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# Interoperability Test Guideline
# For Multimedia Communication Systems
# Based on RFC3261(SIP)
# - STEP1 -

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

Interoperability Test Guideline For Multimedia Communication Systems Based on RFC3261(SIP) - STEP1 -

Revision history

| Version | Date of revision | Content of revision | Person in charge |
|---|---|---|---|
| 1.0 | 2006/6/2 | First edition | Nozawa |

Contents

# 1. Background and Purpose

## 1-1. Background

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 and its subgroup, the SIP Interoperability SWG" of HATS administered by the "Communication and Information network Association of Japan: CIAJ", and "VoIP Deployment Consortium Interoperability WG" administered by the "TELECOM SERVICE ASSOCIATION". These tests attempts 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 will increase and when planning new standards, it could be used as a reference.

## 1-3. Scope of Interoperability Test

The scope of the interoperability test in this Guideline, is the equipment based on the specification RFC3261(SIP) or RFC2543(SIP) specified by IETF(The Internet Engineering Task Force). This STEP1 Guideline does not include all of the RFC3261(SIP) or RFC2543(SIP) specifications. This Guideline provides the minimum necessary procedure for an actual interconnectivity test environment. Other function tests will be added in the future if necessary.

## 2. Preconditions of Tests

### 2-1. Standards to be conformed

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

＜ The Internet Engineering Task Force : IETF Specifications ＞

(1) RFC3261 “SIP: Session Initiation Protocol”
(2) RFC2543 “SIP: Session Initiation Protocol”
(3) RFC2327 “SDP: Session Description Protocol”
(4) RFC1889 “RTP: A Transport Protocol for Real-Time Applications”

< THE TELECOMMUNICATION TECHNOLOGY COMMITTEE : TTC Standards >

(5) JT-H261 “Video Codec for Audiovisual Services at p x 64kbit/s”
(6) JT-H263 “Video Coding for Low Bitrate Communication”
(7) JT-G711 “Pulse Code Modulation (PCM) of Voice Frequencies
    Note : 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.
(8) JT-G722 “7 kHz Audio Coding within 64 kbit/s”
(9) JT-G728 “Coding of Speech at 16 kbit/s Using Low-Delay Code Excited Linear Prediction (LD-CELP)”
(10) JT-G723.1 “Dual Rate Speech Coder for Multimedia Telecommunication Transmitting at 5.3 & 6.3 kbit/s”
(11) JT-G729 “Coding of Speech at 8kbit/s using Conjugate-Structure Algebraic-Code-Excited Linear Prediction(CS-ACELP) “

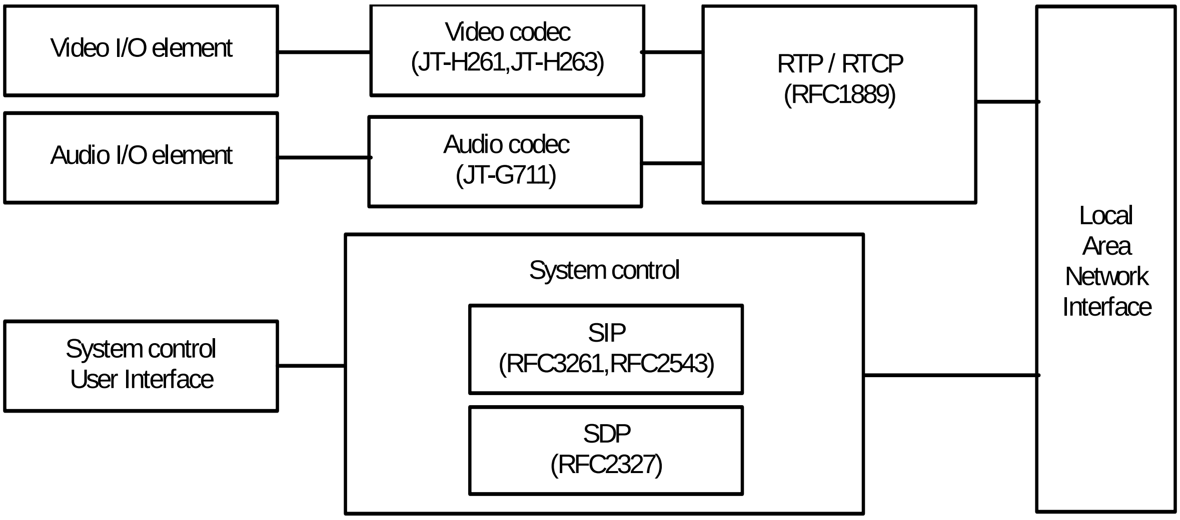


Figure 2.1 SIP multimedia communication terminal

## 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. 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.

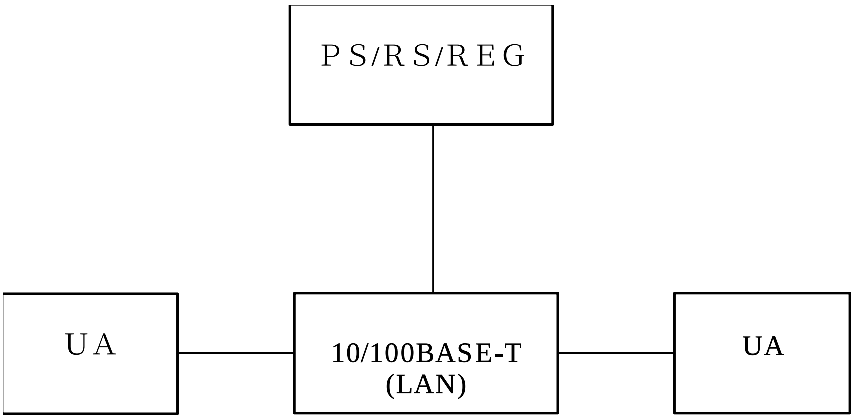


Figure 3.1 Connection between components

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, these components should not affect the performance of the components being tested.

(4) Prepare telephones for contact if the components are set up at different locations.

## 3-2. Execution method

(1) On the date arranged beforehand, the test must be 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.

The table of the sample combinations is shown in Appendix 2.

Note that the interoperability test between the products made by the same manufacturer is basically assumed to have been completed and is not included in the combination.

## 3-3. 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 other test items from the other terminal in accordance with the items listed in Appendix 1. Also, the encoding mode that has executed the communication for the caller or the receiver must be recorded respectively for 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 must confirm that the communication can be disconnected properly.

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

### 3-4. Test results handling

On completion of the test, after both the caller and the receiver check the results, the receiver has to fill in the check sheet in Appendix 1. If errors occur during the test, it is preferable to describe the situation as detailed as possible in the check sheet (phenomena, causes, actions, etc.).

If a re-test is required, indicate it in the MEMO section in the check sheet.

### 3-5. Test items

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) are optional.

(1) Confirmation of digital communications

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 may be added or modified if necessary.

### 3-6. 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 need for such tests are closely linked to the improvement of the terminal capabilities.

# 4. Results Handling and Future Issues

## 4-1. 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 must be recorded for future reference.

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

## 4-2. 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.

## Appendix1 Check Sheet
## SIP 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

<table>
<tr><th>№</th><th colspan="2">Item</th><th>Judging standard</th><th>Result<br>(Yes or No)</th><th colspan="2">Remarks (problem etc.)</th></tr>
<tr><td>1</td><td rowspan="6">Sending side(Terminal A)</td><td>Confirmation of audio communications</td><td rowspan="2">Confirm the communication of audio and the video in each mode.<br>Record the mode used.</td><td></td><td></td><td>Sending side encoding mode<br>Receiving side encoding mode</td></tr>
<tr><td>2</td><td>Confirmation of video communications</td><td></td><td></td><td>Sending side encoding mode<br>Receiving side encoding mode</td></tr>
<tr><td>3</td><td>Transmission rate of the video</td><td>Record the maximum transmission rate capability that was exchanged.</td><td>bps</td><td></td><td></td></tr>
<tr><td>4</td><td>Other</td><td></td><td></td><td></td><td></td></tr>
<tr><td>5</td><td>Disconnection by B</td><td>Confirm that Terminal A disconnected properly when Terminal B disconnected.</td><td></td><td></td><td></td></tr>
<tr><td>6</td><td>Disconnection by A</td><td>Confirm that Terminal A disconnected properly when Terminal A disconnected.</td><td></td><td></td><td></td></tr>
<tr><td>7</td><td rowspan="6">Sending side(Terminal B)</td><td>Confirmation of audio communications</td><td rowspan="2">Confirm the communication of audio and the video in each mode.<br>Record the mode used.</td><td></td><td></td><td>Sending side encoding mode<br>Receiving side encoding mode</td></tr>
<tr><td>8</td><td>Confirmation of video communications</td><td></td><td></td><td>Sending side encoding mode<br>Receiving side encoding mode</td></tr>
<tr><td>9</td><td>Transmission rate of the video</td><td>Record the maximum transmission rate capability that was exchanged.</td><td>bps</td><td></td><td></td></tr>
<tr><td>10</td><td>Other</td><td></td><td></td><td></td><td></td></tr>
<tr><td>11</td><td>Disconnection by A</td><td>Confirm that Terminal B disconnected properly when Terminal A disconnected.</td><td></td><td></td><td></td></tr>
<tr><td>12</td><td>Disconnection by B</td><td>Confirm that Terminal B disconnected properly when Terminal B disconnected.</td><td></td><td></td><td></td></tr>
</table>

- MEMO -

Appendix2 SIP Interoperability Test Combinations Matrix (Sample)

| | | 1<br>entry ABC<br>(model name:UA-A/serverB) | 2<br>entry XYZ<br>(model name:UA-C/serverD) | 3<br>entry XYZ<br>(model name:UA-E/serverD) | 4<br>entry ***<br>(model name:UA-F) | 5<br>entry ###<br>(model name:UA-G) |
|---|---|---|---|---|---|---|
| 1 | entry ABC<br>(model name:UA-A/serverB) | | A-1(serverB)<br>A-2(serverD) | C-1(serverB)<br>C-2(serverD) | D(serverB) | F(serverB) |
| 2 | entry XYZ<br>(model name:UA-C/serverD) | | | —<br>(Note2) | C(serverD) | E(serverD) |
| 3 | entry XYZ<br>(model name:UA-E/serverD) | | | | A(serverD) | D(serverD) |
| 4 | entry ***<br>(model name:UA-F) | | | | | B(w/o server) |
| 5 | entry ###<br>(model name:UA-G) | | | | | |

**[Test Date]**

| | Date/Month/Year | time schedule |
|---|---|---|
| A : | / / | 10:30—11:00 |
| B : | / / | 11:15—11:45 |
| C : | / / | 12:00—12:30 |
| D : | / / | 13:00—13:30 |
| E : | / / | 13:45—14:15 |
| F : | / / | 14:30—15:00 |

Note1) Both caller and callee tests are executed at each intersection of the matrix.

Note2) The interoperability test between the products made by the same manufacturer is assumed to have been completed and is not included in the combination.