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SERIES G: TRANSMISSION SYSTEMS AND MEDIA, DIGITAL SYSTEMS AND NETWORKS

International telephone connections and circuits – General definitions

The transmission plan

ITU-T Recommendation G.101

ITU-T G-SERIES RECOMMENDATIONS

TRANSMISSION SYSTEMS AND MEDIA, DIGITAL SYSTEMS AND NETWORKS

For further details, please refer to the list of ITU-T Recommendations.

ITU-T Recommendation G.101

The transmission plan

Summary

This Recommendation provides guidance for transmission planning of modern telecommunication networks. The primary emphasis is on provisioning of telephone connections. The transmission plan established by this Recommendation takes into account transmission parameters and impairments, different network configurations and elements and modern transmission techniques, as well as the effects resulting from the combination of the various factors having influence on the resulting transmission quality.

Source

ITU-T Recommendation G.101 was approved by ITU-T Study Group 12 (2001-2004) under the ITU-T Recommendation A.8 procedure on 13 November 2003.

FOREWORD

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

The World Telecommunication Standardization Assembly (WTSA), which meets every four years, establishes the topics for study by the ITU-T study groups which, in turn, produce Recommendations on these topics.

The approval of ITU-T Recommendations is covered by the procedure laid down in WTSA Resolution 1.

In some areas of information technology which fall within ITU-T's purview, the necessary standards are prepared on a collaborative basis with ISO and IEC.

NOTE

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

Compliance with this Recommendation is voluntary. However, the Recommendation may contain certain mandatory provisions (to ensure e.g., interoperability or applicability) and compliance with the Recommendation is achieved when all of these mandatory provisions are met. The words "shall" or some other obligatory language such as "must" and the negative equivalents are used to express requirements. The use of such words does not suggest that compliance with the Recommendation is required of any party.

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CONTENTS

Introduction

The goal of this transmission plan is to facilitate interconnection of all communication-related devices (terminals, network elements, public networks, private networks, etc.), independent of technology, so that end-user applications experience no unsatisfactory or annoying impairments. For speech applications this means the delivery of highly intelligible, natural-sounding speech signals at near-optimum acoustic levels, and free of bothersome levels of distortion, echo, and delay. For non-speech applications, this means that associated transactions (e.g., fax, interactive data, e-mail, web browsing, video streaming) can be completed satisfactorily.

This plan allows for service provider solutions that meet or exceed these criteria. Conversely, if an end user or network operator chooses network elements that cannot support this criterion, they do so with the understanding that all applications may not perform satisfactorily and thus interconnection to their network by other network operators may not be desirable.

ITU-T Recommendation G.101

The transmission plan

1 Scope

This Recommendation applies to the transmission planning necessitated by telecom deregulation, and in particular, the division of responsibility if multiple network operators are involved in a given connection.

The aim of this Recommendation is to provide guidance for transmission planning of telecommunication networks used primarily for narrow-band services. Traditionally this involved a subdivision of an international connection into "national systems" and an "international chain", usually in a regulated environment. As with previous versions of ITU-T Rec. G.101, this version can be used to do transmission planning for such an environment; however, this version can also be applied to the deregulated multi-operator environment in which there is no clear division of responsibilities. The reference model in this Recommendation is focused on supporting this latter case.

The transmission plan established by this Recommendation takes into account transmission parameters and impairments, different network configurations and elements, modern transmission techniques as well as effects resulting from the combination of the various factors having influence on the resulting transmission quality. The main focus of this Recommendation is transmission planning of voice services. Implications associated with the planning of other kind of services, e.g., data services, will also be discussed.

The transmission plan allows for the adjustment, i.e., staying within acceptable limits, of important transmission parameters and implementation of network configurations and components in order to guarantee adequate end-to-end transmission performance at all times and under all network-operating conditions. The aim is to allow for flexible control of transmission parameters instead of allocating defined limits. The various transmission parameters are not considered on an individual basis, rather the combined effects of variations in transmission parameters are assessed.

This Recommendation forms a framework for transmission planning. The intention is to: explain basic planning objectives and rules applicable to modern telecommunication networks, list the main technical (transmission) parameters of importance, and provide reference to relevant ITU-T Recommendations.

2 References

The following ITU-T Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published. The reference to a document within this Recommendation does not give it, as a stand-alone document, the status of a Recommendation.

– ITU-T Recommendation G.100 (2001), *Definitions used in Recommendations on general characteristics of international telephone connections and circuits*.

– ITU-T Recommendation G.100.1 (2001), *The use of the decibel and of relative levels in speechband telecommunications*.

– ITU-T Recommendation G.107 (2003), *The E-model, a computational model for use in transmission planning*.

– ITU-T Recommendation G.108 (1999), *Application of the E-model: A planning guide*.
– ITU-T Recommendation G.108.1 (2000), *Guidance for assessing conversational speech transmission quality effects not covered by the E-model*.
– ITU-T Recommendation G.108.2 (2003), *Transmission planning aspects of echo cancellers*.
– ITU-T Recommendation G.109 (1999), *Definition of categories of speech transmission quality*.
– ITU-T Recommendation G.111 (1993), *Loudness ratings (LRs) in an international connection*.
– ITU-T Recommendation G.113 (2001), *Transmission impairments due to speech processing*.
– ITU-T Recommendation G.113 Appendix I (2002), *Provisional planning values for the equipment impairment factor Ie and packet-loss robustness factor Bpl*.
– ITU-T Recommendation G.114 (2003), *One-way transmission time*.
– ITU-T Recommendation G.115 (1996), *Mean active speech level for announcement and speech synthesis systems*.
– ITU-T Recommendation G.116 (1999), *Transmission performance objectives applicable to end-to-end international connections*.
– ITU-T Recommendation G.117 (1996), *Transmission aspects of unbalance about earth*.
– ITU-T Recommendation G.120 (1998), *Transmission characteristics of national networks*.
– ITU-T Recommendation G.121 (1993), *Loudness ratings (LRs) of national systems*.
– ITU-T Recommendation G.122 (1993), *Influence of national systems on stability and talker echo in international connections*.
– ITU-T Recommendation G.126 (1993), *Listener echo in telephone networks*.
– ITU-T Recommendation G.131 (2003), *Talker echo and its control*.
– ITU-T Recommendation G.136 (1999), *Application rules for automatic level control devices*.
– ITU-T Recommendation G.142 (1998), *Transmission characteristics of exchanges*.
– ITU-T Recommendation G.161 (2002), *Interaction aspects of signal processing network equipment*.
– ITU-T Recommendation G.164 (1988), *Echo suppressors*.
– ITU-T Recommendation G.165 (1993), *Echo cancellers*.
– ITU-T Recommendation G.167 (1993), *Acoustic echo controllers*.
– ITU-T Recommendation G.168 (2002), *Digital network echo cancellers*.
– ITU-T Recommendation G.169 (1999), *Automatic level control devices*.
– ITU-T Recommendation G.172 (1988), *Transmission plan aspects of international conference calls*.
– ITU-T Recommendation G.173 (1993), *Transmission planning aspects of the speech service in digital public land mobile networks*.
– ITU-T Recommendation G.174 (1994), *Transmission performance objectives for terrestrial digital wireless systems using portable terminals to access the PSTN*.

– ITU-T Recommendation G.175 (2000), *Transmission planning for private/public network interconnection of voice traffic*.

– ITU-T Recommendation G.176 (1997), *Planning guidelines for the integration of ATM technology into networks supporting voiceband services*.

– ITU-T Recommendation G.177 (1999), *Transmission planning for voiceband services over hybrid Internet/PSTN connections*.

– ETSI EN 300 462-1-1 V1.1.1 (1998), *Transmission and Multiplexing (TM); Generic requirements for synchronization networks; Part 1-1: Definitions and terminology for synchronization networks.*

– ETSI EN 300 462-6-1 V1.1.1 (1998), *Transmission and Multiplexing (TM); Generic requirements for synchronization networks; Part 6-1: Timing characteristics of primary reference clocks.*

– ISO/IEC 11573:1994, *Information technology – Telecommunications and information exchange between systems – Synchronization methods and technical requirements for Private Integrated Services networks.*

3 Terms and definitions

This Recommendation defines the following term:

3.1 transmission reference point: A hypothetical point at or near the sending end of each channel (preceding the virtual switching point), used as the "zero relative level point", in the computation of nominal relative levels. In the case of a digital end office, the main distribution frame is considered the 0 dBr point.

4 Abbreviations

This Recommendation uses the following abbreviations:

HRC Hypothetical Reference Connection

NTP Network Termination Point

PCM Pulse Code Modulation

TE Terminal Equipment

UNI User-Network Interface

5 Fundamental principles of transmission planning

In general a good transmission plan is set up in order to deliver to users signals that are at a desirable level and free from objectionable amounts of delay, echo and distortion. Thus the transmission plan has to take into account transmission parameters and impairments, different network configurations and elements and provide guidance on adequate adjustment of network settings. Depending on the type of network, e.g., traditional narrow-band telephone networks, mobile networks, packet switched networks, specific transmission plans have to be set up in order to take care of specific transmission impairments and conditions.

Transmission planning is a subset of the total network planning, both when creating new networks or extending existing ones. Modern networks allow a very great flexibility in routing and "intelligent" switching features. It is important, however, that transmission aspects are not forgotten in the planning process.

For complicated networks, one should keep in mind the abilities of the particular signalling that is implemented. Advanced signalling systems could, in addition to performing their normal functions, convey information about certain transmission parameters in connections. (Examples of transmission parameters of interest are accumulated delay, existence of echo cancellers in the path, terminals not needing network echo control, accumulated impairments, choice of particular routes for calls with special requirements for high quality connections, etc.)

In order to set up a transmission plan, one has first to analyse the end-to-end connection(s) to be established by the telecommunication network that is being planned. This is done by defining a reference connection/configuration; depending on the complexity of the network, different reference configurations may be needed. A basic reference configuration is given in 6.1, and typical network configurations (as examples) are given in 7.2.

Based on these network configurations, the network components and the resulting transmission impairments need to be identified. Clause 7.1 provides guidance on network components and clause 8 on transmission impairments.

All this information is summarized in a transmission plan specifying the network configuration and requirements for transmission parameters.

In the following paragraph, a list of fundamental principles applicable to all kind of networks is provided and illustrated in Figure 1. Planning guidance on specific network configurations also follows.

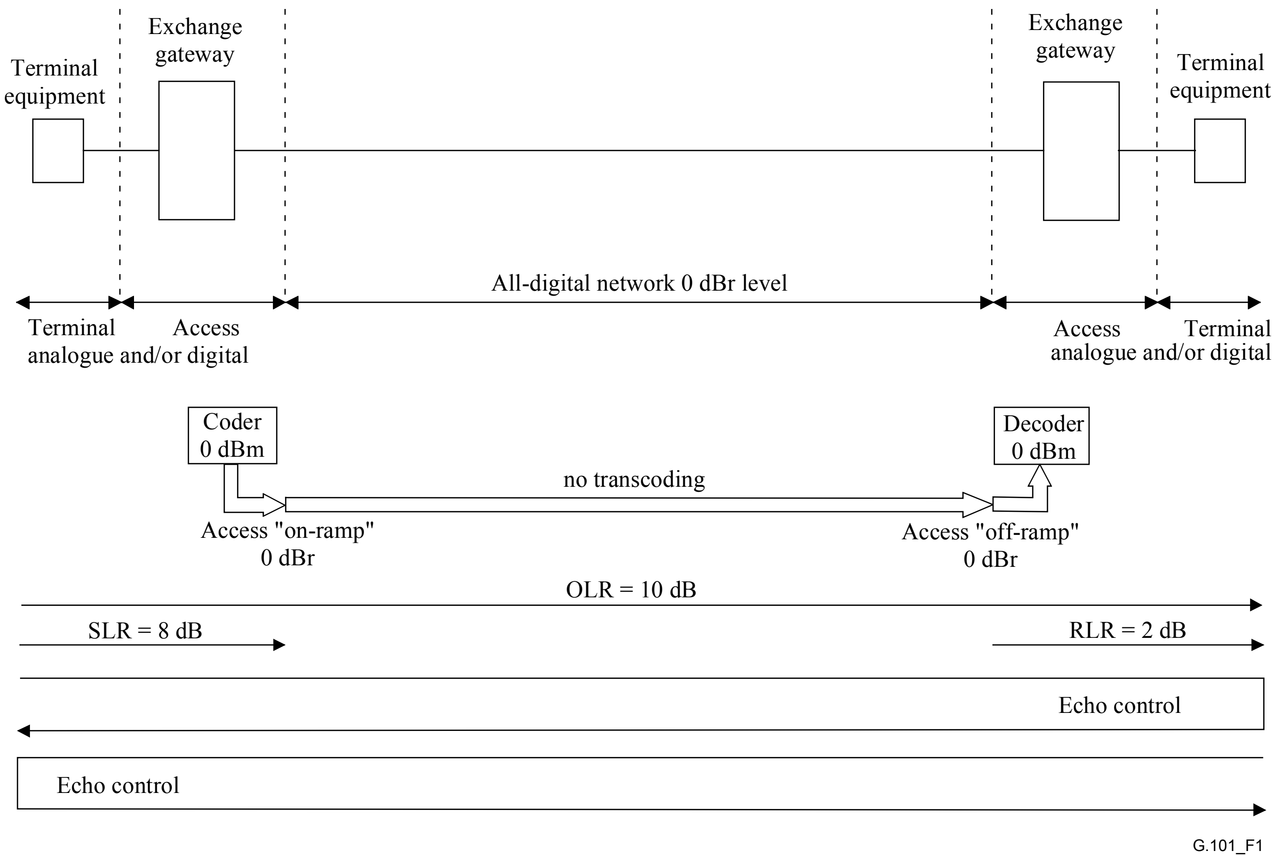


Figure 1/G.101 – Fundamental principles to transmission planning

1) The all-digital network is a lossless bidirectional data highway. This applies equally to TDM and packet switched networks.

2) Access to the all-digital data highway via the on- and off-ramps may be with a digital terminal, a voice gateway (VG)/PBX plus an analogue terminal, a digital local exchange line card plus an analogue terminal, etc.

   NOTE 1 – An on-ramp and off-ramp to the data highway is defined as a 0 dBr point.

3) Ideally, loss/gain is inserted in the analogue domain. Inserting loss/gain in the digital domain can potentially cause quantization distortion.

4) Ideally, loss/gain is inserted in one location. Distributed loss plans can potentially cause increased distortion/reduced dynamic range.

5) The terminal/gateway devices, in conjunction with the digital local exchange loss plan, determine end-to-end acoustic levels.

   The optimum end-to-end loudness is the Overall Loudness Rating OLR = 10 dB.

   Digitally encoded speech levels on the digital data highway should be set to provide optimum dynamic range. This will be achieved if the Send and Receive Loudness Ratings are partitioned as: SLR = 8 dB, and RLR = 2 dB.

   NOTE 2 – The loudness ratings are defined according to ITU-T Rec. P.79.

6) No echo paths shall exist between the on- and off-ramps within the network, i.e., U-turns are not permitted on the data highway.

   Echo paths, between corresponding off- and on-ramps to the data highway, must be controlled to eliminate degradation due to echo.

   - This will be achieved by appropriately specifying TELR values for the access system, in accordance with ITU-T Rec. G.131.
   - Echo may be controlled with passive or active echo control devices or a combination of both.
   - In principle, echo control devices should be located as close as possible to the source of the echo, to minimize echo tailpath delay.

7) Transcoding should be avoided.

For traditional networks supporting narrow-band services (e.g., 64 kbit/s PCM over the PSTN), the existing G.100-series of ITU-T Recommendations (e.g., G.113, G.131) provide guidance. It is recommended to use the E-model (ITU-T Rec. G.107) for confirmation that overall quality is as expected.

NOTE 3 – The E-model is a computational model for assessing the combined effects of variations in several transmission parameters that effect conversational quality of 3.1 kHz handset telephony. The output of the model can be transformed into an estimate of users' perception of end-to-end speech quality categories, allowing for relative comparisons of transmission conditions of various connection scenarios. The combined subjective effects of impairments due to speech compression, delay, packet loss, etc. are captured by the E-model and by no other method.

For networks using mobile access technology, low-bit-rate coding and/or packetized transport, additional guidance is required. The effects of impairments associated with these technologies are uniquely captured by the E-model and therefore it is recommended as the primary transmission planning tool.

Further guidance is given in the following ITU-T Recommendations:

Guidance on specific transmission planning aspects:	G.172 Transmission plan aspects of international conference calls G.173 Transmission planning aspects of the speech service in digital public land mobile networks G.174 Transmission performance objectives for terrestrial digital wireless systems using portable terminals to access the PSTN G.175 Transmission planning for private/public network interconnection of voice traffic G.176 Planning guidelines for the integration of ATM technology into networks supporting voiceband services G.177 Transmission planning for voiceband services over hybrid Internet/PSTN connections
Guidance on echo and automatic level control devices:	G.136 Application rules for automatic level control devices G.164 Echo suppressors G.165 Echo cancellers G.167 Acoustic echo controllers G.168 Digital network echo cancellers G.169 Automatic level control devices
Guidance on important transmission parameters:	G.111 Loudness ratings (LRs) in an international connection G.113 Transmission impairments due to speech processing G.114 One-way transmission time G.121 Loudness ratings (LRs) of national systems G.122 Influence of national systems on stability and talker echo in international connections G.126 Listener echo in telephone networks G.131 Talker echo and its control
Guidance on the E-model and its usage:	G.107 The E-model, a computational model for use in transmission planning G.108 Application of the E-model: A planning guide G.108.1 Guidance for assessing conversational speech transmission quality effects not covered by the E-model G.108.2 Transmission planning aspects of echo cancellers G.109 Definition of categories of speech transmission quality

6 Basic reference model and definitions

This Recommendation defines the following terms:

6.1 Reference model

A complete connection/path includes two terminals connected via one or several telecommunication networks. There are a variety of possible connection types like national, international calls, calls routed over multiple interconnected networks, calls with private networks involved, etc. Furthermore, the type of connection and transmission elements (analogue, digital, circuit/packet switched, wireless, etc.) used in the different constituent parts of the connection needs to be considered. Therefore the variety of possible network configurations is almost infinite, and especially challenging for a transmission plan whose goal is to achieve an adequate end-to-end transmission performance.

Depending on the service that is intended to be offered and the involved network(s), several different connections have to be analysed. Usually the most typical, i.e., predominant connections are taken into account, resulting into a mix of different connections providing a good representation of the real-life situation. The analysis of these connections was traditionally done using well defined, fixed allocation hypothetical reference connections (HRCs). These HRCs were useful in transmission planning in order to obtain an overview of the considered connection and to simplify the identification of all terminal, connection and transmission elements which contribute impairments to the end-to-end transmission performance.

However, today the assumptions underlying the HRCs are likely to be invalid, e.g., the concepts of national networks and fixed allocations no longer apply. Accordingly, the reference model to be used for modern transmission planning must reflect this reality, which means the amount of detail that can be represented in a reference configuration is that represented in Figure 2. This configuration consists of an end-to-end connection with a terminal at each end of the connection and access networks connected by transit networks.

NOTE – Traditionally in transmission planning, it was assumed that national calls are handled by a single national "Administration" and for international calls more than one network operator was involved. This was reflected by dividing an international connection into a national and an international chain. Due to deregulation, this methodology can no longer be applied as a general concept. But in some cases this methodology may still be useful.

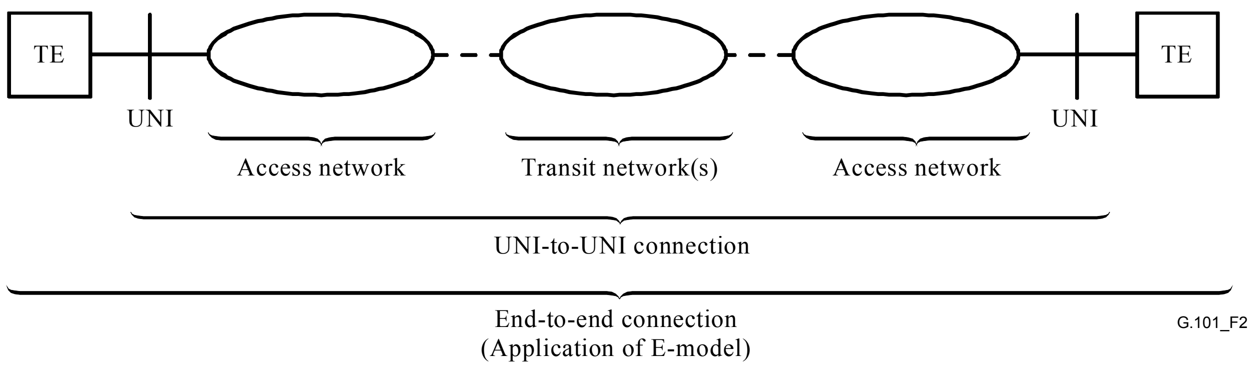


Figure 2/G.101 – Basic network configuration of an end-to-end connection

Starting from this basic configuration, the respective end-to-end (telephone) configuration under consideration can be evaluated. In real life, this configuration will become in most cases more complicated but it may also be simplified. For example, in the case of a local/national call between two end users with telephone sets directly connected to the network, e.g., via an analogue access line, there will be only one network. The public network provides at the same instant the access and transit networks. For international calls, the network configuration will have in most cases the same structure as in Figure 1, but there may be also multiple transit networks.

The reference configuration provides information on all relevant terminal-, connection- and transmission elements having an influence on the end-to-end performance of the connection. The performance is effected, i.e., degraded, by various kinds of impairments. They are identified and assessed by transmission parameters. Depending on the technology used for setting up the end-to-end connection, specific transmission impairments/parameters have to be taken into account. The combined influence of these transmission impairments determine the resulting end-to-end performance of the connection.

6.2 Relative levels

Detailed information on the definition of relative levels and their use and application in telecommunications guidance is given in ITU-T Rec. G.100.1.

6.2.1 PCM Digital Reference Sequence (DRS)

A PCM digital reference sequence is one of the set of possible PCM code sequences that, when decoded by an ideal decoder, produces an analogue sinusoidal signal at the reference frequency (i.e., 1020 Hz) at a level of 0 dBm0.

Conversely an analogue sinusoidal signal at 0 dBm0 at the reference frequency applied to the input of an ideal coder will generate a PCM digital reference sequence.

NOTE 1 – Ideal coders and decoders are assumed to show a relation between analogue and digital signals and vice versa exactly in accordance with the appropriate tables for A-law or μ-law of ITU-T Rec. G.711. "Real" coders and decoders are assumed to be such that the performance characteristics of an encoder/decoder pair between audio frequency ports will meet the requirements of ITU-T Rec. G.712 (see ITU-T Rec. P.310).

NOTE 2 – The digital reference sequence above is a theoretical concept used to describe the conversion between analogue and digital signals in connection with transmission planning. For practical measurements, other Digital Test Sequences (DTS) are used, e.g., as described in ITU-T Rec. P.310.

6.3 Circuits and connections

Definitions and terminology with respect to circuits and connections are given in Annex A.

7 Typical network components and configurations

7.1 Network components

Transmission components can be categorized into three main groups: terminal elements, connection elements and transmission elements.

7.1.1 Terminal elements

In this clause only terminals designed for the transmission of speech signals in real-time are taken into account; however, it is recognized that the transmission plan also supports non-speech terminals.

Voice terminals include all types of telephone sets, digital or analogue, wired, cordless, or mobile, including the acoustical interfaces to the user's mouth and ear.

Telephone sets are characterized, with respect to speech transmission, by their Send Loudness Rating (SLR) and Receive Loudness Rating (RLR) which contribute to the Overall Loudness Rating (OLR) of a connection. Other parameters, such as the Sidetone Masking Rating (STMR), the Listener Sidetone Rating (LSTR), the design of the handset (D-Factor), the frequency response in send and receive directions and the noise floor, also contribute to the end-to-end speech transmission performance.

In case of wireless or IP-based systems, additional distortions and delay may be added, depending on the coding and modulation algorithms used in such devices. As noted, the E-model can account for the effects of these impairments.

7.1.2 Connection elements

Connection elements are all types of switching or routing equipment, such as local exchanges (for the direct connection of terminal elements) and transit exchanges within networks. They may use circuit- or packet-based technology.

Switching systems contribute to the end-to-end delay, due to signal processing, and also to the amount of quantization distortion associated with digital pads and code conversion.

Packet-based routers contribute, in addition, with delay variation versus time and packet loss. Where 4-wire to 2-wire conversions take place within or between switching equipment interfaces, signal reflections contribute to impairments as a source for echo effects.

7.1.3 Transmission elements

Transmission elements are all kinds of media used as the facility between connection elements and between connection elements and terminal elements. The physical media of these elements may be metallic (e.g., copper), fibre-optics or wireless. The signal coming from the user can be either analogue or digital, but between connection elements it is almost allways digital.

Impairments associated with analogue signal transmission include propagation time (generally proportional to distance), loss, frequency response and noise (mainly due to longitudinal interference). For planning purposes, impairments due to frequency response and noise can usually be neglected for short and medium line lengths.

For digital transmission elements, the main transmission impairment is caused by the propagation time via metallic, optic, and radio media. For wireless sections, additional delay is introduced, depending on the coding and modulation algorithm used. Where the transmission element includes analogue-to-digital conversion, loss and distortion are additional impairment factors.

In digital transmission elements, systems use either 64 kbit/s Pulse Code Modulation (PCM) of ITU-T Rec. G.711, or one of the compression techniques based on low bit-rate codecs. Major influence to the transmission quality of these systems may be caused by additional distortions in terms of Equipment Impairment Factor (Ie), and by additional mean one-way delay.

Echo cancellers may also be classified as a type of transmission element when they are deployed in the network. However, some terminals also contain echo cancellers.

Multiplexing is generally used to transport several channels via one single physical media. A variety of multiplexing systems are in use in the existing networks:

– Frequency Division Multiplex (FDM).
– Time Division Multiplex (TDM).
– Digital Circuit Multiplication Equipment (DCME).
– Packet-based facilities:
  • connection oriented (ATM);
  • connectionless (Ethernet, IP).

7.2 Network configurations

The variety of possible network configurations is almost infinite. Network configurations depend on the type of connection – short, average or long national or international calls – and the type of switching and transmission elements used in the different constituent parts of the connection. Only a few examples will be shown here in order to illustrate some important cases.

Figure 3 depicts a fully analogue routing between two analogue sets. The most critical parameters are the overall loudness rating OLR and, in some instances, the noise, provided that for very long connections echo cancellers are employed. (This case was more common in the past.)

In Figure 4, a digital telephone set is connected to an analogue set via a fully digital route. In addition to the conventional PCM systems, a virtual ATM circuit is included in the digital path. (Such configurations are likely to be fairly common in the near future.) At the analogue end, reflections may occur at the hybrid in the exchange. The main impairment in those configurations is caused by talker echo effects at the digital telephone set, due to the transmission time – here increased by the ATM system – and the signal reflections at the far end hybrid. Note, however, that even when ATM circuits are not used in the digital path, there is a possibility for noticeable talker echo.

Although echo effects can be diminished by echo cancellers, a very long transmission time in itself causes impairments to the speech communication quality. A connection subjected to such a risk is shown in Figure 5, depicting a call from a mobile phone (for instance of the GSM type) via a satellite link. Although not shown in the figure, DCMEs are routinely employed on satellite circuits for voice traffic. This type of equipment also increases delay and may produce a particular kind of distortion, if not properly dimensioned.

A further example of the use of DCME systems is shown in Figure 6, where a cordless telephone is used at one end of the connection. There, the tandeming of the low rate codecs in the cordless set and the DCME decreases the transmission quality.

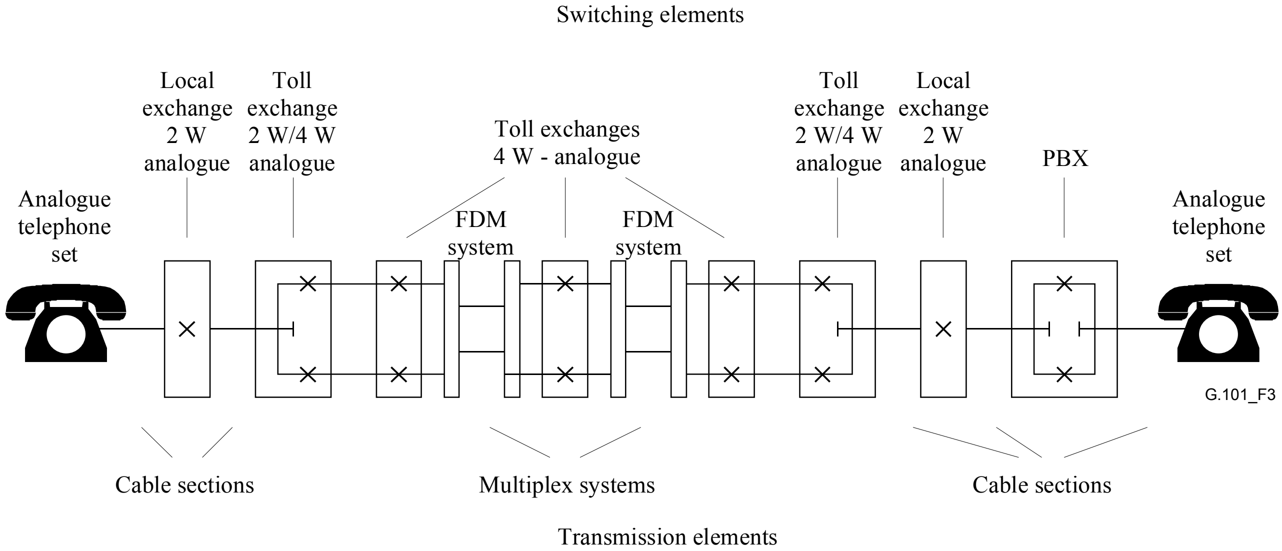


Figure 3/G.101 – Typical configuration for a fully analogue connection

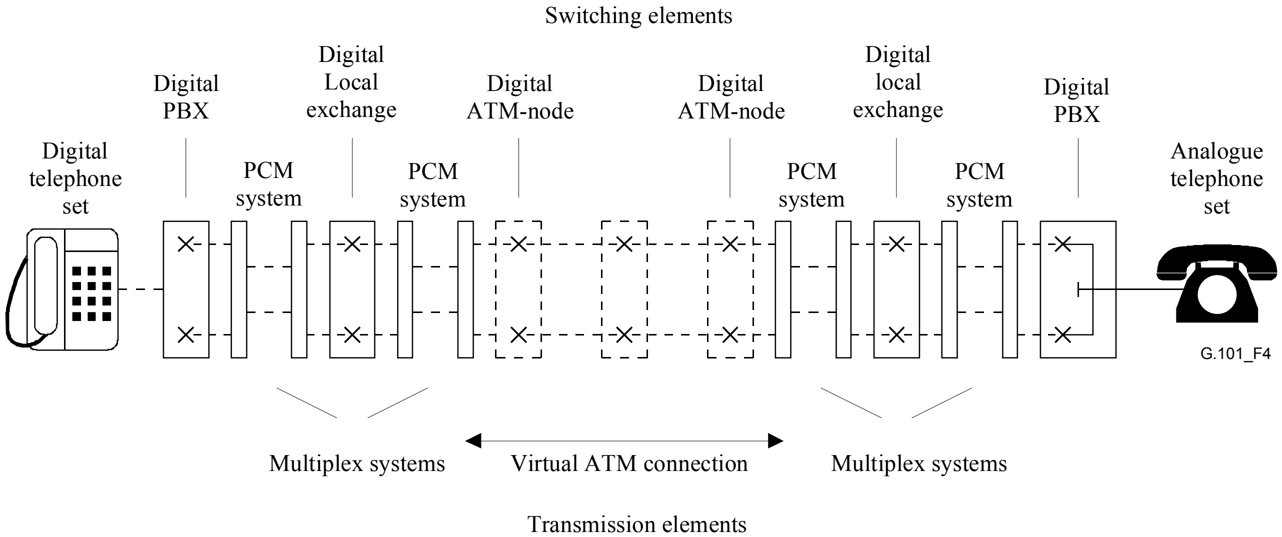
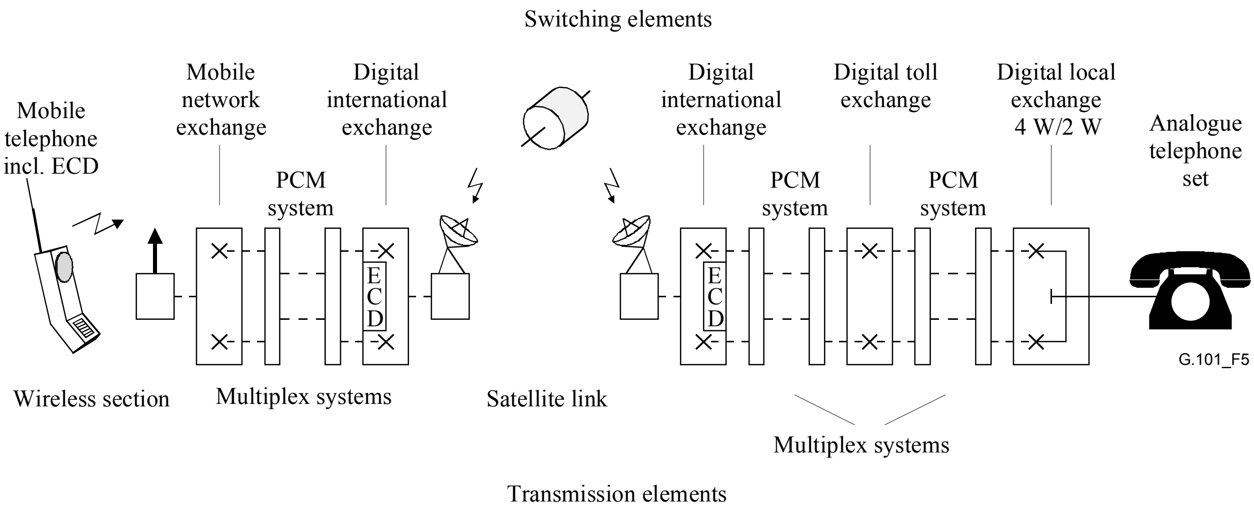
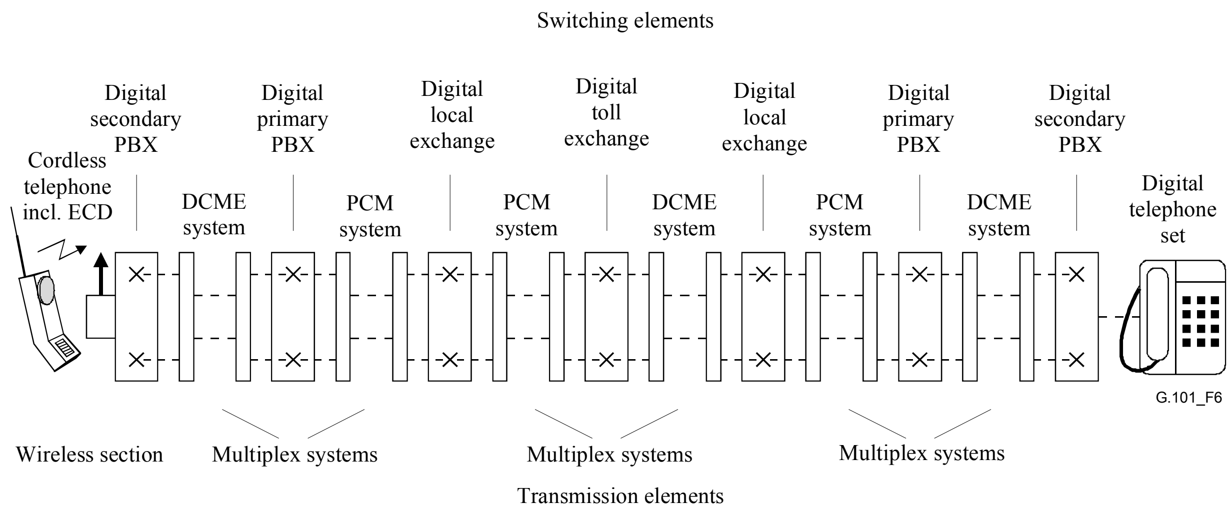


Figure 4/G.101 – Configuration for a fully digital connection, including ATM, between a digital and an analogue set



ECD Echo Control Device

Figure 5/G.101 – Configuration for a mobile telephone connected to the PSTN, with a satellite link included



ECD Echo Control Device

Figure 6/G.101 – Connection for a cordless telephone, the transmission links containing DCMEs

8 Technical requirements/Transmission impairments

In this clause important specific transmission parameters are listed. The subjective effects of some of these impairments are given in ITU-T Rec. P.11. The transmission parameters are not explained in detail but references to relevant ITU-T Recommendations where guidance may be found are given.

8.1 Loudness ratings

The transmission level point at the digital exchange should be 0 dBr. Loss in analogue access sections (i.e., local exchange lines) should be such that at the A/D, D/A conversion point is SLR = 8 dB and RLR = 2 dB, when analogue telephones with the nominal SLR/RLR values defined in national or harmonized standards are attached to the NTPs.

Wherever possible, signal level adjustment should be made in the analogue domain. Digital loss or gain pads limit the available level range and increase signal distortion and should not be used if possible.

ITU-T Rec. G.111 provides nominal values for loudness ratings and ITU-T Rec. P.310 provides SLR = 8 dB and RLR = 2 dB audio performance requirements and associated testing for telephone band digital telephones.

8.2 Noise, crosstalk and group delay distortion

Analogue local access loops and A/D and D/A conversion systems should be designed to achieve performance in respect of noise, crosstalk, and group delay distortion that at least meets the levels recommended in ITU-T Recs Q.551 and Q.552.

The resultant group delay distortion on a connection is a function of the number of translations to voiceband that occur within the network. ITU-T Rec. G.712 provides guidance in this matter.

8.3 Stability control by loss allocation to circuits

More detailed information is given in ITU-T Rec. G.122. Application of these rules for typical circuits is illustrated in Figure 7.

The Type 1 circuit in Figure 7 a) represents the case where digital transmission is used for the entire length of the circuit and digital switching is used at both ends. Such a circuit can generally be operated at a nominal transmission loss of 0 dB as shown because of the transmission properties exhibited by such circuits (e.g., relatively small loss variations with time).

The Type 2 circuit in Figure 7 b) represents the case where the transmission path is established on a digital transmission channel in tandem with an analogue transmission channel. Digital switching is used at the digital end and analogue switching at the analogue end.

It might be possible, in some cases, to operate Type 2 circuits with a nominal loss of 0 dB in each direction of transmission. For example, where the analogue portion could be provided with the necessary gain stability and where the attenuation distortion would permit such operation.

The Type 3 circuit in Figure 7 c) represents the case where the transmission path is established over a tandem arrangement consisting of digital/analogue/digital channels as shown. Digital switching is assumed at both ends.

The Type 4 circuit in Figure 7 d) represents the case where the transmission path is established over a tandem arrangement consisting of analogue/digital/analogue channels as shown. Analogue switching is assumed at both ends.

The Type 5 circuit in Figure 7 e) represents the case where analogue transmission is used for the entire length of the circuit and analogue switching is used at both ends.

For analogue or mixed digital/analogue connections, it is recommended that a loss of $L = 0.5$ dB is inserted.

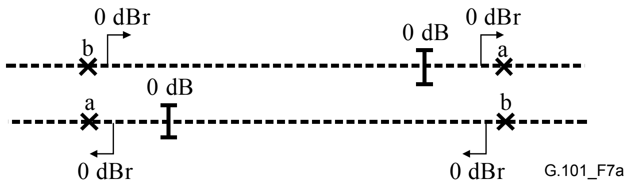
NOTE – General remarks concerning the allocation of losses in the mixed analogue/digital circuits:

In circuit types 2, 3 and 4, the pads needed to control any variability in the analogue circuit sections (arising from loss variations with time or attenuation distortion) are shown in a symmetrical fashion in both directions of transmission. However, in practice, such arrangements may require non-standard levels at the boundaries between circuit sections. Administrations are advised that should they prefer to adopt an asymmetric arrangement, e.g., by putting all the loss into the receive direction at only one end of a circuit (or circuit section); then, provided that the loss is small, e.g., a total of not more than 1 dB, there is no objection on transmission plan grounds.

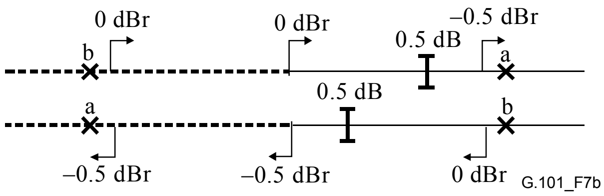
The small amount of asymmetry that results in the international portion of the connection will be acceptable, bearing in mind the small number of international circuits encountered in most actual connections.

As far as national circuits are concerned, Administrations may adopt any arrangements they wish provided that the requirements of 2.2/G.121 are complied with.

In some cases transmultiplexers may be used, in which case the circuits may not be available at audio-frequency at the point at which a pad symbol is used in the diagrams of Figure 7. Should the variability of the analogue portions merit additional loss, the precise way in which this loss can be inserted into the circuits is a matter for Administrations to decide bilaterally.

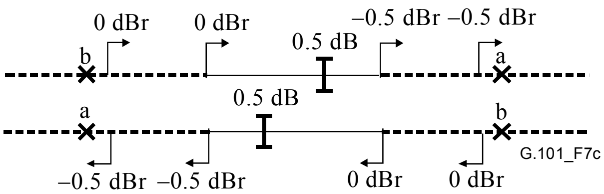


a) Type 1 circuit – All digital circuit with digital switching at both ends



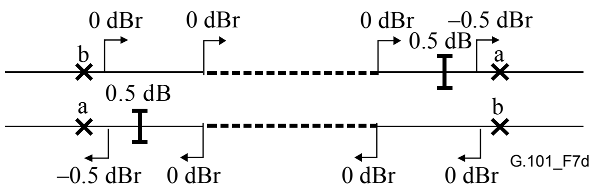
NOTE – The loss is required if the analogue circuit section introduces significant amounts of attenuation distortion or variation of time.

b) Type 2 circuit – Digital/analogue circuit with digital switching at one end and analogue switching at the other end



NOTE – The loss is required if the analogue circuit section introduces significant amounts of attenuation distortion or variation of time.

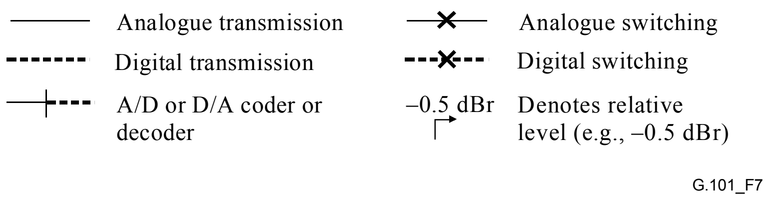
c) Type 3 circuit – Digital/analogue/digital circuit with digital switching at each end



NOTE – The loss is required if the analogue circuit section introduces significant amounts of attenuation distortion or variation of time.

d) Type 4 circuit – Analogue/digital/analogue circuit with analogue switching at each end

e) Type 5 circuit – All analogue circuit with analogue switching at each end



NOTE 1 – The pad symbols in the circuits are not intended to imply that real attenuators are needed. They are a convention of transmission planning engineers.

NOTE 2 – The relative level of a point in a digital link is determined using ideal decoders as described in 6.5.3/G.100.1.

Figure 7/G.101 – Types of international circuits

8.4 Delay/echo

Irrespective of the effect of delay on echo, absolute delay is an impairment and should be controlled. Absolute delay does not impair the intelligibility of speech but if the total delay exceeds around 100 ms from mouth to ear, it begins to affect the interactivity of conversations. Therefore, if possible, large delays should be avoided.

Detailed guidance on one-way transmission time is given in ITU-T Rec. G.114.

Some low bit rate coders introduce high values of delay. Some terminal equipment also introduces high values of delay. Simple terminals may not include echo reduction techniques.

Echo control should normally be provided using echo cancellation rather than echo suppression. The provision by the network of additional echo cancellation facilities (or additional performance by these facilities) should be a matter for commercial negotiation. For example, extra cancellation may be provided at an appropriate extra charge. No echo paths should be introduced within the network.

Terminals that introduce higher delays (e.g., some cordless telephones) normally provide their own echo reduction to compensate for the additional delay. Echo control devices in the network do not normally reduce acoustic echo in terminals because they are designed to reduce electrical echo originating in hybrids. End terminals must satisfy minimum requirements for TCLw to effectively eliminate echo.

For modern network environments echo control is one of the key parameters, since the increased use of digital technology in transmission and switching systems causes lower loss and higher delays for connections. This makes talker echo effects more noticeable.

Listener echo is usually not a problem in modern networks, if talker echo is controlled. (Guidance can be found in ITU-T Rec. G.126.)

Rules for evaluation and control of talker echo are given in ITU-T Rec. G.131. Note that disturbance from talker echo depends not only on the mean transmission time but also on the Talker Echo Loudness Rating (TELR). The latter is a function of the sensitivity of the talker's telephone set, the magnitude of the impedance mismatch causing signal reflection and the loss between the set and the point of reflection. One way to diminish talker echo is to employ a suitable impedance strategy in the 2-wire parts of the network. This is discussed in ITU-T Rec. Q.552.

Echo cancellers should meet the requirements of ITU-T Rec. G.168. Note that for echo cancellers to operate with full efficiency, the echo path should have an essentially linear amplitude characteristic.

8.5 Packet loss

Transmission performance is affected by lost or discarded packets. A packet may be lost due to congestion in the network; a packet may also be discarded at the destination. This would occur, for example, when a packet is sufficiently late that the destination declares the packet lost. Loss of a single packet will result in loss of one or more coded speech frames, depending on the speech coder in use and the number of frames per packet. Therefore the speech coder should be robust with respect to loss of coded frames. In particular, if multiple frames are assembled into a single packet, the performance of the speech coder should be assessed under frame loss conditions that reflect those of the network in use.

Packet systems make typically 6 ms (e.g., ATM) or 20 ms (e.g., VoIP) packets of speech content, so any packet loss is potentially a serious impairment. Many low bit rate speech coders used for packet speech have intrinsic packet loss concealment, which is effective for small amounts of packet loss. However, packet concealment algorithms are not effective for long strings of consecutive lost packets which can occur as a result of congestion in routers.

ITU-T Rec. Y.1541 provides network QoS classes based on packet loss, delay and delay variation. For ATM transport, ITU-T Rec. I.356 provides similar cell-based classes.

The effect of packet loss on transmission quality is covered by the E-model. Impairments resulting from specific codecs may be calculated and can take into account both random and bursty packet loss conditions. For detailed information, see ITU-T Rec. G.107 and Appendix I of ITU-T Rec. G.113.

8.6 The effect of coding and signal processing in the digital path

The impairments in the form of distortions, which are caused by low bit rate encoding and decoding or other forms of signal processing in the digital path, are described and quantified in ITU-T Rec. G.113. Note that digital processing and low bit rate coding in general increase the transmission time.

Depending on the coding technology, tandeming can be especially problematic and should be avoided. Note that the use of G.711 does not cause these problems.

The E-model can be used to assess these low bit rate codec effects and help in choosing a coder for specific configurations and applications.

8.7 Channel compression techniques

Data for such equipment is given in ITU-T Recs G.763 and G.765.

8.8 Bit integrity

Bit integrity is possible across a network only where the path is wholly digital. It may be required for services such as 64 kbit/s unrestricted but is not required for speech. Signalling processing devices such as echo cancellers, low bit rate coders and digital loss and gain pads corrupt bit integrity. If there is to be an option for bit integrity, then it should be possible to disable such devices.

8.9 Bit error performance

Digital transmission equipment should be used to ensure that the error performance specified in the G.820-series of ITU-T Recommendations is exceeded by a substantial margin in normal operating conditions. When faults are not present, errors in transmitted information are normally caused by temporary local electromagnetic phenomena. The levels in these Recommendations allow for the occurrence of these phenomena. Equipment should be designed to perform much better than these requirements in the absence of these phenomena.

8.10 Synchronization

Proper synchronization design is part of the network planning strategy, because synchronization impairments will affect the quality of the calls: the networks should be synchronized as defined in the series of documents ETSI EN 300 462-1-1/6-1 and ISO/IEC 11573 in order to achieve the slip rate objectives defined in ITU-T Rec. G.822. ITU-T Rec. G.810 provides definitions and abbreviations used in timing and synchronization Recommendations. For information on control of jitter and wander within digital networks, guidance is found in ITU-T Recs G.823, G.824 and G.825.

8.11 Attenuation distortion

The attenuation distortion of an end-to-end connection depends on the filtering with respect to the conversion from analogue to digital and vice versa as well as on the electro-acoustical properties of the terminal.

All-digital connections with analogue access interfaces should meet the attenuation distortion requirements as given in ITU-T Rec. G.712 or the Q.550-series of ITU-T Recommendations, respectively.

On all-digital connections which utilize digital telephone sets and all-digital facilities, the attenuation response should meet the attenuation distortion requirements of ITU-T Rec. P.310 for narrow-band handset telephones; or ITU-T Rec. P.311 for wideband handset telephones, or ITU-T Rec. P.341 for wideband hands-free telephones.

8.12 Effect of syllable speech clipping

Syllable speech clipping (i.e., in the time domain) in DCME, PCME, or wireless accesses will affect speech transmission quality to varying degrees based on the length of the clipped speech segments and the total per cent of time that clipping occurs. At the present time, the only meaningful guidance for speech transmission quality in the presence of speech clipping can be derived from subjective evaluations.

8.13 Evaluation of impairments, singly and in combination

ITU-T Rec. G.113 provides planning guidance for designers of networks that would form part of a telephone connection taking into account various transmission impairments introduced by digital speech processing systems. The information provided is for use in conjunction with the transmission planning approach described in ITU-T Recs G.107, G.108 and G.109, i.e., the Impairment Factor Method, which the algorithm of the E-Model (ITU-T Rec. G.107) is based upon. The Impairment Factor Method allows various transmission impairments to be evaluated during transmission planning.

9 Service planning and QoS aspects

The transmission considerations of this Recommendation are a major element of the overall QoS experienced by users. ITU-T Rec. G.1000 gives a framework for QoS planning, and ITU-T Rec. G.1010 gives the requirements for specific applications. Also the G.170 series of ITU-T Recommendations gives transmission planning guidance for some special network configurations e.g., multipoint teleconferences, hybrid ATM/PSTN networks, hybrid IP/PSTN networks, etc.

Annex A

Terminology used in traditional transmission planning

The following definitions and terminologies are used for transmission planning purposes related to analogue transmission elements/sections, primarily analogue accesses. The definitions are still valid and recommended. However, the principles behind them are well understood and in use for many years. Today guidance is needed on the effects on modern digital transmission technology, e.g., packet switched and coding technologies. Therefore, the information on analogue technologies has been summarized in this annex and not in the main body of this Recommendation.

A.1 Circuits and connections

In transmission planning the overall transmission path is divided into sections termed circuits, each having its own 0 dBr Transmission Reference Point (TRP). Circuits are linked together in the exchanges, forming connections. Most often circuits connect switching centres; subscriber lines connected to a local exchange are also termed circuits. Thus, a circuit is constituted by all permanently interconnected equipment. In this way clearly defined segments are obtained with fixed transmission parameters to be adjusted.

In the following, important circuits used in transmission planning are defined.

A.1.1 telephone circuit: In transmission planning, and in the ITU-T G-series Recommendations, a telephone circuit denotes a telecommunication circuit with associated terminating equipment, directly connecting two switching devices or exchanges, in line with Note 2 to the general definition of a circuit (see 1.4/G.100). For simplicity, the term "circuit" is often used instead of "telephone circuit" in the ITU-T G-series Recommendations.

NOTE 1 – Conceptually, (telephone) circuits are those parts of the connections that remain intact and permanently associated with the switches at each end, after a connection is taken down and before a new connection is established. Routine measurements of (telephone) circuits are made in a way approaching the ideal concept as closely as possible, i.e., between circuit access points which between them will include as much of the (telephone) circuit as possible (see 2.1.2/M.565).

NOTE 2 – In some cases, mainly in private networks, the definition of circuit is not applicable. Exchanges within a private network are normally interconnected via leased lines, specified at the interfaces of the transmission systems.

A.1.2 subscriber's (telephone) line; subscriber's loop (in telephony): A link between a public switching entity and a telephone station or a private telephone installation or another terminal using signals compatible with the telephone network.

NOTE – In French, the term "ligne de réseau" is used only when the private telephone installation is a private branch exchange or an internal telephone system.

A.1.3 local (telephone) system; local (telephone) circuit: The combination of subscriber's station, subscriber's line and feeding bridge if present, see Figure A.1.

NOTE 1 – This term is used in the context of transmission planning and performance.

NOTE 2 – In ITU-T English texts, the term "local (telephone) system" is preferred.

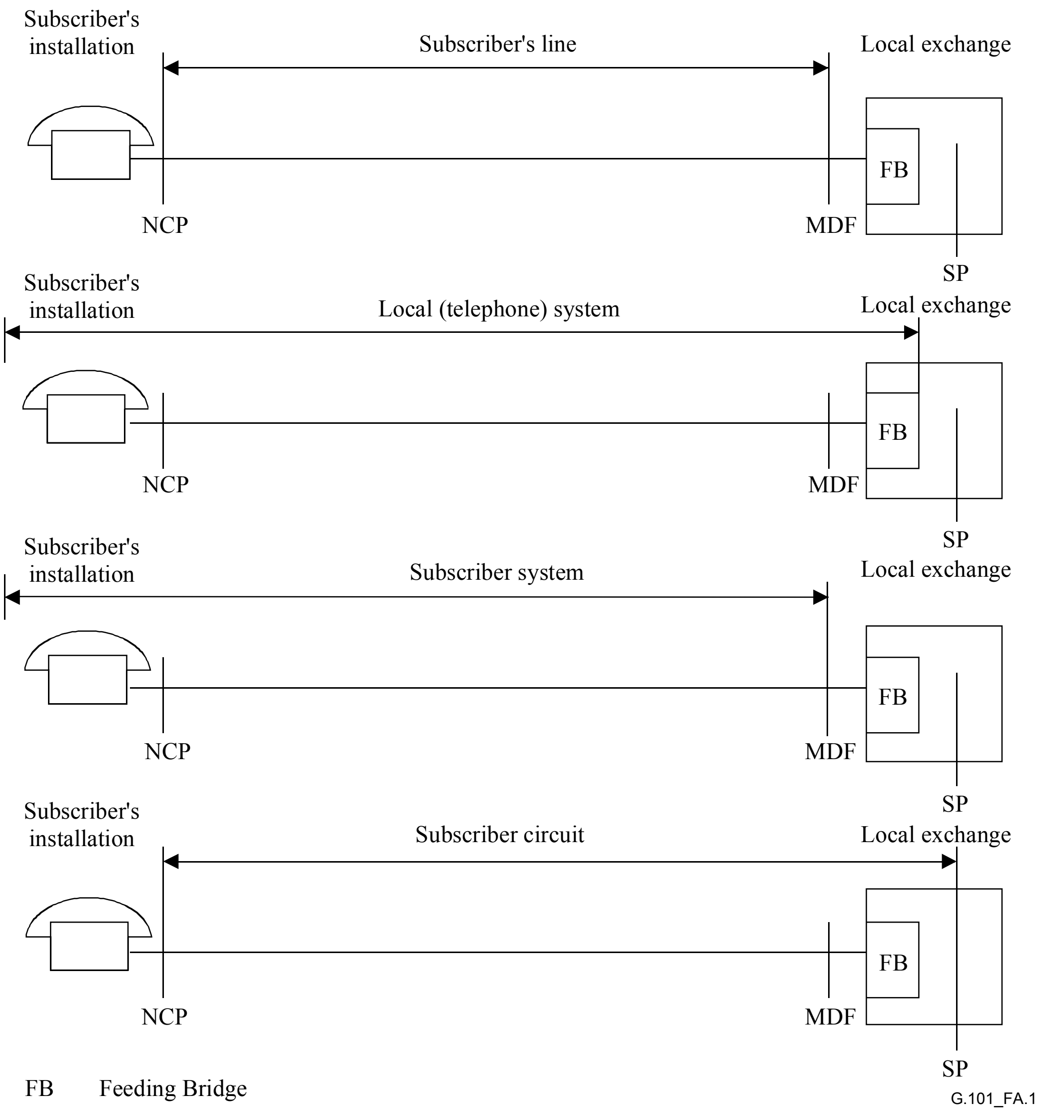
NOTE 3 – A local network includes the local system, the local exchanges and interconnecting circuits.

A.1.4 subscriber system (in transmission planning): A subscriber's line associated with that part of the private telephone installation connected to this line during a telephone call, see Figure A.1.

NOTE – This term is used in the context of transmission planning and performance.

A.1.5 subscriber circuit: The circuit between the local exchange and the Network Connection Point (NCP), i.e., the interface between the public network and the subscriber's installation, see Figure A.1. This interface may for instance be at the MDF of a PBX, at a socket for connecting a telephone set, etc. The location of this interface is dependent on national regulations and practice.

NOTE – In the local exchange, the subscriber circuit usually includes "half" of the exchange in an analogue exchange and in a digital exchange the input and output of the circuit usually will be a digital bit stream corresponding to the "exchange test points" defined in 1.2.1.1/Q.551.



FB Feeding Bridge
MDF Main Distribution Frame
NCP Network Connection Point
SP Switching Points

Figure A.1/G.101 – Subscriber's line, local (telephone) system, subscriber system and subscriber circuit

A.1.6 telephone circuit loss: This is the composite loss at the reference frequency 1020 Hz between the circuit input and its output, as defined in Note 1 below. This will include any loss in the associated terminating equipment of the switching centres.

NOTE 1 – Defined for transmission planning purposes, the input and output of a circuit are hypothetical points in an exchange where circuits are directly interconnected (see 2.3.3/M.560) and are consequently not accessible, e.g., for measurement purposes. To enable the necessary correlation to be made between planning and measured values, "circuit access points" are defined in ITU-T Rec. M.565; their relation to the circuit input and output are shown in Figures 1a and 1b/M.565 for analogue and digital exchanges, respectively. After carrying out the measurement between these points, any necessary correction is made for the effect of circuit access arrangements to allow circuit loss to be determined (see 3.1.2/O.22).

NOTE 2 – For digital exchanges, it will be seen that the circuit input and output correspond to the "exchange test points" as defined in 1.2.1.1/Q.551. Since the levels at these points are defined in terms of the digital bit streams appearing there, neither digital access arrangements nor passage through the digital switchblock will involve any loss or gain, provided the bit sequence is not modified. On the other hand, any recoding, for example such as produced by a "digital pad", will result in circuit loss. To allow at least the mandatory alternative of "bit-transparent" connections (i.e., retaining bit integrity, see 3.1.2/Q.554), the "pad" function must be switchable, i.e., it must be possible:

a) to make measurements under conditions which simulate at will each real traffic condition requiring a different pad value;

b) to check the bit error ratio (see 3.1.1/Q.554), which of course needs to be done in the absence of intentional changes in the bit stream.

NOTE 3 – For analogue exchanges, the assumption is made that nominal switchblock losses (defined in 3.2/Q.45) are divided equally between the two circuits being interconnected in the exchange. The variance of switchblock losses contributes negligibly to the variance of circuit loss by comparison with the objective for loss variations in transmission systems (see 1.1.2/M.160).

NOTE 4 – The circuit access points are not to be confused with the "line access points", usually located at a distribution frame (see ITU-T Rec. M.120). These points are not of interest for transmission planning, but only for the maintenance services for line-up and fault localization purposes.

NOTE 5 – The input and output of international circuits are defined as the Virtual International Connecting Points having defined relative levels. This is necessary to have a defined boundary between the national and international parts of a connection.

A.1.7 connection: A chain of circuits interconnected by switching points, between two different points in the network.

In transmission planning, the loss of a connection is normally the sum of the losses of the circuits making up the connection. (The losses of the switching centres are normally included in the circuit losses.)

NOTE 1 – A complete connection is a connection between two terminal equipment connected to the network.

NOTE 2 – When analogue or mixed analogue/digital circuits are interconnected in the exchanges, "level jumps" often have to be introduced. In a complete connection, the sum of all "level jumps" and digital losses should not exceed 6 dB in the short term and 3 dB in the long term.

Appendix I

Traditional network planning in a regulated environment by subdividing an international connection into "national systems" and an "international chain"

I.1 Subdivision of telephone networks with regard to the interfaces between network operators

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

I.2 General

In the past, most often telephone users had national calls handled by their national Administration. For international calls only, more than one network operator was involved in the connections. This was reflected in ITU-T documents by the division of a complete international connection into "national systems" and "the international chain". Clause I.3 describes this methodology, including the conventions and precautions to be observed.

Although this situation still exists in many cases today, with the advent of deregulation many users will now have the choice between different network operators for long-distance calls and even for local calls. The tendency is toward a future with a multiplicity of interconnected operator networks. In such cases, it is of course more relevant to use other designations for the constituent parts of a connection. Clause I.4 therefore introduces the terms "terminating network" and "transit networks". (Note, however, that many of the conventions and guidelines presented in I.3 also are applicable here.)

To ensure the overall (end-to-end) voice transmission quality, agreement has to be reached between the operators involved in a connection with regard to critical transmission parameters, always keeping in mind the telephone users' actual expectations and needs. For the multi-operator situations covered by I.4, fewer rules of a general character can be given than for the "national network case" described in I.3. However, a discussion of this is outside the scope of this Recommendation.

Note that technical details of networks are touched upon in clause 7, however, only in the form of presenting some typical examples of network components and configurations with comments on which types of transmission impairments they are likely to cause.

I.3 The national systems and the international chain of circuits

I.3.1 Definition of constituent parts

A complete international telephone connection consists of three parts, as shown in Figure I.1. The division between these parts is determined by the Virtual International Connecting Points (VICPs) in the originating/terminating International Switching Centres (ISCs). These are theoretical points with specified relative levels (see I.3.3 and I.3.4).

The three parts of the connection are:

– Two national systems, one at each end. These may comprise of one or more 4-wire national trunk circuits with 4-wire interconnection, as well as circuits with 2-wire connection up to the local exchanges and the subscriber sets with their subscriber lines.

– An international chain made up of one or more 4-wire international circuits. These are interconnected on a 4-wire basis in the international centres which provide for transit traffic and are also connected on a 4-wire basis to national systems in the international centres.

An international 4-wire circuit is delimited by its Virtual International Connecting Points in an International Switching Centre.

NOTE – The Virtual International Connecting Points may not be the same as the points at which the circuit terminates physically in the switching equipment. These latter points are known as the circuit terminals; the exact position of these terminals is decided in each case by the Administration concerned.

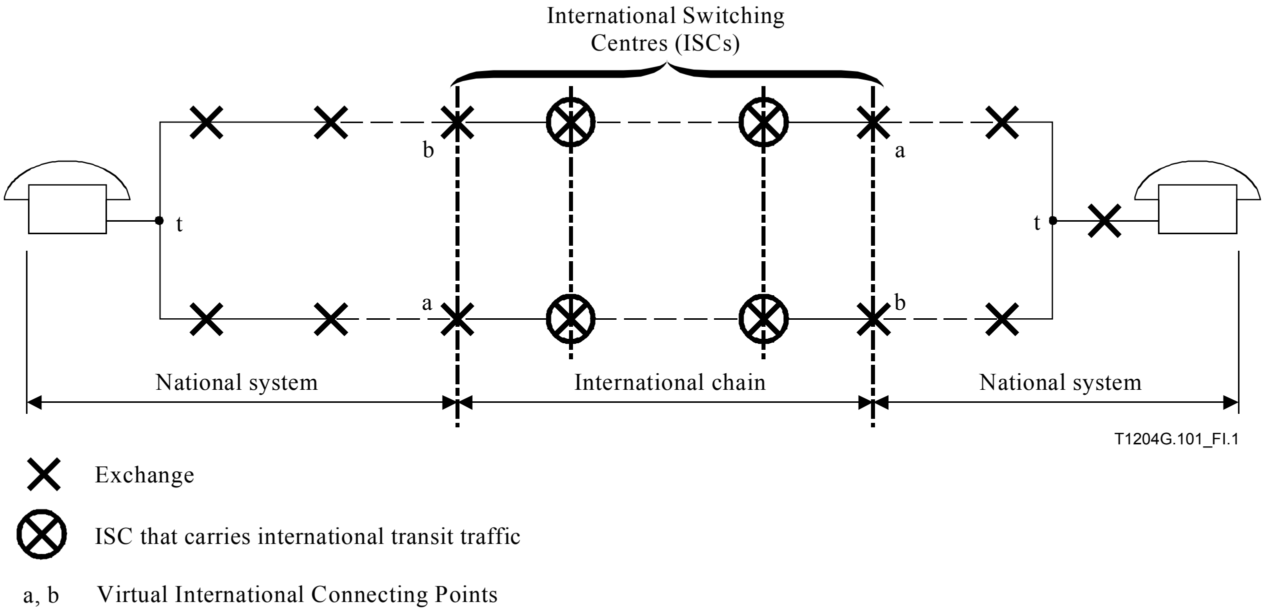


Figure I.1/G.101 – Definition of the constituent parts of an international connection

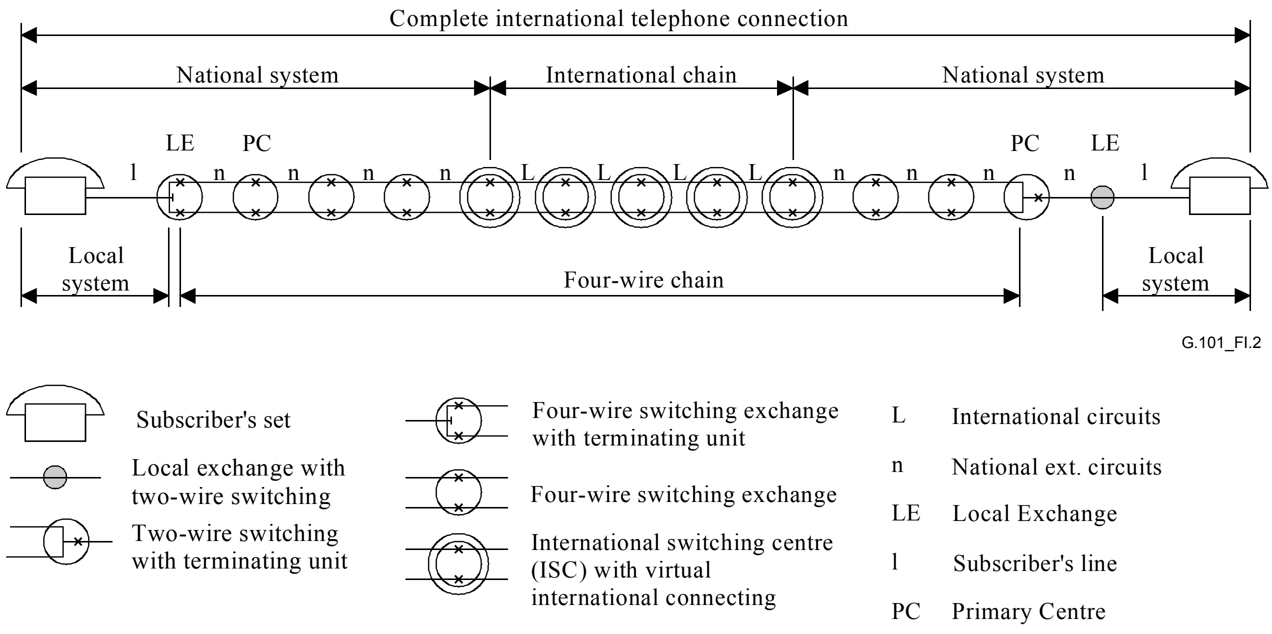
I.3.2 The 4-wire chain

The 4-wire chain (see Figure I.2) denotes the whole unbroken chain of 4-wire national and international circuits in a complete telephone connection, including possible 4-wire circuits between the primary centre and the local exchange and on the subscriber line, e.g., ISDN access and 4-wire or digitally connected PBXs.

I.3.3 Virtual International Connecting Points (VICPs)

The Virtual International Connecting Points define the boundary between the national and international parts of a connection; see Figure I.1. The international connecting points are also used as reference points for transmission quantities recommended for the national and international part of a connection.

NOTE – Earlier, the terms "virtual switching points" and "virtual analogue switching points" were used to define the boundary between the national and international part of a connection. These points, however, were assigned other relative levels.



NOTE – The arrangements shown for the national systems are examples only. In many cases, the local exchange LE (shown as analogue in the figure) is digital.

Figure I.2/G.101 – An international connection to illustrate the nomenclature adopted

I.3.4 Relative levels specified in the Virtual International Connecting Points

The Virtual International Connecting Points of an international 4-wire telephone circuit are by convention fixed to points in the circuit where the nominal relative levels are:

– sending: 0 dBr;

– receiving: 0 dBr for digital circuits or the very short circuits mentioned in Note 4; –0.5 dBr for analogue and mixed analogue digital circuits.

The nominal transmission loss of the international circuits is 0 dB for digital circuits and 0.5 dB for analogue and mixed analogue/digital circuits; see Figure I.3.

NOTE 1 – Usually a 0.5 dB loss has to be introduced in the mixed analogue/digital circuit to satisfy the stability requirements.

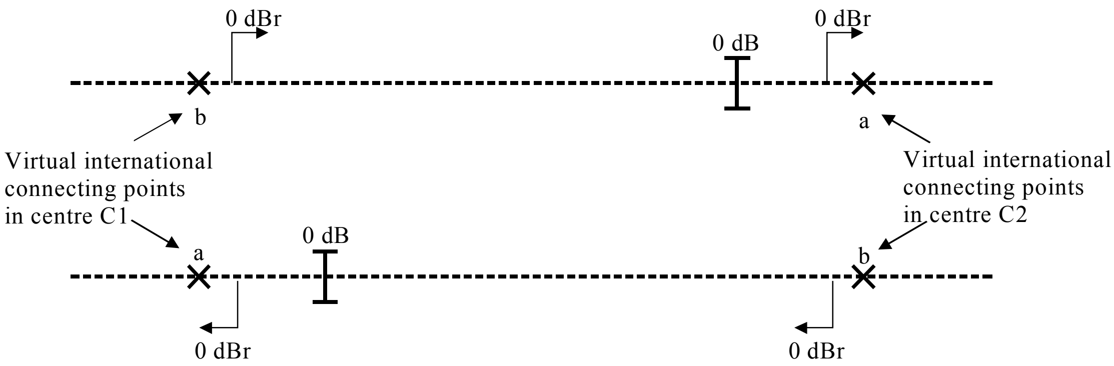
NOTE 2 – The "virtual analogue switching points" used earlier had the relative levels:

– sending: –3.5 dBr;

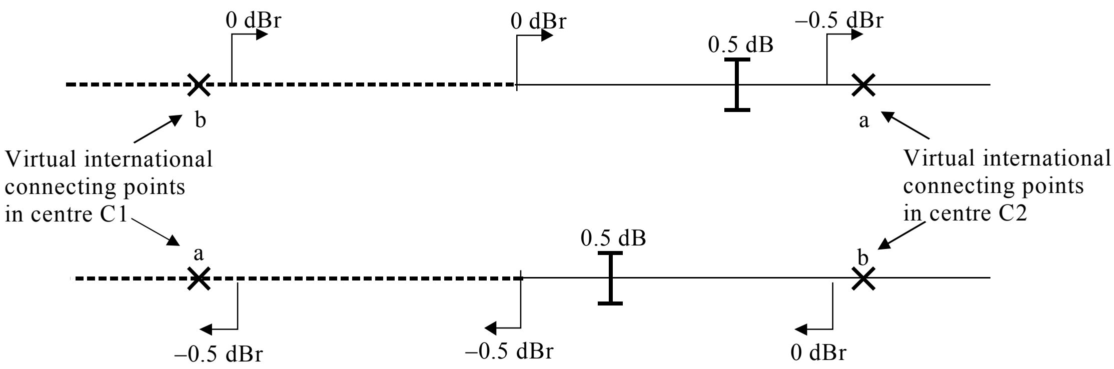
– receiving: –3.5 dBr for digital circuits or the very short circuits mentioned in Note 4; –4 dBr for analogue and mixed analogue/digital circuits.

NOTE 3 – The Virtual International Connecting Points in digital exchanges refer to a digital bit stream, e.g., the exchange test points. In analogue exchanges they often will not be accessible, and will differ from the switching levels nationally used in the ISC.

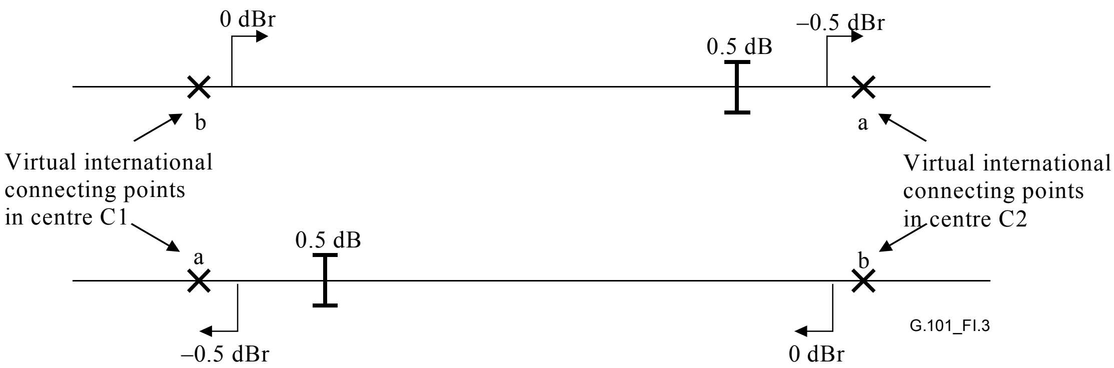
NOTE 4 – If a 4-wire analogue circuit forming part of the 4-wire chain contributes negligible delay and variation of transmission loss with time, it may be operated at zero nominal transmission loss between Virtual International Connecting Points. This relaxation refers particularly to short 4-wire tie-circuits between switching centres – e.g., circuits between two international switching centres in the same city.



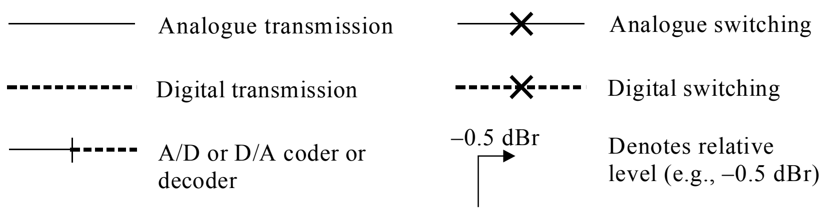
a) Definition of Virtual International Connecting Points for a digital international circuit between digital international centres



b) Definition of Virtual International Connecting Points for a mixed analogue/digital international circuit between an analogue and a digital international centre



c) Definition of Virtual International Connecting Points for an analogue international circuit between analogue international centres



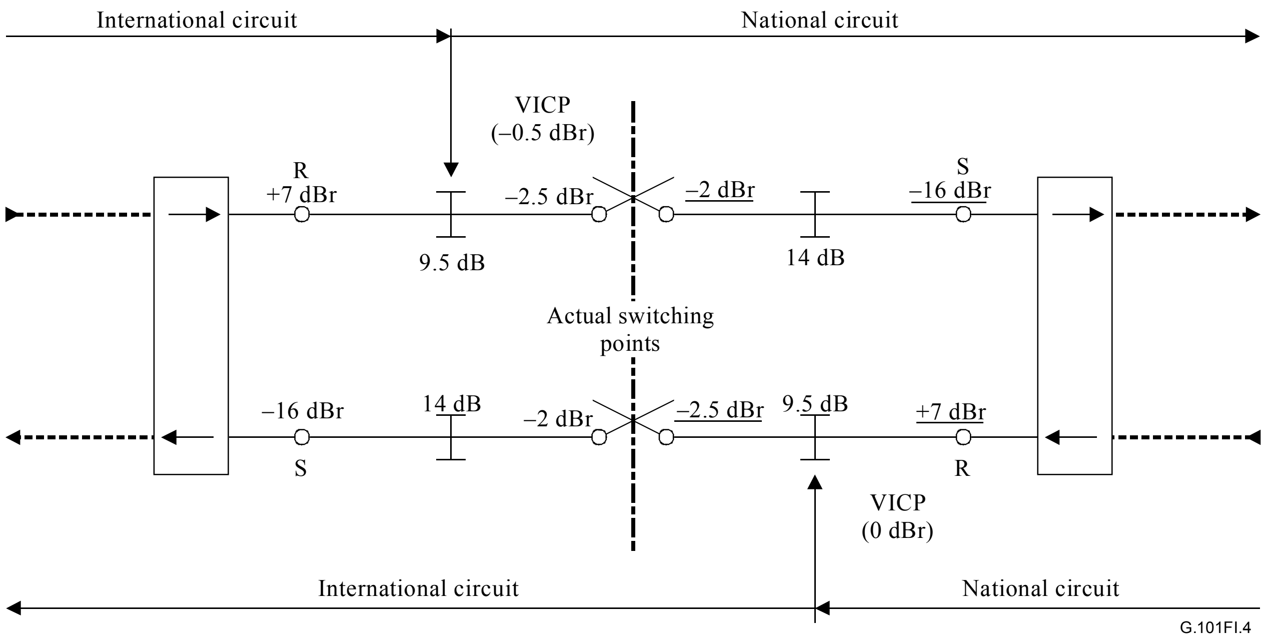
NOTE – The relative level at a point in a digital link is determined using ideal decoders as described in 6.5.3/G.100.1.

Figure I.3/G.101 – Definitions for international circuits

I.3.5 Circuit access point

The ITU-T has defined circuit access points as being "4-wire access points so located that as much as possible of the international circuit is included between corresponding pairs of these access points at the two centres concerned" (see ITU-T Rec. M.565). These points, and their relative level (with reference to the transmission reference point), are determined in each case by the Administration concerned. They are taken as the basic reference points of known relative level to which other transmission measurements will be related. In other words, for measurement and lining-up purposes, the relative level at the appropriate circuit access point is the relative level with respect to which other levels are adjusted.

An example showing an actual arrangement is shown in Figure I.4.



NOTE 1 – Underlined values of relative levels refer to the national circuit. Values of relative levels not underlined refer to the international circuit. In an actual switching centre, the international connecting points may not physically exist. As shown in this figure, the VICP is situated inside a 9.5 dB pad.

NOTE 2 – Each of the 9.5 and 14 dB pads include half the exchange loss.

NOTE 3 – In this example, the national circuit has 0.5 dB loss, giving a 0.5 dB "level jump" in the switch at the input to the international circuit.

Figure I.4/G.101 – Example showing a simplified representation of a transit connection in an international switching centre

I.3.6 Measurement frequency

For international circuits, the reference frequency for maintenance measurements should be within the ranges of 804-860 Hz or 1004-1020 Hz. (See ITU-T Rec. O.6.)

I.4 Multi-operator networks

For multi-operator networks (which implies deregulation), the definition of the constituent parts are "terminating networks" and "transit networks" which all are managed by separate operators. Thus this terminology reflects the *division of responsibility* between the operators involved in connections.

As the name implies, the terminals involved in a connection are to be found in the terminating networks. Each call originates and ends in a terminating network, either the same or a different one. The terminating networks can be interconnected directly or by one or more transit networks. A terminating network may contain cross-connect facilities for routing calls to different switched networks.

Figure I.5 shows an example of a transit network connected to a terminating network.

In modern networks, it can be assumed that transit networks are all digital. At the interconnection points, the conventions, definitions and rules for digital circuits given in I.3 apply.

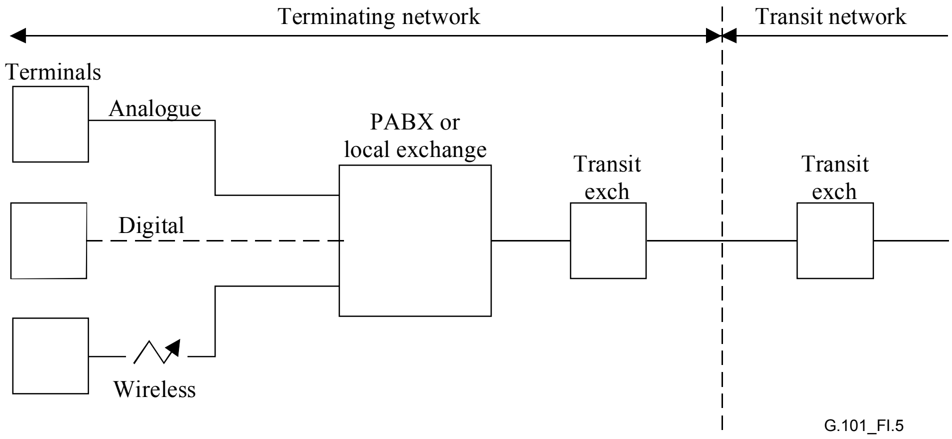


Figure I.5/G.101 – An example showing a simple "terminating network" connected to a "transit network", the constituent parts of connections in a multi-operator market (The terminating and transit networks are managed by different network operators)

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