Special Article on Mobile Multimedia Signal Processing Technologies
Audiovisual Terminal Technology

This article explains system configurations, multiplexing, terminal control and other terminal system technologies that enable audiovisual services under the International Mobile Telecommunications-2000 (IMT-2000) standard, in compliance with 3GPP videophone specifications. It also describes audiovisual terminal technologies in IP packet networks, which are becoming increasingly important, with reference to ITU and IETF standards.

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1. Introduction

Services based on the International Mobile Telecommunications-2000 (IMT-2000) standard will be launched in 2001. Among the wide range of multimedia services that are likely to become available on top of conventional audio communications, real-time interactive audiovisual services are expected to expand the scope of mobile phone calls. The possible implementation of audiovisual services over packet networks is also attracting much attention, following the rapid market penetration of the Internet and the integration of networks with Internet Protocol as a common protocol.

This article explains the various technologies that make real-time interactive audiovisual services possible, as well as the trends in relevant international standards. Chapter 2 describes the terminal technologies that enable audiovisual services through circuit switched transmission based on IMT-2000, with reference to standards established by the International Telecommunication Union (ITU) and the 3rd Generation Partnership Project (3GPP). Chapter 3 introduces the technologies that enable audiovisual services through packet switched networks, with reference to standards set by ITU and the Internet Engineering Task Force (IETF).

2. IMT-2000 Audiovisual Terminals

2.1 History of Standards

Figure 1 illustrates the history of international standards for audiovisual terminals. H.320 is the overview recommendation that specifies an audiovisual terminal for Narrowband Integrated Services Digital Network (N-ISDN) issued by the International Telecommunication Union-Telecommunication Standardization Sector (ITU-T) in 1990. The recommendation ensured interoperability between terminals from different vendors and contributed to the subsequent market penetration of videoconferencing and videotelephony. Later, ITU-T launched programs to develop a standard for terminals and systems dedicated to Broadband Aspects of Integrated Services Digital Network (B-ISDN), Public Switched Telephone Network (PSTN) and Internet Protocol (IP) network, and established standards H.310, H.324 and H.323, respectively, in 1996.

This was followed by the explosive growth of mobile communications and the progress of standardization efforts focusing on third-generation mobile communications, which prompted ITU-T to start studies on audiovisual terminals for mobile communications in 1995. Studies were conducted along with the enhancement of H.324, and the efforts crystallized in the form of Annex C to H.324 in February 1998. In the course of standardizing H.324 Annex C, functional enhancements concentrated on improving robustness against bit errors on the wireless channel.

As an IMT-2000 application, however, H.324 Annex C contains several specifications which are not optimal without modifications: it was originally designed as a general-purpose standard (i.e., it is not dedicated to a particular mobile communications technology), and was prescribed so as to maintain backward compatibility with H.324. Thus, 3GPP's CODEC working group decided to select the optimal Coder-Decoder (CODEC)s and the operation modes for IMT-2000 without any restriction, leading to the establishment of 3G-324M in December 1999. Visual phones, which are being

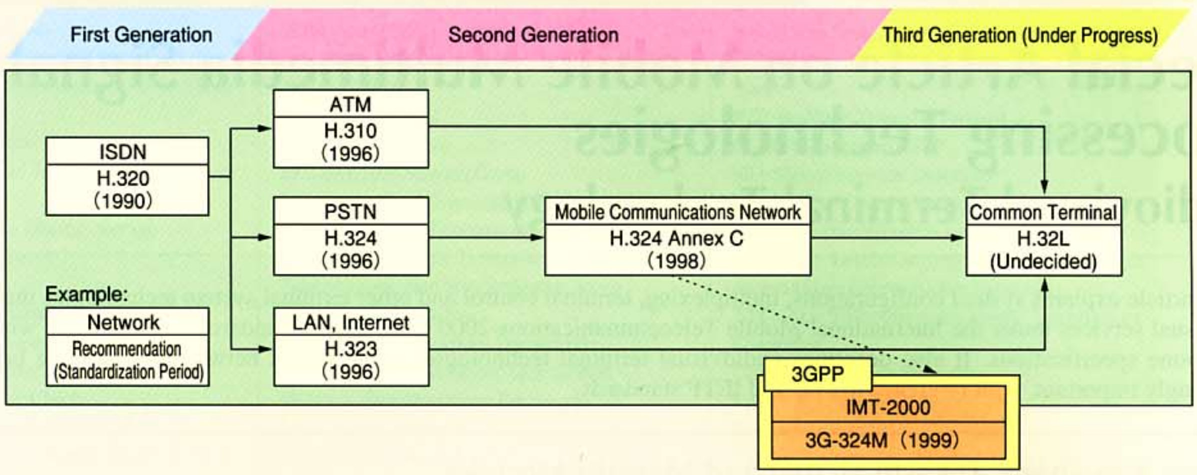


Figure 1 History of Audiovisual Terminal Standards

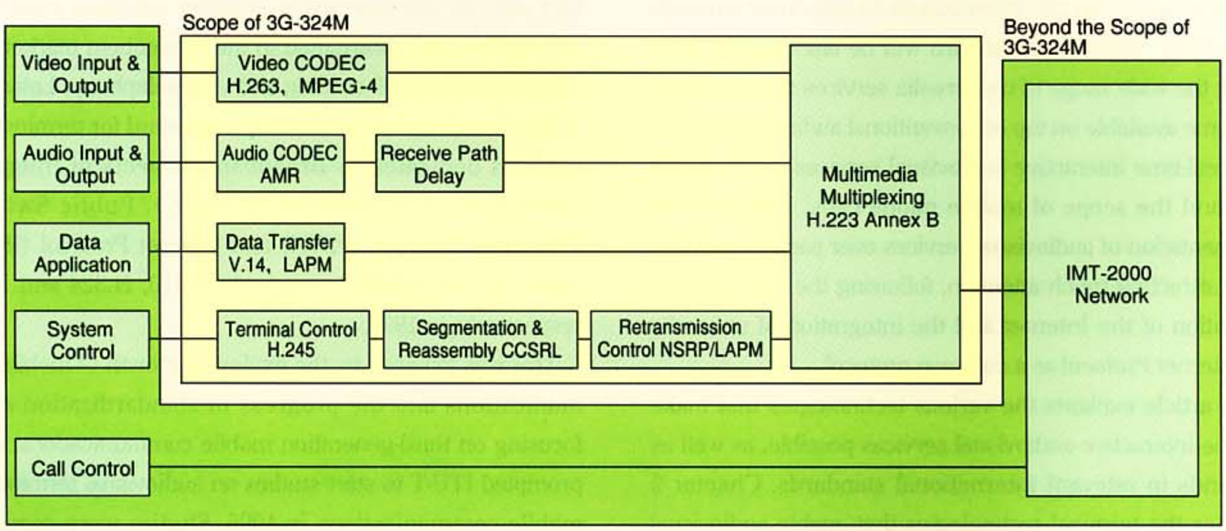


Figure 2 Terminal Configuration of 3G-324M

available from the initial stages of Wideband CDMA (W-CDMA), are in compliance with 3G-324M.

2.2 3G-324M Terminal Configuration

Figure 2 illustrates the 3G-324M terminal configuration and the scope of 3G-324M specifications. 3G-324M is an umbrella standard that consists of four major functional components: audio CODECs, video CODECs, control protocol for multimedia communications and multimedia multiplexing selected from standards from ITU-T and other standardization bodies.

Adaptive Multi Rate (AMR) and H.263 are mandatory CODECs for audio and video, respectively, whereas MPEG-4 (Moving Pictures Experts Group-4) is an optional but recommended CODEC for video. 3G-324M also mandates H.245 as a terminal control protocol for multimedia communications and requires H.223 Annex B support as a multiplexing protocol for mobile multimedia communications over an error-prone channel.

2.3 Media Coding

3G-324M terminals can use various media coding algorithms (CODEC)s through negotiation with the communications control procedures described. Nonetheless, the standard specifies the minimum requirements in regard to CODEC to ensure interoperability between terminals.

For audio CODEC, 3G-324M places considerable importance on the ease of installing the CODEC in the mobile terminal and therefore specifies AMR, the mandatory CODEC for basic audio services, as the mandatory CODEC. It also recommends to the installation of G.723.1, the mandatory CODEC in H.324, even though it is not mandatory for 3G-324M. For the technical specifications of each audio CODEC, please refer to the article titled Speech Coding Technology in this volume.

For image CODEC, 3G-324M specifies H.263 Baseline (i.e., excluding the optional functions) as the mandatory CODEC, as in the case of H.324. It also recommends the use of MPEG-4 Video while specifying its use in detail to tackle transmission errors, which is a phenomenon unique to mobile communications. For the technical specifications of each video CODEC, please refer to the article titled Video Coding Technology in this volume.

2.4 Multimedia Multiplexing

Audio, video, user data and control messages are mapped onto a bitstream through multimedia multiplexing (hereinafter referred to as "multiplexing") before transmission. At the receiver side, each set of information must be extracted properly from the incoming bitstream. Multiplexers must also provide transmission services according to each information type, such as Quality of Service (QoS) and framing.

H.223, a multiplexing protocol geared to H.324, satisfies these requirements with a two-layer approach: an adaptation layer and a multiplexing layer. In mobile communications, however, substantial error robustness is additionally required for multimedia multiplexing. To achieve this additional requirement, ITU-T enhanced H.223 for mobile multimedia communications in the course of standardizing H.324 Annex C.

As a result of the enhancement, the degree of error robustness can be chosen according to the properties of the transmission channel, by adding error-robust tools to H.223 one by one. Currently, 3 types of error-robust tool sets are defined, from level 1 to level 3. Levels 1, 2 and 3 are prescribed by H.223 Annex A, B and C, respectively. A terminal that supports a certain level must also support the lower levels in order to ensure interoperability. 3G-324M requires the support of level 2. The features of levels 0-2 are as follows.

(1) Level 0

This is the main unit of H.223.

Three adaptation layers are defined according to the type of the higher layer.

① AL1: For user data and control information. Error control is performed by the higher layer.

② AL2: For audio. Error detection and sequence numbering are applied.

③ AL3: For video. Error detection is applied. Sequence numbering and Automatic Repeat Request (ARQ) can be applied.

The multiplexing layer realizes functions of both Time Division Multiplexing (TDM) and packet multiplexing, in order to achieve high efficiency and low delay. The media with variable bitrates such as video, is mapped into the multiplexing frames using packet multiplexing, whereas media requiring low delay but with a fixed frame size, such as audio, is mapped using TDM.

To synchronize the multiplexing frame boundaries, an 8-bit High-level Data Link Control (HDLC) flag is used. And, "0" bit insertion is executed to avoid flag pattern emulation in the payload data, which means that the byte alignment cannot be maintained.

(2) Level 1

In order to improve the performance of frame synchronization in the multiplexing layer, the synchronization flag is replaced: from 8 bit HDLC flags to 16 bit Pseudo Noise (PN) sequence. Level 1 abolishes the "0" bit insertion to maintain the byte alignment.

(3) Level 2

On top of Level 1, a Payload Length field is added to the frame header, an error correction code is applied to improve the performance of frame synchronization and the error robustness of header information. Furthermore, an optional field can be added to improve the burst error robustness of header information.

2.5 Terminal Control

3G-324M adopts H.245 as its terminal control protocol, as with H.324. H.245 is commonly applied not only in H.324 but also in ITU-T multimedia terminal standards for various networks. One of the merits of using common control protocol is that it enables the construction of a gateway between different types of networks without much difficulty.

H.245 has the following functions.

(1) Master Slave Determination

H.245 determines each terminal to be the master or slave when communication begins. This information is used to resolve conflicts in succeeding procedures.

(2) Capability Exchange

Terminals at both ends exchange the supported capabilities in order to acquire information concerning acceptable transmission modes and coding modes.

(3) Logical Channel Signaling

H.245 opens logical channels for media transmission, setting their applicable parameters, or closes logical channels. It can also set the relationship between logical channels.

(4) Initialization and Alteration of Multiplexing Table

H.245 adds or deletes entries in the multiplexing table.

(5) Mode Request for Audio, Video and Data

H.245 can request the transmission mode of the other terminal.

(6) Detection of Round Trip Delay

H.245 can measure the round trip delay. This function can also be used to confirm whether or not the other terminal is working properly.

(7) Loop-back Test

(8) Command and Indication

H.245 can request the action of the encoder, request flow control and notify the protocol status.

In order to perform these functions, H.245 defines the messages to be sent and prescribes the control procedures according to these messages.

For the definition of messages, H.245 applies Abstract Syntax Notation 1 (ASN.1; ITU-T X.680 | ISO/IEC IS 8824-1), which ensures the high readability and extensibility of the definition, and for conversion into binary format, it adopts Packed Encoding Rules (PED; ITU-T X.691 | ISO/IEC IS 8825-2), in order to achieve highly efficient message transmission. It adopts Specification and Description Language (SDL) in its control procedures, prescribing status transitions both extensively and visually, including exceptional processing.

2.6 Mobile Multilink

One of the unique features of IMT-2000 is multicall, which is a function for making multiple calls at the same time. The quality of audiovisual communications can be improved by combining this function with multilink transmission, which bundles several physical channels and provides them as one logical channel. ITU-T standardized a mobile multilink operation of H.324 as H.324 Annex H in November 2000. This new annex defines the operation of H.324 over a maximum of 8 physical channels aggregated together on the mobile multilink layer, which is also defined in this annex, to provide a higher total bitrate. H.324 Annex H is primarily designed for use on error-prone wireless channels.

Figure 3 below illustrates the channel aggregation at the mobile multilink layer and the multilink frame format to be used on each physical channel. The mobile multilink layer is located between the physical channel and the H.223 multiplexing layer. It divides the output bitstream from the H.223 multiplexing layer into Sample Size (SS) bytes, and maps them onto each channel. The order of mapping is fixed, according to the order of the Channel Tag (CT), which is assigned to each channel and set to the CT field in the multilink frame header. To achieve the inter-and intra-channel synchronization at the receiver, a multilink frame header (synchronization flag and header information) is inserted into every SPF (Sample Per Frame) sample. At the receiver side, the mobile multilink layer extracts SS-byte samples from each channel in the order of the value of the CT field, and reconstructs the original bitstream.

In H.324 Annex H, two data transmission modes are defined according to the header configuration: full header mode and compressed header mode. Transition between transmission modes is initiated either by the H.245 command message from the receiver or by the transmitter accompanied with the H.245 indication message. A change of the Multilink frame length (SS and SPF) is permitted during the full header mode to minimize frame synchronization errors due to transmission bit errors.

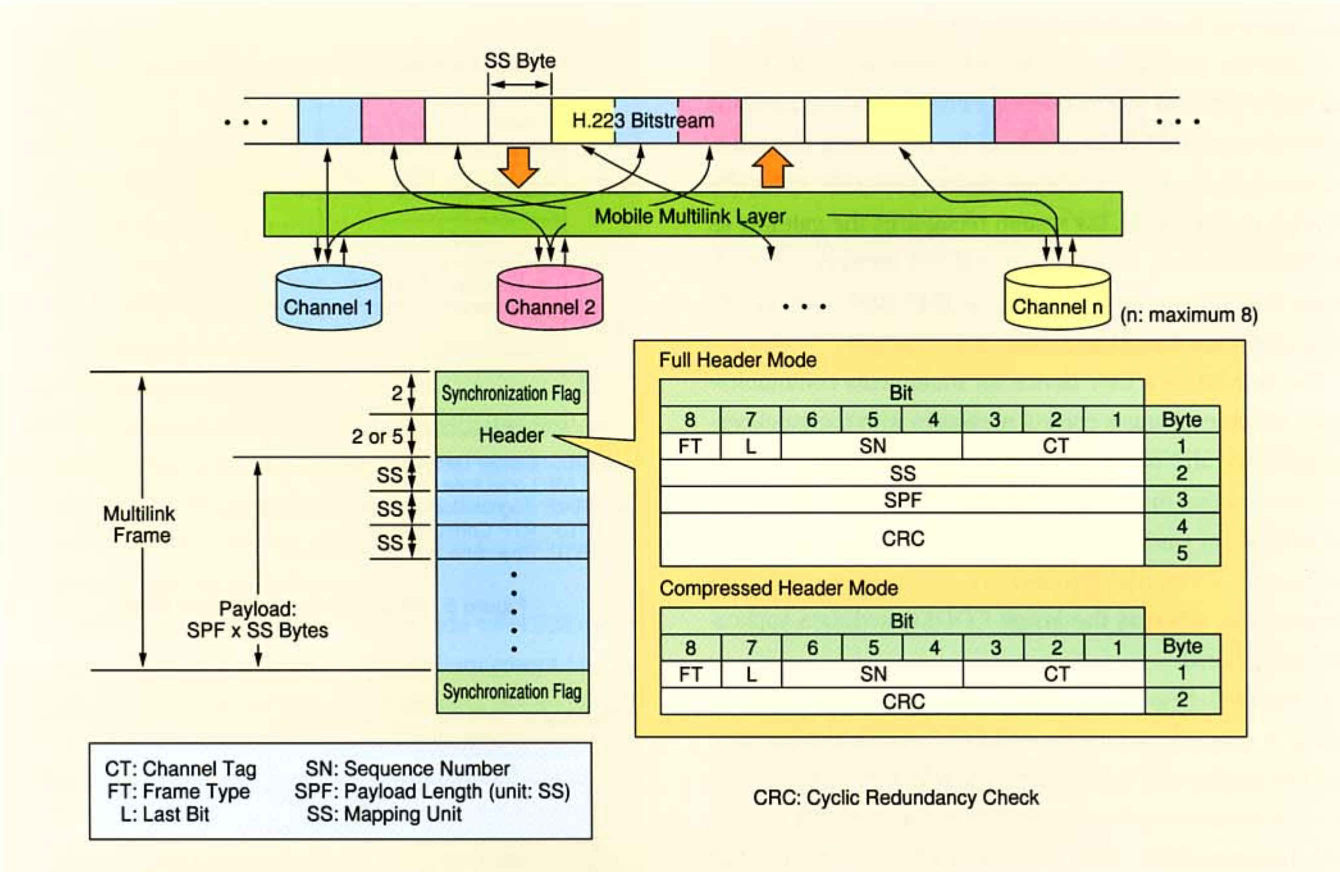
3. IP Network Audiovisual Terminal Technology

The explosive market penetration of the Internet has made a wide range of applications available over IP networks. For audio and video, which have traditionally been transmitted over circuit switched networks, efforts are being made to transmit them over IP networks in order to maximize network efficiency. 3GPP and DoCoMo are making efforts to shift to IP networks.

This chapter provides an overview of multimedia system standards for IP networks.

3.1 Standardization Trends

Real-time multimedia system standards for IP networks include H.323 and Session Initiation Protocol (SIP). H.323



Full Header Mode

Bit 8	7	6	5	4	3	2	1	Byte
FT	L	SN			CT			1
SS								2
SPF								3
CRC								4
								5

Compressed Header Mode

Bit 8	7	6	5	4	3	2	1	Byte
FT	L	SN			CT			1
CRC								2

Figure 3 Operations of the Mobile Multilink Layer

was standardized in conjunction with H.320 by ITU-T, and the first version was released in 1996. This standard has been adopted in Voice over IP (VoIP) equipment, and is widely used over the Internet. SIP was standardized in line with Hyper Text Transfer Protocol (HTTP) by IETF, and was specified as RFC2543 in 1999. In 3GPP, SIP will be used as the call control protocol for their All-IP network specification, which enables services similar to circuit-switched services over All-IP network.

3.2 H.323 Overview

H.323 is an ITU-T standard that prescribes multimedia communications systems for IP networks and other packet-switched networks, and is one of the most successful and most rapidly spreading specifications for Internet telephony systems. Figure 4 illustrates a multimedia communications system defined by H.323. H.323 defines the functional components in the multimedia communications system (i.e., terminal, gateway, gatekeeper and Multipoint Control Unit) and specifies the communication procedures to be taken by each component when a user executes multimedia communications. The control protocol for multimedia communications and the packetization scheme for audio and video are fully defined by H.245 (refer to Section 2.5) and H.225.0.

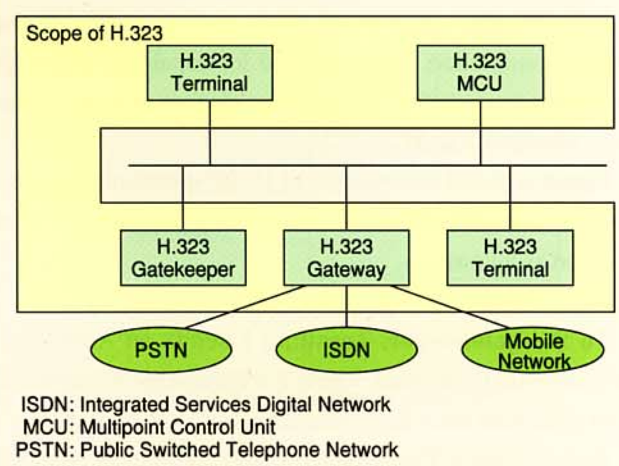


Figure 4 H.323 System Configuration

The function of each component shown in Figure 4 is as follows. The gatekeeper is a component that controls calls between H.323 endpoints (terminal and gateway). It has functions to:

① Translate an alias address into an IP address;

② Approve/reject a terminal's request to start communication; and

③ Approve/reject a terminal's bandwidth request.

A gateway provides connectivity between an H.323 terminal and a terminal in a non-H.323 network (e.g., a 3G-324M terminal in an IMT-2000 network) by converting the transmission formats, communication control protocols and audio & video coding. An H.323 system recognizes the gateway as an H.323 terminal, whereas a non-H.323 network acknowledges it as its own terminal (e.g., an IMT-2000 network will think that it is a 3G-324 terminal).

The terminal is a user device for multimedia communications, which exchanges control messages with the gatekeeper, gateway and other terminals, and also transmits and receives various media data. **Figure 5** depicts the H.323 terminal protocol stack, in which the audio CODEC, the terminal control parts, and the network interface are required components, whereas the image CODEC and data application are optional.

In **Figure 5**, Registration, Admission and Status (RAS) signaling is used to execute procedures between the terminal and the gatekeeper, such as terminal registration, admission to start communication, bandwidth request, status display and disconnection. RAS messages are transmitted and received for these purposes. Call control signaling is used to establish a virtual connection between terminals in a connectionless IP network. H.225.0 specifies call control signaling based on Q.931 including call-setup, ringing, answering and release-completion. H.245 is used for capability exchange between terminals and the opening/closing of logical channels, as with 3G-324M.

Figure 6 shows an example of H.323 communication procedures using a gatekeeper. The terminal executes communication as follows.

(1) Call Setup

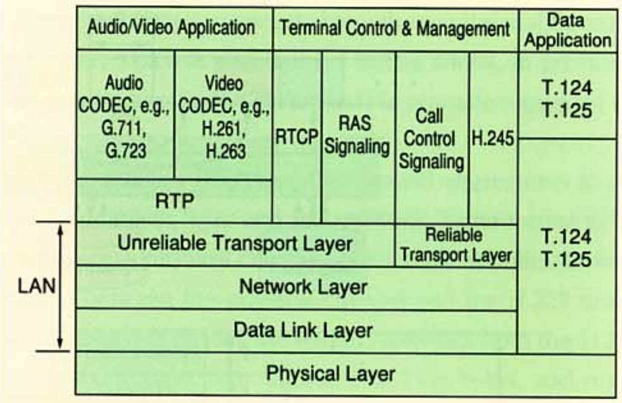
To the gatekeeper, Terminal 1 sends an Admission Request (ARQ) message, which is a request for admission to start communication ①.

If an Admission Confirmation (ACF) message is received, Terminal 1 opens a call-control channel for the address informed in the ACF message ② (in this case, Terminal 2), and sends a Call Setup message to the address ③.

Terminal 2 sends ARQ to the gatekeeper ⑤. In response to the reception of ACF ⑥, it sends a CONNECT message to Terminal 1.

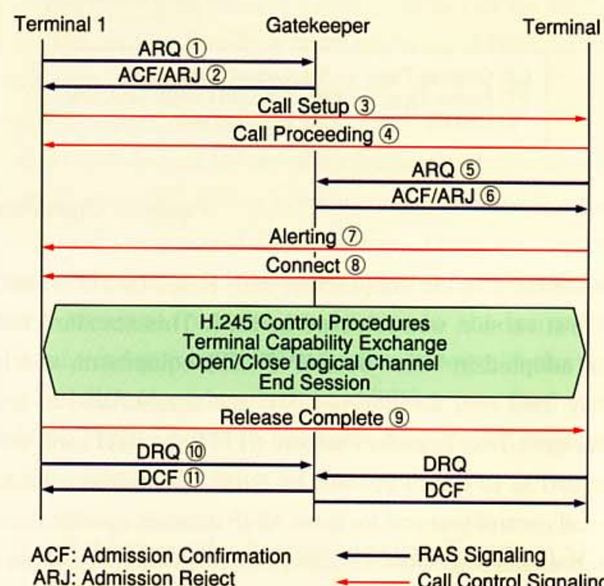
(2) Terminal Capability Exchange

Terminal 1 opens an H.245 control channel for the address informed in the call-control signaling message (in this case, Terminal 2). Then, it sends and receives information on terminal capabilities with the use of an H.245 message.



CODEC: Corder Decorder
LAN: Local Area Network
RAS: Registration, Admission and Status
RTCP: RTP Control Protocol
RTP: Real-time Transport Protocol

Figure 5 H.323 Terminal Protocol Stack



ACF: Admission Confirmation
ARJ: Admission Reject
ARQ: Admission Request
DCF: Disengage Confirmation
DRQ: Disengage Request
RAS: Registration, Admission and Status

Figure 6 H.323 Communication Procedures

(3) Establishing Audiovisual Communication

In response to receiving the other terminal's capability information, audio and video CODECs and parameters that can be used for the communication between the terminals are chosen. With the use of an H.245 message, logical channels for audio and video data transmission are opened.

Real-time Transport Protocol (RTP) is used for transmitting of audio and video data, and RTP Control Protocol (RTCP) for the transmission control of such data.

(4) Ending Calls

The logical channel is closed with the use of an H.245 message.

Terminal 1 sends an H.245 End Session command, and closes the H.245 control channel after receiving an End Session command from Terminal 2.

If the call control channel remains open, Terminal 1 sends a Release Complete message ⑨ and closes the channel.

Then, Terminal 1 sends a Disengage Request (DRQ) message to the gatekeeper ⑩.

When using a gatekeeper, registration with the gatekeeper is required before call setup. Other than the above scenario, it is also possible to have procedures that do not involve the use of a gatekeeper, as well as procedures for sending call control messages via the gatekeeper.

Various enhancements have been made to H.323, since the release of Version 1. One such enhancement is Fast Connect, which is a procedure for shortening the initial connection time. The latest version currently available is Version 4, released in November 2000.

3.3 Overview of Session Initiation Protocol (SIP)

SIP is a call control protocol for multimedia communications over IP networks and other packet based networks. RFC2543, which specifies SIP, sets the system configuration, control message transmission procedures and formats. To achieve SIP-based multimedia communications, SIP must be applied with other protocols like Session Description Protocol (SDP) and RTP. SDP is used to describe the media types in use, network address, and other information concerning communication. RTP is applied to send audio, video and other media data.

(1) SIP System Configuration

As in the example of an SIP system shown in Figure 7, the SIP system consists of User Agent (UA)s, a proxy server and a location server. UAs correspond to user terminals and media storage devices in networks, and each of them has an identifier called SIP URL, similar to an e-mail address. The proxy server relays control messages sent from UA and other devices, and controls the calls. The location server manages the SIP URL and address information.

Figure 8 illustrates the layer structure of an SIP user terminal. For the transmission of control messages, a reliable protocol (e.g., Transmission Control Protocol: TCP) or an unreliable one (e.g., User Datagram Protocol: UDP) may be used. In cases where an unreliable protocol is chosen, reliability is ensured by retransmitting data to higher layers.

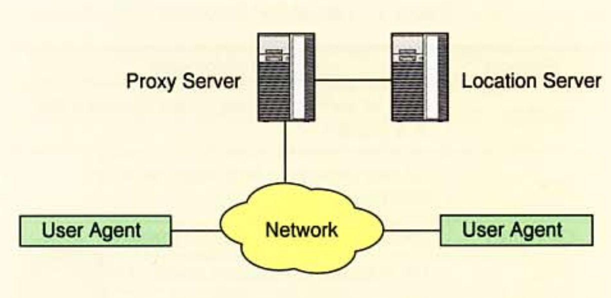


Figure 7 SIP System Configuration

Call Control & Terminal Control	Media		
Session Description SDP	Audio CODEC	Video CODEC	Data Application
SIP	Media Transmission Protocol RTP/RTCP		
e.g., TCP, UDP	UDP		
IP			

CODEC: Corder Decoder
IP: Internet Protocol
RTP: Real-time Transport Protocol
SDP: Session Description Protocol
SIP: Session Initiation Protocol
TCP: Transmission Control Protocol
UDP: User Datagram Protocol

Figure 8 SIP Protocol Stack

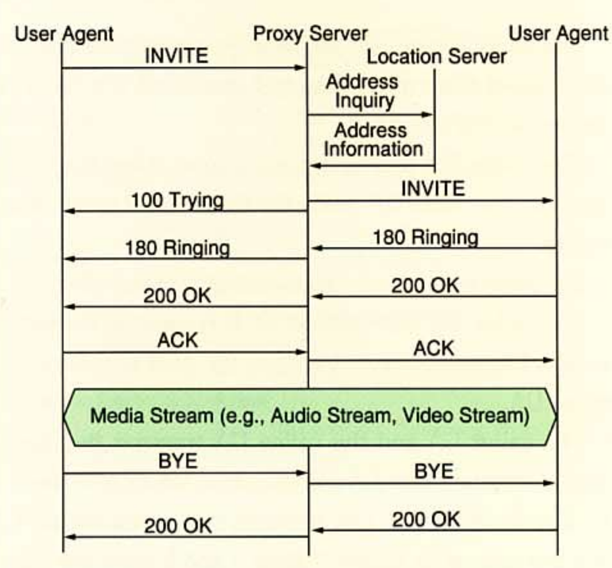


Figure 9 SIP Call Control Procedures

(2) Call Setup Procedures

Figure 9 shows an example of a call setup procedure by SIP. Control messages consist of text. Each control unit is composed of a request and a response.

① The caller UA transmits an INVITE request to the proxy server.

Table 1　List of SIP Request

Request	Nature of Request
INVITE	User or service is being invited to participate in a session.
ACK	UA has received a final response to an INVITE request.
OPTIONS	UA is being queried about capabilities.
BYE	UA indicates to server to release the call.
CANCEL	UA cancels a pending request.
REGISTER	UA registers address with location service.

Table 2　List of SIP Response Types

Status-Code	Response-Phrase	Nature of Response
1xx	Infomational	Request received, continuing to process the request.
2xx	Success	The action was successfully received, understood, and accepted.
3xx	Redirection	Further action needs to be taken in order to complete the request.
4xx	Client Error	The request contains bad syntax or cannot be fulfilled at this server.
5xx	Server Error	The server failed to fulfill an apparently valid request.
6xx	Global Failure	The request cannot be fulfilled at any server.

② The proxy server uses the location server to identify the address of the callee UA, and transfers the INVITE request to that callee UA.

③ The callee UA makes a ringing tone. After the callee answers, the callee UA sends an OK response to the proxy server.

④ The proxy server sends an OK response to the caller UA.

⑤ The caller UA transmits an ACK request to the proxy server. The proxy server transfers the ACK request to the callee UA.

⑥ The caller UA and the callee UA transmit the media stream and the caller UA and the callee UA receive them.

In this manner, SIP can perform call setup within 1.5 round trip time at its fastest. Tables 1 and 2 show the list of requests and response types, respectively.

(3) Overview of Session Description Protocol (SDP)

As mentioned in (2), SDP describes the information for transmitting the media stream in the INVITE request and the OK response, such as the type of media to be transmitted/received, its attributes, the network address and port number. As in the example shown in Figure 10, the description in SDP is in text format, as with the SIP control message. Each line of SDP consists of a field like <type>=<value>, in which <type> represents the type of description by alphabet, and <value> shows the description according to <type>. The order of the appearance of the field is predefined.

V=0	→Protocol Version
o=- 2890844256 2890842807 IN IP4 192.168.0.1	→Owner/Creator and Session Identifier
s=Let's talk	→Session Name
t=0 0	→Time the session is active
c=IN IP4 192.168.0.1	→Connection Information
m=audio 49120 RTP/AVP 97	→Audio Description
a=rtpmap: 97 AMR/8000	→Audio Attribute
a=fmtp: 97 maxframes=2	→Audio Attribute
m=video 49170 RTP/AVP 98	→Video Description
a=rtpmap: 98 MP4V/90000	→Video Attribute
a=fmtp: 98 profile-level-id=1	→Video Attribute

Figure 10　Example of SDP Description

SDP was originally designed as a standard for notifying the terminal capabilities and other information in one direction. As it is not a standard for terminals notifying their capabilities to each other when starting communication, the following problems have been pointed out.

① The CODEC in use cannot be determined until the media stream arrives.

② Due to the poor description, terminal capabilities cannot be explained sufficiently.

In response, IETF is engaged in standardization activities under the name SDP Next Generation (SDPng), aimed at solving these problems and adding new functions.

4. Conclusion

This article explained the audiovisual terminal technologies used in IMT-2000 and IP packet networks, with reference to standards established by ITU-T, 3GPP and IETF. In regard to audiovisual terminal technologies geared to IMT-2000, it provided an overview of the 3G-324M standard specified by 3GPP and described the component technologies. With respect to the technologies for packet networks, it outlined H.323 (the ITU-T standard which is widely used today in VoIP equipment) and SIP (the IETF standard which enables circuit switching services using 3GPP packets). The scope of audiovisual terminal technologies is expected to expand further, including applications in wireless packet networks. Close attention must be paid to technology trends in the future.

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Glossary

ACF: Admission Confirmation
AMR: Adaptive Multi Rate
ARJ: Admission Reject
ARQ: Admission Request
ATM: Asynchronous Transfer Mode
B-ISDN: Broadband Aspects of Integrated Services Digital Network
CCSRL: Control Channel Segmentation and Reassembly Layer
CODEC: Corder Decorder
CRC: Cyclic Redundancy Check
DCF: Disengage Confirmation
DRQ: Disengage Request
HDLC: High-Level Data Link Control Procedure
HTTP: Hyper Text Transfer Protocol
IETF: Internet Engineering Task Force
IMT-2000: International Mobile Telecommunications-2000
IP: Internet Protocol
ISDN: Integrated Services Digital Network
ITU: International Telecommunication Union
ITU-T: International Telecommunication Union-Telecommunication Standardization Sector
LAN: Local Area Network
LAPM: Link Access Prosedure for Modem
MCU: Multipoint Control Unit
MPEG: Moving Picture Experts Group
N-ISDN: Narrowband Integrated Services Digital Network
NSRP: Numbered Simple Retransmission Protocol
PN: Pseudo Noise
PSTN: Public Switched Telephone Network
QoS: Quality of Service
RASvRegistration, Admission and Status
RTPvReal-time Transport Protocol
RTCP: RTP Control Protocol
SDP: Session Description Protocol
SDPng: SDP Next Generation
SIP: Session Initiation Protocol
TCP: Transmission Control Protocol
UA: User Agent
UDP: User Datagram Protocol
VoIP: Voice over IP
W-CDMA: Wideband Code Division Multiple Access
3GPP: 3rd Generation Partnership Project