

# TS 101 319 V1.6.4 (1998-12)

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*Technical Specification*

**Telecommunications and Internet Protocol  
Harmonization Over Networks (TIPHON);  
Signalling for basic calls from an H.323 terminal to a terminal  
in a Switched-Circuit Network (SCN)**

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Pursuant to the ETSI IPR Policy, no investigation, including IPR searches, has been carried out by ETSI. No guarantee can be given as to the existence of other IPRs not referenced in SR 000 314 (or the updates on the ETSI Web server) which are, or may be, or may become, essential to the present document.

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## Foreword

This Technical Specification (TS) has been produced by ETSI Project Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON).

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# 1 Scope

The present document describes interactions between ITU-T Recommendation H.323 [10] Terminals, Gatekeepers, Gateways, and Switched-Circuit Networks (SCN) for support of TIPHON scenario 1, see TR 101 306 [18].

The present document specifies a profile to be applied to the ITU-T Recommendation H.323 [10] for the purposes of TIPHON compliant systems.

Call trace protocols, e.g. for use in tracing malicious calls, are outside the scope of the present document.

The present document is applicable to Basic Call for voice only telephone calls from an H.323 Terminal on a network using Internet Protocol transport to Terminals on SCNs, e.g. Public Switched Telephone Network (PSTN), Integrated Services Digital Network (ISDN) or GSM Public Land Mobile Network (PLMN).

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# 2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies.
- A non-specific reference to an ETS shall also be taken to refer to later versions published as an EN with the same number.

- [1] EN 300 092-1: "Integrated Services Digital Network (ISDN); Calling Line Identification Presentation (CLIP) supplementary service; Digital Subscriber Signalling System No. one (DSS1) protocol".
- [2] Void.
- [3] EN 300 403-1: "Integrated Services Digital Network (ISDN); Digital Subscriber Signalling System No. one (DSS1) protocol; Signalling network layer for circuit-mode basic call control; Part 1: Protocol specification [ITU-T Recommendation Q.931 (1993), modified]".
- [4] ETS 300 659-1 (1997): "Public Switched Telephone Network (PSTN); Subscriber line protocol over the local loop for display (and related) services; Part 1: On hook data transmission".
- [5] ISO/IEC 11572 (1996): "Information Technology - Telecommunications and information exchange between systems - Private Integrated Services Network - Circuit mode bearer services - Inter-exchange signalling procedures and protocol".
- [6] ITU-T Recommendation H.225.0 (1998): "Call signalling protocols and media stream packetization for packet based multimedia communication systems".
- [7] ITU-T Recommendation H.235 (1998): "Security and encryption for H Series (H.323 and other H.245 based) multimedia terminals".
- [8] ITU-T Recommendation H.245 (1998): "Control protocol for multimedia communication".
- [9] Void.
- [10] ITU-T Recommendation H.323 (1998): "Packet-based multimedia communications systems".
- [11] ITU-T Recommendation Q.731.3 (1993): "Calling Line Identification Presentation (CLIP)".
- [12] ITU-T Recommendation Q.761 (1997): "Signalling System No. 7 – ISDN User Part functional description".

- [13] ITU-T Recommendation Q.762 (1993): "Specifications of Signalling System No. 7; General Function of Messages and Signals of the ISDN User Part of Signalling System No. 7".
- [14] ITU-T Recommendation Q.763 (1993): "Specifications of Signalling System No. 7; Formats and Codes of the ISDN User Part of Signalling System No. 7".
- [15] ITU-T Recommendation Q.764 (1993): "Specifications of Signalling System No. 7; Signalling System No. 7 - ISDN User Part Signalling Procedures".
- [16] ITU-T Recommendation Q.931 (1993): "ISDN user-network interface layer 3 specification for basic call control".
- [17] ITU-T Recommendation Q.932 (1993): "Digital Subscriber Signalling System No. 1 (DSS1) – Generic procedures for the control of ISDN supplementary services".
- [18] TR 101 306 (1998): "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON); Requirements for service interoperability; Scenario 1".
- [19] TS 101 312 (1998): "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON); Network architecture and reference configurations; Scenario 1".

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## 3 Definitions and abbreviations

### 3.1 Definitions

For the purposes of the present document, the following terms and definitions apply:

**access token:** An octet string which may be present in Registration, Admission, and Status (RAS) messages and the Setup message. It is used to specify the identity, authorization, or other security characteristics of a network element or a user.

**call:** See "telephone call".

**endpoint:** An H.323 Terminal or Gateway. An endpoint can call and be called. It generates and/or terminates information streams.

**gatekeeper:** The Gatekeeper is an H.323 entity on the network that provides address translation and controls access to the network for H.323 Terminals, Gateways and Multipoint Control Units (MCU). The Gatekeeper may also provide other services to the H.323 Terminals, Gateways and MCUs such as bandwidth management and locating Gateways.

**gateway:** An H.323 Gateway is an endpoint on the network which provides for real-time, two-way communications between H.323 Terminals on the packet based network and other Terminals on a switched circuit network.

**H.323 Terminal:** An entity which provides audio and optionally video and data communications capability in point-to-point or multipoint conferences in packet-based networks.

**telephone call:** Two-way speech communication between two users by means of Terminals via network infrastructure.

## 3.2 Abbreviations

In addition to the abbreviations given in TR 101 306 [18] and TS 101 312 [19], the following abbreviations apply:

ACF	AdmissionConfirm (RAS message)
ARJ	AdmissionReject (RAS message)
ARQ	AdmissionRequest (RAS message)
CLI	Calling Line Identification
CLIP	Calling Line Identification Presentation
DSS1	Digital Subscriber Signalling System No. one
DTMF	Dual Tone Multi Frequency
GSM	Global System for Mobile communication
IN	Intelligent Network
IP	Internet Protocol
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part (of CCITT number 7 signalling)
MCU	Multipoint Control Units
NNI	Network-to-Network Interface
PLMN	Public Land Mobile Network
PSTN	Public Switched Telephone Network
RAS	Registration, Admission, and Status
RCF	RegistrationConfirm (RAS message)
RRQ	RegistrationRequest (RAS message)
SCN	Switched-Circuit Network
UNI	User-to-Network Interface

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## 4 Entities participating in the setup of a basic call

The TIPHON system is a multi-component environment system. It consists of H.323 Terminals, Gateways, and Gatekeepers. For more details about these entities and the scenarios in which they are acting, refer to TS 101 312 [19].

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## 5 Registration, call setup and call release

### 5.1 Overview

The procedures of ITU-T Recommendations H.323 [10], H.225.0 [6] and H.245 [8] shall apply with the limitations or extensions described in the present document. The procedures of ITU-T Recommendations H.235 [7] shall apply unchanged.

In accordance with ITU-T Recommendation H.323 [10], the protocols defined in ITU-T Recommendations H.225.0 [6], H.245 [8] shall be used as the protocols for call control signalling and logical channel controlling within the packet based data network. The use of ITU-T Recommendations H.225.0 [6] and H.245 [8] is specified in more detail in the following subclauses.

NOTE: ITU-T Recommendation H.225.0 defines two protocols: Registration, Admission, and Status (RAS) and a ITU-T Recommendations Q.931-like protocol.

Annex A describes changes to the ITU-T Recommendations Q.931 [16] messages and the contents of messages carried by the User-to-user information element in ITU-T Recommendation H.225.0 [6].

The RAS protocol shall apply unchanged.

The ASN.1 definitions given in annex H of ITU-T Recommendation H.225.0 [6] shall be used.



The precise translation of ITU-T Recommendation H.323 [10] messages to the SCN shall depend on the nature of the SCN (an ISDN, a PSTN, etc.) and the level of connectivity to be supported. This translation shall provide support, as necessary, of:

- the semantics of the protocol specified in ITU-T Recommendations Q.761 [12], Q.762 [13], Q.763 [14], Q.764 [15] and Q.931 [16] with variations in implementation appropriate to the region in which the SCN operates; or
- the protocol specified in ISO/IEC 11572 [5].

## 5.2 Security principles

TIPHON compliant systems shall support security services. They are not required to always be activated.

## 5.3 Registration

### 5.3.1 Registration basics

Both H.323 Terminals and Gateways shall register with Gatekeepers. Registration shall be in accordance with ITU-T Recommendation H.323 [10] procedures.

NOTE: Gateway performance may be enhanced if the Gatekeeper issues the **PreGrantedARQ** indication in the RegistrationConfirm (RCF) message on successful registration of the Gateway.

Two types of registration shall be supported:

- authenticated registration with the Gatekeeper;
- anonymous registration with the Gatekeeper.

### 5.3.2 Authenticated registration

Authenticated registrations shall be in accordance with the security profiles in annex B.

### 5.3.3 Anonymous registration

In this case the RRQ message shall not contain an access token.

NOTE: The intention here is to allow certain types of call, e.g. Freephone. The types of call are service provider dependent.

### 5.3.4 Registration keep alive

Endpoints shall support the keep alive procedure as specified in subclauses 7.2.2 and 8.4.2 of ITU-T Recommendation H.323 [10]. If the registration ceases to be valid (e.g. the connection between the H.323 Terminal and the Gatekeeper goes down for some reason), the Gatekeeper shall remove the registration and release all ongoing calls. A new registration may be established using the procedures of ITU-T Recommendation H.323 [10].

## 5.4 Call setup

### 5.4.1 Prerequisites

Calls between users in the IP network and users in the SCN shall be setup only after successful registration according to subclause 5.3.

The sequences in annex C are built on the principles described in the following subclauses.

### 5.4.2 The en-bloc procedure

If a CALL PROC message is returned to the H.323 Terminal, the "Sending complete" information element shall be inserted, if not already there, in the SETUP message towards the next network element (e.g. next Gatekeeper or a Gateway).

### 5.4.3 Overlap sending

On receipt of such a SETUP message, the Gatekeeper starts timer T302 (the value of timer T302 is specified in ITU-T Recommendation Q.931 [16]) and sends a SETUP ACKNOWLEDGE message to the H.323 Terminal.

The Gatekeeper shall restart timer T302 on the receipt of every INFORMATION message not containing a sending complete indication and containing the called party information element with at least one valid character.

### 5.4.4 Establishment of media channels

#### 5.4.4.1 Fast Connect procedure

TIPHON-compliant systems shall use the Fast Connect procedure of ITU-T Recommendations H.323 [10], subclause 8.1.7.

NOTE 1: This includes the capability to negotiate media channels using ITU-T Recommendation H.245 or to fall back to ITU-T Recommendation H.245 signalling at any time of the call.

NOTE 2: This enables in-band information to be passed prior to call establishment.

#### 5.4.4.2 Encapsulation of H.245 messages within H.225.0 messages

TIPHON-compliant systems shall support encapsulation of ITU-T Recommendation H.245 [8] messages within ITU-T Recommendation H.225.0 [6] messages according to ITU-T Recommendations H.323 [10], subclause 8.2.1.

NOTE: Encapsulation of H.245 messages within H.225.0 is preferred to a separate H.245 channel because of its greater efficiency.

### 5.4.5 Dual Tone Multi Frequency (DTMF) signalling

Transfer of the DTMF tones shall be by means of the ITU-T Recommendation H.245 [8] message **userInputIndication**.

NOTE: The **userInputIndication** could be tunnelled via the FACILITY message if necessary.

## 5.4.6 Basic call setup

Calls shall be setup using the procedures defined in ITU-T Recommendations H.323 [10] with the following changes/clarifications:

- within the context of this specification, call setup uses only one user channel towards the SCN. Calls requiring a number of user channels will not be supported;
- the Gatekeeper and the Gateway shall support both the en-bloc and the overlap sending procedure.

NOTE 1: Annex C, "Message flows for call setup", gives one possible mapping of messages to and from the SCN for overlap sending with fast connect.

If an element or a message not allowed to be used within the context of this specification is received, the receiver shall pass on, but otherwise ignore, the message or the element i.e. the receiver shall act as if the message or the element was not received.

If a element is received with a value, not allowed within the context of this specification, the receiver shall, if the element is optional, pass on, but otherwise ignore, the element (act as if the element is not received) or if the element is mandatory act as if the default value was received.

NOTE 2: The security policy of an operator's network or the security policy implemented in a network element may override the error handling as described above.

## 5.4.7 Carrier selection

Carrier selection shall be performed by transmitting the necessary information to the Gatekeeper and Gateway within the called party number field. This procedure shall not preclude overlap sending.

## 5.5 Active phase

### 5.5.1 General aspects

The active phase of the call shall commence when the called party answers and the Gateway issues the **Connect** message as a result.

Upon detection of failure of the call in the IP network, accounting and charging functions shall be informed.

NOTE 1: In order to detect call failures in the IP network, which affect accounting and charging functions, Gatekeepers should specify an **irrFrequency** value inside the AdmissionConfirm (ACF) message.

Transfer of the DTMF tones shall be by means of the ITU-T Recommendation H.245 [8] message **userInputIndication**.

NOTE 2: The **userInputIndication** could be tunnelled via the FACILITY message if necessary.

### 5.5.2 Exceptional cases during the active phase

If the Gatekeeper detects a failure, the Gatekeeper shall initiate clearing of the call as described in subclause 5.6.

If the Gateway detects a failure, the Gateway shall initiate clearing towards the SCN, and clear the call in the IP network as described in subclause 5.6, towards the IP network.

If the H.323 Terminal detects a failure, the H.323 Terminal shall initiate clearing of the call as described in subclause 5.6.

## 5.6 Call release

Call release may be initiated by the H.323 Terminal, the Gateway, or the Gatekeeper. The reason for initiating call release may, e.g. be normal disconnection of a call or a call failure.

Message collisions shall be handled by H.323 Terminals, Gatekeepers and Gateways.

NOTE: Message collision occurs when the endpoint and the Gateway initiate clearing at the same time.

## 5.7 Calling Line Identification (CLI)

CLI information may be provided by calling users and may be received by called users. When a user wishes to provide a calling number it shall be provided using the signalling elements described in ITU-T Recommendation H.225.0 [6].

Users may provide a number using the optional Calling Party Number information element in the Setup message.

NOTE: The procedures and protocols for handling these information elements may be defined in regional and national regulations and/or codes of practice. Wherever such requirements exist they may supersede the requirements expressed here.

According to ITU-T Recommendation H.225.0 [6] the number cannot have qualifications defined in accordance with octet 3a information (Presentation Indicator and Screening Indicator) as in tables 4 to 11 of ITU-T Recommendation Q.931 [16]. Accordingly, in the absence of octet 3a information, the calling party number information element shall be treated as if octet 3a contained the following values:

- 1) "presentation allowed"; and
- 2) "user-provided not screened".

## Annex A (normative): Required support of ITU-T Recommendation H.225.0 call control protocol

ITU-T Recommendation H.323 [16], uses a call control protocol derived from ITU-T Recommendations Q.931 [16] which is specified in the ITU-T Recommendation H.225.0 [6].

This annex defines a profile which is intended to clarify the use of ITU-T Recommendation H.225.0 [6] in TIPHON-compliant systems. Whenever the contents of this annex are in conflict with ITU-T Recommendation H.225.0 [6] this annex shall take precedence.

The current version of the ITU-T Recommendations Q.931 [16] part of ITU-T Recommendation H.225.0 [6] includes a number of optional messages and information elements. The EN 300 403-1 [3] standard forbids some of the information elements which are optional in ITU-T Recommendation H.225.0 [6], listed below. Gateways that interconnect with ETSI DSS1 networks shall not send these information elements.

Table A.1 identifies changes to entries in table 4 of ITU-T Recommendation H.225.0 [6].

The changes in the table shall apply to TIPHON Phase 1 systems.

**Table A.1: Modification to ITU-T Recommendation H.225.0 [6] and Usage of Q.931 [16] and Q.932 [17] Messages**

Call Establishment Messages	Transmit (M, CM)	Receive and act on (M, CM)
Call Proceeding	M	M
Progress	M	M
Setup Acknowledge	M	CM (see note)
Miscellaneous Messages		
Information	CM	M
NOTE : Only mandatory if the H.323 Terminal indicates <b>canOverlapSend</b> in the Setup message. M = mandatory, CM = conditionally mandatory.		

## A.1 Message definitions

### A.1.1 Alerting

Table A.2 identifies changes to entries in table 5 of ITU-T Recommendation H.225.0 [6].

**Table A.2: Modifications to ITU-T Recommendation H.225.0 [6] Usage of the Alerting message**

Information element	H.225.0 status	Length in H.225.0
Signal	F (see note)	NA
NOTE : F = forbidden Na = not applicable		

## A.1.2 Information

Table A.3 identifies changes to entries in table 9 of ITU-T Recommendation H.225.0 [6].

**Table A.3: Modifications to ITU-T Recommendation H.225.0 [6] Usage of the Information message**

Information element	H.225.0 status	Length in H.225.0
Signal	F	NA

## A.1.3 Release Complete

Table A.4 identifies changes to entries in table 10 of ITU-T Recommendation H.225.0 [6].

**Table A.4: Modifications to ITU-T Recommendation H.225.0 [6] Usage of the Release Complete message**

Information element	H.225.0 status	Length in H.225.0
Signal	F	NA

## A.1.4 Setup

Table A.5 identifies changes to entries in table 11 of ITU-T Recommendation H.225.0 [6].

**Table A.5: Modifications to ITU-T Recommendation H.225.0 [6] Usage of the Setup message**

Information element	H.225.0 status	Length in H.225.0
Signal	F	NA

## A.1.5 Setup Acknowledge

The contents and semantics of a SETUP ACKNOWLEDGE message received from the network are defined in tables 3 to 16 of ITU-T Recommendations Q.931 [16] with the single modification that the Signal information element is not allowed.

## Annex B (normative): Interoperable security profiles

Compliant systems shall provide security services according to one or more of the following interoperable security profiles. These profiles shall identify their purpose, and they shall specify the method, cryptographic algorithms, and cryptographic parameters (e.g. key lengths) for:

- the five security services (authentication, access control, non-repudiation, confidentiality, and integrity);
- four or more protocol components (RAS, H.225.0, H.245, RTP);
- the security information flows identified in TS 101 312 [19] (S1-S17).

As a convenient reference, profiles may include a summary matrix derived from the following form for each security information flow.

**Table B.1: Profile summary matrix**

Security Services	Call Functions				
	RAS	H.225.0	H.245	RTP	Other(s)
Authentication					
Access Control					
Non-Repudiation					
Confidentiality					
Integrity					

Each element in the matrix shall identify the security mechanism (e.g. Not applicable, None, IPSEC, TLS/SSL, Access Token, H.235, or Other) and the cryptographic algorithm(s) and parameters supported.

In addition, each profile description shall include the detailed information sufficient to ensure interoperability, and lists the attacks it counters, the provided security level, and the potential consequences of a breach in its security.

If a profile includes multiple tables (e.g. for multiple security information flows), then the security mechanisms specified for each service shall not contradict each other.

NOTE: All security profiles are referred to by their clause number in this annex (e.g. "Profile B.1") rather than a descriptive term.

### B.1 Security Profile B.1

Security profile B.1 provides no security services whatsoever. All elements in the summary matrix list a security mechanism of "None." It is appropriate for networks where security is not provided or not permitted or provided by other means (e.g. physical isolation).

**Table B.2: Security Profile B.1**

Security Services	Call Functions				
	RAS	H.225.0	H.245	RTP	
Authentication	None	None	None	None	None
Authorization	None	None	None	None	None
Non-Repudiation	None	None	None	None	None
Privacy	None	None	None	None	None
Integrity	None	None	None	None	None

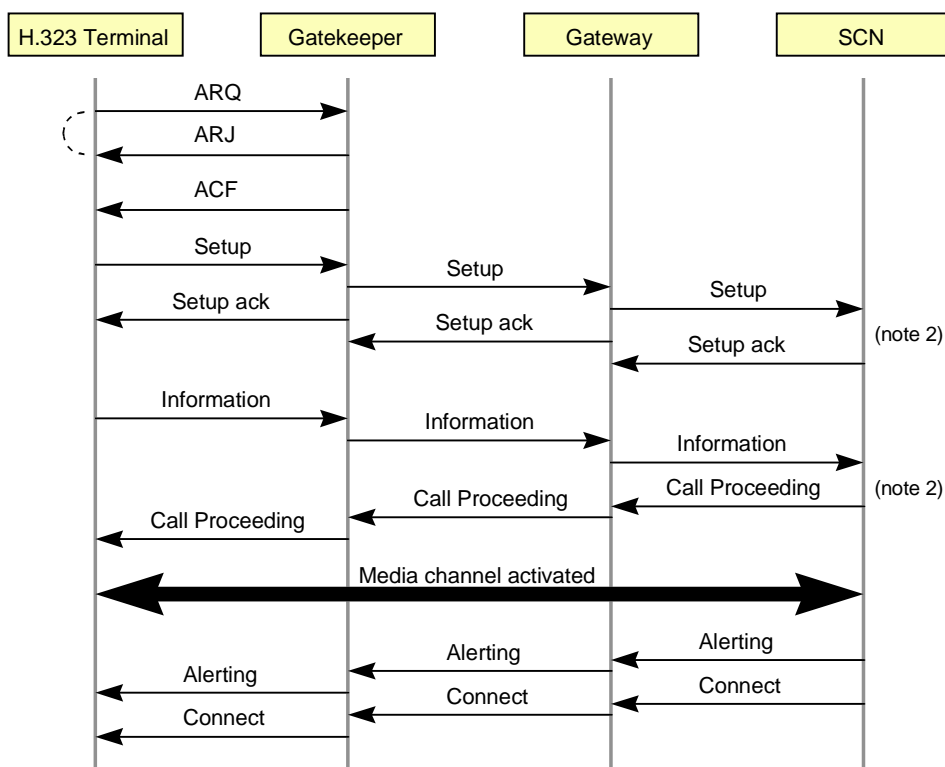
Additional profiles will be defined in the future.

## Annex C (informative): Message flows for call setup

### C.1 Gatekeeper routed calls

The signalling procedures, described in the following subclause, are based on the Gatekeeper routed model.

#### C.1.1 H.323 Terminal and Gateway registered to same Gatekeeper, fast connect procedure, overlap procedure



NOTE 1: This figure is given only for informative purposes and reflects ITU-T Recommendation Q.931 [16] cases only.

NOTE 2: If the SCN has received sufficient digits to complete number analysis, Call Proceeding shall be sent instead of Setup Ack in response to Setup and the Information message will be discarded.

NOTE 3: In this flow diagram the ARQ/ACF procedure between Gateway and Gatekeeper is not performed since preGrantedARQ is assumed.

**Figure C.1: Fast call setup using overlap sending and Gatekeeper routed call model within a Zone**



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## Annex D (informative): Basic call scenario definitions

### D.1 Successful calls

A successful call is one in which the SCN receives the called address and routes the call to a network Terminal which can accept two way speech communication. The network signalling will normally have sent an Answer signal back from the terminating exchange. The generally understood meaning of this is that the calling user is able to speak to a user handling calls directed at the called address or to some automatic call answering device. The normal expectation is that the caller will be required to pay for such calls.

In addition to the simple meaning above there are cases, notably associated with mobile telephones, where network operators send back an answer signal even though two way speech has not been established. The normal expectation is that the caller will be required to pay for such calls.

---

### D.2 Unsuccessful calls

Unsuccessful calls do not reach the two way or one way speech condition which follows Answer. The reasons for this are for example:

- caller clears prior to ringing;
- caller clears prior to answer;
- called party is engaged on another call;
- called party is not accepting calls;
- called party does not answer;
- network congestion is encountered.

In all of these cases the user is not offered the service associated with a successful call but there is no fault in the network.

---

### D.3 Call failures

A call failure occurs when the network fails to complete a call and is therefore neither successful or unsuccessful. A call failure also occurs when an established call, whether or not answer has occurred, is broken down as a result of equipment failure. The normal expectation is that customers do not pay for calls after the point at which the fault is detected.

---

### D.4 Calling line identification (CLI)

The following subclauses deal with two kinds of calling line identity. The general purpose is to ensure that a CLI is offered to the SCN for use by the terminating network. At present there is no definition of a mechanism for withholding CLI information derived by authentication or given as a presentation number. The behaviour of a TIPHON Gateway is that the presentation indicator is set to Presentation Allowed in the SETUP and IAM messages. If a Gateway does not have any number as a result of authentication or supplied presentation number the network number alone is supplied. The presentation indicator for network number is set to Presentation Restricted. In the case where no network number is available for whatever reason the value of the indicator is set to Number not available due to inter-working.

Thus when a user either supplies a presentation number or arranges for the network number to be displayed the relevant presentation indicators are set to presentation allowed.

## D.4.1 Presentation number

If a user wants the called party to receive a presentation number it can be achieved by putting a suitable E.164 number in the Calling party number parameter of the SETUP message. If the calling user has a special arrangement with the service provider, as described in EN 300 092-1 [1]. The type of number field may be a "subscriber number", "national number" or "international number" as specified in EN 300 092-1 [1] with code point values defined in subclause 4.5.10 of EN 300 403-1 [3].

The Gatekeeper may use the number supplied by the H.323 Terminal as well as the identification of the user determined by the authentication process. If the number appears to be a valid one, the user is authorized to provide presentation numbers and the interface from the Gateway is an NNI, the number can be forwarded to the public network as "User provided, verified, and passed". If the interface is a UNI, the number will be passed to the network in exactly the same form as it was supplied in the ITU-T Recommendation H.225.0 [6] SETUP message. This number will be converted into a presentation number if the ISDN exchange allows the facility.

There is another way of generating a CLI to send to the called party. The user need not provide a number in the calling party field instead the Gatekeeper accesses a presentation number database indexed by the identification derived from the authentication operation. If the connection forwards into the narrow-band network is ISUP the number may be passed as "user provided, verified, and passed". This behaviour is specified in the Stage 3 Description for CLIP in ISUP, ITU-T Recommendation Q.731.3 [11]. If the interface to the narrow-band network is a UNI the calling party field is passed on in the same form as it was supplied by the user. The range of CLI values which can be passed over such an interface will make verification by the public network impossible. The UNI should therefore be registered with the service provider as one which uses the special arrangement described in subclause 6.2 of the CLIR protocol specification, EN 300 092-1 [1]. The consequence of this is that all calling numbers passed over this interface will be delivered as "User provided, not verified".

Whichever way it is generated the calling number is passed over the network to the terminating exchange where it is passed to the called party. If the called party is connected using ISDN the calling number information is passed unchanged in accordance with EN 300 403-1 [3]. For PSTN lines with the CLI display service the number is sent in accordance with ETS 300 659-1 [4] for on hook signalling. In ISDN the calling party number information element is passed on unchanged to the called party only if he subscribes to the CLIP service. In the case of an analogue with caller display the number is always delivered but the qualification, see below, of the calling number is not necessarily sent. This is because the qualification parameter is not in a normative part of the specification. Nevertheless the qualification type values are identical to the ISDN case, i.e.:

- 00: User provided not verified.
- 01: User provided verified, and passed.
- 10: User provided verified, and failed.
- 11: Network provided.

Numbers in the category "00: User provided, not verified" convey the meaning that the recipient shall treat the number as unreliable. If the called party does not want to receive calls from uncertain numbers he need not answer the call. Where the interface from the Gateway is an NNI the and authentication allows it the number may be given as either verified and passed, for a limited range of presentation numbers. Since this range of uncertainty is already conveyed it is not necessary to generate a new category solely for internet use. There is no greater uncertainty about an internet telephony number than about one which is user provided, verified and failed. The issue of whether that extra information should be displayed on analogue display devices is a matter for national regulators.

## D.4.2 Network number

For a normal telephone call the Network number is usually the same as the calling party number. In some cases the caller will choose not to release that number to the called party, nevertheless the callers number is transmitted to the terminating exchange with an indicator that it is not to be released. Certain customer lines, such as emergency services, are provided with an override facility which allows the calling number to be released irrespective of the caller's preference. The protocol used and the means of display is a national matter.

When a presentation number is provided either by the user or by a Gatekeeper the type of number is not set to network number unless the actual identity of the user is verified and can be treated with the same certainty as an equivalent fixed network calling number. The purpose of this subclause is to describe how to find a calling party number which can be used by law enforcement agencies in order to commence investigations in the same way as for a normal fixed network call.

There are a number of different configurations which need to be considered but the main issue is that of trust. For the scheme to meet its objective there shall be trust between the parties involved in the call which is equivalent to that between network operators providing calling line identities in today's network. The configurations include:

- directly connected IP links to the Gateway provider;
- indirectly connected permanent IP links to the Gateway provider;
- dial-in links to the Gateway provider;
- dial-in links to an indirectly connected Gateway provider;
- connections where the other providers act only third parties conveying IP traffic as if via a private data switch.

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## D.5 In-band information

The SCN may provide in-band information before the CONNECT message is sent. In-band information can be sent in the form of tones or announcements.

Some reasons for in-band information are as follows:

- information to the calling user that the SCN telephone is ringing;
- SCN wants to give information to the calling user about the progress of the call, e.g. "You have now dialled the police please disconnect or wait and the call will be set-up";
- progress tone is sent to avoid silence while routing over low speed routes;
- the called party number has changed and the new number is announced (changed number interception service);
- different types of interception services giving announcements;
- different types of interception services routing the call to an operator;
- prompt the calling user for more information. This case applies for some IN services and it takes place before the active phase.

To allow in-band information from the SCN to be transported to the terminal, the audio channel may be set-up before the SETUP message is sent to the SCN.

## D.5.1 Call to operator/Non-chargeable calls

If the user places a call to an operator who is connected to the SCN, or if the call is routed to an operator because of the invocation of a service, it might happen that the operator does not return a B-answer, i.e. there will be no CONNECT message sent from the SCN to the gateway. The reason for this behaviour is that the operator does not want to start charging.

## D.5.2 IN Services

The user shall be able to utilize IN services within the SCN. Some of the IN services are simple number translation services while other IN services require user interaction.

When user interaction is required the IN node prompts the user, using in-band information, and the user answers, using DTMF tones.

The IN node may operate in two different modes:

- answered mode, i.e. the IN node sends a B-answer towards the calling user and then continues with the interactive part;
- non-answered mode, i.e. the IN node requests the SCN to establish the voice channels in both directions and then continues with the interactive mode.

The answered mode will be experienced by the Gateway as a normal call while the non-answered mode will be experienced as a progress indicator No. 8 "*In-band information or appropriate pattern is available*" in a CALL PROC or a PROGRESS message.

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## Bibliography

The following material, though not specifically referenced in the body of the present document or not yet publicly available, gives supporting information.

TR 101 300: "Telecommunications and Internet Protocol Harmonization Over Network (TIPHON); Description of technical issues".

EN 300 356: "Integrated Services Digital Network (ISDN); Signalling System No. 7; ISDN User Part (ISUP) version 3 for the international interface".

ITU-T Recommendation H.246 (1998): "Interworking of H-Series multimedia terminals with H-Series multimedia terminals and voice/voiceband terminals on GSTN and ISDN".

ITU-T Recommendation Q.731 (1993): "Stage 3 description for number identification supplementary services using Signalling System No. 7".

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## History

<b>Document history</b>		
V1.6.4	December 1998	Publication