SLOVENSKI STANDARD
SIST-V ETSI/EG 202 090 V1.1.2:2005
01-januar-2005

Širokopasovno digitalno omrežje z integriranimi storitvami (B-ISDN) – Medobratovalnost signalizacij H.323/B-ISDN

Broadband Integrated Services Digital Network (B-ISDN); H.323/B-ISDN signalling interoperability

Ta slovenski standard je istoveten z: EG 202 090 Version 1.1.2

ICS:

33.080	Digitalno omrežje z integriranimi storitvami (ISDN)	Integrated Services Digital Network (ISDN)

SIST-V ETSI/EG 202 090 V1.1.2:2005 en

EG 202 090 V1.1.2 (1999-03)

ETSI Guide

Broadband Integrated Services Digital Network (B-ISDN); H.323/B-ISDN signalling interoperability

Reference

DEG/SPS-05173 (jco00ide.PDF)

Keywords

ATM, audio, B-ISDN, broadband, interoperability, IP, ISDN, multimedia, speech, video

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Foreword

This ETSI Guide (EG) has been produced by ETSI Technical Committee Signalling Protocols and Switching (SPS).

1 Scope

The present document is to investigate the interoperability of H.323 and B-ISDN signalling environments with a view to identifying possible standardization activities in this area. The following aspects of interoperability are investigated:

- the interworking of H.323 and B-ISDN environments, both separated and combined, via a gateway;
- the connection of remote H.323 environments via B-ISDN;
- the use of H.323 annex C (H.323 on ATM), including the co-existence of H.323 annex C and B-ISDN signalling in the same network and possible future enhancement of annex C;
- the application of the B-ISDN separated call control protocol to gatekeeper-to-gatekeeper communication;
- the impact of H.323 on future mobility protocols.

For each of the above, due consideration is given to the following: addressing impact (including use of IP addressing); conferencing models; T.120 data impact; basic call procedures; generic functional procedures (GFP) and supplementary services; end-to-end resource and session level protocols.

Possible future standardization activities resulting from the present document will include H.323 work items as well as B-ISDN signalling work items.

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, subsequent revisions do apply.
- 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] ECMA-155 (1997): "Private Integrated Services Networks - Addressing".

[2] ECMA-254 (1996): "B-PISN - Inter-Exchange Signalling Protocol - Generic Functional Protocol (B-QSIG-GF)".

[3] ATM Forum P-NNI V1.0 (1996): "Private Network-Network Interface Specification Version 1.0 (af-pnni-0055.000)".

[4] ATM Forum UNI 4.0 (1996): "ATM User-Network Interface (UNI) Signalling Specification Version 4.0 (af-sig-0061.000)".

[5] Void.

[6] TR 101 300 (V1.1): "Telecommunications and Internet Protocol Harmonization Over Network (TIPHON); Description of technical issues".

[7] TS 101 313: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON); Network architecture and reference configurations; Scenario 2".

[8] TS 101 324 (V1.1): "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON); Naming and addressing; Scenario 1".

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[10] IETF RFC 1577 (1994): "Classical IP and ARP over ATM".

[11] ISO/IEC 13247 (1997): "Information technology - Telecommunications and information exchange between systems - Broadband Private Integrated Services Network - Inter-exchange signalling protocol - Basic call/connection control".

[12] ISO/IEC 11571: "Information technology - Telecommunications and information exchange between systems - Numbering and sub-addressing in private integrated services networks".

[13] ITU-T Recommendation H.225.0 (1998): "Call signalling protocols and media stream packetization for packet based multimedia communication systems".

[14] ITU-T Recommendation H.245 (1998): "Control protocol for multimedia communication".

[15] ITU-T Recommendation H.323 (1998): "Packet based multimedia communication systems".

[16] ITU-T Recommendation H.450.1 (1998): "Generic functional protocol for the support of supplementary services in H.323".

[17] ITU-T Recommendation H.450.2 (1998): "Call transfer supplementary service for H.323".

[18] ITU-T Recommendation H.450.3 (1998): "Call diversion supplementary service for H.323".

[19] ITU-T Recommendation Q.2726.1 (1996): "B-ISDN user part - ATM end system address".

[20] ITU-T Recommendation Q.2931 (1995): "Digital Subscriber Signalling system No. 2 (DSS 2) - User-Network Interface (UNI) Layer 3 specification for basic call/connection control".

[21] ITU-T Recommendation Q.2932.1 (1996): "Digital subscriber signalling system No. 2 - Generic functional protocol: Core functions".

[22] ITU-T Recommendation Q.931 (1993): "ISDN user-network interface layer 3 specification for basic call control".

[23] ITU-T Recommendation T.120 (1996): "Data protocols for multimedia conferencing".

[24] Void.

[25] Void.

[26] Void.

[27] ITU-T Recommendation H.235: "Security and encryption for H-Series (H.323 and other H.245-based) multimedia terminals".

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

[29] ITU-T Recommendation G.723.1: "Dual rate speech coder for multimedia communications transmitting at 5.3 and 6.3 kbit/s".

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

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

[32] ITU-T Recommendation H.261: "Video codec for audiovisual services at p x 64 kbit/s".

[33] ITU-T Recommendation H.263: "Video coding for low bit rate communication".

[34] ITU-T Recommendation E.164: "The international public telecommunication numbering plan".

[35] ITU-T Recommendation Q. 2761: "Functional description of the B-ISDN user part (B-ISUP) of signalling system No. 7".

[36] ITU-T Recommendation Q. 2762: "General Functions of messages and signals of the B-ISDN user part (B-ISUP) of Signalling System No. 7".

[37] ITU-T Recommendation Q. 2763: "Signalling System No. 7 B-ISDN User Part (B-ISUP) - Formats and codes".

[38] ITU-T Recommendation Q. 2764: "Signalling System No. 7 B-ISDN User Part (B-ISUP) - Basic call procedures".

[39] ITU-T Recommendation H.246: "Interworking of H-Series multimedia terminals with H-Series multimedia terminals and voice/voiceband terminals on GSTN and ISDN".

[40] ATM Forum UNI 3.1: "ATM User-Network Interface Specification V3.1".

[41] ITU-T Recommendation E.191: "B-ISDN numbering and addressing".

3 Abbreviations

For the purposes of the present document, the following abbreviations apply:

AESA	ATM End System Address
AFI	Authority and Format Identifier
ATM	Asynchronous Transfer Mode
BC	Bearer Control
B-HLI	Broadband6 High Layer Information
B-ISDN	Broadband-ISDN
B-ISUP	Broadband-ISDN User Part
B-PISN	Broadband-PISN
CC	Call Control
CN	Core Network
DCC	Designated Country Code
GFP	Generic Functional Protocol
GFT	Generic Functional Transport
GSM	Global System Mobile
GSTN	General Switched Telephony Network
ICD	International Code Designator
IETF	International Engineering Task Force
IMT	International Mobile Telecommunications
IN	Intelligent Network
IP	Internet Protocol
ISDN	Integrated Services Digital Network
LAN	Local Area Network
LANE	LAN Emulation
MC	Multipoint Controller
MCU	Multipoint Control Unit
MGCP	Media Gateway Control Protocol
MIB	Management Information Base
MP	Multipoint Processor
MPLS	Multi-Protocol Label Switching
MPOA	Multi-Protocol Over ATM
MT	Mobile Terminal
NAVDEC	Network Access Server and Voice on IP Device Control
NSAP	Network Service Access Point
PISN	Private Integrated Services Network
PNNI	Private Network-to-Network Interface
PNP	Private Numbering Plan
PUM	Personal User Mobility
PVC	Permanent Virtual Circuit
QoS	Quality of Service
RAN	Radio Access Network

RAS	Registration, Admission and Status
RMOA	Real-time Multimedia Over ATM
RTCP	Real-Time Control Protocol
RTP	Real-Time Protocol
SCN	Switched Circuit Network
SDP	Session Description Protocol
SIP	Session-Initiated Protocol
SNMP	Simple Network Management Protocol
SS7	Signalling System N°7
SVC	Switched Virtual Circuit
TCP	Transmission Control Protocol
TE	Terminal Equipment
TIPHON	Telecommunications and Internet Protocol Harmonization Over Networks
UDP	User Datagram Protocol
UIM	User Identity Module
UNI	User-to-Network Interface
UPT	Universal Personal Telephony
URL	Universal Resource Locator
VHE	Virtual Home Environment

4 Introduction

4.1 Overview of H.323 and related recommendations

4.1.1 H.323 concept and entities

H.323 specifies multimedia conferencing over packet networks. This sentence already contains the keywords for noting major characteristics of H.323:

- *Multimedia:* The media mix in a call can consist of audio, video and data streams. Audio communication has to be supported, video and data are optional. Media can be added, dropped or replaced dynamically during a call.

NOTE 1: A telefax service is being drafted for the next version of H.323.

- *Conference:* Even a two-party call is considered a special case of a multiparty conference. With regard to audio or video, multipoint conferencing can be centralized (media processing is done by a central multipoint processor, MP), decentralized (media are multicast directly from the sending party to all receiving parties), or mixed (combination of the two). Hybrid conferences combining centralized audio with decentralized video, or vice versa, are possible. Control information is always centralized via a Multipoint Controller (MC). Data traffic is also distributed centrally.

- *Packet network:* Although H.323 is independent of the specific transport substructure, its main sphere seems to be TCP/IP based networks (LANs/intranets or internet). The network provides a transparent transport service between communicating entities, i.e. the entities communicate end-to-end without any "exchange" or "switch" between them.

H.323 entities are terminals, gateways, gatekeepers, MCs, MPs and multipoint control units (MCUs). An MCU consists of an MC and optionally one or more MPs.

Terminals, gateways and MCUs are endpoints, which can place and accept calls (i.e. they are "callable").

A gatekeeper is not an endpoint and is "addressable" but not callable. It performs for endpoints in its "zone" tasks such as address resolution, admission control and bandwidth control.

MCs and MPs are neither addressable nor callable, but are part of an endpoint or gatekeeper which is addressable/callable.

H.323 uses the terms *call* and *conference* in the following way:

- A call is the point-to-point multimedia communication between two H.323 endpoints, either direct or via gatekeeper(s) and/or MC(s). In case of interworking, the H.323 call consists of the section between the gateway and another H.323 endpoint.
- A point-to-point conference is the multimedia communication between two terminals irrespective of their location. In case of interworking, it contains a section between the gateway and another H.323 endpoint, and also a section in the other network.
- A multiparty conference is the multimedia communication between three or more terminals irrespective of their location, but including at least one MC. A point-to-point conference becomes a multiparty conference by adding parties, and vice versa by dropping parties.

NOTE 2: The term 'conference' in H.323 corresponds to the term 'session' in the B-ISDN information model.

4.1.2 Channels defined in H.323

H.323 uses the concept of channels to structure the information exchange between communicating entities. The following channels are part of the communication process:

- *Call signalling channel*. This reliable channel carries information for call control and supplementary service control. The Q.931-like protocol used over this channel is specified in H.225.0 and in H.450.x (for supplementary services).
- *H.245 control channel*. This reliable channel carries the H.245 protocol for media control.
- *RAS channel*. This unreliable channel provides for communication between an endpoint and its gatekeeper. The RAS (Registration, Admission and Status) protocol is specified in H.225.0.
- *Logical channels for media*. Usually each real-time medium is carried in a separate pair of uni-directional unreliable channels, one for each direction, using the RTP and RTCP protocols. A call with audio and video therefore involves at least four logical channels. Data traffic, however, uses a bi-directional reliable channel and a protocol stack according to T.120.

In this list, a 'reliable channel' means connection-mode transport, while 'unreliable' refers to connectionless transport. In an IP-based scenario, this corresponds to TCP and UDP, respectively.

Annex C of H.323 provides a means of carrying real-time media information (audio or video) over an ATM SVC instead of over UDP/IP when an ATM path exists endstation-to-endstation. This allows the real-time media to take advantage of the QoS guarantees that ATM provides. Other H.323-related protocols are still carried over TCP/IP or UDP/IP, using an IP/ATM (e.g. MPLS, MPOA, LANE, RFC 1577, RFC 1483) technique. If audio or video is carried over ATM according to annex C of H.323, a bi-directional logical channel may be used instead of a pair of uni-directional channels, thereby resulting in a single SVC used bi-directionally instead of two SVCs used uni-directionally.

4.1.3 Protocols and related standards

H.323 itself does not specify protocols but normatively refers to a number of other ITU-T Recommendations H.225.0, H.245, H.235, H.450.x, T.120 etc.

H.225.0 covers three areas:

- *Call signalling*. A protocol loosely based on Q.931 is specified for call establishment. Its main purpose is to obtain a transport address for an H.245 connection. Except in the fast connect case (see 4.1.5), where no H.245 connection exists, a call is not cleared when a call signalling connection terminates. Call clearing is normally done by H.245 signalling. However, for the sake of supplementary service usage, the H.225.0 call signalling connection may be kept until call release.

- *RAS protocol.* This protocol between endpoint and gatekeeper provides the following major functions:
  - *Gatekeeper discovery*. Enables an endpoint to find its responsible gatekeeper.
  - *Endpoint Registration/Unregistration*. A terminal or gateway registers with a gatekeeper to enable participation in calls. An endpoint may unregister if it is not to take part in further calls.
  - *Admission*. Enables a terminal or gateway to get permission to set up or accept a call.
  - *Bandwidth changes*. Instructs or permits an endpoint to change the bandwidth for an existing call.
  - *Endpoint Location*. Locates endpoints based on their alias addresses and returns a transport address (IP address + port number) where they can be called.
  - *Disengage*. Informs the gatekeeper of call release or instructs the endpoint to terminate a call.
  - *Status.* Keeps the gatekeeper informed of the status of a call.
- *RTP/RTCP*. H.225.0 specifies the use of the RTP and RTCP protocols for audio and video streams in an H.323 call.

H.245 operates between two endpoints or between an endpoint and an MCU and provides the following functionality:

- capabilities exchange;
- opening and closing of logical channels;
- mode requests;
- master-slave determination;
- flow control;
- call clearing;
- media loop.

Also there are various other H.245 messages that are indicated as applicable in annex A of H.323 but for which there are no H.323-specific detailed procedures. These include round trip delay determination (can be used as a heartbeat mechanism) and various conference commands, requests and indications.

H.235 deals with security issues for H.323 calls.

H.450.x series of recommendations specifies generic procedures (H.450.1) and specific protocols for supplementary services (currently call transfer and call diversion; several others in the drafting state).

T.120 is used as data conferencing protocol. A T.120 conference is treated as a logical channel with its own protocol suite.

Another important aspect is the coding of user plane information. A number of recommendations apply here, including G.711, G.723.1, G.728, and G.729 for audio coding and H.261 and H.263 for video coding.

4.1.4 Call procedures

This subclause summarizes the steps involved in a simple H.323 call where gatekeepers are in use. A terminal may participate in a call if it has registered with its associated gatekeeper. The actual procedures may be more complex than described here since there are various options for routeing the call control and H.245 control channels either directly between endpoints or via one or more gatekeepers. Also the adding or dropping of other parties is not considered here.

The calling terminal wishing to place a call may either know the transport address (IP address) or have an alias address of the destination (e.g. an E.164 number). In the latter case the terminal may ask the gatekeeper(s) for endpoint location, passing the alias address to the gatekeeper(s), which will return a transport address. This explicit location procedure allows multicasting of the request to many gatekeepers. Usually, however, the address resolution is implicit in the admission procedure between an endpoint and its associated gatekeeper, as described below.

The first step of call establishment is taken by the terminal by asking the gatekeeper for call admission, passing the destination (alias and/or transport) address to the gatekeeper. If granted admission, the terminal will send a SETUP message to the transport address returned by the gatekeeper. At the destination side, the called endpoint asks its gatekeeper for admission to accept the incoming call. If granted, the call will be accepted, and a transport address for the H.245 control channel is returned to the calling endpoint. The H.245 control channel is then established, and the further call related signalling is done via H.245: exchange of capabilities between calling and called endpoint, master-slave determination, opening and closing of logical channels etc. Once a logical channel is open, user data can be sent on it. During the lifetime of a call an endpoint and its associated gatekeeper may exchange status messages or bandwidth change messages. A call is cleared by closing all logical channels, sending a command to end the session (call) on the H.245 control channel and releasing the H.245 control channel. The call control channel can be released at any time after the H.245 control channel is established, at the latest at call clearing.

4.1.5 Fast connect

Fast connect is an option specified in 8.1.7 of H.323 that reduces the number of round trip delays involved in establishing a call and initial media streams by including H.245 open logical channel information in the SETUP and CONNECT messages. A separate H.245 channel using its own TCP connection may or may not subsequently be established, depending on the need for further channel opening and closing during the call. Fast connect is also mandated in the new H.323 annex F ("Single Use Device"), currently determined in Study Group 16.

Related to this is the use of UDP instead of TCP for signalling, as specified in new H.323 annex E. Both annex E and annex F will appear in H.323 version 3 shortly.

4.1.6 Inter-domain signalling

A new annex G to H.225.0, currently in a "determined" state in SG16, covers inter-domain address resolution aspects. The protocol operates between administrative domain border gatekeepers and between border gatekeepers and clearing houses. It allows an administrative domain to obtain routing addresses for calls to destinations in other domains.

4.2 Overview of modelling work relating to B-ISDN signalling

Work under Question 6 in ITU-T Study Group 11 has developed in document TRQ.2001an information model for future B-ISDN signalling capabilities. The information model comprises a number of object classes and associations. Object classes are grouped into four service levels as follows:

- session service level;
- resource service level;
- call service level;
- bearer service level.

The majority of the object classes involved are shown in figure x/TRQ.2001 reproduced in Figure 1.