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Interoperability Test Guideline

For SIP/MPEG-4 Multimedia Communication System

HATS Conference
（Promotion Conference of Harmonization of Advanced Telecommunication Systems）
Multimedia Communication Test Implementation Liaison Committee

Interoperability Test Guideline for SIP/MPEG-4 Multimedia Communication System

Revision History

Version	Date of revision	Content of revision	Person in charge
1.0	2006/3/15	First edition	Nakabayashi
1.1	2006/6/2	Add description for JT-G711	Nakabayashi
1.2	2006/6/13	Add some of the explanations on H.264	Nakabayashi
1.3	2006/9/11	Complete adding all of the explanations on H.264	Nakabayashi

Contents

1. Background and Purpose

1-1. Background

ISO/IEC 14496 “Information technology — Coding of audio-visual objects — Part 2: Visual” is a compression encoding method, which we expect to be widely applied in the future. It encodes effectively, has the enhanced error tolerance, and supports the object-based encoding. On the other hand, ITU-T H.264 | ISO/IEC 14496-10 “Information technology — Coding of audio-visual objects — Part 10: Advanced Video Coding” is a compression method which achieves a higher compression rate. It is expected to be applied in the wide range from internet streaming applications with low bit rates to HDTV broadcasting and digital movie applications. RFC3261 (SIP: Session Initiation Protocol), which was standardized by IETF (Internet Engineering Task Force), allows multimedia communications with rapidly spreading LANs. For the sound development of these technologies, it is necessary to resolve various problems regarding interoperability between terminals and reflect the results to the standard.

1-2. Purpose

With the market share of the products based on the above standard growing, it is essential to ensure interoperability between the products in order to facilitate utilization of digital TV phone/conference systems. However, we assume that various kinds of functionalities will be added to the various products in the future, and therefore, there might be a situation in which the interoperability is not ensured between different products based on the same standard. To handle this situation, we need to test and ensure their interoperability.

In this Guideline, the contents and the procedures are provided to conduct such tests which check the minimum interoperability between the devices made by different manufacturers.

The specific interoperability tests are conducted by the “Multimedia Communication Test Implementation Liaison Committee: MMC TILC” of HATS administered by the “Communication and Information network Association of Japan: CIAJ”. These tests attempt to ensure interoperability between each product and consequently, it is expected that the infrastructure to put the digital TV phone/conference system in practice would be improved. We also hope that the effectiveness of the standard itself would increase and when planning new standards, it could be used as a reference.

Note : The words "MUST", "SHOULD", "RECOMMENDED", "MAY", and "OPTIONAL" stated in this document are used according to the descriptions in RFC2119.

2. Preconditions of Tests

2-1. Standards to be conformed

Figure 2.1 shows the multimedia communication terminal of SIP. The common standards that MUST be conformed for the interoperability of the system are as follows:

(1) RFC3261 "SIP: Session Initiation Protocol"

(2) RFC2327 "SDP: Session Description Protocol"

(3) RFC3550 "RTP: A Transport Protocol for Real-Time Applications"
"RTP Control Protocol -- RTCP"

(4) RFC3551 "RTP Profile for Audio and Video Conferences with Minimal Control"

(5) RFC3016 "RTP Payload Format for MPEG-4 Audio/Visual Streams"

(6) RFC3264 "An Offer/Answer Model with the Session Description Protocol (SDP)"

(7) ISO/IEC14496-2(2004)
"Information technology — Coding of audio-visual objects — Part 2: Visual"

(8) JT-G711 "Pulse Code Modulation (PCM) of Voice Frequencies"

Note: This standard was established by TTC (Telecommunication Technology Committee) of Japan according to the ITU-T G.711 standard.

In the JT-G711 standard, all the descriptions relevant to A-law PCM are deleted from the ITU-T recommendation G.711 because µ-law PCM has been adopted as standard in Japan.

(9) RFC2119 "Key words for use in RFCs to Indicate Requirement Levels"

(10) RFC3984 "RTP Payload Format for H.264"

(11) H.264|ISO/IEC14496-10(2005)

H.264 | "Information technology — Coding of audio-visual objects — Part 10: Advanced Video Coding"

Note: In this Guideline, unless otherwise stated, "H.264" refers to ITU-T H.264 | ISO/IEC 14496-10 "Information technology — Coding of audio-visual objects — Part 10: Advanced Video Coding".

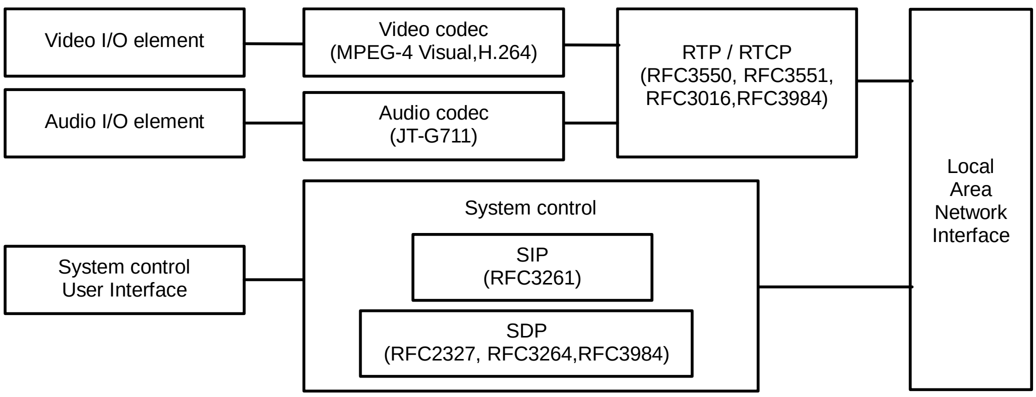

Figure 2.1 SIP multimedia communication terminal

2-2. Pre-test

Connect the required components for the interoperability test to the 10/100BASE-T local area network and ensure that the test items listed in Chapter 3 work properly between your own components.

3. Interoperability Tests

3-1. Definition of the test profiles

The interoperability test is intended for the devices which meet the pre-test conditions in Chapter 2. However, as there are too many target devices, the standard test profiles have been decided. This test SHOULD basically be conducted subject to these profiles.

For the interoperability test, two test profiles are defined in each video coding method:

・High rate profile

・Low rate profile

3-1-1 MPEG-4 Test Profiles

Table 3.1 MPEG-4 test profiles

Item	Specification(for high rate)	Specification(for low rate)
Call control Capability exchange	SIP (RFC3261) SDP (RFC2327) Capability exchange (RFC3264)	Same as for high rate
Media transmission	RTP (RFC3550, RFC3551) RTCP (RFC3550 Option) MPEG-4 packetization (RFC3016)	Same as for high rate
Video: Video encoding method Frame size(RECOMMENDED)	 MPEG-4 Visual (SP@L3) CIF	 MPEG-4 Visual (SP@L0) QCIF
Audio: Audio encoding method (RECOMMENDED)	JT-G711μ-law	Same as for high rate

(1) Call control: RFC3261-complied SIP is used. SIP is not expanded such as RFC3262. SDP is used for the capability exchange and RFC3264 is used for its negotiation.

(2) Media transmission: RFC3550- and RFC3551-complied RTP is used. RTCP is OPTIONAL. For the packetization of MPEG-4 Visual bitstream, RFC3016 is used.

(3) Video: MPEG-4 Visual is used. The test profile for the high rate is SP@L3 and for the low rate is SP@L0. In both cases, it is run with the encoding rate and the frame size within the standard. To improve the interoperability, CIF is RECOMMENDED for the high rate, and QCIF for the low rate. For the number of the objects, “1” is RECOMMENDED. However, you MAY use any other parameters within the range of the profile level if possible.

(4) Audio: There is no fixed audio encoding method, but JT-G711μ-law is RECOMMENDED for the test profile. The tests for the other audio encoding methods MAY be conducted if possible.

3-1-2 H.264 Test Profiles

Table 3.2 H.264 test profiles

Item	Specification(for high rate)	Specification(for low rate)
Call control Capability exchange	SIP (RFC3261) SDP (RFC2327) Capability exchange (RFC3264/3984)	Same as for high rate
Media transmission	RTP (RFC3550, RFC3551) RTCP (RFC3550 Option) H.264 packetization (RFC3984)	Same as for high rate
Video: Video encoding method Frame size(RECOMMENDED)	H.264 Baseline Profile Level 1.2 CIF	H.264 Baseline Profile Level 1 QCIF
Audio: Audio encoding method (RECOMMENDED)	JT-G711μ-law	Same as for high rate

(1) Call control: RFC3261-complied SIP is used. SIP is not expanded such as RFC3262. SDP is used for the capability exchange and RFC3264 and RFC3984 are used for its negotiation.
(2) Media transmission: RFC3550- and RFC3551-complied RTP is used. RTCP is OPTIONAL. For the packetization of H.264 stream, RFC3984 is used.
(3) Video: H.264 is used. The test profile for the high rate is Baseline Profile Level 1.2 and for the low rate is Baseline Profile Level 1. In both cases, it is run with the encoding rate and the frame size within the standard. To improve the interoperability, CIF is the RECOMMENDED image size for the high rate, and QCIF for the low rate. For the number of the objects, “1” is RECOMMENDED. However, you MAY use any other parameters within the range of the profile level if possible.
(4) Audio: There is no fixed audio encoding method, but JT-G711μ-law is RECOMMENDED for the test profile. The connection tests for the other audio encoding methods MAY be conducted if possible.

3-2. Test environment

(1) Use the private environment separated from the local area network that is usually operated.
(2) Figure 3.1 shows the connection between the components of the test.

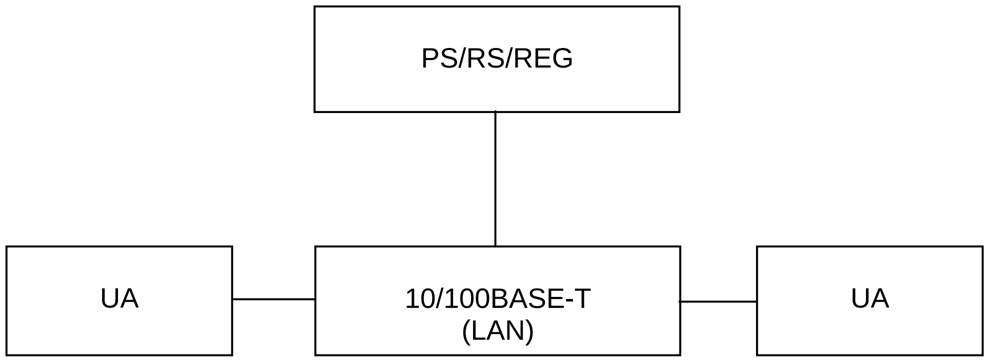

Figure 3.1 Connection between components

UA: User agent
PS: Proxy server
RS: Redirect server
REG: Registrar server

(3) The components used for the test is connected to the test LAN. At this time, more than one component which is not used for the test MAY be connected to the same LAN. However, the caution needs to be taken so that these components do not affect each other's performance such as the bandwidth used.
(4) Prepare telephones for contact if the components are set up at different locations.

3-3. Execution method

(1) On the date arranged beforehand, the test is conducted according to the procedures described in this chapter.
(2) The combination of the connection is round robin.
The test scenarios are as follows:
・Scenario 1: Connect a UA without the server.
・Scenario 2: Connect a UA through the server.
The scenario to be used is decided before the test by the testing vendors.
In Scenario 2, when Company A's UA and Company B's UA are connected for example, there are two combinations of the test; the connection through Company A's server and the connection through Company B's server.

Note that each manufacturer is responsible for the interoperability between the products made by itself and the test is assumed to have been completed. Therefore, it is not included in the combination.

3-4. Testing procedures

(1) Register a UA in the server (not necessary in Scenario 1).

(2) “Calling UA” calls “Receiving UA”.

(3) If the call is not received, try calling again up to 3 times. If the call is still not received, check the communication conditions, such as the registration information. If something is wrong with the conditions, then retry from (1); otherwise, consider this as a communication error and conduct procedure (7).

(4) After confirming the connection, the receiving UA checks that it can properly receive the audio, the video, and the other test items from the other terminal in accordance with the items listed in Appendix 1 for MPEG-4 and Appendix 2 for H.264. Also, record the encoding mode that has executed the communication for the caller or the receiver respectively in Calling UA or Receiving UA.

(5) Continue the communication for at least 3 minutes. Then, check if all the items have been tested.

(6) Both the caller and the receiver confirm that the communication can be disconnected properly.

(7) Switch the role of the caller and the receiver, and repeat from (1) to (6).

3-5. Confirmation Details / Result Assessment for Test

In this Guideline, the test items are determined for only audio and video communications. Checking for other mode changes during the communication (such as video format, parameters, and still pictures) is OPTIONAL. The test is passed if the details of the test procedures are conformed and the confirmations of the followings are successful.

(1)Confirmation of the digital communication

Follow the test procedure, and check that the connection is done with the appropriate transfer rate according to the call connection and the receiving capability of both terminals.

(2)Confirmation of video and audio communications

Confirm the audio and video communications and check the mode for the receiving capability of the both terminals.

(3)Confirmation of the communication disconnection

Follow the test procedures and check if the call can be disconnected properly.

Note that test items SHOULD be added or modified if necessary.

3-6. Test results handling

On completion of the test, after both the caller and the receiver check the results, the receiver fills in the check sheet in Appendix 1 for MPEG-4 and Appendix 2 for H.264. If errors occur during the test, describe the situation as detailed as possible in the check sheet (phenomena, causes, actions, etc) after the discussion between the testing vendors or with the Secretariat office.

If you wish to re-test, indicate it in the MEMO section in the check sheet.

3-7. Optional test items (reference)

It is preferable to conduct more advanced connectivity tests when both terminals obviously have more capabilities.

Whether to conduct the OPTIONAL tests or not SHOULD be considered when requested. The needs for such tests are closely linked to the improvement of the terminal capabilities.

4. Recommended Specification

4-1. Basic connection sequences

The basic sequences of SIP/MPEG-4 interoperability with SIP Protocol are shown below. Figure 4.1 is the connection sequence without the server and Figure 4.2 is the connection sequence through the server.

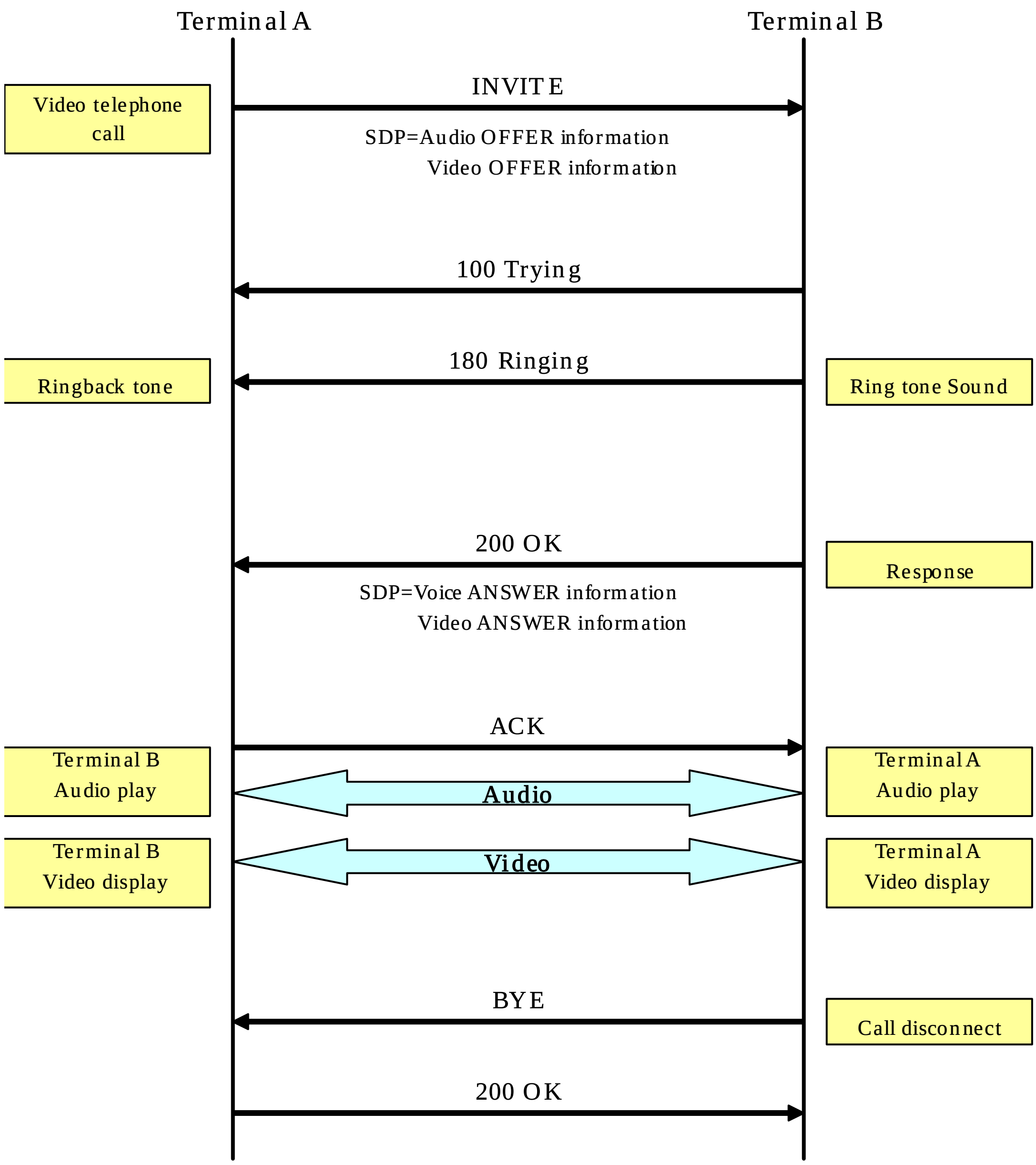

Figure 4.1 Basic sequence of the video communication
(UA connection without the server)

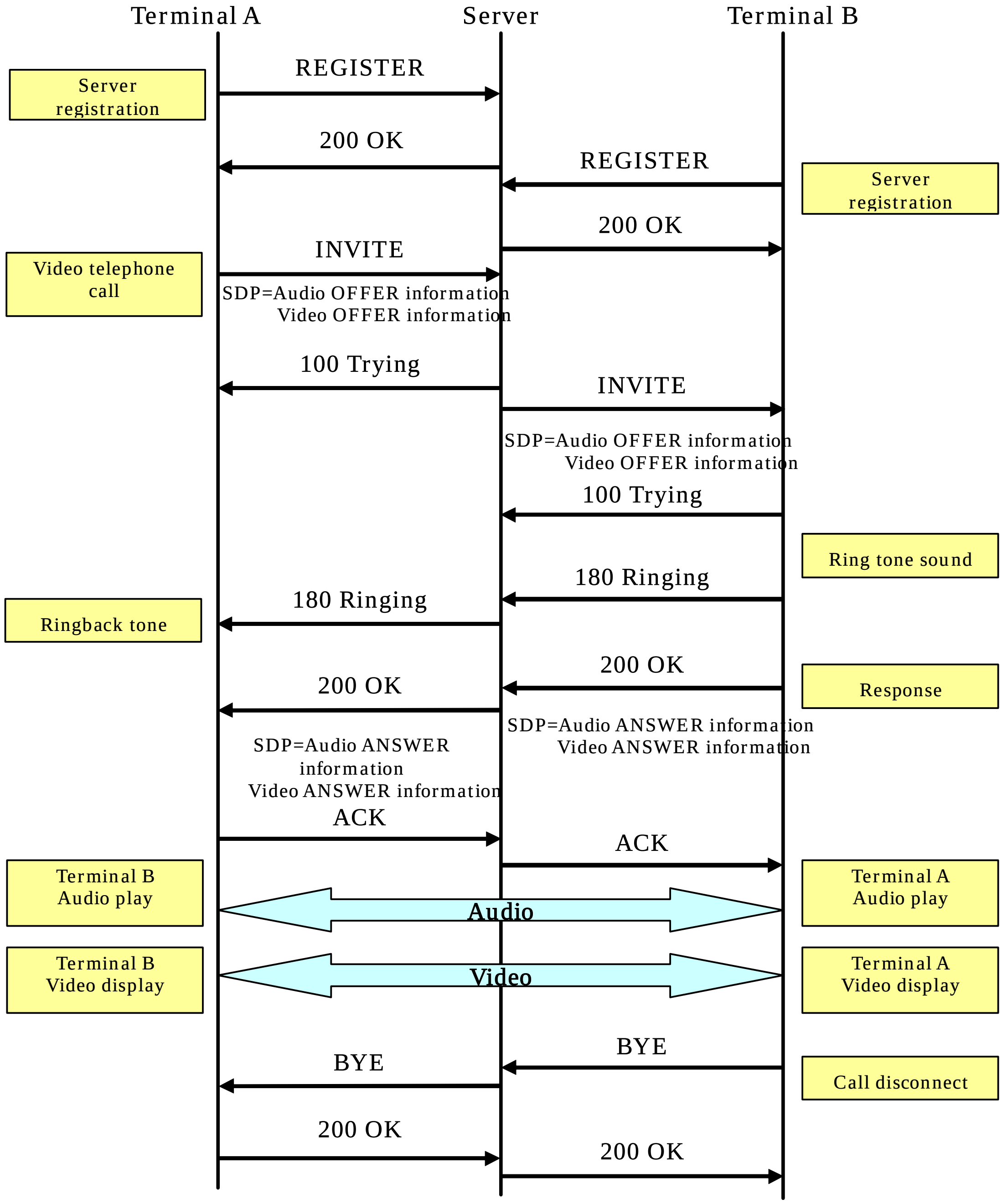

Figure 4.2 Basic sequence of the video communication
(UA connection through the server)

4-2. SIP message regulations

Table 4.1 shows the information which MUST be described in the INVITE request for the SIP/MPEG-4 video communication negotiation.

Table 4.1 INVITE request regulations outline

Item	Content	Remarks
Request Line	Method (＝INVITE)	
	Request-URI	
	SIP-Version	
Header Field	Via	
	From	
	To	
	Call-ID	
	CSeq	
	Max-Forwards	
	Contact	
	Content-Type	Necessary when SDP is used
	Content-Length	same as the above

4-3. SDP parameter regulations

4-3-1 SDP regulations outline

Below is the SDP regulations outline. Only the essential parameters for the SIP/MPEG-4 video communication negotiation are shown.

Table 4.2 SDP regulations outline

<table>
<tr><th colspan="3">Line type、Parameter</th><th>Regulations</th><th>Remarks</th></tr>
<tr><td rowspan="4">m</td><td colspan="2"><media></td><td>Fixed as “Video"</td><td>Media type used</td></tr>
<tr><td colspan="2"><port></td><td>Set the port number for RTP stream reception</td><td>RTCP Reception port number＝This port number＋1</td></tr>
<tr><td colspan="2"><transport></td><td>Fixed as “RTP/AVP"</td><td></td></tr>
<tr><td colspan="2"><fmt list></td><td>”96”-”127” (*)</td><td></td></tr>
<tr><td rowspan="7">a</td><td rowspan="3">（Value attribute）
rtpmap</td><td><payload type></td><td>Specify the RTP payload type value of (*)</td><td></td></tr>
<tr><td><encoding name></td><td>Fixed as “MP4V-ES" for MPEG-4
Fixed as “H264” for H.264</td><td></td></tr>
<tr><td><clock rate></td><td>Fixed as “90000"</td><td></td></tr>
<tr><td rowspan="4">（Value attribute）
fmtp</td><td><payload type></td><td>Specify the RTP payload type value of (*)</td><td>RTP dynamic payload value</td></tr>
<tr><td rowspan="2"><profile-level-id></td><td>MPEG-4:
For the high rate：SP@L3
For the low rate：SP@L0</td><td></td></tr>
<tr><td>H.264:
For the high rate:
Baseline Profile Level 1.2
For the low rate:
Baseline Profile Level 1</td><td></td></tr>
<tr><td><config></td><td>Specify "Config" that wishes to send</td><td>Encoder config information (mandatory for MPEG-4)</td></tr>
</table>

4-3-2 Details of the SDP regulations

(1) m

Specify the attribute of the video media you wish to use.

Description format:

m=<media><port><transport><fmt list>

Example of the setting：

m=video 18624 RTP/AVP 98

The port example has to be an even number.

The OFFER side can specify more than one encoding format of the video media. The ANSWER side can choose only one encoding format from the OFFER side's requests and return it to the OFFER side.
If there is no encoding format available, the ANSWER side MUST change only the port number in m line to 0 and return it to the OFFER side.

・<media>

Only “Video" is allowed.

・<port>

Specify the port number for the RTP stream reception.

Specify the even port number for the RTP reception number.

The RTCP reception port number is the RTP reception port number + 1, which is an odd number.

・<transport>

Only "RTP/AVP" is allowed. (Specify the transport protocol)

・<fmt list>

Specify the RTP dynamic payload type value.

The OFFER side can define the RTP dynamic payload type values of the video encoding. More than one value can be specified.

The OFFER side describes the value from the left to the right, from the high priority to the low priority.

The ANSWER side can choose only one payload type value from the values specified by the OFFER side. The ANSWER side MUST specify the dynamic payload value without changing anything and ANSWER to the OFFER side.

(2) a

Specify the video session information for each media.

Description format:

(In the case of the value attribute)

a=<attribute>:<value>

Example of the setting:

(In the case of the value attribute)

a=rtpmap:<payload type> <encoding name>/<clock rate>
[/<encoding parameter>]

a=fmtp:<payload type> <profile-level-id> <config>

・How to handle the value attribute of rtpmap :

－Describe <payload type> with the RTP dynamic payload value indicated in <fmt list> of the m line.

－Describe <encoding name> with the name of the video encoding. It is fixed as “MP4V-ES" for MPEG-4 Visual and as “H264” for H.264.

－Specify <clock rate> with the clock rate. It is fixed as “90000".
－It is not necessary to describe <encoding parameter>. When described, it would be ignored.

・How to handle the value attribute of fmtp :
－Specify the parameter of the MPEG-4 stream or the H.264 stream to be sent.
This will not be the target of the negotiation.
－Specify <payload type> with the RTP dynamic payload value which is indicated in both the <fmt list> of the m line and rtpmap of the a line.
－Specify <profile-level-id> with the level to be supported. In MPEG-4 Visual, set SP@L3 for the high rate and SP@L0 for the low rate. In H.264, set Baseline Profile Level 1.2 for the high rate and Baseline Profile Level 1 for the low rate.
－Specify <config> with the MPEG-4 encoder configuration information. This is not necessary for H.264.

4-3-3 SDP negotiation method

It is RECOMMENDED to use the method in which you can decide the video encoding format by specifying the m line and the a line with the video encoding formats on the OFFER side and by choosing only one format on the ANSWER side.
The typical sequence is shown below.

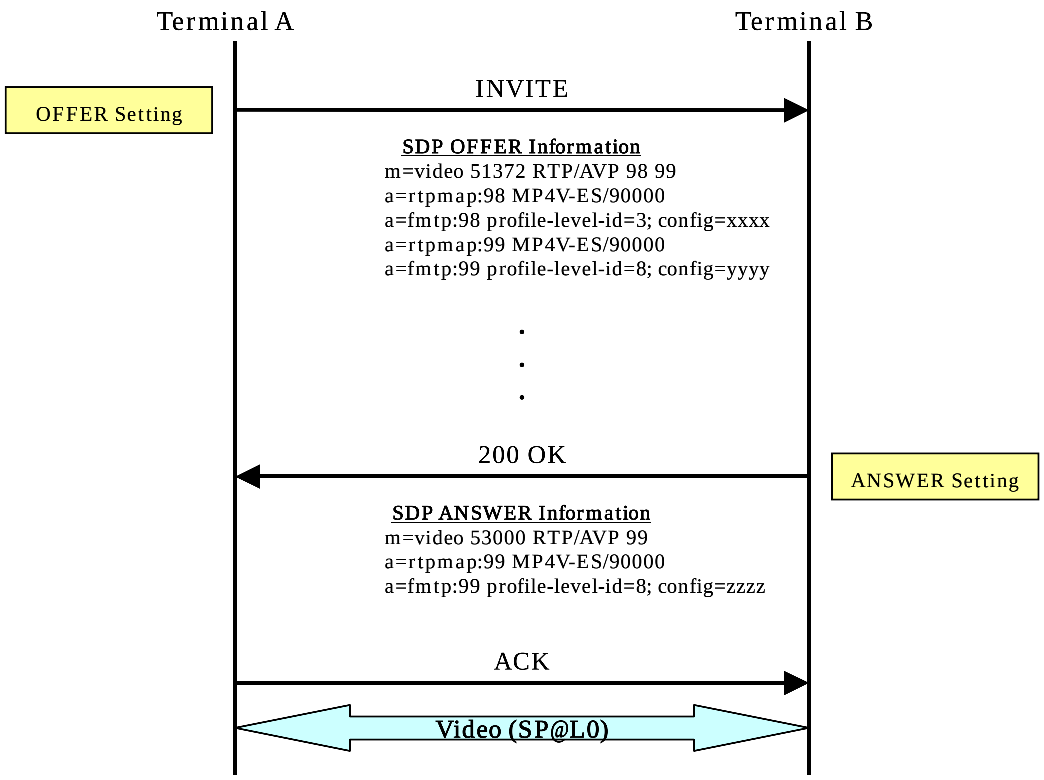

Figure 4.3 Example of the sequence of the SDP negotiation

In the example above, the sending side (terminal A) sends the video stream with the capability at the lower profile level (Simple Profile@Level 0) than the level which the receiving side wishes to use.
For the details of the description of the each line, please refer to 4-3-2.

4-4. Video regulations

4-4-1 Basic operations of encoding and decoding

・MPEG-4 Encoding

Both the OFFER side and the ANSWER side encode the MPEG-4 Visual according to the parameters below.

- It is RECOMMENDED that the MPEG-4 profile and the level SHOULD be encoded with the lower level than the level that the receiving side wishes to use.
- The MPEG-4 profile is specified as Simple Profile with the levels 0-3.
- For the CONFIG parameters, it is encoded with the parameters that the sending side wishes to use.
- When encoding the bit rate which is lower than the bit rate that OFFER/ANSWER wish to use is used (It is RECOMMENDED to select a lower bit rate).

・MPEG-4 Decoding

Both the OFFER side and the ANSWER side decode the MPEG-4 Visual according to the parameters below.

- It is RECOMMENDED that the MPEG-4 profile and the level SHOULD be decoded with the level that sending side wishes to use.
- For the decode CONFIG parameters, the parameters which the sending side wishes to use are used.

・H.264 Encoding

Both the OFFER side and the ANSWER side encode the H.264 according to the parameters below.

- It is RECOMMENDED that the H.264 profile and the level SHOULD be encoded with the lower level than the level that the receiving side wishes to use.
- The H.264 profile is specified as Baseline with the levels 1-1.2.
- When encoding the bit rate which is lower than the bit rate that OFFER/ANSWER wish to use is used (It is RECOMMENDED to select a lower bit rate).

・H.264 Decoding

Both the OFFER side and the ANSER side decode the H.264 according to the parameters below.

- It is RECOMMENDED that the H.264 profile and the level SHOULD be decoded with the level that sending side wishes to use.

4-4-2 Video stream

The CONFIG parameters MUST be included in the MPEG-4 video stream to be sent. The information on PPS/SPS MUST be included in the H.264 video stream.

For the CONFIG in the video stream, the CONFIG information which has been negotiated under "4-3-3 SDP negotiation method" MUST be used.

It is RECOMMENDED to insert I frames regularly; especially for the H.264, it is RECOMMENDED to insert IDR frames. When inserting IDR frames, they MUST be sent with the PPS/SPS information added before them.

4-4-3 How to insert the video stream into the RTP payload

・MPEG-4

It MUST comply with one of the division methods defined in RFC3016.

The division method a) is RECOMMENDED.

・H.264

It MUST comply with one of the payload structures defined in RFC3984.

The payload structure of Single NAL Unit Packet is RECOMMENDED.

4-5. Audio regulations

Each product can choose the audio codec to implement. It is, however, RECOMMENDED to implement JT-G711(μ-Law) which is used mainly for the SIP(VoIP) interoperability tests of the HATS.

5. Results Handling and Future Issues

5-1. Results handling

The results of the interoperability tests submitted by each company are collected and compiled by the Secretariat office. The organized results, in principle, are to be published accordingly.

In order to improve the efficiency of the tests, the test procedures, the methods, the locations, and the results are recorded for future reference.

If any requests or suggestions for this Guideline arise upon conducting the interoperability tests, they can be submitted at any time to the MMC TILC, which will deliberate on whether to accept them.

5-2. Others

If any problems arise about the contents of the standard regulations during the interoperability tests, they will be examined and if necessary, will be reflected in future standardization efforts.

Appendix 1　Check Sheet　(MPEG-4)

SIP/MPEG4 Interoperability Test Check Sheet

[Filled in by]

Company/Organization	
Person in charge	
TEL	
FAX	

Test date [(year) (month) (date) : ～ :]
Test location [ ]
UA A [Company/Organization: Model type:]
UA B [Company/Organization: Model type:]
Server C [Company/Organization: Model type:]

List of test items

№		Item	Judging standard	Result (Yes or No)	Remarks (problem etc.)	
1	Sending side (Terminal A)	Confirmation of audio communication	Confirm the communication of audio and the video in each mode.			Sending side encoding mode Receiving side encoding mode
2		Confirmation of video communication	Record the mode used.			Sending side encoding mode (Profile@Level) Receiving side encoding mode (Profile@Level)
3		Transmission rate of the video	Record the maximum transmission rate capability that was exchanged.	bps bps		Sending side transmission rate Receiving side transmission rate
4		Confirmation of the RTP(a) mode	Confirm that the DCI information is sent through RFC-3016 (a).			When transmitted through (a), fill in Yes. When not transmitted through (a), fill in No.
5		Disconnection by B	Confirm that Terminal A disconnected properly when Terminal B disconnected.			
6		Disconnection by A	Confirm that Terminal B disconnected properly when Terminal A disconnected.			
7	Sending side (Terminal B)	Confirmation of the audio communication	Confirm the communication of audio and the video in each mode.			Sending side encoding mode Receiving side encoding mode
8		Confirmation of the video communication	Record the mode used.			Sending side encoding mode (Profile@Level) Receiving side encoding mode (Profile@Level)
9		Transmission rate of the video	Record the maximum transmission rate capability that was exchanged.	bps bps		Sending side transmission rate Receiving side transmission rate
10		Confirmation of the RTP(a) mode	Confirm that the DCI information is sent through RFC-3016 (a).			When transmitted with (a), fill in Yes. When not transmitted with (a), fill in No.
11		Disconnection by A	Confirm that Terminal B disconnected properly when Terminal A disconnected.			
12		Disconnection by B	Confirm that Terminal A disconnected properly when Terminal B disconnected.			

— MEMO —

[Details of problems]

Appendix 2　Check Sheet　(H.264)

SIP/H.264　Interoperability Test Check Sheet

[Filled in by]

Company/Organization	
Person in charge	
TEL	
FAX	

Test date　[(year)　(month)　(date)　:　～　:　]
Test location　[　]
UA A　[Company/Organization:　Model type:　]
UA B　[Company/Organization:　Model type:　]
Server C　[Company/Organization:　Model type:　]

List of test items

№	Item		Judging standard	Result (Yes or No)	Remarks (problem etc.)	
1	Sending side (terminal A)	Confirmation of audio communication	Confirm the communication of audio and the video in each mode. Record the mode used.			Sending side encoding mode Receiving side encoding mode
2		Confirmation of video communication				Sending side encoding mode (Profile@Level) Receiving side encoding mode (Profile@Level)
3		Transmission rate of the video	Record the maximum transmission rate capability that was exchanged.	bps bps		Sending side transmission rate Receiving side transmission rate
4		RTP confirmation	Confirm the packetization mode of RFC-3984			When transmitted with Single NAL Unit, fill in Yes.Otherwise, fill in No.
			Confirm that the PPS/SPS is transmitted.			When transmitted, fill in Yes. When not transmitted, fill in No.
5		Disconnection by B	Confirm that Terminal A disconnected properly when Terminal B disconnected.			
6		Disconnection by A	Confirm that Terminal B disconnected properly when Terminal A disconnected.			
7	Sending side (terminal B)	Confirmation of the audio communication	Confirm the communication of audio and the video in each mode. Record the mode used.			Sending side encoding mode Receiving side encoding mode
8		Confirmation of the video communication				Sending side encoding mode (Profile@Level) Receiving side encoding mode (Profile@Level)
9		Transmission rate of the video	Record the maximum transmission rate capability that was exchanged.	bps bps		Sending side transmission rate Receiving side transmission rate
10		RTP confirmation	Confirm the packetization mode of RFC-3984			When transmitted with Single NAL Unit, fill in Yes.Otherwise, fill in No.
			Confirm that the PPS/SPS is transmitted.			When transmitted, fill in Yes. When not transmitted, fill in No.
11		Disconnection by A	Confirm that Terminal B disconnected properly when Terminal A disconnected.			
12		Disconnection by B	Confirm that Terminal A disconnected properly when Terminal B disconnected.			

— MEMO —

[Details of problems]