



**Emergency Communications (EMTEL);
Interoperability testing of core elements for
network independent access to emergency services**

Reference

DTS/EMTEL-00038

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ETSI

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Foreword

This Technical Specification (TS) has been produced by ETSI Special Committee Emergency Communications (EMTEL).

Modal verbs terminology

In the present document "**shall**", "**shall not**", "**should**", "**should not**", "**may**", "**need not**", "**will**", "**will not**", "**can**" and "**cannot**" are to be interpreted as described in clause 3.2 of the [ETSI Drafting Rules](#) (Verbal forms for the expression of provisions).

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Executive summary

The present document specifies the Interoperability Test Descriptions (TD) for core elements involved in network-independent access to emergency services. It provides a standardized methodology to verify end-to-end interoperability between various emergency service network functions, including the Emergency Services IP network (ESInet), Public Safety Answering Points (PSAPs), and originating networks like VoIP and IMS. The test descriptions focus on validating key functionalities such as location delivery (By-Value and By-Reference), Advanced Mobile Location (AML), multimedia session management (audio, video, real-time text), and emergency call routing.

Introduction

Ensuring that emergency communications components from different vendors can interoperate is critical for the reliable delivery of life-saving services. While conformance testing validates individual components, interoperability testing ensures that the entire chain from the caller's device through the core network to the PSAP functions correctly in a multi-vendor environment. The present document defines the necessary test supporting functional elements, configurations and data sets required to perform lab-based interoperability testing. By following these standardized test descriptions, operators and vendors can validate the deployment of Next Generation emergency services independently of the access network.

1 Scope

The present document defines the Interoperability Test Descriptions (TD) for core elements providing network-independent access to emergency services. These test descriptions are intended to:

- Verify end-to-end interoperability between originating services (VoIP, IMS, PSTN/PABX) and Next Generation core services such as the Border Control Function (BCF), Emergency Service Routing Proxy (ESRP), and Emergency Call Routing Function (ECRF).
- Cover procedures for location acquisition and transport, including Location-By-Value and By-Reference, as well as Advanced Mobile Location (AML) via HTTPS and SMS.
- Validate multimedia capabilities, ensuring successful transmission and reception of audio, video, Real-Time Text (RTT), and total conversation across different network configurations.
- Test routing and policy functions, including the use of the LoST protocol for mapping locations to service URNs and handling SIP headers for emergency call management.
- Provide a basis for standardized lab testing by specifying the baseline configurations, reference points, and interfaces (such as SIP, HTTP, HELD, and RTP) required for a complete test environment.

The tests described in the present document are focused on functional interoperability and do not include performance or security testing protocols.

2 References

2.1 Normative references

References are either specific (identified by date of publication and/or edition number or version number) or non-specific. For specific references, only the cited version applies. For non-specific references, the latest version of the referenced document (including any amendments) applies.

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The following referenced documents are necessary for the application of the present document.

- [1] [ETSI TS 103 479](#): "Emergency Communications (EMTEL); Core elements for network independent access to emergency services".
- [2] [ETSI TS 123 167](#): "Universal Mobile Telecommunications System (UMTS); LTE; IP Multimedia Subsystem (IMS) emergency sessions; (3GPP TS 23.167)".
- [3] [ETSI TS 123 333](#): "Digital cellular telecommunications system (Phase 2+) (GSM); Universal Mobile Telecommunications System (UMTS); LTE; Multimedia Resource Function Controller (MRFC) - Multimedia Resource Function Processor (MRFP) Mp interface: Procedures descriptions; (3GPP TS 23.333)".
- [4] [ETSI TS 103 625](#): "Emergency Communications (EMTEL); Transporting Handset Location to PSAPs for Emergency Communications - Advanced Mobile Location".
- [5] [ETSI TS 103 698](#): "Emergency Communications (EMTEL); Lightweight Messaging Protocol for Emergency Service Accessibility (LMPE)".
- [6] [IETF RFC 7946](#): "The GeoJSON Format", (August 2016).
- [7] [ETSI TS 103 683](#): "Mobile Standards Group (MSG); Testing; Next Generation eCall High Level Application Protocol (HLAP) Interoperability Testing".

2.2 Informative references

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The following referenced documents may be useful in implementing an ETSI deliverable or add to the reader's understanding, but are not required for conformance to the present document.

Not applicable.

3 Definition of terms, symbols and abbreviations

3.1 Terms

Void.

3.2 Symbols

Void.

3.3 Abbreviations

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

AML	Advanced Mobile Location
APP	Application
BCF	Border Control Function
CA	Certification Authority
CAP	Common Alerting Protocol
CPE	Call Processing Equipment
CSCF	Call Session Control Function
DID	Decentralised Identifier
DLT	Distributed Ledger Technology
DNS	Domain Name System
DUT	Device under Test
ECRF	Emergency Call Routing Function
ECSCF	Emergency Call Session Control Function
ESInet	Emergency Services IP network
ESRP	Emergency Service Routing Proxy
ETSI	European Telecommunications Standards Institute
EUT	Equipment Under Test
FG	Forest Guide
GNSS	Global Navigation Satellite System
GW	Gateway
HELD	HTTP Enabled Location Delivery
HTTP	Hypertext Transfer Protocol
HTTPS	Hypertext Transfer Protocol Secure
IETF	Internet Engineering Task Force
IBCF	Interconnection Border Control Function
ID	Identity
IFS	Interoperable Function Statement
IF	Interface
IM	Instant Messaging

IMS	IP Multimedia Subsystem
IP	Internet Protocol
IVS	In-Vehicle System
LbR	Location by Reference
LbV	Location by Value
LCS	Location Services (3GPP)
LIF	Location Interwork Function
LIS	Location Information Server
LMPE	Lightweight Messaging Protocol for Emergency Service Accessibility
LNG	Legacy Network Gateway
LRF	Location Retrieval Function
LO	Location Object
LOST	Location to Service Translation
LS	Location Server
MRFP	Multimedia Resource Function Processor
MSD	Minimum Set of Data
NAPTR	Name Authority Pointer
NG	Next Generation
NIF	Next Generation Specific Interwork Function
PABX	Private Automatic Branch Exchange
PCSCF	Proxy Call Session Control Function
PIDF	Presence Information Data Format
PIDF-LO	Presence Information Data Format - Location Object
PIF	Protocol Interworking Function
PSTN	Public Switched Telephone Network
PRF	Policy Routing Function
PSAP	Public Safety Answering Point
RTCP	Real-time Transport Control Protocol
RTP	Real-time Transport Protocol
RTT	Real-Time Text
SCSCF	Serving Call Session Control Function
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SIPS	Session Initiation Protocol Secure
SMS	Short Message Service
SMSC	Short Message Service Centre
SUT	System Under Test
TC	Total Conversation
TCP	Transmission Control Protocol
TD	Test Description
TLS	Transport Layer Security
TR	(ETSI) Technical Report
TS	(ETSI) Technical Specification
UA	User Agent
UC	Unified Communication
UDP	User Datagram Protocol
UE	User Equipment
URI	Uniform Resource Identifier
URL	Uniform Resource Locator
URN	Uniform Resource Name
UTF	Unicode Transformation Format
VoIP	Voice over IP WGS World Geodetic System
XML	eXtensible Markup Language

4 Conventions

4.1 Test Description Pro forma

A Test Description (TD) is a detailed description of the process that needs to be followed to test one or more inter operable functionalities between two or more vendor implementations. A TD should include as a minimum the following elements: The following different types are defined.

Interoperability Test Description			
Identifier	<i>a unique test description identifier should follow a well-defined naming convention, e.g.: TD_XX_00</i>		
Test Objective	<i>a concise summary of the test, which should reflect its purpose and allow readers to easily distinguish this test from any other test in the document</i>		
Configuration	<i>list of all the required equipment for running this test, possibly also including a (reference to) an illustration of a test architecture or test configuration</i>		
References	<i>list of references to the base specification clause(s), use case(s), requirement(s), etc. which are either used in the test or define the functionality being tested</i>		
Applicability	<i>list of features and capabilities in the IFS which are required to be supported by the SUT in order to execute this test (e.g. if this list contains an optional feature to be supported, then the test is optional)</i>		
Pre-test conditions	<i>list of test specific pre-conditions that need to be met by the SUT including information about equipment configuration, i.e. precise description of the initial state of the SUT prior to start executing the test sequence</i>		
Test Sequence	Step	Type	Description
	1	<type>	step description
	2		
	3		
Notes	<i>optional list of explanatory notes</i>		

- A **stimulus** corresponds to an event that triggers an EUT to proceed with a specific protocol action, like sending a message for instance.
- A **check** step consists of verifying that the EUT behaves according to the expected behaviour (for instance the EUT behaviour shows that it receives the expected message).
- A **configure** corresponds to an action to modify the EUT configuration.
- A **verify** step consists of verifying that the tested scenario provides expected results (for instance an emergency call is received at the correct PSAP and media is transmitted).

Each check step consists of the receipt of protocol messages on reference points, with valid content. The check should be performed using a trace created by a monitor tool.

4.2 Interoperable Functions Statement

The "Interoperable Functions Statement" (IFS) identifies the standardised functions of a DUT. These functions can be mandatory, optional or conditional (depending on other functions), and depend on the role played by the DUT. The IFS can also be used as a pro forma by a vendor to identify the functions that its DUT will support when interoperating with corresponding functions from other vendors.

5 Configurations and Interfaces

5.1 ESInet

5.1.1 General

The clauses below describe the baseline ESInet configuration required to execute tests as defined in the present document. The given architecture describes the minimum setup necessary to test a complete end-to-end path including relevant core services. Necessary extensions to consider different hierarchical levels in the test scenarios are out of scope.

5.1.2 Architecture

Core services connecting to an ESInet are introduced in ETSI TS 103 479 [1] and listed as follows:

- Border Control Function (BCF);
- Emergency Call Routing Function (ECRF);
- Call bridging function (BRIDGE);
- Public Safety Answering Point (PSAP);
- Emergency Services Routing Proxy (ESRP); and
- Location Information Service (LIS).

Figure 1 illustrates the base testing environment including various originating services that interconnect to core services via a BCF. Optionally, LIS and ESRP can be used to address missing functionality in the originating network such as location and service URN support. Detailed description of originating services can be found in the following clauses.

NOTE: The use of LIS and ESRP in the origination network does not suggest a specific architecture, rather it helps to support either different test combinations based on location, or to use standard VoIP devices for test purposes.

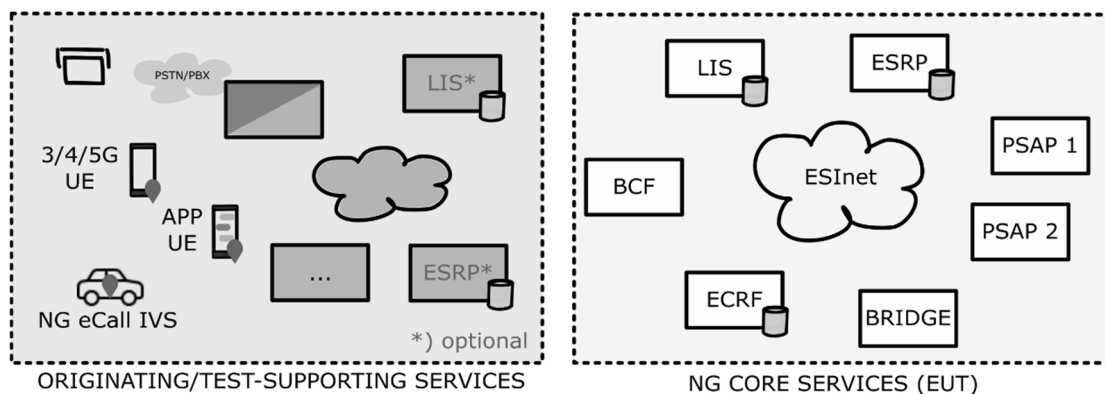


Figure 1: ESInet configuration

5.1.3 Interfaces

Mandatory interface reference points as shown in Figure 2 are introduced in ETSI TS 103 479 [1] and define media and signalling capabilities of core services connected to an ESInet.

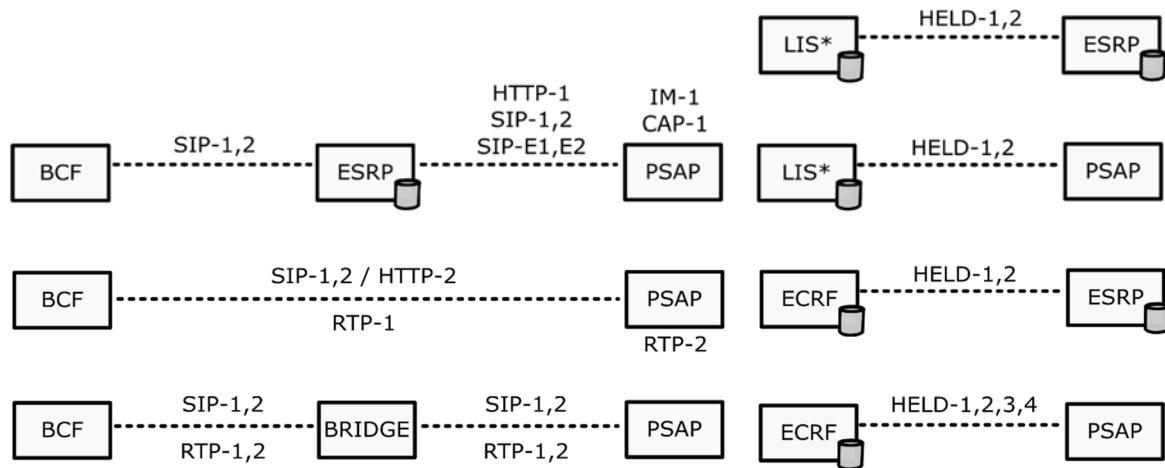


Figure 2: Reference points

Interface identifiers, as listed below, are used in the test description to list required interfaces for the respective test.

SIP-1, SIP-2:

Interface between BCF, ESRP and PSAP elements that defines SIP transport and signalling capabilities.

SIP-E1, SIP-E2:

Interface between ESRP and PSAP elements that defines domain specific SIP event notification capabilities.

HTTP-1:

Interface between ESRP and PSAP elements that defines domain specific web service capabilities.

HTTP-2:

Interface between BCF and PSAP elements that defines domain specific web service capabilities.

LOST-1, LOST-2:

Interface between ESRP, PSAP and ECRF elements that defines LoST signalling capabilities.

LOST-3, LOST-4:

Interface between PSAP and ECRF elements that defines LoST signalling capabilities.

HELD-1, HELD-2:

Interface between ESRP or PSAP and LIS elements that defines location dereference and HELD signalling capabilities.

RTP-1, RTP-2:

Interface between BCF and PSAP elements that defines media transport capabilities and media types for audio, video and real-time text.

IM-1:

PSAP call handling capabilities to support instant messaging.

CAP-1:

PSAP call handling capabilities to support the common alerting protocol.

5.2 VoIP

5.2.1 General

The clauses below describe the baseline VoIP configuration required to execute tests as defined in the present document. The given architecture describes the minimum setup necessary to test a complete end-to-end path including relevant core services with emergency calls originating from a VoIP service provider infrastructure including fixed or mobile devices and call origination (PSTN). Necessary extensions to consider different hierarchical levels in the test scenarios are out of scope.

5.2.2 Architecture

Core services connecting to an ESInet are introduced in ETSI TS 103 479 [1] and listed as follows:

- Border Control Function (BCF);
- Emergency Call Routing Function (ECRF);
- Call bridging function (BRIDGE);
- Public Safety Answering Point (PSAP);
- Emergency Services Routing Proxy (ESRP); and
- Location Information Service (LIS).

Originating services comprising the VoIP configuration are:

- VoIP service provider infrastructure (SIP and GW in Figure 3)
- Call originating devices (UE, APP, IP in Figure 3)

Figure 3 illustrates the VoIP testing environment including originating services that interconnect to core services via a BCF. Optionally, LIS and ESRP can be used to address missing functionality in the originating network such as location or service URN support. ESInet peering is either local (lab) or remote with appropriate security mechanisms. Further information regarding NG eCall tests can be found in ETSI TS 103 683 [7].

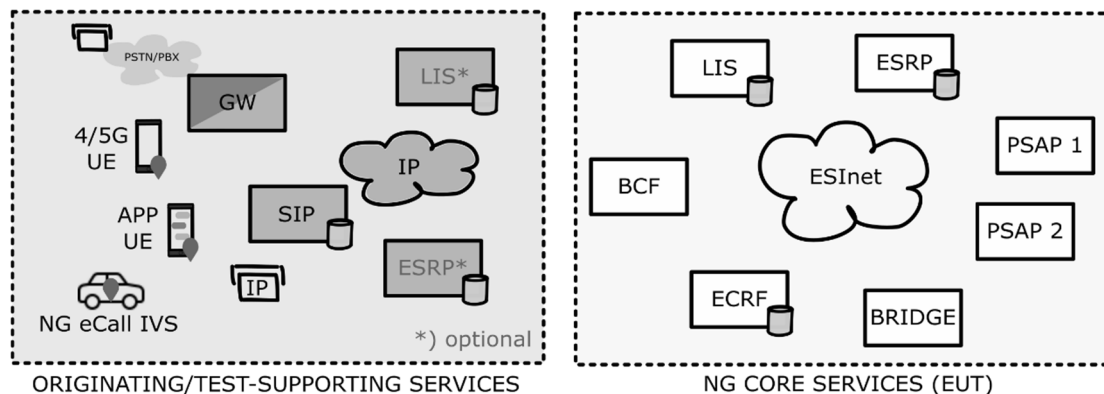


Figure 3: VoIP configuration

5.2.3 Interfaces

Mandatory interface reference points as shown in Figure 4 are introduced in ETSI TS 103 479 [1] and define media and signalling capabilities of the VoIP services connected to an ESInet.

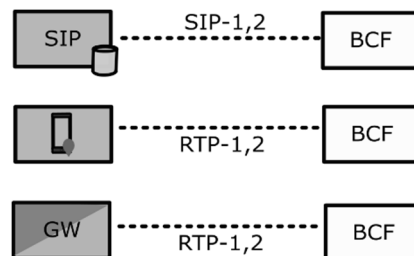


Figure 4: Reference points

Interface identifiers, as listed below, are used in the test description to list required interfaces for the respective test.

SIP-1, SIP-2:

Interface between BCF, ESRP and PSAP elements that defines SIP transport and signalling capabilities.

RTP-1, RTP-2:

Interface between BCF and PSAP elements that defines media transport capabilities and media types for audio, video and real-time text.

Optional interface reference points as shown in Figure 5 are introduced in ETSI TS 103 698 [5] and in ETSI TS 103 479 [1] define chat messaging capabilities of an APP service connected to an ESInet and location refresh via SIP.

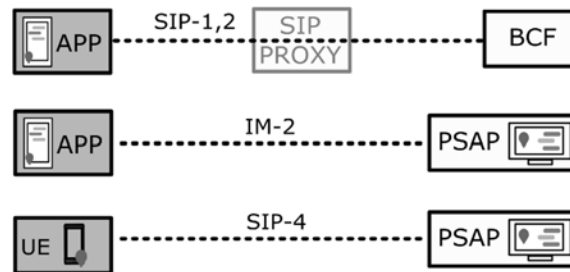


Figure 5: Reference points

Interface identifiers, as listed below, are used in the test description to list required interfaces for the respective test.

SIP-1, SIP-2:

Interface between APP, BCF, ESRP and PSAP elements that defines SIP transport and signalling capabilities.

IM-2:

Interface between APP and PSAP that defines chat handling capabilities to support instant messaging.

SIP-4:

Interface between VoIP UE and PSAP that defines in-dialog location refresh via SIP.

5.3 IMS

5.3.1 General

The clauses below describe the baseline IMS configuration required to execute tests as defined in the present document. The given architecture describes the minimum setup necessary to test a complete end-to-end path including relevant core services with emergency calls originating from an IMS service provider infrastructure including mobile devices attached to a 4/5G mobile network. Necessary extensions to consider different hierarchical levels in the test are out of scope.

5.3.2 Architecture

Core services connecting to an ESInet are introduced in ETSI TS 103 479 [1] and listed as follows:

- Border Control Function (BCF);
- Emergency Call Routing Function (ECRF);
- Call bridging function (BRIDGE);
- Public Safety Answering Point (PSAP);
- Emergency Services Routing Proxy (ESRP); and LRF, SCSCF, PCSCF, ECSCF
- Location Information Service (LIS).

Originating services comprising the IMS configuration are:

- IMS services (in Figure 5)
- IMS interconnection functions (IBCF and MRFP in Figure 6)

Figure 6 illustrates the IMS testing environment including originating services that interconnect to core services via a BCF. ESInet peering is assumed to be or remote with appropriate security mechanisms. Further information regarding NG eCall tests can be found in ETSI TS 103 683 [7].

NOTE: In possible exceptional cases, a local 4/5G test infrastructure can also be connected locally.

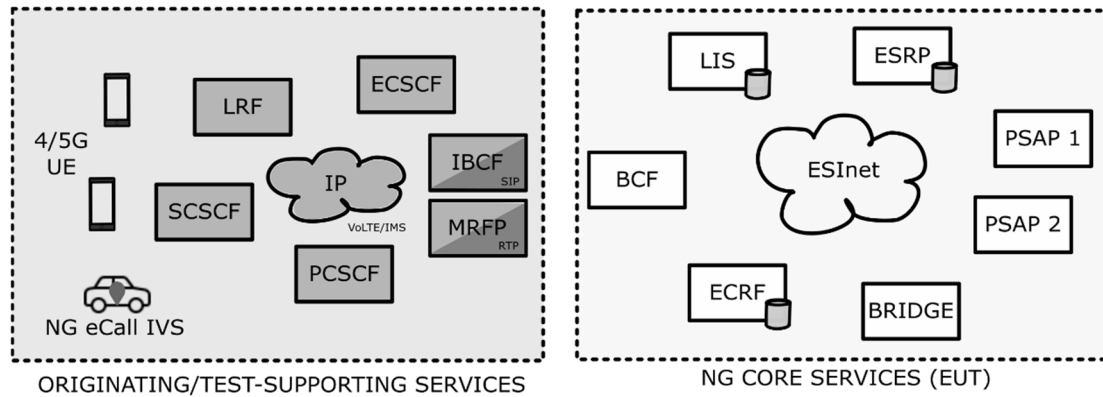


Figure 6: IMS configuration

5.3.3 Interfaces

Mandatory interface reference points as shown in Figure 7 are introduced in ETSI TS 123 167 [2] and ETSI TS 123 333 [3], respectively, and define media and signalling capabilities of the IMS services connected to an ESInet.

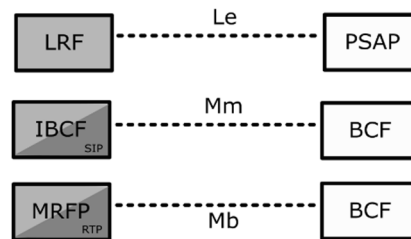


Figure 7: Reference points

Interface identifiers, as listed below, are used in the test description to list required interfaces for the respective test.

Le:

Interface used by the external LCS clients (in this case the PSAP) to retrieve location information.

Mm:

IP interface between CSCF/IBCF and IP networks used, for example, to receive a session request from another SIP server or terminal.

Mb:

Interface to terminate bearer traffic on the MRFP including Real Time Protocol (RTP) packets, along with associated Real Time Control Protocol (RTCP) information.

Optional interface reference points as shown in Figure 8 are introduced in ETSI TS 103 479 [1] and define location refresh via SIP.

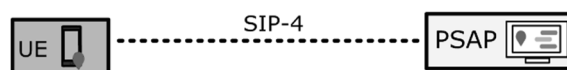


Figure 8: Reference points

Interface identifiers, as listed below, are used in the test description to list required interfaces for the respective test.

SIP-4:

Interface between VoIP UE and PSAP that defines in-dialog location refresh via SIP.

5.4 PSTN/PABX

5.4.1 General

The clauses below describe the baseline PSTN/PABX configuration required to execute tests as defined in the present document. The given architecture describes the minimum setup necessary to test a complete end-to-end path including relevant core services with emergency calls originating from a PSTN/PABX provider infrastructure including fixed or mobile devices and call origination (PSTN). Necessary extensions to consider different hierarchical levels in the test scenarios are out of scope.

5.4.2 Architecture

Core services connecting to an ESInet are introduced in ETSI TS 103 479 [1] and listed as follows:

- Border Control Function (BCF);
- Emergency Call Routing Function (ECRF);
- Call bridging function (BRIDGE);
- Public Safety Answering Point (PSAP); and
- Emergency Services Routing Proxy (ESRP).

Originating services comprising the VoIP configuration are:

- PSTN/PABX provider infrastructure (LNG in Figure 8)
- Call originating devices (UE or phones illustrated in Figure 8)

Figure 9 illustrates the PSTN/PABX testing environment including originating services that interconnect to core services via a BCF. LNG is synonym for any embedded functionality (covering signalling, media and location information) necessary to interwork with core services connected to an ESInet. ESInet peering is either local (lab) or remote with appropriate security mechanisms.

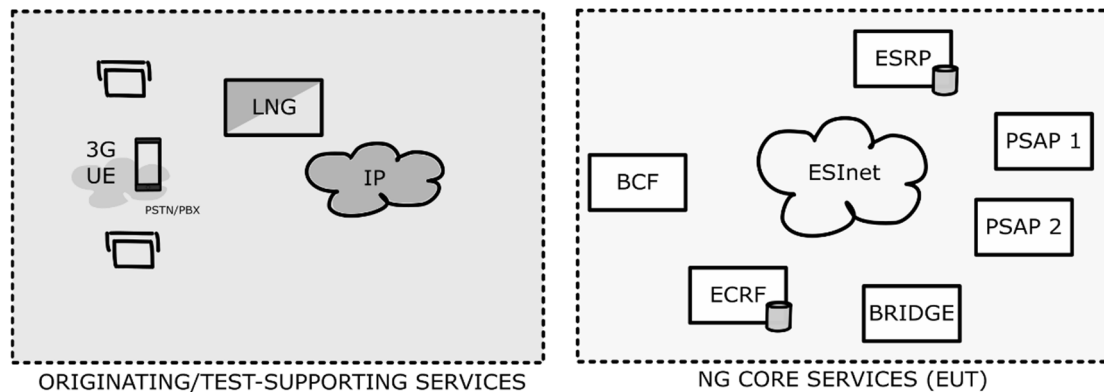


Figure 9: PABX/PSTN configuration

5.4.3 Interfaces

Mandatory interface reference points as shown in Figure 10 are introduced in ETSI TS 103 479 [1] and define media and signalling capabilities of the PSTN/PABX service connected to an ESInet.



Figure 10: Reference points

Interface identifiers, as listed below, are used in the test description to list required interfaces for the respective test.

SIP-1, SIP-2:

Interface between BCF, ESRP and PSAP elements that defines SIP transport and signalling capabilities.

HELD-1, HELD-2:

Interface between ESRP or PSAP and LNG elements that defines location dereference and HELD signalling capabilities.

RTP-1, RTP-2:

Interface between BCF and PSAP elements that defines media transport capabilities and media types for audio, video and real-time text.

5.5 AML

5.5.1 General

The clauses below describe the baseline AML configuration required to execute tests as defined in the present document. The given architecture describes the minimum setup necessary to test a complete end-to-end path including relevant core services with emergency calls originating from a mobile network provider infrastructure that has enabled AML services including mobile devices and call origination (PSTN). Necessary extensions to consider different hierarchical levels in the test scenarios are out of scope.

5.5.2 Architecture

Core services connecting to an ESInet are introduced in ETSI TS 103 479 [1] and listed as follows:

- Border Control Function (BCF);
- Emergency Call Routing Function (ECRF);
- Public Safety Answering Point (PSAP);
- Emergency Services Routing Proxy (ESRP); and
- Location Information Service (LIS).

Originating services comprising the AML configuration are:

- Mobile network provider infrastructure (SMSC and GW in Figure 11)
- Call originating and AML triggering devices (UE, APP in Figure 11)

Figure 11 illustrates the AML testing environment including originating services that interconnect to core services via a BCF. A specific LIS is configured as AML location hub to receive SMS and HTTPS notifications. Optionally, LIS and ESRP can be used to address missing functionality in the originating network such as location (e.g. for routing purposes) or service URN support. ESInet peering is either local (lab) or remote with appropriate security mechanisms.

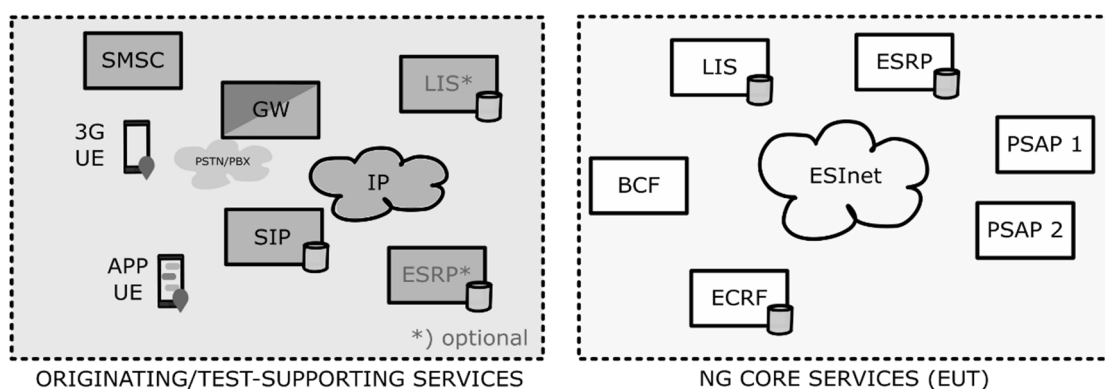


Figure 11: AML configuration

5.5.3 Interfaces

Mandatory interface reference points as shown in Figure 12 are introduced in ETSI TS 103 479 [1] and define media and signalling capabilities required to run AML test scenarios.

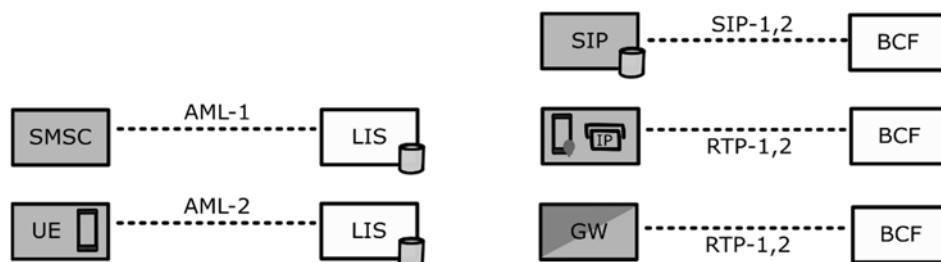


Figure 12: Reference points

Interface identifiers, as listed below, are used in the test description to list required interfaces for the respective test.

SIP-1, SIP-2:

Interface between BCF, ESRP and PSAP elements that defines SIP transport and signalling capabilities.

RTP-1, RTP-2:

Interface between BCF and PSAP elements that defines media transport capabilities and media types for audio, video and real-time text.

AML-1:

Interface between SMSC and LIS that defines AML via SMS.

AML-2:

Interface between UE and LIS that defines AML via HTTPS push.

6 Data Sets

6.1 General

The clauses below define a baseline dataset to support tests as defined in the present document. The data sets allow for a limited number of PSAPs and can easily be supplemented if necessary. The model chosen to determine location information and service regions is essential.

6.2 PSAP service regions

Service regions are defined by polygons and in this case are simple rectangles as shown in Figure 13 below. Service regions in combination with service URNs typically map to a PSAP SIP URI (e.g. a location within service region B + urn:service:sos = sip:psap-2@test.org). A suitable reference point is the location where the tests are carried out (test site). This provides the possibility to test outdoors (e.g. with an IMS or AML test configuration) and to access GNSS-based location information. In principle, the areas are freely selectable as long as the locations are within the given service regions (refer to clause 6.3).

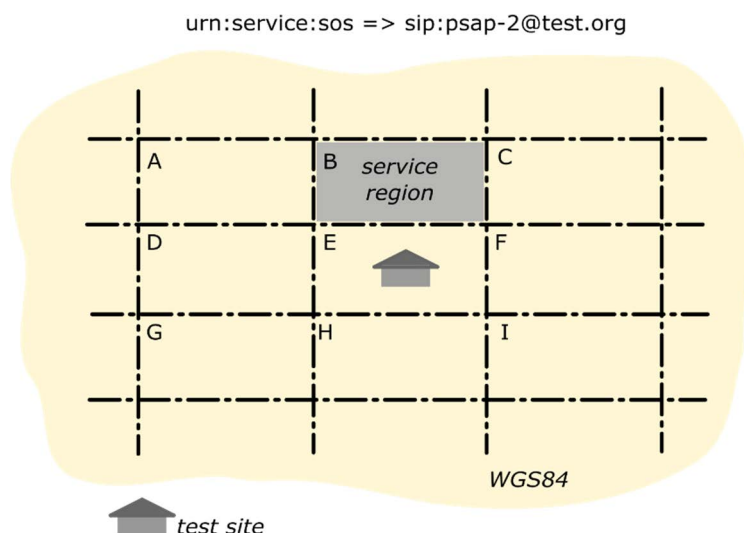


Figure 13: PSAP service regions

The following example describes a service region identified as PSAP01 and encoded as GeoJSON [6]:

```
{
  "type": "FeatureCollection",
  "features": [
    {
      "type": "Feature",
      "properties": {
        "Name": "PSAP01",
        "Description": ""
      },
      "geometry": {
        "type": "Polygon",
        "coordinates": [
          [
            [7.044231891632079, 43.620248285299674],
            [7.0513129234313965, 43.620248285299674],
            [7.0513129234313965, 43.62436460453863],
            [7.044231891632079, 43.62436460453863],
            [7.044231891632079, 43.620248285299674]
          ]
        ]
      }
    }
  ]
}
```

6.3 Test locations

Test locations are defined as point or circle (point and radius) as shown in Figure 14 below. A location associated with an identity (tel or sip URI) can be configured in the LIS to provide an easy way to distinguish between different PSAPs when testing.

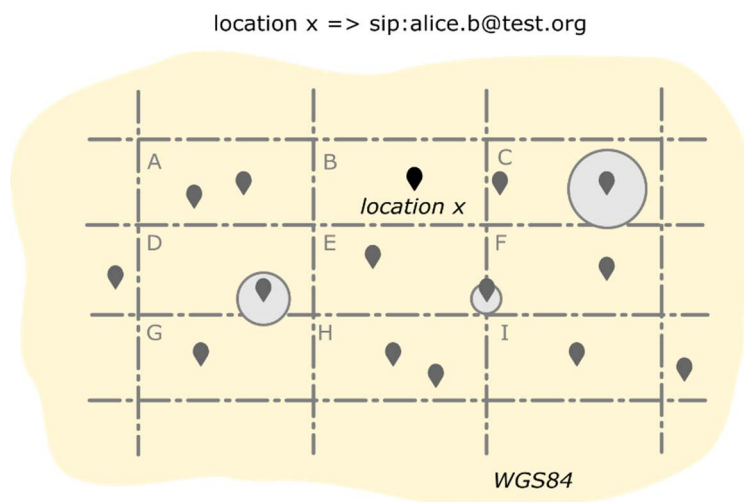


Figure 14: Test locations

The following example describes a location identified as ALICE01 and encoded as GeoJSON [6]:

```
{
  "type": "FeatureCollection",
  "features": [
    {
      "type": "Feature",
      "properties": {
        "Name": "ALICE01",
        "Description": ""
      },
      "geometry": {
        "type": "Point",
        "coordinates": [
          7.046184539794921,
          43.623013240241434
        ]
      }
    }
  ]
}
```

6.4 Domain Name System

The use of DNS is highly recommended in order to simplify the entry of test data and to be able to issue certificates on names instead of IP addresses. Test scenarios described in the present document require at least A records and NAPTR U-records (where the result is an URI). NAPTR service strings are `LoST:http` or `LoST:https` to lookup ECRF/FG services and `LIS:HELD` to lookup LIS services.

The following naming scheme should be used for A records:

`<functional element>.<vendor>.<zone>`

Examples are:

`fg1.vendor-a.plugtests.net`

`psap2.vendor-b.plugtests.net`

`esrp1.vendor-c.plugtests.net`

`ecrf.vendor-a.plugtests.net`

7 Test Descriptions

7.1 Location

7.1.1 Location-By-Value (VoIP)

7.1.1.1 General

Identifier: LOC_LBV_01

This test is intended to verify the standard-compliant transmission of location information from the originating network (VoIP) to the control centre (PSAP) using location-by-value.

This test shall use the configuration as described in clause 5.2 VoIP of the present document.

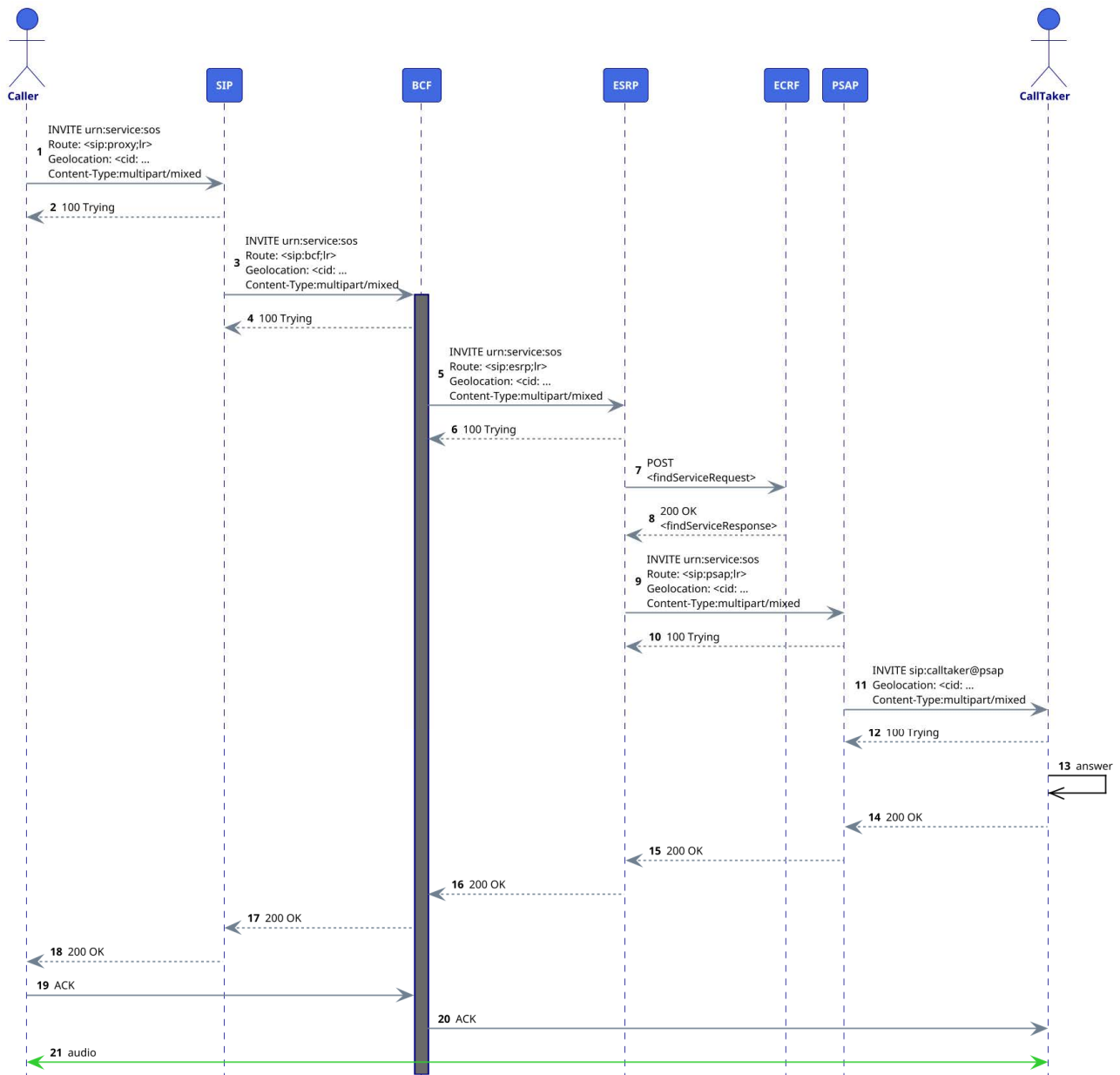
The originating network infrastructure shall support location-by-value, which transmits either geodetic (e.g. point, circle) or civic location information in a PIDF-LO as defined in ETSI TS 103 479 [1].

NOTE: This scenario can be combined with the test description according to clause 7.1.4, Location-By-Reference (VoIP) in a single test run.

7.1.1.2 Description

Interoperability Test Description			
Identifier	LOC_LBV_01		
Test Objective	Verify the standard-compliant transmission of location information from the originating network to the control centre using Location By Value		
Configuration	VoIP		
References	ETSI TS 103 479 [1]		
Applicability	UE (SIP-1, SIP-2, RTP-1, RTP-2) BCF (SIP-1, SIP-2, RTP-1, RTP-2) ESRP (SIP-1, SIP-2, LOST-1, LOST-2) ECRF (LOST-1, LOST-2) PSAP (SIP-1, SIP-2, RTP-1, RTP-2)		
Pre-test conditions	• IP connectivity among all elements of the specific scenario • UE configured to register with SIP Proxy/Registrar • UE has location information (value) available • SIP Proxy trigger points for emergency call routing (to BCF) • BCF trigger points for emergency call routing (to ESRP) • ESRP trigger points for emergency call routing • ESRP configured to query the ECRF • ECRF configured with correct mapping		
Test Sequence	Step	Type	Description
	1	stimulus	User dials emergency number (e.g. 112)
	2	check	Dialog creating INVITE (LbV) received at BCF
	3	check	Dialog creating INVITE (LbV) received at ESRP
	4	check	LoST request received at ECRF
	5	check	ESRP emergency call routing based on location
	6	check	Dialog creating INVITE (LbV) received at PSAP
	7	check	SIP dialog established
	8	verify	PIDF/LO (LbV) received at PSAP
	9	verify	Call connected and location displayed

7.1.1.3 Message Sequence Chart



7.1.2 Location-By-Value (IMS)

7.1.2.1 General

Identifier: LOC_LBV_02

This test is intended to verify the standard-compliant transmission of location information from the originating network (IMS) to the control centre (PSAP) using location-by-value.

This test shall use the configuration as described in clause 5.3 IMS of the present document.

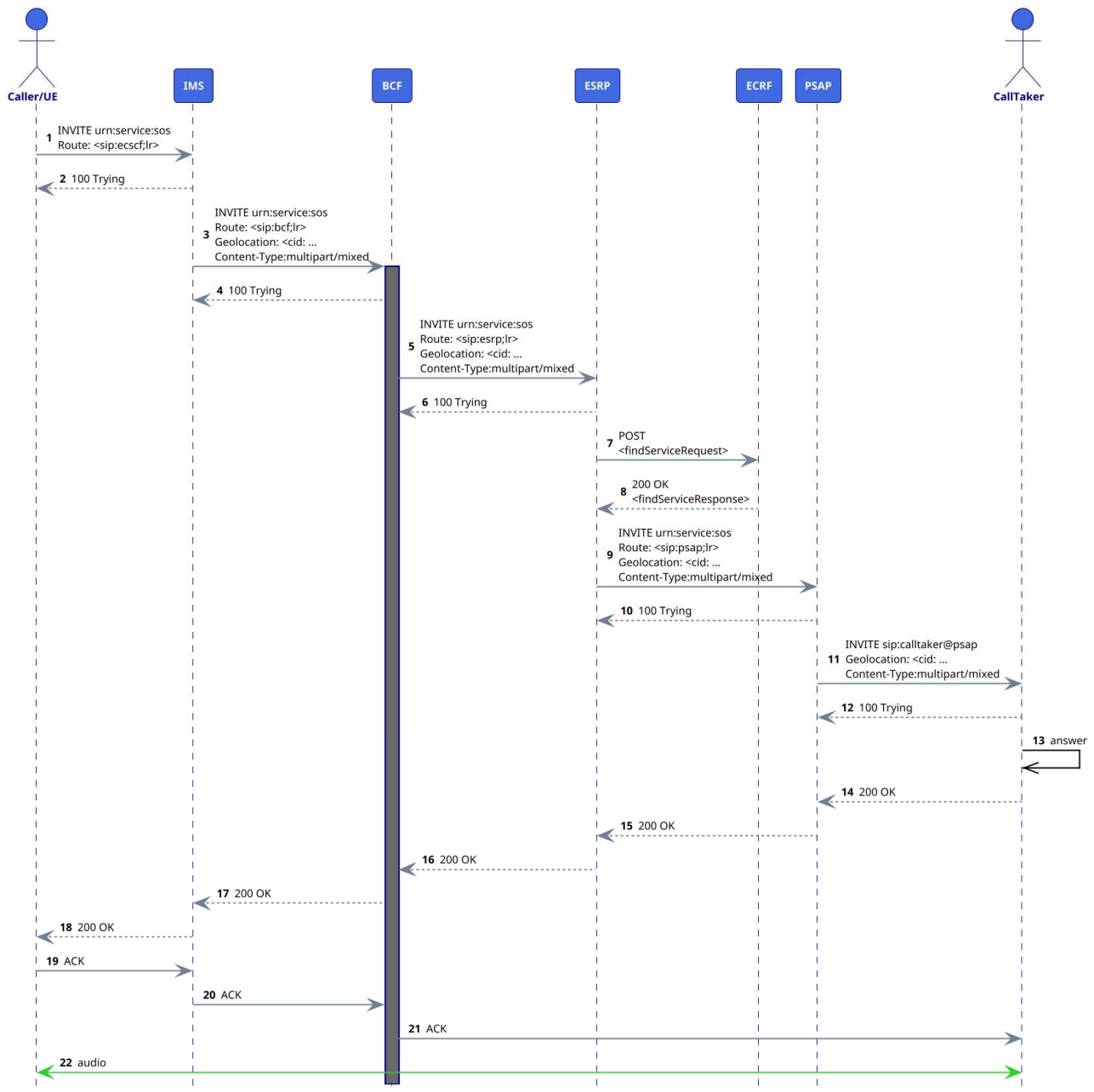
The originating network infrastructure shall support location-by-value, which transmits geodetic (e.g. point, circle, polygon) in a PIDF-LO as defined in ETSI TS 103 479 [1].

NOTE: This scenario can be combined with the test description according to clause 7.1.5 in a single test run.

7.1.2.2 Description

Interoperability Test Description			
Identifier	LOC_LBV_02		
Test Objective	Verify the standard-compliant transmission of location information from the originating network to the control centre using Location By Value		
Configuration	IMS		
References	ETSI TS 103 479 [1], ETSI TS 123 167 [2], ETSI TS 123 333 [3]		
Applicability	IMS (Le, Mm, Mb) BCF (SIP-1, SIP-2, RTP-1, RTP-2) ESRP (SIP-1, SIP-2, LOST-1, LOST-2) ECRF (LOST-1, LOST-2) PSAP (SIP-1, SIP-2, RTP-1, RTP-2)		
Pre-test conditions	• IP connectivity among all elements of the specific scenario • ECSCF trigger points for emergency call routing • LRF has location information (value) available • BCF trigger points for emergency call routing (to ESRP) • ESRP trigger points for emergency call routing • ESRP configured to query the ECRF • ECRF configured with correct mapping		
Test Sequence	Step	Type	Description
	1	stimulus	User dials test emergency number
	2	check	Dialog creating INVITE (LbV) received at BCF
	3	check	Dialog creating INVITE (LbV) received at ESRP
	4	check	LoST request received at ECRF
	5	check	ESRP emergency call routing based on location
	6	check	Dialog creating INVITE (LbV) received at PSAP
	7	check	SIP dialog established
	8	verify	PIDF/LO (LbV) received at PSAP
	9	verify	Call connected and location displayed

7.1.2.3 Message Sequence Chart



7.1.3 Location-By-Value (PSTN/PABX)

7.1.3.1 General

Identifier: LOC_LBV_03

This test is intended to verify the standard-compliant transmission of location information from the originating network (PSTN/PABX) to the control centre (PSAP) using Location-By-Value.

This test shall use the configuration as described in clause 5.4 PSTN/PABX of the present document.

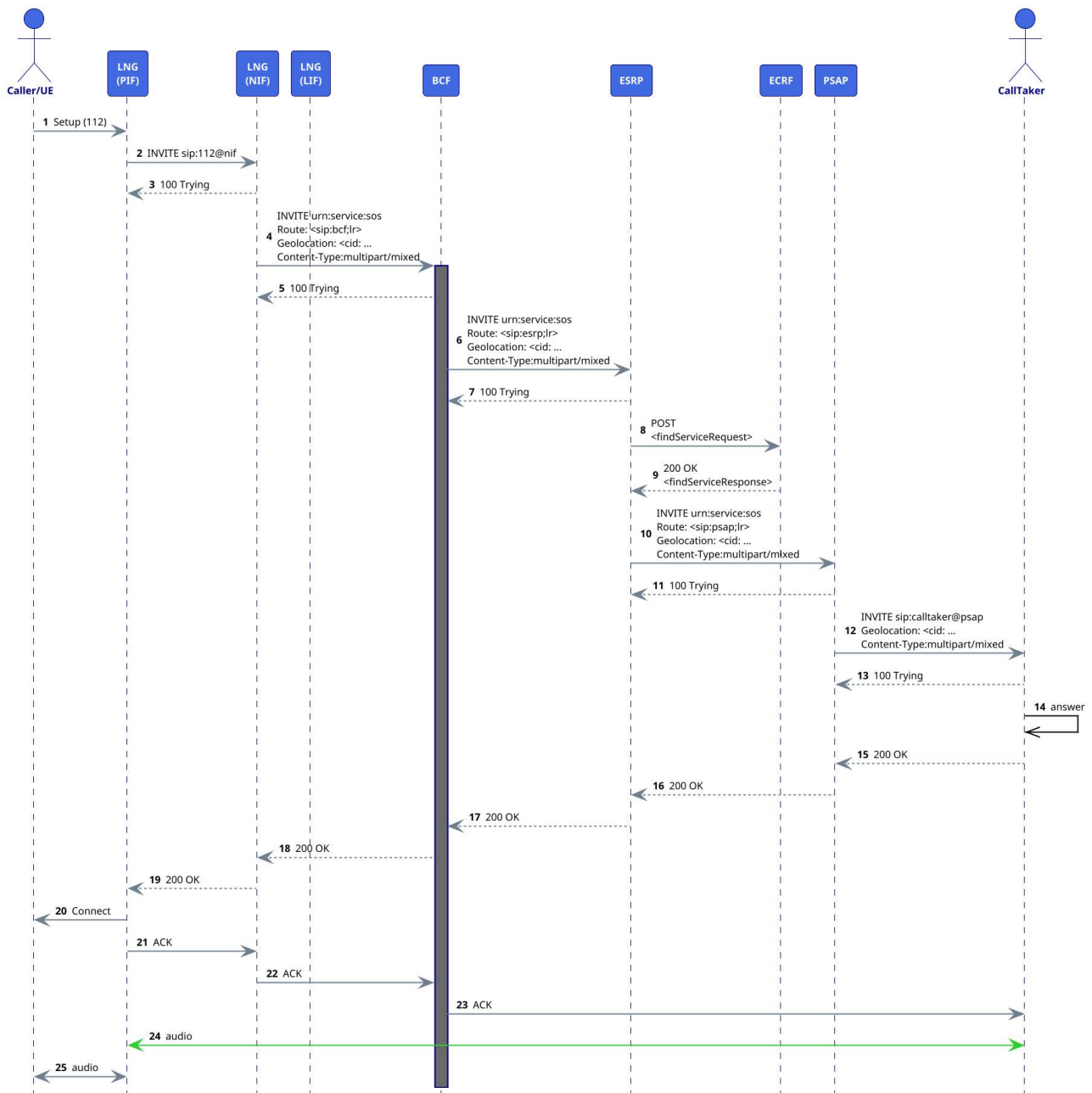
The originating network infrastructure shall support location-by-value, which transmits civic location information in a PIDF-LO as defined in ETSI TS 103 479 [1].

NOTE: This scenario can be combined with the test description according to clause 7.1.6 in a single test run.

7.1.3.2 Description

Interoperability Test Description			
Identifier	LOC_LBV_03		
Test Objective	Verify the standard-compliant transmission of location information from the originating network to the control centre using Location By Value		
Configuration	PSTN/PABX		
References	ETSI TS 103 479 [1]		
Applicability	LNG (SIP-1, SIP-2, RTP-1, RTP-2) BCF (SIP-1, SIP-2, RTP-1, RTP-2) ESRP (SIP-1, SIP-2, LOST-1, LOST-2) ECRF (LOST-1, LOST-2) PSAP (SIP-1, SIP-2, RTP-1, RTP-2)		
Pre-test conditions	• IP connectivity among all elements of the specific scenario • ECSCF trigger points for emergency call routing • LRF has location information (value) available • BCF trigger points for emergency call routing (to ESRP) • ESRP trigger points for emergency call routing • ESRP configured to query the ECRF • ECRF configured with correct mapping		
Test Sequence	Step	Type	Description
	1	stimulus	User dials test emergency number
	2	check	Dialog creating INVITE (LbV) received at BCF
	3	check	Dialog creating INVITE (LbV) received at ESRP
	4	check	LoST request received at ECRF
	5	check	ESRP emergency call routing based on location
	6	check	Dialog creating INVITE (LbV) received at PSAP
	7	check	SIP dialog established
	8	verify	PIDF/LO (LbV) received at PSAP
	9	verify	Call connected and location displayed

7.1.3.3 Message Sequence Chart



7.1.4 Location-By-Reference (VoIP)

7.1.4.1 General

Identifier: LOC_LBR_01

This test is intended to verify the standard-compliant transmission of location information from the originating network (VoIP) to the control centre (PSAP) using location-by-reference.

This test shall use the configuration as described in clause 5.2 VoIP of the present document.

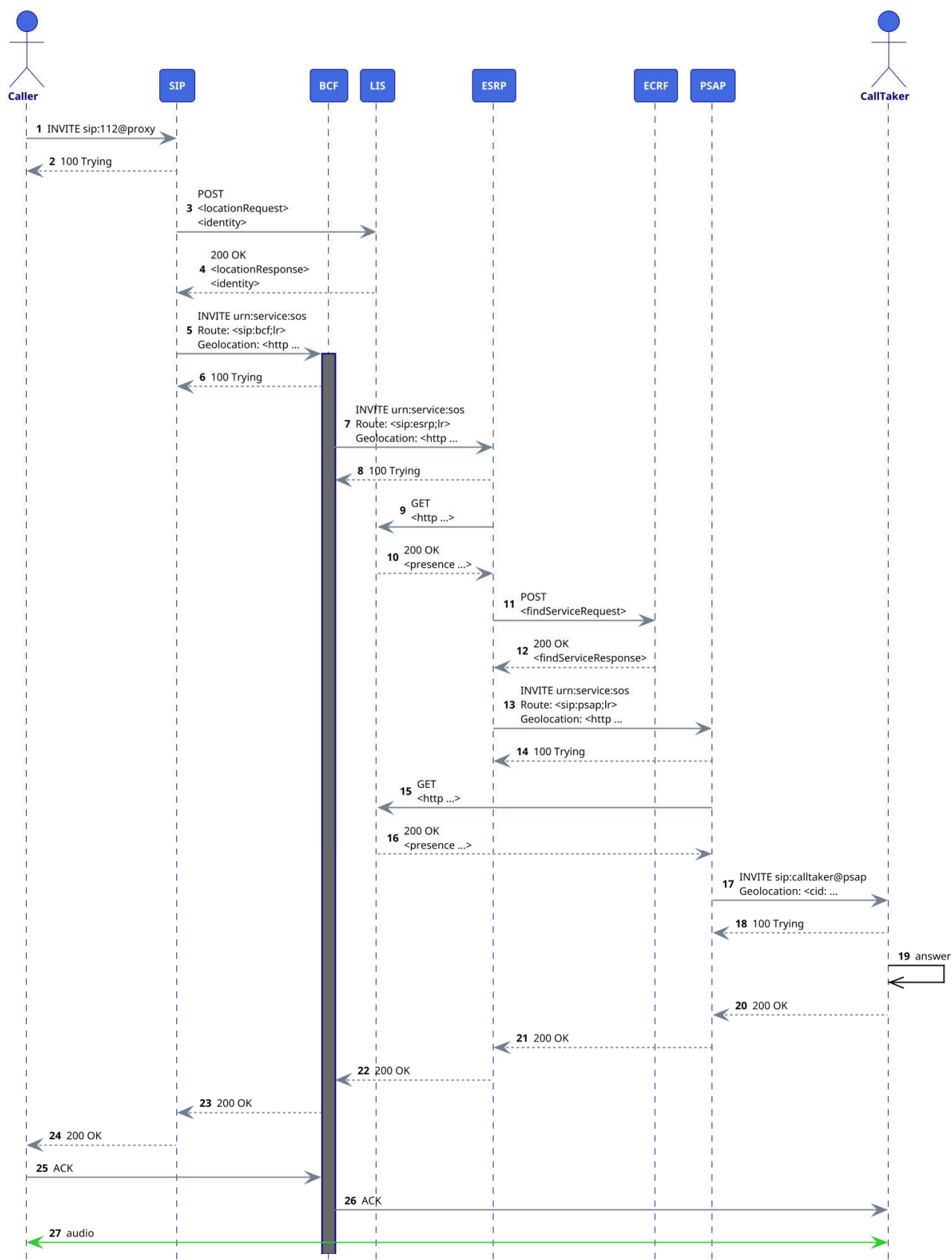
The originating network infrastructure shall support location-by-reference, which transmits a SIP Geolocation header value that can be dereferenced by any authorized entity that receives the reference as defined in ETSI TS 103 479 [1].

NOTE: This scenario can be combined with the test description according to clause 7.1.1 in a single test run.

7.1.4.2 Description

Interoperability Test Description			
Identifier	LOC_LBR_01		
Test Objective	Verify the standard-compliant transmission of location information from the originating network to the control centre using Location By Reference		
Configuration	VoIP		
References	ETSI TS 103 479 [1]		
Applicability	UE (SIP-1, SIP-2, RTP-1, RTP-2, HELD-1) BCF (SIP-1, SIP-2, RTP-1, RTP-2) LIS (HELD-1, HELD-2) ESRP (SIP-1, SIP-2, LOST-1, LOST-2, HELD-2) ECRF (LOST-1, LOST-2) PSAP (SIP-1, SIP-2, RTP-1, RTP-2, HELD-2)		
Pre-test conditions	• IP connectivity among all elements of the specific scenario • UE configured to register with SIP Proxy/Registrar • UE has location information (reference) available • SIP Proxy trigger points for emergency call routing (to BCF) • BCF trigger points for emergency call routing (to ESRP) • ESRP trigger points for emergency call routing • ESRP configured to query LIS and ECRF • ECRF configured with correct mapping		
Test Sequence	Step	Type	Description
	1	stimulus	User dials emergency number (e.g. 112)
	2	check	Dialog creating INVITE (LbR) received at BCF
	3	check	Dialog creating INVITE (LbR) received at ESRP
	4	check	ESRP dereferences location information
	5	check	LoST request received at ECRF
	6	check	ESRP emergency call routing based on location
	7	check	Dialog creating INVITE (LbR) received at PSAP
	8	check	SIP dialog established
	9	verify	PSAP dereferences location information
	10	verify	Call connected and location displayed

7.1.4.3 Message Sequence Chart



7.1.5 Location-By-Reference (IMS)

7.1.5.1 General

Identifier: LOC_LBR_02

This test is intended to verify the standard-compliant transmission of location information from the originating network (IMS) to the control centre (PSAP) using location-by-value.

This test shall use the configuration as described in clause 5.3 IMS of the present document.

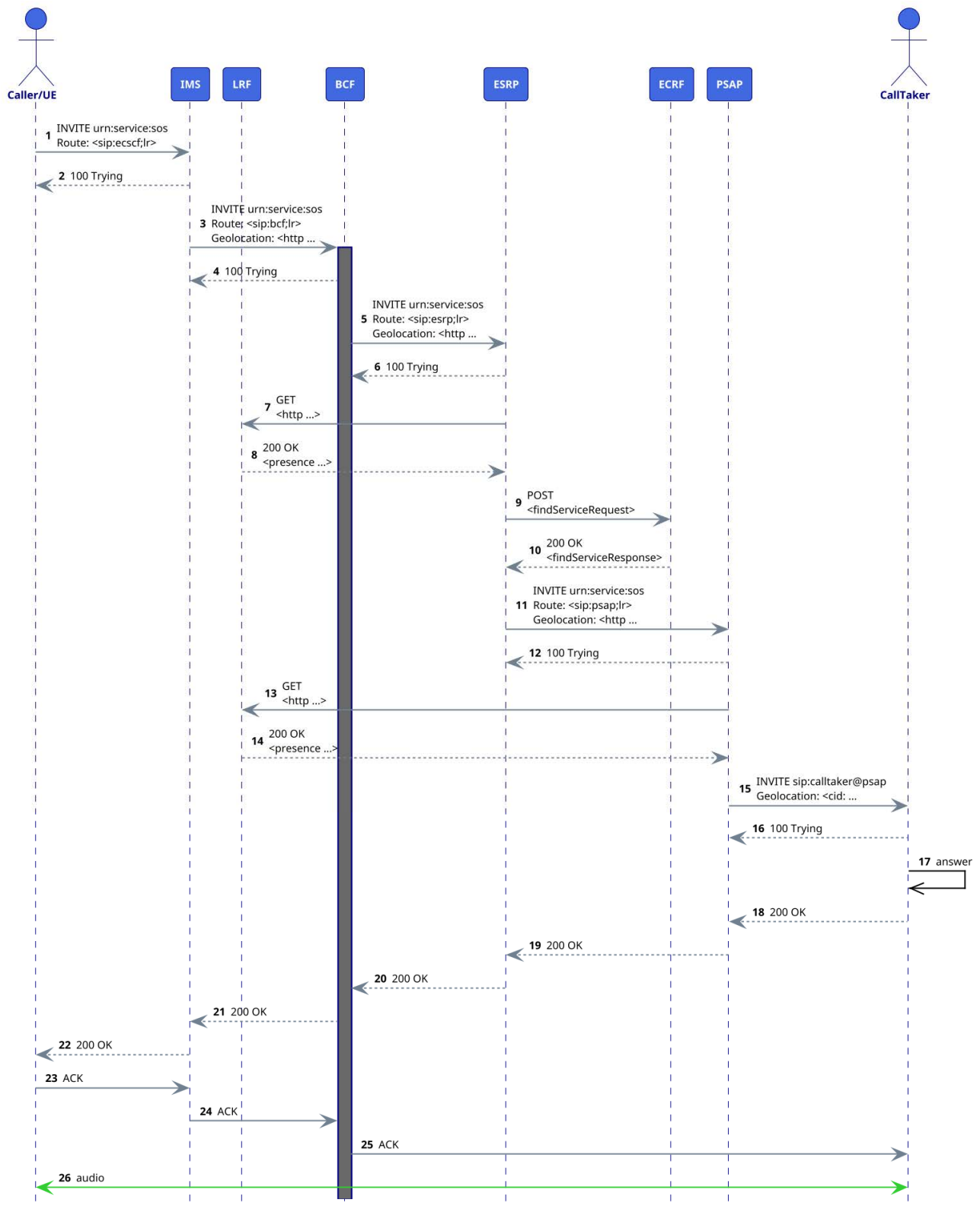
The originating network infrastructure shall support location-by-reference, which transmits a SIP Geolocation header value that can be dereferenced by any authorized entity that receives the reference as defined in ETSI TS 103 479 [1].

NOTE: This scenario can be combined with the test description according to clause 7.1.2 in a single test run.

7.1.5.2 Description

Interoperability Test Description			
Identifier	LOC_LBR_02		
Test Objective	Verify the standard-compliant transmission of location information from the originating network to the control centre using Location By Reference		
Configuration	IMS		
References	ETSI TS 103 479 [1], ETSI TS 123 167 [2], ETSI TS 123 333 [3]		
Applicability	IMS (Le, Mm, Mb) BCF (SIP-1, SIP-2, RTP-1, RTP-2) LIS (HELD-1, HELD-2) ESRP (SIP-1, SIP-2, LOST-1, LOST-2, HELD-2) ECRF (LOST-1, LOST-2) PSAP (SIP-1, SIP-2, RTP-1, RTP-2, HELD-2)		
Pre-test conditions	• IP connectivity among all elements of the specific scenario • ECSCF trigger points for emergency call routing • LRF has location information (reference) available • BCF trigger points for emergency call routing (to ESRP) • ESRP trigger points for emergency call routing • ESRP configured to query LIS and ECRF • ECRF configured with correct mapping		
Test Sequence	Test Sequence		
	Step	Type	Description
	1	stimulus	User dials test emergency number
	2	check	Dialog creating INVITE (LbR) received at BCF
	3	check	Dialog creating INVITE (LbR) received at ESRP
	4	check	ESRP dereferences location information
	5	check	LoST request received at ECRF
	6	check	ESRP emergency call routing based on location
	7	check	Dialog creating INVITE (LbR) received at PSAP
	8	check	SIP dialog established
	9	verify	PSAP dereferences location information
	10	verify	Call connected and location displayed

7.1.5.3 Message Sequence Chart



7.1.6 Location-By-Reference (PSTN/PABX)

7.1.6.1 General

Identifier: LOC_LBR_03

This test is intended to verify the standard-compliant transmission of location information from the originating network (PSTN/PABX) to the control centre (PSAP) using Location-By-Value.

This test shall use the configuration as described in clause 5.4 PSTN/PABX of the present document.

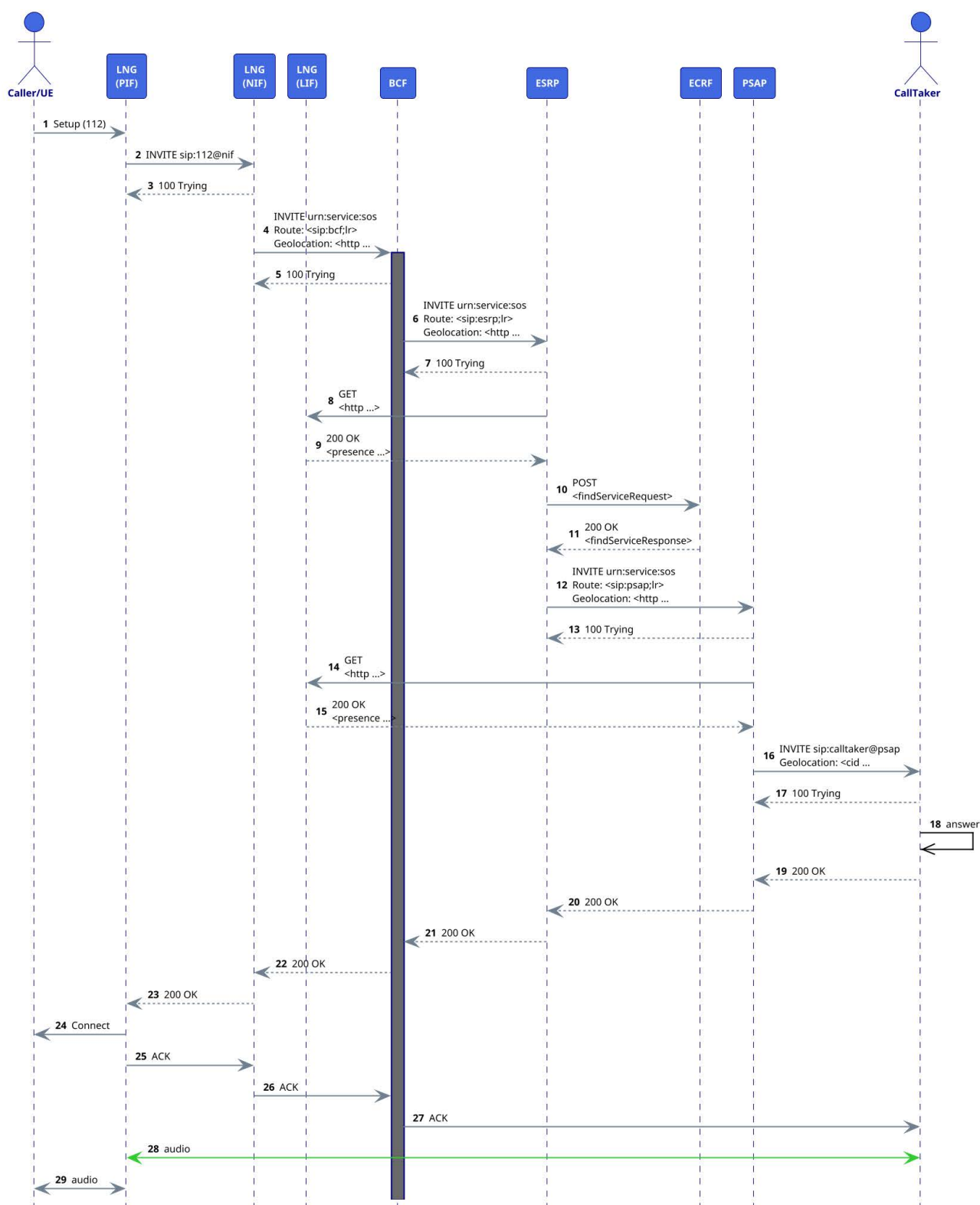
The originating network infrastructure shall support location-by-reference, which transmits a SIP Geolocation header value that can be dereferenced by any authorized entity that receives the reference as defined in ETSI TS 103 479 [1].

NOTE: This scenario can be combined with the test description according to clause 7.1.3 in a single test run.

7.1.6.2 Description

Interoperability Test Description			
Identifier	LOC_LBR_03		
Test Objective	Verify the standard-compliant transmission of location information from the originating network to the control centre using Location By Reference		
Configuration	PSTN/PABX		
References	ETSI TS 103 479 [1]		
Applicability	LNG (SIP-1, SIP-2, RTP-1, RTP-2, HELD-2) BCF (SIP-1, SIP-2, RTP-1, RTP-2) ESRP (SIP-1, SIP-2, LOST-1, LOST-2, HELD-2) ECRF (LOST-1, LOST-2) PSAP (SIP-1, SIP-2, RTP-1, RTP-2, HELD-2)		
Pre-test conditions	• IP connectivity among all elements of the specific scenario • LNG trigger points for emergency call routing • LNG has location information (value) available • BCF trigger points for emergency call routing (to ESRP) • ESRP trigger points for emergency call routing • ESRP configured to query LIS and ECRF • ECRF configured with correct mapping		
Test Sequence	Step	Type	Description
	1	stimulus	User dials test emergency number
	2	check	Dialog creating INVITE (LbR) received at BCF
	3	check	Dialog creating INVITE (LbR) received at ESRP
	4	check	ESRP dereferences location information
	5	check	LoST request received at ECRF
	6	check	ESRP emergency call routing based on location
	7	check	Dialog creating INVITE (LbR) received at PSAP
	8	check	SIP dialog established
	9	verify	PSAP dereferences location information
	10	verify	Call connected and location displayed

7.1.6.3 Message Sequence Chart



7.1.7 Advanced-Mobile-Location (HTTPS)

7.1.7.1 General

Identifier: LOC_AML_01

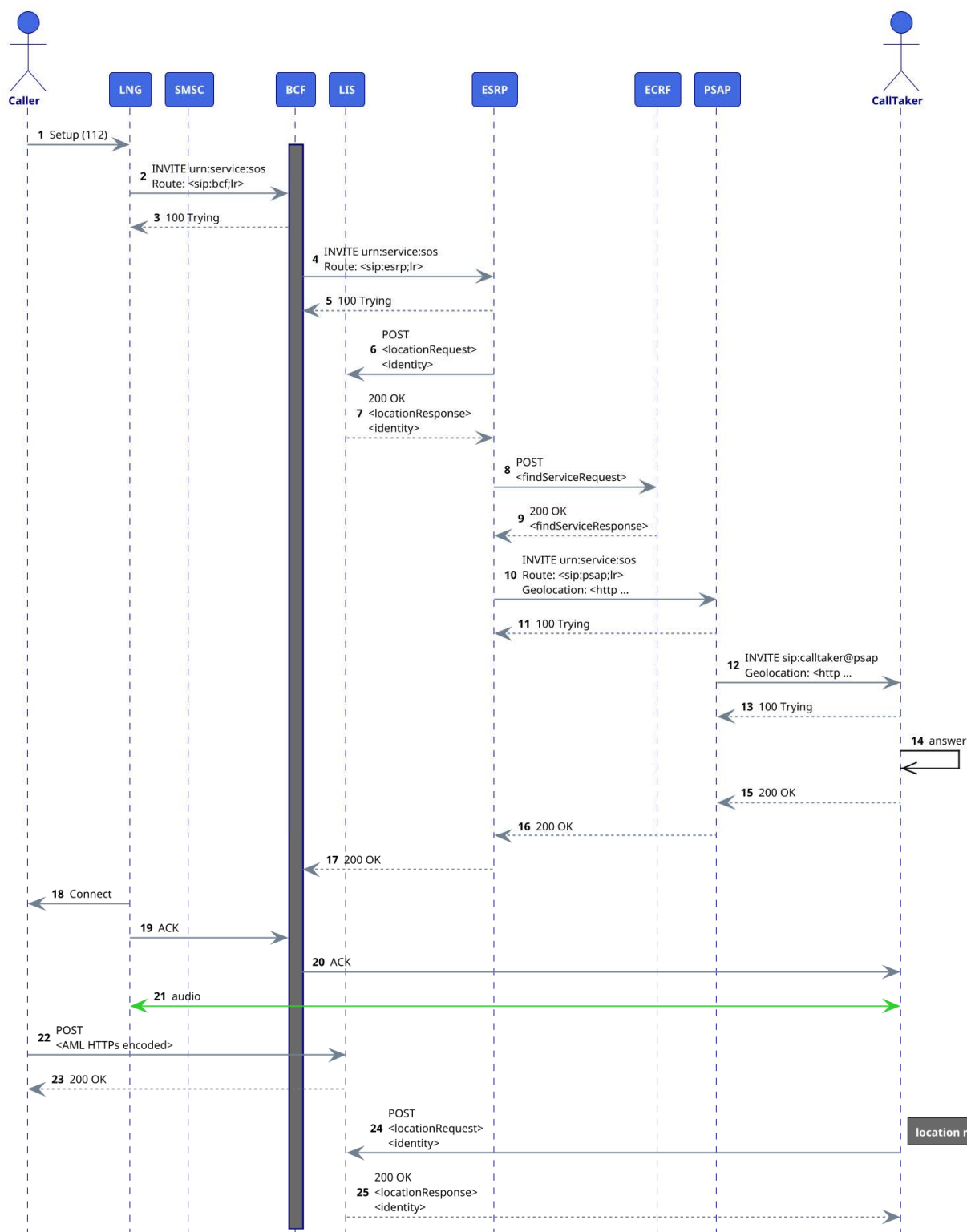
This test is intended to verify the standard-compliant transmission of location information from an emergency call originating mobile network (VoIP) to the location hub (LIS) using AML HTTPS push.

This test shall use the configuration as described in clause 5.5 AML of the present document.

7.1.7.2 Description

Interoperability Test Description			
Identifier	LOC_AML_01		
Test Objective	Verify the standard-compliant transmission of location information from an emergency call originating mobile network (VoIP) to the location hub (LIS) using AML HTTPS push		
Configuration	AML		
References	ETSI TS 103 479 [1], ETSI TS 103 625 [4]		
Applicability	UE (AML-2) BCF (SIP-1, SIP-2, RTP-1, RTP-2) LIS (HELD-1, HELD-2, AML-2) ESRP (SIP-1, SIP-2, LOST-1, LOST-2, HELD-2) ECRF (LOST-1, LOST-2) PSAP (SIP-1, SIP-2, RTP-1, RTP-2, HELD-2)		
Pre-test conditions	• IP connectivity among all elements of the specific scenario • UE configured to trigger AML and send HTTPS push notification to LIS • LIS has static (routable) location information of calling identity • SIP Proxy trigger points for emergency call routing (to BCF) • BCF trigger points for emergency call routing (to ESRP) • ESRP trigger points for emergency call routing • ESRP configured to query LIS and ECRF • ECRF configured with correct mapping		
Test Sequence	Step	Type	Description
	1	stimulus	User dials test emergency number
	2	check	Dialog creating INVITE received at BCF
	3	check	Dialog creating INVITE received at ESRP
	4	check	ESRP queries location information (by identity)
	5	check	LoST request received at ECRF
	6	check	ESRP emergency call routing based on location
	7	check	Dialog creating INVITE (LbR) received at PSAP
	8	check	SIP dialog established
	9	verify	Call connected and location displayed
	10	verify	LIS receives AML HTTPS push message
	11	verify	PSAP repeatedly dereferences location information

7.1.7.3 Message Sequence Chart



7.1.8 Advanced-Mobile-Location (SMS)

7.1.8.1 General

Identifier: LOC_AML_02

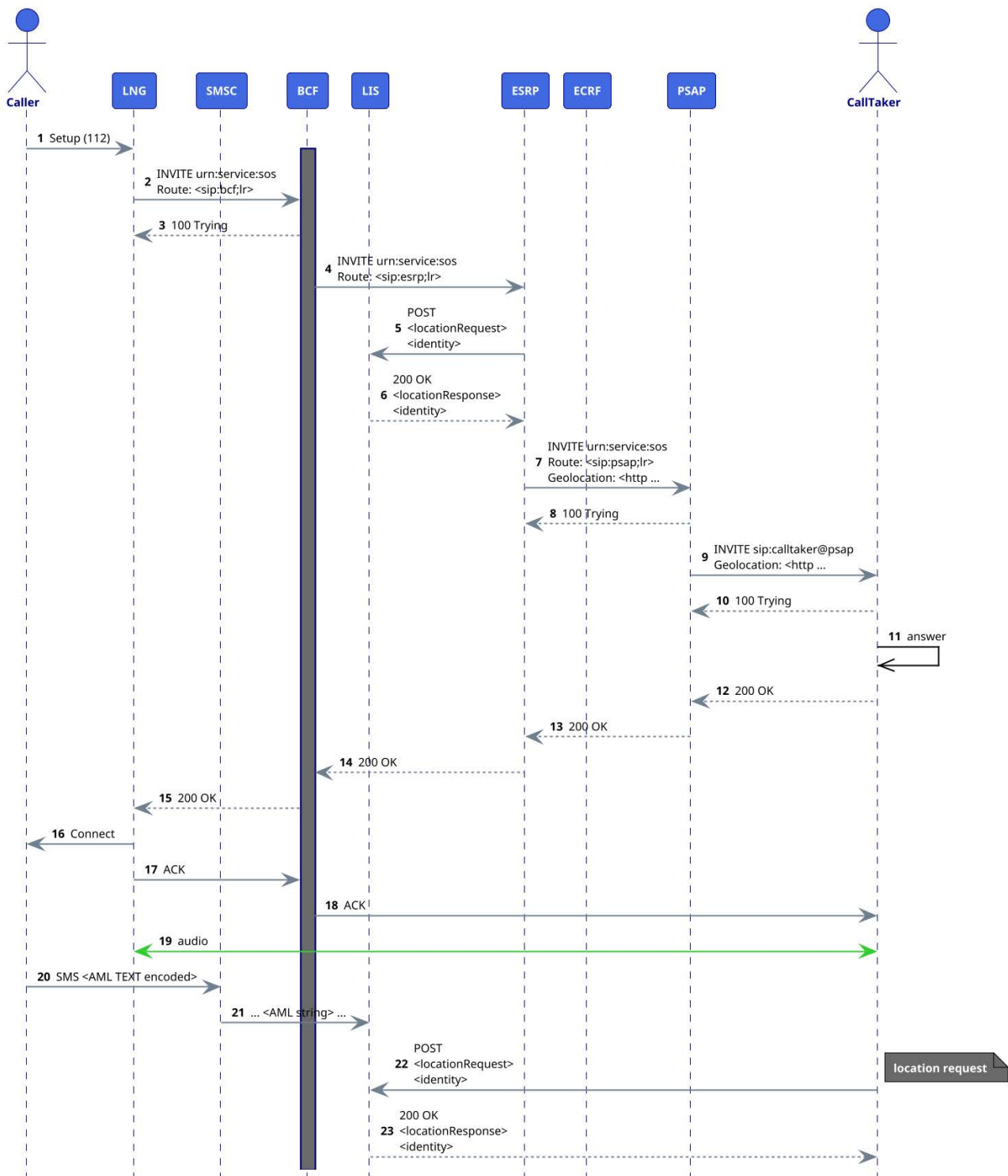
This test is intended to verify the standard-compliant transmission of location information from an emergency call originating mobile network (VoIP) to the location hub (LIS) using AML SMS push.

This test shall use the configuration as described in clause 5.5 AML of the present document.

7.1.8.2 Description

Interoperability Test Description			
Identifier	LOC_AML_02		
Test Objective	Verify the standard-compliant transmission of location information from an emergency call originating mobile network (VoIP) to the location hub (LIS) using AML SMS		
Configuration	AML		
References	ETSI TS 103 479 [1], ETSI TS 103 625 [4]		
Applicability	UE (AML-1 / sms) BCF (SIP-1, SIP-2, RTP-1, RTP-2) LIS (HELD-1, HELD-2, AML-1 / sms) ESRP (SIP-1, SIP-2, LOST-1, LOST-2, HELD-2) ECRF (LOST-1, LOST-2) PSAP (SIP-1, SIP-2, RTP-1, RTP-2, HELD-2)		
Pre-test conditions	• IP connectivity among all elements of the specific scenario • UE configured to trigger AML and send SMS notification to SMSC • LIS has static (routable) location information of calling identity • SIP Proxy trigger points for emergency call routing (to BCF) • BCF trigger points for emergency call routing (to ESRP) • ESRP trigger points for emergency call routing • ESRP configured to query LIS and ECRF • ECRF configured with correct mapping		
Test Sequence	Step	Type	Description
	1	stimulus	User dials test emergency number
	2	check	Dialog creating INVITE received at BCF
	3	check	Dialog creating INVITE received at ESRP
	4	check	ESRP queries location information (by identity)
	5	check	LoST request received at ECRF
	6	check	ESRP emergency call routing based on location
	7	check	Dialog creating INVITE (LbR) received at PSAP
	8	check	SIP dialog established
	9	verify	Call connected and location displayed
	10	verify	LIS receives AML SMS message
	11	verify	PSAP repeatedly dereferences location information

7.1.8.3 Message Sequence Chart



7.1.9 Advanced-Mobile-Location (DATA SMS)

7.1.9.1 General

Identifier: LOC_AML_03

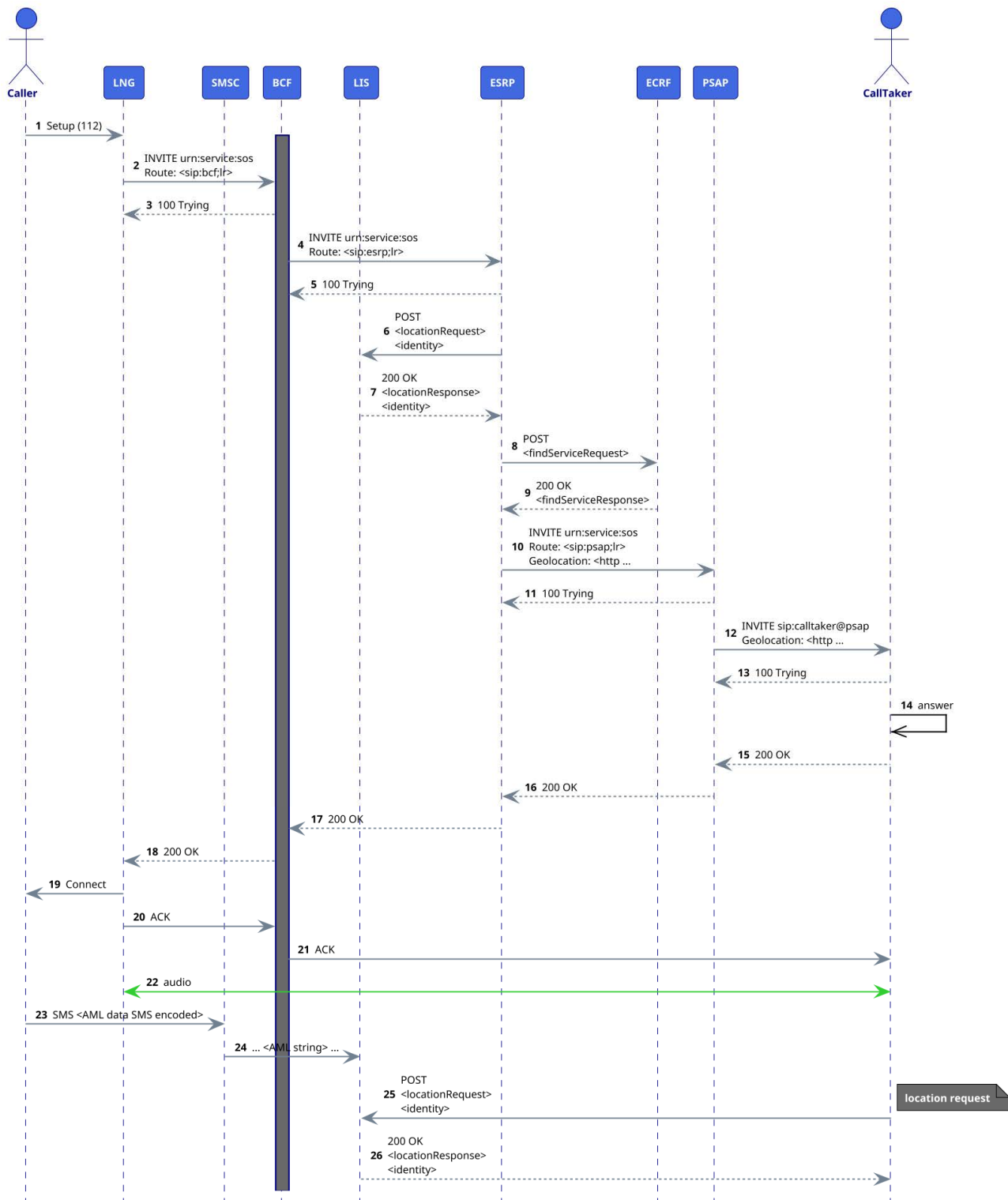
This test is intended to verify the standard-compliant transmission of location information from an emergency call originating mobile network (VoIP) to the location hub (LIS) using AML DATA SMS push.

This test shall use the configuration as described in clause 5.5 AML of the present document.

7.1.9.2 Description

Interoperability Test Description			
Identifier	LOC_AML_03		
Test Objective	Verify the standard-compliant transmission of location information from an emergency call originating mobile network (VoIP) to the location hub (LIS) using AML DATA SMS		
Configuration	AML		
References	ETSI TS 103 479 [1], ETSI TS 103 625 [4]		
Applicability	UE (AML-1 / data sms) BCF (SIP-1, SIP-2, RTP-1, RTP-2) LIS (HELD-1, HELD-2, AML-1 / data sms) ESRP (SIP-1, SIP-2, LOST-1, LOST-2, HELD-2) ECRF (LOST-1, LOST-2) PSAP (SIP-1, SIP-2, RTP-1, RTP-2, HELD-2)		
Pre-test conditions	• IP connectivity among all elements of the specific scenario • UE configured to trigger AML and send DATA SMS notification to SMSC • LIS has static (routable) location information of calling identity • SIP Proxy trigger points for emergency call routing (to BCF) • BCF trigger points for emergency call routing (to ESRP) • ESRP trigger points for emergency call routing • ESRP configured to query LIS and ECRF • ECRF configured with correct mapping		
Test Sequence	Step	Type	Description
	1	stimulus	User dials test emergency number
	2	check	Dialog creating INVITE received at BCF
	3	check	Dialog creating INVITE received at ESRP
	4	check	ESRP queries location information (by identity)
	5	check	LoST request received at ECRF
	6	check	ESRP emergency call routing based on location
	7	check	Dialog creating INVITE (LbR) received at PSAP
	8	check	SIP dialog established
	9	verify	Call connected and location displayed
	10	verify	LIS receives AML DATA SMS message
	11	verify	PSAP repeatedly dereferences location information

7.1.9.3 Message Sequence Chart



7.1.10 Location-Refresh (VoIP)

7.1.10.1 General

Identifier: LOC_RFH_01

This test is intended to verify the standard-compliant location refresh from the originating device (VoIP) to the control centre (PSAP) using SIP methods UPDATE or INVITE.

This test shall use the configuration as described in clause 5.2 VoIP of the present document.

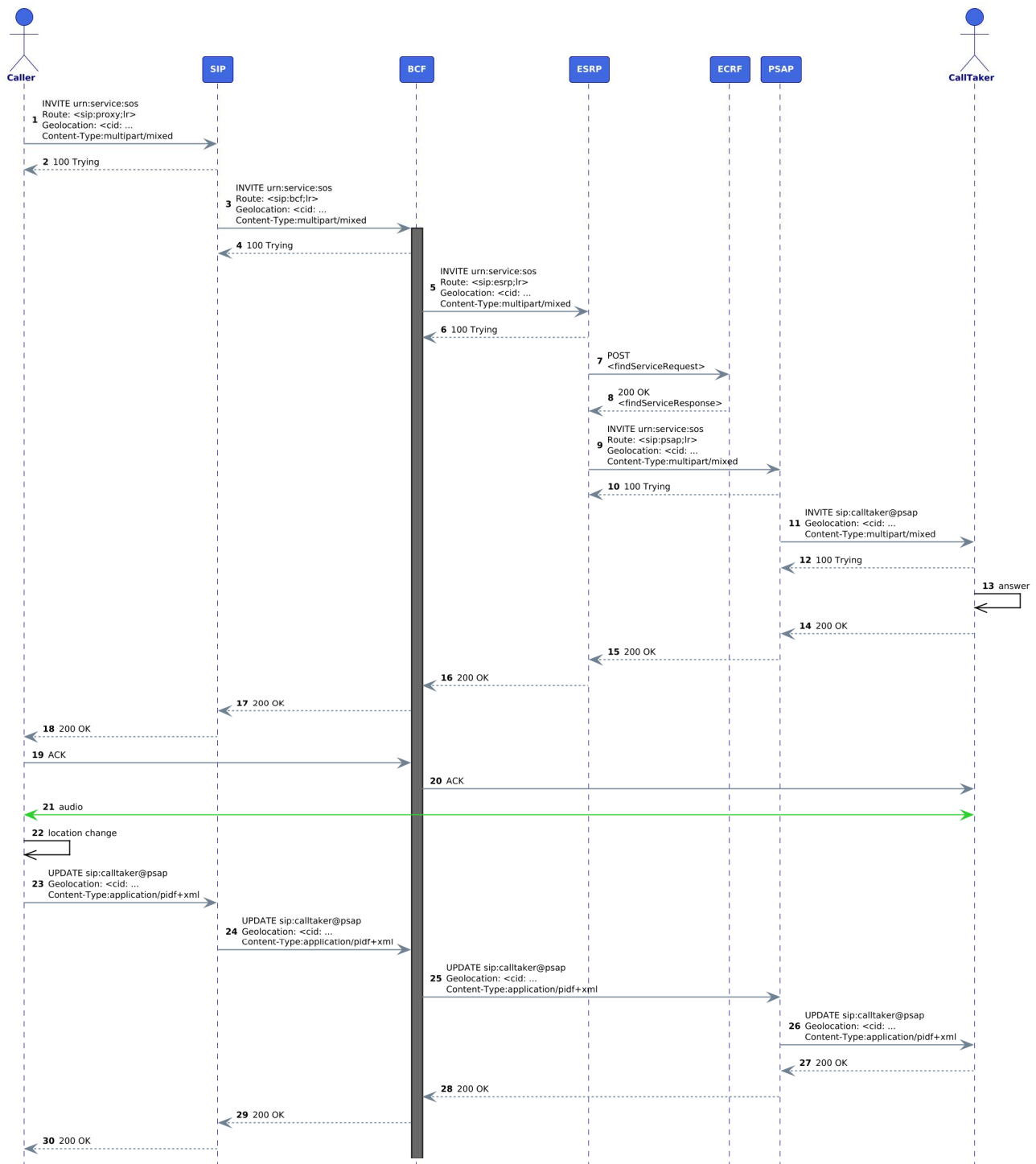
The originating network infrastructure shall support location-by-value, which transmits either geodetic (e.g. point, circle) or civic location information in a PIDF-LO as defined in ETSI TS 103 479 [1].

NOTE: This scenario can be combined with the test description according to clause 7.1.1 in a single test run.

7.1.10.2 Description

Interoperability Test Description			
Identifier	LOC_RFH_01		
Test Objective	Verify the standard-compliant location refresh from the originating device (VoIP) to the control centre (PSAP) using SIP methods UPDATE or INVITE		
Configuration	VoIP		
References	ETSI TS 103 479 [1]		
Applicability	UE (SIP-1, SIP-2, SIP-4, RTP-1, RTP-2) BCF (SIP-1, SIP-2, RTP-1, RTP-2) ESRP (SIP-1, SIP-2, LOST-1, LOST-2) ECRF (LOST-1, LOST-2) PSAP (SIP-1, SIP-2, SIP-4, RTP-1, RTP-2)		
Pre-test conditions	• IP connectivity among all elements of the specific scenario • UE configured to register with SIP Proxy/Registrar • UE has location information (value) available • SIP Proxy trigger points for emergency call routing (to BCF) • BCF trigger points for emergency call routing (to ESRP) • ESRP trigger points for emergency call routing • ESRP configured to query the ECRF • ECRF configured with correct mapping		
Test Sequence	Step	Type	Description
	1	stimulus	SIP dialog established
	2	check	PIDF/LO (LbV) received at PSAP
	3	check	Call connected and location displayed
	4	verify	In_dialog UPDATE or INVITE received
	5	verify	PIDF/LO (LbV) UPDATE received at PSAP
	6	verify	Call still connected and new location displayed

7.1.10.3 Message Sequence Chart



7.1.11 Location-Refresh (IMS)

7.1.11.1 General

Identifier: LOC_RFH_02

This test is intended to verify the standard-compliant location refresh from the originating device (UE) to the control centre (PSAP) using SIP methods UPDATE or INVITE.

This test shall use the configuration as described in clause 5.3 IMS of the present document.

The originating network infrastructure shall support location-by-value, which transmits geodetic (e.g. point, circle, polygon) in a PIDF-LO as defined in ETSI TS 103 479 [1].

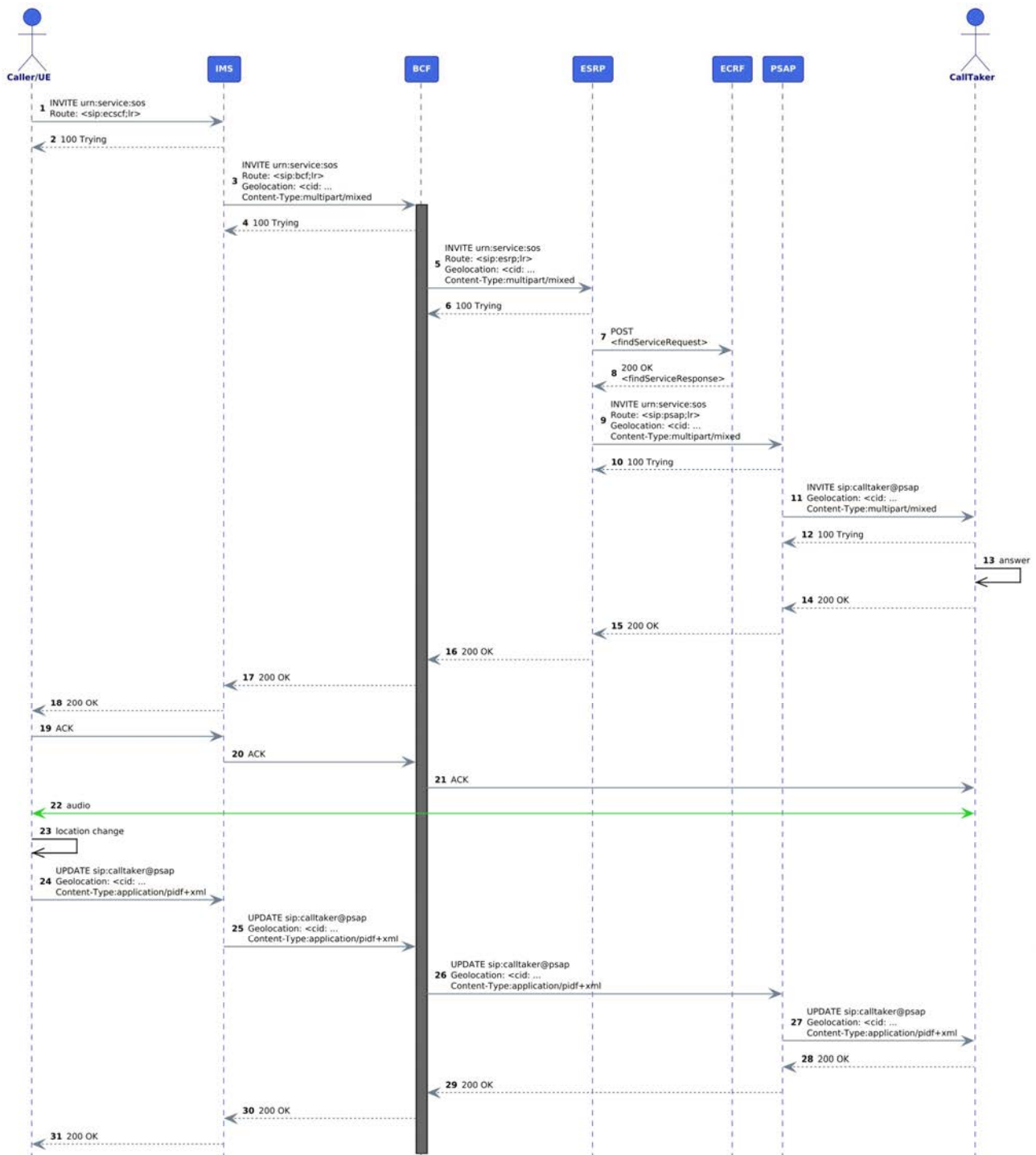
NOTE 1: This scenario can be combined with the test description according to clause 7.1.2 in a single test run.

NOTE 2: For NG eCall tests, it is assumed that the IVS provides a PIDF/LO in a SIP UPDATE or INVITE to refresh the location. Other methods for requesting location updates via MSD are outside the scope of the present document.

7.1.11.2 Description

Interoperability Test Description			
Identifier	LOC_RFH_02		
Test Objective	Verify the standard-compliant location refresh from the originating device (VoIP) to the control centre (PSAP) using SIP methods UPDATE or INVITE		
Configuration	IMS		
References	ETSI TS 103 479 [1], ETSI TS 123 167 [2], ETSI TS 123 333 [3]		
Applicability	UE (SIP-1, SIP-2, SIP-4, RTP-1, RTP-2) IMS (Le, Mm, Mb) BCF (SIP-1, SIP-2, RTP-1, RTP-2) ESRP (SIP-1, SIP-2, LOST-1, LOST-2) ECRF (LOST-1, LOST-2) PSAP (SIP-1, SIP-2, SIP-4, RTP-1, RTP-2)		
Pre-test conditions	• IP connectivity among all elements of the specific scenario • UE configured to register with SIP Proxy/Registrar • UE has location information (value) available • SIP Proxy trigger points for emergency call routing (to BCF) • BCF trigger points for emergency call routing (to ESRP) • ESRP trigger points for emergency call routing • ESRP configured to query the ECRF • ECRF configured with correct mapping		
Test Sequence	Test Sequence		
	Step	Type	Description
	1	stimulus	SIP dialog established
	2	check	PIDF/LO (LbV) received at PSAP
	3	check	Call connected and location displayed
	4	verify	In_dialog UPDATE or INVITE received
	5	verify	PIDF/LO (LbV) UPDATE received at PSAP
	6	verify	Call still connected and new location displayed

7.1.11.3 Message Sequence Chart



7.2 Multimedia

7.2.1 Audio (VoIP)

7.2.1.1 General

Identifier: MME_AUD_01

This test is intended to verify the standard-compliant transmission of audio from the originating network (VoIP) to the control centre (PSAP) transmitted via RTP.

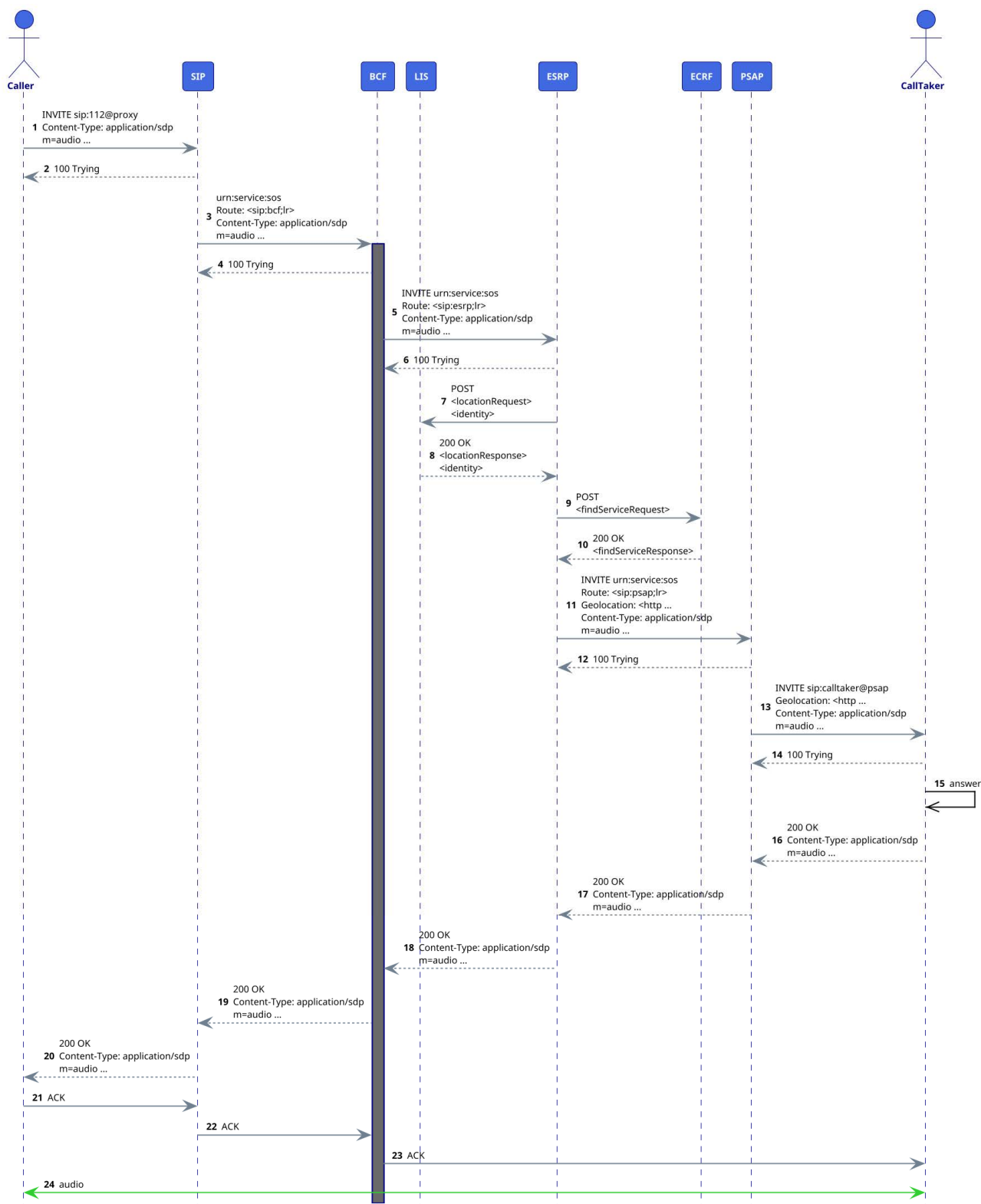
This test shall use the configuration as described in clause 5.2 VoIP of the present document. In the case an UE does not support either LbV or LbR, the location information can also be determined via LIS and a UE identity (tel or sip URI) from the ESRP. Optionally, LIS* and ESRP* can be operated outside the ESInet for this purpose.

The originating network infrastructure shall support proper audio codecs as defined in ETSI TS 103 479 [1].

7.2.1.2 Description

Interoperability Test Description			
Identifier	MME_AUD_01		
Test Objective	Verify the standard-compliant transmission of audio (a) from the originating network (VoIP) to the control centre (PSAP) transmitted via RTP		
Configuration	VoIP		
References	ETSI TS 103 479 [1]		
Applicability	UE (SIP-1, SIP-2, RTP-1, RTP-2) LIS (HELD-1, HELD-2) BCF (SIP-1, SIP-2, RTP-1, RTP-2) ESRP (SIP-1, SIP-2, LOST-1, LOST-2, HELD-2) ECRF (LOST-1, LOST-2) PSAP (SIP-1, SIP-2, RTP-1, RTP-2, HELD-2)		
Pre-test conditions	• IP connectivity among all elements of the specific scenario • UE configured to register with SIP Proxy/Registrar • UE has location information (reference) available • SIP Proxy trigger points for emergency call routing (to BCF) • BCF trigger points for emergency call routing (to ESRP) • ESRP trigger points for emergency call routing • ESRP configured to query LIS and ECRF • ECRF configured with correct mapping		
Test Sequence	Description		
	Step	Type	Description
	1	stimulus	User dials emergency number (e.g. 112)
	2	check	Dialog creating INVITE (a) received at BCF
	3	check	Dialog creating INVITE (a) received at ESRP
	4	check	ESRP queries location information (by identity)
	5	check	LoST request received at ECRF
	6	check	ESRP emergency call routing based on location
	7	check	Dialog creating INVITE (a) received at PSAP
	8	check	SIP dialog established
	9	verify	Call (a) connected and location displayed

7.2.1.3 Message Sequence Chart



7.2.2 Audio (IMS)

7.2.2.1 General

Identifier: MME_AUD_02

This test is intended to verify the standard-compliant transmission of audio from the originating network (IMS) to the control centre (PSAP) transmitted via RTP.

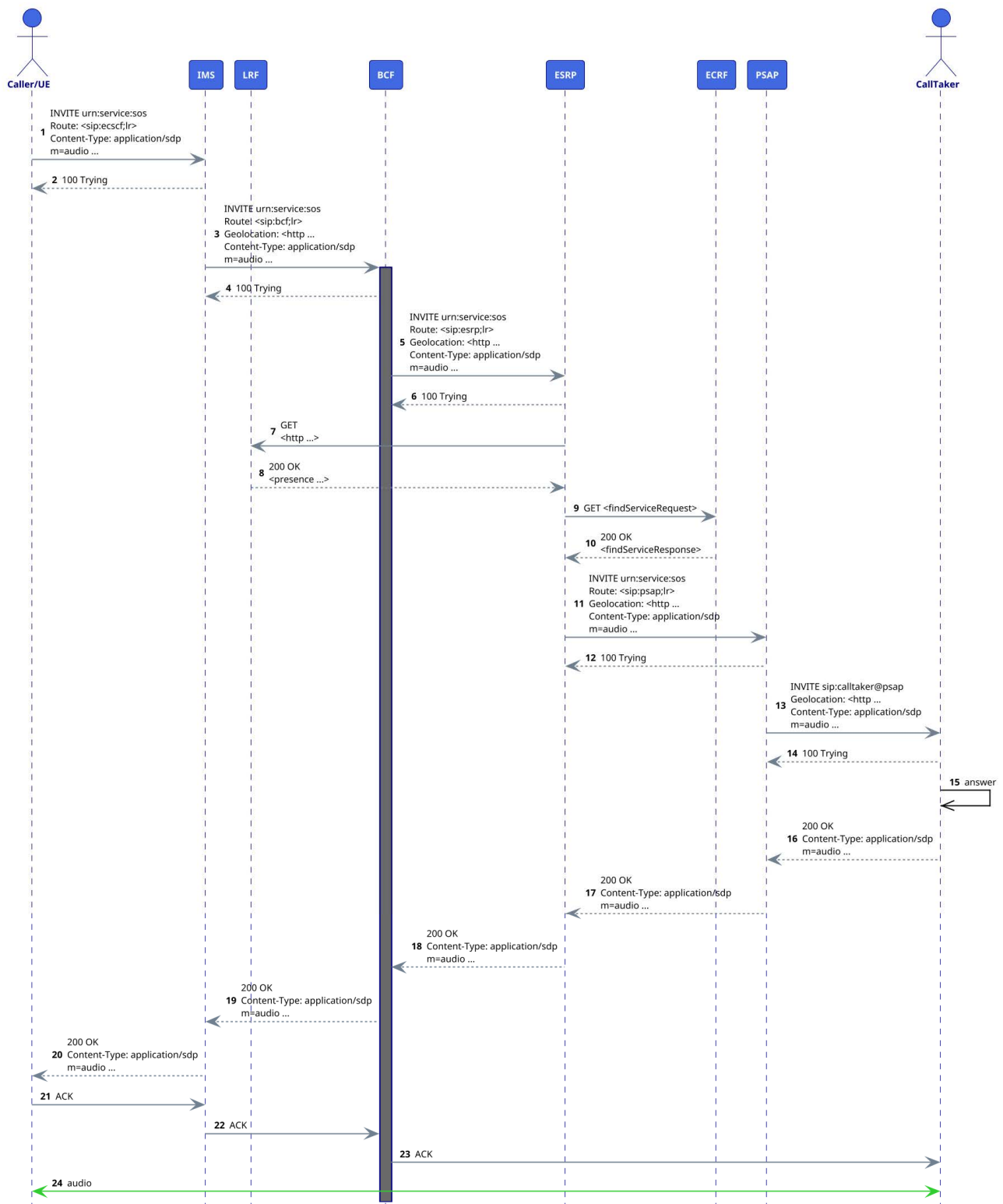
This test shall use the configuration as described in clause 5.3 IMS of the present document.

The originating network infrastructure shall support proper audio codecs as defined in ETSI TS 103 479 [1].

7.2.2.2 Description

Interoperability Test Description			
Identifier	MME_AUD_02		
Test Objective	Verify the standard-compliant transmission of audio (a) from the originating network (IMS) to the control centre (PSAP) transmitted via RTP		
Configuration	IMS		
References	ETSI TS 103 479 [1], ETSI TS 123 167 [2], ETSI TS 123 333 [3]		
Applicability	IMS (Le, Mm, Mb) BCF (SIP-1, SIP-2, RTP-1, RTP-2) ESRP (SIP-1, SIP-2, LOST-1, LOST-2) ECRF (LOST-1, LOST-2) PSAP (SIP-1, SIP-2, RTP-1, RTP-2)		
Pre-test conditions	• IP connectivity among all elements of the specific scenario • ECSCF trigger points for emergency call routing • LRF has location information (value) available • BCF trigger points for emergency call routing (to ESRP) • ESRP trigger points for emergency call routing • ESRP configured to query the ECRF • ECRF configured with correct mapping		
Test Sequence	Step	Type	Description
	1	stimulus	User dials test emergency number
	2	check	Dialog creating INVITE (a) received at BCF
	3	check	Dialog creating INVITE (a) received at ESRP
	4	check	LoST request received at ECRF
	5	check	ESRP emergency call routing based on location
	6	check	Dialog creating INVITE (a) received at PSAP
	7	check	SIP dialog established
	8	verify	Call (a) connected and location displayed

7.2.2.3 Message Sequence Chart



7.2.3 Audio (PSTN/PABX)

7.2.3.1 General

Identifier: MME_AUD_03

This test is intended to verify the standard-compliant transmission of audio from the originating network (PSTN/PABX) to the control centre (PSAP) transmitted via RTP.

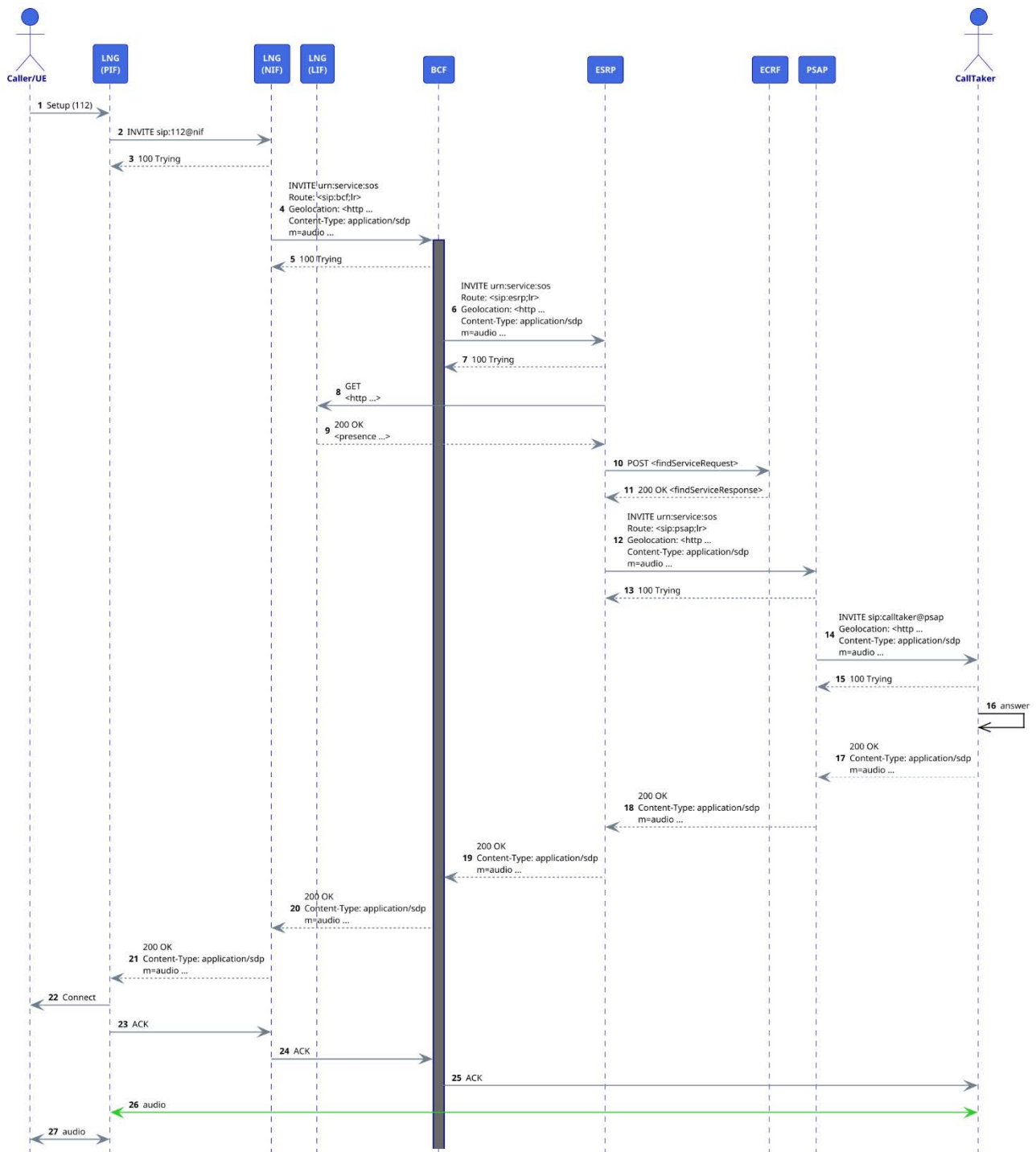
This test shall use the configuration as described in clause 5.4 VoIP of the present document.

The originating network infrastructure shall support proper audio codecs as defined in ETSI TS 103 479 [1].

7.2.3.2 Description

Interoperability Test Description			
Identifier	MME_AUD_03		
Test Objective	Verify the standard-compliant transmission of audio (a) from the originating network (PSTN/PABX) to the control centre (PSAP) transmitted via RTP		
Configuration	PSTN/PABX		
References	ETSI TS 103 479 [1]		
Applicability	LNG (SIP-1, SIP-2, RTP-1, RTP-2, HELD-2) BCF (SIP-1, SIP-2, RTP-1, RTP-2) ESRP (SIP-1, SIP-2, LOST-1, LOST-2, HELD-2) ECRF (LOST-1, LOST-2) PSAP (SIP-1, SIP-2, RTP-1, RTP-2, HELD-2)		
Pre-test conditions	• IP connectivity among all elements of the specific scenario • LNG trigger points for emergency call routing • LNG has location information (value) available • BCF trigger points for emergency call routing (to ESRP) • ESRP trigger points for emergency call routing • ESRP configured to query LIS and ECRF • ECRF configured with correct mapping		
Test Sequence	Step	Type	Description
	1	stimulus	User dials test emergency number
	2	check	Dialog creating INVITE (a) received at BCF
	3	check	Dialog creating INVITE (a) received at ESRP
	4	check	ESRP dereferences location information
	5	check	LoST request received at ECRF
	6	check	ESRP emergency call routing based on location
	7	check	Dialog creating INVITE (a) received at PSAP
	8	check	SIP dialog established
	9	verify	Call connected and location displayed

7.2.3.3 Message Sequence Chart



7.2.4 Video (VoIP)

7.2.4.1 General

Identifier: MME_VID_01

This test is intended to verify the standard-compliant transmission of audio and video from the originating network (VoIP) to the control centre (PSAP) transmitted via RTP.

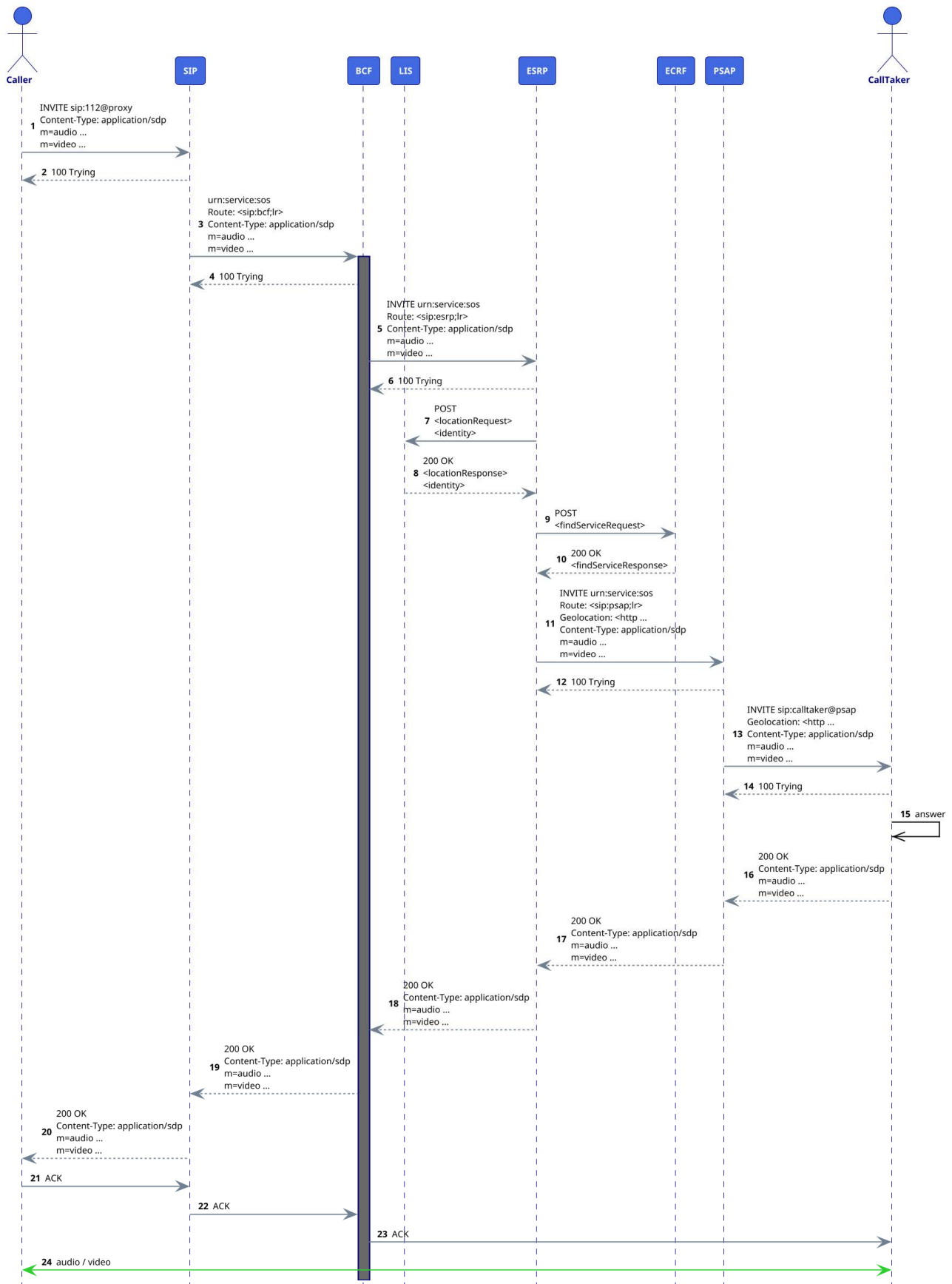
This test shall use the configuration as described in clause 5.2 VoIP of the present document. In the case an UE does not support either LbV or LbR, the location information can also be determined via LIS and a UE identity (tel or sip URI) from the ESRP. Optionally, LIS* and ESRP* can be operated outside the ESInet for this purpose.

The originating network infrastructure shall support proper audio and video codecs as defined in ETSI TS 103 479 [1].

7.2.4.2 Description

Interoperability Test Description			
Identifier	MME_VID_01		
Test Objective	Verify the standard-compliant transmission of audio (a) and video (v) from the originating network (VoIP) to the control centre (PSAP) transmitted via RTP		
Configuration	VoIP		
References	ETSI TS 103 479 [1]		
Applicability	UE (SIP-1, SIP-2, RTP-1, RTP-2) LIS (HELD-1, HELD-2) BCF (SIP-1, SIP-2, RTP-1, RTP-2) ESRP (SIP-1, SIP-2, LOST-1, LOST-2, HELD-2) ECRF (LOST-1, LOST-2) PSAP (SIP-1, SIP-2, RTP-1, RTP-2, HELD-2)		
Pre-test conditions	• IP connectivity among all elements of the specific scenario • UE configured to register with SIP Proxy/Registrar • UE has location information (reference) available • SIP Proxy trigger points for emergency call routing (to BCF) • BCF trigger points for emergency call routing (to ESRP) • ESRP trigger points for emergency call routing • ESRP configured to query LIS and ECRF • ECRF configured with correct mapping		
Test Sequence	Step	Type	Description
	1	stimulus	User dials emergency number (e.g. 112)
	2	check	Dialog creating INVITE (a/v) received at BCF
	3	check	Dialog creating INVITE (a/v) received at ESRP
	4	check	ESRP queries location information (by identity)
	5	check	LoST request received at ECRF
	6	check	ESRP emergency call routing based on location
	7	check	Dialog creating INVITE (a/v) received at PSAP
	8	check	SIP dialog established
	9	verify	Call (a/v) connected and location displayed

7.2.4.3 Message Sequence Chart



7.2.5 Video (IMS)

7.2.5.1 General

Identifier: MME_VID_02

This test is intended to verify the standard-compliant transmission of audio and video from the originating network (IMS) to the control centre (PSAP) transmitted via RTP.

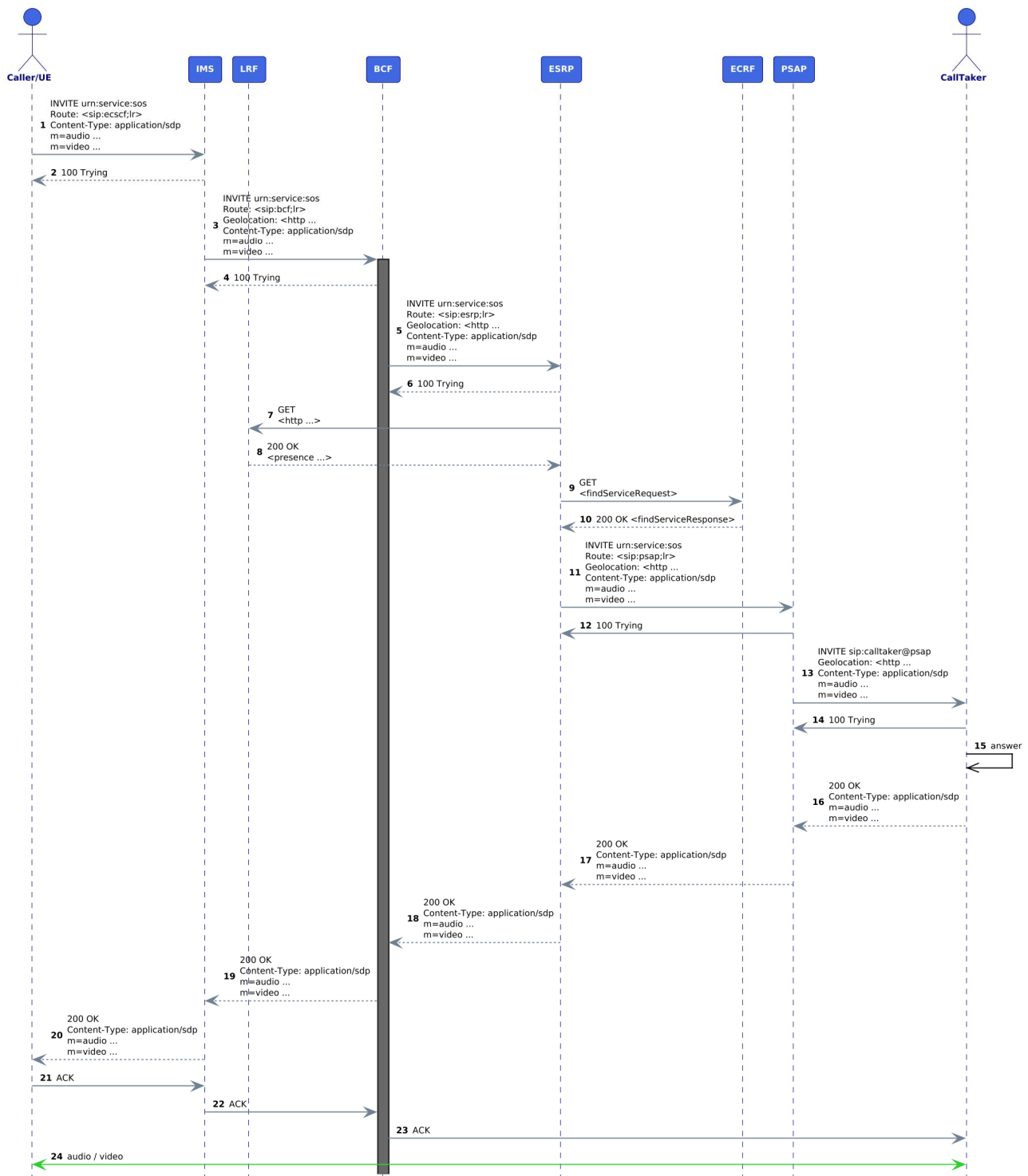
This test shall use the configuration as described in clause 5.3 IMS of the present document.

The originating network infrastructure shall support proper audio and video codecs as defined in ETSI TS 103 479 [1].

7.2.5.2 Description

Interoperability Test Description			
Identifier	MME_VID_02		
Test Objective	Verify the standard-compliant transmission of audio (a) and video (v) from the originating network (IMS) to the control centre (PSAP) transmitted via RTP		
Configuration	IMS		
References	ETSI TS 103 479 [1], ETSI TS 123 167 [2], ETSI TS 123 333 [3]		
Applicability	IMS (Le, Mm, Mb) BCF (SIP-1, SIP-2, RTP-1, RTP-2) ESRP (SIP-1, SIP-2, LOST-1, LOST-2) ECRF (LOST-1, LOST-2) PSAP (SIP-1, SIP-2, RTP-1, RTP-2)		
Pre-test conditions	• IP connectivity among all elements of the specific scenario • ECSCF trigger points for emergency call routing • LRF has location information (value) available • BCF trigger points for emergency call routing (to ESRP) • ESRP trigger points for emergency call routing • ESRP configured to query the ECRF • ECRF configured with correct mapping		
Test Sequence	Step	Type	Description
	1	stimulus	User dials test emergency number
	2	check	Dialog creating INVITE (a/v) received at BCF
	3	check	Dialog creating INVITE (a/v) received at ESRP
	4	check	LoST request received at ECRF
	5	check	ESRP emergency call routing based on location
	6	check	Dialog creating INVITE (a/v) received at PSAP
	7	check	SIP dialog established
	8	verify	Call (a/v) connected and location displayed

7.2.5.3 Message Sequence Chart



7.2.6 Real-time Text (VoIP)

7.2.6.1 General

Identifier: MME_RTT_01

This test is intended to verify the standard-compliant transmission of real-time text from the originating network (VoIP) to the control centre (PSAP) transmitted via RTP.

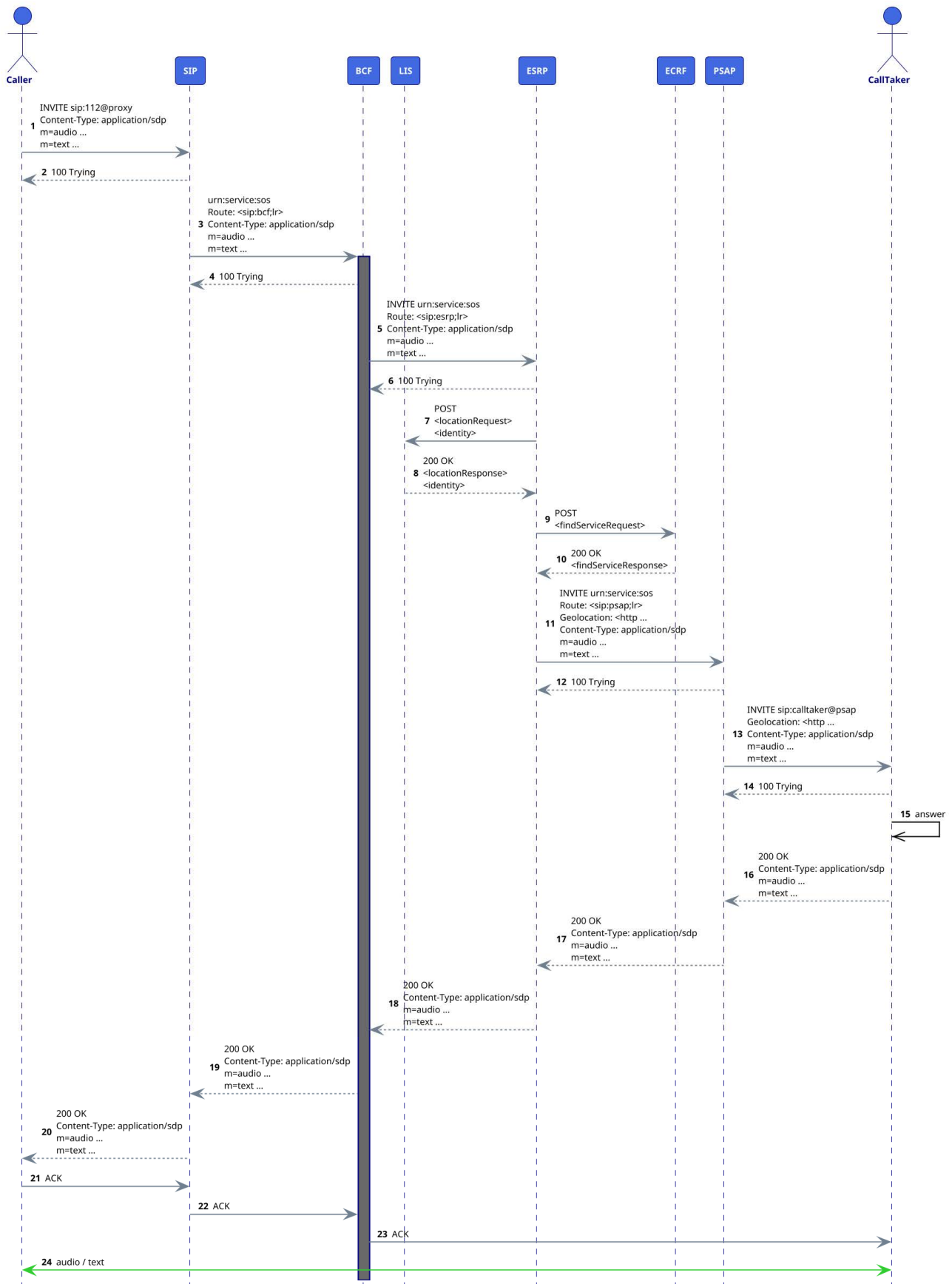
This test shall use the configuration as described in clause 5.2 VoIP of the present document. In the case an UE does not support either LbV or LbR, the location information can also be determined via LIS and a UE identity (tel or sip URI) from the ESRP. Optionally, LIS* and ESRP* can be operated outside the ESInet for this purpose.

The originating network infrastructure shall support proper text encoding as defined in ETSI TS 103 479 [1].

7.2.6.2 Description

Interoperability Test Description			
Identifier	MME_RTT_01		
Test Objective	Verify the standard-compliant transmission of audio (a) and real-time text (t) from the originating network (VoIP) to the control centre (PSAP) transmitted via RTP		
Configuration	VoIP		
References	ETSI TS 103 479 [1]		
Applicability	UE (SIP-1, SIP-2, RTP-1, RTP-2) LIS (HELD-1, HELD-2) BCF (SIP-1, SIP-2, RTP-1, RTP-2) ESRP (SIP-1, SIP-2, LOST-1, LOST-2, HELD-2) ECRF (LOST-1, LOST-2) PSAP (SIP-1, SIP-2, RTP-1, RTP-2, HELD-2)		
Pre-test conditions	• IP connectivity among all elements of the specific scenario • UE configured to register with SIP Proxy/Registrar • UE has location information (reference) available • SIP Proxy trigger points for emergency call routing (to BCF) • BCF trigger points for emergency call routing (to ESRP) • ESRP trigger points for emergency call routing • ESRP configured to query the ECRF • ECRF configured with correct mapping		
Test Sequence	Step	Type	Description
	1	stimulus	User dials emergency number (e.g. 112)
	2	check	Dialog creating INVITE (a/t) received at BCF
	3	check	Dialog creating INVITE (a/t) received at ESRP
	4	check	ESRP queries location information (by identity)
	5	check	LoST request received at ECRF
	6	check	ESRP emergency call routing based on location
	7	check	Dialog creating INVITE (a/t) received at PSAP
	8	check	SIP dialog established
	9	verify	Call (a/t) connected and location displayed

7.2.6.3 Message Sequence Chart



7.2.7 Real-time Text (IMS)

7.2.7.1 General

Identifier: MME_RTT_02

This test is intended to verify the standard-compliant transmission of real-time text from the originating network (IMS) to the control centre (PSAP) transmitted via RTP.

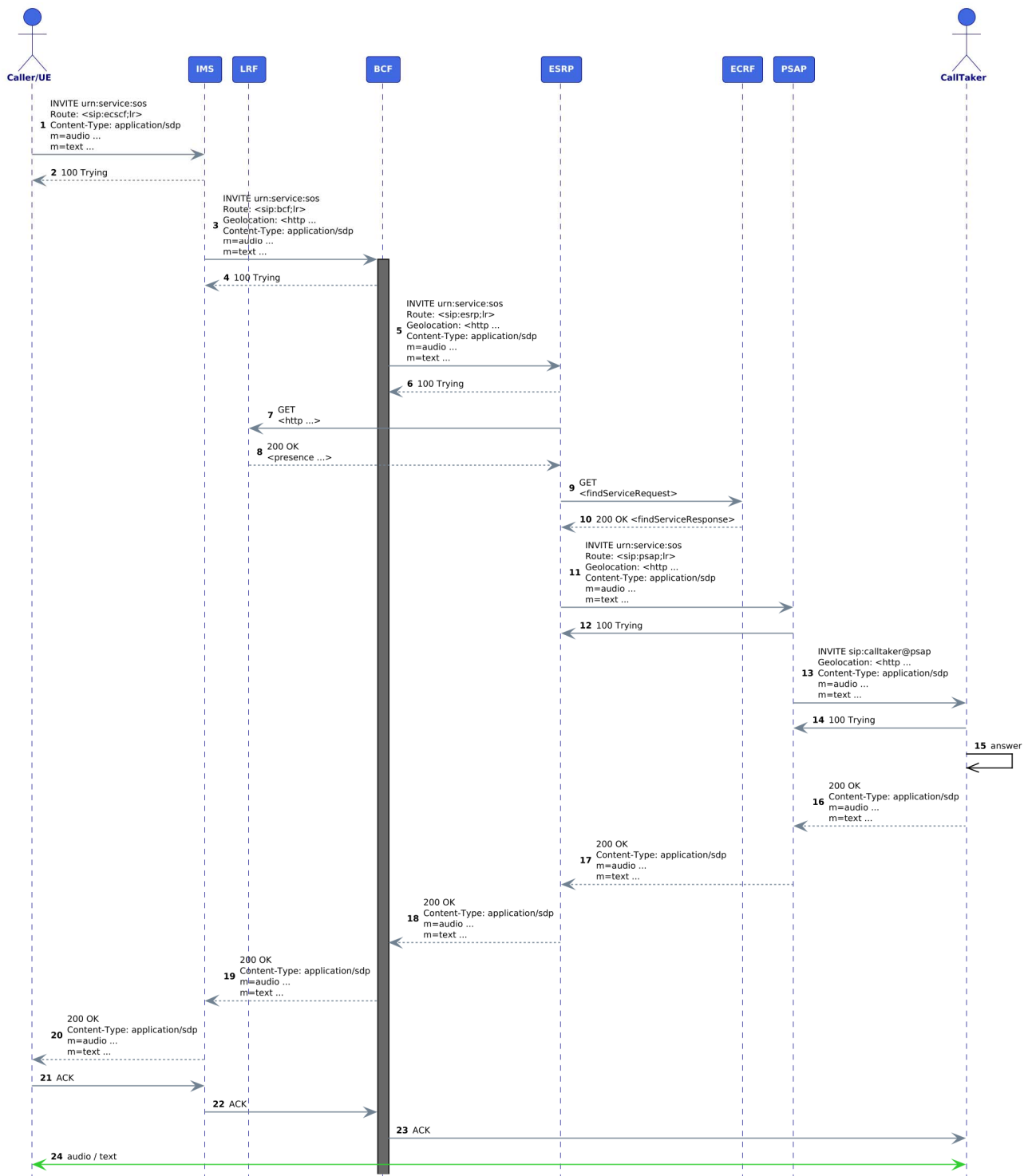
This test shall use the configuration as described in clause 5.2 VoIP of the present document. In the case an UE does not support either LbV or LbR, the location information can also be determined via LIS and a UE identity (tel or sip URI) from the ESRP. Optionally, LIS* and ESRP* can be operated outside the ESIInet for this purpose.

The originating network infrastructure shall support proper text encoding as defined in ETSI TS 103 479 [1].

7.2.7.2 Description

Interoperability Test Description			
Identifier	MME_RTT_02		
Test Objective	Verify the standard-compliant transmission of audio (a) and real-time text (t) from the originating network (IMS) to the control centre (PSAP) transmitted via RTP		
Configuration	IMS		
References	ETSI TS 103 479 [1], ETSI TS 123 167 [2], ETSI TS 123 333 [3]		
Applicability	IMS (Le, Mm, Mb) BCF (SIP-1, SIP-2, RTP-1, RTP-2) ESRP (SIP-1, SIP-2, LOST-1, LOST-2) ECRF (LOST-1, LOST-2) PSAP (SIP-1, SIP-2, RTP-1, RTP-2)		
Pre-test conditions	• IP connectivity among all elements of the specific scenario • ECSCF trigger points for emergency call routing • LRF has location information (value) available • BCF trigger points for emergency call routing (to ESRP) • ESRP trigger points for emergency call routing • ESRP configured to query the ECRF • ECRF configured with correct mapping		
Test Sequence	Step	Type	Description
	1	stimulus	User dials test emergency number
	2	check	Dialog creating INVITE (a/t) received at BCF
	3	check	Dialog creating INVITE (a/t) received at ESRP
	4	check	LoST request received at ECRF
	5	check	ESRP emergency call routing based on location
	6	check	Dialog creating INVITE (a/t) received at PSAP
	7	check	SIP dialog established
	8	verify	Call (a/t) connected and location displayed

7.2.7.3 Message Sequence Chart



7.2.8 Total Conversation (VoIP)

7.2.8.1 General

Identifier: MME_TOC_01

This test is intended to verify the standard-compliant transmission of audio, video and real-time text from the originating network (VoIP) to the control centre (PSAP) transmitted via RTP.

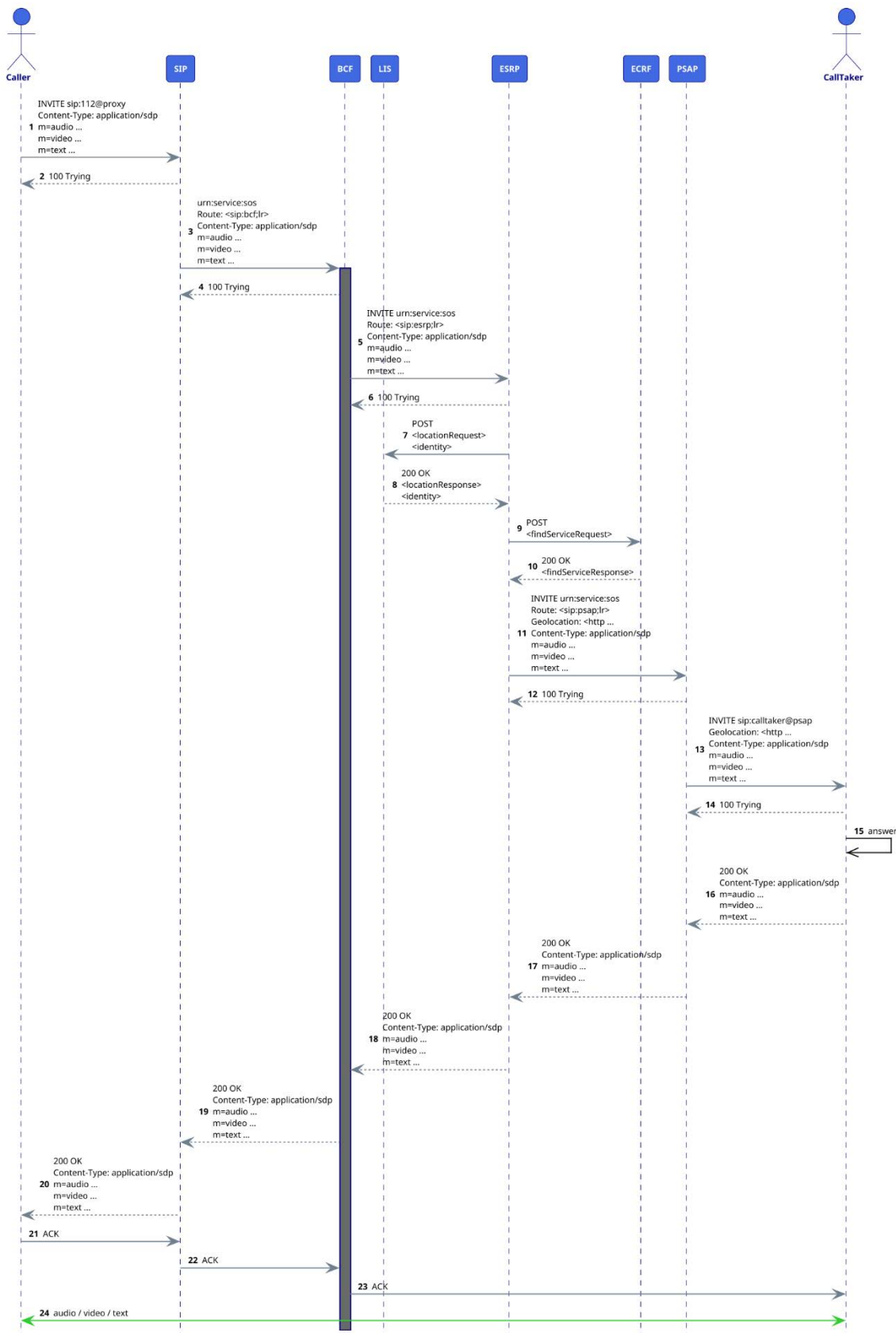
This test shall use the configuration as described in clause 5.2 VoIP of the present document. In the case an UE does not support either LbV or LbR, the location information can also be determined via LIS and a UE identity (tel or sip URI) from the ESRP. Optionally, LIS* and ESRP* can be operated outside the ESInet for this purpose.

The originating network infrastructure shall support proper audio, video codec and text encoding as defined in ETSI TS 103 479 [1].

7.2.8.2 Description

Interoperability Test Description			
Identifier	MME_TOC_01		
Test Objective	Verify the standard-compliant transmission of audio (a), video (v) and real-time text (t) from the originating network (VoIP) to the control centre (PSAP) transmitted via RTP		
Configuration	VoIP		
References	ETSI TS 103 479 [1]		
Applicability	UE (SIP-1, SIP-2, RTP-1, RTP-2) LIS (HELD-1, HELD-2) BCF (SIP-1, SIP-2, RTP-1, RTP-2) ESRP (SIP-1, SIP-2, LOST-1, LOST-2, HELD-2) ECRF (LOST-1, LOST-2) PSAP (SIP-1, SIP-2, RTP-1, RTP-2, HELD-2)		
Pre-test conditions	• IP connectivity among all elements of the specific scenario • UE configured to register with SIP Proxy/Registrar • UE has location information (reference) available • SIP Proxy trigger points for emergency call routing (to BCF) • BCF trigger points for emergency call routing (to ESRP) • ESRP trigger points for emergency call routing • ESRP configured to query LIS and ECRF • ECRF configured with correct mapping		
Test Sequence	Step	Type	Description
	1	stimulus	User dials emergency number (e.g. 112)
	2	check	Dialog creating INVITE (a/v/t) received at BCF
	3	check	Dialog creating INVITE (a/v/t) received at ESRP
	4	check	ESRP queries location information (by identity)
	5	check	LoST request received at ECRF
	6	check	ESRP emergency call routing based on location
	7	check	Dialog creating INVITE (a/v/t) received at PSAP
	8	check	SIP dialog established
	9	verify	Call (a/v/t) connected and location displayed

7.2.8.3 Message Sequence Chart



7.2.9 Total Conversation (IMS)

7.2.9.1 General

Identifier: MME_TOC_02

This test is intended to verify the standard-compliant transmission of audio, video and real-time text from the originating network (IMS) to the control centre (PSAP) transmitted via RTP.

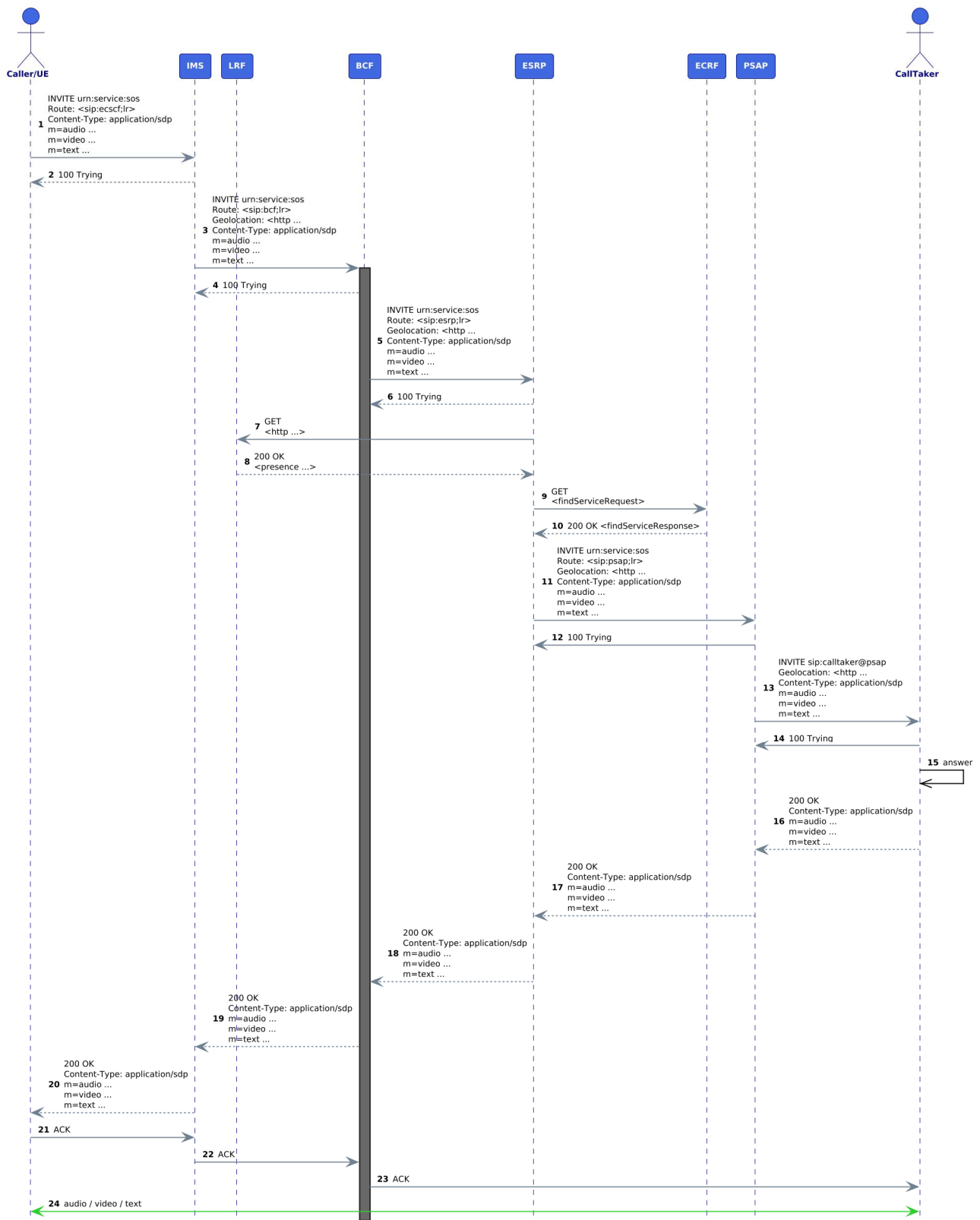
This test shall use the configuration as described in clause 5.3 IMS of the present document.

The originating network infrastructure shall support proper audio, video codec and text encoding as defined in ETSI TS 103 479 [1].

7.2.9.2 Description

Interoperability Test Description			
Identifier	MME_TOC_02		
Test Objective	Verify the standard-compliant transmission of audio (a), video (v) and real-time text (t) from the originating network (IMS) to the control centre (PSAP) transmitted via RTP		
Configuration	IMS		
References	ETSI TS 103 479 [1], ETSI TS 123 167 [2], ETSI TS 123 333 [3]		
Applicability	IMS (Le, Mm, Mb) BCF (SIP-1, SIP-2, RTP-1, RTP-2) ESRP (SIP-1, SIP-2, LOST-1, LOST-2) ECRF (LOST-1, LOST-2) PSAP (SIP-1, SIP-2, RTP-1, RTP-2)		
Pre-test conditions	• IP connectivity among all elements of the specific scenario • ECSCF trigger points for emergency call routing • LRF has location information (value) available • BCF trigger points for emergency call routing (to ESRP) • ESRP trigger points for emergency call routing • ESRP configured to query the ECRF • ECRF configured with correct mapping		
Test Sequence	Step	Type	Description
	1	stimulus	User dials test emergency number
	2	check	Dialog creating INVITE (a/v/t) received at BCF
	3	check	Dialog creating INVITE (a/v/t) received at ESRP
	4	check	LoST request received at ECRF
	5	check	ESRP emergency call routing based on location
	6	check	Dialog creating INVITE (a/v/t) received at PSAP
	7	check	SIP dialog established
	8	verify	Call (a/v/t) connected and location displayed

7.2.9.3 Message Sequence Chart



7.3 Policy

7.3.1 Time and Date (VoIP, IMS, PSTN/PABX)

7.3.1.1 General

Identifier: POL_TAD_01

This test is intended to verify standard-compliant policy routing of emergency calls originating from different networks (VoIP, IMS, PSTN/PABX) and forwarded to a control centre (PSAP) based on location and date and/or time conditions.

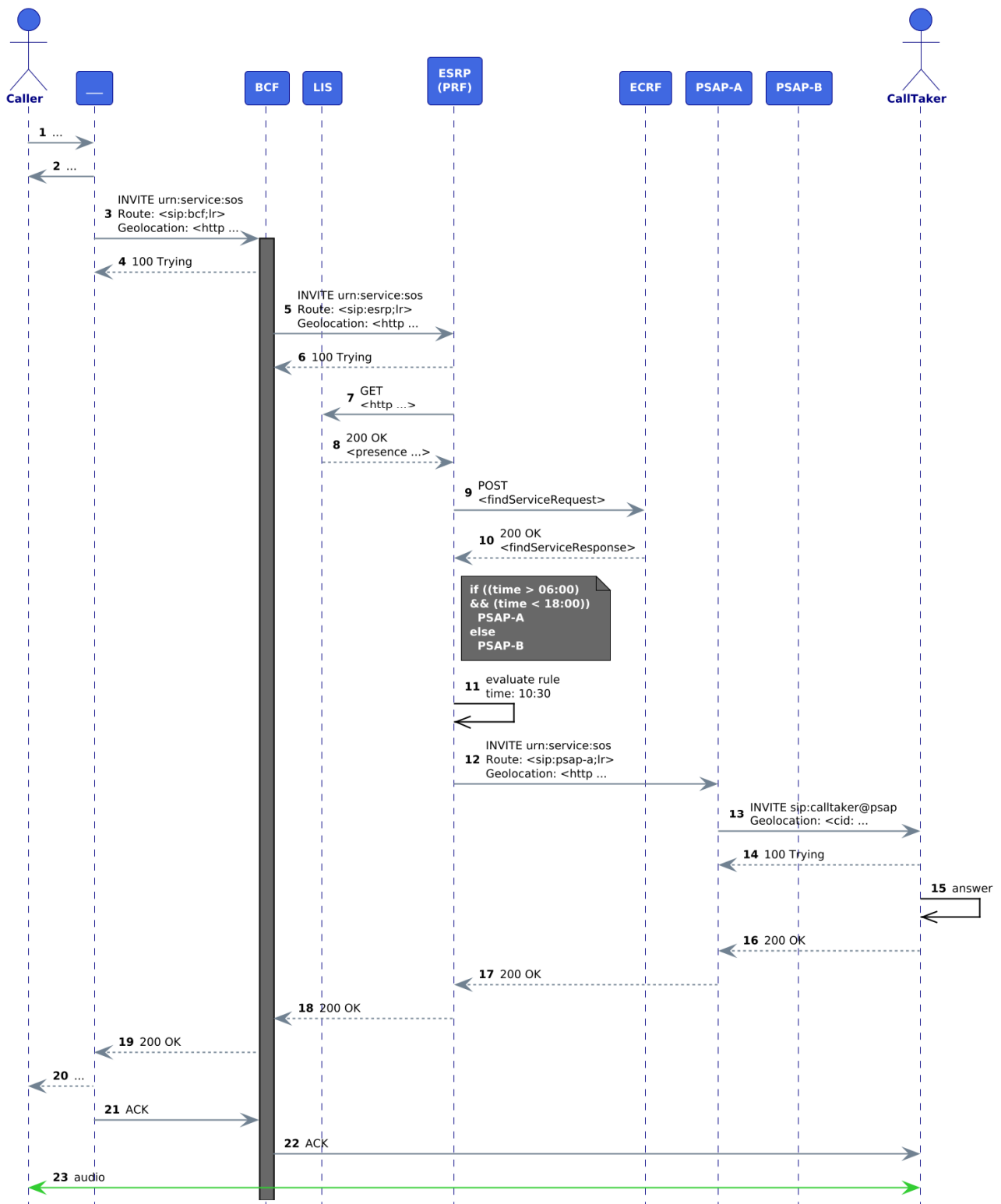
This test shall use the configuration as described in clauses 5.2 VoIP, 5.3 IMS, and 5.4 PSTN/PABX of the present document. In the case an UE does not support either LbV or LbR, the location information can also be determined via LIS and a UE identity (tel or sip URI) from the ESRP. Optionally, LIS* and ESRP* can be operated outside the ESInet for this purpose.

The originating network infrastructure shall support proper audio codecs as defined in ETSI TS 103 479 [1].

7.3.1.2 Description

Interoperability Test Description			
Identifier	POL_TAD_01		
Test Objective	Verify standard-compliant policy routing of emergency calls originating from different networks (VoIP, IMS, PSTN/PABX) and forwarded to a control centre (PSAP) based on location and date and/or time conditions		
Configuration	VoIP, IMS, PSTN/PABX		
References	ETSI TS 103 479 [1]		
Applicability	UE (SIP-1, SIP-2, RTP-1, RTP-2) IMS (Le, Mm, Mb) LNG ([UE] + HELD-2) LIS (HELD-1, HELD-2) BCF (SIP-1, SIP-2, RTP-1, RTP-2) ESRP (SIP-1, SIP-2, LOST-1, LOST-2, HELD-2) ECRF (LOST-1, LOST-2) PSAP (SIP-1, SIP-2, RTP-1, RTP-2, HELD-2)		
Pre-test conditions	• IP connectivity among all elements of the specific scenario • UE configured to register with SIP Proxy/Registrar • UE has location information (reference) available • SIP Proxy trigger points for emergency call routing (to BCF) • BCF trigger points for emergency call routing (to ESRP) • PRF configured with proper ruleset (date/time condition) • ESRP trigger points for emergency call routing • ESRP configured to query LIS and ECRF • ECRF configured with correct mapping		
Test Sequence	Step	Type	Description
	1	stimulus	User dials emergency number
	2	check	Dialog creating INVITE received at BCF
	3	check	Dialog creating INVITE received at ESRP
	4	check	ESRP queries location information (by identity)
	5	check	LoST request received at ECRF
	6	check	PRF applies specific rule
	7	check	ESRP emergency call routing based on condition
	8	check	Dialog creating INVITE received at PSAP
	9	check	SIP dialog established
	10	verify	Call connected and location displayed

7.3.1.3 Message Sequence Chart



7.3.2 SIP Header (VoIP, IMS, PSTN/PABX)

7.3.2.1 General

Identifier: POL_SIP_01

This test is intended to verify standard-compliant policy routing of emergency calls originating from different networks (VoIP, IMS, PSTN/PABX) and forwarded to a control centre (PSAP) based on location and certain SIP header value conditions (e.g. SIP From header value).

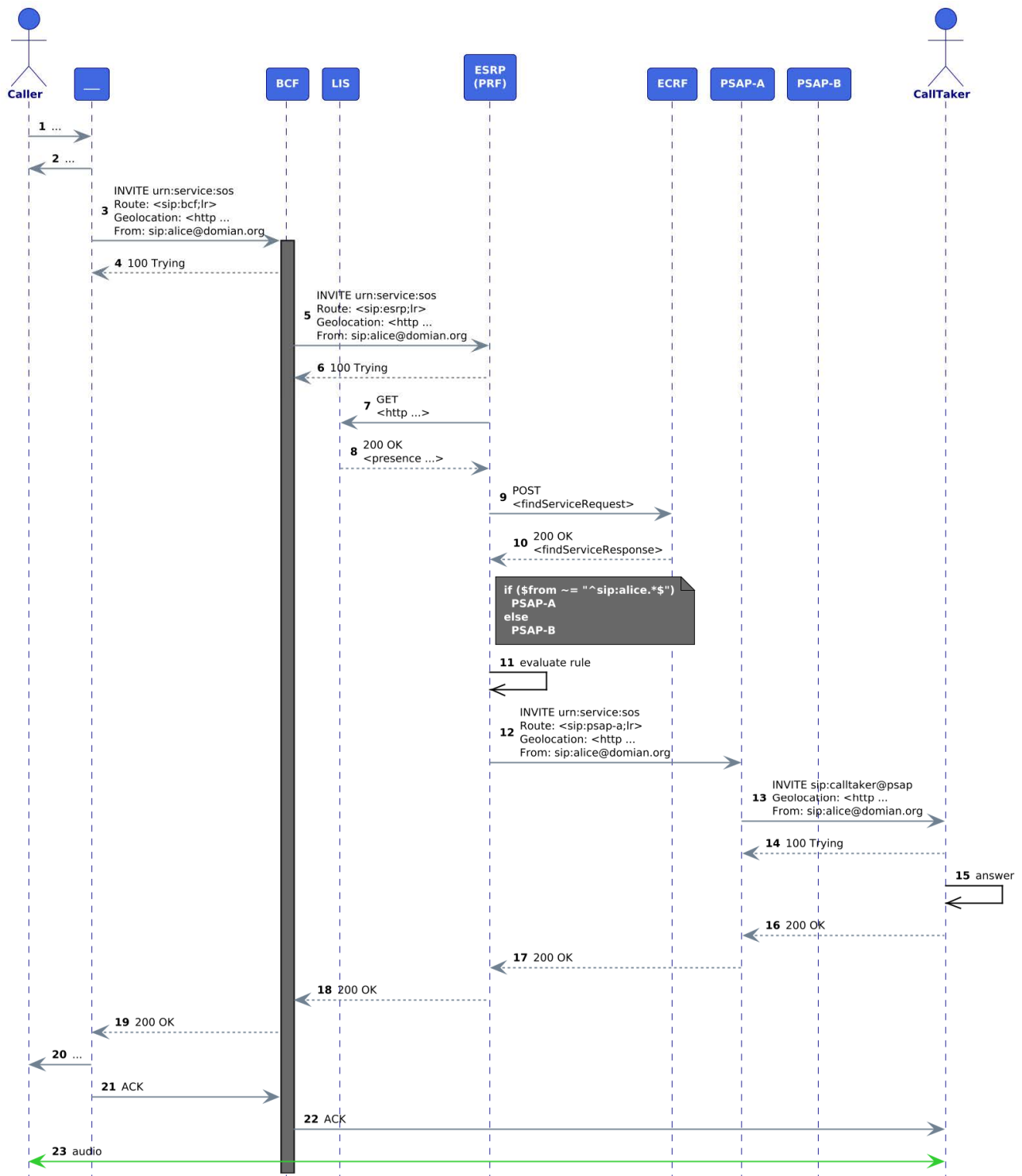
This test shall use the configuration as described in clauses 5.2 VoIP, 5.3 IMS, and 5.4 PSTN/PABX of the present document. In the case an UE does not support either LbV or LbR, the location information can also be determined via LIS and a UE identity (tel or sip URI) from the ESRP. Optionally, LIS* and ESRP* can be operated outside the ESInet for this purpose.

The originating network infrastructure shall support proper audio codecs as defined in ETSI TS 103 479 [1].

7.3.2.2 Description

Interoperability Test Description			
Identifier	POL_SIP_01		
Test Objective	Verify standard-compliant policy routing of emergency calls originating from different networks (VoIP, IMS, PSTN/PABX) and forwarded to a control centre (PSAP) based on location and certain SIP header value conditions		
Configuration	VoIP, IMS, PSTN/PABX		
References	ETSI TS 103 479 [1]		
Applicability	UE (SIP-1, SIP-2, RTP-1, RTP-2) IMS (Le, Mm, Mb) LNG ([UE] + HELD-2) LIS (HELD-1, HELD-2) BCF (SIP-1, SIP-2, RTP-1, RTP-2) ESRP (SIP-1, SIP-2, LOST-1, LOST-2, HELD-2) ECRF (LOST-1, LOST-2) PSAP (SIP-1, SIP-2, RTP-1, RTP-2, HELD-2)		
Pre-test conditions	• IP connectivity among all elements of the specific scenario • UE configured to register with SIP Proxy/Registrar • UE has location information (reference) available • SIP Proxy trigger points for emergency call routing (to BCF) • BCF trigger points for emergency call routing (to ESRP) • PRF configured with proper ruleset (SIP header value condition) • ESRP trigger points for emergency call routing • ESRP configured to query LIS and ECRF • ECRF configured with correct mapping		
Test Sequence	Step	Type	Description
	1	stimulus	User dials emergency number
	2	check	Dialog creating INVITE received at BCF
	3	check	Dialog creating INVITE received at ESRP
	4	check	ESRP queries location information (by identity)
	5	check	LoST request received at ECRF
	6	check	PRF applies specific rule
	7	check	ESRP emergency call routing based on condition
	8	check	Dialog creating INVITE received at PSAP
	9	check	SIP dialog established
	10	verify	Call connected and location displayed

7.3.2.3 Message Sequence Chart



7.4 Lost Hierarchy

7.4.1 Iteration (ESRP)

7.4.1.1 General

Identifier: LST_ITE01

This test is intended to verify the standard-compliant handling of LoST Redirect responses and iterative LoST querying on the ESRP.

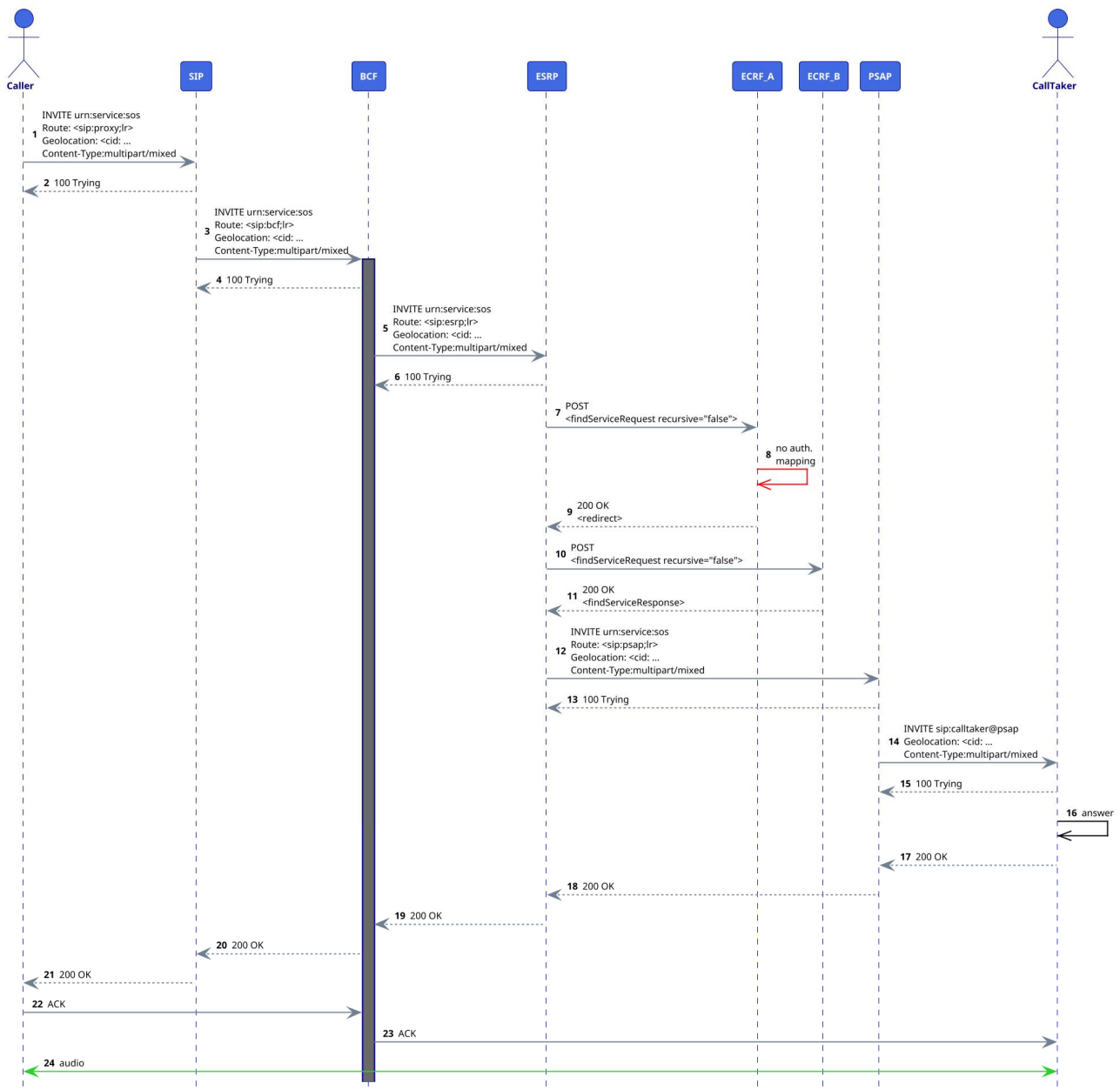
This test shall use the configuration as described in clause 5.2 VoIP of the present document.

The originating network infrastructure shall support proper audio codecs as defined in ETSI TS 103 479 [1].

7.4.1.2 Description

Interoperability Test Description			
Identifier	LST_ITE_01		
Test Objective	Verify the standard-compliant handling of LoST Redirect responses and iterative LoST querying on the ESRP.		
Configuration	VoIP		
References	ETSI TS 103 479 [1]		
Applicability	UE (SIP-1, SIP-2, RTP-1, RTP-2) BCF (SIP-1, SIP-2, RTP-1, RTP-2) ESRP (SIP-1, SIP-2, LOST-1, LOST-2) ECRF (LOST-1, LOST-2) PSAP (SIP-1, SIP-2, RTP-1, RTP-2)		
Pre-test conditions	• IP connectivity among all elements of the specific scenario • UE configured to register with SIP Proxy/Registrar • UE has location information (value) available • SIP Proxy trigger points for emergency call routing (to BCF) • BCF trigger points for emergency call routing (to ESRP) • ESRP trigger points for emergency call routing • ESRP configured to query the ECRF_A • ECRF_A configured with no mapping and ECRF_B as the higher level ECRF • ECRF_B configured with correct mapping		
Test Sequence	Test Sequence		
	Step	Type	Description
	1	stimulus	User dials emergency number (e.g. 112)
	2	check	Dialog creating INVITE (LbV) received at BCF
	3	check	Dialog creating INVITE (LbV) received at ESRP
	4	check	LoST request received at ECRF_A
	5	check	LoST request received at ECRF_B (from ESRP)
	6	check	ESRP emergency call routing based on location
	7	check	Dialog creating INVITE (LbV) received at PSAP
	8	check	SIP dialog established
	9	verify	PIDF/LO (LbV) received at PSAP
	10	verify	Call connected and location displayed

7.4.1.3 Message Sequence Chart



7.4.2 Recursion (ESRP)

7.4.2.1 General

Identifier: LST_REC_01

This test is intended to verify the standard-compliant recursive handling of LoST requests on the ECRF.

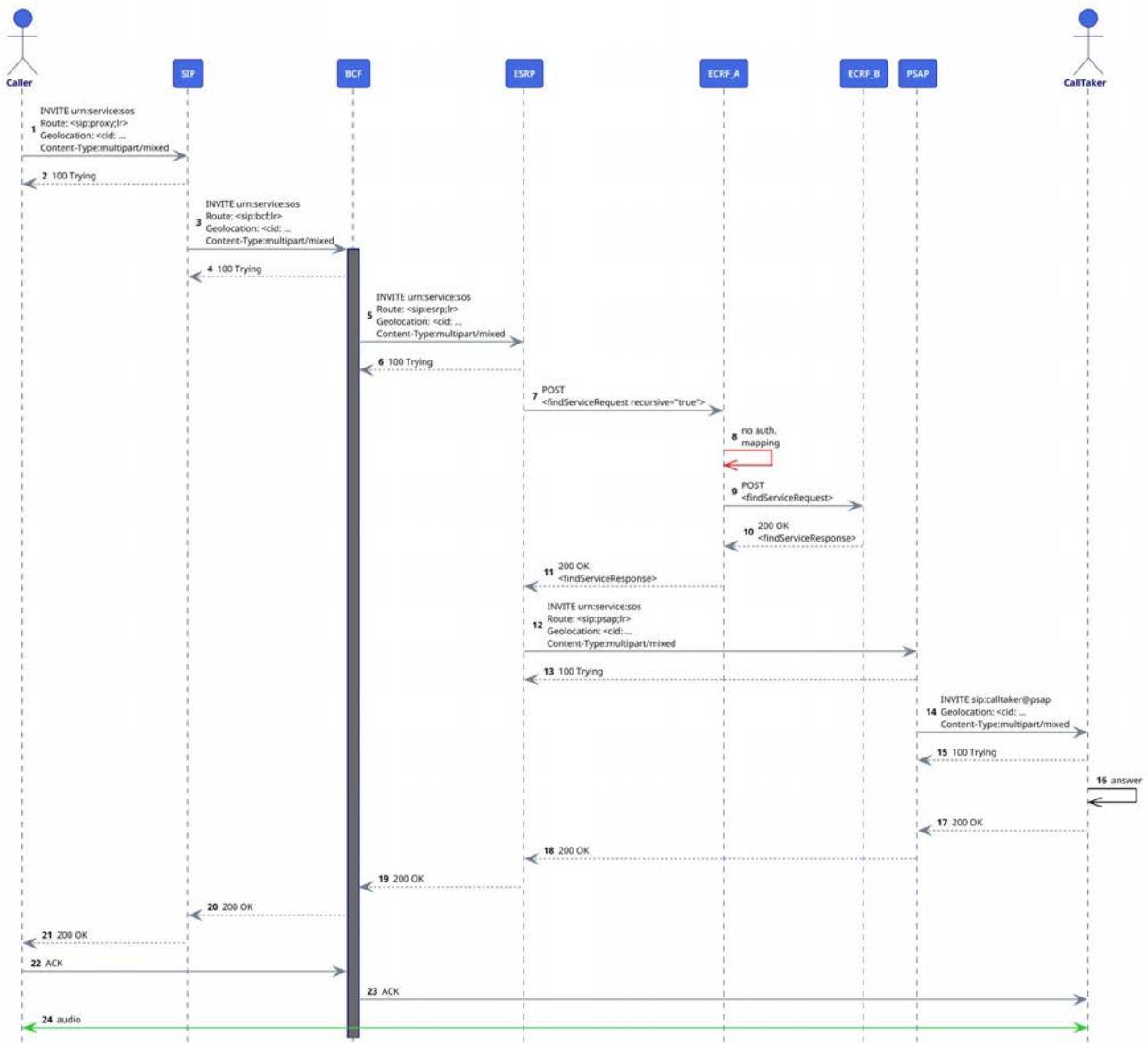
This test shall use the configuration as described in clause 5.2 VoIP of the present document.

The originating network infrastructure shall support proper audio codecs as defined in ETSI TS 103 479 [1].

7.4.2.2 Description

Interoperability Test Description			
Identifier	LST_REC_01		
Test Objective	Verify the standard-compliant recursive handling of LoST requests on the ECRF.		
Configuration	VoIP		
References	ETSI TS 103 479 [1]		
Applicability	UE (SIP-1, SIP-2, RTP-1, RTP-2) BCF (SIP-1, SIP-2, RTP-1, RTP-2) ESRP (SIP-1, SIP-2, LOST-1, LOST-2) ECRF (LOST-1, LOST-2) PSAP (SIP-1, SIP-2, RTP-1, RTP-2)		
Pre-test conditions	• IP connectivity among all elements of the specific scenario • UE configured to register with SIP Proxy/Registrar • UE has location information (value) available • SIP Proxy trigger points for emergency call routing (to BCF) • BCF trigger points for emergency call routing (to ESRP) • ESRP trigger points for emergency call routing • ESRP configured to query the ECRF_A • ECRF_A configured with no mapping and ECRF_B as the higher level ECRF • ECRF_B configured with correct mapping		
Test Sequence	Step	Type	Description
	1	stimulus	User dials emergency number (e.g. 112)
	2	check	Dialog creating INVITE (LbV) received at BCF
	3	check	Dialog creating INVITE (LbV) received at ESRP
	4	Check	LoST request received at ECRF_A
	5	Check	LoST request received at ECRF_B (from ECRF_A)
	6	check	ESRP emergency call routing based on location
	7	check	Dialog creating INVITE (LbV) received at PSAP
	8	check	SIP dialog established
	9	verify	PIDF/LO (LbV) received at PSAP
	10	verify	Call connected and location displayed

7.4.2.3 Message Sequence Chart



7.4.3 Recursion (ECRF) with FG

7.4.3.1 General

Identifier: LST_REC_02

This test is intended to verify the standard-compliant recursive handling of LoST requests on the ECRF when it receives a LoST redirect from a FG.

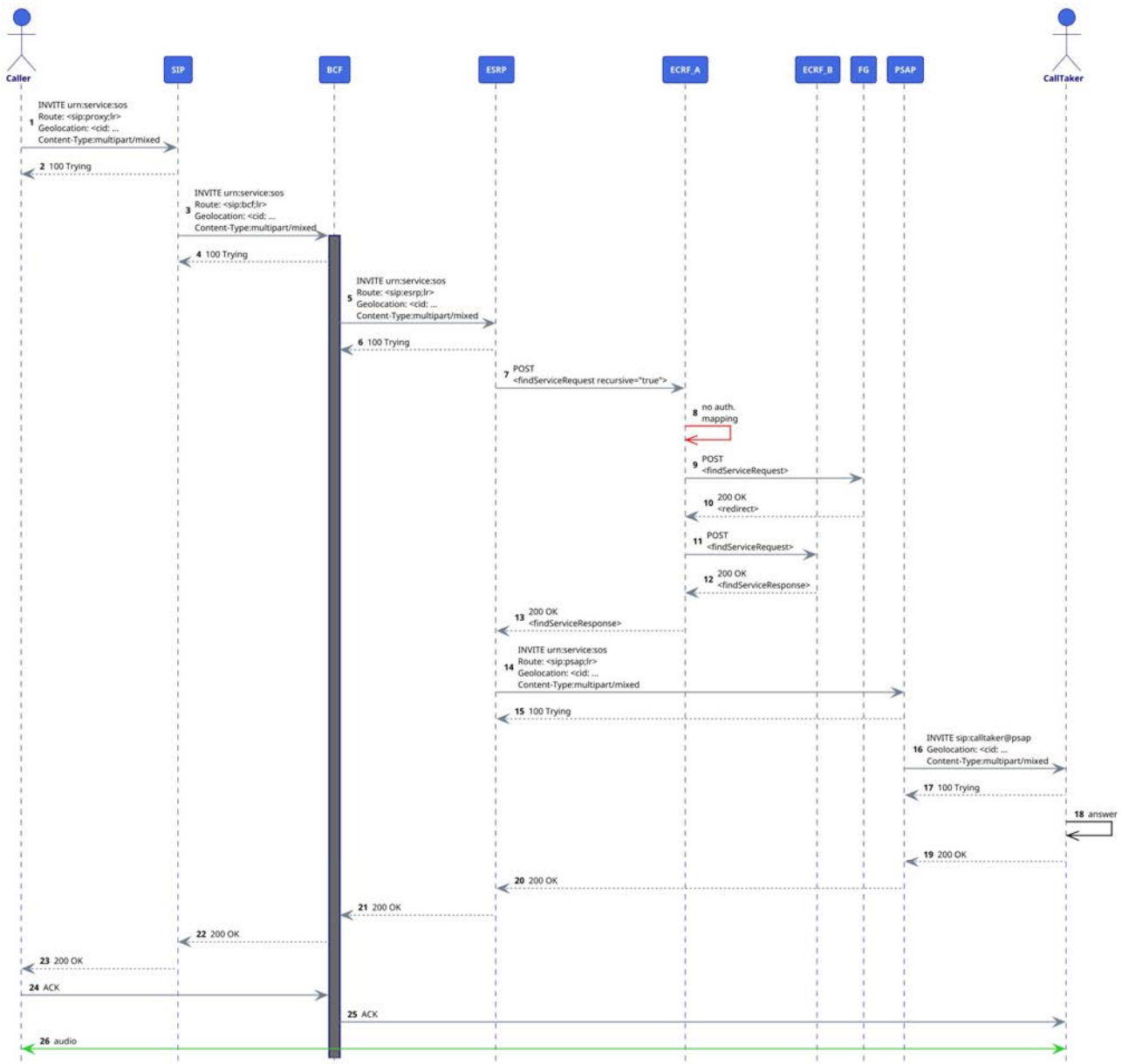
This test shall use the configuration as described in clause 5.2 VoIP of the present document.

The originating network infrastructure shall support proper audio codecs as defined in ETSI TS 103 479 [1].

7.4.3.2 Description

Interoperability Test Description			
Identifier	LST_REC_02		
Test Objective	Verify the standard-compliant recursive handling of LoST requests on the ECRF.		
Configuration	VoIP		
References	ETSI TS 103 479 [1]		
Applicability	UE (SIP-1, SIP-2, RTP-1, RTP-2) BCF (SIP-1, SIP-2, RTP-1, RTP-2) ESRP (SIP-1, SIP-2, LOST-1, LOST-2) ECRF (LOST-1, LOST-2) PSAP (SIP-1, SIP-2, RTP-1, RTP-2)		
Pre-test conditions	• IP connectivity among all elements of the specific scenario • UE configured to register with SIP Proxy/Registrar • UE has location information (value) available • SIP Proxy trigger points for emergency call routing (to BCF) • BCF trigger points for emergency call routing (to ESRP) • ESRP trigger points for emergency call routing • ESRP configured to query the ECRF_A • ECRF_A configured with no mapping and and FG as Forest Guide • FG configured to return a redirect to ECRF_B • ECRF_B configured with correct mapping		
Test Sequence	Step	Type	Description
	1	stimulus	User dials emergency number (e.g. 112)
	2	check	Dialog creating INVITE (LbV) received at BCF
	3	check	Dialog creating INVITE (LbV) received at ESRP
	4	check	LoST request received at ECRF_A
	5	check	LoST request received at FG (from ECRF_A)
	6	check	LoST request received at ECRF_B (from ECRF_A)
	7	check	ESRP emergency call routing based on location
	8	check	Dialog creating INVITE (LbV) received at PSAP
	9	check	SIP dialog established
	10	verify	PIDF/LO (LbV) received at PSAP
	11	verify	Call connected and location displayed

7.4.3.3 Message Sequence Chart



7.5 Messaging

7.5.1 Basic Chat (VoIP)

7.5.1.1 General

Identifier: MSG_LMP_01

This test is intended to verify the standard-compliant transmission of lightweight chat messages from the originating network (VoIP) to the control centre (PSAP) using LMPE.

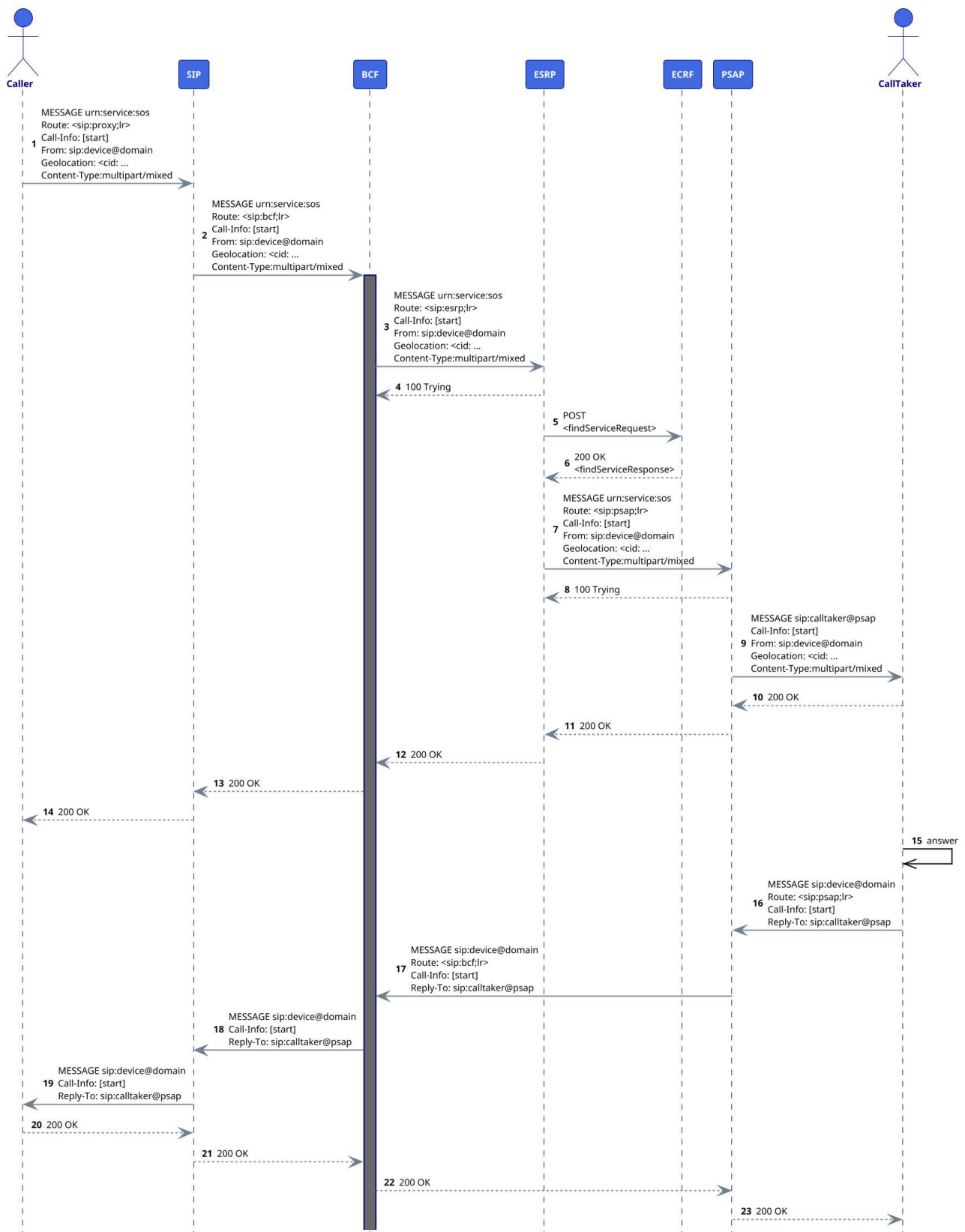
This test shall use the configuration as described in clause 5.2 VoIP of the present document.

The originating network infrastructure and the PSAP shall support LMPE as defined in ETSI TS 103 698 [5].

7.5.1.2 Description

Interoperability Test Description			
Identifier	MSG_LMP_01		
Test Objective	Verify the standard-compliant transmission of lightweight chat messages from the originating network (VoIP) to the control centre (PSAP)		
Configuration	VoIP		
References	ETSI TS 103 479 [1], ETSI TS 103 698 [5]		
Applicability	UE (SIP-1, SIP-2, IM-2) BCF (SIP-1, SIP-2, RTP-1, RTP-2) ESRP (SIP-1, SIP-2, LOST-1, LOST-2) ECRF (LOST-1, LOST-2) PSAP (SIP-1, SIP-2, IM-2)		
Pre-test conditions	• IP connectivity among all elements of the specific scenario • UE configured to register with SIP Proxy/Registrar • UE has location information (value) available • SIP Proxy trigger points for emergency call routing (to BCF) • BCF trigger points for emergency call routing (to ESRP) • ESRP trigger points for emergency call routing • ESRP configured to query the ECRF • ECRF configured with correct mapping • UE and PSAP support LMPE		
Test Sequence	Step	Type	Description
	1	stimulus	User dials emergency number
	2	check	Initial SIP MESSAGE (LbV) received at BCF (<i>start</i>)
	3	check	Initial SIP MESSAGE (LbV) received at ESRP (<i>start</i>)
	4	check	LoST request received at ECRF
	5	check	ESRP emergency call routing based on location
	6	check	Initial SIP MESSAGE (LbV) received at PSAP (<i>start</i>)
	7	check	PSAP responds with SIP MESSAGE (Reply-To, <i>start</i>)
	8	verify	PIDF/LO (LbV) received at PSAP
	9	verify	Chat connected (<i>in-chat</i>) and location displayed

7.5.1.3 Message Sequence Chart



7.6 NG eCall

7.6.1 Basic Communication (IMS)

7.6.1.1 General

Identifier: ECL_BAS_01

This test is intended to verify the standard-compliant transmission of MSD data and audio from the IVS via the originating network (IMS) to the control centre (PSAP) transmitted via SIP and RTP.

This test shall use the configuration as described in clause 5.3 IMS of the present document.

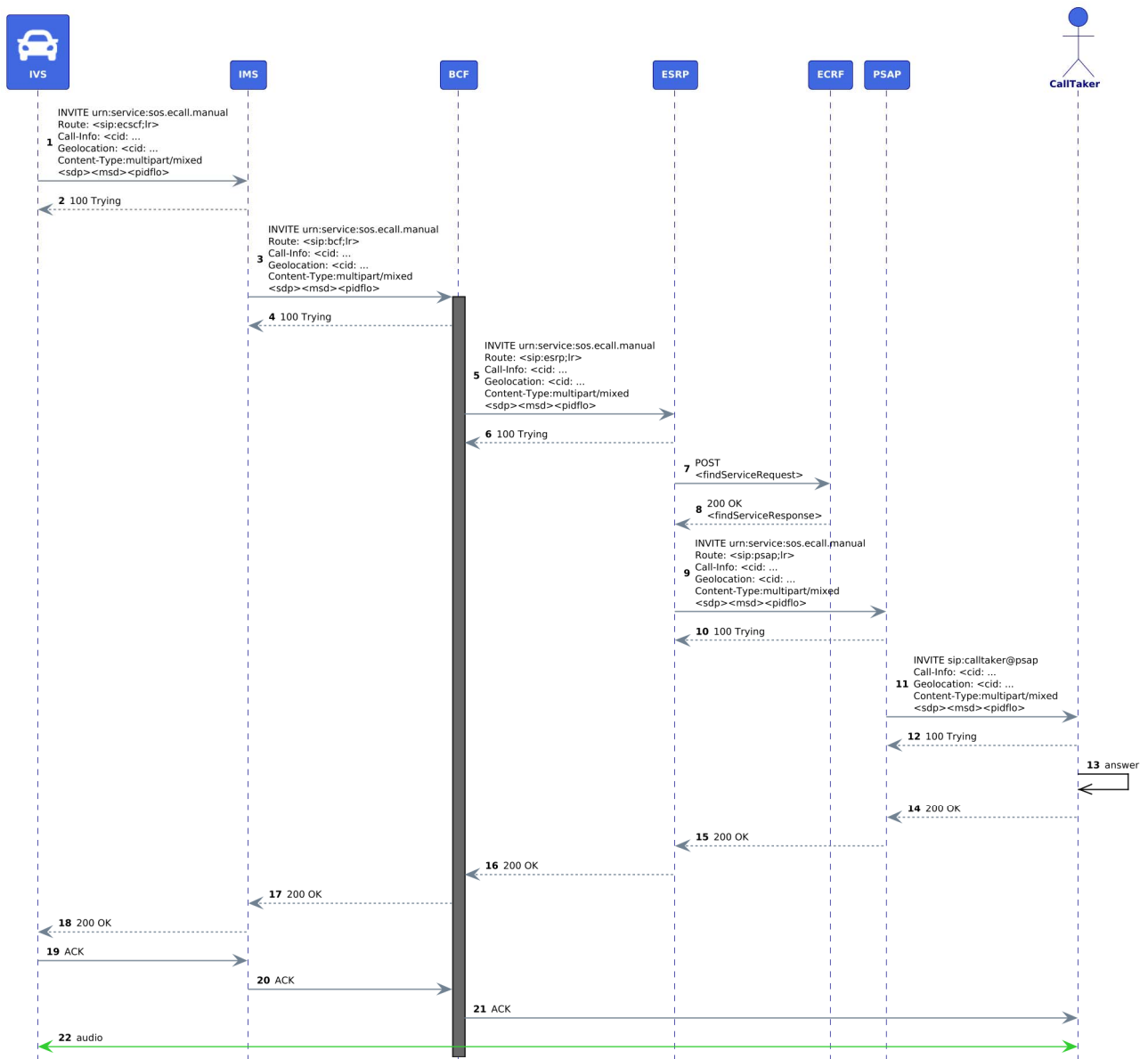
The originating network infrastructure shall support proper audio codecs as defined in ETSI TS 103 479 [1].

NOTE: For NG eCall tests, it is assumed that the IVS provides a PIDF/LO in the INVITE to be used for routing purposes. In the case a PIDF/LO is missing, other mechanisms such as policy routing rules in the ESRP can be used to forward to the correct PSAP.

7.6.2.2 Description

Interoperability Test Description			
Identifier	ECL_BAS_01		
Test Objective	Verify that the IVS is able to initiate a manual eCall containing MSD and the correct eCall initiation indicator for a manually triggered eCall		
Configuration	IMS		
References	ETSI TS 103 479 [1], ETSI TS 123 167 [2], ETSI TS 123 333 [3], ETSI TS 103 683 [7]		
Applicability	IMS (Le, Mm, Mb) BCF (SIP-1, SIP-2, RTP-1, RTP-2) ESRP (SIP-1, SIP-2, LOST-1, LOST-2) ECRF (LOST-1, LOST-2) PSAP (SIP-1, SIP-2, RTP-1, RTP-2)		
Pre-test conditions	• IP connectivity among all elements of the specific scenario • ECSCF trigger points for emergency call routing • LRF has location information (value) available • BCF trigger points for emergency call routing (to ESRP) • ESRP trigger points for emergency call routing • ESRP configured to query the ECRF • ECRF configured with correct mapping		
Test Sequence	Step	Type	Description
	1	stimulus	IVS initiates a manual eCall
	2	check	Dialog creating INVITE (a) received at BCF
	3	check	Dialog creating INVITE (a) received at ESRP
	4	check	LoST request received at ECRF
	5	check	ESRP emergency call routing based on location
	6	check	Dialog creating INVITE (a) received at PSAP
	7	check	SIP dialog established
	8	verify	Call (a) connected, MSD and location displayed

7.6.2.3 Message Sequence Chart



Annex A (normative): Interoperable Functions Statement (IFS)

A.1 Entities

Table A.1: Entities

Item	Which entity do you support?	Status	Support
1	UE		
2	IMS		
3	UC		
4	LNG (PIF)		
5	LNG (NIF)		
6	LNG (LIF)		
7	BCF		
8	LIS		
9	ESRP		
10	ECRF		
11	FG		
12	BRIDGE		
13	PSAP		

A.2 UE Features

Table A.2: UE features

Item	Feature	ID	Ref	Status	Support
1	Does the UE support SIP?	UE_SIP	[1]		
2	Does the UE support RTP?	UE_RTP	[1]		
3	Does the UE support G.711?	UE_G711	[1]		
4	Does the UE support H.264?	UE_H264	[1]		
5	Does the UE support Real-time Text?	UE_RTT	[1]		
6	Does the UE support PIDF/LO?	UE_PFL	[1]		
7	Does the UE support service URNs?	UE_URN	[1]		
8	Does the UE support AML (HTTP)?	UE_AMLH	[1]		
9	Does the UE support AML (SMS)?	UE_AMLS	[1]		
10	Does the UE support AML (DATA SMS)?	UE_AMLD	[1]		
11	Does the UE support LMPE?	UE_LMPE	[5]		

A.3 IMS Features

Table A.3: IMS features

Item	Feature	ID	Ref	Status	Support
1	Does the IMS support SIP?	IMS_SIP	[1]		
2	Does the IMS support RTP?	IMS_RTP	[1]		
3	Does the IMS support G.711?	IMS_G711	[1]		
4	Does the IMS support H.264?	IMS_H264	[1]		
5	Does the IMS support Real-time Text?	IMS_RTT	[1]		
6	Does the IMS support HELD?	IMS_HELD	[1]		
7	Does the IMS support NG specific SIP Header?	IMS_NGS	[1]		
8	Does the IMS support PIDF/LO?	IMS_PFL	[1]		
9	Does the IMS support service URNs?	IMS_URN	[1]		

A.4 LNG (PIF) Features

Table A.4: PIF features

Item	Feature	ID	Ref	Status	Support
1	Does the PIF support SIP?	PIF_SIP	[1]		
2	Does the PIF support RTP?	PIF_RTP	[1]		
3	Does the PIF support G.711?	PIF_G711	[1]		

A.5 LNG (NIF) Features

Table A.5: NIF features

Item	Feature	ID	Ref	Status	Support
1	Does the NIF support SIP?	NIF_SIP	[1]		
2	Does the NIF support RTP?	NIF_RTP	[1]		
3	Does the NIF support G.711?	NIF_G711	[1]		
4	Does the NIF support NG specific SIP Header?	NIF_NGS	[1]		
5	Does the NIF support service URNs?	NIF_URN	[1]		

A.6 LNG (LIF) Features

Table A.6: LIF features

Item	Feature	ID	Ref	Status	Support
1	Does the LIF support PIDF/LO?	LIF_PFL	[1]		
2	Does the LIF support HELD?	LIF_HELD	[1]		

A.7 BCF Features

Table A.7: BCF features

Item	Feature	ID	Ref	Status	Support
1	Does the BCF support SIP?	BCF_SIP	[1]		
2	Does the BCF support RTP?	BCF_RTP	[1]		
3	Does the BCF support G.711?	BCF_G711	[1]		
4	Does the BCF support H.264?	BCF_H264	[1]		
5	Does the BCF support HELD?	BCF_HELD	[1]		
6	Does the BCF support NG specific SIP Header?	BCF_NGS	[1]		
7	Does the BCF support PIDF/LO?	BCF_PFL	[1]		
8	Does the BCF support service URNs?	BCF_URN	[1]		
9	Does the BCF support Real-time Text?	BCF_RTT	[1]		
10	Does the BCF support emergency call transfer?	BCF_ECT	[1]		

A.8 LIS Features

Table A.8: LIS features

Item	Feature	ID	Ref	Status	Support
1	Does the LIS support HELD?	LIS_HELD	[1]		
2	Does the LIS support SIP SUBSCRIBE/NOTIFY location update?	LIS_SIPLO	[1]		
3	Does the LIS support PIDF/LO?	LIS_PFL	[1]		
4	Does the LIS support AML (HTTP)?	LIS_AMLH	[1]		
5	Does the LIS support AML (SMS)?	LIS_AMLS	[1]		
6	Does the LIS support AML (DATA SMS)?	LIS_AMLD	[1]		

A.9 ESRP Features

Table A.9: ESRP features

Item	Feature	ID	Ref	Status	Support
1	Does the ESRP support SIP?	ESRP_SIP	[1]		
2	Does the ESRP support SIP SUBSCRIBE/NOTIFY location update?	ESRP_SIPLO	[1]		
3	Does the ESRP support SIP SUBSCRIBE/NOTIFY queue events?	ESRP_SIPQU	[1]		
4	Does the ESRP support HELD?	ESRP_HELD	[1]		
5	Does the ESRP support dequeue registration?	ESRP_DEQU	[1]		
6	Does the ESRP support LoST?	ESRP_LOST	[1]		
7	Does the ESRP support policy routing function?	ESRP_PRF	[1]		
8	Does the ESRP support PIDF/LO?	ESRP_PFL	[1]		
9	Does the ESRP support service URNs?	ESRP_URN	[1]		
10	Does the ESRP support LoST iteration/redirection?	ESRP_LRE	[1]		

A.10 ECRF Features

Table A.10: ECRF features

Item	Feature	ID	Ref	Status	Support
1	Does the ECRF support LoST?	ECRF_LOST	[1]		
2	Does the ECRF support PIDF/LO?	ECRF_PFL	[1]		
3	Does the ECRF support service URNs?	ECRF_URN	[1]		
4	Does the ECRF support LoST recursion?	ECRF_LRE	[1]		
5	Does the ECRF support LoST iteration?	ECRF_LIT	[1]		
6	Does the ECRF support location validation?	ECRF_VAL	[1]		

A.11 BRIDGE Features

Table A.11: BRIDGE features

Item	Feature	ID	Ref	Status	Support
1	Does the PSAP emergency call transfer?	BRIDGE_ECT	[1]		

A.12 PSAP Features

Table A.12: PSAP features

Item	Feature	ID	Ref	Status	Support
1	Does the PSAP support SIP?	PSAP_SIP	[1]		
2	Does the PSAP support RTP?	PSAP_RTP	[1]		
3	Does the PSAP support G.711?	PSAP_G711	[1]		
4	Does the PSAP support H.264?	PSAP_H264	[1]		
5	Does the PSAP support Real-time Text?	PSAP_RTT	[1]		
6	Does the PSAP support SIP SUBSCRIBE/NOTIFY location update?	PSAP_SIPLO	[1]		
7	Does the PSAP support SIP SUBSCRIBE/NOTIFY queue events?	PSAP_SIPQU	[1]		
8	Does the PSAP support HELD?	PSAP_HELD	[1]		
9	Does the PSAP support PIDF/LO?	PSAP_PFL	[1]		
10	Does the PSAP support dequeue registration?	PSAP_DEQU	[1]		
11	Does the PSAP support service URNs?	PSAP_URN	[1]		
12	Does the PSAP emergency call transfer?	PSAP_ECT	[1]		
13	Does the PSAP support LMPE?	PSAP_LMPE	[5]		

Annex B (informative): Change history

Date	Version	Information about changes
January 2026	1.0.1	Initial version of the stable draft

History

Version	Date	Status
V1.1.1	March 2026	Publication