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

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ITU-T Recommendation G.799.1/Y.1451.1

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ITU-T Recommendation G.799.1/Y.1451.1

Functionality and interface specifications for GSTN transport network equipment for interconnecting GSTN and IP networks

Summary

Voice and voiceband traffic in international networks has traditionally been transported by circuit switched systems and equipment. With the advent of the networks optimized for the transport of Internet Protocol (IP), and as a result of its considerable growth and pervasive nature, more and more voice traffic is expected to be carried over IP networks.

Given that voice and voiceband services remain a significant part of telecommunications, there is a need to ensure a high quality of service for voice carried in part, or wholly, via IP. This Recommendation defines interfaces and functionality for equipment that interconnect GSTN networks at the TDM interface and networks optimized for the transport of IP, such that they will provide the degree of voice quality and interoperability required.

Source

ITU-T Recommendation G.799.1/Y.1451.1 was approved on 13 June 2004 by ITU-T Study Group 15 (2001-2004) under the ITU-T Recommendation A.8 procedure.

Keywords

Bearer interface, echo canceller, end-to-end transmission performance, fax over IP, gateway, Internet gateway, Internet protocol, IP gateway, media gateway, media gateway controller, quality of service, signalling interface, speech coding, signalling interface, TDM, TDM-IP gateway, TIGIN, VoIP, voice gateway, voice over IP, voice quality.

FOREWORD

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As of the date of approval of this Recommendation, ITU had not received notice of intellectual property, protected by patents, which may be required to implement this Recommendation. However, implementors are cautioned that this may not represent the latest information and are therefore strongly urged to consult the TSB patent database.

CONTENTS

ITU-T Recommendation G.799.1/Y.1451.1

Functionality and interface specifications for GSTN transport network equipment for interconnecting GSTN and IP networks

1 Scope

This Recommendation covers the requirements of equipment that interconnects GSTN networks at the TDM interface and networks optimized for the transport of IP. Such equipment is referred to in this Recommendation as a TIGIN gateway. This Recommendation describes functionality such as voiceband coding and echo control, transport interfaces, signalling interfaces, and OA&M interfaces of these devices. Support is provided for calls from IP to GSTN, GSTN to IP, IP to IP and GSTN to GSTN. While such TIGIN gateways may support multimedia traffic and Remote Access Server (RAS) functionality, this Recommendation covers only voice, voiceband data, facsimile, narrow-band digital data, address and signalling tones. Support of other media such as digital data and video are outside the scope of this Recommendation.

This Recommendation describes bearer and signalling-related aspects of a TDM to IP interworking media gateway. It does not describe the requirements on a Media Gateway Controller, with the exception that the MGC is required to provide Q.115.1 Echo Control Logic. This Recommendation does not define any new protocols or network architectures, but rather refers to existing protocols and architectures. It also does not specify a level of performance, as this will be covered by other Recommendations referenced by this Recommendation. Operation and management procedures are outside the scope of this Recommendation.

TIGIN gateways are assumed to support bulk interconnection between GSTN and IP networks. They are considered to be a trunking and not an access gateway. Support of ATM on the GSTN interface is not included in this Recommendation.

2 References

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

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3 Terms and definitions

This Recommendation defines the following terms.

For terms and definitions not appearing in this clause, see ITU-T Rec. G.701 (1993), *Vocabulary of digital transmission and multiplexing, and pulse code modulation (PCM) terms*.

3.1 access gateway: A type of gateway that provides a User-Network Interface (UNI) such as ISDN.

3.2 general switched telephone network (GSTN): This network includes ATM, PSTN, ISDN, wireless networks and private networks.

3.3 media gateway (MG): The media gateway converts media format provided in one type of network to the format required in another type of network. For example, a MG could terminate bearer channels from a switched circuit network (e.g., DS0s) and media streams from a packet network (e.g., RTP streams in an IP network). This gateway may be capable of processing audio, video and T.120 multimedia signals alone or in any combination, and will be capable of full duplex media format translations. The MG may also play audio/video messages and performs other interactive voice response (IVR) functions, or may perform media conferencing. For the purpose of this Recommendation, the term media gateway refers to a voice gateway.

3.4 media gateway controller (MGC): Controls the parts of the call state that pertain to connection control for media channels in a media gateway.

3.5 media gateway control (Megaco): The term Megaco is the IETF name for ITU-T Rec. H.248.1. Megaco is defined in IETF RFC 3525. The protocol defined in RFC 3525 is common text with ITU-T Rec. H.248.1 Version 1. However, the latest version is H.248.1 Version 2 which is not published as an IETF RFC. When used in this Recommendation, Megaco is meant to be synonymous with the protocol defined in ITU-T Rec. H.248.1 Version 2 and is referred to throughout the Recommendation as "H.248/Megaco".

3.6 signalling gateway (SG): A signalling gateway contains the SCN Signalling Interface that terminates SS7 or other signalling links where the call control channels are separated from bearer channels.

3.7 simple packet relay transport (SPRT): A transport protocol defined in ITU-T Rec. V.150.1.

3.8 TIGIN gateway: A voice gateway complying with this Recommendation.

3.9 trunking gateway: A gateway between an SCN network and packet network that typically terminates a large number of digital circuits.

3.10 UDP transport layer (UDPTL): A transport protocol defined in ITU-T Rec. T.38.

3.11 voiceband data mode (VBD mode): A mode of operation characterized by use of a codec (e.g., G.711) that passes voiceband modulated signals with minimal distortion, has constant end-to-end latency, disables Voice Activity Detection, Comfort Noise Generation and any DC removal filters that may be integral with the speech encoder.

3.12 voice gateway (VG): A voice gateway is a subset of a gateway that deals with voice and voiceband traffic only, and not data or video traffic.

4 Abbreviations

This Recommendation uses the following abbreviations.

ATM Asynchronous Transfer Mode

CAS Channel Associated Signalling

CDMA Code Division Multiple Access

DHCP Dynamic Host Configuration Protocol

DiffServ Differentiated Services

DS0 Digital Signal level 0

DS1 Digital Signal level 1

DS3 Digital Signal level 3

DTMF Dual Tone Multi-Frequency

DTX Discontinuous Transmission

E11 (T1) Digital interface signal 1544 kbit/s (see ITU-T Rec. G.703)

E12 (E1) Digital interface signal 2048 kbit/s (see ITU-T Rec. G.703)

E32 (T3) Digital interface signal 44 736 kbit/s (see ITU-T Rec. G.703)

E31 (E3) Digital interface signal 34 368 kbit/s (see ITU-T Rec. G.703)

ESP Encapsulating Security Payload

FAX Facsimile

GSTN General Switched Telephone Network

IETF Internet Engineering Task Force

IKE Internet Key Exchange

IP Internet Protocol

ISDN Integrated Services Digital Network

ITU International Telecommunication Union

IVR Interactive Voice Response

M2UA A protocol, as defined in RFC 3331, for the backhauling of Signalling System 7 Message Transfer Part 2 (SS7 MTP2) User signalling messages over IP using the Stream Control Transmission Protocol (SCTP).

M3UA A protocol, as defined in RFC 3332, for supporting the transport of any SS7 MTP3-User signalling (e.g., ISUP and SCCP messages) over IP using the services of the stream control transmission protocol.

MF Multi-Frequency

MG Media Gateway

MGC Media Gateway Controller

MR Modem Relay

MTP Message Transfer Part

NAT Network Address Translation

NLP Non-Linear Processor

NMS Network Management System

OC3 Optical Carrier level 3

OC12 Optical Carrier level 12

OC48 Optical Carrier level 48

OC192 Optical Carrier level 192

PCC Per Call Control

PCM Pulse Code Modulation

PDH Plesiochronous Digital Hierarchy

PPP Point-to-Point Protocol

PSTN Public Switched Telephone Network

QoS Quality of Service

RAS Remote Access Server

RTCP RTP Control Protocol

RTP Real-time Transport Protocol

SCN Switched Circuit Network

SCTP Stream Control Transmission Protocol

SDH	Synchronous Digital Hierarchy
SG	Signalling Gateway
SID	Silence Insertion Descriptor
SIGTRAN	Signalling Transport
SNMP	Simple Network Management Protocol
SONET	Synchronous Optical Network
SPRT	Simple Packet Relay Transport
SS7	Signalling System No. 7
STM1	Synchronous Transport Module 1
STM4	Synchronous Transport Module 4
STM16	Synchronous Transport Module 16
STM64	Synchronous Transport Module 64
TCP	Transmission Control Protocol
TDM	Time Division Multiplex(ing)
TFO	Tandem Free Operation
TIGIN	**T**ransport Network Equipment for **I**nterconnecting **G**STN and **I**P **N**etworks
TMN	Telecommunication Management Network
UDP	User Datagram Protocol
UDPTL	Facsimile UDP Transport Layer Protocol
VBD	VoiceBand Data
VG	Voice Gateway
VoIP	Voice over IP

5 Conventions

In order to be compliant with this Recommendation, an implementation must provide functionality that is defined as mandatory. The words "shall" and "must", and their negatives, "shall not" and "must not" are used to express mandatory provisions.

Compliance with this Recommendation is achieved only when all mandatory provisions are met. However, the inclusion of mandatory provisions in a Recommendation does not imply, of itself, that compliance with the Recommendation is required of any party.

6 General description of GSTN transport network equipment for interconnecting GSTN and IP networks (TIGIN)

This Recommendation covers the requirements of equipment that interconnects GSTN networks and networks optimized for the transport of IP. TIGIN gateway is a physical entity that comprises the voice and voiceband data-related part of the media gateway function and, in some cases, the signalling gateway function as described in the decomposed gateway architecture (ITU-T Rec. H.248.1). The mapping of the functional entities to the physical entity is shown in Figure 1.

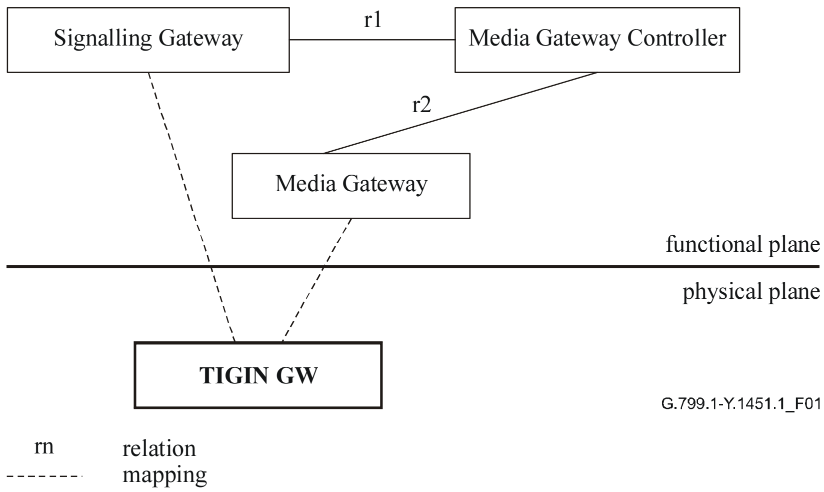
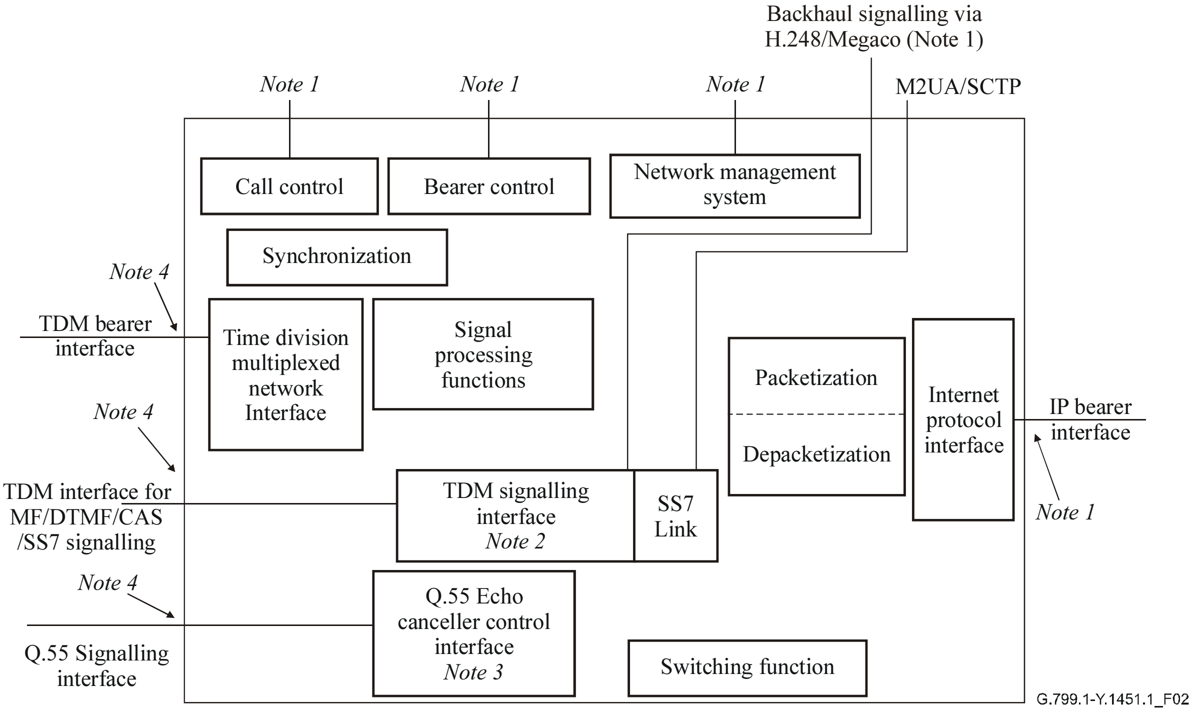


Figure 1/G.799.1/Y.1451.1 – Entity relationship diagram

In practice, a TIGIN gateway may be composed of multiple pieces of equipment, each with specialized functions, such as signalling interfaces, voice compression/decompression, packetization, etc. Figure 2 illustrates some of the functions performed in a gateway. This Recommendation does not specify how these functions are to be performed, nor does it specify the specific interconnections which may be implemented between functions. It is recognized that the functions of a signalling gateway may be combined with those of a TIGIN gateway in the same physical device. In this situation, certain interfaces described in this Recommendation may not be present. This Recommendation does not describe the requirements on a media gateway controller, with the exception that the MGC is required to provide Q.115.1 echo control logic.

The TIGIN gateway supports network configurations with SS7-associated signalling on TDM interface. In network configurations with quasi-associated signalling, a stand-alone signalling gateway, as shown in Figure 3, is required. The use of both quasi-associated signalling and associated SS7 signalling links within one cluster of TIGIN gateway, media gateway controller and signalling gateway is not possible due to SS7 constraints.



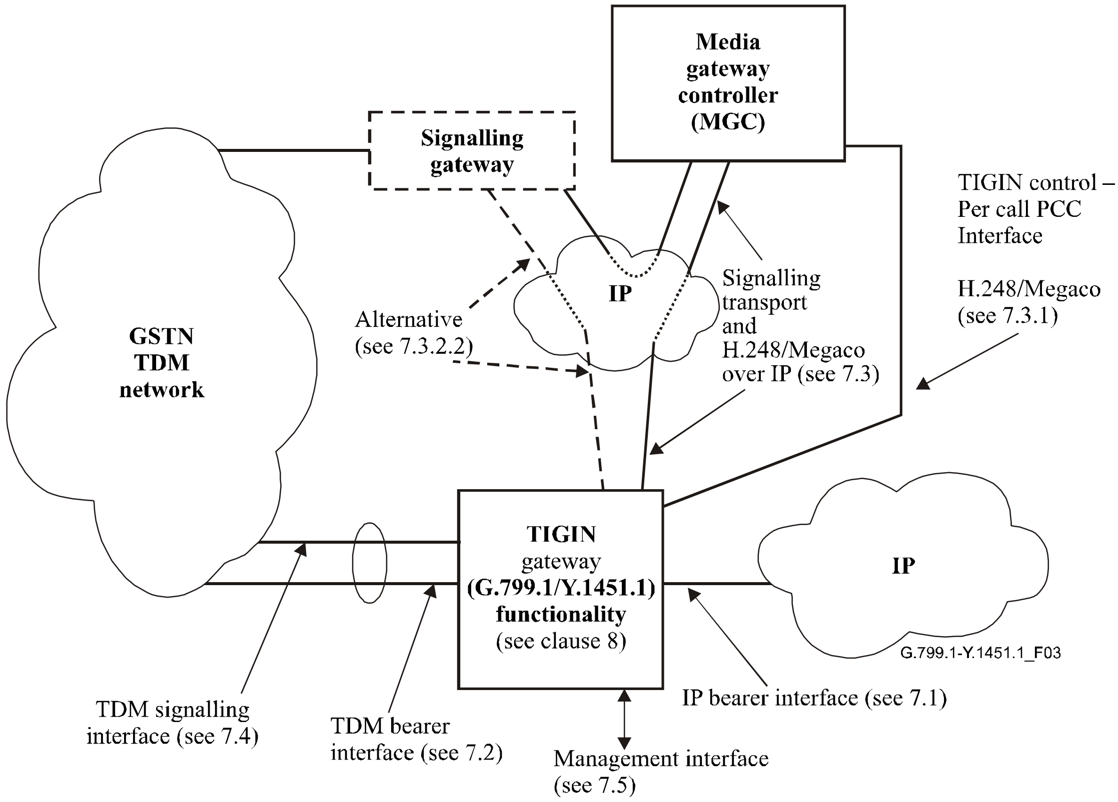
NOTE 1 – These may or may not share the same external interface (see 7.1 and 7.3).
NOTE 2 – Optional. Only required if SS7 associated or in-band (MF/DTMF/CAS) signalling on TDM interface is supported.
NOTE 3 – Optional.
NOTE 4 – These may or may not share the same external interface.

Figure 2/G.799.1/Y.1451.1 – Illustration of functions performed by a TIGIN gateway

Security functions which may be provided at several protocol layers of the various interfaces are not shown in Figure 2 above. Each of the elements included in the above block diagram is described in detail in the following clauses.

7 Interfaces

The overall relationship between the TIGIN gateway and other devices and protocols in the network is illustrated in Figure 3.



NOTE – In cases where the SS7 signalling networks consist of quasi-associated links, a separate signalling gateway is required as shown in the dotted-line box at upper left of this figure. It is not the intent of this figure to suggest the co-existence of quasi-associated and associated signalling links (see clause 7).

Figure 3/G.799.1/Y.1451.1 – Relationship between TIGIN gateway and other devices and protocols

The support of NAT and firewall traversal in networks with TIGIN gateways is for further study.

A TIGIN gateway should implement a mechanism for differentiating Quality of Service for bearer stream, call setup signalling, per call control signalling and management control signalling.

This can be achieved by, e.g.:

- Physical separation of interfaces;
- DiffServ (see IETF RFC 2475);
- Separation at Layer 2 (e.g., MPLS, VLAN, ATM VCC multiplexing, etc.).

7.1 IP bearer interfaces

The IP interface block in the block diagram in Figure 2 provides the transmission facility interface for the network element to the IP network.

Table 1 shows layers of the TIGIN IP bearer interface model.

Table 1/G.799.1/Y.1451.1 – IP bearer interface protocol layers

Application	Transparent	Voice	Tones	Modem (Note 1)		FAX (Note 1)	
Upper layers	RTP			SPRT	RTP (VBD mode)	UDPTL or RTP (Note 2)	RTP (VBD mode)
Transport layers	UDP (User datagram protocol)						
Network layer	IP (Internet protocol)						
Lower layers	Physical and data link						
NOTE 1 – See 8.1.5 for Modem and FAX support.							
NOTE 2 – The use of RTP provides call statistics through RTCP and allows a smoother transition between speech and non-speech communication in a call.							

Table 1 shows only basic stacks. Additional function for Quality of Service (QoS), security, header compression, etc. may be added.

7.1.1 Lower layers

This Recommendation does not restrict the use of any lower layer protocol intended for IP packet transmission and approved by an internationally recognized standards body.

Lower layer interface performs link protection, alarm detection and performance monitoring of the interface layers, as defined in appropriate standards.

7.1.2 IP layer

IP protocol shall conform to IETF RFC 791, RFC 950 and RFC 919.

Support of IPv6 (IETF RFC 2460) is for further study.

IP network topology, IP packet distribution and routing protocols are not the subject of this Recommendation.

For a given call, a TIGIN gateway shall transmit and receive packets on the same IP address.

7.1.3 Transport layer

UDP protocol shall be used as transport layer protocol and it shall conform to IETF RFC 768. For a given call, a TIGIN gateway shall include in the Source UDP port field of the UDP port header of all its transmitted packets, pertaining to a particular media stream, the same UDP port number as the one announced to its peer for that media stream.

7.1.4 Upper layers

RTP protocol is used for transparent signal transmission and for voice and tones transmission as IETF RFC 3550 describes. IETF RFC 3550 also defines RTP Control Protocol (RTCP) as a companion control protocol for RTP.

7.2 GSTN (TDM) bearer interfaces

The TDM interface block in the block diagram in Figure 2 provides the transmission facility interface for the network element to the GSTN network. The physical layer of the TDM bearer interface shall consist of at least one of the interface types in Table 2. Voice and voiceband coding on this interface shall conform to ITU-T Rec. G.711.

Table 2/G.799.1/Y.1451.1 – TDM bearer interfaces

Electrical	E1	G.703, G.704	DS1	ANSI T1.107
	E3	G.703, G.704	DS3	ANSI T1.107
	STM1	G.707/Y.1322, G.703		
Optical	STM1	G.707/Y.1322, G.957	OC3	ANSI T1.105
	STM4	G.707/Y.1322, G.957	OC12	ANSI T1.105
	STM16	G.707/Y.1322, G.957	OC48	ANSI T1.105
	STM64	G.707/Y.1322, G.957	OC192	ANSI T1.105

7.3 TIGIN control interfaces

7.3.1 TIGIN control-per call

Per call control (PCC) describes the interface between the TIGIN and the MGC that provides for real-time per call control of the TIGIN. The MGC is the control element that interworks between the signalling network and the TIGIN, providing the TIGIN with the real-time per-call control signals.

The PCC protocol defines the messaging between the MGC and the TIGIN and allows for the following capabilities such as:

- Establish, modify, and release connections within the TIGIN, including originating and terminating addresses (e.g., TDM-TDM, TDM-IP).
- Query the status of connections or resources within the TIGIN.
- Notification of call-related alarms or failures.
- Continuity Check.
- Transfer of certain GSTN Signalling information for MF and CAS-based protocols carried on the TDM link.

PCC allows the MGC to specify the internal TIGIN resources required in the bearer stream (Echo control, DTMF detection, Modem/Fax detection, etc.). It also allows for the setting of per call features and parameters required, based on call type and interoperability needs. These parameters may include some or all of the following:

- Real-time call processing event notification, including the control of echo cancellers;
- Silence suppression activation;
- Comfort noise generation;
- Jitter buffer size;
- Selection of and/or negotiation of codec (e.g., ITU-T Recs G.711, G.723.1, G.729).

  NOTE 1 – The presence of different codecs on the GSTN and IP interfaces implies performing a transcoding function (e.g., ITU-T Recs G.711 to G.729).
- Packet size selection.

Selection of these mechanisms will affect voice quality. See 8.1, 8.2 and 8.3 for more information on this topic.

Upon completion of a call, as specified in PCC protocols, a TIGIN shall be capable of supplying, upon request, statistics related to the call. A minimal set of statistics is listed below. These statistics can be provided using RTCP Sender and Receiver reports as outlined in RFC 3550.

- Number of packets sent;
- Number of octets sent;

- Number of packets received;
- Number of octets received;
- Average inter-arrival jitter.

The upper layer protocol used shall conform to H.248/Megaco.

The IP-based PCC interface may share the same physical media as the IP bearer interface.

The options for transport of H.248/Megaco control protocol between MGC and the TIGIN gateway are shown in Table 3 below. The transport over UDP and TCP is defined in Annex D/H.248.1.

Transport over SCTP is defined in ITU-T Rec. H.248.4.

Table 3/G.799.1/Y.1451.1 – TIGIN control protocol layers

<table>
<tr><th>Application</th><th colspan="5">TIGIN control</th></tr>
<tr><td>Upper layers</td><td colspan="5">H.248/Megaco</td></tr>
<tr><td>Transport layer</td><td>SCTP</td><td rowspan="2">MTP L3</td><td>M3UA
SCTP</td><td>UDP</td><td>TCP</td></tr>
<tr><td>Network layer</td><td>IP (Internet Protocol)</td><td colspan="3">IP (Internet Protocol)</td></tr>
<tr><td rowspan="2">Lower layers</td><td rowspan="2">Physical and data link</td><td>MTP L2</td><td colspan="3" rowspan="2">Physical and data link</td></tr>
<tr><td>MTP L1</td></tr>
</table>

NOTE 2 – If SS7-associated signalling links are used, then TIGIN and MGC should have the same Signalling Point Code. However, if H.248/Megaco is used over M3UA or MTPL3, then TIGIN and MGC must have different Signalling Point Codes. If H.248/Megaco signalling carried over M3UA or MTPL3 is used, then the use of SS7-associated signalling links is not possible. Conversely, in a network with SS7-associated links, the use of M3UA or MTPL3 as the transport layer for H.248/Megaco between TIGIN and MGC is, therefore, not possible.

Note that for networks that do not provide highly reliable connections, but where highly reliable transport is required SCTP, MTP3 or M3UA over SCTP should be used.

The protocols used in various stacks shall meet the requirements specified in the publications referenced below:

1) IP protocol shall conform to IETF RFC 791, RFC 950 and RFC 919.
2) Support of IPv6 (IETF RFC 2460) is for further study.
3) IP network topology, IP packet distribution and routing protocols are not the subject of this Recommendation.
4) UDP protocol is used as transport layer protocol and it shall conform to IETF RFC 768.
5) SCTP protocol shall conform to IETF RFC 2960 and IETF RFC 3309.
6) TCP protocol (RFC 793) may be optionally used for non-time-critical applications.
7) M3UA protocol shall conform to IETF RFC 3332.
8) MTP L1 protocol shall conform to Q.702.
9) MTP L2 protocol shall conform to Q.703.
10) MTP L3 protocol shall conform to Q.704.

7.3.2 Signalling over IP

A TIGIN gateway should support the SIGTRAN framework architecture for signalling transport over IP networks, defined by IETF RFC 2719.

7.3.2.1 In-band tones and signals

A TIGIN gateway shall have the capability to detect DTMF digits and other in-band signalling tones, as defined in 8.1.4, and generate messages to the media gateway controller, as described in H.248/Megaco.

When the call control used in an IP network does not mandate an out-of-band transport of DTMF and other in-band signals, a TIGIN gateway shall support the undistorted transport of DTMF and other in-band signals across the IP network. If a TIGIN gateway is using a codec that does not support the undistorted transport of DTMF, it shall transport those events either by other means provided by that codec for undistorted transport if available, or alternatively by IETF RFC/2833, if mutually negotiated.

7.3.2.2 Support for SS7, MF, and CAS signalling

The signalling gateway function may be integrated in the TIGIN Gateway. Its purpose is to transfer GSTN signalling information that arrives at the TDM interfaces of the TIGIN Gateway to the MGC, and vice versa.

Alternatively, the signalling interface for support of SS7 signalling may exist between the TIGIN gateway and the signalling gateway. Signalling support exists for two families of GSTN signalling, namely common channel signalling and MF/CAS-based signalling. Each case is handled in a different manner.

7.3.2.2.1 Support for Signalling System No. 7

A TIGIN gateway shall support SS7 by one of the following methods (see Table 4):

- Backhaul of common channel signalling using M2UA over SCTP, as described in RFC 2960 and RFC 3309.
- Use of M3UA, as described in IETF RFC 3332.
- SS7 signalling transport over RTP using 64 bit/s clear channel method (for further study).

Table 4/G.799.1/Y.1451.1 – Signalling transport control protocol layers

Upper layers	M2UA/M3UA	RTP
Transport layer	SCTP	UDP
Network layer	IP (Internet Protocol)	IP (Internet Protocol)
Lower layers	Physical and data link	Physical and data link

Note that for networks that do not provide highly reliable connections, but where highly reliable transport is required, M2UA or M3UA over SCTP should be used.

7.3.2.2.2 Support for MF and CAS-based signalling

A TIGIN gateway should support MF and CAS-based GSTN signalling protocols using the H.248/Megaco packages. The basic CAS packages are defined in ITU-T Rec. H.248.25.

7.4 TDM signalling interface

TDM signalling interfaces supported by equipment conforming to this Recommendation shall conform to national or international standards.

7.5 Management interface

Both TMN and SNMP management interfaces are for further study.

8 TIGIN functionality

8.1 Signal processing functions

The network element supports voice-related features as described in this clause. Some or all of the voice feature functions listed in this clause may be utilized during a call. The PCC message specifies which functions shall be used on a call.

As an example, silence suppression and comfort noise generation may be requested on a call that uses G.711 coding. The network element, in that case, would take the GSTN data, interpret it as G.711 coding and generate packets to the IP network using G.711 coding only if it detects speech activity. In the reverse direction, it would receive packets from the IP network, interpret them as G.711-encoded PCM, convert the packets into a synchronous (TDM) stream and send it to the GSTN network. During the time packets are not received it would generate comfort noise G.711 PCM values and send it to the GSTN network.

In addition, other signal processing features supported by the network element could include the following:

- Continuity check tone generation and detection;
- Continuity check loop-back.

Signal processing functions may affect voice quality. See 7.3.1 and 8.2 for more information on this topic.

8.1.1 Echo control

In compliance with ITU-T Rec. G.177, a voice-over-IP connection traversing a hybrid is required to include a G.168-compliant echo canceller. Since the echo canceller needs to have a constant delay for its echo path, it shall be placed so that the echo path is on the TDM side of the interface that includes the hybrid.

Network operators and service providers should ensure that echo cancellation is applied in the most appropriate location for the specific configuration. The echo canceller shall be associated with the TIGIN gateway in one of three possible configurations, as outlined in the following clauses.

It should be reinforced that all voice connections involving GSTN-to-IP interworking involving a hybrid, require echo cancellation and the network operator or service provider, choosing to utilize either configuration B or C below, should ensure that an echo canceller is provisioned on the TDM bearer side of the hybrid Internet/GSTN connection.

8.1.1.1 TIGIN internal echo control – Configuration A

In this configuration, the echo canceller is integrated into the TIGIN gateway itself, as shown in Figure 4. With this configuration, H.248/Megaco messages convey the echo canceller control information from the MGC to the TIGIN gateway. The TIGIN gateway then directly controls the internal echo canceller function in accordance with the messages received. For this configuration, the echo canceller function within the TIGIN gateway is required and the Q.55/Q.52 process is optional but not used.

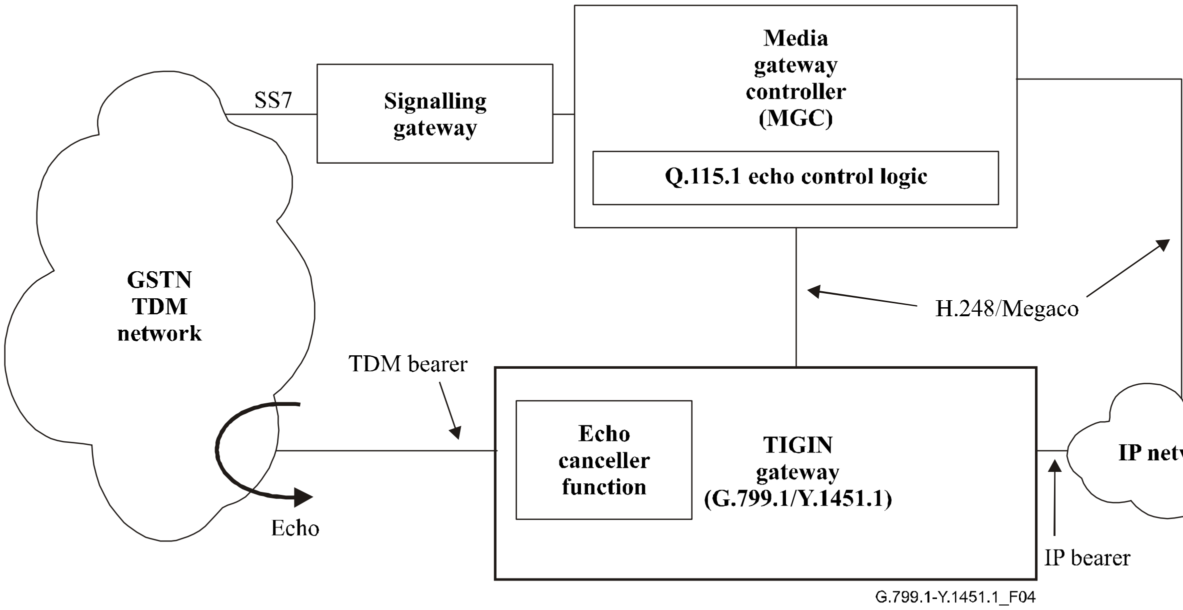


Figure 4/G.799.1/Y.1451.1 – Echo canceller integrated into TIGIN gateway – Configuration A

8.1.1.2 TIGIN external echo control with Q.55/Q.52 control – Configuration B

In this configuration, the echo canceller is directly associated with and controlled by the TIGIN gateway, as shown in Figure 5. With this configuration, H.248/Megaco messages convey the echo canceller control information from the MGC to the TIGIN gateway. The TIGIN gateway then converts the messages into appropriate protocol data units and sends them to the associated external echo canceller. The conversion processes are described in 8.1.1.2.1. For this configuration, the echo canceller function within the TIGIN gateway is optional but not used, and the Q.55 process is mandatory.

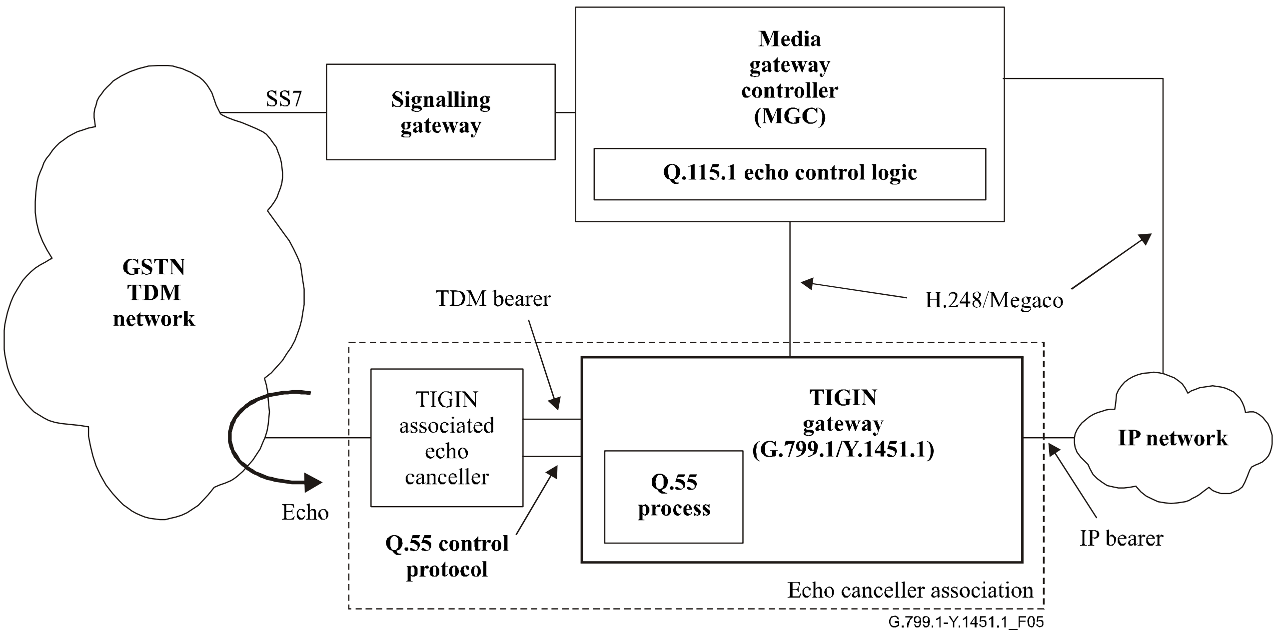


Figure 5/G.799.1/Y.1451.1 – TIGIN external echo control with Q.55 control – Configuration B

8.1.1.2.1 Echo canceller control procedures

When the echo cancellation function is not integrated into the TIGIN gateway, but is located on a TDM trunk of the gateway and is controlled by the TIGIN gateway, the following procedures shall be used. These procedures define the mapping between H.248/Megaco messages received by the TIGIN gateway from the media gateway controller, and Q.55 protocol data units transmitted to the associated external echo canceller.

Echo canceller enable procedure

When the H.248/Megaco package responsible for controlling echo cancellers is received with the value set to "on" for a specific TDM DS0 channel, the Q.55 protocol data unit "enable echo cancellation for channel n", as defined in Annex H/Q.55, where 'n' is the DS0 channel specified by the H.248/Megaco message, is transmitted on the Q.55 echo control interface of the TIGIN gateway.

Echo canceller disable procedure

When the H.248/Megaco package responsible for controlling echo cancellers is received with the value set to "off" for a specific TDM DS0 channel, the Q.55 protocol data unit "disable echo cancellation for channel n", as defined in Annex H/Q.55, where 'n' is the DS0 channel specified by the H.248/Megaco message, is transmitted on the Q.55 echo control interface of the TIGIN gateway.

8.1.1.3 GSTN switch-associated echo control – Configuration C (Informational)

In this configuration, the echo canceller is not directly associated with or controlled by the TIGIN gateway. Instead, the echo canceller is associated with and may be controlled by a switch in the GSTN, as shown in Figure 6.

If the GSTN switch controls the echo canceller, it may either be controlled via a Q.55 interface, or, via some other control mechanism. If the GSTN switch controls the associated echo canceller via Q.55, echo control messages are generated as a result of the Q.115.1 echo control logic information in the GSTN switch and are sent to the GSTN switch-associated echo canceller. For this configuration, while the echo canceller function and/or Q.55 process within the TIGIN gateway may exist, they are not used.

It shall be noted that the Q.115.1 echo control logic in the MGC will request the disabling of a canceller associated with the TIGIN gateway, if any is present, as in this particular case, one has already been provisioned in the GSTN.

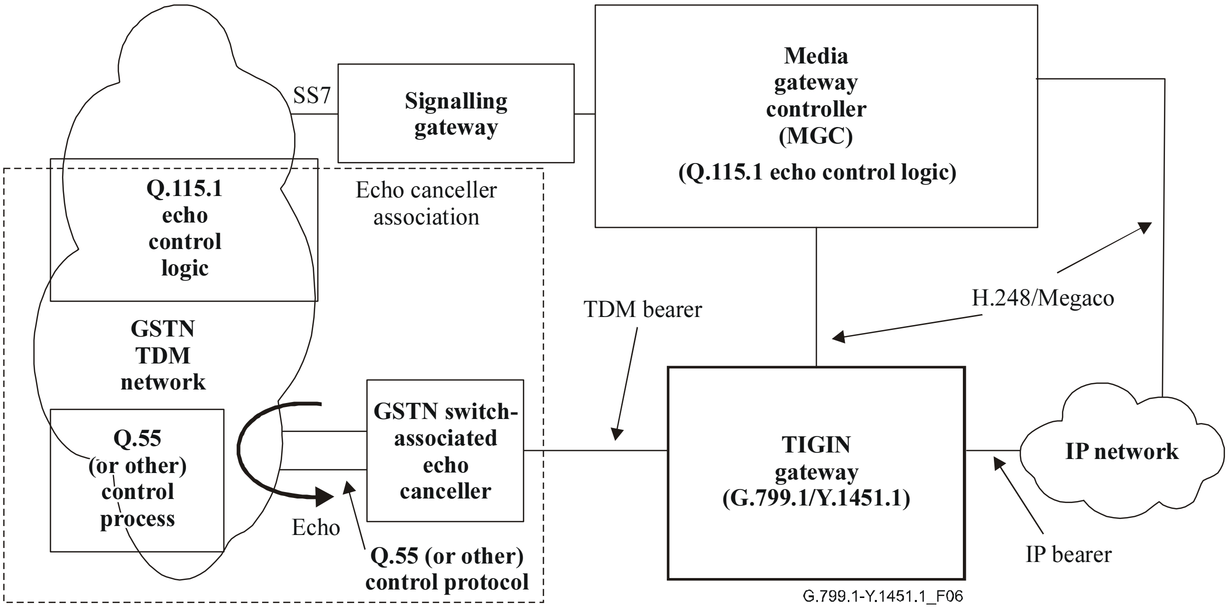


Figure 6/G.799.1/Y.1451.1 – Non-TIGIN-associated echo control – Configuration C

8.1.2 Voice coding

As a minimum, a TIGIN gateway shall include a G.711 codec (for both A and μ-laws). The default G.711 codec (e.g., A or μ-law) shall be selected based upon the physical location of the TIGIN and mechanisms described in ITU-T Rec. H.248.1.

A TIGIN gateway may optionally include codecs other than G.711 (e.g., G.728, G.729, G.723.1, codecs commonly used in GSM, CDMA, TDMA, etc. networks). Delay caused by voice coding should be kept to a minimum (see Annex A/G.114 for calculation of delay figures).

8.1.3 Digital speech interpolation

Silence suppression and comfort noise generation can be either internal or external to the codec.

8.1.3.1 Internal silence suppression and comfort noise generation

8.1.3.1.1 Silence suppression

Silence suppression shall be supported according to the speech codec used, e.g., Annex B/G.729 for G.729, or Annex A/G.723.1 for G.723.

8.1.3.1.2 Comfort noise generation

Comfort noise generation shall be supported according to the speech codec used, e.g., Appendix II/G.711 for G.711, Annex B/G.729 for G.729, or Annex A/G.723.1 for G.723.1.

8.1.3.2 External silence suppression and comfort noise generation

There should be a method of ensuring synchronous reset capability between the encoder and decoder of G.726, G.727, G.728 and G.729 when external comfort noise generation is used with these codecs. In these cases, an external synchronous reset will be beneficial to voice quality, particularly at the transient from-silence-to-active-speech when the codec is used in the DTX condition. If generic SID comfort noise insertion is used with a low bit rate coder, synchronous reset between encoder and decoder should be employed.

8.1.4 Tones and announcements

The following functions may be hosted by a TIGIN gateway:

- Voice announcements;
- Supervisory tones;
- Tone detection.

If any of these features are provided, they shall be provided in the manner described in this clause. The following clauses describe each function in detail and give some examples of when they are required.

8.1.4.1 Voice announcements

Voice announcements fall into two categories:

- fixed announcements;
- announcements with variable parts.

Fixed announcements are used by basic PSTN calls in circumstances where, for example, it is necessary to play a rejection announcement to a caller such as: ***The number you have dialled has not been recognized, please check and try again***.

Announcements with variable parts are often required for PSTN supplementary services where, for example, a date, time or telephone number needs to be inserted into a standard announcement e.g., for a call that has been diverted: ***Telephone number <TN> will receive your incoming calls***. In this case, <*TN*> is the variable part that identifies the diverted-to telephone number.

All voice announcements shall be controlled via H.248/Megaco from the Media Gateway Controller using H.248.7.

8.1.4.2 Supervisory tones

In addition to the voice announcements described in 8.1.4.1, a small range of supervisory tones, e.g., Dial, Ringing, Busy, etc., need to be generated to support PSTN supplementary services. For example, these are necessary to cover supplementary service invocation confirmations such as: ***If this is incorrect press # after the dial tone*** **<Dial Tone>**.

All supervisory tones shall be controlled via H.248/Megaco from the Media Gateway Controller using Annex E/H.248.1.

Technical characteristics of the tone generation shall conform to relevant national and international standards.

8.1.4.3 DTMF tone detection

DTMF receivers are required to collect end-user responses for PSTN supplementary services that involve interactive dialogues. Generally, this functionality is required for Intelligent Network (IN) services with interactive menu dialogues, but it is also needed for some of the more basic PSTN services such as call diversion. For example, a DTMF receiver is necessary to collect the invited response to the call diversion service prompt: ***If this is incorrect press # after the dial tone***.

Tone detection shall be controlled via H.248/Megaco from the Media Gateway Controller using Annex E/H.248.1.

8.1.5 Facsimile, voiceband data modem, and text telephony handling

A TIGIN gateway shall be capable of transporting facsimile data, voiceband modem data and text telephony data defined in ITU-T Rec. V.18 over IP networks. National preferences shall dictate which of the V.18 modulation schemes are supported. At a minimum, a TIGIN gateway shall support facsimile, voiceband data modem and text telephony transmission via VBD Mode. A standardized mechanism for transport of this type is under study within ITU-T.

Support for the following relay mechanisms is optional.

If facsimile transmission is supported via facsimile relay, it shall be done by standardized mechanisms. For Group 3 Facsimile Equipment, T.38 shall be used. To ensure a minimum expectation of performance, a TIGIN gateway that supports T.38 shall support the T.38 default and optional transitioning mechanisms defined in Annex E/T.38 for transitioning between Voice and T.38 states. The choice of the transitioning mechanism is made by the MGC.

If modem transmission is supported via modem relay, it shall be done by V.150.1. Note that ITU-T Rec. V.150.1 also supports VBD mode. If V.150.1 is a mutually negotiated option, then VBD mode can be supported by the procedures defined by that Recommendation.

To ensure a minimum expectation of performance, a TIGIN gateway that supports a V.150.1 modem relay shall:

- Only use a universal MR gateway type.
- Use either single or double trans-compression.

If text transmission is supported via text relay, it shall be supported via applicable national and international standards. A standardized mechanism for transport of this type is under study within ITU-T.

8.1.6 Speech reconstruction

Voice coders in a TIGIN gateway should follow established rules for speech reconstruction under conditions of lost packets. Packet loss concealment techniques shall be provided for all codecs supported by the TIGIN Gateway, e.g., Appendix I/G.711 and Annex I/G.728. It should be noted that ITU-T Recs G.729 and G.723.1 have packet loss concealment capabilities embedded into their respective decoders.

8.1.7 Voice encryption

Methods of supporting voice encryption are for further study.

8.1.8 Distributed speech recognition and distributed speaker verification

Methods of supporting distributed speech recognition and distributed speaker verification are for further study.

8.1.9 Jitter buffer

A TIGIN gateway shall include a jitter buffer that compensates for jitter (or delay variation) introduced by the IP network. Jitter (delay variation) can occur for a number of different reasons including network congestion, queuing, etc. and, if the jitter buffer is not dimensioned properly, then voice quality degradation can result. If the jitter buffer is too small to accommodate the maximum delay experienced by some packets, then packet loss will occur. If the jitter buffer is too large, the end-to-end delay experienced by the user is increased.

For voice traffic the jitter buffer shall be adaptive to minimize the delay while keeping packet loss rate below a reasonable limit.

For VBD mode, the jitter buffer shall provide constant latency. For example, this may be achieved either by fixed jitter buffer, or by freezing the adaptive jitter buffer.

The default jitter buffer settings shall be provisionable via a management interface. It shall be possible to override these settings on a per-ephemeral basis using H.248/Megaco control via the media gateway controller.

It should be possible to set jitter buffer parameters to support reliable transport of 64 kbit/s ISDN data. Jitter buffer requirements for supporting 64 kbit/s ISDN data are for further study

8.2 Speech quality

In providing the following functions, TIGIN gateway design should be optimized to minimize degradation of speech transmission performance:

- Echo cancellation;
- Adaptive jitter buffering;
- Packet loss concealment;
- Silence suppression;
- Voice activity detection;
- Comfort noise generation.

See 8.1.3 for handling of silence suppression and comfort noise generation.

Support for tandem coder avoidance, other than Tandem Free Operation (TFO), is for further study. As outlined in ITU-T Rec. G.177, it is recommended that only one encode/decode of speech be performed, and that speech coder tandem processing be eliminated whenever possible.

ITU-T Rec. P.1010 provides fundamental speech transmission objectives for VoIP gateways and provides references to other Recommendations which need to be taken into account for gateways to support acceptable end-to-end speech transmission performance. Additionally, ITU-T Rec. G.177 should be followed with respect to end-to-end transmission performance and planning guidance for voiceband services.

Note that a methodology to objectively predict and evaluate end-to-end speech quality is described in ITU-T Rec. P.862. Additionally, ITU-T Rec. P.502 provides objective test methods for evaluation of speech transmission quality of terminals and speech transmission systems, and utilizes test signals defined in ITU-T Rec. P.501. Specific objective requirements for these methodologies are for further study.

Definitions of categories of voice transmission quality may be found in ITU-T Rec. G.109.

8.2.1 TIGIN support for tandem free operation (Optional)

As different packet networks will be interfaced with one another, the tandeming of speech codecs operating at low to medium rates (4 to 16 kbit/s) will be very common. Because tandem speech codecs degrade speech quality, it is highly recommended that a single coded pair be used end-to-end.

In calls where TIGIN gateways connect two transcoding mobile networks, the TIGIN gateways become in-path equipment with respect to the two networks. In order to avoid speech codecs in tandem operation and to avoid degradation of voice quality, G.799.1/Y.1451.1 gateways could optionally support, on a call-by-call basis depending on the bandwidth available in the IP network, one or more TFO protocols such as defined in the following:

- TIA/EIA-895-A, *CDMA Tandem Free Operation*;
- ETSI TS 128 062, *Inband Tandem Free Operation (TFO) of Speech Codecs, Stage 3 – Service Description*.

The support for other possible TFO standards is for further study.

The above two TFO standards were based on the same in-band signalling protocol and are intended for Wireless services over circuit-switched networks. The mechanism for supporting TFO over IP-based networks is for further study.

8.2.2 Packet size selection

The choice of packet size is a trade-off between transport efficiency, quality and delay. The delay associated with codec processing and packetization should be kept as short as possible. Gateways should adhere to ITU-T Rec. G.177 in this regard. When multiple frames of coded speech are allocated to the same packet, packet loss concealment techniques become less effective and, as a result, may possibly lower end-to-end speech quality when packet loss is encountered.

Where codec frame size permits, a TIGIN gateway shall support the packetization times of 10 and 20 ms. Otherwise, a TIGIN gateway shall support a packetization time of one frame size (e.g., 30 ms for G.723.1). These packetization times shall be supported in both transmit and receive directions. Other packetization times may be supported and negotiated under control of H.248.1.

Packet size shall be an integer multiple of the codec frame length. To accomplish this objective when G.729 or G.729A is used, two frames per packet shall be considered as the maximum packet size.

Similarly, G.711 and G.726 may be used with packet sizes of 10 ms (80 samples) or 20 ms (160 samples) to achieve this objective.

Finally, when G.723.1 is used, only one frame shall be included in each packet. The 7.5 ms look-ahead and the 30 ms frame size of G.723.1 results in a combined voice buffering and processing delay of between 37.5 ms and 67.5 ms (depending on the computational power available), thus contributing to the difficulty of interactive communications.

8.3 Switching

The TIGIN gateway shall provide the ability to switch telephone calls in the direction of the PSTN and in the direction of the IP network. Switch control is based on general call control and allows the establishment, modification and release of the connection within the TIGIN. The TIGIN architecture is based on separation of the logical control plane from the transport resources. The TIGIN architecture should be suitable for high availability networks, which includes non-blocking paths through the TIGIN.

TIGIN switching functions include:

- Providing basic cross-connect functionality between TDM and IP logical ports.
- Forwarding user information.
- Supporting a multiplicity of switching elements under the domain of a single controller.
- Providing switching element reliability.
- Replicating data for point-to-multipoint connections.
- Providing a common switch control interface to one or more controllers.

8.4 Clock recovery and synchronization

A TIGIN gateway requires a clock recovery circuit that allows synchronization to a reference clock and generation of a precise local clock. This synchronization circuit shall support primary and secondary synchronization inputs for reference clocks, and may support additional reference inputs.

A TIGIN gateway node clock shall be traceable to an external ITU-T Rec. G.811 source. This may be achieved using external clock ports or traffic ports as described below:

- Any E1 or DS1 TDM or SONET/SDH interface;
- G.703 E1/T1 External Clock Port;
- 2048 kHz synchronization signal according to clause 13/G.703.

In addition, the TIGIN gateway node clock shall meet the requirements of ITU-T Rec. G.813 that specifies slave clock performance applicable to SDH equipment. The following references on Jitter and Wander are applicable to signal interfaces and synchronization signals. ITU-T Rec. G.823 is applicable for PDH signals based on the 2048 kbit/s hierarchy and synchronization signals, and ITU-T Rec. G.824 is applicable for PDH signals based on the 1544 kbit/s hierarchy. ITU-T Rec. G.825 is applicable for SDH signals. The physical layer for these signals is defined in ITU-T Rec. G.703. Specifically, 2048 kHz is specified in clause 13/G.703.

Similar to the case of hierarchical master-slave SCN synchronization methods, it is recommended that a primary and a secondary synchronization reference input be provided to each clock.

NOTE – If a TIGIN gateway connects to a VoIP terminal then there may be clock rate differences between the local clock in the VoIP terminals and the clock rate in the GSTN network. This frequency difference may result in slips and packet discards/duplications.

8.5 Security

8.5.1 Security on management interfaces

Security on management interfaces is for further study.

8.5.2 Per call control and signalling interfaces security

If a TIGIN gateway supports security on IP-based per call control and signalling interfaces, then it shall be supported in the following manner. A TIGIN that utilizes IP for the transport of per call control and signalling shall support IPSec and must be able to secure the traffic flows using ESP (see IETF RFC 2406) with null encryption and IPSec transport and tunnel mode encapsulation as defined in IETF RFC 2401. All other methods of security defined in IETF RFC 2401 are optional and shall be negotiated via IKE.

8.6 Additional features

8.6.1 Packet filtering

A TIGIN gateway may provide support for source IP address and source UDP port filtering. This support must be provisionable. This is based on source and destination transport addresses being equal as per 7.1.2 and 7.1.3.

Note that when connecting to devices that are not TIGIN gateways, this may not be possible as this may lead to RTP non-interoperability if the remote device does not send and receive from the same transport address.

8.6.2 Lawful interception (optional)

Support for lawful interception is optional.

If lawful interception is supported by a TIGIN gateway then it shall not degrade the Quality of Service of any intercepted call in a detectable way.

NOTE – The following is a list of possible (non-exhaustive) ways that lawful intercept could degrade performance in a detectable way:

- impact on latency (e.g., large or fixed increase);
- impact on loss (e.g., large or fixed increase);
- impact on jitter.

Ways of providing lawful interception in this Recommendation are for further study.

8.7 Management functions and management system interfaces

The management system interface performs the functions to administer the system on a non-real-time (non-per-call) basis with respect to the following subclauses. Further details for each of the management functions outlined in the following subclauses are included in the general (e.g., ITU-T Rec. M.3100) and technology specific (e.g., ITU-T Rec. G.784, IETF RFC 2579) Recommendations.

Support of TMN and SNMP management systems are for further study.

8.7.1 Configuration management

For further study. Issues to be considered include the management of:

- TDM interface features such as line rates, frame formats;
- IP interface features;
- Packetizer options such as packet loss mitigation scheme, which are not needed to be controlled on a per call basis;
- Echo canceller options;
- Voice feature options;
- Clock recovery;
- All other configurable parameters.

8.7.2 Fault management

For further study. Issues to be considered include the management of:

- Equipment faults;
- Facility faults at TDM and IP sides.

8.7.3 Performance management

Performance management includes items such as facility and connection performance monitoring, congestion control, and traffic management.

Clause 8.7.3.1 provides requirements for call-quality monitoring and diagnostic capabilities.

Clause 8.7.3.2 provides requirements for congestion control and traffic management.

8.7.3.1 Call-quality monitoring and diagnostics

The TIGIN gateway shall provide the capability to monitor and report on individual call-quality and causes of voice or signal degradation using RTCP Extended Reports (RTCP XR) parameters and reporting capabilities. ITU-T Rec. H.248.30 provides a method of reporting those parameters to MGC. A generic RTCP monitoring and reporting capability that meets these requirements is described in IETF RFC 3611. The following listing of performance parameters shall be supported by the TIGIN gateway. Note that the performance parameters/metrics listed below are a subset of those listed in IETF RFC 3611. The definitions for each of the listed parameters/metrics are provided in IETF RFC 3611.

a) Network Packet Loss Rate – See IETF RFC 3611, section 4.7.1 for the definition of this parameter/metric. The proportion of packets lost during transmission.

b) Jitter Buffer Discard Rate – See IETF RFC 3611, section 4.7.1 for the definition of this parameter/metric. The proportion of packets discarded by the receiving jitter buffer. This may be high if the jitter level is high or if the jitter buffer is too small.

c) Burst Loss Density – See IETF RFC 3611, section 4.7.2 for the definition of this parameter/metric. The average proportion of packets both lost and discarded during burst periods. A burst is a period during which a high proportion of packets are either lost in transit or discarded due to late arrival. In general, a burst is likely to result in audible degradation to call quality. A burst is defined as a longest sequence that:

   1) starts with a lost or discarded packet;

2) does not contain any occurrences of Gmin or more consecutively received (and not discarded) packets; and

3) ends with a lost or discarded packet.

d) Gap Loss Density – See IETF RFC 3611, section 4.7.2 for the definition of this parameter/metric. The average proportion of packets lost and discarded during gaps. A gap is a period between bursts; the above burst definition means that during gaps the packet loss rate is low and lost/discarded packets are isolated and hence can be effectively masked by packet loss concealment algorithms.

e) Burst Duration – See IETF RFC 3611, section 4.7.2 for the definition of this parameter/metric. The average length of burst periods expressed in milliseconds.

f) Gap Duration – See IETF RFC 3611, section 4.7.2 for the definition of this parameter/metric. The average length of gap periods expressed in milliseconds.

g) RTCP Round Trip Delay – See IETF RFC 3611, section 4.7.3 for the definition of this parameter/metric. The round trip delay between RTP instances, expressed in milliseconds.

h) End System Delay – See IETF RFC 3611, section 4.7.3 for the definition of this parameter/metric. The end system delay, comprising encode, decode and jitter buffer delay, expressed in milliseconds. This may be combined with the RTCP Round Trip Delay to estimate the overall Voice over IP segment round trip delay.

i) Signal Level – See IETF RFC 3611, section 4.7.4 for the definition of this parameter/metric. The received signal level measured during talkspurts, expressed in dBm0.

j) Noise Level – See IETF RFC 3611, section 4.7.4 for the definition of this parameter/metric. The received noise level measured during silence periods, expressed in dBm0.

k) Residual Echo Return Loss – See IETF RFC 3611, section 4.7.4 for the definition of this parameter/metric. The echo return loss after the effects of echo cancellation, expressed in dB. Both Local and Remote RERL values are specified, although only the Local RERL may be available (from the Gateway's Echo Canceller).

l) Gmin – See IETF RFC 3611, section 4.7.6 for the definition of this parameter/metric. A parameter used to define bursts. (See c above for the definition of "burst".) This is, by default, set to 16, which sets the threshold packet loss rate between bursts and gaps to approximately 6%.

m) R Factor – See IETF RFC 3611, section 4.7.5 for the definition of this parameter/metric. A calculated R factor representing receiving end call quality.

The R-factor is dependent on the availability of a number of different parameters. The minimum set of parameters that shall be measured or confidently estimated in order to calculate a meaningful R-factor is as follows:

- Talker echo loudness rating;
- Mean one-way delay of the echo path;
- Round-trip delay in a 4-wire loop;
- Absolute delay in echo-free connections;
- Equipment impairment factor;
- Packet-loss robustness factor;
- Random packet-loss probability;
- Circuit noise referred to 0 dBr-point.

All other parameters that cannot be measured or estimated may be set to their default values defined in ITU-T Rec. G.107. This includes the case where an echo canceller is not integrated with the TIGIN gateway as per configuration A in 8.1.1.1.

n) External R Factor – See IETF RFC 3611, section 4.7.5 for the definition of this parameter/metric. An R factor representing the effects of an externally connected network.

o) Estimated MOS-LQ – See IETF RFC 3611, section 4.7.5 for the definition of this parameter/metric. An estimated receiving end Listening Quality MOS.

p) Estimated MOS-CQ – See IETF RFC 3611, section 4.7.5 for the definition of this parameter/metric.

q) Packet Loss Concealment Type – See IETF RFC 3611, section 4.7.6 for the definition of this parameter/metric. Type of packet loss concealment algorithm in use. Indicates unknown, silence insertion, "standard" (i.e., per appropriate ITU-T Recommendation) and "enhanced".

r) Jitter Buffer Type – See IETF RFC 3611, section 4.7.6 for the definition of this parameter/metric. Indication of whether the receiving jitter buffer is fixed or adaptive or unknown.

s) Jitter Buffer Adaptation Rate – See IETF RFC 3611, section 4.7.6 for the definition of this parameter/metric. For adaptive jitter buffers – the adaptation rate. This is defined as the time taken to adjust to a step of 70 ms increase in jitter divided by twice the frame size in milliseconds. This value need only be provided if the Jitter Buffer Type is specified.

t) Jitter Buffer Nominal Size – See IETF RFC 3611, section 4.7.7 for the definition of this parameter/metric. For fixed and adaptive jitter buffers – the current nominal jitter buffer size. This is expressed in milliseconds. This value need only be provided if the Jitter Buffer Type is specified.

u) Jitter Buffer Max Size – See IETF RFC 3611, section 4.7.7 for the definition of this parameter/metric. For fixed and adaptive jitter buffers – the maximum jitter buffer size in milliseconds. This value need only be provided if the Jitter Buffer Type is specified.

v) Jitter Buffer Absolute Max – See IETF RFC 3611, section 4.7.7 for the definition of this parameter/metric. For adaptive jitter buffers – the maximum achievable size of the buffer in milliseconds. This value need only be provided if the Jitter Buffer Type is specified.

8.7.3.2 Congestion control and traffic management

The TIGIN gateway shall be robust against excessive H.248 messaging from MGC, e.g., it must not reset itself nor shall calls in-progress be terminated in an uncontrolled manner.

The TIGIN gateway shall support H.248.10, media gateway resource congestion handling package or H.248.11, media gateway overload control package, or both, to protect the gateway from call processing overload.

When SS7 messages enter the TIGIN via a TDM interface and get forwarded to the MGC (using M2UA or M3UA over SCTP), overload protection against signalling congestion at the TIGIN may be required, depending upon how the TIGIN is engineered internally. If it is required, then the TIGIN shall be capable of detecting that it has internal signalling congestion, and shall apply SS7 signalling management controls:

- For M3UA scenarios, control using Transfer Controlled (TFC) and Transfer Prohibited (TFP) procedures defined in ITU-T Recs Q.703 and Q.704 shall be used.
- For M2UA scenarios, congestion control using MTP2 messaging and M2UA primitives shall be used.

Overload controls and traffic management required for the other signalling interfaces (MF, DTMF, CAS) are for further study.

Monitoring of TIGIN internal resource usage for performance and capacity management purposes is for further study.

8.7.4 Security management

8.7.4.1 Key management

If a TIGIN gateway supports security on IP-based per call control and signalling interfaces, then security management shall be supported in the following manner:

A TIGIN gateway shall support the Internet Key Exchange (IKE) protocol for security key management (see IETF RFC 2409) with the minimum support of pre-shared keys as defined in IETF RFC 2409.

A TIGIN gateway may additionally authenticate itself to the remote site using either one of public key encryption authentication forms or digital signature as described in IETF RFC 2409. For the case of digital signature authentication, X.509 certificates shall be used (see IETF RFC 3280).

Security management, including key management for all other IP traffic, is for further study.

8.7.5 Accounting

No accounting functions are supported in a TIGIN gateway. Note that this does not imply that signals such as on-hook, off-hook, etc., that support or provide information necessary for certain accounting type functions (e.g., call billing) do not need to be conveyed or supported. Rather, that the TIGIN gateway itself does not provide any direct accounting functionality such as call billing, etc.

8.7.6 Maintenance testing

The TIGIN Gateway shall provide the ability to monitor calls and place test calls in the direction of the GSTN and in the direction of the IP network.

The monitor and intrusive access point shall provide a test access point on a per TIGIN gateway basis and not on a per-shelf or per-rack basis, i.e., one per TIGIN gateway. This test access capability shall support all features and services provided by the TIGIN gateway.

8.7.6.1 Non-intrusive monitor access

The monitor access shall provide the ability to connect to a specific port/address/call and monitor one or both directions of an active call simultaneously (Figure 7). It shall be possible to monitor a specific port or connection within the TIGIN gateway allowing a port to be monitored before a call is active on the port. When an active call is placed on the port, the monitor shall remain active. The monitor access may provide either a TDM or IP output and will depend on the actual implementation of the TIGIN gateway.

In the non-intrusive monitoring case, there is no requirement that signalling be monitored.

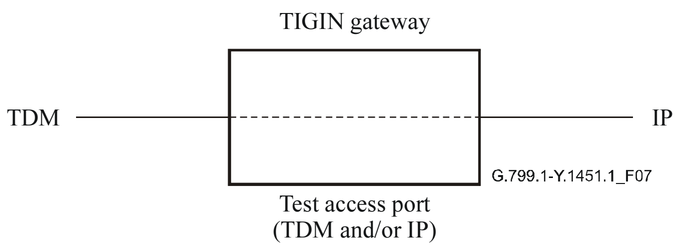


Figure 7/G.799.1/Y.1451.1 – Non-intrusive monitor access

8.7.6.2 Intrusive monitor access

The intrusive test access shall allow test calls to be made in either direction. The intrusive test access shall allow the user to connect to a specific port/address and place a call to a destination, either in the direction of the GSTN, or the IP network. The intrusive test access may provide either a TDM or IP output and will depend on the actual implementation of the TIGIN gateway.

In the case where the test access port is TDM (e.g., Figure 8a), a TDM/IP conversion function is required to access the IP data port or stream.

The specific point within the IP path where access is placed shall be the last IP point prior to any speech processing functions.

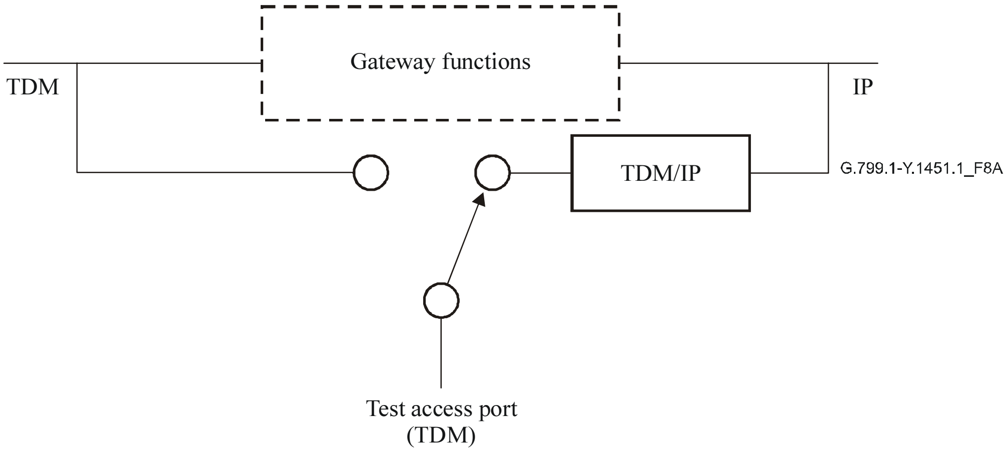


Figure 8a/G.799.1/Y.1451.1 – Intrusive test access – TDM test access port

In the case where the Test Access Port is IP (e.g., Figure 8b), a TDM/IP conversion function is required to access the TDM data port or stream.

The specific point within the TDM path where access is placed shall be prior to any speech processing functions.

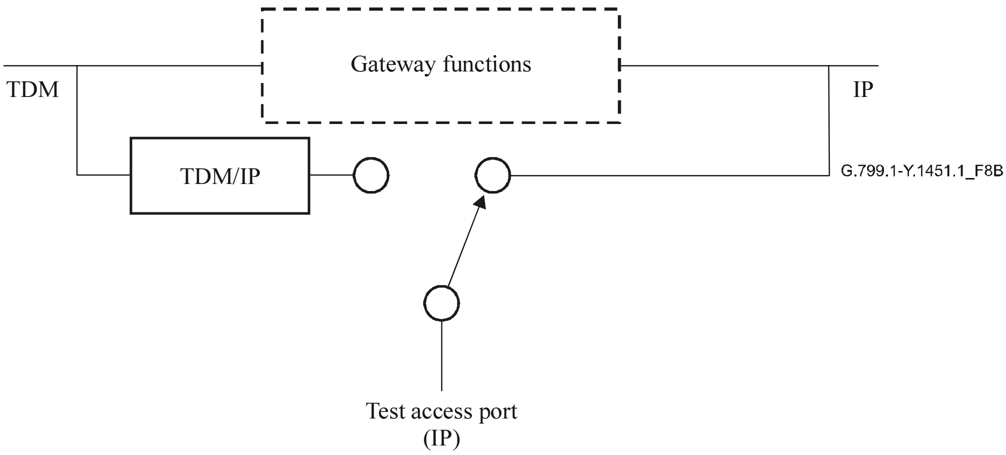


Figure 8b/G.799.1/Y.1451.1 – Intrusive test access – IP test access port

If TDM is provided, the interface shall be E1/T1 (G.703, G.704) and signalling shall conform to 7.2. If an IP output is provided, the interface shall be as specified within 7.1 (G.799.1/Y.1451.1) and signalling shall be accomplished via H.248/Megaco signalling messages sent to/from the TIGIN gateway, as with any other IP port.

In the case of SS7 signalling arrangements, the signalling to set up calls is done via the H.248/Megaco path to the MGC.

The functions provided by the management system should not overlap with those provided by per-call control.

Appendix I

Overview of the location of TIGIN gateways in end-to-end connections

Figure I.1 shows a functional block diagram of transport network equipment for interconnecting GSTN terminals and VoIP terminals (hereinafter abbreviated TIGIN). The MGC is present although, for simplicity, is not shown in Figure I.1.

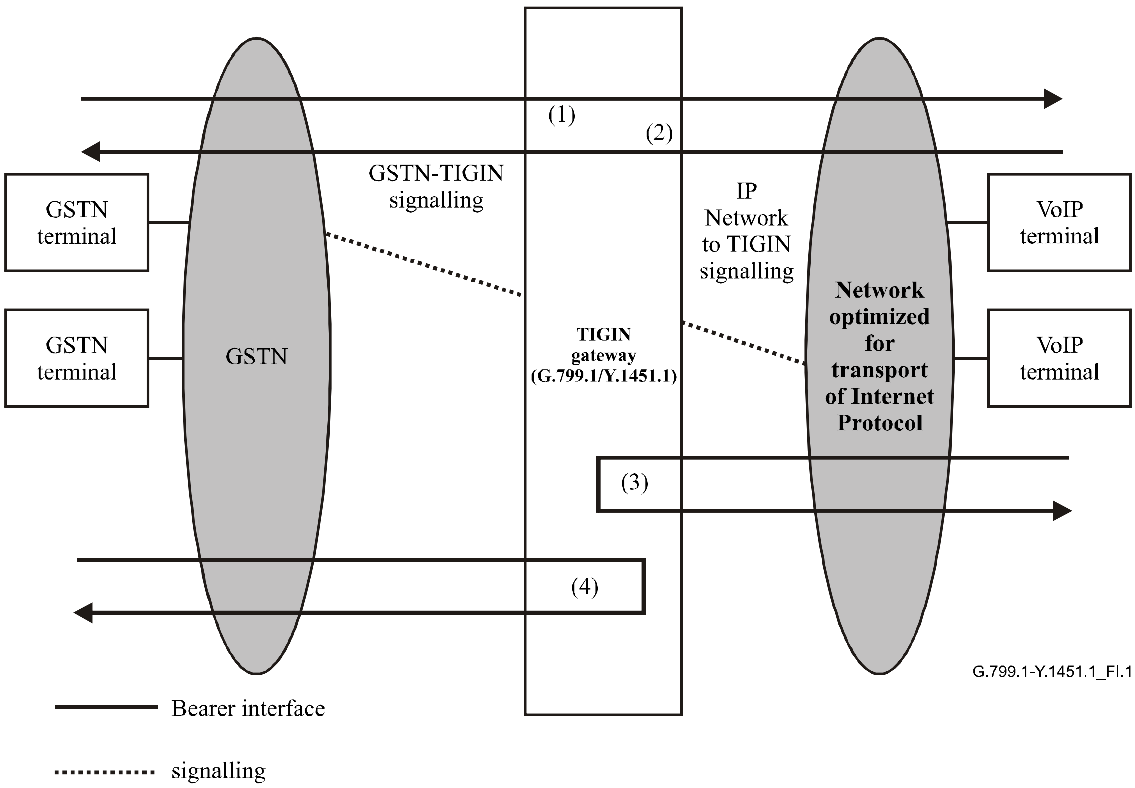


Figure I.1/G.799.1/Y.1451.1 – Location of TIGIN and possible connection types of interest in this Recommendation

This appendix describes the four connection types illustrated in Figure I.1.

The terminals attached to the Internet are assumed to have H.323 functionality from the point of view of voice and voiceband data transmission. These terminals may be connected to the IP network via a direct connection (e.g., Ethernet, Token Ring, etc.) or a dial-up connection (e.g., modem and PPP link). The IP and GSTN sections are interconnected through a GSTN/IP gateway, called a TIGIN. For convenience, this gateway is designated with a single box in Figure I.1.

The specific functions of the gateway will depend on whether the direction of transmission is from the IP Network to the GSTN or vice versa. In particular, the functions in the gateway include (but are not limited to):

Call originating on the IP network and terminating on GSTN

- Reception, interpretation and generation of relevant signalling messages;
- Packet disassembly (including "IP stack");
- Speech decoder (including error concealment, comfort noise, silence insertion, etc.);
- Management or regulation of delay variation;
- Echo cancellation.

Call originating on the GSTN and terminating on IP network

- Reception, interpretation and generation of relevant signalling messages;
- Speech encoder (including silence removal, comfort noise decisions, etc.);
- Packet assembly (including "IP stack");
- Management or regulation of delay variation;
- Echo cancellation.

Four connection arrangements are considered in this Recommendation, each of which is shown in Figure I.1. They are:

1) GSTN terminal → VoIP terminal (GSTN terminal → GSTN → TIGIN → IP network → VoIP terminal);
2) VoIP terminal → GSTN terminal (VoIP terminal → IP network → TIGIN → GSTN → GSTN terminal);
3) VoIP terminal → VoIP terminal (VoIP terminal → IP network → TIGIN → IP network → VoIP terminal);
4) GSTN terminal → GSTN terminal (GSTN terminal → GSTN → TIGIN → GSTN → GSTN terminal).

Each of these four connection arrangements requires the use of one gateway. Hence, connections that are purely GSTN-based or are strictly VoIP-to-VoIP using only the Internet are not the subject of this Recommendation. Situations in which multiple or tandem gateways are used are covered by combinations of connections 1 through 4 (e.g., a combination of 1 and 2 would be GSTN terminal → GSTN → TIGIN #1 → IP network → TIGIN #2 → GSTN → GSTN terminal). More complex arrangements may be envisaged but it is not considered necessary to describe them in detail as they can be made up of these above cases.

Appendix II

Further information on performance/interworking and synchronization

II.1 Performance/interworking

Performance/interworking issues include:

a) Ability to vary the packet size on a per call basis.

   Voice samples are accumulated and buffered into a packet, and a packet is sent when it becomes full. The size of the packet can be controlled by PCC messages, but the network must ensure that they are the same on both ends of a connection. In addition, the capability to vary the packet size on a per call basis must be supported by the PCC protocol, and is outside the scope of this Recommendation.

b) Silence suppression/comfort noise generation when used in conjunction with codecs that do not describe them.

c) Real-time control protocol (RTCP).

d) Packet loss mitigation scheme.

e) Jitter buffer size.

   Variation in the arrival of packets into the TIGIN (packet delay variation) must be accommodated by a jitter buffer. The size of the jitter buffer must be selected based on the quality goals for the network. In addition, the type of data being transported may affect the selection of the jitter buffer size. For example, voice transmission is tolerant to packet loss but not to excessive delay, whereas voiceband data or data is tolerant to delay but not to packet loss.

f) Transmission delay.

   Delay through the TIGIN should be minimized. In addition, the TIGIN should be tolerant to the overall delay produced by the network.

II.2 RTP timestamps

As described in IETF RFC 3550, the randomly initialized RTP timestamp is a 32-bit field that must be updated such that it reflects the sampling instance of the first octet in the RTP data packet. Consequently, there is a direct relation between the rate of the TDM clock at the sending TIGIN and the rate of the RTP packets being filled up, sent over the IP network, and the rate of their arrival at the receiving TIGIN. At the receive end, the arrival of an RTP packet could be timestamped based on the TDM clock of the circuit switch domain interfacing to the local (receiving) TIGIN.

The difference between the packet arrival times and the difference between RTP packet timestamps could be algorithmically processed to extract information for adjusting the local TDM clock (see IETF RFC 2833).

Appendix III

TIGIN equipment arrangements

The basic interworking arrangements between IP networks and GSTN networks are shown in Figure III.1. Note that the figures shown in this appendix are only examples and are not exhaustive. The MGC is present although, for simplicity, is not shown in the figures below.

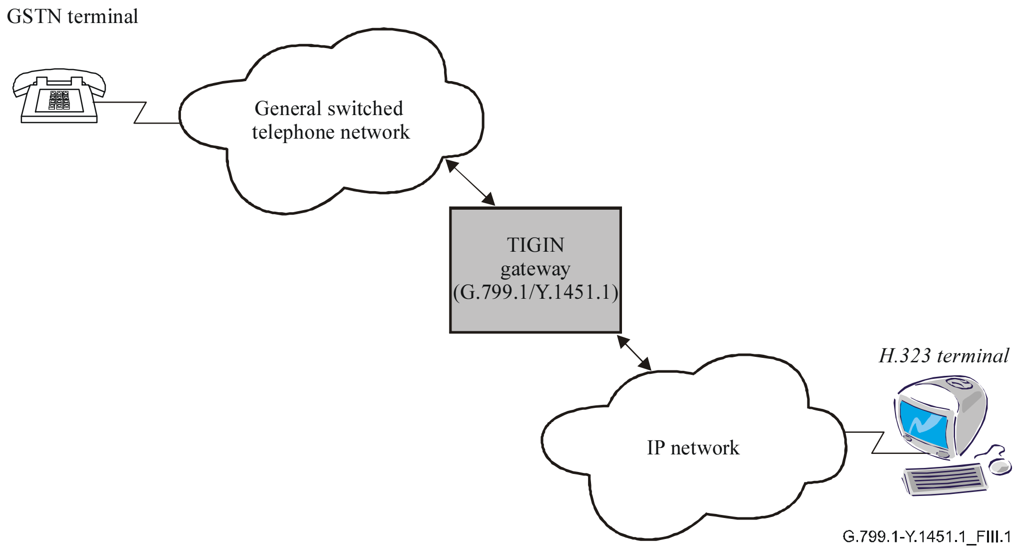


Figure III.1/G.799.1/Y.1451.1 – GSTN terminal interworking with H.323 terminal

Other cases may exist as shown in Figures III.2 and III.3.

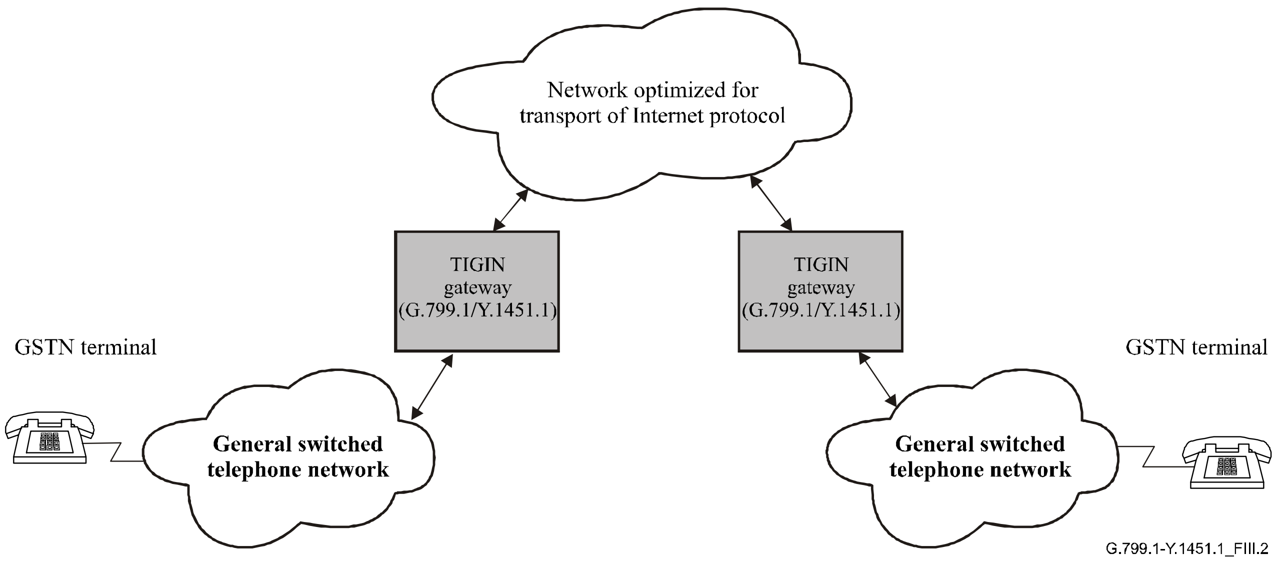


Figure III.2/G.799.1/Y.1451.1 – GSTN Terminals interconnected through network optimized for transport of IP

or

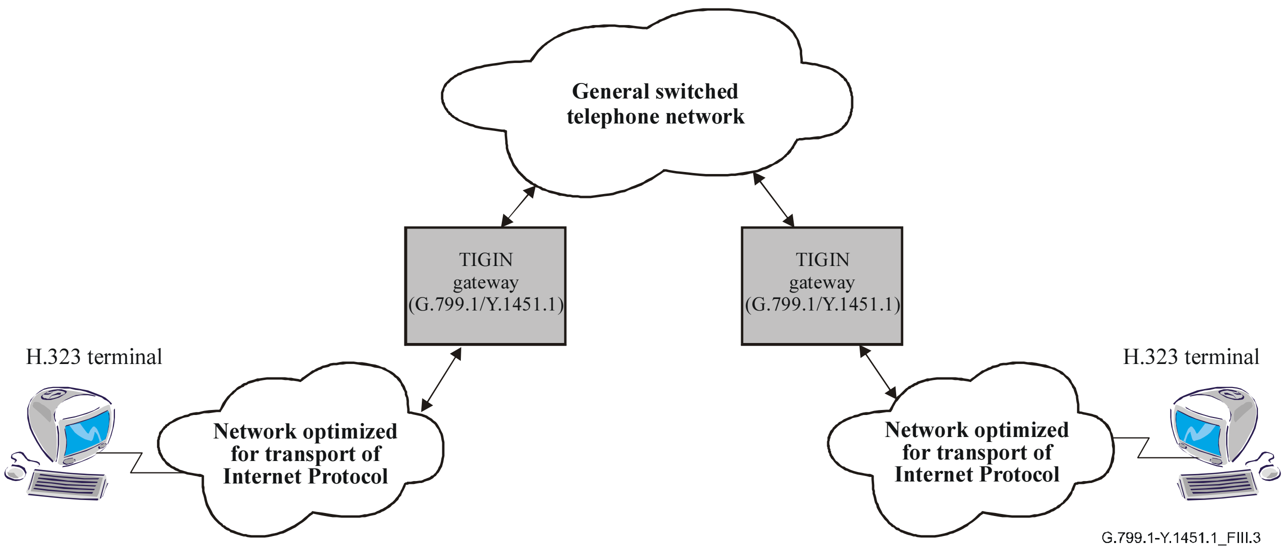


Figure III.3/G.799.1/Y.1451.1 – H.323 terminals interconnected through GSTN network

In the latter two cases, the interworking functions will remain the same so will not be dealt with further.

Appendix IV

Mapping of E-model (G.107) parameters from IETF RFC 3611 metrics and determining R-factor

IV.1 The E-model

The E-model (ITU-T Rec. G.107) was developed as a planning tool, but may, however, be used to estimate transmission quality based on measured network parameters. The E-model defines a method for determining an R-factor, which may be translated into an estimated Mean Opinion Score (MOS). The E-model was developed under the assumption that parameters were stable. However, some parameters, notably packet loss rate, may vary considerably during a call. Extensions to the E-model, such as ETSI TS 101 329-5 Annex E, have been developed in order to permit time-varying parameters to be incorporated into calculated transmission quality values.

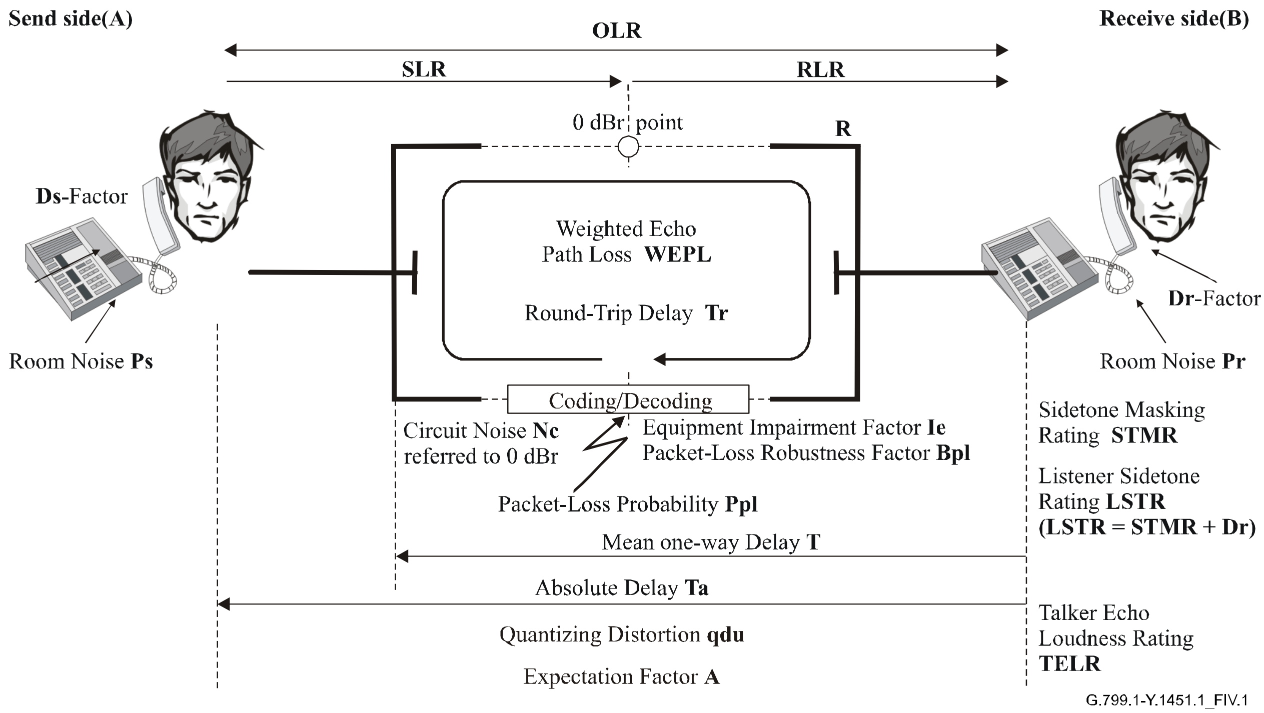


Figure IV.1/G.799.1/Y.1451.1 – G.107 Reference Model (figure from ITU-T Rec. G.107)

Figure IV.1 is taken from ITU-T Rec. G.107 and is incorporated into this appendix to facilitate the following description of the mappings between RFC 3611 parameters and E-model parameters. In the scenario shown, an R-factor is being determined for the "Receive Side", and hence the diagram would be reversed for the other direction of communication. To facilitate the explanation below, the send side will be denoted A and the receive side denoted B, and hence an R-factor is being determined for the B end of the connection (in this case gateway "B"). The E-model parameters below are all "B" side parameters unless otherwise noted.

It should also be noted that ITU-T Rec. G.108 provides more detailed guidance on the application of the E-model and also makes use of the same "A" and "B" notations for the two endpoints.

Some factors, for example the handset Ds, Dr and STMR parameters, RLR and Pr are generally not observable or measurable at a trunking gateway and, hence, should be set to known values for the specific network or G.107 default values.

IV.2 Parameters related to delay

ITU-T Rec. G.107 specifies three delay parameters:

– Ta the mean end-to-end one way delay;
– T the mean one way delay from the receive side to the point at which talker echo would occur if the receive side were talking; and
– Tr the round trip delay between sources of echo on the send and receive sides.

IETF RFC 3611 specifies two delay parameters, the RTCP round trip delay and the end system delay, denoted T_{RTCP} and T_{ESD}(A or B) below.

Gateway B may measure or estimate T_{RTCP} and T_{ESD}(B) and may receive T_{ESD}(A) in an RTCP XR message from the remote gateway A. In addition, gateway B may determine the local echo path one way delay T_{EPD}(B) from the gateway's echo canceller, however, it should be noted that this delay differs from the G.107 T parameter as it represents only the echo path delay over the PCM or analog path.

The E-model parameters may be estimated from the parameters known to gateway B as follows – (Note that in ITU-T Rec. G.107, Ta is shown in Figure 1 as "absolute delay", however, is referred to in Table 2/G.107 and 8.7.3.1 m) fourth bullet, as "absolute delay in echo-free connections"):

$$\mathrm{Ta} = \frac{\left(\mathrm{T_{RTCP}} + \mathrm{T_{ESD}}(\mathrm{A}) + \mathrm{T_{ESD}}(\mathrm{B}) + \mathrm{T_{EPD}}(\mathrm{B}) \text{.........} + \mathrm{T_{EPD}}(\mathrm{A}) \text{if known}\right)}{2}$$

NOTE – If an RTCP XR message has not been received from gateway A, then assume that

$$\mathrm{T_{ESD}}(\mathrm{A}) = \mathrm{T_{ESD}}(\mathrm{B})$$

T is approximately equal to Ta

Tr is approximately equal to 2 × T

IV.3 Parameters related to packet loss

ITU-T Rec. G.107 provides a formula for calculating the effective Ie (Ie-eff) factor under conditions of independent (non-bursty) packet loss using the Packet Loss Probability (Ppl) and the Packet-loss Robustness Factor (Bpl). ITU-T Rec. G.107 omits to mention that the packet loss rate should be measured after the de-jitter buffer in the end system i.e., should use the probability of *lost and discarded* packets. The base (zero packet loss) Ie factor and packet-loss robustness factor is tabulated in Appendix I/G.113 for a range of common codec types.

ETSI TS 101 329-5 Annex E defines algorithms for measuring the distribution of lost and discarded packets, and for calculating Ie-eff in the presence of time-varying (bursty) packet loss and discard. This approach has been shown to provide improved correlation with subjective test data for time-varying conditions (e.g., correlation coefficient of about 0.84 instead of 0.71).

ETSI TS 101 329-5 Annex E requires that Ie-eff be separately calculated for the burst and gap condition, giving Ie-eff(burst) and Ie-eff(gap), and then combined to give a single Ie-eff value. The Bpl method defined in ITU-T Rec. G.107 for mapping packet loss rate to Ie-eff should be used to determine Ie-eff(gap) and Ie-eff(burst) based respectively on the packet loss rate in gaps and bursts.

IETF RFC 3611 incorporates the TS 101 329-5 Annex E algorithm for packet loss distribution measurement and hence the Burst Loss Density, and Gap Loss Density can be input directly to the ETSI TS 101 329-5 Annex E Ie-eff(burst) and Ie-eff(gap) calculations and used to produce an overall Ie-eff factor.

a) *Approach based on bursty/time varying packet loss rate*

Ppl (burst) = Burst Loss Density

Ppl (gap) = Gap Loss Density

Ie-eff(burst) calculated using Ppl (burst) and Bpl for Codec

Ie-eff(gap) calculated using Ppl (burst) and Bpl for Codec

Ie-eff calculated using ETSI TS 101 329-5 Annex E (section E.7.1)

If the Ie-eff factor is calculated based only on average packet loss and discard rate, then the G.107 Ppl parameter can be calculated as the weighted average of the RTCP XR gap and burst loss densities.

b) *Approach based on assumed independent/constant packet loss rate*

$$Ppl = \frac{(Burst\ \ Loss\ Density \times Burst\ Duration + Gap\ Loss\ Density \times Gap\ Duration\ )}{(Burst\ Duration + Gap\ Duration)}$$

Ie-eff calculated using Ppl and Bpl for codec.

IV.4 Parameters related to noise

ITU-T Rec. G.107 defines the noise level as the sum of the noise powers for circuit and room noise. If no information on noise level is measured or available, then the default G.107 parameters should be used.

IETF RFC 3611 defines noise level as the received noise level during silence periods expressed in dBm0. This corresponds to the sum of the sending end room noise and circuit noise, and may be directly input into the E-model as parameter No(B).

IV.5 Parameters related to signal level and loudness

The remote end G.107 Send Loudness Rating SLR(A) can be estimated from the locally measured IETF RFC 3611 signal level (B) parameter, and the local SLR(B) value from the signal level (A), reported via a received RFC 3611 message.

SLR(A) = –15 – Signal Level (B) .. direct input to R-factor calculation

SLR(B) = –15 – Signal Level (A) .. used to determine TELR

IV.6 Parameters related to Echo

The G.107 talker echo loudness rating is the sum of the remote echo return loss and local send and receive loudness ratings.

$$TELR(B) = SLR(B) + ERL(A) + RLR(B)$$

The echo return loss value should be the estimated ERL **after** the effects of the echo canceller (defined as "Residual" echo return loss in IETF RFC 3611) from the remote gateway, and should be the value reported via a received RFC 3611 message. If the "residual echo return loss" value(s) are not reported via the RFC 3611 messages, then the G.107 default value for TELR should be used.

IV.7 Calculating R, MOS-LQ and MOS-CQ values

ITU-T Rec. G.107 defines the R-factor as:

$$R = Ro - Is - Id - Ie + A$$

Where Ro is primarily related to the basic signal-to-noise ratio, Is represents impairments that occur simultaneously with speech, Id represents impairments that are delayed with respect to speech, Ie represents the equipment impairment factor and A the advantage factor.

R is a *conversational* quality metric, and can be translated to an estimated conversational quality MOS score (MOS-CQ in IETF RFC 3611) using the R to MOS conversion function in ITU-T Rec. G.107. By omitting the Id parameter, an estimated listening quality MOS score (MOS-LQ in IETF RFC 3611) can be determined.

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ITU-T Y-SERIES RECOMMENDATIONS

GLOBAL INFORMATION INFRASTRUCTURE, INTERNET PROTOCOL ASPECTS AND NEXT GENERATION NETWORKS

GLOBAL INFORMATION INFRASTRUCTURE	
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Network aspects	Y.300–Y.399
Interfaces and protocols	Y.400–Y.499
Numbering, addressing and naming	Y.500–Y.599
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