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# Transmission and Multiplexing (TM); Considerations on transmission delay and transmission delay values for components on connections supporting speech communication over evolving digital networks


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## Foreword

This ETSI Technical Report (ETR) has been produced by the Transmission and Multiplexing (TM) Technical Committee of the European Telecommunications Standards Institute (ETSI).

ETRs are informative documents resulting from ETSI studies which are not appropriate for European Telecommunication Standard (ETS) or Interim European Telecommunication Standard (I-ETS) status. An ETR may be used to publish material which is either of an informative nature, relating to the use or the application of ETSs or I-ETSs, or which is immature and not yet suitable for formal adoption as an ETS or an I-ETS.

## Introduction

In the past, the approach to transmission planning for speech communications was based on the concept of a single public network that provided connections between individual terminals or between relatively simple private networks. In the future, following the liberalisation of voice telephony and the introduction of competing telecommunications infrastructures, the situation will become much more complex with many alternative options.

In parallel with these regulatory changes, new fixed and mobile technologies are being introduced and these technologies in many cases introduce additional transmission delay. Examples are:

- Synchronous Digital Hierarchy (SDH) multiplexing systems;
- Asynchronous Transfer Mode (ATM) switching and cell based transmission;
- new mobile network air interfaces with special framing structures, sometimes including interleaving;
- new codecs for speech transmission;
- optical fibre transmission both in trunk and local access networks;
- radio systems for the local loop.

As a result of these changes, there is a need for a thorough review of the approach to transmission planning. The various elements of transmission planning and their inter-relations are identified in figure 1.

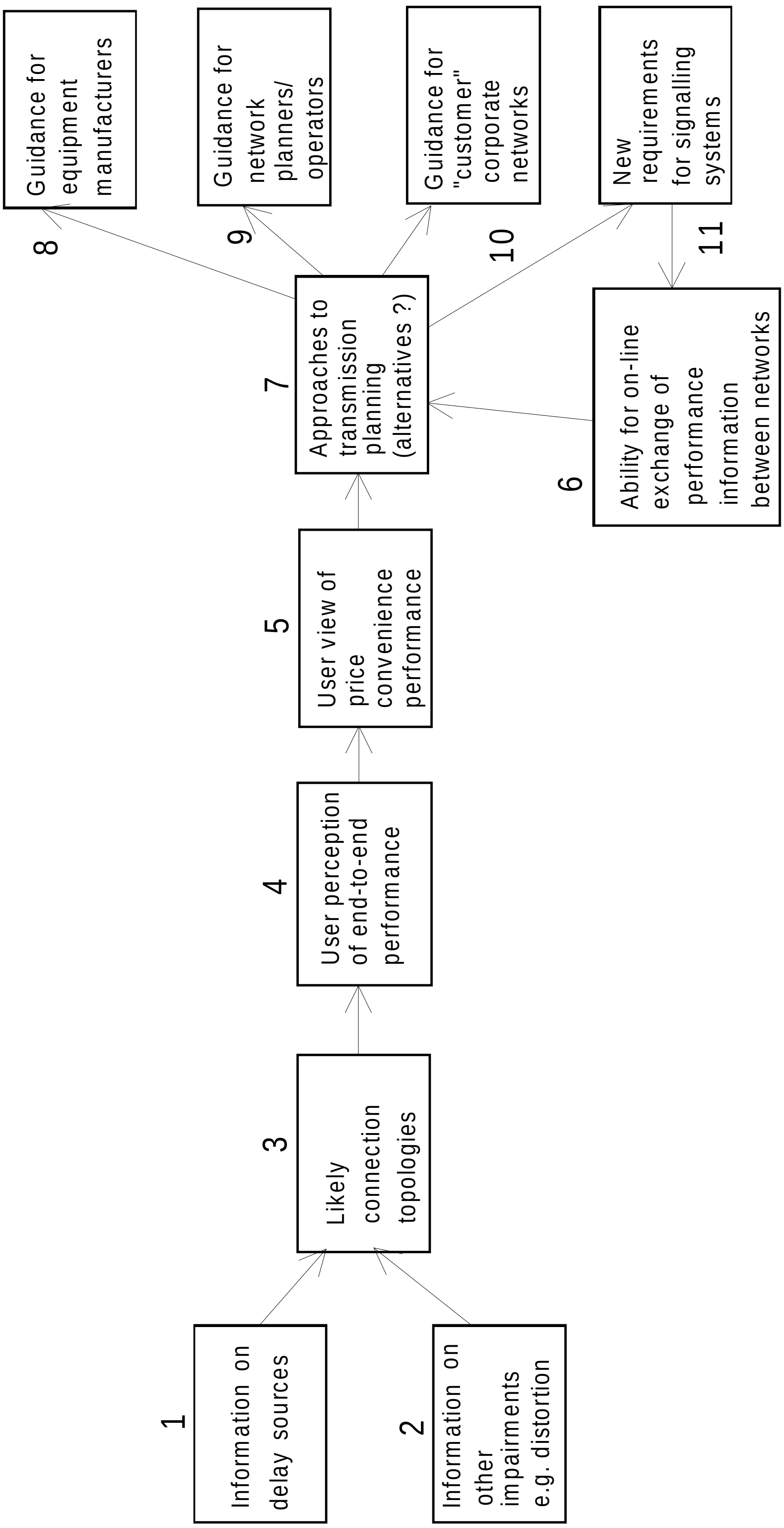


**Figure 1: Transmission planning elements and their inter-relationships**

The following gives the status of the various activities identified in figure 1.

1 The provision of an accurate and comprehensive set of information on the delay contribution of various technologies is the main purpose of this report.

2 No work on distortion is currently in progress, although it is addressed to some extent in ETR 250 [2].

3 No document has yet been produced. New reference models are needed. This work is linked to item 7.

4 This item has been addressed fully in ETR 250 [2].

5 No coherent set of information is currently available, although sources of information are increasing through marketing surveys of user views of some mobile and corporate systems. The model contained in ETR 250 [2] includes an expectation factor that can be used.

6 &11 The ability of the currently defined signalling systems is limited, new requirements for new facilities may be identified.

7 ETSI Technical Sub-committee BTC 2 has a work programme item addressing this issue and the intention is to produce an ETR.

8 Some guidance is provided in various ETSs.

9 Guidance on User Network Interface-User Network Interface (UNI-UNI) performance and interconnection arrangements is urgently needed. Guidance hitherto has been based on the public-private model and the 25 ms rule M of ITU-T Recommendation G.131 [4].

10 Some guidance based on the old public-private network model has been produced in ETR 004 [8] (under review by BTC 2) and ETS 300 283 [1].

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# 1 Scope

The purpose of this ETR is to examine those components which contribute to the transmission delay on connections supporting speech traffic over evolving digital networks. It provides typical values which can be used to develop echo and delay guidance documents for network operators/planners.

# 2 References

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

[1] ETS 300 283: "Business TeleCommunications (BTC); Planning of loudness rating and echo values for private networks digitally connected to the public network".

[2] ETR 250: "Transmission and Multiplexing (TM); Speech communication quality from mouth to ear for 3,1 kHz handset telephony across networks".

[3] ITU-T Recommendation G.114: "One-way transmission time".

[4] ITU-T Recommendation G.131: "Stability and echo".

NOTE: A totally revised version of ITU-T Recommendation G.131: Control of talker echo (without the section on stability) is expected to be approved in 1996.

[5] ITU-T Recommendation G.165: "Echo cancellers".

[6] ETS 300 540: "Digital cellular telecommunications system (Phase 2); Transmission planning aspects of the speech service in the GSM Public Land Mobile Network (PLMN) system (GSM 03.50)".

[7] ITU-T Recommendation G.113: "Transmission impairments".

[8] ETR 004: "Business Telecommunications (BT); Overall transmission plan aspects of a private branch network for voice connections with access to the public network".

[9] ITU-T Recommendation Q.551: "Transmission characteristics of digital exchanges".

[10] ITU-T Recommendation Q.543: "Digital exchange performance design objectives".

[11] ITU-T Recommendation Q.512: "Digital exchange interfaces for subscriber access".

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

[13] TBR 008: "Integrated Services Digital Network (ISDN); Telephony 3,1 kHz teleservice Attachment requirements for handset terminals".

[14] I-ETS 300 245-3: "Integrated Services Digital Network (ISDN); Technical characteristics of telephony terminals Part 3: Pulse Code Modulation (PCM) A-law, loudspeaking and handsfree telephony".

[15] I-ETS 300 245-5: "Integrated Services Digital Network (ISDN); Technical characteristics of telephony terminals; Part 5: Wideband (7 kHz) handset telephony".

[16] I-ETS 300 245-6: "Integrated Services Digital Network (ISDN); Technical characteristics of telephony terminals; Part 6: Wideband (7 kHz), loudspeaking and hands free telephony".

[17] I-ETS 300 131: "Radio Equipment and Systems (RES); Common air interface specification to be used for the interworking between cordless telephone apparatus in the frequency band 864,1 MHz to 868,1 MHz, including public access services".

[18] TBR 010: "Radio Equipment and Systems (RES); Digital European Cordless Telecommunications (DECT) General terminal attachment requirements: telephony applications".

[19] ETS 300 175: "Radio Equipment and Systems (RES); Digital European Cordless Telecommunications (DECT) Common interface".

[20] TBR 009: "European digital cellular telecommunications system; Attachment requirements for Global System for Mobile communications (GSM) mobile stations; Telephony".

[21] ITU-T Recommendation G.173: "Transmission planning aspects of the speech service in digital public land mobile networks".

[22] ITU-T Recommendation G.174: "Transmission performance objectives for terrestrial digital wireless systems using portable terminals to access the PSTN".

[23] ETS 300 326: "Radio Equipment and Systems (RES); Terrestrial Flight Telephone System (TFTS)".

[24] ITU-T Recommendation G.763: "Digital circuit multiplication equipment using ADPCM (Recommendation G.726) and digital speech interpolation".

[25] ITU-T Recommendation G.764: "Voice packetization - Packetized voice protocols".

[26] ETS 300 010-1: "Transmission and Multiplexing (TM); Synchronous cross connect equipment 64 and n x 64 kbit/s cross connection rate 2 048 kbit/s access ports Part 1: Core functions and characteristics".

[27] ITU-T Recommendation I.731: "dont exist".

[28] prETS 300 463: "Transmission and Multiplexing (TM); Requirements of passive Optical Access Networks (OANs) to provide services up to 2 Mbit/s bearer capacity".

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

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

[31] ITU-T Recommendation G.726: "40, 32, 24, 16 kbit/s adaptive differential pulse code modulation (ADPCM)".

[32] ITU-T Recommendation G.727: "5-, 4-, 3- and 2-bits/sample embedded adaptive differential pulse code modulation (ADPCM)".

[33] ITU-T Recommendation G.722: "7 kHz audio-coding within 64 kbit/s".

[34] ITU-T Recommendation G.728: "Coding of speech at 16 kbit/s using low-delay code excited linear prediction".

[35] ETS 300 580: "European digital cellular telecommunications system (Phase 2); Full rate speech processing functions (GSM 06.01)".

[36] ITU-T Recommendation G.729: "Coding of speech at 8 kbit/s using conjugate structure algebraic-code-excited linear-prediction".

[37] ITU-T Recommendation G.723.1: "Dual rate speech coder for multimedia telecommunication at 5,3 and 6,3 kbit/s".

[38] I-ETS 300 281: "Integrated Services Digital Network (ISDN); Telephony 7 kHz teleservice Terminal requirements necessary for end-to-end compatibility".

[39] I-ETS 300 302-1: "Integrated Services Digital Network (ISDN); Videotelephony teleservice Part 1: Electroacoustic characteristics for handset telephony function when using Pulse Code Modulation (PCM) encoding".

[40] TIA IS-54: "Cellular system dual-mode mobile station - Base station compatibility standard".

[41] TIA IS-96: "Speech Service Option Standard for Wideband Spread Spectrum Digital Cellular System".

[42] TIA IS-136: "800 MHz TDMA Cellular-Radio Interface - Mobile Station - Base Station Compatibility".

[43] ETR 152 (1995): "Transmission and Multiplexing (TM); High bitrate Digital Subscriber Line (HDSL) transmission system on metallic local lines; HDSL core specification and applications for 2 048 kbit/s based access digital sections including HDSL dual-duplex Carrierless Amplitude Phase Modulation (CAP) based system".

[44] T1.413: "American National Standard for Telecommunications; Network and Customer Installation Interfaces; Asymetrical Digital Subscriber Line (ADSL) metalic interface".

# 3 Definition and abbreviations

## 3.1 Definition

For the purposes of this ETR, the following definition applies:

**transmission delay:** Comprises the propagation time determined by distance together with the processing time of different technologies and equipment.

## 3.2 Abbreviations

For the purposes of this ETR, the following abbreviations apply:

| | |
|---|---|
| ACELP | Algebraic Code Excited Linear Prediction |
| ADPCM | Adaptive Differential Pulse Code Modulation |
| ADSL | Asymetrical Digital Subscriber Line |
| ATM | Asynchronous Transfer Mode |
| CAI | Common Air Interface |
| CDMA | Code Division Multiple Access (spread spectrum) |
| CELP | Code Excitation Linear Prediction |
| CFP | Cordless Fixed Part |
| CPP | Cordless Portable Part |
| CT2 | second generation Cordless Telephone |
| DAMPS | Digital Advanced Mobile Phone System |
| DCME | Digital Circuit Multiplication Equipment |
| DCS | Digital Cellular System |
| DCT | Discrete Cosine Transformation |
| DECT | Digital European Cordless Telecommunications |
| ERP | Ear Reference Point |
| FDM | Frequency Division Multiplexing |
| GSM | Global System for Mobile communication |
| HDSL | High bitrate Digital Subscriber Line |
| IN | Intelligent Network |
| ISDN | Integrated Services Digital Network |
| LD-CELP | Low Delay Code Excited Linear Predictor |
| LE | Local Exchange |
| MPLPC | MultiPulse Linear Predictive Coding |
| MRP | Mouth Reference Point |
| NTP | Network Termination Point |
| PABX | Private Automatic Branch Exchange |
| PCM | Pulse Code Modulation |
| PCME | Packet Circuit Multiplication Equipment |
| PCN | Personal Communications Network |
| PDH | Plesiochronous Digital Hierarchy |
| PLMS | Public Land Mobile System |
| PON | Passive Optical Network |
| POTS | Plain Old Telephony Service |
| PSTN | Public Switched Telephone Network |
| QCELP | Q-Code Excitation Linear Prediction |
| QDU | Quantization Distortion Unit (see ITU-T Recommendation G.113 [7]) |
| RPE-LTP | Regular Pulse Excitation with Long Term Predictor |
| SDH | Synchronous Digital Hierarchy |
| TCLw | Terminal Coupling Loss (weighted) |
| TE | Terminal Equipment |
| TFTS | Terrestrial-Flight Telecommunications System |
| VSELP | Vector Sum Excited Linear Prediction |
| WPCS | Wireless Personal Communication System |

# 4 General considerations

Any consideration of delay must take into account all the key drivers which range from customer requirements, through technology trends to possible regulatory influences. This clause outlines these drivers.

Telephone connections with long transmission delay and with negligible echo cause significant user difficulty due to the interruption of normal conversational flow, and for this reason excessive delays are best avoided. The presence of short delays on telephone connections does not cause conversational difficulty due to interruption of conversational flow, but can cause significant difficulties in the presence of echo. Delays as short as 1 ms can cause major impairments if the echo is not at a sufficiently low level, due to cancellation between local sidetone and the delayed sidetone (echo) causing significant ripples or nulls in the speech response. It is therefore necessary to understand the echo levels as well as the delay times in order to predict and plan the overall connection performance.

For connections terminated in 2-wire analogue terminals acceptable echo performance may be obtained by good impedance matching of the Terminal Equipment (TE) to the network. This can often be more economic, and provide better performance than the use of additional echo control devices. Evolution of local access systems such as street corner multiplexers, radio drop-wires etc., is leading towards shorter copper local loops, and together with the acceptance of a harmonised impedance makes good impedance matching of 2-wire terminals a more viable proposition. This leads to the possibility of a trade-off of impedance matching against delay in the local access system, which could give significant technical and commercial advantages to new types of local access systems. (The harmonised impedance consists of a 270 Ω resistor in series with a parallel combination of a 750 Ω resistor and a capacitor of 150 nF).

Most of the echo is generated within the local access part of an overall connection, and annex B attempts to summarise the delay, echo and quantisation performance of various TE types and local access delivery systems.

NOTE 1: Transmission delay has a major impact on customer perception of speech quality; where relatively small increases in delay in a connection can cause echo to become objectionable to customers, even on national connections, when the loss of the path is not sufficiently high enough to counter the increased echo.

NOTE 2: Connections with long delay times will require some form of echo control, but the delay may still affect customer opinion due to the disruption in conversational fluency.

## 4.1 End-user requirements

Most users of European services have little experience of delay and echo effects, hence they will be seen as new impairments that worsen the overall service quality. User expectations are increasing in many areas and there will be little support for new networks with significant delay or echo. Many subjective studies have been carried out (as referenced in draft ETR 250 [2]) to derive appropriate end-to-end limits for delay and echo.

The expectation is that users will require network operators to maintain delay and echo performance at its current (generally imperceptible) level for the large majority of calls.

## 4.2 Network operator requirements

The network operator wishes to provide a cost-effective network using appropriate technology that meets all the end-user requirements. Operators may also have other network operators as customers and there is likely to be a requirement for some statement of the delay/echo contribution of the respective networks.

### 4.3 Network technology and services

The emerging transport technologies required to support end-to-end multiservice networks appear to be introducing additional transmission delays in comparison with traditional networks. Access network systems such as Passive Optical Networks (PONs) and radio systems are likely to require additional delay allowances, whilst ATM will introduce a delay into the connection which is dependent upon the bit rate used by the terminals.

The increasing use of Intelligent Network (IN) control to provide both new and existing services may result in extra transmission delay if calls are routed through additional transmission links and switches.

Availability of cost-effective network technology will be a major driver for network operators which needs to be reflected in evolving delay limits.

### 4.4 Customer equipment and networks

Delay and echo effects are perceived by customers on an end to end basis which includes the contributions from customer-owned equipment and networks. As public networks evolve to providing a digital presented service to customers then the delay and echo contribution of the terminal equipment becomes more significant. These issues have been addressed in ETS 300 283 [1] (Planning of loudness rating and echo values for private networks digitally connected to the public network) and ETR 250 [2].

### 4.5 Multi-operator networks

The changing European regulatory environment is gradually resulting in an increase in the number of operators who are able to provide a service supporting speech communication in any one country. Interconnection of those networks will become increasingly common and end to end connections will frequently involve more than one operator. Details of network characteristics which affect performance (including transmission delay) need to be included in the agreements and contracts between the operators.

As a minimum, such agreements should include a common understanding of the end-end transmission delay objectives that are being followed. They should also include an understanding of who has overall responsibility for ensuring that the objectives are met. For example the operator providing a service to the customer may take overall responsibility for quality and negotiate appropriately with the other operators to meet the end-end objectives.

## 5 Transmission delay values

### 5.1 Overall (end-user to end-user)

Delay values need to be set in conjunction with the echo control mechanism that is employed (see ITU-T Recommendation G.131 [4]). Echoes may be controlled by providing a sufficiently high echo loss (the inherent echo loss in analogue hybrids or the acoustic to acoustic loss designed into digital handsets) or purpose designed echo control devices (usually echo cancellers).

An objective is normally to make maximum use of the inherent echo loss of network and customer equipment to minimise the cost of employing additional echo control devices. However, this approach requires consistent performance specification for customer equipment.

The following delay ranges are derived from ITU-T Recommendation G.114 [3] and include both public and private network contributions.

#### 5.1.1 Range 0 - 25 ms

25 ms is generally considered as the maximum one way delay for which echo can be made tolerable by providing circuit loss in addition to the trans-hybrid loss. This range is expected to cover the majority of national calls within an average sized country.

### 5.1.2 Range 25 - 150 ms

Connections with delays greater than 25 ms will generally require echo control devices. Appropriately installed devices meeting ITU-T Recommendation G.165 [5] will result in little impairment for delays up to 150 ms.

### 5.1.3 Range 150 - 400 ms

Within this range there is an increasing risk of difficulties from disruption of conversational flow as the delay increases even assuming appropriate echo cancellers are fitted.

### 5.1.4 Above 400 ms

Connections with values above 400 ms should be avoided for general network planning.

## 6 Delay allocation

Public and private networks have traditionally been treated separately, as is exemplified by ETS 300 283 [1]. The treatment of delay in that standard assumes that, in order to meet Rule M of CCITT Recommendation G.131 [4], public networks have delays in the range of 15 - 20 ms, and that private networks have delays limited to 5 ms (unless special echo control provisions are made).

As networks are evolving into a multi operator environment, such a simplistic approach becomes inappropriate, and alternative ways for dealing with delay are necessary.

ETR 250 [2] provides guidance on the treatment of network delays (and other network impairments) arising from various sources, and provides a tool to determine their effect on perceived speech quality.

## 7 Echo control

There will be an increasing number of cases where the system delays are sufficiently long to require echo control for all calls, an example being Global System for Mobile communications (GSM) mobile systems. A general principle should be that the system contributing the additional delay should be responsible for controlling the resultant echo. This principle has been adopted for GSM where the GSM network is responsible for echo control on calls to fixed networks with a defined delay budget (ETS 300 540 [6]).

Delay limits on connections with additional echo control should ideally meet the requirements of subclause 5.1.2 above, i.e. a maximum delay of 150 ms. However it is recognised that some systems will require the extended limits in subclause 5.1.3. Apportionment of these limits between network operators will need to be agreed on a case by case basis using the reference connections in clause 8 as a guide.

Planning and installation of multiple echo control devices is not straightforward as the devices need to operate in a controlled network environment. This includes delay limits for access to international switches and the characteristics of connected private networks.

These issues are discussed in more detail in ETR 250 [2].

# 8 Connection arrangements

In the multi network operator environment, end to end connections may be provided by a variety of means. An example of a connection path showing the various contributing elements is shown below.

Examples of the delays contributed by terminal elements, connection elements and transmission media are given in annex A.

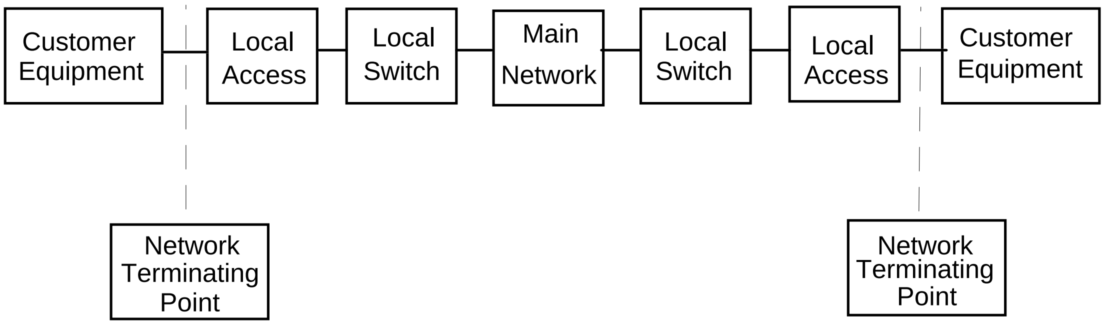


**Figure 2: Overall connection segments (national)**

The local access segment may consist of a variety of transmission methods, e.g. analogue over copper pairs, digital over copper pairs or optical fibre, wireless.

The customer equipment may consist of e.g. analogue telephone set, digital telephone set, cordless terminal apparatus, Private Automatic Branch eXchange (PABX) or a private network.

Normally a mobile system (e.g. GSM) may consist of the local access segment together with the customer equipment segment.

# Annex A: Delay contributions of network elements

The values given in this annex are intended to give guidance to transmission planners. It should be noted that some values are taken from ITU-T Recommendations or standards, others are calculated worst case values and some are typical values derived from practical experience. The remarks column indicates the source when known.

## A.1 Terminal elements

**Table A.1: Terminal delay**

| Terminal elements | Mean one-way delay | Remarks |
|---|---|---|
| ISDN: 3,1 kHz handset | 1 ms | I-ETS 300 245-2 [12], TBR 008 [13] |
| ISDN: 3,1 kHz handsfree | 4 ms | I-ETS 300 245-3 [14] |
| ISDN: 7 kHz handset | 3,5 ms | I-ETS 300 245-5 [15] |
| ISDN: 7 kHz handsfree | 5 ms | I-ETS 300 245-6 [16] |
| CT2 (ADPCM 32 kbit/s) | 2,5 ms | I-ETS 300 131 [17],<br>Common Air Interface (CAI) |
| DECT (ADPCM 32 kbit/s) | 14 ms | TBR 010 [18], ETS 300 175 [19] |
| DAMPS (VSELP 8 kbit/s) | 95 ms | TIA IS-54 [40], USA |
| GSM (RPE-LTP 13 kbit/s) | 95 ms | TBR 009 [20] (75 ms), ETS 300 540 [6] |
| DCS 1800 (RPE-LTP 13 kbit/s) | 95 ms | |
| DCT 900 (ADPCM 32 kbit/s) | 20 ms | Scandinavian digital cordless |
| PLMS | 110 ms | ITU-T Recommendation G.173 [21]<br>(Delay objective 40 ms) |
| WPCS | 40 ms | ITU-T Recommendation G.174 [22] |
| TFTS (MPLPC 9,6 kbit/s) | 95 ms | ETS 300 326 [23] |

## A.2 Connection elements

The values for digital exchanges are given in ITU-T Recommendation Q.551 [9] (where A = analogue and D = digital) and are applicable under reference load A as defined in ITU-T Recommendation Q.543 [10].

The interfaces A, B, Z etc. are given in ITU-T Recommendations Q.511 [9] and Q.512 [11] respectively.

**Table A.2: Connection delay**

| Connection systems | Mean one-way delay | Remarks |
|---|---|---|
| FDM channel modulator or demodulator | 0,75 ms | |
| FDM compandored channel modulator or demodulator | 0,5 ms | |
| Transmuliplexer | 1,5 ms | In earth stations 3,3 ms |
| Digital transit exchange D/D | 0,45 ms | From / to interface A or B |
| Digital local exchange A/A | 1,5 ms | From / to interface Z |
| Digital local exchange A/D | 1,05 ms | From interface Z to interface A or B |
| Digital local exchange D/A | 0,9 ms | From interface A or B to interface Z |
| Digital local exchange A/D | 1,425 ms | From interface Z to interface V1 |
| Digital local exchange D/A | 1,275 ms | From interface V1 to interface Z |
| Digital local exchange D/D | 1,2 ms | From / to interface V1 |
| Access digital section | 1,25 ms | From interface V1 to S/T bus |

(continued)

**Table A.2 (concluded): Connection delay**

| Connection systems | Mean one-way delay | Remarks |
|---|---|---|
| Access digital section, repeater | 0,75 ms | 1 allowed |
| Remote concentrator of LE | 0,325 ms | See note |
| Digital PABX | Equal to digital local exchange | |
| Echo cancellers | 0,5 ms | ITU-T Recommendation G.165 [5] |
| Conference devices | 2,0 ms | Must be time invariant |
| DCME (ADPCM n kbit/s) | 30 ms | ITU-T Recommendation G.763 [24] with speech + non-remodulated voice band data |
| DCME (ADPCM n kbit/s) | 350 ms | ITU-T Recommendation G.763 [24] with remodulated voice band data |
| DCME (16 or 8 kbit/s) | 60 ms | This is an example of a proprietary implementation and is not in accordance with ITU-T Recommendation G.763 [24] |
| DCME (4,8 kbit/s) | 120 ms | This is an example of a proprietary implementation and is not in accordance with ITU-T Recommendation G.763 [24] |
| PCME | 35 ms | ITU-T Recommendation G.764 [25] + with speech + non-remodulated voice band data |
| PCME | 70 ms | ITU-T Recommendation G.764 [25] with remodulated voice band data |
| PDH higher order Multiplexer (pair) | 20 μs (worst case) | See note |
| PDH 1/0 digital cross-connect | 650 μs | ETS 300 010-1 [26] |
| SDH digital cross-connect, compound 4/3/1 (synchronous/synchronous) | 60 μs (worst case) | See note |
| SDH digital cross-connect, compound 4/3/1 (plesiochronous and synchronous) | 110 μs (worst case) | See note |
| SDH ADM STM-1/STM-4/STM-16 (aggregate/aggregate) | 60 μs (worst case) | See note |
| SDH ADM STM-1/STM-4/STM-16 (aggregate/tributary) | 110 μs (worst case) | See note |
| ATM with 64 kbit/s input | 6 ms | Packetisation / depacketisation delay |
| ATM switch | 300 μs (maximum)<br>150 μs (99 percentile)<br>100 μs (mean)<br><br>250 μs | Cell Transfer Delay, ITU-T Recommendation I.731 [27]<br><br><br><br>Cell Delay Variation, ITU-T Recommendation I.731 [27] |
| PON, D/A and A/D | 1,5 ms | Interface V5.x to/from interface Z, prETS 300 463 [28] |
| PON, D/D if NT1 is an integral part of ONU | 1,5 ms | Interface V5.x to/from S/T bus, prETS 300 463 [28] |
| PON, D/D if NT1 is remote from the ONU | 2,0 ms | Interface V5.x to/from S/T bus, prETS 300 463 [28] |
| NOTE: This value has not been standardised yet but is provided for guidance. It is therefore subject to change pending the outcome of ongoing standardisation work. | | |

## A.3 Transmission media

The values in the table below are stated in ITU-T Recommendation G.114 [3].

**Table A.3: Transmission media delay**

| Transmission media | Mean one-way delay | Remarks |
|---|---|---|
| Terrestrial coaxial cable or radio-relay system; FDM and digital transmission | 4 µs/km | Allows for delay in repeaters and regenerators |
| Optical fibre cable system | 5 µs/km | Allows for delay in repeaters and regenerators |
| Submarine coaxial cable system | 6 µs/km | Allows for delay in repeaters and regenerators |
| Satellite system, 1 400 km | 12 ms | Distance delay between earth stations only |
| Satellite system, 14 000 km | 110 ms | Distance delay between earth stations only |
| Satellite system, 36 000 km | 260 ms | Distance delay between earth stations only |

## A.4 Connection coding devices

The algorithmic delay of the coding devices given in the table below is already included in the delay of the terminal elements (table A.1) and connection elements (table A.2). It should therefore **not** be added to the delay value given there.

The coding algorithm used by terminal elements and connection elements is given in tables A.1 and A.2 respectively to allow the reader to find the equipment impairment value "k" within the ETR 250 [2].

**Table A.4: Coding delay**

| Connection coding devices | Algorithmic delay | Remarks |
|---|---|---|
| PCM | 0,75 ms | ITU-T Recommendation G.711 [29] and ITU-T Recommendation G.712 [30] |
| PCM | | ITU-T Recommendation G.711 [29], truncated to 7 bits (56 kbit/s) |
| ADPCM (40 kbit/s) | 0,125 ms | ITU-T Recommendation G.726 [31] and ITU-T Recommendation G.727 [32] |
| ADPCM (32 kbit/s) | 0,125 ms | ITU-T Recommendation G.726 [31] and ITU-T Recommendation G.727 [32] (G.721:1988) |
| ADPCM (24 kbit/s) | 0,125 ms | ITU-T Recommendation G.726 [31] and ITU-T Recommendation G.727 [32] |
| ADPCM (16 kbit/s) | 0,125 ms | ITU-T Recommendation G.726 [31] and ITU-T Recommendation G.727 [32] |
| ADPCM (64 kbit/s) (7 KHz wideband) | 4 ms | ITU-T Recommendation G.722 [33] |
| PCM/ADPCM/PCM transcoder | 0,5 ms | |
| LD-CELP (16 kbit/s) | 0,625 ms | ITU-T Recommendation G.728 [34] |
| LD-CELP (12,8 kbit/s) | 0,625 ms | ITU-T Recommendation G.728 [34] |
| RPE-LTP (13 kbit/s) | 20 ms | GSM Full-rate (ETS 300 580 [35]) |
| ACELP (13 kbit/s) | 20 ms | NPAG for PCS 1900 |
| MPLPC (9,6 kbit/s) | ~ 20 ms | TFTS |
| VSELP (8 kbit/s) | 28 ms | TIA IS-54 [40], USA |
| VSELP (6,7 kbit/s) | 27 ms | JDC Full-rate |
| QCELP (8 kbit/s) | 20 ms | TIA IS-96 [41], USA |
| CS-ACELP (8 kbit/s) | 15 ms | ITU-T Recommendation G.729 [36], TIA IS-136 [42], USA |
| CELP (3,45 kbit/s) | 45 ms | JDC Half-rate |
| CELP+ (6,8 kbit/s) | ~ 20 ms | |
| AV.25Y (5,27 - 6,3 kbit/s) | 37,5 ms | ITU-T Recommendation G.723 [37], dual rate speech coder for multimedia |

# Annex B: Delay contributions for terminal equipment and local access delivery systems

## B.1 Wired terminals

### B.1.1 3,1 kHz handset - analogue (2-wire)

Nominal 1-way delay: 0 ms
Coding: none
QDUs: none
TCLw: n/a - The telephone set impedance is usually defined
Source Specification: none

- The delay introduced by an ordinary 3,1 kHz analogue telephone can be assumed to be zero;

- the telephone set impedance is usually defined by approval requirements. A harmonised European impedance has been agreed (The harmonised impedance consists of a 270 Ω resistor in series with a parallel combination of a 750 Ω resistor and a capacitor of 150 nF);

- the SideTone Masking Rating (STMR) is usually defined. Sometimes the zero sidetone impedance $Z_{SO}$ for the telephone is also defined.

### B.1.2 3,1 kHz handsfree - analogue (2-wire)

Nominal 1-way delay: 1,5 ms (at 500 mm test position)
Coding: none
QDUs: none
TCLw: n/a - The telephone set impedance is usually defined
Source Specification: none

- The delay introduced by a 3,1 kHz analogue handsfree telephone arises almost entirely from the acoustic transmission delay of the air path (velocity of sound = 33 cm/ms);

- the telephone set impedance is usually defined by approval requirements. A harmonised European impedance has been agreed (The harmonised impedance consists of a 270 Ω resistor in series with a parallel combination of a 750 Ω resistor and a capacitor of 150 nF).

### B.1.3 3,1 kHz handset - digital (4-wire)

Nominal 1-way delay: 1 ms
Coding: ITU-T Recommendation G.711 [29]
QDUs: 0,5
TCLw: > 40 dB at normal sensitivities, > 35 dB at all volume settings
Source Specification: ETS 300 245-2 [12], TBR 008 [13]

- Specified limit: The sum of the delays (round trip delay) from the Mouth Reference Point (MRP) to the line interface, and from the line interface to the Ear Reference Point (ERP) should be less than 2 ms;

- the source of the delay is mostly the signal processing arising from the speech encoding algorithm.

### B.1.4 3,1 kHz handsfree - digital (4-wire)

Nominal 1-way delay: 4 ms
Coding: ITU-T Recommendation G.711 [29]
QDUs: 1
TCLw:
Source Specification: ETS 300 245-3 [14]

- Specified limit: The sum of the delays (round trip delay) from the MRP to the network connection point , and from the network connection point to the ERP should be less than 8 ms;

- the sum of the send and receive delays allows 5 ms for the digital signal processing and 3 ms for the air path

### B.1.5 7 kHz handset - digital (4-wire)

Nominal 1-way delay: 3,5 ms
Coding: ADPCM according to ITU-T Recommendation G.722 [33] - 7 KHz, 64 kb/s
QDUs:
TCLw:
Source Specification: ETS 300 281 [38] (ETS 300 245-5 [15])

- Specified limit: The sum of the delays (round trip delay) from the MRP to the network connection point , and from the network connection point to the ERP should be less than 7 ms;

- the delay is mainly due to the signal processing time arising from the speech encoding (which should not exceed 4 ms).

### B.1.6 7 kHz handsfree - digital (4-wire)

Nominal 1-way delay: 5 ms
Coding: ADPCM according to ITU-T Recommendation G.722 [33] - 7 KHz, 64 kb/s
QDUs:
TCLw: > 35 dB
Source Specification: ETS 300 245-6 [16]

- Specified limit: The sum of the delays (round trip delay) from the MRP to the network connection point, and from the network connection point to the ERP should be less than 10 ms;

- the sum of the send and receive delays allows 7 ms for the digital processing and 3 ms for the air path (assumed to be 500 mm).

### B.1.7 Low Delay Code Excited Linear Predictor (LD-CELP) handset - digital (4-wire)

Nominal 1-way delay: 5 ms
Coding: ITU-T Recommendation G.728 [34]
QDUs: 3,5
TCLw: > 40 dB at nominal sensitivities, > 35 dB at all volume control settings
Source Specification: ETS 300 245-6 [16]

- Specified limit: The sum of the delays (round trip delay) from the MRP to the digital interface and from the digital interface to the ERP should be less than 10 ms;

- the source of the delay is mainly the signal processing arising from the speech encoding.

## B.2 Cordless terminals

Cordless terminals typically consist of a handset or Cordless Portable Part (CPP) together with a base station or Cordless Fixed Part (CFP). Since a CPP or CFP cannot be used on its own, the following details summarise the performance of a CPP interworking with a CFP. The CPP or CFP may be a part of some larger apparatus e.g. the CFP may be part of a cordless PABX, in which case additional impairments to those identified below may be present due to the additional functionality.

### B.2.1 Second generation Cordless Telephone (CT2) handset

| | |
|---|---|
| Nominal 1-way delay: | < 2,5 ms acoustic reference point to network termination point |
| Coding: | ITU-T Recommendation G.726 [31] - 32 kb/s ADPCM |
| QDUs: | 3,5 |
| TCLw: | > 34 dB (free field measurement) |
| Source Specification: | I-ETS 300 131 [17] |

- The sum of the delay (round trip delay) from the MRP to the line interface and the delay from the line interface to the ERP should not exceed 5,0 ms when averaged over the frequency range 500 Hz to 2 500 Hz;

- local sidetone within the handset (CPP) may be switched on or off by control signals from the CFP carried via the D channel depending on whether the CFP is connected to a 2-wire or 4-wire Network Terminating Point (NTP);

- for cordless terminals where the CFP is connected to a 4-wire connection, it may be necessary to introduce an artificial echo path within the CFP to ensure any network provided echo canceller is active. More details may be found in annex K to I-ETS 300 131 [17];

- the Quantization Distortion Unit (QDU) value is derived from: 0,5 for the A-law convertor plus 2,5 for the PCM>ADPCM>PCM tandem conversion plus 0,5 for the A-law convertor;

- for cordless terminals where the CFP is connected to a 4-wire digital connection, the quantisation distortion and 1-way delay may be reduced by up to 0,5 QDU and 0,5 ms due to the absence of A-law and A to D convertors within the CFP.

### B.2.2 Digital European Cordless Telecommunications (DECT) handset

| | |
|---|---|
| Nominal 1-way delay: | <14 ms acoustic reference point to network termination point |
| Coding: | ITU-T Recommendation G.726 [31] - 32 kb/s ADPCM |
| QDUs: | 3,5 |
| TCLw: | type a) > 46 dB, type b) > 34 dB (free field measurement) |
| Source Specification: | ETS 300 175-8 [19], TBR 010 [18] |

- The sum of the delay from the MRP to the line interface and the delay from the line interface to the ERP (round trip delay) should not exceed 28 ms for an analogue line interface and 27,5 ms for a digital line interface;

- the DECT CPP indicates to the CFP at call setup whether it is type a) or type b) apparatus;

- for cordless terminals where the CFP is connected to a 4-wire connection, it may be necessary to introduce an artificial echo path within the CFP to ensure any network provided echo canceller is active. More details may be found in ETS 300 175-8 [19] subclauses 7.4.1.2 and 7.10;

- the QDU value is derived from: 0,5 for the A-law convertor plus 2,5 for the PCM>ADPCM>PCM tandem conversion plus 0,5 for the A-law convertor;

- for cordless terminals where the CFP is connected to a 4-wire digital connection, the quantisation distortion may be reduced by up to 0,5 QDU due to the absence of A-law convertors within the CFP;

- for details of DECT handsets in tandem with GSM, see ETS 300 175-8 [19], subclause 8.2.

### B.2.3 Global System for Mobile communication (GSM) handset

This also applies to Personal Communications Network (PCN) handsets.

| | |
|---|---|
| Nominal 1-way delay: | 95 ms acoustic reference point to PSTN point of interconnect |
| Coding: | RPE-LTP 13 kb/s |
| QDUs: | 7 (between uniform PCM interfaces) |
| TCLw: | > 34 dB (sealed ear measurement) |
| Source Specification: | TBR 009 [20], ETS 300 540 [6] |

- The main components of the delay are; interleaving delay on the air interface, collection time of the frame samples, and frame transmission time between the base station controller and the base transceiver station;

- for mobile to mobile calls within the same mobile network, the delay is approximately twice the delay value given above i.e. 190 ms;

- a terminal with a particular TCLw value measured under sealed ear conditions will in general provide inferior echo performance to a terminal with the same TCLw value measured under free field conditions.

### B.2.4 Code Division Multiple Access (CDMA) handset

| | |
|---|---|
| Nominal 1-way delay: | 10 ms |
| Coding: | ITU-T Recommendation G.711 [29] - PCM<br>ITU-T Recommendation G.726 [31]- 40, 32, 24, 16 kb/s ADPCM |
| QDUs: | dependent on coding used |
| TCLw: | > 34 dB (sealed ear measurement) |
| Source Specification: | Not yet available |

- The round trip delay (including transmission delay) should not exceed 20 ms for voice encoded 64 kb/s time slot.

### B.2.5 Wireless Personal Communication System (WPCS) handset

| | |
|---|---|
| Nominal 1-way delay: | < 40 ms |
| Coding: | ITU-T Recommendation G.174 [22] |
| QDUs: | |
| TCLw: | |
| Source Specification: | For further study |

## B.3 Videophones

### B.3.1 Videophone with analogue access

Nominal 1-way delay:
Coding:
QDUs:
TCLw:
Source Specification: Not yet available

- Work in progress within ETSI STC TE 5 and ITU-T Study Group 15 Special Rapporteurs group on Low Bitrate Coding.

### B.3.2 Videophone with digital access 2x64 kb/s

| | |
|---|---|
| Nominal 1-way delay: | 187,5 ms |
| Coding: | ITU-T Recommendation G.711 [29] truncated to 7 bits (56 kb/s) |
| QDUs: | |
| TCLw: | > 40 dB at nominal sensitivities (free field test)<br>> 35 dB at all volume control settings |
| Source Specification: | I-ETS 300 302-1 [39] |

- The maximum sum of delays (round trip delay) from the MRP to the digital interface and from the digital interface to the ERP should be less than 375 ms;

- additional delay is place in the speech path in order to achieve "lip Synchronism" with the video image.

### B.3.3 Videophone with digital access 2 048 kb/s

Nominal 1-way delay:
Coding:
QDUs:
TCLw:
Source Specification: For further study

## B.4 Other systems

### B.4.1 Radio drop wires

| | |
|---|---|
| Nominal 1-way delay: | Not specified |
| Coding: | Not specified |
| QDUs: | Not specified |
| TCLw: | Not applicable |
| Source Specification: | No generic specification identified |

- Most of the radio drop wire schemes identified so far are based to a large extent on CT2 or DECT. Some proprietary systems exist but those identified fall within the limits for DECT;

- some information on the use of DECT in local loop applications may be found in annex B of ETS 300 175-8 [19].

### B.4.2 DECT repeater

| | |
|---|---|
| Nominal 1-way delay: | 5 ms per cascaded CRFP, and maximum 2,5 ms per chain of cascaded REP links |
| Coding: | none |
| QDUs: | none |
| TCLw: | not applicable |
| Source Specification: | ETS 300 175-10 [19] |

- The DECT repeater re-transmits data received in a physical channel from one DECT termination to a physical channel for a different DECT termination, without additional processing of the bearer channel information;

- see ETS 300 175-8 [19], subclause 7.10 for additional information on echo control.

### B.4.3 High bitrate Digital Subscriber Line (HDSL)

| | |
|---|---|
| Nominal 1-way delay: | 1,25 ms |
| Coding: | none |
| QDUs: | none |
| TCLw: | not applicable |
| Source Specification: | ETR 152 [43] |

- Provides ISDN primary rate access on 2 or 3 subscriber copper pairs.

### B.4.4 Asymetrical Digital Subscriber Line (ADSL)

| | |
|---|---|
| Nominal 1-way delay: | 2 ms for ISDN basic rate access; 0,2 ms maximum for Plain Old Telephony Service (POTS) |
| Coding: | none |
| QDUs: | none |
| TCLw: | not applicable |
| Source Specification: | European annex to T1.413 [44] on ADSL |

- The delay to the analogue POTS service is due to group delay in the POTS splitting filters.

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