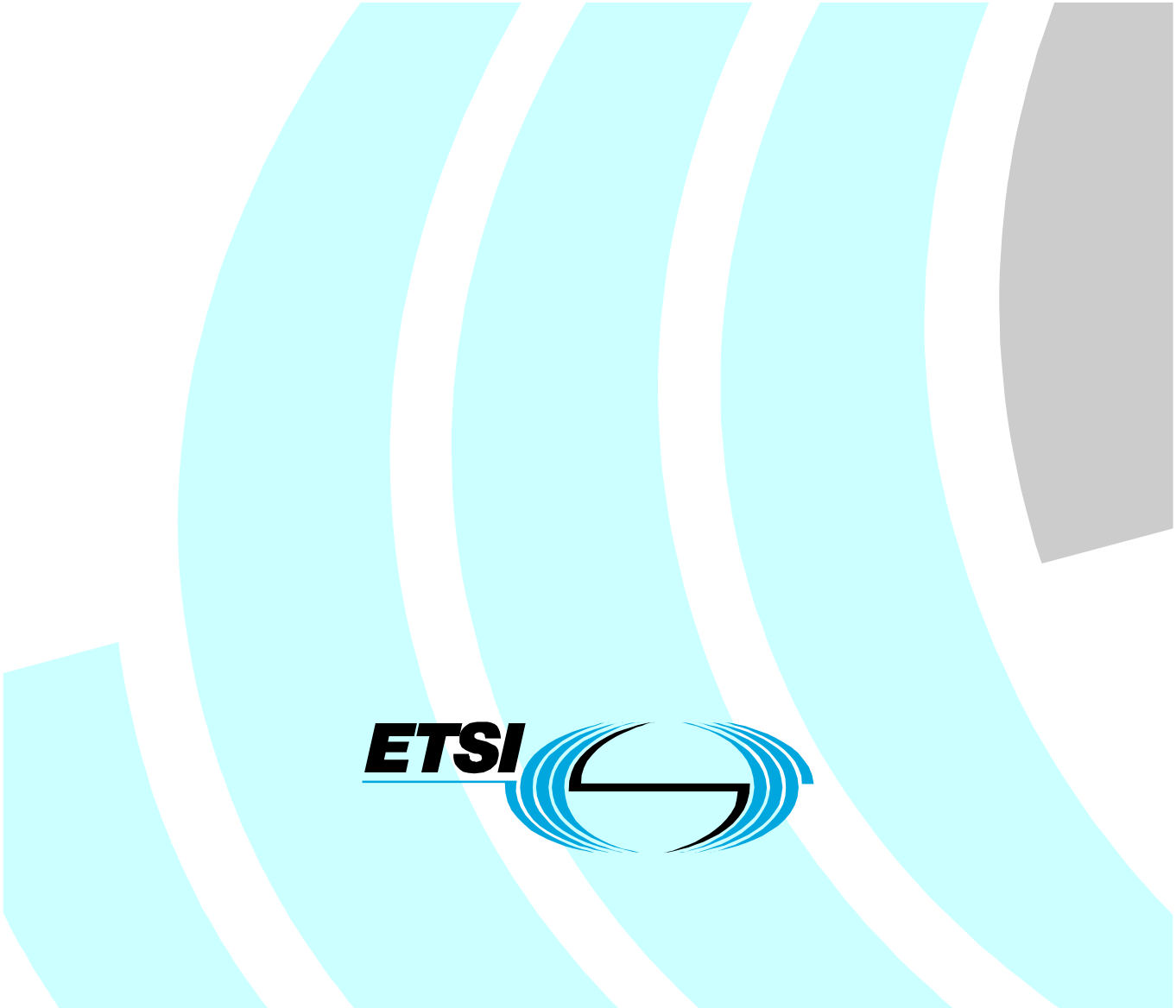
ETSI TS 102 237-2 V4.1.1 (2003-12)

Technical Specification

Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 4; Interoperability test methods and approaches; Part 2: H.323-SIP interoperability test scenarios to support multimedia communications in NGN environments



Reference
DTS/TISPAN-06009-02-TIPHON

Keywords
H.323, interoparability, IP, SIP, telephony,
testing, VoIP

ETSI

650 Route des Lucioles
F-06921 Sophia Antipolis Cedex - FRANCE

Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

Siret N° 348 623 562 00017 - NAF 742 C
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Foreword

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

The present document is part 2 of a multi-part deliverable covering Interoperability test methods and approaches, as identified below:

Part 1: "Generic approach to interoperability testing";

Part 2: "H.323-SIP interoperability test scenarios to support multimedia communications in NGN environments".

Introduction

The objective of the project team that produced the present document was to develop TIPHON Release 4 - H.323 - SIP Interoperability Test Scenarios". The other objective was to provide information, based on this particular exercise, for further development/enhancement/correction of the meta-protocol and the relevant profiles of the TIPHON technical specifications.

The initial analysis showed that the TIPHON documentation (H.323 and SIP profiles) for Release 4 was not finalized at the time of writting the present document. So the present document concentrates on the TIPHON Release 3 documentation.

To prepare the present document, the project team used the following publications as input: TR 101 308 [14], TS 101 314 [5], TS 101 315 [6], TS 101 882 [8], TS 101 883 [11], TS 101 884 [12], TS 101 878 [7], ITU-T Recommendation H.323 [1], RFC 2327 [10] and RFC 3261 [9] standards.

1 Scope

The present document describes an H.323-SIP Interoperability Test Suite focusing on test cases for functionality that include protocol interworking. It is based on the documents TS 101 883 [11] (H.323 Profile) and TS 101 884 [12] (SIP Profile) of TIPHON Release 3.

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 and/or edition number or version number) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies.

Referenced documents which are not found to be publicly available in the expected location might be found at http://docbox.etsi.org/Reference.

[1] ITU-T Recommendation H.323 (2000): "Packet-based multimedia communications systems".

[2] ITU-T Recommendation H.225.0 (2000): "Call signalling protocols and media stream packetization for packet-based multimedia communication systems".

[3] ITU-T Recommendation H.245 (2000): "Control protocol for multimedia communication".

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

[5] ETSI TS 101 314: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 3; Abstract Architecture and Reference Points Definition; Network Architecture and Reference Points".

[6] ETSI TS 101 315: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 3; Functional Entities, Information Flow and Reference Point Definitions; Guidelines for application of TIPHON functional architecture to inter-domain services".

[7] ETSI TS 101 878: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 3; Service Capability Definition; Service Capabilities for a simple call".

[8] ETSI TS 101 882: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 3; Protocol Framework Definition; General (meta-protocol)".

[9] IETF RFC 3261: "SIP: Session Initiations Protocol".

[10] IETF RFC 2327: "SDP: Session Description Protocol".

[11] ETSI TS 101 883: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 3; Technology Mapping; Implementation of TIPHON architecture using H.323".

[12] ETSI TS 101 884: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 3; Technology Mapping; Implementation of TIPHON architecture using SIP".

[13] IETF RFC 2833:"RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals".

[14] ETSI TR 101 308: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON); Requirements Definition Study, SIP and H.323 Interworking".

3 Definitions and abbreviations

3.1 Definitions

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

End-Point (EP): can be a terminal or a gateway

In-Band DTMF: DTMF signals are sent on the media channel according to RFC 2833 [13]

Out-Band DTMF: DTMF signals are sent in a H.245 IndicationMessage, in field userInput

SIP Server: SIP entitiy comprising a SIP Proxy and a SIP Registrar

3.2 Abbreviations

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

BC	Basic Call
CC	Call Clearing
CEE	Call Establishment EnBLoc-sending
CEO	Call Establishment Overlap-sending
CLIP	Calling Line Identity Presentation
CLIR	Calling Line Identity Restriction
COLP	COnnected Line identity Presentation
COLR	COnnected Line identity Restriction
CPB	Called Party Busy
CPNA	Called Party does Not Answer
DR	DeRegistration
DTMF	Dual Tone Multi-Frequency
EP	EndPoint
EUT	Equipment Under Test
GK	GateKeeper
HO	H.323 Originated
IWF	InterWorking Function
NRIS	No Registration Indication Support
QE	Qualified Equipment
RE	REgistration
RIS	Registration Indication Support
SO	SIP Originated
SS	Supplementary Services
TP	Test Purposes
TSS	Test Suite Structure

4 Methodology

The process used in this work is outlined in the diagram below. Initially the H.323 [1] and SIP [9] standards are used as references to study and verify the protocol profiles of TIPHON H.323, TS 101 883 [11], and TIPHON SIP, TS 101 884 [12]. The results were used to generate the interoperability testing specifications.

Output of this work may lead to contributions to other working groups in TIPHON on any deficiencies and problems discovered whilst performing this work.

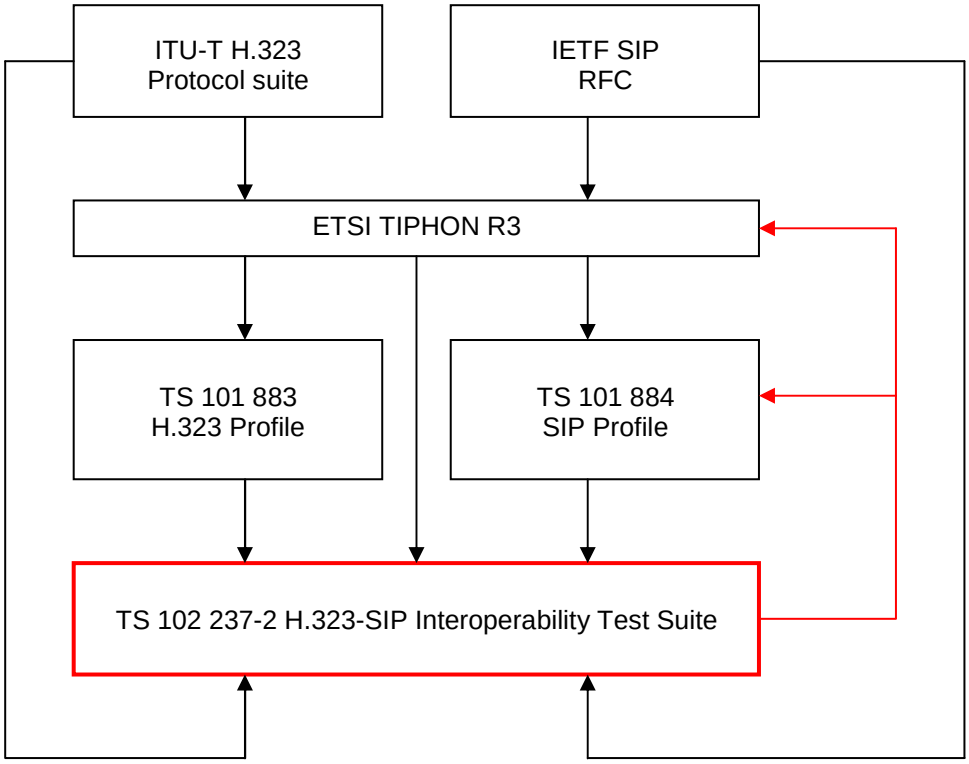

Figure 1: Methodology

5 Abstract architectures

The abstract architectures show the individual entities of the test architecture, as well as the human test operators.

We recommend that interoperability testing is performed in a way that one entity Equipment Under Test (EUT) is tested with equipment that already proved interoperability Qualified Equipment (QE). There are no restrictions in regard to which entity of the test architecture actually may be the EUT.

The following architectures give examples for a possible division into EUT and QE.

5.1 H.323-administered architecture

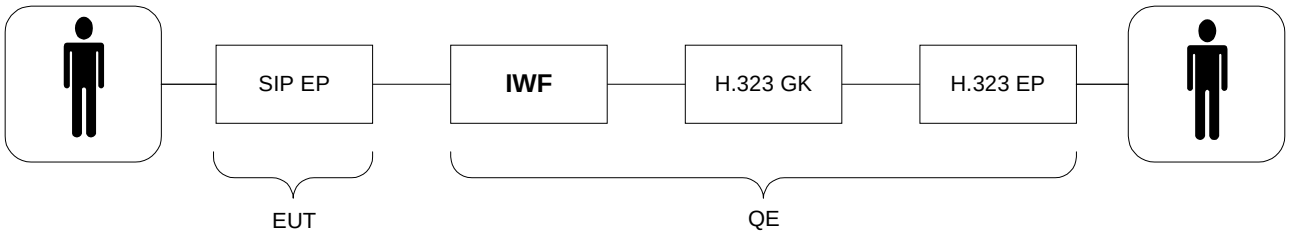

Figure 2: H.323-administered architecture

5.2 SIP-administered architecture

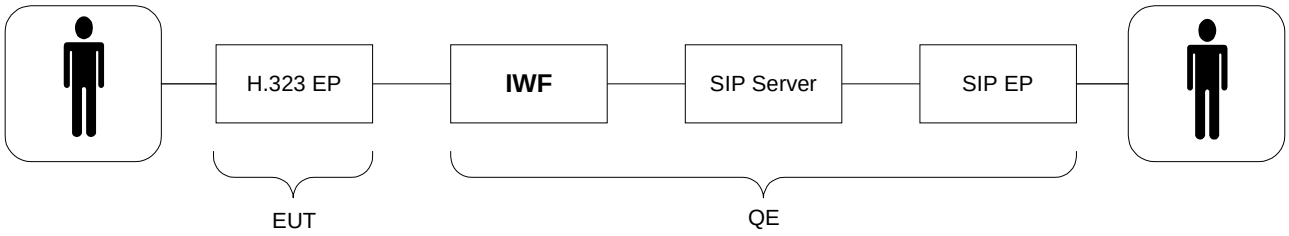

Figure 3: SIP-administered architecture

5.3 Architectures with two administration authorities

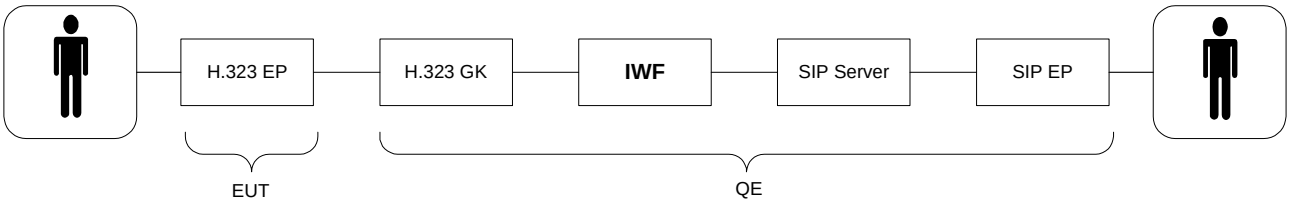

Figure 4: Architecture with two administration authorities

6 Test Suite Structure (TSS)

The Test Suite Structure (TSS) follows the network architecture and the protocol architecture. The first two levels are determined by the desired functionality. The third level is defined by the protocol of the originating side.

Table 1: H.323-SIP interoperability TSS

Functionality	Sub-Functionality	Originating Protocol
Registration (RE)		
	Registration Indication Support (RIS)	
		H.323 Originated (HO)
		SIP Originated (SO)
	No Registration Indication Support (NRIS)	
		H.323 Originated (HO)
		SIP Originated (SO)
	Deregistration (DR)	
		H.323 Originated (HO)
		SIP Originated (SO)
Basic Call (BC)		
	Call Establishment EnBLoc-Sending (CEE)	
		H.323 Originated (HO)
		SIP Originated (SO)
	Call Establishment Overlap-Sending (CEO)	
		H.323 Originated (HO)
		SIP Originated (SO)
	Call Clearing (CC)	
		H.323 Originated (HO)
		SIP Originated (SO)
	Called Party Busy (CPB)	
		H.323 Originated (HO)
		SIP Originated (SO)

Functionality	Sub-Functionality	Originating Protocol
	Called Party does not answer (CPNA)	
		H.323 Originated (HO)
		SIP Originated (SO)
Supplementary Services (SS)		
	Calling Line Identity presentation (CLIP)	
		H.323 Originated (HO)
		SIP Originated (SO)
	Calling Line Identity Restriction (CLIR)	
		H.323 Originated (HO)
		SIP Originated (SO)
	COnnected Line identity Presentation (COLP)	
		H.323 Originated (HO)
		SIP Originated (SO)
	COnnected Line identity Restriction (COLR)	
		H.323 Originated (HO)
		SIP Originated (SO)
	Dual Tone Multi-Frequency (DTMF)	
		H.323 Originated (HO)
		SIP Originated (SO)

7 Test Purposes (TP)

7.1 TP naming convention

The naming scheme was chosen to be able to easily enter additional test cases in case of adaptions.

Table 2: Numbering scheme

Identifier: **<funct>_<nnn>**

<funct> = RIS, NRIS, DR, CEE, CEO, CC, CBA, CPB. CPNA, CLIP, CLIR, COLP, COLR, DTMF

<nnn> = sequential number (01 to 99)

7.2 Test strategy

As the TIPHON profiles for H.323 and SIP contain no explicit requirements for testing, the TPs were generated as a result of the profiles' analysis. The TPs are based on functional requirements for interoperability testing.

7.3 Interoperability TPs

7.3.1 REgistration (RE)

7.3.1.1 Registration Indication Supported (RIS)

RIS_01
To verify that a user whose terminal indicates registration is able to register at a SIP Server using a H.323 EP with manual GK discovery

RIS_02
To verify that a user whose terminal indicates registration is able to register at a SIP Server using a H.323 EP with automatic GK discovery

RIS_03
To verify that a user whose terminal indicates registration is able to register to the H.323 GK by means of a manually configured SIP EP

RIS_04
To verify that a user whose terminal indicates registration is able to register to the H.323 GK by means of a SIP EP using multicast discovery

7.3.1.2 No Registration Indication Supported (NRIS)

NRIS_01
To verify that a user whose terminal does not indicate registration is able to register at a SIP Server using a H.323 EP with manual GK discovery

NRIS_02
To verify that a user whose terminal does not indicate registration is able to register at a SIP Server using a H.323 EP with automatic GK discovery

NRIS_03
To verify that a user whose terminal does not indicate registration is able to register to the H.323 GK by means of a manually configured SIP EP

NRIS_04
To verify that a user whose terminal does not indicate registration is able to register to the H.323 GK by means of a SIP EP using multicast discovery

7.3.1.3 DeRegistration (DR)

DR_01
To verify that a H.323 EP is able to deregister from a SIP Server

DR_02
To verify that a SIP EP is able to deregister from a H.323 GK

7.3.2 Basic Call (BC)

7.3.2.1 Call Establishment EnbBloc-sending (CEE)

CEE_01
To verify that a call can be established successfully from a H.323 EP to a SIP EP using EnBloc-sending, that the media channels are opened after the call establishment and that a ring tone is heard at the calling party

CEE_02
To verify that a call can be established successfully from a SIP EP to a H.323 EP using EnBloc-sending, that the media channels are opened after the call establishment and that a ring tone is heard at the calling party

7.3.2.2 Call Establishment Overlap-sending (CEO)

CEO_01
To verify that a call can be established successfully from a H.323 EP to a SIP EP using Overlap-sending, that the media channels are opened after the call establishment and that a ring tone is heard at the calling party

CEO_02
To verify that a call can be established successfully from a SIP EP to a H.323 EP using Overlap-sending, that the media channels are opened after the call establishment and that a ring tone is heard at the calling party

7.3.2.3 Call Clearing (CC)

CC_01
To verify that an established call, originated by a H.323 EP, can be released by the calling H.323 party and that releasing the call also clears the media channel

CC_02
To verify that an established call, originated by a H.323 EP, can be released by the called SIP EP and that releasing the call also clears the media channel

CC_03
To verify that an established call, originated by a SIP EP, can be released by the calling SIP party and that releasing the call also clears the media channel

CC_04
To verify that an established call, originated by a SIP EP, can be released by the called H.323 EP and that releasing the call also clears the media channel

CC_05
To verify that a call attempt from the H.323 EP may be released by the calling H.323 EP before the called SIP EP answers

CC_06
To verify that a call attempt from the SIP EP may be released by the calling SIP EP before the called H.323 EP answers

7.3.2.4 Called Party Busy (CPB)

CPB_01
To verify that a call attempt from a H.323 EP to a busy SIP user delivers a busy indication at the calling party

CPB_02
To verify that a call attempt from a SIP EP to a busy H.323 user delivers a busy indication at the calling party

7.3.2.5 Called Narty does Not Answer (CPNA)

CPNA_01
To verify that a call attempt from a H.323 EP to a SIP EP is cleared and that the calling H.323 party receives a "No answer"-indication if the SIP EP does not answer in a certain time period

CPNA_02
To verify that a call attempt from a H.323 EP to a SIP EP is cleared and that the calling SIP party receives a "No answer"-indication if the H.323 EP does not answer in a certain time period

7.3.3 Supplementary Services (SS)

7.3.3.1 CLIP

CLIP_01
To verify that the line identity of a calling H.323 EP is presented to the called SIP EP

CLIP_02
To verify that the line identity of a calling SIP EP is presented to the called H.323 EP

7.3.3.2 CLIR

CLIR_01
To verify that the transmitted line identity of a calling H.323 EP is not presented to the called SIP EP if CLIR is enabled for the calling party

CLIR_02
To verify that the transmitted line identity of a calling SIP EP is not presented to the called H.323 EP if CLIR is enabled for the calling party

7.3.3.3 COLP

COLP_01
To verify that the line identity of a SIP EP called from a H.323 EP is presented to the calling H.323 EP on connection

COLP_02
To verify that the line identity of a H.323 EP called from a SIP EP is presented to the calling SIP EP on connection

7.3.3.4 COLR

COLR_01
To verify that the line identity of a SIP EP called from a H.323 EP is not presented to the calling H.323 EP on connection if COLR is enabled for the called party

COLR_02
To verify that the line identity of a H.323 EP called from a SIP EP is not presented to the calling SIP EP on connection if COLR is enabled for the called party

7.3.3.5 DTMF

DTMF_01
To verify that the calling party in a H.323 originated and established call is able to send In-Band DTMF to the called party

DTMF_02
To verify that the called SIP EP in a H.323 originated and established call is able to send In-Band DTMF to the calling party

DTMF_03
To verify that the calling party in a SIP originated and established call is able to send In-Band DTMF to the called party

DTMF_04
To verify that the called H.323 EP in a SIP originated and established call is able to send In-Band DTMF to the calling party

DTMF_05
To verify that the calling party in a H.323 originated and established call is able to send Out-Band DTMF to the called party

DTMF_06
To verify that the called SIP EP in a H.323 originated and established call is able to send Out-Band DTMF to the calling party

DTMF_07
To verify that the calling party in a SIP originated and established call is able to send Out-Band DTMF to the called party

DTMF_08
To verify that the called H.323 EP in a H.323 originated and established call is able to send Out-Band DTMF to the calling party

8 Interoperability test suite

8.1 Configuration

All entities of the test architecture shall be configured according to the TIPHON H.323 profile TS 101 883 [11] and the TIPHON SIP profile TS 101 884 [12].

Rough configuration information concerning the individual test cases is given in the section "Pre-Test Condition" in each test case.

8.2 Endpoint RE

8.2.1 EP supports registration indication

8.2.1.1 H.323 EP registers at SIP server using manual GK discovery

Test: RIS_01	**Selection Criteria:** Optional	**Selected:** Yes No
Title:	H.323 EP registers at SIP Server using manual GK discovery	
Test Purpose:	To verify that a user whose terminal indicates registration is able to register at a SIP Server using a H.323 EP with manual GK discovery	
Abstract Architecture:	SIP-administered architecture, clause 5.2, figure 3	
Pre-test conditions:	• Provide H.323 EP with the IWF's RAS address • Configure the H.323 EP to use manual GK discovery • Configure SIP Server with the user's registration information • H.323 EP supports registration indication	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate registration**		
2	*Check: Is H.323EP indicating registration (visual indicator)?*	Yes	No
Observations:			

8.2.1.2 H.323 EP registers to SIP server using automatic GK discovery

Test: RIS_02	**Selection Criteria:** Optional	**Selected:** Yes No
Title:	H.323 EP registers at SIP Server using automatic GK discovery	
Test Purpose:	To verify that a user whose terminal indicates registration is able to register at a SIP Server using a H.323 EP with automatic GK discovery	
Abstract Architecture:	SIP-administered architecture, clause 5.2, figure 3	
Pre-test conditions:	• Configure the H.323 EP to use automatic GK discovery • Configure SIP Server with the user's registration information • H.323 EP supports registration indication	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate registration**		
2	*Check: Is H.323EP indicating registration (visual indicator)?*	Yes	No
Observations:			

8.2.1.3 SIP EP registers to H.323 GK using manual configuration

Test: RIS_03	**Selection Criteria:** Optional	**Selected:** Yes No
Title:	SIP EP registers to H.323 GK using manual configuration	
Test Purpose:	To verify that a user whose terminal indicates registration is able to register to the H.323 GK by means of a manually configured SIP EP	
Abstract Architecture:	H.323-Administered architecture, clause 5.1, figure 2	
Pre-test conditions:	• Provide SIP EP with the IWF's registrar address • Configure H.323 GK with user's registration information • SIP EP supports Registration Indication	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate registration**		
2	*Check: Is SIP EP indicating registration (visual indicator)?*	Yes	No
Observations:			

8.2.1.4 SIP EP registers to H.323 GK using multicast discovery

Test: RIS_04	**Selection Criteria:** Optional	**Selected:** Yes No
Title:	SIP EP registers to H.323 GK	
Test Purpose:	To verify that a user whose terminal indicates registration is able to register to the H.323 GK by means of a SIP EP using multicast discovery	
Abstract Architecture:	H.323-Administered architecture, clause 5.1, figure 2	
Pre-test conditions:	• Configure SIP EP to use multicast discovery • Configure H.323 GK with user's registration information • SIP EP supports Registration Indication	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate registration**		
2	*Check: Is SIP EP indicating registration (visual indication)?*	Yes	No
Observations:			

8.2.2 EP does not support registration indication

8.2.2.1 H.323 EP registers at SIP server using manual GK discovery

Test: NRIS_01	**Selection Criteria:** Optional	**Selected:** Yes No
Title:	H.323 EP registers at SIP Server using manual GK discovery	
Test Purpose:	To verify that a user whose terminal does not indicate registration is able to register at a SIP Server using a H.323 EP with manual GK discovery	
Abstract Architecture:	SIP-administered architecture, clause 5.2, figure 3	
Pre-test conditions:	• Provide H.323 EP with the IWF's RAS address • Configure the H.323 EP to use manual GK discovery • Configure SIP Server with the user's registration information • Configure testbed to initiate a call • H.323 EP does not support registration indication	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call**		
2	*Check: Call attempt is permitted?*	No	Yes
3	**Initiate registration**		
4	**Initiate call**		
5	*Check: Call attempt is permitted?*	Yes	No
Observations:			

8.2.2.2 H.323 EP registers at SIP server using automatic GK discovery

Test: NRIS_02	**Selection Criteria:** Optional	**Selected:** Yes No
Title:	H.323 EP registers at SIP Server using automatic GK discovery	
Test Purpose:	To verify that a user whose terminal does not indicate registration is able to register at a SIP Server using a H.323 EP with automatic GK discovery	
Abstract Architecture:	SIP-administered architecture, clause 5.2, figure 3	
Pre-test conditions:	• Configure the H.323 EP to use automatic GK discovery • Configure SIP Server with the user's registration information • Configure testbed to initiate a call • H.323 EP does not support registration indication	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call**		
2	*Check: Call attempt is permitted?*	No	Yes
3	**Initiate registration**		
4	**Initiate call**		
5	*Check: Call attempt is permitted?*	Yes	No
Observations:			

8.2.2.3 SIP EP registers to H.323 GK using manual configuration

Test: NRIS_03	**Selection Criteria:** Optional	**Selected:** Yes No
Title:	SIP EP registers to H.323 GK using manual configuration	
Test Purpose:	To verify that a user whose terminal does not indicate registration is able to register to the H.323 GK by means of a manually configured SIP EP	
Abstract Architecture:	H.323-Administered architecture, clause 5.1, figure 2	
Pre-test conditions:	• Provide SIP EP with the IWF's registrar address • Configure H.323 GK with user's registration information • Configure Testbed to initiate a call • SIP EP does not support registration Indication	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call**		
2	*Check: Call attempt is permitted?*	No	Yes
3	**Initiate registration**		
4	**Initiate call**		
5	*Check: Call attempt is permitted?*	Yes	No
Observations:			

8.2.2.4 SIP EP registers to H.323 GK using multicast discovery

Test: NRIS_04	**Selection Criteria:** Optional	**Selected:** Yes No
Title:	SIP EP registers to H.323 GK	
Test Purpose:	To verify that a user whose terminal does not indicate registration is able to register to the H.323 GK by means of a SIP EP using multicast discovery	
Abstract Architecture:	H.323-Administered architecture, clause 5.1, figure 2	
Pre-test conditions:	• Configure SIP EP to use multicast discovery • Configure H.323 GK with user's registration information • Configure Testbed to initiate a call • SIP EP supports Registration Indication	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call**		
2	*Check: Call attempt is permitted?*	No	Yes
3	**Initiate registration**		
4	**Initiate call**		
5	*Check: Call attempt is permitted?*	Yes	No
Observations:			

8.2.3 Deregistration

8.2.3.1 H.323 EP deregisters

Test: DR_01	**Selection Criteria:** Mandatory	**Selected:** Yes ~~No~~
Title:	H.323 EP deregisters from SIP Server	
Test Purpose:	To verify that a H.323 EP is able to deregister from a SIP Server	
Abstract Architecture:	SIP-administered architecture, clause 5.2, figure 3	
Pre-test conditions:	• H.323 EP must be registered at SIP Server	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate deregistration at H.323 EP**		
2	*Check: Is terminal indicating deregistration (visual(indicator) or audible indication(dial tone))?*	Yes	No
Observations:			

8.2.3.2 SIP EP deregisters

Test: DR_02	**Selection Criteria:** Mandatory	**Selected:** Yes ~~No~~
Title:	SIP EP deregisters from H.323 GK	
Test Purpose:	To verify that a SIP EP is able to deregister from a H.323 GK	
Abstract Architecture:	H-323-administered architecture, clause 5.1, figure 2	
Pre-test conditions:	• SIP EP must be registered at H.323 GK	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate deregistration at SIP EP**		
2	*Check: Is terminal indicating deregistration (visual(indicator) or audible indication(dial tone))?*	Yes	No
Observations:			

8.3 Basic Call establishment

The test cases in this clause are valid for all test architectures. Therefore, there is no section "abstract architecture" in the test cases.

The test cases of clauses 8.3.1 and 8.3.2 shall be run for each supported codec.

8.3.1 Call Establishment with EnBloc-Sending

8.3.1.1 Call Establishment from H.323 EP to SIP EP

Test: CEE_01	**Selection Criteria:** Mandatory	**Selected:** Yes ~~No~~
Title:	Call establishment from H.323 EP to SIP EP	
Test Purpose:	To verify that a call can be established successfully from a H.323 EP to a SIP EP using EnBloc-Sending, that the media channels are opened after the call establishment and that a ring tone is heard at the calling party	
Pre-test conditions:	• Configure equipment to use EnBloc-Sending • Both EPs must have successfully registered	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call from H.323 EP to SIP EP**		
2	*Check: Is Ringing Tone heard at H.323 EP?*	Yes	No
3	*Check: Is SIP EP alerting (visual or audible indication)?*	Yes	No
4	*Check: Can voice from SIP EP be heard at H.323 EP?*	No	Yes
5	*Check: Can voice from H.323 EP be heard at SIP EP?*	No	Yes
6	**Accept call at SIP EP**		
7	*Check: Is Ringing Tone heard at H.323 EP?*	No	Yes
8	*Check: Is SIP EP alerting?*	No	Yes
9	*Check: Can voice from SIP EP be heard at H.323 EP?*	Yes	No
10	*Check: Can speech from H.323 EP be heard at SIP EP?*	Yes	No
11	**Clear call at H.323 EP or SIP EP**		

Observations:

8.3.1.2 Call Establishment from SIP EP to H.323 EP

Test: CEE_02	**Selection Criteria:** Mandatory	**Selected:** Yes ~~No~~
Title:	Call establishment from SIP EP to H.323 EP	
Test Purpose:	To verify that a call can be established successfully from a SIP EP to a H.323 EP using EnBloc-Sending, that the media channels are opened after the call establishment and that a ring tone is heard at the calling party	
Pre-test conditions:	• Configure equipment to use EnBloc-Sending • Both EPs must have successfully registered	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call from SIP EP to H.323 EP**		
2	*Check: Is Ringing Tone heard at SIP EP?*	Yes	No
3	*Check: Is H.323 EP alerting (visual or audible indication)?*	Yes	No
4	*Check: Can voice from SIP EP be heard at H.323 EP?*	No	Yes
5	*Check: Can voice from H.323 EP be heard at SIP EP?*	No	Yes
6	**Accept call at H.323 EP**		
7	*Check: Is Ringing Tone heard at SIP EP?*	No	Yes
8	*Check: Is H.323 EP alerting?*	No	Yes
9	*Check: Can voice from SIP EP be heard at H.323 EP?*	Yes	No
10	*Check: Can speech from H.323 EP be heard at SIP EP?*	Yes	No
11	**Clear call at SIP EP or H.323 EP**		

Observations:

8.3.2 Call Establishment with Overlap-Sending

8.3.2.1 Call Establishment from H.323 EP to SIP EP

Test: CEO_01	**Selection Criteria:** Mandatory	**Selected:** Yes ~~No~~
Title:	Call establishment from H.323 EP to SIP EP	
Test Purpose:	To verify that a call can be established successfully from a H.323 EP to a SIP EP using Overlap-Sending, that the media channels are opened after the call establishment and that a ring tone is heard at the calling party	
Pre-test conditions:	• Configure equipment to use Overlap-Sending • Both EPs must have successfully registered	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call from H.323 EP to SIP EP**		
2	*Check: Is Ringing Tone heard at H.323 EP?*	Yes	No
3	*Check: Is SIP EP alerting (visual or audible indication)?*	Yes	No
4	*Check: Can voice from SIP EP be heard at H.323 EP?*	No	Yes
5	*Check: Can voice from H.323 EP be heard at SIP EP?*	No	Yes
6	**Accept call at SIP EP**		
7	*Check: Is Ringing Tone heard at H.323 EP?*	No	Yes
8	*Check: Is SIP EP alerting (visual or audible indication)?*	No	Yes
9	*Check: Can voice from SIP EP be heard at H.323 EP?*	Yes	No
10	*Check: Can voice from H.323 EP be heard at SIP EP?*	Yes	No
11	**Clear call at H.323 EP or SIP EP**		
Observations:			

8.3.2.2 Call Establishment from SIP EP to H.323 EP

Test: CEO_02	**Selection Criteria:** Mandatory	**Selected:** Yes ~~No~~
Title:	Call establishment from SIP EP to H.323 EP	
Test Purpose:	To verify that a call can be established successfully from a SIP EP to a H.323 EP using Overlap-Sending, that the media channels are opened after the call establishment and that a ring tone is heard at the calling party	
Pre-test conditions:	• Configure equipment to use Overlap sending • Both EPs must have successfully registered	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call from SIP EP to H.323 EP**		
2	*Check: Is Ringing Tone heard at SIP EP?*	Yes	No
3	*Check: Is H.323 EP alerting (visual or audible indication)?*	Yes	No
4	*Check: Can voice from SIP EP be heard at H.323 EP?*	No	Yes
5	*Check: Can voice from H.323 EP be heard at SIP EP?*	No	Yes
6	**Accept call at H.323 EP**		
7	*Check: Is Ringing Tone heard at SIP EP?*	No	Yes
8	*Check: Is H.323 EP alerting (visual or audible indication)?*	No	Yes
9	*Check: Can voice from SIP EP be heard at H.323 EP?*	Yes	No
10	*Check: Can voice from H.323 EP be heard at SIP EP?*	Yes	No
11	**Clear call at SIP EP or H.323 EP**		
Observations:			

8.3.3 Call Clearing

8.3.3.1 H.323 EP calls SIP EP and H.323 EP clears the call

Test: CC_01	**Selection Criteria:** Mandatory	**Selected:** Yes ~~No~~
Title:	H.323 EP calls SIP EP and H.323 EP clears the call	
Test Purpose:	To verify that an established call, originated by a H.323 EP, can be released by the calling H.323 party and that releasing the call also clears the media channel	
Pre-test conditions:	• Configure equipment to be able to establish a call • Both parties must be registered	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call from H.323 EP to SIP EP**		
2	**Accept call at SIP EP**		
3	**Clear call at H.323 EP**		
4	*Check: Is call released(call signalling)?*	Yes	No
5	*Check: Are voice channels cleared?*	Yes	No
Observations:			

8.3.3.2 H.323 EP calls SIP EP and SIP EP clears the call

Test: CC_02	**Selection Criteria:** Mandatory	**Selected:** Yes ~~No~~
Title:	H.323 EP calls SIP EP and SIP EP clears the call	
Test Purpose:	To verify that an established call, originated by a H.323 EP, can be released by the called SIP EP and that releasing the call also clears the media channel	
Pre-test conditions:	• Configure equipment to be able to establish a call • Both parties must be registered	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call from H.323 EP to SIP EP**		
2	**Accept call at SIP EP**		
3	**Clear call at SIP EP**		
4	*Check: Is call released(call signalling)?*	Yes	No
5	*Check: Are voice channels cleared?*	Yes	No
Observations:			

8.3.3.3 SIP EP calls H.323 EP and SIP EP clears the call

Test: CC_03	**Selection Criteria:** Optional	**Selected:** Yes No
Title:	SIP EP calls H.323 EP and SIP EP clears the call	
Test Purpose:	To verify that an established call, originated by a SIP EP, can be released by the calling SIP party and that releasing the call also clears the media channel	
Pre-test conditions:	• Configure equipment to be able to establish a call • Both parties must be registered	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call from SIP EP to H.323 EP**		
2	**Accept call at H.323 EP**		
3	**Clear call at SIP EP**		
4	*Check: Is call released(call signalling)?*	Yes	No
5	*Check: Are voice channels cleared?*	Yes	No
Observations:			

8.3.3.4 SIP EP calls H.323 EP and H.323 EP clears the call

Test: CC_04	**Selection Criteria:** Optional	**Selected:** Yes No
Title:	SIP EP calls H.323 EP and H.323 EP clears the call	
Test Purpose:	To verify that an established call, originated by a SIP EP, can be released by the called H.323 EP and that releasing the call also clears the media channel	
Pre-test conditions:	• Configure equipment to be able to establish a call • Both parties must be registered	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call from SIP EP to H.323 EP**		
2	**Accept call at H.323 EP**		
3	**Clear call at H.323 EP**		
4	*Check: Is call released(call signalling)?*	Yes	No
5	*Check: Are voice channels cleared?*	Yes	No
Observations:			

8.3.3.5 H.323 EP calls SIP EP and H.323 EP clears call before SIP EP answers

Test: CC_05	**Selection Criteria:** Mandatory	**Selected:** Yes ~~No~~
Title:	H.323 EP calls SIP EP and H.323 EP clears call before SIP EP answers	
Test Purpose:	To verify that a call attempt from the H.323 EP may be released by the calling H.323 EP before the called SIP EP answers	
Pre-test conditions:	• Configure equipment to be able to establish a call • Both parties must be registered	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call from H.323 EP to SIP EP**		
2	**Clear call at H.323 EP**		
3	*Check: Is the call released(call signalling)?*	Yes	No
4	*Check: Are the voice channels established?*	No	Yes
Observations:			

8.3.3.6 SIP EP calls H.323 EP and SIP EP clears call before H.323 EP answers

Test: CC_07	**Selection Criteria:** Mandatory	**Selected:** Yes ~~No~~
Title:	SIP EP calls H.323 EP and SIP EP clears call before H.323 EP answers	
Test Purpose:	To verify that a call attempt from the SIP EP may be released by the calling SIP EP before the called H.323 EP answers	
Pre-test conditions:	• Configure equipment to be able to establish a call • Both parties must be registered	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call from SIP EP to H:323 EP**		
2	**Clear call at SIP EP**		
3	*Check: Is the call released(call signalling)?*	Yes	No
4	*Check: Are the voice channels established?*	No	Yes
Observations:			

8.3.4 Called Party is busy

8.3.4.1 H.323 EP calls SIP EP and SIP EP is busy

Test: CPB_01	**Selection Criteria:** Mandatory	**Selected:** Yes ~~No~~
Title:	H.323 EP calls SIP EP and SIP EP is busy	
Test Purpose:	To verify that a call attempt from a H.323 EP to a busy SIP user delivers a busy indication at the calling party	
Pre-test conditions:	• Configure equipment to be able to establish a call to a busy user • Both parties must be registered	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1 2	**Initiate call from third party to SIP EP or vice versa** **Accept call at called party**		
3 4	**Initiate call from H.323 EP to SIP EP** *Check: Is Busy indication received (audible or visual) at H.323 EP?*	Yes	No
5 6	**Clear call at H.323 EP** **Clear call at SIP EP or Third party**		
Observations:			

8.3.4.2 SIP EP calls H.323 EP and H.323 EP is busy

Test: CPB_02	**Selection Criteria:** Mandatory	**Selected:** Yes ~~No~~
Title:	SIP EP calls H.323 EP and H.323 EP is busy	
Test Purpose:	To verify that a call attempt from a SIP EP to a busy H.323 user delivers a busy indication at the calling party	
Pre-test conditions:	• Configure equipment to be able to establish a call to a busy user • Both parties must be registered	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1 2	**Initiate call from third party to H.323 EP or vice versa** **Accept call at called party**		
3 4	**Initiate call from SIP EP to H.323 EP** *Check: Is Busy indication received (audible or visual) at SIP EP?*	Yes	No
5 6	**Clear call at SIP EP** **Clear call at H.323 EP or Third party**		
Observations:			

8.3.5 Called Party does not answer

8.3.5.1 H.323 EP calls SIP EP and SIP EP does not answer

Test: CPNA_01	**Selection Criteria:** Mandatory	**Selected:** Yes ~~No~~
Title:	H.323 EP calls SIP EP and SIP EP does not answer	
Test Purpose:	To verify that a call attempt from a H.323 EP to a SIP EP is cleared and that the calling H.323 party receives a "No answer"-indication if the SIP EP does not answer in a certain time period	
Pre-test conditions:	• Configure equipment to be able to establish a call • Both parties must be registered • Configure call initiation timeout	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call from H.323 EP to SIP EP**		
2	*Check: Is "No Answer" indication received (audible or visual) at H.323 EP?*	Yes	No
3	*Check: Does H.323 EP stop call attempt after timeout?*	Yes	No
4	*Check: Is the SIP EP still alerting?*	No	Yes
Observations:			

8.3.5.2 SIP EP calls H.323 EP and H.323 EP does not answer

Test: CPNA_02	**Selection Criteria:** Mandatory	**Selected:** Yes ~~No~~
Title:	SIP EP calls H.323 EP and H.323 EP does not answer	
Test Purpose:	To verify that a call attempt from a H.323 EP to a SIP EP is cleared and that the calling SIP party receives a "No answer"-indication if the H.323 EP does not answer in a certain time period	
Pre-test conditions:	• Configure equipment to be able to establish a call • Both parties must be registered • Configure call initiation timeout	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call from SIP EP to H.323 EP**		
2	*Check: Is "No Answer" indication received (audible or visual) at SIP EP?*	Yes	No
3	*Check: Does SIP EP stop call attempt after timeout?*	Yes	No
4	*Check: Is the H.323 EP still alerting?*	No	Yes
Observations:			

8.4 Supplementary Services (SS)

8.4.1 CLIP

8.4.1.1 Call Establishment from H.323 EP to SIP EP

Test: CLIP_01	**Selection Criteria:** Optional	**Selected:** Yes No
Title:	Call establishment from H.323 EP to SIP EP	
Test Purpose:	To verify that the line identity of a calling H.323 EP is presented to the called SIP EP	
Pre-test conditions:	• Configure equipment to be able to establish a call • Configure equipment to use CLIP • Both parties must be registered	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call from H.323 EP to SIP EP**		
2	*Check: Is H.323 EP's identity shown at SIP EP?*	Yes	No
3	**Clear the call at either party**		
Observations:			

8.4.1.2 Call Establishment from SIP EP to H.323 EP

Test: CLIP_02	**Selection Criteria:** Optional	**Selected:** Yes No
Title:	Call establishment from SIP EP to H.323 EP	
Test Purpose:	To verify that the line identity of a calling SIP EP is presented to the called H.323 EP	
Pre-test conditions:	• Configure equipment to be able to establish a call • Configure equipment to use CLIP • Both parties must be registered	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call from SIP EP to H.323 EP**		
2	*Check: Is SIP EP's identity shown at H.323 EP?*	Yes	No
3	**Clear the call at either party**		
Observations:			

8.4.2 CLIR

8.4.2.1 Call Establishment from H.323 EP to SIP EP

Test: CLIR_01	**Selection Criteria:** Optional	**Selected:** Yes No
Title:	Call establishment from H.323 EP to SIP EP	
Test Purpose:	To verify that the transmitted line identity of a calling H.323 EP is not presented to the called SIP EP if CLIR is enabled for the calling party	
Pre-test conditions:	• Configure equipment to be able to establish a call • Configure administrative entities to use CLIR for calling party • Configure CLIP for called party • Both parties must be registered	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call from H.323 EP to SIP EP**		
2	*Check: Is H.323 EP's identity shown at SIP EP?*	No	Yes
3	**Clear the call at either party**		
Observations:			

8.4.2.2 Call Establishment from SIP EP to H.323 EP

Test: CLIR_02	**Selection Criteria:** Optional	**Selected:** Yes No
Title:	Call establishment from SIP EP to H.323 EP	
Test Purpose:	To verify that the transmitted line identity of a calling SIP EP is not presented to the called H.323 EP if CLIR is enabled for the calling party	
Pre-test conditions:	• Configure equipment to be able to establish a call • Configure administrative entities to use CLIR for calling party • Configure CLIP for called party • Both parties must be registered	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call from SIP EP to H.323 EP**		
2	*Check: Is SIP EP's identity shown at H.323 EP?*	No	Yes
3	**Clear the call at either party**		
Observations:			

8.4.3 COLP

8.4.3.1 Call Establishment from H.323 EP to SIP EP with COLP

Test: COLP_01	**Selection Criteria:** Optional	**Selected:** Yes No
Title:	Call establishment from H.323 EP to SIP EP with COLP	
Test Purpose:	To verify that the line identity of a SIP EP called from a H.323 EP is presented to the calling H.323 EP on connection	
Pre-test conditions:	• Configure equipment to be able to establish a call • Configure equipment to use COLP • Both parties must be registered	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call from H.323 EP to SIP EP**		
2	**Accept call at SIP EP**		
3	*Check: Is SIP EP's identity shown at H.323 EP?*	Yes	No
4	**Clear call at either party**		
Observations:			

8.4.3.2 Call Establishment from SIP EP to H.323 EP with COLP

Test: COLP_02	**Selection Criteria:** Optional	**Selected:** Yes No
Title:	Call establishment from SIP EP to H.323 EP with COLP	
Test Purpose:	To verify that the line identity of a H.323 EP called from a SIP EP is presented to the calling SIP EP on connection	
Pre-test conditions:	• Configure equipment to be able to establish a call • Configure equipment to use COLP • Both parties must be registered	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call from SIP EP to H.323 EP**		
2	**Accept call at H.323 EP**		
3	*Check: Is H.323 EP's identity shown at SIP EP?*	Yes	No
4	**Clear call at either party**		
Observations:			

8.4.4 COLR

8.4.4.1 Call Establishment from H.323 EP to SIP EP with COLR

Test: COLR_01	**Selection Criteria:** Optional	**Selected:** Yes No
Title:	Call establishment from H.323 EP to SIP EP with COLR	
Test Purpose:	To verify that the line identity of a SIP EP called from a H.323 EP is not presented to the calling H.323 EP on connection if COLR is enabled for the called party	
Pre-test conditions:	• Configure equipment to be able to establish a call • Configure H.323 EP to use COLP • Configure SIP EP to use COLR • Both parties must be registered	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call from H.323 EP to SIP EP**		
2	**Accept call at SIP EP**		
3	*Check: Is SIP EP's identity shown at H.323 EP?*	No	Yes
4	**Clear call at either party**		
Observations:			

8.4.4.2 Call Establishment from SIP EP to H.323 EP with COLR

Test: COLR_02	**Selection Criteria:** Optional	**Selected:** Yes No
Title:	Call establishment from SIP EP to H.323 EP with COLR	
Test Purpose:	To verify that the line identity of a H.323 EP called from a SIP EP is not presented to the calling SIP EP on connection if COLR is enabled for the called party	
Pre-test conditions:	• Configure equipment to be able to establish a call • Configure SIP EP to use COLP • Configure H.323 EP to use COLR • Both parties must be registered	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call from SIP EP to H.323 EP**		
2	**Accept call at H.323 EP**		
3	*Check: Is H.323 EP's identity shown at SIP EP?*	No	Yes
4	**Clear call at either party**		
Observations:			

8.4.5 DTMF

8.4.5.1 H.323 EP calls SIP EP and calling EP sends In-Band DTMF

Test: DTMF_01	**Selection Criteria:** Optional	**Selected:** Yes No
Title:	H.323 EP calls SIP EP and calling EP sends In-Band DTMF	
Test Purpose:	To verify that the calling party in a H.323 originated and established call is able to send In-Band DTMF to the called party	
Pre-test conditions:	• Configure equipment to be able to establish a call • Configure calling EP to send In-Band DTMF • Configure called EP to receive In-Band DTMF • Both parties must be registered	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call from H.323 EP to SIP EP**		
2	**Accept call at SIP EP**		
3	**Press DTMF key on H.323 EP**		
4	*Check: Is DTMF indicated(visual or audible) at H.323 EP?*	Yes	No
5	*Check: Is DTMF indicated(visual or audible) at SIP EP?*	Yes	No
6	**Clear call at either party**		
Observations:			

8.4.5.2 H.323 EP calls SIP EP and called EP sends In-Band DTMF

Test: DTMF_02	**Selection Criteria:** Optional	**Selected:** Yes No
Title:	H.323 EP calls SIP EP and called EP sends In-Band DTMF	
Test Purpose:	To verify that the called SIP EP in a H.323 originated and established call is able to send In-Band DTMF to the calling party	
Pre-test conditions:	• Configure equipment to be able to establish a call • Configure calling EP to receive In-Band DTMF • Configure called EP to send In-Band DTMF • Both parties must be registered	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call from H.323 EP to SIP EP**		
2	**Accept call at SIP EP**		
3	**Press DTMF key on SIP EP**		
4	*Check: Is DTMF indicated(visual or audible) at H.323 EP?*	Yes	No
5	*Check: Is DTMF indicated(visual or audible) at SIP EP?*	Yes	No
6	**Clear call at either party**		
Observations:			

8.4.5.3 SIP EP calls H.323 EP and calling EP sends In-Band DTMF

Test: DTMF_03	**Selection Criteria:** Optional	**Selected:** Yes No
Title:	SIP EP calls H.323 EP and calling EP sends In-Band DTMF	
Test Purpose:	To verify that the calling party in a SIP originated and established call is able to send In-Band DTMF to the called party	
Pre-test conditions:	• Configure equipment to be able to establish a call • Configure calling party to send In-Band DTMF • Configure called party to receive In-Band DTMF • Both parties must be registered	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call from SIP EP to H.323 EP**		
2	**Accept call at H.323 EP**		
3	**Press DTMF key on SIP EP**		
4	*Check: Is DTMF indicated(visual or audible) at H.323 EP?*	Yes	No
5	*Check: Is DTMF indicated(visual or audible) at SIP EP?*	Yes	No
6	**Clear call at either party**		
Observations:			

8.4.5.4 SIP EP calls H.323 EP and called EP sends In-Band DTMF

Test: DTMF_04	**Selection Criteria:** Optional	**Selected:** Yes No
Title:	SIP EP calls H.323 EP and called EP sends In-Band DTMF	
Test Purpose:	To verify that the called H.323 EP in a SIP originated and established call is able to send In-Band DTMF to the calling party	
Pre-test conditions:	• Configure equipment to be able to establish a call • Configure calling party to receive In-Band DTMF • Configure called party to send In-Band DTMF • Both parties must be registered	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call from SIP EP to H.323 EP**		
2	**Accept call at H.323 EP**		
3	**Press DTMF key on H.323 EP**		
4	*Check: Is DTMF indicated(visual or audible) at H.323 EP?*	Yes	No
5	*Check: Is DTMF indicated(visual or audible) at SIP EP?*	Yes	No
6	**Clear call at either party**		
Observations:			

8.4.5.5 H.323 EP calls SIP EP and calling EP sends Out-Band DTMF

Test: DTMF_05	**Selection Criteria:** Optional	**Selected:** Yes No
Title:	H.323 EP calls SIP EP and calling EP sends Out-Band DTMF	
Test Purpose:	To verify that the calling party in a H.323 originated and established call is able to send Out-Band DTMF to the called party	
Pre-test conditions:	• Configure equipment to be able to establish a call • Configure calling party to send Out-Band DTMF • Configure called party to receive Out-Band DTMF • Both parties must be registered	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call from H.323 EP to SIP EP**		
2	**Accept call at SIP EP**		
3	**Press DTMF key on H.323 EP**		
4	*Check: Is DTMF indicated(visual or audible) at H.323 EP?*	Yes	No
5	*Check: Is DTMF indicated(visual or audible) at SIP EP?*	Yes	No
6	**Clear call at either party**		
Observations:			

8.4.5.6 H.323 EP calls SIP EP and called EP sends Out-Band DTMF

Test: DTMF_06	**Selection Criteria:** Optional	**Selected:** Yes No
Title:	H.323 EP calls SIP EP and called EP sends Out-Band DTMF	
Test Purpose:	To verify that the called SIP EP in a H.323 originated and established call is able to send Out-Band DTMF to the calling party	
Pre-test conditions:	• Configure equipment to be able to establish a call • Configure calling party to receive Out-Band DTMF • Configure called party to send Out-Band DTMF • Both parties must be registered	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call from H.323 EP to SIP EP**		
2	**Accept call at SIP EP**		
3	**Press DTMF key on SIP EP**		
4	*Check: Is DTMF indicated(visual or audible) at H.323 EP?*	Yes	No
5	*Check: Is DTMF indicated(visual or audible) at SIP EP?*	Yes	No
6	Clear call at either party		
Observations:			

8.4.5.7 SIP EP calls H.323 EP and calling EP sends Out-Band DTMF

Test: DTMF_07	**Selection Criteria:** Optional	**Selected:** Yes No
Title:	SIP EP calls H.323 EP and calling EP sends Out-Band DTMF	
Test Purpose:	To verify that the calling party in a SIP originated and established call is able to send Out-Band DTMF (UserInputIndication) to the called party	
Pre-test conditions:	• Configure equipment to be able to establish a call • Configure calling EP to send Out-Band DTMF • Configure called EP to receive Out-Band DTMF • Both parties must be registered	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call from SIP EP to H.323 EP**		
2	**Accept call at H.323 EP**		
3	**Press DTMF key on SIP EP**		
4	*Check: Is DTMF indicated(visual or audible) at H.323 EP?*	Yes	No
5	*Check: Is DTMF indicated(visual or audible) at SIP EP?*	Yes	No
6	**Clear call at either party**		
Observations:			

8.4.5.8 SIP EP calls H.323 EP and called EP sends Out-Band DTMF

Test: DTMF_08	**Selection Criteria:** Optional	**Selected:** Yes No
Title:	SIP EP calls H.323 EP and called EP sends Out-Band DTMF	
Test Purpose:	To verify that the called H.323 EP in a H.323 originated and established call is able to send OutBand DTMF (UserInputIndication) to the calling party	
Pre-test conditions:	• Configure equipment to be able to establish a call • Configure calling EP to receive Out-Band DTMF • Configure called EP to send Out-Band DTMF • Both parties must be registered	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call from SIP EP to H.323 EP**		
2	**Accept call at H.323 EP**		
3	**Press DTMF key on H.323 EP**		
4	*Check: Is DTMF indicated(visual or audible) at H.323 EP?*	Yes	No
5	*Check: Is DTMF indicated(visual or audible) at SIP EP?*	Yes	No
6	**Clear call at either party**		
Observations:			

Annex A (informative): Additional test cases

A.1 Call clearing

The following two test cases are not included in the official test suite, because the functionality they cover is not part of the specifications, the test suite is based on. However, they provide value to manufacturers of soft-phones.

A.1.1 H.323 EP calls SIP EP and SIP EP clears call before answering

Test:	**Selection Criteria:** Optional	**Selected:** Yes No
Title:	H.323 EP calls SIP EP and SIP EP clears call before it answers	
Test Purpose:	To verify that a call attempt from the H.323 EP may be released by the called SIP EP before answering	
Pre-test conditions:	• Configure equipment to be able to establish a call • Both parties must be registered	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call from H.323 EP to SIP EP**		
2	**Clear call at SIP EP**		
3	*Check: Is the call released(call signalling)?*	Yes	No
4	*Check: Are the voice channels established?*	No	Yes
Observations:			

A.1.2 SIP EP calls H.323 EP and H.323 EP clears call before answering

Test:	**Selection Criteria:** Optional	**Selected:** Yes No
Title:	SIP EP calls H.323 EP and H.323 EP clears call	
Test Purpose:	To verify that a call attempt from the SIP EP may be released by the called H.323 EP	
Pre-test conditions:	• Configure equipment to be able to establish a call • Both parties must be registered	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate call from SIP EP to H:323 EP**		
2	**Clear call at H.323 EP**		
3	*Check: Is the call released(call signalling)?*	Yes	No
4	*Check: Are the voice channels established?*	No	Yes
Observations:			

A.2 Miscellaneous

A.2.1 Refresh bindings/Keep alive of SIP EP with H.323 GK

Test:	**Selection Criteria:** optional	**Selected:** Yes No
Title:	Refresh bindings/Keep Alive of SIP EP with H.323 GK	
Test Purpose:	To verify that once an SIP EP has been registered it stays registered with the H.323 GK	
Pre-test conditions:	Configure EUT to set TTL(Time To Live) to a value(e.g. 30 s)	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate registration SIP EP**		
2	*Check B: Is terminal indicating registration (visual(indicator) or audible indication(dial tone))?*	Yes	No
3	**Start countdown timer(30")**		
4	*Check B: Is terminal indicating registration (visual(indicator) or audible indication(dial tone)) after countdown timer expires?*	Yes	No
Observations:			

A.2.2 Voice call from H.323 EP to SIP EP using silence suppression/comfort noise

Test:	**Selection Criteria:** Optional	**Selected:** Yes No
Title:	Voice call establishment from H.323 EP to SIP EP using silence suppression/comfort noise	
Test Purpose:	To verify that a call can be established successfully to SIP EP by H.323 EP and that speech communication is possible between H.323 EP and SIP EP using silence suppressing/comfort noise	
Pre-test conditions:	Configure EUT and QE to support at least codec G.711 Configure EUT and QE to support silence suppression and comfort noise	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate new call at H.323 EP to the address of SIP EP**		
2	*Check A: Is dial Tone heard?*	Yes	No
3	**Make address SIP EP**		
4	*Check A: Is Ringing Tone heard?*	Yes	No
5	*Check B: Is terminal alerting (visual or audible indication)?*	Yes	No
6	**Accept call at SIP EP**		
7	*Check A: Is Ringing Tone heard?*	No	Yes
8	*Check B: Is terminal alerting?*	No	Yes
9	**Apply speech at H.323 EP**		
10	*Check B: Can speech from H.323 EP be heard and understood?*	Yes	No
11	**Apply speech at SIP EP**		
12	*Check A: Can speech from SIP EP be heard and understood?*	Yes	No
13	**Clear the call at H.323 EP and SIP EP**		
Observations:			

A.2.3 G3 Fax call establishment from H.323 EP to SIP EP using T.38

For this test case additional hardware (G3 Fax) is needed. The SIP EP and H.323 EP have to be able to connect a G3 Fax.

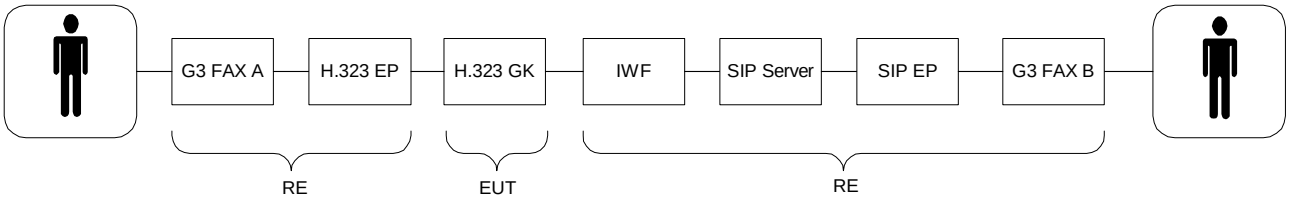

Figure 5: Configuration for FAX call

Test:	**Selection Criteria:** Optional	**Selected:** Yes No
Title:	G3 Fax call establishment from H.323 EP to SIP EP using T.38	
Test Purpose:	To verify that a G3 Fax call can be established successfully to SIP EP by H.323 EP and that Fax communication is possible between H.323 EP and SIP EP.	
Pre-test conditions:	Configure EUT and QE to support at least codec G.711 Configure EUT and QE to support T.38	

Step	**Test description**	**Verdict**	
		Pass	**Fail**
1	**Initiate new G3 Fax call at H.323 EP to the address of SIP EP**		
2	**Make address SIP EP**		
3	*Check B: Is G3 Fax alerting (visual or audible indication)?*	Yes	No
4	**Accept Fax call at SIP EP**		
5	*Check B: Is terminal alerting?*	No	Yes
6	**Apply Fax sheet at G3 Fax A**		
7	*Check A: Is Fax sheet completely send?*	Yes	No
8	*Check B: Is Fax sheetcompletely received?*	Yes	No
9	**Clear the call at H.323 EP and SIP EP**		
Observations:			

History

Document history		
V4.1.1	December 2003	Publication