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Manual on Air Traffic Services (ATS) Ground-Ground Voice Switching and Signalling

Approved by the Secretary General
and published under his authority

First Edition — 2002

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AMENDMENTS

The issue of amendments is announced regularly in the *ICAO Journal* and in the monthly *Supplement to the Catalogue of ICAO Publications and Audio-visual Training Aids*, which holders of this publication should consult. The space below is provided to keep a record of such amendments.

RECORD OF AMENDMENTS AND CORRIGENDA

AMENDMENTS		
No.	Date	Entered by

CORRIGENDA		
No.	Date	Entered by

FOREWORD

This manual was developed with the assistance of the ATS Voice Switching and Signalling Study Group (AVSSSG) formed by the Air Navigation Commission in 1998 with the task of updating the ICAO provisions on air traffic services (ATS) ground-ground voice communications to support the introduction of digital technology.

The purpose of this manual is to provide technical guidance, specifications and reference material to assist in the engineering of the ground voice communications facilities specified in Annex 10, Volume III, Part II, Chapter 4,[8] in order to meet the ground-ground communications requirements specified in Annex 11, Chapter 6, Section 6.2.[9]

This manual is not a complete guide to the planning and deployment of networks since there is already ample relevant material generally available. Rather, the manual aims to provide general indications of the issues that are considered to be important in the context of deploying ground voice communications networks to support air traffic management and to provide guidance on the direction in which such networks should develop.

Historically, voice communications between air traffic services (ATS) units have been carried out via direct analogue circuits and, more recently, via switched analogue circuits. The economics of telecommunications service provision have now altered in favour of digital technologies. This manual, therefore, provides recommendations and technical guidance for administrations wishing to migrate their international ATS communications to make use of switched digital networking.

This manual contains two chapters: Chapter 1, dealing with Operational Requirements and Chapter 2, dealing with Engineering Requirements. Chapter 1 describes in detail the facilities specified in Annex 10, Volume III, Part II, Chapter 4,[8] and explains how these are used to meet the requirements for ground-ground communications specified in Annex 11, Chapter 6, Section 6.2.[9] Chapter 2 considers the following aspects of ATS communications:

a) network planning, implementation and management;

b) quality of service (QoS) and network performance issues;

c) network numbering;

d) details of the signalling systems needed to support the defined telephone facilities;

e) supervisory tone;

f) security aspects; and

g) migration to digital voice switching and signalling technology.

Appendix A describes a method for assigning signalling delay to network elements. Appendix B lists all the publications referenced in the text.

The manual incorporates, by dated or undated reference, provisions from other publications necessary to assist administrations with the deployment of digital switched network technologies as the means of providing the ground voice facilities needed for air traffic management purposes.

It is intended that the manual be kept up to date. Future editions will most likely be improved on the basis of experience gained and of comments and suggestions received from users of this manual. Therefore, readers are invited to give their views, comments and suggestions on this edition. These should be directed to the Secretary General of ICAO.

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GLOSSARY

Address. A string or combination of decimal digits, symbols and additional information that identifies the specific termination points of a connection in a network.

Administration. An organization responsible for providing international air traffic management communication services.

Air traffic services (ATS) ground voice network. A telecommunications network providing telecommunication services to support operational air traffic management processes.

Air traffic services unit. A generic term meaning variously, air traffic control unit, flight information centre or air traffic services reporting office.

Bilateral agreement. An agreement between two administrations to provide a certain capability or act in a particular manner.

Circuit. A combination of two transmission channels permitting bidirectional transmission of signals between two points to support a single communication.

Class of service. The privileges, priorities, limitations and restrictions determining a party's ability to initiate outgoing calls, to receive incoming calls and to invoke/use relevant supplementary facilities.

Controller working position (CWP). In the context of this manual, one particular type of terminal equipment used specifically for the purpose of performing operational duties of air traffic management.

Digital leased line. A point-to-point digital circuit rented by an administration from a public network operator or telecommunications service provider for exclusive use by the administration for the provision of air traffic management communication services.

E1/T1. The primary rates of the plesiochronous digital hierarchy. E1 refers to the 2 048 kbit/s primary rate. T1 refers to the 1 544 kbit/s primary rate.

End VCS. In the context of a particular call, an originating or terminating VCS. It can also be a gateway VCS; depending on the capabilities of the signalling system with which interworking occurs.

Equipment impairment factor (Ie). A number allocated to a network element, in units of "eif", that indicates the anticipated incremental level of impairment that would result when this element is inserted into a connection.

Incoming gateway VCS. A gateway VCS that routes an incoming call from a route employing one signalling system on to an inter-VCS link employing a different system.

Inter-VCS link (transmission link). A link between two VCSs comprising the totality of the signalling transfer means (i.e. a signalling channel) and the user information transfer (i.e. speech channels) means.

Key. A device that not only includes a key but also any means of activating a facility, including push-buttons, computer mouse, screen icons and touch-sensitive panels, etc.

Name. A string of characters used for the name identification of a party involved in a call.

Network. A set of equipment (terminal equipment, switching equipment, call-processing equipment, etc.) located at geographically dispersed locations and interconnected via transmission links to provide telecommunication services to a defined group of users.

Network performance. The ability of a network or network portion to provide the functions related to communications between users.

Number. An address restricted to containing numerical values, as defined by a numbering plan.

Originating VCS. In the context of a call, the VCS to which the A-party's terminal equipment is attached.

Outgoing gateway VCS. A gateway VCS that routes an incoming call from an inter-VCS link employing one signalling system on to a route employing another signalling system.

Party (A-party, B-party and C-party). The users involved, sequentially, in a telephone call as follows:

A-party: the user who initiates a call (the calling party);

B-party: the user who first receives the call (the called party and/or the connected party); and

C-party: the user who joins an established call, e.g. in a conference.

Private integrated services network (PISN). An integrated services digital network providing services to a specific set of users (different from a public network, which provides services to the general public).

Private integrated services network exchange (PINX). A PISN nodal entity that provides automatic connection handling functions used for the provision of telecommunication services. A private automatic branch exchange (PABX) is a typical example of a PINX. In the context of this manual, a voice communication system (VCS) is one particular type of PINX. In some countries a VCS is known as a voice switching and control system (VSCS).

Private signalling system no. 1 (PSS1). An internationally standardized digital signalling protocol, used between the nodes of a PISN. PSS1 is also known as QSIG. This document refers to it as PSS1 (QSIG).

Quality of service. The degree of satisfaction experienced by the user of a service, brought about by the collective effect of the mechanisms employed (support performance, operability performance, serviceability performance, security performance) to ensure adequate performance of that service.

Terminal equipment. An item of equipment attached to a telecommunications network to provide access for a user to one or more services of that network. A telephone is a typical example of terminal equipment.

Terminating VCS. The VCS to which the B-party's terminal equipment is attached.

Toll quality (PSTN quality). The average quality of long-distance public switched telephone network connections, i.e. good intelligibility, good speaker identification, naturalness, only minor disturbing impairments.

Transit VCS. In the context of a call, any VCS through which the call passes, excluding the originating VCS, the terminating VCS, an incoming gateway VCS and an outgoing gateway VCS.

User (controller). An air traffic controller or other person using, via terminal equipment, the services provided by ATS ground voice networks to undertake the operational duties of air traffic management.

Note.— This definition does not include personnel carrying out administrative functions. Such persons are considered to be normal telephone users without special requirements arising from the function of air traffic management.

Virtual private network (VPN). A private network based upon the use of shared public switched network infrastructures where those shared infrastructures take the place of dedicated transmission links and transit switching nodes in a private network and incorporate mechanisms to prevent users in one VPN from interacting with the users of a different VPN.

Voice communication system (VCS). See the definition of private integrated services network exchange (PINX) above.

Voice switching and control system (VSCS). See the definition of private integrated services network exchange (PINX) above.

ACRONYMS/ABBREVIATIONS

ACSE	Association control service element
ADPCM	Adaptive differential pulse code modulation
AGVN	ATS ground voice network
AMR	Adaptive multi-rate
ANF	Additional network feature
APDU	Application protocol data unit
ATM	Air traffic management
ATS	Air traffic services
CC	Call control
CCA	Call control agent
CPE	Customer premises equipment
CS-ACELP	Conjugate-structure algebraic-code-excited linear-prediction
CWP	Controller working position
DA	Direct access
DS0	Digital signal level 0
DSE	Dialogue service element
ECAC	European Civil Aviation Conference
EG	ETSI Guide
eif	Equipment impairment factor (unit of measure for)
EN	Norme Européenne (European standard)
ES	ETSI standard
ETR	ETSI technical report
ETSI	European Telecommunications Standards Institute
FIC	Flight information centre
GFP	Generic functional protocol
GoS	Grade of service
GSM	Global system for mobile communications
ICD	Interface control document
Ie	Equipment impairment factor
IEC	International Electrotechnical Commission
IP	Internetworking protocol
ISDN	Integrated services digital network
ISO	International Organization for Standardization
ITU	International Telecommunication Union
ITU-T	International Telecommunication Union — Telecommunication Standardization Sector
LAPD	Link access procedure D
LD-CELP	Low-delay code excited linear prediction
ms	Millisecond
MSC	Message sequence chart
ORIG	Originating
PABX	Private automatic branch exchange
PCM	Pulse-code modulation
PIN	Personal identification number
PSS1	Private signalling system no. 1
PSTN	Public switched telephone network
QDU	Quantization distortion unit
QoS	Quality of service
QSIG	Signalling at the “Q” reference point
SS	Supplementary service
TERM	Terminating
TIA	Telecommunications Industry Association (USA)
TR	Technical report
VAD	Voice activity detection
VCS	Voice communication system
VSCS	Voice switching and control system

Chapter 1

OPERATIONAL REQUIREMENTS

1.1 CORRELATION TO THE REQUIREMENTS OF ANNEX 11, SECTION 6.2

1.1.1 Air traffic services (ATS) requirements for communications within and between flight information regions are set out in Section 6.2 (“Aeronautical fixed service, ground-ground communications”) of Annex 11.[9] This section identifies “communications by direct speech alone . . ., whereby for the purpose of transfer of radar control, the communications can be established instantaneously and for other purposes the communications can normally be established within fifteen seconds”, as being the two major components making up the ATS ground telephone facilities.

1.1.2 The requirements make it apparent that except for the purpose of transferring radar control where “instantaneous” establishment of communications is required, a switched network in which communications could normally be established within fifteen seconds is acceptable for ground telephone purposes. In the case of transferring radar control, Annex 11[9] describes “instantaneous” as meaning “immediate access between controllers”, i.e. no appreciable delay between one controller activating a key and another controller receiving an indication of the call. In the past this has meant the use of dedicated circuits between controllers. Today, however, modern switched digital networks are sufficiently fast and have the capacity and the functionality sufficient to meet the “instantaneous” requirement.

1.1.3 In addition to the operational requirements contained in Annex 11,[9] this manual takes into account current, well-established ground-ground telephone facilities (such as “hotlines”) as well as the performance criteria specified for modern air traffic management (ATM) procedures, such as the reduced radar separation minima applicable in the European Civil Aviation Conference (ECAC)[7] area. In Annex 10,Volume III, Part II, Chapter 4,[8] three different communication mechanisms, or basic call types, are stipulated to fulfil the broad spectrum of known operational requirements, including those specified in Annex 11.[9] The three basic call types are the following:

a) instantaneous access;

b) direct access; and

c) indirect access.

Annex 10[8] also recommends the provision of supplementary facilities in order to meet the requirements of Annex 11,[9] namely:

a) means of indicating the calling/called party identity;

b) means of initiating urgent/priority calls; and

c) conference capabilities.

Sections 1.3 and 1.4, respectively, describe these basic call types and supplementary facilities in detail.

1.2 IMPLEMENTATION

1.2.1 Means of provision of telecommunication services

Implementation of a switched network should be undertaken in an evolutionary manner. Where two or more administrations are involved, adequate coordination should be established through the appropriate ICAO regional offices. Until all administrations concerned have a high level of confidence in a newly implemented network, an adequate contingency plan should be in place. Administrations should agree on a target date for full implementation of a proposed network. It should be noted, however, that even partial implementation of a new digital network will represent an improvement over the existing arrangements in many cases.

1.2.2 Applicability to domestic operational requirements

Switched networks can also be used to meet domestic operational requirements, and the concepts and techniques outlined in this manual, although intended to meet international requirements, may also be applicable to domestic networks. Adequate security should be in place to prevent misuse of the international network by un-authorized users.

1.3 BASIC CALL TYPES (PRIMARY USER GROUND TELEPHONE FACILITIES)

1.3.1 General

1.3.1.1 As has already been noted in Section 1.1, several different basic call types (telephone facilities) are needed to meet the operational ground communications requirements of ATM. These different facilities are generically referred to as the "primary user ground telephone facilities". They are the facilities used to conduct routine and urgent communications between ATS units.

1.3.1.2 Each of the following sections describes, from the user point of view, the operation of one of the primary ground telephone facilities. To assist the description, the term "user" is used to refer to an air traffic controller or other operational person using the facility concerned. To distinguish users who have different roles in the communication, the terms A-party, B-party and C-party are used. The term "VCS" (voice communication system) is used to refer to the equipment providing the facility concerned. This manual makes no assumptions about what form a VCS will take, except to say that a VCS normally has attached to it some kind of terminal equipment (e.g. a controller working position (CWP)) through which a user interacts with the system.

Note.— The supervisory tones referred to in 1.3.2, 1.3.3 and 1.3.4 are defined in Section 2.6.

1.3.2 Instantaneous access facility

1.3.2.1 The operation of a single key by the A-party is all that is required to initiate a call to the B-party. The B-party address is assigned and fixed semi-permanently in the A-party VCS. It is thus uniquely associated with a particular key and each key is labelled as such.

Note.— Some administrations may prefer that it be necessary for the A-party to sustain the key operation for the duration of the call.

1.3.2.2 Dial tone and outgoing signalling tones are not given to the A-party. Ringing/ringback tone is not given to the A-party. Number unobtainable/reorder tone is given to the A-party if the call fails for any reason, including any busy conditions encountered.

1.3.2.3 The arrival of the call from the A-party to the B-party causes, simultaneously, the events detailed in a) to d):

a) The A-party identity is indicated to the B-party either by association with a key assigned and fixed semi-permanently in the B-party VCS or by means of a dynamic display. Due to the usually urgent nature of instantaneous access calls, any visual (and/or audible) alerts should be distinctive from other types of calls.

b) An audible alert is generated at the B-party VCS in accordance with the following options:

   1) no audible alert;

   2) an alert of fixed duration; or

   3) a continuous alert requiring a silencing action by the B-party.

c) The B-party VCS automatically accepts/answers the incoming call without any intervention required by the user. This occurs regardless of the B-party being engaged on any other type of call. Thus, B-party busy is an abnormal situation and should result in number unobtainable/reorder tone being given to the A-party (see 1.3.2.2). At this stage the speech channel from the A-party to the B-party is established. How speech from the A-party is managed (i.e. conferenced with other speech at the B-party working position, switched to a loudspeaker or to a split-headset) is a matter for the B-party administration to decide.

d) By bilateral agreement, the establishment of the call as detailed in c) may also result in the A-party having some monitoring facilities of the B-party's working position, including ground-ground and air-to-ground radiotelephony. This enables the A-party to exercise discretion before passing the message.

1.3.2.4 The B-party may respond to the A-party by activation of a key associated with the incoming call. This action only enables the return speech path and is not a new instantaneous access call. The call is cleared by the A-party only with no effect on other calls in progress at the B-party.

1.3.3 Direct access facility

1.3.3.1 The operation of a single key by the A-party is all that is required to initiate a call to the B-party. The B-party address is assigned and fixed semi-permanently in the A-party VCS. It is, thus, uniquely associated with a particular key and each key is labelled as such.

1.3.3.2 Dial tone and outgoing signalling tones are not given to the A-party. Ringing/ringback tone is optional by bilateral agreement between the A-party and B-party administrations. Busy tone shall be given, if appropriate. However, due to either the exclusive, one-to-one assignments of the keys between the 'A'- and 'B'-parties or the reserved capacity in the B-party dynamic display, it is abnormal for the A-party to encounter the B-party busy. This is a fundamental attribute of the direct access facility.

1.3.3.3 A suitable mechanism (i.e. number unobtainable/reorder tone) shall be provided to inform the A-party, should the call fail for any reason other than because the B-party is busy.

1.3.3.4 The B-party is alerted to the presence of the incoming call by audio and/or visual means, as determined by the B-party VCS. The A-party identity is indicated to the B-party either by association with a key assigned and fixed semi-permanently in the B-party VCS or by means of a dynamic display. The B-party accepts the incoming call by means of a single action associated with a key or dynamic display.

1.3.3.5 Under normal conditions the B-party can receive one or more direct access calls simultaneously and by using the A-party identities, together with a defined operational procedure or experience, the B-party will deal with each call appropriately.

1.3.3.6 At the end of a call, either the A-party or the B-party shall be required to deselect/clear the call. Some implementations may require both parties to deselect/clear the call.

1.3.4 Indirect access facility

1.3.4.1 By entering the full address of the B-party (e.g. by dialling it on a keypad or by entering a short code/speed call number), the A-party can initiate a call to the required B-party.

Note.— As part of the indirect access facility, some administrations permit communication to be established with normal telephone users of the wider private telephone network and/or the public switched telephone network (PSTN) as well as to users with operational responsibility, such as other air traffic controllers.

1.3.4.2 The design of some VCSs may be such that a user action equivalent to "off-hook" may be needed prior to entering the desired B-party address. In this case, dial tone may be given to the A-party. Signalling tones will not normally be given. Ringing/ringback tone and busy tone shall be given to the A-party, as appropriate. A suitable mechanism (i.e. number unobtainable/reorder tone) shall be provided to inform the A-party, should the call fail for any reason other than because the B-party is busy.

1.3.4.3 The B-party is alerted to the presence of the incoming call by audio and/or visual means, as determined by the B-party VCS. The A-party identity may be indicated to the B-party (e.g. by means of a dynamic display) but this could be limited to an indication of the network over which the call was made. The B-party accepts the incoming call by means of a user action equivalent to "off-hook".

1.3.4.4 The B-party may be able to receive an indication of one or more indirect access calls at the same time. By using the A-party identity (if available), the B-party can select which call to answer/handle next. At the end of a call, either the A-party or the B-party can deselect/clear the call by an appropriate user action (equivalent to "on-hook").

1.3.5 Performance requirements of primary user ground telephone facilities (see Table 1-1)

1.4 SUPPLEMENTARY FACILITIES

1.4.1 General

1.4.1.1 Some supplementary facilities are needed to support the primary user ground telephone facilities (see Section 1.1). Each of the following sections describes, from the user point of view, the operation of one of these supplementary facilities.

Note.— Many signalling protocols are not capable of supporting all of the supplementary facilities described below. However, they can all be supported by the PSS1 (QSIG) signalling protocol (see Section 2.4.2).

Table 1-1. Performance requirements for primary user ground telephone facilities

Facility	*Establishment of communication should occur:*
Instantaneous access	Within 1 second or less for 99% of call attempts. The interval of 1 second is the delay between the A-party initiating the call and the A-party to B-party speech path being established.
Direct access	Within 2 seconds or less for 99% of call attempts. The interval of 2 seconds is the delay between the A-party initiating the call and the B-party being alerted to the presence of the call.
Indirect accesss	Within 15 seconds or less for 99% of call attempts. The interval of 15 seconds is the delay between the end of dialling by the A-party and the B-party being alerted to the presence of the call.

1.4.2 Indication of calling, called and connected party identity

1.4.2.1 Indications of calling, called and connected party identity allow the users involved in a call to receive information concerning the identity of the other users in the call. The identity of a user normally consists of that user's telephone number and, optionally, a name.

1.4.2.2 When a call is established (either directly via a direct access key or indirectly via the telephone keypad), the identity of the A-party should be indicated at the B-party VCS.

1.4.2.3 When a call is alerting, the identity of the B-party should be indicated at the A-party VCS.

1.4.2.4 When a call is answered, the identity of the B-party should be indicated at the A-party VCS.

1.4.3 Operational implementation — model for the display of caller identity

1.4.3.1 In the most basic networks (i.e. those without inter-VCS switching and signalling), the identity of the caller will be implicit through hardwiring of a particular key at the CWP to a transmission circuit (line port) of the VCS. No addressing information is passed across the line interface. If the line port detects an incoming call, the key illuminates and an audible warning is given. The key is labelled in some way to indicate the identity of the correspondent on the far end of the line. In this instance, there will be a one-to-one mapping between the VCS line ports and the operator keys.

1.4.3.2 In networks where switching and signalling are used, a VCS can use the calling and called party numbers, transmitted in the inter-VCS signalling protocol, to ring a unique direct access key at the CWP. A common implementation is for the called party number to address a particular CWP and the calling party number to address a predefined key at that CWP. Where no key has been defined at the addressed position, a call queue or indirect access indicator lamp can be used to indicate the presence of an incoming call. An alphanumeric display at the CWP can provide either the calling party number or the actual identity through use of a look-up table within the VCS.

1.4.4 Indication of urgent/priority calls

1.4.4.1 The priority facility is a means of attaching an indicator (or flag) to a telephone call to show that it is "urgent" as opposed to "routine". It is intended for use when it is necessary to make an urgent call concerning the safety of aircraft (i.e. an emergency situation) and to enable, if necessary, the interruption of less urgent calls in progress at the time. The use of priority is generally agreed by bilateral agreement between administrations. The ultimate decision and responsibility as to whether a call has priority rests with the A-party in accordance with local operational procedures. There are three ways in which priority can be set:

a) manually, before the call is made: before making the call, a priority key is operated on the VCS to set the priority of the call to "urgent". This method is used when the call is already known to be urgent;

b) during call set-up: at any time during call set-up, the operation of a priority key will change the priority of the call from “routine” to “urgent”. This method would be used as a reaction to an urgent operational situation that has arisen, including a delay in answering at the far end or on receipt of a busy tone;

c) automatic setting of priority: the priority of all calls from a particular CWP or set of keys is pre-programmed in the VCS to be “urgent”. This method can be used for operational reasons when calls made from a particular CWP or key are always to be treated as urgent. An example of this is to use the priority facility to distinguish between instantaneous access and direct access calls.

1.4.4.2 Equally, the B-party VCS should react to an incoming priority call in the following manner:

a) provide some means of indicating that a priority call has been received (e.g. special visual and/or audible indications); and

b) allow the priority call to intrude in a call already established (see Section 1.4.5), preceded by a warning indication.

1.4.4.3 If a priority call cannot proceed due to congestion (all available circuits, links or channels are busy), the priority call should interrupt an established routine call (should one exist), thus allowing the priority call to proceed. Before the established routine call is interrupted, all parties engaged in that call should receive an interrupt warning tone.

1.4.5 Conference capabilities

1.4.5.1 There are two kinds of telephone conference: add-on conference and conferences arising from call intrusion (e.g. as a result of using the priority facility). Each of these is described below. The use of conference is generally by bilateral agreement between administrations.

Add-on conference

1.4.5.2 The (telephone) add-on conference facility is a means of allowing three or more parties to speak together. When a call has already been established between two parties (A and B), either one may choose to include another party in the call. For example, if the B-party decides to set up a conference, the B-party first places the A-party on hold (if required) and then makes a new call to the C-party. The B-party can then choose to set up the conference or to speak with the A-party or C-party individually and discreetly. The B-party can withdraw from the conference, leaving the other parties in conversation. Alternatively, it may be preferred that when the B-party withdraws from the conference, calls to all the parties involved are cleared.

1.4.5.3 Conferencing between internal users (i.e. on the same VCS) and external users (i.e. via external lines) is available with most modern VCSs. Most systems allow many parties to be included in a single conference. Ultimately, however, it is the responsibility of the administration to determine the maximum number of parties that is acceptable and manageable.

Conferences resulting from call intrusion

1.4.5.4 Conferences can also arise from the use of the intrusion facility. In the ATS context, a conference can be caused by the arrival of an urgent (priority) call (see Section 1.4.4) when the B-party is already engaged (busy) with another call. Subject to the intrusion being permitted by the B-party VCS (due to authorization by the local administration), a conference will be established usually after a warning indication is provided to the parties being intruded upon.

1.4.6 Automatic recording

As with all operational telephone services, and in accordance with Annex 11, Chapter 6,[9] a VCS should provide a facility whereby all telephone conversations, together with call originating and management data, are automatically recorded.

Chapter 2

ENGINEERING REQUIREMENTS

2.1 NETWORK PLANNING, IMPLEMENTATION AND MANAGEMENT

2.1.1 General

2.1.1.1 The main task faced by planners of ATS communications systems is to implement the primary user ground telephone facilities (basic call types) and supplementary facilities required by Annex 10, Volume III,[8] in a cost-effective, reliable and flexible manner and to subsequently manage those facilities. To do this in a manner that ensures interoperability and appropriate levels of service across national/regional boundaries demands the use of common technologies and adherence to a set of general planning rules.

2.1.1.2 This chapter describes the possible technical means to implement the facilities and functions described earlier. It identifies the switching technologies, transmission systems and signalling protocols that should be used. Emphasis is given to digital technologies and the planning rules necessary to avoid degradation of speech quality, to ensure signalling interoperability and to enable optimized call routing (e.g. to avoid congested links, to reduce costs). ATS administrations should also give consideration to the management of the facilities.

2.1.2 Transmission media and technologies

2.1.2.1 Developments in electronics, optical transmission techniques, and radio communications have expanded the range of transmission media to include:

a) copper cables;

b) optical fibre;

c) satellite transmission; and

d) microwave links.

2.1.2.2 Many techniques have been developed for processing the information to be sent over these transmission media in order to maximize usage of the available bandwidth. Most techniques can be categorized into generalized classes, namely:

a) frequency division multiplexing;

b) time division multiplexing;

c) statistical multiplexing;

d) data compression techniques;

e) connection-oriented, packet-based techniques; and

f) connectionless, packet-based techniques.

2.1.2.3 In practice, most real networks employ combinations of transmission media and several processing mechanisms to create economical and efficient networks that offer high levels of network performance and good quality of service (QoS).

2.1.3 Use of switching elements and signalling protocols

2.1.3.1 To avoid having to fully interconnect each possible source of information with each possible destination, switching elements also have to be incorporated in the network design. Switching elements use either circuit-oriented switching or packet-/cell-oriented switching mechanisms, the choice being determined to a large extent by the transmission technologies employed in the network.

2.1.3.2 The use of switching elements in a network implies the capability to convey instructions (i.e. signalling) from the calling party to a switching element and between switching elements to permit calls to be routed via shared transmission resources. The capability of the signalling protocol employed influences the way in which the telephone facilities described in Chapter 1 can be provided.

2.1.4 Speech compression technologies

2.1.4.1 Speech signals contain much redundant information due to the repetitive waveforms and the periods of silence during which no data need be transmitted. To remove these inefficiencies, many algorithms (e.g. LD-CELP, conjugate structure algebraic-code-excited linear prediction (CS-ACELP), global system for mobile communications (GSM) full-rate, GSM adaptive multi-rate (AMR)) have been developed to compress speech to a much lower bit rate. In all cases, the objective has been to reduce the required bandwidth without any loss of speech quality. Many of these sophisticated algorithms introduce additional delay due to the large number of speech samples required to process the signal correctly.

2.1.4.2 As well as introducing additional delay, some of these algorithms affect the speech quality; therefore, care has to be taken over the choice of algorithm. For example, it is not recommended to choose a very low bit rate algorithm in order to maximize bandwidth usage if the result is only radio-quality speech when "toll- or phone-quality" speech is required. If disparate coding algorithms are adopted at different points within a voice network, the resulting system will be unusable as a network and only suitable for single hop connections.

2.1.5 Recommended basis of ATS ground voice networks (AGVNs)

Replacement of analogue technologies with digital technologies

2.1.5.1 Many administrations currently employ analogue technologies and signalling protocols for their AGVNs. While existing implementations of these technologies are still acceptable for use in ATS operations, the technologies are now largely obsolete. New installations should therefore be based on digital technologies.

Core technology

2.1.5.2 An AGVN will be constructed from transmission links connected by switching nodes (i.e. VCS). Each element influences the QoS provided by the network. The greater the number of conversions and the amount of processing introduced into the speech path, the higher the degradation of the original speech signal and the throughput delay. The following general guidelines are recommended for the development of a new AGVN:

a) AQVNS should use transmission links and core-switching capabilities based on 64 kbit/s channels of the digital transmission hierarchy defined in ITU-T Recommendation G.702;[30]

Note.— In some regions of the world, these channels are referred to as "DS0" (digital signal level 0) channels.

b) Speech should be encoded according to the rules for 64 kbit/s A-law or µ-law pulse-code modulation (PCM) encoding, with transcoding as appropriate at national/regional boundaries, as specified in ITU-T Recommendation G.711;[32]

c) Signalling within AGVNs should be closely based on the private signalling system number 1 (PSS1), also known as "QSIG", and defined in ISO/IEC 11572[12] and related ISO/IEC international standards. The PSS1 (QSIG) signalling system is intended for signalling between nodes of a private integrated services network (PISN), i.e. between PINXs. The way in which this signalling system should be used in AGVNs is described in detail in Section 2.4.2.

The use of alternative technologies is not precluded, provided that the required level of performance is achieved.

Use of sub-multiplexing and low bit rate speech encoding schemes

2.1.5.3 Where a bilateral agreement has been reached between the administrations, AGVNs may make use of sub-multiplexing and low bit rate speech encoding schemes in the following circumstances:

a) where no more than two "hops" (transmission links in series) are envisaged between the originating VCS and the terminating VCS, i.e. one transit VCS; or

b) where the use of a satellite is required, e.g. on a link of transoceanic or transcontinental length.

2.1.5.4 In general, a single stage of speech compression and decompression using one of today's low bit rate algorithms has little noticeable effect on the speech quality perceived by the end-user. However, where several stages of compression and decompression occur in series (as might be the case in a switched tandem connection), the perceived quality quickly degrades to an unacceptable level.

Therefore, the combined use of various techniques (transit switching, multiple stages of compression, satellite links, etc.) has to be carefully planned. Furthermore, network planners are advised to be aware of the potential future use of low bit rate vocoders in air-ground communications and the consequent impact on speech quality that could arise from the use of these in tandem with low bit rate speech encoding schemes in ground communications. Subject to these considerations, the following arrangements can be used:

a) 16 kbit/s sub-multiplexing in conjunction with LD-CELP speech encoding (ITU-T Recommendation G.728);[33] and

b) 8 kbit/s sub-multiplexing in conjunction with CS-ACELP speech encoding (ITU-T Recommendation G.729)[34] speech encoding.

2.1.5.5 These arrangements are defined in international standards ISO/IE 17310[20] and ISO/IEC 17311,[21] respectively. Compression and decompression only take place once. Transit modes complying with the requirements of these international standards are required to switch the compressed bit stream without decompressing and recompressing it. The characteristics of the two speech encoding algorithms used are shown in Table 2-1, together with those of the full 64 kbit/s PCM recommended in Section 2.1.5.2.

2.1.5.6 The influence of low bit rate encoding algorithms on speech transmission quality can be evaluated by associating each type of algorithm with a specific Ie. The influence of several low bit rate encoding algorithms in tandem is characterized by the sum of the individual impairment factors for the algorithms in the chain. Table 2-1 shows the Ie associated with different encoding algorithms.

2.1.5.7 Network planners should be aware of the impairments introduced by low bit-rate encoders or other voice signal processing and should take these into account in order that the desired level of speech quality may be met in the network.

Equipment configurations

2.1.5.8 There are many different ways in which switching equipment (VCS), multiplexing equipment, compression equipment and transmission links can be combined to construct an AGVN.

Note.— Various "scenarios" are described in detail in a technical report (TR) available from ISO/IEC, T 14475.[22] *This TR is also available from ECMA (an international, Europe-based industry association for standardizing information and communication systems) as TR/76.*[2]

2.1.5.9 In determining the configuration to use, network planners have to consider many factors, including:

a) whether the transmission links between VCSs are to be wholly dedicated to voice or whether they will be links shared with other applications (e.g. aeronautical data applications);

b) the anticipated level of traffic between VCSs and, hence, the bandwidth required of the transmission links. For instance, are several 64 kbit/s lines sufficient or is a wider bandwidth transmission structure containing many channels (e.g. E1/T1) necessary?;

c) the cost of transmission links and whether it is advantageous to use sub-multiplexing and speech compression techniques;

Table 2-1. Characteristics of speech encoding algorithms

Encoding algorithm	Rate (kbit/s)	Delay (ms) (Note 1)	Ie
PCM (G.711)	64	0.75	0
LD-CELP (G.728)	16	2	7 (Note 2)
CS-ACELP (G.729)	8	35	10
CS-ACELP (G.729 Annex A and VAD)	8	35	11

Note 1.— All delay values are one-way.
Note 2.— This Ie is the same as for 32 kbit/s adaptive differential pulse code modulation (ADPCM).

d) the need for any overflow arrangements to cope with traffic overload situations (e.g. routing via alternative transmission links, routing via PSTN); and

e) the need for redundancy in transmission links or other back-up/standby arrangements (e.g. PSTN).

2.1.5.10 Various configurations are possible and they are not necessarily mutually exclusive. More than one configuration can be used in any given network. The matter can be rendered more complex by other factors, such as:

a) whether the interface characteristics of the VCS match the interface characteristics of the transmission link; and

b) whether a sub-multiplexing and compression capability is integrated with a VCS or performed separately from it.

2.1.5.11 When the interface characteristics of the VCS match the interface characteristics of the transmission link, the VCS can be directly connected to the transmission link. This configuration is illustrated in Figure 2-1. Points A and B are referred to in Table 2-3.

2.1.5.12 When the interface characteristics of the VCS do not match the interface characteristics of the transmission link, the VCS must be indirectly connected to the transmission link via another device such as a multiplexer. This configuration is illustrated in Figure 2-2. Points A and B are referred to in Table 2-3. If a sub-multiplexing and compression capability is required, this function can either be performed as an integral function of the VCS, or it can be performed by separate equipment, such as a multiplexer. Table 2-2 shows the possible configurations based on these aspects.

2.1.5.13 When sub-multiplexing and compression are performed separately from the VCS (i.e. as part of an indirectly connected configuration), restrictions have to be applied to the network topology to prevent excessive degradation of speech quality. Table 2-3 gives some examples of these configurations and how they determine the possibilities at each of the interface points A and B in Figures 2-1 and 2-2. The table identifies the relevant ISO/IEC international standards and ITU-T recommendations associated with each configuration.

2.1.6 Planning guidelines

2.1.6.1 In addition to the general guidelines for core technology given above, a number of other planning guidelines should be applied to the design of AGVNs in order to provision networks that are reliable, of high quality and easy to maintain. These rules are listed below and elaborated on in subsequent sections.

a) The configuration of the network should be stable. That is to say, the network must be able to handle an increasing volume of traffic and changes in the route pattern without major re-engineering.

b) Networks should normally be based on the use of dedicated digital circuits. These should be commensurate in number with the operational requirements for grade of service (GoS), reliability and availability. As an alternative to the use of leased lines, the use of virtual private network (VPN) services may be considered.

c) VCSs should normally be linked in a polygonal configuration so that traffic has a choice of routes by which the destination can be reached. VCSs should have the capability for automatically routing a call via an alternative route if the primary route is congested or unavailable.

d) Special care should be taken to maintain a uniform quality level throughout the length of transmission links, with special attention being paid to inter-connections between telecommunications service providers and ATM facilities.

e) The overall propagation delay for any specific connection through the network should be kept to a minimum.

f) The through-switching (tandem) function should normally be limited such that no more than four VCSs (i.e. the originating VCS, the terminating VCS and no more than two transit VCSs) may be connected in tandem for any through-connection.

g) When planning for the use of satellite channels, care should be taken to maintain the propagation time within the limits specified for the network as a whole.

h) It should be possible to assign a specific “class of service” to each network resource (e.g. trunk lines, terminal equipment).

i) Networks should not be permitted to have a single point of failure that would impede critical ATS operations or propagate primary failures to end systems.

Table 2-2. Possible configurations based on considerations of interface matching and compression capability

Interface characteristics of the VCS	Sub-multiplexing and compression not required	Sub-multiplexing and compression required
Match interface characteristics of the transmission link	*Directly connected* No compression capability needed	*Directly connected* Compression capability integral with VCS
Do not match interface characteristics of the transmission link	*Indirectly connected via a multiplexer* No compression capability needed	*Indirectly connected via a multiplexer* Compression capability integral with multiplexer

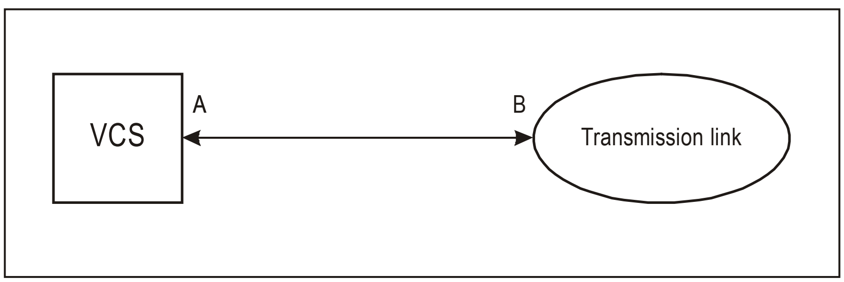

Figure 2-1. Directly connected configuration

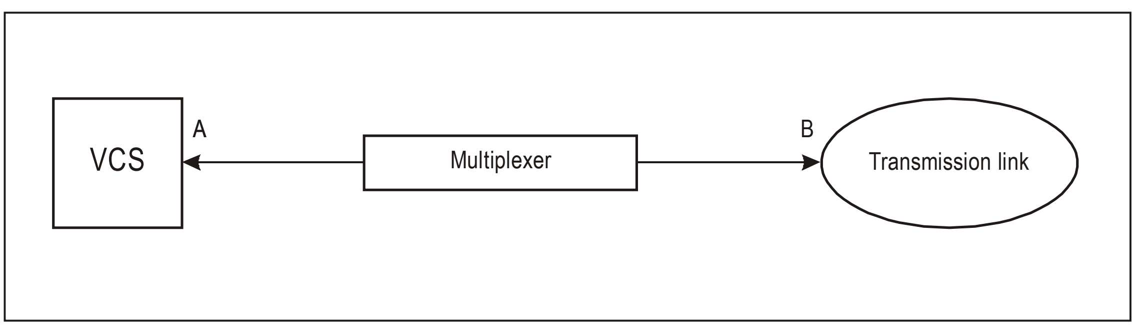

Figure 2-2. Indirectly connected configuration

Table 2-3. Example configurations

	Directly connected, no compression	Directly connected, with integral compression	Indirectly connected, no compression	Indirectly connected, with separate compression
Bit rate of voice channels				
– at point A	64 kbit/s	16 kbit/s	64 kbit/s	64 kbit/s
– at point B	64 kbit/s	16 kbit/s	64 kbit/s	16 kbit/s
Speech encoding				
– at point A	A/μ-law PCM	16 kbit/s LD-CELP	A/μ-law PCM	A/μ-law PCM
– at point B	A/μ-law PCM	16 kbit/s LD-CELP	A/μ-law PCM	16 kbit/s LD-CELP
Decompression required for tandem switching	No	No (Note 2)	No	Yes
Interface at point A	6 kbit/s G.703 or E1/T1 (Note 1)	64 kbit/s G.703	Fractional E1/T1	Fractional E1/T1
Interface at point B (type of transmission link)	64 kbit/s or E1/T1 digital leased lines (Note 1)	64 kbit/s G.703 digital leased line	E1 or T1 digital leased line (Note 3)	64 kbit/s G.703 digital leased line
Bit rate of signalling channel				
– at point A	64 kbit/s	16 kbit/s	64 kbit/s	64 kbit/s
– at point B	64 kbit/s	16 kbit/s	64 kbit/s	16 kbit/s
Relevant specifications	ITU-T Rec. G.703 ITU-T Rec. G.711 ISO/IE 11474	ITU-T Rec. G.703 ITU-T Rec. G.728 ISO/IEC 17310	ITU-T Rec. G.703 ITU-T Rec. G.711 ISO/IEC 11474	ITU-T Rec. G.703 ITU-T Rec. G.711 ITU-T Rec. G.728

Note 1.— Several possibilities are available based on the expected level of traffic. It is possible to use a single 64 kbit/s digital leased line, several 64 kbit/s digital leased lines or an E1 or T1 digital leased line.

Note 2.— No decompression is required because the compression and sub-multiplexing capability is integral to the VCS. In this case the VCS is able to determine whether or not it should perform an end VCS role or a transit VCS role for each particular call. For calls for which it acts as a transit VCS, the compressed bit stream is switched transparently from the input port to the output port without the need for decompression and recompression. The compressed bit stream is rate-adapted to, and switched at, the normal channel rate of the VCS.

Note 3.— Only a fraction of the channels available in this type of leased line are allocated for use as voice channels between VCSs. The remaining channels are allocated to other applications.

j) Network availability is dependent upon the availability values of individual circuits which, typically, could be in the range of 98 per cent to 99.5 per cent (although there may be significant variations from these values in some areas). The overall target availability of an AGVN should be 99.98 per cent or better if the prevailing circumstances permit.

k) When planning an AGVN, network planners should give consideration to how the network is to be managed on a day-to-day basis.

2.1.6.2 These guidelines are in addition to any tele-communication regulatory requirements applicable to private telephone networks in the State of operation.

2.2 QoS AND NETWORK PERFORMANCE

2.2.1 General

2.2.1.1 The QoS offered by a network is the degree of satisfaction experienced by the user of a service, brought about by the collective effect of the mechanisms employed to ensure adequate performance of that service. In any telephone network there are many performance mechanisms that have to be considered to ensure an acceptable QoS, i.e.:

a) rapid call set-up times (in order to meet the performance requirements specified for the primary user ground telephone facilities (Section 1.3.5);

b) intelligibility of speech; and

c) minimum delay in the voice path.

2.2.1.2 This section covers the following mechanisms, each of which contribute to the overall QoS:

a) traffic engineering (GoS, availability) aspects;

b) transmission planning;

c) allocation of signalling delay;

d) the use of satellite links;

e) network synchronization; and

f) alternate routing.

2.2.1.3 As a general principle, ATS networks have to be carefully planned to take account of the expected levels of telephone traffic while at the same time maintaining an acceptable QoS and minimizing network operating costs. Specifying the appropriate QoS and delivering that level of service is the responsibility of the network planner. QoS influences the route that voice traffic will take over a network. Routing strategies should be specified so as to minimize impairments to speech quality and cost. Other determinants, such as priority and security levels, should also be considered.

2.2.2 Traffic engineering

Normal methods of traffic calculation for telephone networks can be used for the design of AGVNs. It should be noted, however, that telephone calls used for the purpose of ATM are normally of a shorter duration than ordinary telephone calls. To avoid over-provisioning (and hence, excessive cost) or under-provisioning (and hence, overload) of resources (switching equipment and trans-mission links), networks are usually dimensioned to a chosen GoS. GoS is defined as the probability that a call will be lost during the busy hour due to the shortage of switching resources or transmission links. Mathematically, it is equal to the proportion of calls lost. In the case of AGVNs, the recommended GoS value is 0.001. It is recommended that traffic statistics be used regularly to ensure that this GoS be achieved.

2.2.3 Transmission planning

Recommended planning methodology

2.2.3.1 One of the most important aspects of QoS in ATS voice networks is the intelligibility of speech. It can affect the ability of ATS personnel to accurately judge the disposition (e.g. mood, level of stress) of the other party in a conversation. Intelligibility depends on speech quality which in turn is determined by bandwidth, lowest usable audio frequency, linear frequency response and other transmission characteristics.

2.2.3.2 To ensure optimum speech quality, planners must pay careful attention to the speech encoding algorithm, other transmission parameters and the values (limits) allowed for these parameters in given network configurations. A brief introduction to some of the terms is provided in this section.

2.2.3.3 Administrations are strongly recommended to use a modern transmission planning methodology such as that described in ITU-T Recommendations G.108[26] and G.109.[27] This methodology, addressing the overall transmission plan aspects for telephony in a private network, has been adopted by ITU-T as the basis for transmission planning in private networks with global scope. It was developed jointly by the European Telecommunications Standards Institute ETSI and the America-based Telecommunications Industry Association (TIA). It is also described in ETSI guide EG 201 050,[3] and TIA/EIA publication TIA/EIA-TSB-32-A.[47] This methodology uses a model (the E-model) for assessing the mouth-to-ear performance of voice telephony across networks of all types. The E-model is described in ITU-T Recommendation G.107.[25] It is also described in ETR 250.[6]

2.2.3.4 ITU-T Recommendation G.113 gives guidance on the effect of impairments on end-to-end speech quality. The Ie method allocates a value of distortion to each network element and then allows the simple addition of these impairments to determine the overall impairment introduced by all the elements of a connection. An expectation factor (*A*) is then subtracted from the overall impairment value (I_{tot}) to generate the calculated planning impairment factor (I_{cpif}). The expectation factor (*A*) represents the "access advantage", i.e. the effect on the overall transmission quality (change in I_{cpif}), as perceived by the user, caused by the ease or difficulty with which a connection can be established. For example, satellite links confer an advantage in that they allow the provision of service to a remote location, and as a result, the user may discount the speech impairments resulting from the satellite system. Typically, a satellite link will result in a value of twenty for *A*. In normal wired connections, *A* has a value of zero. ITU-T Recommendation G.113 also provides recommendations for upper limits for the total impairment value with regard to different perceived levels of speech quality, as shown in Table 2-4.

2.2.3.5 For practical planning purposes, it is recommended that normal connections within the planned network, or between the planned network and the public or other networks, should result in an upper value of fifteen for the total impairment. Planned values above this limit should be subject to careful analysis to ensure that the operational requirements will be met. For exceptional configurations (e.g. satellite links to rural areas), higher values (not exceeding 45) are acceptable.

Transmission levels

2.2.3.6 Transmission levels are not as relevant in the digital environment as they are in the analogue environment. In a wholly digital environment, signals traverse a network without any loss. However, speech is still delivered to the user in analogue format. Therefore, even in a modern digital network, transmission levels must still be considered where an analogue-to-digital or digital-to-analogue conversion occurs at the edges of the network. For an interim period, some networks may contain a mixture of digital and analogue transmission links or switches. In these cases, transmission levels must be given more careful consideration. Today, relative rather than absolute transmission levels form the basis of network transmission planning.

Delay

2.2.3.7 Each element (transmission link, processing element, switching element) involved in a networked telephone call introduces an amount of delay into the call. Such delays can be attributed to the processing and transmission (propagation) of the speech signal and to the processing and transmission (propagation) of the signalling information required to control the call's progress through the network. The former can be taken account of in transmission planning for the network (this section). The latter is not catered for by normal transmission planning rules and is dealt with in Section 2.2.4.

2.2.3.8 Speech delay manifests itself in two ways: first, in the forward path where the speech from the caller

Table 2-4. Quality levels as a function of the total impairment value (I_{cpif})

Upper limit for total impairment value (I_{cpif})	Speech quality
5	Very good
10	Good
20	Adequate
30	Limiting case

takes a significant time to reach the called party, and second, through echoed signals returning from the far end. The greater the delay, the more difficult it becomes to sustain normal conversation. ITU-T Recommendation G.114[29] gives guidance on the acceptable end-to-end, one-way transmission time delays for international telephone connections as follows:

0–150 ms = acceptable for most applications;

150–400 ms = acceptable, but with an impact on transmission quality, depending on the degree of user interactivity required in the transmission; and

>400 ms = unacceptable for general planning purposes.

2.2.3.9 As a planning value, the maximum one-way transmission time for 64 kbit/s and E1/T1 leased lines in an AGVN can be assumed to be (10 + 0.01 × G) ms, where G is distance in kilometres between the endpoints. This value is a maximum delay budget and will rarely be met in practice. It takes account of the fact that many leased lines are routed over fibre, which introduces slightly more delay than other transmission media. It is based on geographical rather than routed distance between the endpoints and includes an allowance for delay introduced by intermediate transmission and switching equipment. This allowance is sufficient to also cater for the initial delay encountered with those speech compression algorithms where a finite period is needed to gather the initial samples of the caller's speech before onward transmission commences. Delay on satellite links is covered in Section 2.2.5.

Echo loss and stability

2.2.3.10 Echo loss and stability relate to how much of the originator's speech is heard by the originator, reflected from the remote end. In other words, it is the level of the user's voice (side-tone) heard in the headset that is not locally generated (see Figure 2-3). Most voice systems provide a local side-tone to callers to provide confidence that their speech is being transmitted. This is added to the echo return signal to provide a composite side-tone.

2.2.3.11 Reflection of the signal can occur at any point in the transmission path. This can result, for example, from a four-wire to a two-wire conversion, from cross-talk in the electronics or from the remote user's microphone picking up the output of the speaker or headset. Where the returned level is particularly high, a feedback loop can be established with the resultant whistling and howling characteristic of the feedback condition. Echo loss has to be carefully controlled where there is an appreciable round trip delay in the speech path. If the echo loss is low, the round trip delay must also be low to ensure that the caller does not perceive an echo. Callers will only tolerate very low levels of echo before it starts to affect the flow of conversation. The higher the echo loss, the longer the round trip delay that can be tolerated by the user. For many long-distance links, echo cancellers have to be installed to cancel out the returned echo signal if the voice path is to be usable.

Quantization distortion

2.2.3.12 Quantization distortion results from differences between the actual signal level and the granularity of the sampling levels in the processing of analogue-to-digital and digital-to-analogue conversions. It also arises in the transcoding of a digitally encoded signal from one coding scheme to another. Various speech encoding/transcoding algorithms have differing amounts of quantization distortion, expressed in terms of quantization distortion units (QDUs), associated with them.

2.2.3.13 For a particular speech path, the quantization distortion associated with the processing elements forming the path are totalled to give a quantization distortion figure for the route. A high number of QDUs implies that the received speech will be of poor quality. However, the QDU totalling method, while providing a good approximation for

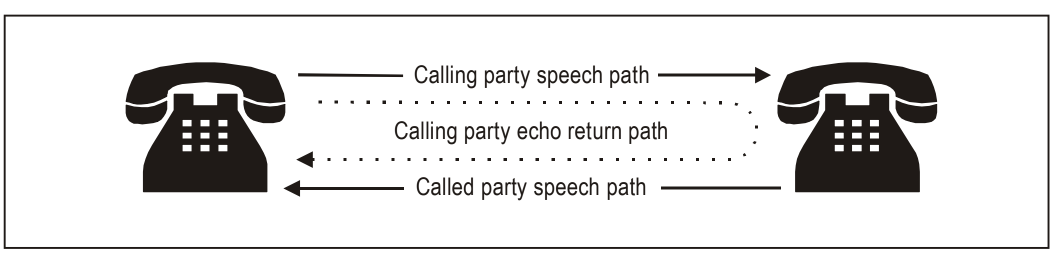

Figure 2-3. Echo return path

older processing techniques, does not work well with more modern coding algorithms (e.g. low bit rate encoding algorithms of the ITU-T G.72x series of recommendations). Such algorithms contribute distortions (resulting in a decrease of the perceived voice quality) that cannot be readily quantified using a QDU as the measure. To deal with these distortions, a new measure, the Ie, has been introduced (ref. ITU-T G.113,[28] EG 201 050,[3] TIA/EIA-TSB-32-A).[47]

2.2.4 Allocation of signalling delay

2.2.4.1 The amount of signalling delay acceptable in AGVNs is constrained by the performance requirements of the primary user ground telephone facilities (see Section 1.3.5). The objective, therefore, should be to establish each call in less than one second. This is the requirement deriving from the instantaneous access facility, as it is generally not possible to distinguish from the signalling whether a call results from the use of this facility or from the use of another facility with less stringent performance requirements.

2.2.4.2 Clearly, there will be circumstances where some calls take longer than one second to establish, and some circumstances where the call fails to be established at all. For network planning purposes, therefore, the following guidelines can be stated:

a) 99 per cent of calls during the busiest hour of a typical day should be established within one second;

b) one per cent of calls during the busiest hour of a typical day can take longer than one second to establish; and

c) of the one per cent of calls taking longer than one second to establish, one in ten can fail to be established at all.

These guidelines support both the specified instantaneous access performance requirement (see Section 1.3.5) and the recommended GoS value of 0.001 (see Section 2.2.2).

2.2.4.3 To assist with the correct provisioning of equipment to meet these signalling performance requirements, Appendix A describes a method that can be used to plan the assignment of the signalling delay budget to network elements.

2.2.5 The use of satellite links

2.2.5.1 ITU-T Recommendation G.114[29] details the various delays to be expected in the network and gives delays of 12 ms for low earth orbit (1 400 km) (LEO) satellites and 260 ms for geostationary satellites (36 000 km). These figures are for the space segment only. For planning purposes, they should be increased to 95 ms and 350 ms, respectively, to include ground station delays.

2.2.5.2 The link delay increases the call establishment delay in addition to the degradation of voice quality. For geostationary satellites, the delay may cause the call set-up time to exceed the one second maximum allowed for instantaneous access, especially if a large terrestrial segment is also involved in reaching the satellite ground station. It is clear that two or more satellite hops should be avoided in an ATS environment.

2.2.5.3 Geostationary satellite links should not be planned for normal ATS communications unless there is no economic alternative and the resulting degraded performance is judged to be acceptable in the planned context (e.g. low traffic, remote locations). This does not preclude the use of these links as emergency back-up facilities, where the degraded performance is of secondary importance. Low earth orbit satellites have delays comparable to terrestrial links; therefore, for network planning purposes, they may be treated as normal circuits.

2.2.6 Network synchronization

2.2.6.1 Proper synchronization within a network is important for the control of bit slip. Nodes in a network can derive their timing from a variety of sources:

a) another node in the network;

b) another network; or

c) an internal clock source.

2.2.6.2 As a general rule, network nodes should be synchronized to an external clock source provided via an interface to a digital network. The clock source can be derived from the AGVN or from an external source, such as the PSTN. In the event that no external clock source is available, a node may take its system timing from an internal clock source. However, network planners should be aware that use of an inaccurate clock source can lead to a high level of slips between the node and the rest of the network. In particular, clock accuracies of $\pm 1.10^{-6}$, commonly used in commercially available equipment, normally would not meet an administration's requirements. Further guidance on synchronization strategies is given in ISO/IE 11573.[13]

2.2.7 Alternate routing

Alternate routing provides a path for a call when the primary path is out of service or congested. The capability for alternate routing may or may not exist in the network, depending on its configuration. In some cases it may not be economical to provide alternate routing capabilities, or it simply may not be possible. The PSTN may be used as a means of providing an alternative routing capability (depending on the class of service required).

2.3 NUMBERING

2.3.1 General

The numbering plan is an essential element in a switched communication system. It identifies all users and provides necessary information to the switching equipment for the routing of the traffic. Numbering plans in general have to balance the desire to keep the number of digits dialled for a call to the minimum while including the possibility of expansion beyond the planned capacity without changing the basic structure of the plan.

2.3.2 Recommended numbering plan

The following characteristics are recommen-ded for the numbering plan, but it is advisable to check that there are no overriding regional plans in force:

a) numbers should consist of six digits, whereby the first two digits identify the area, the third and fourth digits, the ATS unit, and the fifth and sixth digits, the CWP or correspondent within the ATS unit. This format is illustrated in Figure 2-4;

b) up to two additional digits may be added following the sixth digit to allow a larger number of CWPs within an ATS unit to be uniquely addressed; and

c) the area identifier may be used to identify either a single country or a group of countries.

2.3.3 Future numbering principles

2.3.3.1 The recommended numbering plan is adequate to meet current and near-future network requirements. In the longer term, the development of a numbering plan allowing addressing of a greater number of users will be required. In view of the expected future requirement for additional digits, it is recommended that when new equipment is to be procured, it should be capable of accommodating these changes.

2.3.3.2 Worldwide, AGVNs operate collectively to provide communication services for ATM. They are, therefore, a federation of individual networks, usually separated along national or regional boundaries. This structure imposes several requirements on future global numbering schemes as follows:

a) a future global numbering scheme should be based on internationally approved standards for numbering and addressing. In particular, it should meet the requirements for PISN numbering plans specified in ISO/IE 11571;[11]

b) each AGVN is self-standing but has to cooperate with other AGVNs (i.e. route calls to or accept calls from another AGVN). Therefore, a coordinated numbering scheme is required. Such a numbering scheme should permit the identification and addressing of individual networks as well as the identification and addressing of entities within networks;

c) it should not be necessary to use the full global number in order to address another entity within the same network. Thus, the global numbering scheme should be hierarchical;

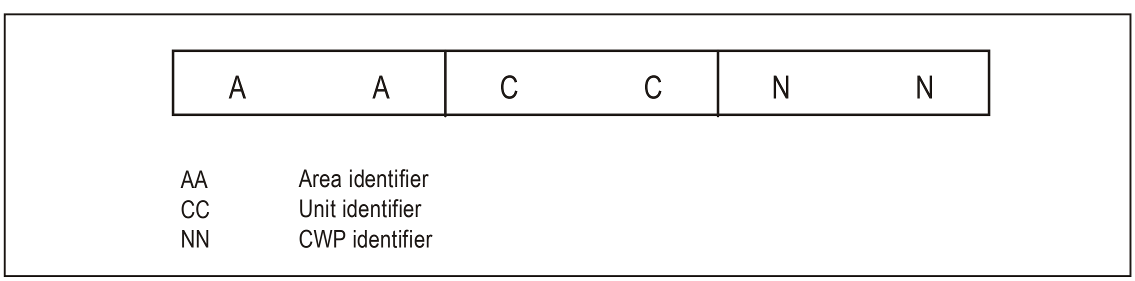

Figure 2-4. Format of numbers in the recommended numbering plan

d) each AGVN is independently administered. Therefore, a mechanism is needed to control the allocation of blocks of numbers to the administrations in order to avoid numbering conflicts. ICAO should be the responsible authority;

e) backward compatibility with the current recommended ICAO numbering plan is a desirable objective; and

f) the maximum number length in a future global numbering scheme should be fixed (i.e. closed numbering scheme) and should not exceed 15 digits (as recommended in ITU-T Recommendation E.164);[23]

g) due to the increasing growth in internetworking protocol (IP) technology, it is highly probable that a CWP will need an IP address even though a numbering plan may also be in use.

2.4 DIGITAL SIGNALLING SYSTEMS

2.4.1 General

Telecommunications networks evolve in response to the demand for new features and services. The essential components in current voice communication network evolution are digital transmission technology and the use of common channel signalling systems. Digital technology can offer many benefits including the use of digital voice compression techniques to save bandwidth and reduce the number of circuits, improved voice quality, well-standardized services, vendor independence, flexible connection and routing options, integration of voice and data, and easier transition in the future. The greatest advantage of digital technology, however, is likely to be the flexibility and range of supplementary services (SSs) that can be provided to the individual user by the use of sophisticated common channel signalling systems.

2.4.2 Recommended signalling system

2.4.2.1 The use of PSS1 is recommended in AGVNs. PSS1 (QSIG) is an internationally standardized signalling system for use in corporate voice networks. It is an ISDN-based common channel system suitable for peer-to-peer signalling. It provides the ability to connect separate communication systems together in a way that allows users to share network resources and to use the features of those communication systems (e.g. call transfer) in a network-wide manner. In particular, PSS1 (QSIG) allows products from different vendors to interoperate with one another.

2.4.2.2 In addition to the ability of the PSS1 (QSIG) to set up a basic call, an extensive range of SSs and additional network features (ANFs) specify how telephony features, such as call transfer and call forwarding, operate in a networked multi-vendor environment. This range of capabilities is enabled by the generic functional protocol (GFP), a mechanism that supports the addition of new functionality (either standardized or proprietary) in a straightforward and backward compatible manner. PSS1 (QSIG) is defined by a number of international standards published by ISO/IEC.

Note.— Standards for PSS1 (QSIG) have been mainly developed by ECMA, an international, Europe-based industry association for standardizing information and communication systems, and are available as ECMA publications. Subsequently, they were adopted as a series of international standards by ISO/IEC and as a series of ENs by ETSI. Appendix B contains a table of equivalence for the publications available from the three organizations.

2.4.2.3 The manner in which PSS1 (QSIG) should be used in AGVNs to provide the facilities identified in Chapter 1 is described in Section 2.4.3.

2.4.3 Use of PSS1 (QSIG) in AGVNs

PSS1 (QSIG) capabilities required to support the facilities defined in Chapter 1

2.4.3.1 Provision of the facilities described in Chapter 1 can be achieved by supporting a subset of the complete PSS1 (QSIG) signalling system in conjunction with one or more of the configurations described in Section 2.1.5.8. Tables 2-5 and 2-6 relate the facilities described in Chapter 1 with the capabilities of PSS1 (QSIG) that have to be supported.

2.4.3.2 The use of additional capabilities of PSS1 (QSIG) over and above those identified in Tables 2-5 and 2-6 is not precluded.

Implementation of PSS1 (QSIG) in AGVNs

2.4.3.3 To implement PSS1 (QSIG) in AGVNs, the following capabilities are required:

Table 2-5. PSS1 (QSIG) capabilities required to support primary user ground telephone facilities

Primary user ground telephone facilities	PSS1 (QSIG) capabilities required
Instantaneous access (see Section 1.3.2)	Basic call, transit counter
Direct access (see Section 1.3.3)	
Indirect access (see Section 1.3.4)	

Table 2-6. PSS1 (QSIG) SSs required to support supplementary user ground telephone facilities

Supplementary user ground telephone facilities	PSS1 (QSIG) SSs required (Note 1)
Indication of calling, called and connected party identity (see Section 1.4.2)	Number identification, name identification
Indication of urgent/priority calls (see Section 1.4.4)	Call priority interruption and protection, call intrusion
Add-on conference (see 1.4.5.2 and 1.4.5.3)	Basic call with appropriate VCS functionality
Conferences arising from call intrusion (see 1.4.5.4)	Call intrusion
Automatic recording (see Section 1.4.6)	(Note 2)

Note 1.— Support of basic call and the GFP is required for the use of PSS1 (QSIG) SSs.
Note 2.— Support of this facility is unrelated to PSS1 (QSIG) capabilities.

a) A-law/μ-law PCM speech encoding in accordance with ITU-T Recommendation G.711;[32]

b) E1/T1 interface structures in accordance with ITU-T Recommendation G.703;[31]

c) static circuit-mode inter-PINX connections in accordance with ISO/IE 14474;[16]

d) symmetrical LAPD procedures in the data link layer in accordance with Amendment 1 to ITU-T Recommendation Q.921;[46]

e) PSS1 (QSIG) basic call procedures in accordance with ISO/IE 11572;[12]

Note.— Basic call procedures include procedures for number identification.

f) PSS1 (QSIG) GFP in accordance with ISO/IE 11582[14] (excluding procedures for connectionless application protocol data unit (APDU) transport, association control service element (ACSE) protocol handling and DSE protocol handling);

g) PSS1 (QSIG) transit counter ANF in accordance with ISO/IEC 15056;[18]

h) PSS1 (QSIG) call priority interruption and call priority interruption protection SSs in accordance with ISO/IEC 15992;[19] and

i) PSS1 (QSIG) call intrusion SS in accordance with ISO/IEC 14846.[17]

Regional variants of PSS1 for ATS use

2.4.3.4 Some regional specifications for the use of PSS1 (QSIG) in ATS voice ground networks exist, in particular:

a) profile standard EN 301 846[4] (also published by ECMA as ECMA-312),[1] which has been adopted by EUROCONTROL for use in the ECAC area; and

b) regional ICD[10] for ATS speech digital signalling system, adopted for use in the Asia/Pacific Region.

Note.— Both of the above specifications make use of sub-multiplexing and voice compression techniques.

2.5 ANALOGUE SIGNALLING SYSTEMS

Signalling systems No. 5 and R2 are defined in the ITU-T Recommendations of the Q.1xx series[36,37,38,39] and of the Q.4xx series,[40,41,42,43] respectively. There are still many instances of installations that use analogue signalling systems. In particular, a special variant of signalling system R2[48] is in widespread use in some AGVNs. Signalling system No. 5 is also used but to a lesser extent. It is expected that existing installations of both signalling systems No. 5 and R2 will continue to be used for an extended period of time, at least until they have been completely replaced by digital signalling installations. However, the deployment of new installations of either No. 5 or R2 or other analogue technologies (e.g. SS1, SS3) is no longer recommended.

2.6 SUPERVISORY TONES

The characteristics of supervisory tones (ringing, busy, number unobtainable, etc.) should conform to the requirements of ITU-T Recommendation E.180/Q.35,[24] except where indicated in Table 2-7.

Note.— In the case where the tones are provided locally by the end system, administrations have the flexibility of changing them without affecting other users of the network.

2.7 NETWORK MANAGEMENT

2.7.1 Functional areas

2.7.1.1 Network management is classified into five functional areas briefly described in 2.7.1.2 to 2.7.1.6.

Fault management

2.7.1.2 Fault management is the detection, isolation and correction of the abnormal operation of the network and its environment. Typically, it is concerned with failure monitoring (alarm surveillance), fault reporting, fault isolation and service restoration.

Configuration management

2.7.1.3 Configuration management consists of the functions that are necessary to exercise control over, identify, collect data from and provide data to equipment in the network for the purpose of controlling the arrangement of network resources. Typically, configuration management involves provisioning, installation, commissioning, programming of network specific data, identification of parts, reporting and record-keeping.

Accounting

2.7.1.4 Accounting is the function necessary for tracking, allocating and controlling the costs of using network resources. Typically, it is concerned with call logging and call accounting.

Performance management

2.7.1.5 Performance management is the measurement, reporting and evaluation of equipment and network behaviour to permit assessment of the long-term operational effectiveness of the network (QoS) and to aid future network planning. Typically, it is concerned with performance monitoring (data collection), traffic measurement, monitoring of resource status and QoS monitoring and control. Performance management also includes planning, procedures and mechanisms for network back-up and disaster recovery.

Security management

2.7.1.6 Security management is the function necessary to ensure adequate security of the installation and the prevention of both malicious and inadvertent damage that compromises the operational integrity of the network. Typically, it is concerned with physical security, system security and network security. Security is treated separately in Section 2-8.

2.7.2 Network management standards

There are currently no network management standards that are explicitly accepted by the telecommunications industry for the management of private telephone networks. Although some standard protocols are used, almost all private telephone network management is performed in a proprietary manner with proprietary products.

Table 2-7. Supervisory tones

Tone	Purpose	Frequency (Hz)	Period
Dial	Returned to a user attempting to call without using a direct access key when that user indicates to the system readiness to dial (e.g. pressing a dedicated key to select a line or placing the telephone set off-hook).	425	Continuous tone
Ringing/ringback	Returned to the A-party after successful call establishment and prior to call acceptance.	425	(1 s on, 4 s off), repeated
Terminal busy/engaged	Returned to the A-party if all available signalling means to the B-party are occupied.	425	(0.5 s on, 0.5 s off), repeated
Congestion	Returned to the A-party if a call cannot be completed to the required B-party due to all appropriate network links being occupied or otherwise unavailable.	425	(0.5 s on, 0.5 s off), repeated
Number unobtainable/ reorder (Note 1)	Returned to the A-party if a terminal is "out of service" or the B-party address is unassigned.	1 000	(0.5 s on, 0.5 s off), repeated
Interrupt warning (Note 1)	Injected into the voice path to warn a party of the imminent priority interruption of an established call.	1 000	(40 ms on, 0.5 s off) repeated for approximately 5–15 s, until disconnection
Forced conference warning (Note 1)	Injected into the voice path to warn a party of the imminent priority conferencing of an established call.	1 000	(1 s on)

Note 1.— Not specified in ITU-T Recommendation E.180.

Note 2.— The recommendation for congestion tone is the same as for busy tone. If users wish to distinguish between the two tones, a dual frequency congestion tone can be used.

2.8 SECURITY ASPECTS

2.8.1 General

2.8.1.1 The emplacement of adequate security measures is an obvious duty of administrations, and as international networks evolve toward an open digital architecture, administrations will need to establish and implement a robust security policy. Safeguards are necessary to protect their own ATM as well as other (external) users connected to the network. This section relates to the security of all systems and services that contribute to the provision of ground telephone services, although there will be commonality with the security of other services.

2.8.1.2 Security can be considered from the perspective of three general categories: physical security, system security and network security. They are discussed in Sections 2.8.2, 2.8.3 and 2.8.4.

2.8.2 Physical security

2.8.2.1 Appropriate physical security measures, similar to those usually employed throughout facilities associated with the management of air traffic, should be taken to protect the areas and systems used to provide communication services.

2.8.2.2 Guidance on safeguarding against physical risks is outside the scope of this manual, but in protecting communication services, consideration also needs to be given where there is a dependency on the services of third parties, such as telecommunications service providers. It is unlikely that the safeguarding of these can be assured; therefore, independent contingency measures need to be considered and put in place as required. The scenario that should be considered is a total and sustained loss of one or all third-party services.

2.8.2.3 All areas of the communications system (including VCS, transmission links, PSTN links and the related power supply, network management and maintenance centres) should be considered as being vulnerable to physical tampering, including an electrical or electronic attack. Where it is possible for an administration to do so, the following is a list of measures that can be taken to reduce the effect of physical damage:

a) duplicate equipment;

b) physically split systems; and

c) separate environments/locations for equipment with independent power and other services.

2.8.2.4 Of particular importance are those areas where the external communications services are provided, including patch bays, line transmission equipment and external cable ducts. In cases where there is no alternative to the use of a single telecommunications service provider, consideration should be given to connecting to completely separate network access points via completely separate physical routes. The ideal is to mitigate against the effects of all common points of potential failure.

2.8.2.5 The provision of standby links is a common practice to reduce the effects of physical damage and other causes of loss of service. From the point of view of achieving an improved overall availability, however, standby links must be provided with as much physical separation and independence from the main links as is practicable.

2.8.2.6 Consideration of the consequences of loss of service due to physical effects should also be extended to the operational level and planned in conjunction with the users. It might be possible for one user or working position to provide contingency services for another user or working position, but this could be severely impaired if the contingency was due to the loss of a link they both shared.

2.8.3 System security

2.8.3.1 Within this category, all items of communications equipment within a particular operational location are included, such as VCS and line transmission equipment. System security includes ensuring that management and maintenance interfaces are not vulnerable to unauthorized use. VCSs can be attacked through their maintenance and network management interfaces as well as through interfaces to the outside world. Such attacks may manifest themselves as blockage of port accessibility, unusual traffic patterns or modification/corruption of configuration files.

2.8.3.2 Where many items of equipment are physically interconnected, steps need to be taken to prevent either deliberate or inadvertent unauthorized access to operational VCSs and external links. VCSs should have very tightly controlled call-barring and class-of-service mechanisms to prevent such occurrences while still permitting flexibility in their use. The use of personal identification number (PIN) techniques to restrict access to operational services is not usually considered to be adequate.

2.8.3.3 VCSs should, ideally, provide support for identification and authentication of authorized users, in particular, management and maintenance personnel. Mechanisms to prevent misuse of the trunk network by unauthorized users (i.e. call barring) should also be provided.

2.8.4 Network security

2.8.4.1 AGVNs, each under different administrative control, can be interconnected at national or regional boundaries. These networks will make use of facilities leased/rented from telecommunications service providers. They may also have interfaces to the PSTN to permit calls to be made to destinations outside the network to provide link back-up or simply as an alternative to the leased-line scenario.

2.8.4.2 Attacks on the network could come from within the network itself, via the infrastructure rented from telecommunications service providers or via interfaces to the PSTN. Such attacks can be physical, logical or both and could affect the availability, integrity or confidentiality of voice transport.

2.8.4.3 Where a VCS provides external (public) access, this functionality must be restricted to ensure that external callers cannot gain access to trunk lines used for operational ATM. Where a VCS does not support all of the functionality listed above, consideration must be given to an alternative means of providing the same level of protection.

2.8.4.4 AGVNs should provide accountability, access control and assurance features through the use of mechanisms such as authentication of authorized users and the maintenance of an audit log of all actions.

2.9 TRANSITION ARRANGEMENTS

2.9.1 The degree of difficulty experienced in the transition of the analogue network to the digital network will depend on the size and complexity of the analogue network being replaced. If the existing network is based on an analogue VCS and dedicated point-to-point circuits with no routing capability, then the digital VCS may be introduced with temporary analogue interfaces that are converted to digital when enough digital VCSs are commissioned. The routing functionality can be utilized for parts of the network as the number of digital VCSs grow, thus benefiting from the reductions in network costs. In cases where analogue VCSs are providing routing functionality, the transition is more complex. The options for transition include:

a) modifying the external interfaces of the old switches to coexist with digital VCSs, which may or may not be possible or economically feasible;

b) using the new digital VCS with analogue interfaces to emulate the analogue-routing functionality until enough digital VCSs are commissioned to start transitioning to digital trunks and routing functionality; and

c) reverting to dedicated point-to-point circuits during transition until enough digital VCSs are commissioned.

2.9.2 Because of the variety of different analogue VCSs and routing capabilities in existence, careful operational and economic analysis will be required to determine the best transition strategy, and a high degree of coordination will be required with other administrations for international links.

Appendix A

METHOD FOR ASSIGNING SIGNALLING DELAY TO NETWORK ELEMENTS

A.1 GENERAL

This appendix describes a method that can be used to plan the assignment of a signalling delay budget to elements of an AGVN in order to achieve the correct provisioning of equipment to meet the signalling performance requirements. It also describes a general model for apportioning call establishment and clearing delays to a call through a telephone network and identifies potential sources of signalling delay. It then shows how this general model can be refined by integrating the signalling for a particular telephone facility (in this example, the direct access facility) to deliver a specific delay model. Network planners can develop similar models for real network configurations.

A.2 GENERAL MODEL FOR A BASIC TELEPHONE CALL

Functional elements of a basic telephone call
(see Figure A-1)

A.2.1 The call control agent (CCA) functional entity acts on behalf of the user and calls upon the call control (CC) functional entities for the provision of the service requested by the user. CCAs are located within the CWP involved in the basic call, i.e. the CWP of the user who requests the service (the A-party) and the CWP of the destination user (the B-party). Specific forms of CCA exist for the A-party and the B-party.

A.2.2 The CC functional entities cooperate to provide the service requested by a CCA. Specific forms of CC are located at the network nodes (typically, VCSs) through which the call is routed, including the node serving the A-party (originating CC), the node serving the B-party (terminating CC) and any intermediate nodes (transit CC). With specialization as described, the model becomes similar to that shown in Figure A-2.

A.2.3 A notional transmission system (indicated by the horizontal lines) is considered to appear between each pair of functional entities. Such a transmission system may or may not actually exist, depending on the realization of the model in the physical world. The model is independent of the network or networks involved in the call, i.e. it applies equally to situations where interworking between networks occurs. In this case there are other specific forms of CC (incoming gateway CC, outgoing gateway CC) located at the interworking point.

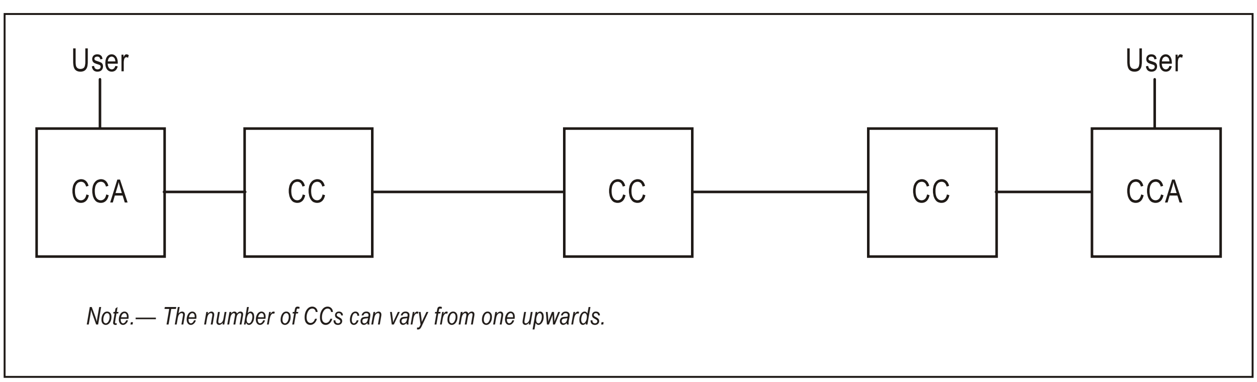

Figure A-1. Functional entities involved in a basic call

Use of a functional model for apportioning call establishment/clearing delays

A.2.4 Each component of the functional model introduces an element of delay into the processing of a call. Delays due to the processing and transmission of signalling information are of particular interest. Elements of delay can be apportioned to each component of the functional model, as shown in Figure A-3. The definition of each delay element is as follows:

T_{ccaA} In the case of call establishment, the delay caused by the time taken by CC “A” to process the A-party’s request for a service. In the case of call clearing, the delay caused by the time taken to process the A-party’s request to clear a call.

T_{localA}, T_{localB} The delay for the transmission of signalling information between a CCA and its serving CC.

T_{ccORIG} In the case of call establishment, the delay caused by the time taken by the originating CC to process the A-party’s request for a service. In the case of call clearing, the delay caused by the time taken to process the A-party’s request to clear a call.

T_{txnY} The delay for the transmission of signalling information between two CCs. Where a connection involves one or more transit CCs, there will be multiple transmission segments represented by T_{txnY} where Y = 1 to N + 1 (N = number of transit CCs). The delay imposed by all connections (sum of all T_{txnY}) is represented by T_{txnTOT}.

$T_{ccTRANSIT}$ In the case of call establishment, the delay caused by the time taken by a transit CC to process a transit call (i.e. an incoming call and onward-routed outgoing call). In the case of call clearing, the delay caused by the

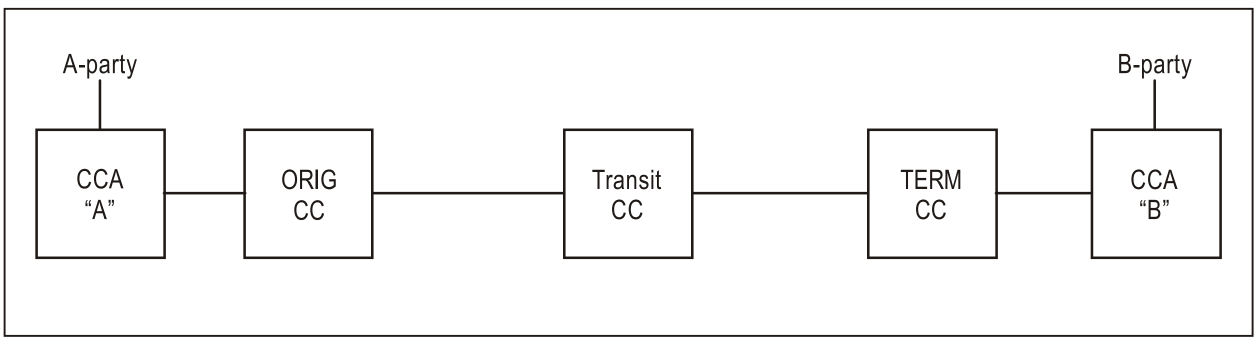

Figure A-2. Functional model for a basic call

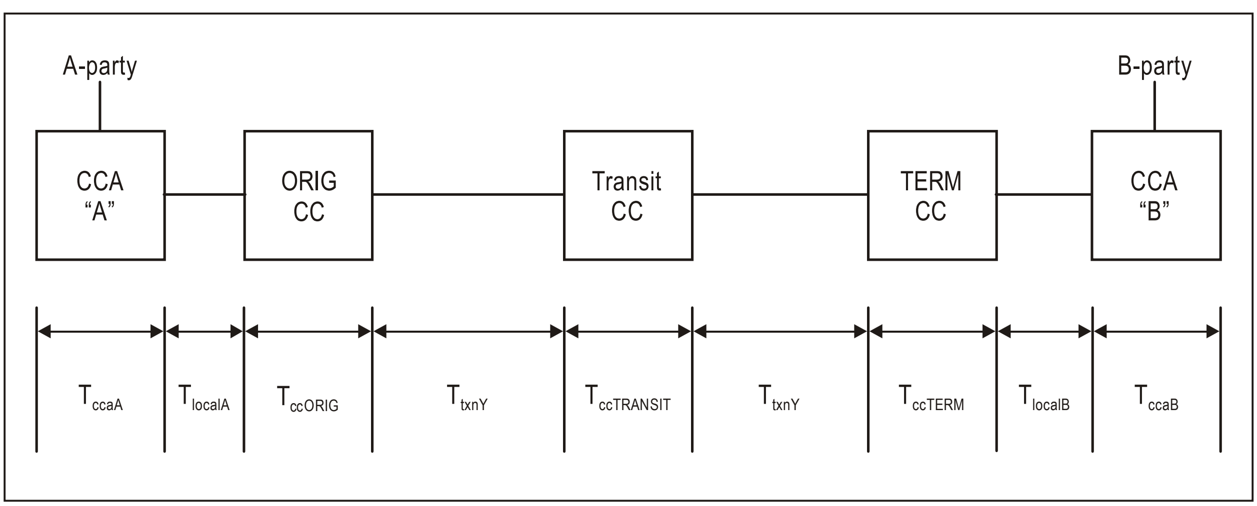

Figure A-3. Apportionment of delay elements

time taken to process a request to clear a transit call. Where a connection involves more than one transit CC, the delay imposed by all transit CCs is represented by $N \times T_{ccTRANSIT}$.

T_{ccTERM} In the case of call establishment, the delay caused by the time taken by the terminating CC to present the incoming call to the B-party. In the case of call clearing, the delay caused by the time taken to process the A-party's request to clear a call.

T_{ccaB} In the case of call establishment, the delay caused by the time taken by CC "B" to present the incoming call to the B-party. In the case of call clearing, the delay caused by the time taken by CCA "B" to process the A-party's request to clear a call.

A.3 SOURCES OF SIGNALLING DELAY

Signalling delay arising in CCAs and CCs

A.3.1 Delays arising in CCAs and CCs can only be calculated accurately for specific signalling systems implemented on specific items of equipment (e.g. CWPs, VCSs) from particular manufacturers. It is difficult to calculate generally applicable values because much depends on the overall system architecture, the microprocessors used and the efficiency of the software design. For network planning purposes, therefore, typical values that are maxima that must not be exceeded have to be chosen.

A.3.2 ITU-T Recommendation Q.543[44] deals with performance design objectives for digital exchanges and is intended to guide the design of public exchange equipment. Table A-1, which is based on this ITU-T Recommendation, summarizes the basic design objectives for public exchange equipment in integrated services digital networks (ISDN).

A.3.3 As can be seen from the table, the design objectives for public exchanges are quite relaxed. Typical modern customer premises equipment (CPE) has little difficulty in meeting or exceeding objectives twice as stringent as those shown. By selecting and manipulating the appropriate delay elements (T_{ccaA}, T_{ccORIG}, $T_{ccTRANSIT}$, etc.) in the delay model, performance parameters similar to those in Table A-1 can be determined for connections in AGVNs.

Signalling delay resulting from the transmission system

A.3.4 Similarly, it is difficult to assign accurate one-way delay values for the T_{txnY} parameters. This is due to the general unavailability of accurate information for specific connection segments. Network planners are thus forced to use values that apply to the general case and, as such, include conservative allowances. Although primarily intended for calculating delay in speech and data transmission, the values specified in Section 2.2.3.7 can also be applied to the transmission of signalling information.

A.3.5 As already noted, ITU-T Recommendation Q.543[44] gives maximum values for processing signalling information in digital exchanges. Based on this, it is reasonable to expect that intermediate signalling equipment in a transmission link might introduce some additional delay. However, in the context of leased lines, there is little dynamic switching overhead associated with the line itself and consequently, the effect of signalling processing delay is minimal. No additional allowance needs to be made.

Table A-1. Q.543 performance design objectives for public exchanges

Performance parameter	Design objective
User signalling acknowledgement delay	$\leq$400 ms
Signalling transfer delay	$\leq$200 ms
Call set-up delay (overlap sending)	$\leq$400 ms
(en-bloc sending)	$\leq$600 ms
Through-connection delay	$\leq$250 ms
Incoming call indication sending delay (overlap receiving)	$\leq$400 ms
(en-bloc receiving)	$\leq$600 ms
Connection release delay	$\leq$250 ms

A.4 INTEGRATION OF SIGNALLING TO GIVE A SPECIFIC DELAY MODEL

Refinement of the model to produce a specific delay model

A.4.1 To be of practical value, it is necessary to refine the general model described above by:

a) mapping the signalling for a particular telephone facility to deliver a specific delay model;

b) identifying the specific call-processing tasks to be performed at each stage of signalling in the specific delay model; and

c) attaching a delay element for each identified task.

A.4.2 By mapping appropriate signalling exchanges to the delay model, planning values can be calculated for each specific ground-user telephone facility. This can be done for a particular network topology, thus allowing network planners to optimize a network design to avoid excessive delay.

Example delay model for the direct access facility

A.4.3 Figure A-4 shows a typical signalling model for the direct access facility. The three-digit numbers represent specific processing tasks (and hence, delay) to be performed by CCAs and CCs at each stage of signalling. These delays are defined in Table A-2.

Note.— *Since a CCA and its serving CC will almost always be co-located (i.e. usually in the same building), the delay between them can effectively be ignored.*

A.4.4 Total call set-up delay, the parameter underpinning the performance requirement for direct access connections can be obtained by summing delays as follows:

Call set-up delay =

$$\underset{000}{T_{ccaA}} + \underset{000}{T_{ccORIG}} + (N^*T_{ccTRANSIT}) + T_{txnTOT} + \underset{300}{T_{ccTERM}} + \underset{400}{T_{ccaB}}$$

where N is the number of transit nodes in the connection and T_{txnTOT} is the total delay arising from the transmission systems, expressed as

$$\sum_{Y=1}^{Y=N+1} TtxnY$$

where T_{txnY} is the delay between a pair of CCs, calculated as (10 + 0.01 × G) ms, and G is the geographical distance in kilometres.

Note.— A model can also be constructed for the instantaneous access facility. However, since the required signalling is similar, it does not differ significantly from the model for direct access.

Allocation of values

A.4.5 Allocating the previously given values to the derived formulae shows that there is ample scope for network planners to specify connection topologies that fall well within the performance requirements of both the direct access and instantaneous access facilities. Taking the case of a direct access connection involving a single transit node with distances of 1 500 km and 4 100 km between the adjacent nodes would give an allowance as follows:

$$100 + 300 + (1 \times 300) + (10 + 0.01 \times 1\,500)$$
$$+ (10 + 0.01 \times 4\,100) + 300 + 100 = 1\,176 \text{ ms}$$

This value is well within the two-second performance required for direct access.

A.4.6 At first sight it appears that an instantaneous access connection involving a single transit node could not meet the performance requirement (one second). However, as previously stated, available modern equipment generally has no difficulty in exceeding the allowances by a wide margin. Moreover, using relatively old fashioned microprocessors, actual values for T_{ccORIG}, etc., are typically nearer to 150 ms than to 300 ms, and values for T_{ccaA}, etc., should be nearer to 20–30 ms than to 100 ms. Thus, the allowance of 1 176 ms drops to 726 ms or less.

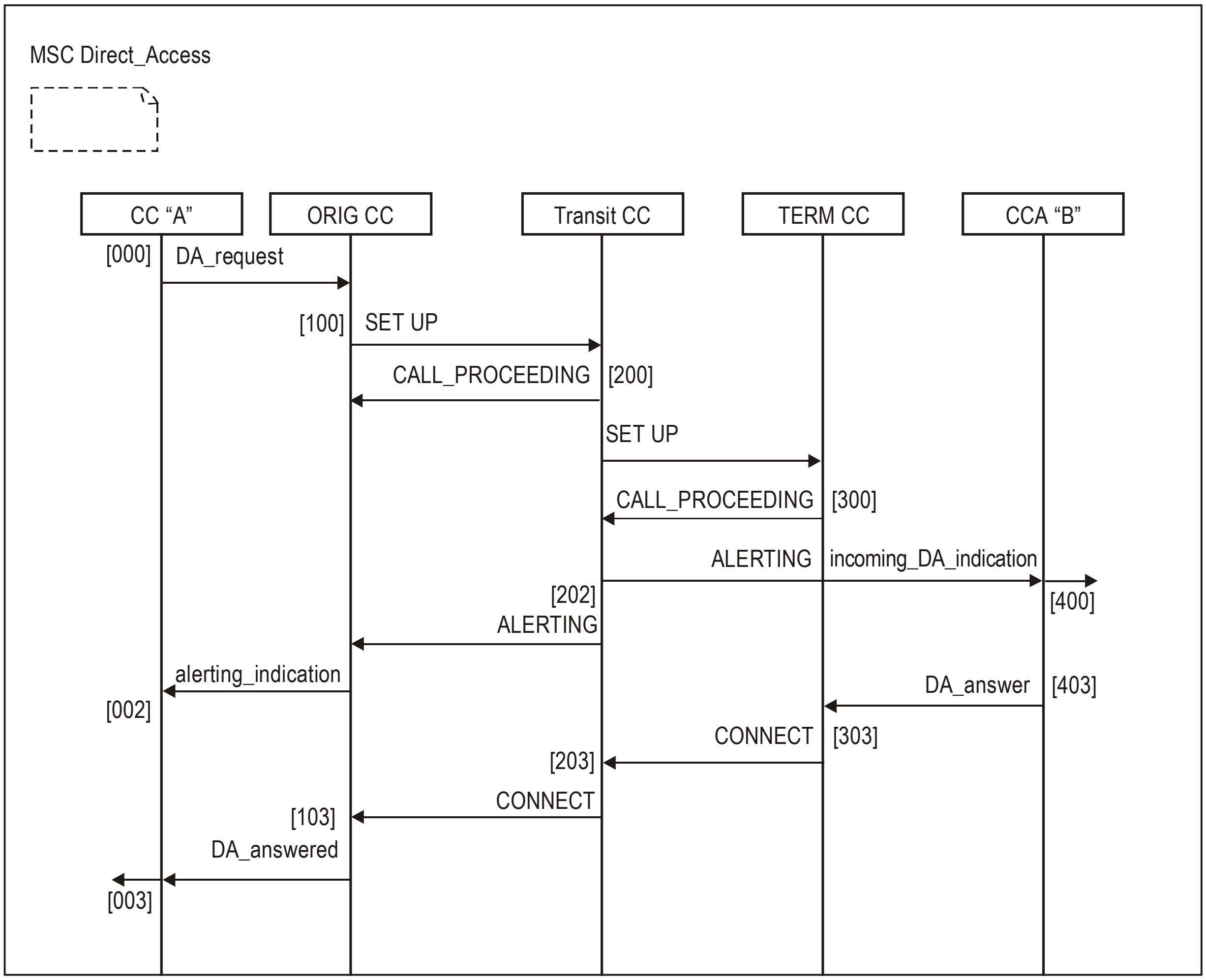

Figure A-4. Signalling model for the direct access facility

Table A-2. Call-processing tasks and delay elements for the direct access facility

Call-processing task definition	Delay element (Note 1)
Between A-party requesting a service and CCA "A" sending a DA_request to the originating CC.	T_{ccaA} 000
Between CCA "A" receiving an indication that the B-party is alerting (alerting_indication) and the A-party being given an indication of that.	T_{ccaA} 002
Between CCA "A" receiving an indication that the B-party has answered (DA_answered) and the A-party being given an indication of that.	T_{ccaA} 003
Between an originating CC receiving a DA_request from CCA "A" and sending SET UP to the next CC.	T_{ccORIG} 100 (Note 2)
Between an originating CC receiving an indication that the B-party is alerting (ALERTING) and passing that indication on to CCA "A" (alerting_indication).	T_{ccORIG} 102
Between an originating CC receiving an indication that the B-party has answered (CONNECT) and passing that indication on to CCA "A" (DA_answered).	T_{ccORIG} 103
Between a transit CC receiving SET UP from the preceding CC and sending SET UP to the subsequent CC.	$T_{ccTRANSIT}$ 200 (Note 2)
Between a transit CC receiving an indication from the preceding CC that the B-party is alerting (ALERTING) and passing that indication on to the subsequent CC.	$T_{ccTRANSIT}$ 202
Between a transit CC receiving an indication from the preceding CC that the B-party has answered (CONNECT) and passing that indication on to the subsequent CC.	$T_{ccTRANSIT}$ 203
Between a terminating CC receiving SET UP and sending an incoming_DA_ indication to CCA "B".	T_{ccTERM} 300 (Note 3)
Between a terminating CC receiving an indication (DA_answer) from CCA "B" that the B-party has answered and sending a CONNECT message to the next CC.	T_{ccTERM} 303
Between CCA "B" receiving an indication of an incoming call (incoming_DA_ indication) and the B-party being given an indication of that call.	TccaB 400
Between the B-party answering and CCA "B" giving an indication (DA_answer) of that to the terminating CC.	T_{ccaB} 403

Note 1.— The signalling transfer delay imposed by a node (equivalent to the signalling transfer delay parameter shown in Table A-1) is the average of all the delays for that node.

Note 2.— Equivalent to the call set-up delay parameter shown in Table A-1.

Note 3.— Equivalent to the incoming call indication sending delay parameter shown in Table A-1.

Appendix B

REFERENCED PUBLICATIONS AND INFORMATION SOURCES

B.1 GENERAL

The manual incorporates, by reference, provisions from other publications. These references are cited at the appropriate places in the text and the publications are listed in Section B.2, together with other publications that may be useful for gaining a fuller understanding of the subject matter of this manual. Section B.3 provides a table of equivalence for PSS1 specifications generated by different standardization bodies. Web site addresses of standardization bodies are provided in Section B.4.

B.2 REFERENCES

For dated references, subsequent amendments to any of these publications apply to this manual only when incorporated by amendment. For undated references, the latest edition of the publication referred to applies.

1. ECMA-312 (2000): "Private Integrated Services Network (PISN) — Profile Standard for the Use of PSS1 (QSIG) in Air Traffic Services Networks" (also published by ETSI as EN 301 846).

2. ECMA TR/76 (1999): "Private Integrated Services Network — Architecture and Scenarios for Private Integrated Services Networking".

   Note.— The content of ECM TR/76 is fully aligned with ISO/IEC TR 14475, Second Edition.

3. EG 201 050 (1999): "Speech Processing, Transmission and Quality Aspects (STQ); Overall Transmission Plan Aspects for Telephony in a Private Network", V1.2.2.

4. EN 301 846 (2001): "Private Integrated Services Network (PISN) — Profile Standard for the Use of PSS1 (QSIG) in Air Traffic Services Networks", (also published by ECMA as ECMA-312).

5. ES 201 168 (1998): "Corporate Networks (CN); Transmission characteristics of digital Private Branch eXchanges (PBXs)".

6. ETR 250 (1996): "Transmission and Multiplexing (TM); Speech communication quality from mouth to ear for 3,1 kHz handset telephony across networks".

7. EUROCONTROL ASM.ET1.ST18.1000-REP-01.00 (1998): "Guidelines for the Application of the ECAC Radar Separation Minima", Edition 2.0.

8. ICAO Annex 10, Volume III, Part II, Chapter 4, "Aeronautical Speech Circuits".

9. ICAO Annex 11, Chapter 6, "Air Traffic Services Requirements for Communications".

10. ICAO State letter T8/3.1, dated 13 May 1998, on the "Asia/Pacific Regional Interface Control Document (ICD) for Signalling System for Digital ATS Speed Circuit Network".

11. ISO/IEC 11571 (1998): "Information technology, Telecommunications and information exchange between systems — Private Integrated Services Networks — Addressing".

12. ISO/IEC 11572: (2000) "Information technology — Telecommunications and information exchange between systems — Private Integrated Services Network (PISN) — Circuit mode bearer services — Inter-exchange signalling procedures and protocol".

13. ISO/IEC 11573 (1994): "Information technology — Telecommunications and information exchange between systems — Synchronization methods and technical requirements for Private Integrated Services Networks".

14. ISO/IEC 11582 (1995): "Information technology — Telecommunications and information exchange between systems — Private Integrated Services Network — Generic functional protocol for the support of supplementary services — Inter-exchange signalling procedures and protocol".

15. ISO/IEC 13868 (1995): "Information technology — Telecommunications and information exchange between systems — Private Integrated Services Network — Inter-exchange signalling protocol — Name identification supplementary services".

16. ISO/IEC 14474 (1998): "Information technology — Telecommunications and information exchange between systems — Private Integrated Services Network — Functional requirements for static circuit-mode inter-PINX connections".

17. ISO/IEC 14846 (1996): "Information technology — Telecommunications and information exchange between systems — Private Integrated Services Network — Inter-exchange signalling protocol — Call intrusion supplementary service".

18. ISO/IEC 15056 (1997): "Information technology — Telecommunications and information exchange between systems — Private Integrated Services Network — Inter-exchange signalling protocol — Transit counter additional network feature".

19. ISO/IEC 15992 (1998): "Information technology — Telecommunications and information exchange between systems — Private Integrated Services Network — Inter-exchange signalling protocol — Call priority interruption and call priority interruption protection supplementary services".

20. ISO/IEC 17310 (2000): "Information technology — Telecommunications and information exchange between systems — Private Integrated Services Network — Mapping functions for the employment of 64 kbit/s circuit mode connections with 16 kbit/s sub-multiplexing".

21. ISO/IEC 17311 (2000): "Information technology — Telecommunications and information exchange between systems — Private Integrated Services Network — Mapping functions for the employment of 64 kbit/s circuit mode connections with 8 kbit/s sub-multiplexing".

22. ISO/IEC TR 14475 (2001): "Information technology — Telecommunications and information exchange between systems — Private Integrated Services Network — Architecture and Scenarios for Private Integrated Services Networking", Second Edition.

Note.— The content of ISO/IEC TR 14475, Second Edition, is also published by ECM as ECMA TR/76.

23. ITU-T Recommendation E.164 (1997): "The international public telecommunication numbering plan".

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

25. ITU-T Recommendation G.107 (2000): "The E-Model, a computational model for use in transmission planning".

26. ITU-T Recommendation G.108 (1999): "Application of the E-model: A planning guide".

27. ITU-T Recommendation G.109 (1999): "Definition of categories of speech transmission quality".

28. ITU-T Recommendation G.113 (1996): "Transmission impairments".

29. ITU-T Recommendation G.114 (1996): "Transmission systems and media — General characteristics of international telephone connections and international telephone circuits — One-way transmission time".

30. ITU-T Recommendation G.702 (1988): "Digital hierarchy bit rates".

31. ITU-T Recommendation G.703 (1998): "General aspects of digital transmission systems — Terminal equipments physical/electrical characteristics of hierarchical digital interfaces."

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

33. ITU-T Recommendation G.728 (1992): "Coding of speech at 16 kbit/s using low-delay code excited linear prediction".

34. ITU-T Recommendation G.729 (1996): "Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear-prediction (CS-ACELP)".

35. ITU-T Recommendation G.729 — Annex A (1996): "C source code and test vectors for implementation verification of the G.729 reduced complexity 8 kbit/s CS-ACELP speech coder".

36. ITU-T Recommendation Q.140 (1988): "Specifications of signalling system No 5 — Definition and function of signals".

37. ITU-T Recommendation Q.141 (1988): "Specifications of signalling system No 5 — Signal code for line signalling".

38. ITU-T Recommendation Q.151 (1988): "Specifications of signalling system No 5 — Signal code for register signalling".

39. ITU-T Recommendation Q.152 (1988): "Specifications of signalling system No 5 — End-of-pulsing conditions — register arrangements concerning ST (end-of-pulsing) signal".

40. ITU-T Recommendation Q.400 (1988): "Specifications of signalling system R2 — Definition and function of signals — Forward line signals".

41. ITU-T Recommendation Q.411 (1988): "Specifications of signalling system R2 — Line signalling, analogue version — Line signalling code".

42. ITU-T Recommendation Q.440 (1988): "Specifications of signalling system R2 — Interregister signalling — General".

43. ITU-T Recommendation Q.441 (1988): "Specifications of signalling system R2 — Interregister signalling — Signalling code".

44. ITU-T Recommendation Q.543 (1993): "Digital Exchanges — Digital exchange performance design objectives".

45. ITU-T Recommendation Q.921 (1997): "ISDN user-network interface — Data link layer specification".

46. ITU-T Recommendation Q.921 — Amendment 1 (2000): "Digital Subscriber Signalling System No.1 — Data link layer".

47. TIA/EIA-TSB-32-A (1998): "Overall Transmission Plan Aspects for Telephony in a Private Network".

48. EUROCONTROL document COM-GUI-01-00: Guidelines for the Implementation of the Automatic Voice Communication Network, Edition 2.0.

B.3 EQUIVALENCES FOR PSS1 (QSIG) SPECIFICATIONS

Specifications for PSS1 (QSIG) are mainly developed by ECMA, an international Europe-based industry association dedicated to the standardization of information and communications technology systems. They are then submitted to ISO/IEC for adoption as international standards. Once an international standard is published, the specification is also submitted to the ETSI for publication as an EN. Table B-1 lists the equivalent publications from each of the three organizations, ECMA, ISO/IEC and ETSI. Generally, the publications of the three organizations are technically aligned with one another. However, in some instances there are some technical differences between the publications of ISO/IEC and those of the other two organizations that reflect specific requirements of the European marketplace.

B.4 WEB SITES OF STANDARDIZATION BODIES

ECMA: www.ecma.ch

ETSI: www.etsi.org

ICAO: www.icao.int

ISO: www.iso.ch

ITU: www.itu.int

Table B-1. Equivalent publications for PSS1 (QSIG)

Service	ECMA standard	International standard	European standard
Basic call	ECMA-143	ISO/IEC 11572	EN 300 172
GFP	ECMA-165	ISO/IEC 11582	EN 300 239
Number identification	(Note)	(Note)	(Note)
Name identification	ECMA-164	ISO/IEC 13868	EN 300 238
Call intrusion	ECMA-203	ISO/IEC 14846	EN 300 426
Transit counter	ECMA-225	ISO/IEC 15056	EN 301 048
Call priority interruption and protection	ECMA-264	ISO/IEC 15992	EN 301 656
Addressing	ECMA-155	ISO/IEC 11571	EN 300 189
Mapping/static circuit mode	ECMA-226	ISO/IEC 14474	EN 301 765
Mapping/16	ECMA-253	ISO/IEC 17310	EN 301 039
Mapping/8	ECMA-289	ISO/IEC 17311	EN 301 924

Note.— The protocol specification for the identification SSs is contained within the basic call protocol standards (ECMA-143, ISO/IEC 11572 and EN 300 172).

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