NG112 Plugtests 2026

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ETSI

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

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

Siret N° 348 623 562 00017 - NAF 742 C
Association à but non lucratif enregistrée à la
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Contents

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1 Executive Summary

ETSI, in partnership with EENA (the European Emergency Number Association), has organized the seventh NG112 Emergency Communications Plugtests™ event. This event was held in Ljubljana, Slove, from 26th May to 29th May 2026.
The aim of the event was to trial independently and jointly all components of the 112 communication chain based on Next Generation networks. Different topics were addressed, including Location Based Emergency Call Routing, Policy Based Emergency Call Routing, NG eCall and Next Generation Media Types such as Real Time Text (RTT).
Special attention was paid to the complete end-to-end emergency communication chain from originating mobile device, through an IMS and the ESInet to the PSAP.
11 organizations from Europe, and North America, had the opportunity to connect their equipment to the test infrastructure and validate the interoperability and conformity of their market solutions using different scenarios. In addition, 2 organizations participated as observers. In total, 13 organizations were involved in the Plugtests.

The scope of the event included content-rich emergency calling, location- and policy-based routing, secure transport via TLS, end-to-end emergency calling as well as RTT and NG eCall through the ESInet. Participants put their products to the test, gaining valuable insights from experiencing a variety of scenarios.

The results of the tests show that the NG112 technology is mature and that routing NG eCall over the ESInet provides many benefits. With the technical foundation of ETSI TS 103 479 and its accompanying standards, the conditions for procurement and deployment are reached.

2 References

The following base specifications were validated in the Plugtests event.

[i.1] ETSI TS 103 479 (v1.3.1): “Emergency Communications (EMTEL); Core elements for network independent access to emergency services”.
[i.2] ETSI TS 103 698 (v1.2.1): “Emergency Communications (EMTEL); Lightweight Messaging Protocol for Emergency Service Accessibility (LMPE)”
[i.3] RFC 4103: “RTP Payload for Text Conversation”
[i.4] IETF RFC 8147: “Next-Generation Pan-European eCall”
[i.5] CEN/TS 17184:2022: "Intelligent transport systems - eSafety - eCall High level application Protocols (HLAP) using IMS packet switched networks"
[i.6] CEN EN 15722:2020: "Intelligent transport systems - eSafety - eCall Minimum Set of Data”

The following test specifications were used and validated in the Plugtests event.

[i.7] ETSI TS 103 480 (v1.1.1): "Emergency Communications (EMTEL); Interoperability testing of core elements for network independent access to emergency services"
[i.8] ETSI TS 103 650 (v1.2.1): “Emergency Communications (EMTEL); Testing - Conformance test specifications for core elements for network independent access to emergency services (NG112)”
[i.9] ETSI TS 103 683 (v2.1.1): Mobile Standards Group (MSG); Testing; Next Generation eCall High Level Application Protocol (HLAP) Interoperability Testing
[i.10] ETIS TS 103 795-2 (v1.1.1): Core Network and Interoperability Testing (INT); Network Interoperability Test Description for emergency services over VoLTE (3GPP™ Release 15); Part 2: Test Descriptions

[i.11] CEN/TS 17240:2024 (draft): Intelligent transport systems - ESafety - ECall end to end conformance testing for IMS packet switched based systems.

3 Abbreviations

BCF	Border Control Function
ECRF	Emergency Call Routing Function
ESInet	Emergency Services IP Network
ESRP	Emergency Services Routing Proxy
HELD	HTTP-Enabled Location Delivery
HTTP	Hyper-text Transfer Protocol
IMS	IP Multimedia Subsystem
IP	Internet Protocol
IUT	Implementation Under Test
IVS	In-Vehicle System
LbR	Location by Reference
LbV	Location by Value
LIS	Location Information Server
mTLS	Mutual TLS
MSD	Minimum Set of Data
NGCS	Next Generation Core Services
PSAP	Public Safety Answering Point
RFC	Request for Comments
SDP	Session Description Protocol
SIP UA	Session Initiation Protocol User Agent
TLS	Transport Layer Security
TS	Technical Specification
UE	User Equipment

4 Participants

20 participants from 14 companies and organisations of 9 countries, including observers and the technical experts, participated in the 7th NG112 and NGeCall Emergency Communications Plugtests™ event. They were supported by an additional 4-member ETSI team.

Participating organizations in the Plugtests who signed the Rules of Engagement (RoE) and Non-Disclosure Agreement (NDA) are listed below:

Company name		Role
Centro Español de Servicios Telemáticos, S.A. (CESTEL)	ES	Vendor
Frequentis AG	AT	Vendor / Expert
GILAWA Ltd	CY	Vendor
GridGears GmbH	AT	Vendor / Expert
Intersec SA	FR	Vendor
Mobileum Inc	US	Vendor
Omda	SE	Vendor
Oracle Corporation	US	Vendor
Qualcomm Germany GmbH	DE	Vendor
Special Telecommunications Service (STS)	RO	Vendor
Valid8.com, Inc.	US	Vendor
Telekom Slovenije d.d.	SI	Host

Observers:

Company name		Role
GSM Association	GB	Observer
European Emergency Number Association (EENA)	BE	Observer

Table 1: List of organizations

5 Scope of the event

5.1 Objectives

The main objectives of this event were to validate the market interoperability of different solutions for end-to-end emergency services communications, as defined in TS 103 479 (Version 1.3.1) and TS 103 480 (Version 1.1.1). Additionally, the event aimed to validate the interoperability of various next-generation (NG) eCall solutions through end-to-end tests based on TS 103 683 (Version 2.1.1) or TS 103 795-2 (Version 1.1.1).

It also provided a valuable opportunity for developers from different companies to collaborate, test their implementations, and ensure seamless product interoperability. Finally, the event served to evaluate how closely several implementations conformed to established interface specifications, including various RFCs and Technical Specifications (TS).

To validate this interoperability, specific features and services were integrated into the testing scenarios, supported by the test infrastructure (covering IMS and VoIP originating service provider) shown in Figure 1. These scenarios primarily evaluated location conveyance by both value and reference, alongside location- and policy-based routing.

Testing also focused on secure transport using certificates and mutual authentication, as well as core end-to-end emergency communications. Furthermore, the participants evaluated Real-Time Text (RTT) capabilities, emergency callbacks featuring both audio and RTT, and the routing of NG eCall emergency as well as test calls over the Emergency Services IP Network (ESInet).

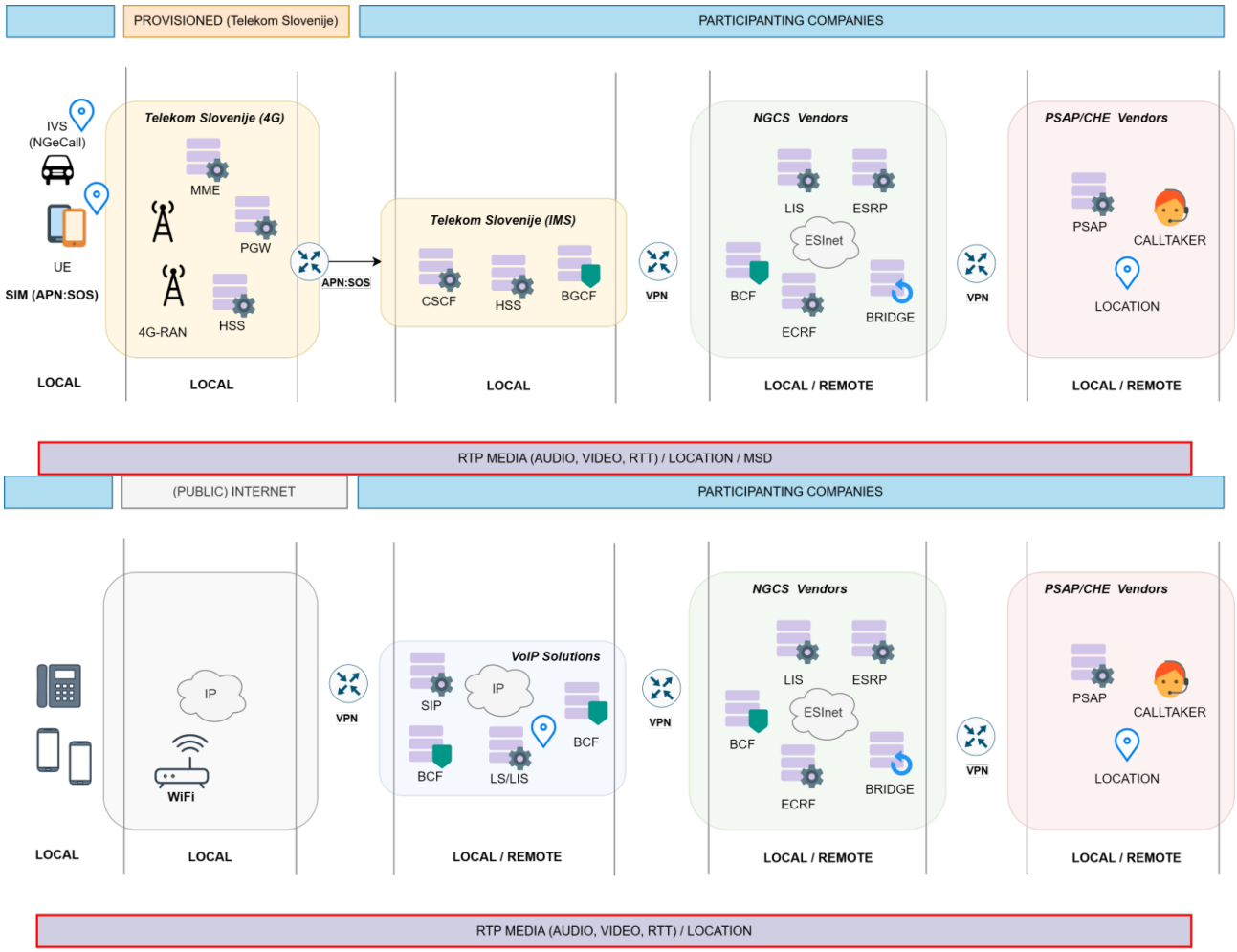

Figure 1: Test infrastructure

5.2 Description

During this event, testing was divided into two main domains: Next Generation Core Services (NGCS) and Next Generation eCall (NG eCall). Within these domains, participants executed four distinct groups of tests covering various scenarios and test cases, such as location-based call routing, accessibility features, and diverse originating networks.

The evaluation was structured into the following four focus areas:

- **Generic NGCS Testing**: This group established baseline NGCS functionality, focusing on basic media transport verification, routing, Location by Value (LbV), and Location by Reference (LbR).
- **Media Communications**: These scenarios validated advanced end-to-end multi-media capabilities, including audio, video, and Real-Time Text (RTT) transmissions.
- **Security**: This focus area evaluated secure transport mechanisms, testing mutual authentication behaviours with both valid and invalid certificates.
- **eCall Functionality**: These tests specifically targeted NG eCall-related features, verifying Minimum Set of Data (MSD) transmissions and specialized NG eCall call scenarios.

5.3 NG112 Emergency Communication

5.3.1 General

As in the previous editions of the NG112 Communications Plugtest event, the NG112 components and their interfaces, as shown in Figure 2, of different vendors were tested working together. Scenarios and vendors combinations were planned, and tests were executed.

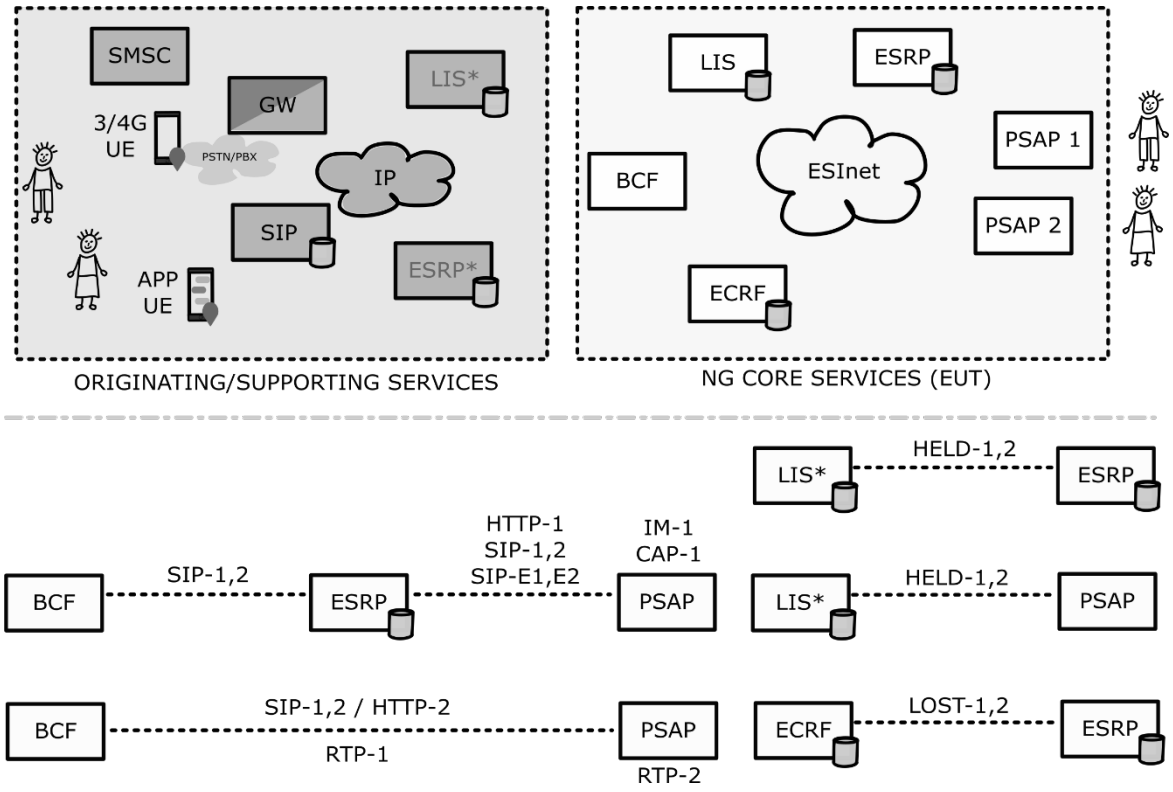

Figure 2: Functional elements and interfaces (refer to ETSI TS 103 479 [i.1])

The tests were executing using basic groups of scenarios:

Mobile Network Attachment Procedures
UE Initialization, Network Authentication, Attach Acceptance, Location Update, Emergency Number Download (MME; download local emergency numbers to the UE)

IMS Procedures
IMS Registration, Capability Negotiation (whether the IMS is capable of handling RTT sessions), Emergency Call Setup, Media Capability Exchange, Callback

IMS or VoIP / ESInet Interconnect
E-CSCF Routing, LRF/LIS Location Retrieval

ESInet / NGCS / PSAP
Basic Emergency Communication, Certificates, STIR/SHAKEN, Routing, Mapping, Media: Audio/Video/RTT

5.3.2 Test Data

Testing several scenarios required creating simple polygons that define PSAP areas (or service boundaries). Provided location information represents 48 locations (4 per service area): ALICE01, BOB01, CAROL01, ALICE02, ... with ALICExx and BOBxx as point, CAROLxx as circle location type, and DAVEXX as civic location, see Figure 3.

Location Information Services (LIS) were preconfigured with locations (pin icons in Figure 3). For instance, sip:alice-01@plugtests.app resolves to a location within the top-left PSAP's region (area shown in Figure 3) and therefore calls originated by alice-01 shall route to the PSAP configured for that region.

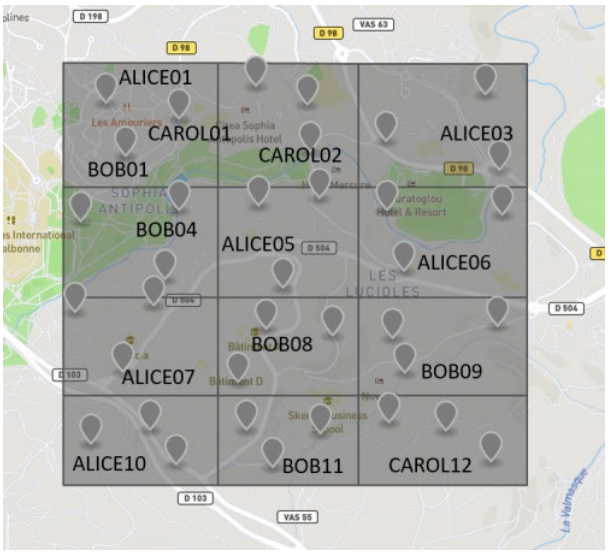

Figure 3: Pre-configured test locations and SIP identities (source: geojson.io)

5.3.3 Test Tools

To simplify remote testing and to be able to perform simple tests without an originating device as part of a test scenario, a tool (RoboDialer) was used to trigger emergency communication. This tool can be operated via the command line with curl or via the web browser or *links* (see Figure 4). To trigger a call, the following attributes are required:

- **Caller = a2** (sip:alice-02@plugtests.app); used to define the location used for testing
- **Service = 112|911**; used to set proper service URNs
- **BCF = vendor_1**; used to select the BCF (can be omitted)
- **ESRP = vendor_2**; used to select an ESRP (indicated in a pre-loaded route-set)

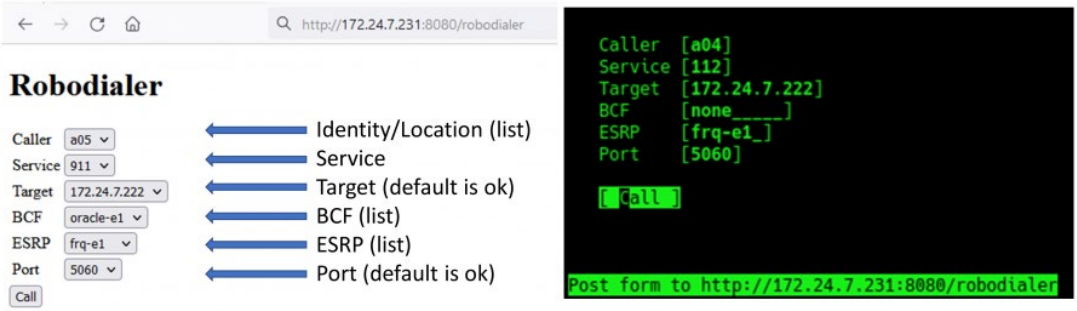

Figure 4: RoboDialer (Web UI and CL Tool)

5.3.4 Configurations

Different test configurations were used to test interoperability among different service instances from different vendors. The basic configuration, as briefly shown in Figure 6 below, supports Next Generation Core Service (NGCS) testing scenarios, multimedia communication (audio, video and text) and location delivery using identities (sip and tel URIs). Most test calls were either placed from Android or Apple devices or in case of NG eCall originated from an IVS. The IMS forwarded the calls to the border control function (BCF).

The BCF used static routing to forward calls to the ESRP used in specific scenarios or combinations. To route to the correct PSAP based on the location received or requested at the LIS via HELD, the ESRP was requesting routing information at the configured ECRF and finally forwarded the emergency call to the PSAP serving the location at which the caller is located.

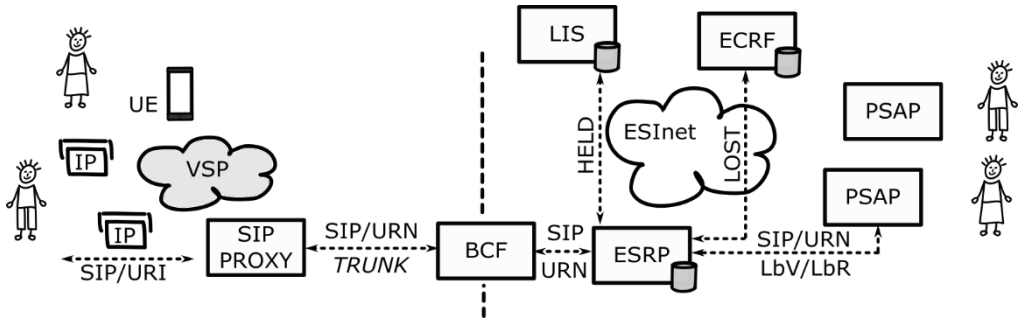

Figure 5: General NGCS Configuration

5.3.5 Emergency Communications Test Scenarios

5.3.5.1 Call Started with Voice/Video and RTT Simultaneously

Scenarios of this kind are designed to assess the ability of the network and PSAP to accept various media types (including 'Total Conversation') from the very first signalling packet onwards, without the need for any changes or modifications to be made during the call. A brief overview is provided below; further details on the individual scenarios can be found in TS 103 480 (Version 1.1.1).

Test-Setup & Precondition

- **Originating UE**: Configured to initiate a mixed-media emergency session containing both voice and RTT immediately upon pressing the emergency button.
- **NGCS / ESInet & PSAP**: Configured to support multi-stream SDP profiles from call arrival.

Sequence of Steps & Signalling (simplified)

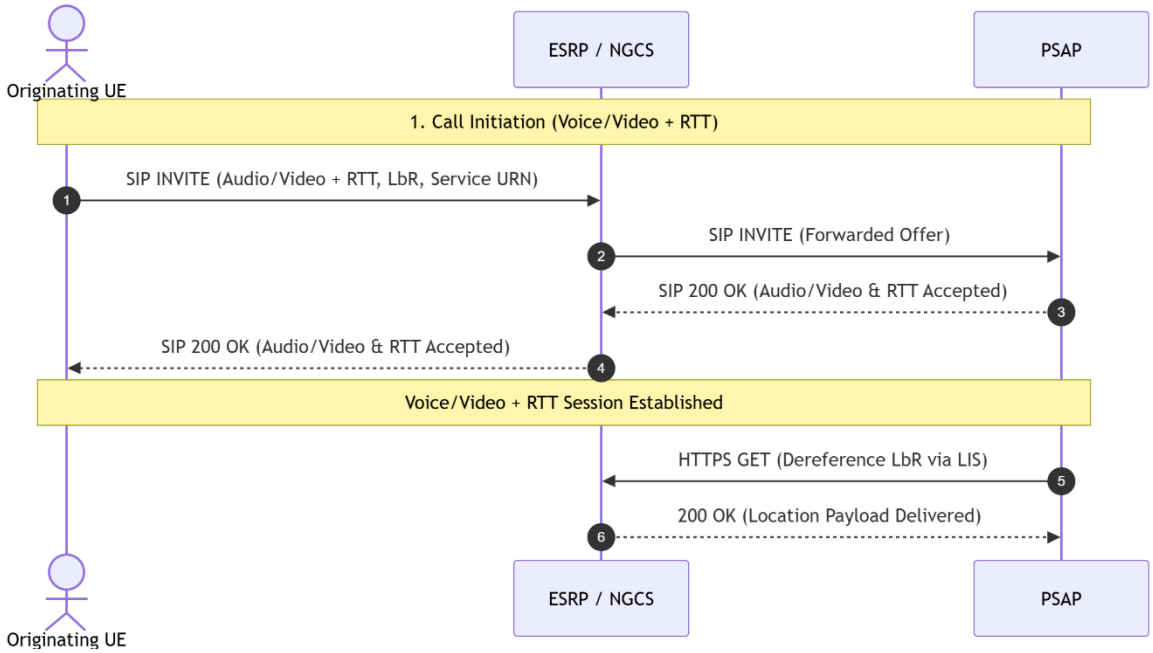

Figure 6: Call Started with Voice/Video and RTT Simultaneously

Expected Pass Criteria

- The call routes properly to the correct jurisdiction based on the LbR data.
- The PSAP dereferences the LbR data successfully upon call arrival.
- The PSAP indicates call data, the call is manually accepted, and any offered media is active

- No signalling fallback occurs (e.g., the PSAP does not reject the text stream with port 0 in the SDP answer)

5.3.5.2 Voice/Video Call with RTT Added Mid-Call

Scenarios of this kind are designed to assess the ability of the network and PSAP to dynamically renegotiate session parameters (SIP re-INVITE) to introduce a Real-Time Text (RTT) media stream into an active emergency voice/video call without dropping the session. A brief overview is provided below; further details on the individual scenarios can be found in TS 103 480 (Version 1.1.1).

Test Setup & Preconditions

- **Originating UE**: Configured to initiate a voice/video call with a Service URN (e.g., `urn:service:sos`). It includes a Geolocation header containing a Location-by-Reference URL (HTTPS URI pointing to a Location Information Server / LIS).
- **NGCS / ESInet**: Emergency Services Routing Proxy (ESRP) and Emergency Call Routing Function (ECRF) are operational.
- **PSAP**: An IP-capable PSAP terminal ready to accept voice/video and dynamically handle RTT renegotiation.

Sequence of Steps & Signalling (simplified)

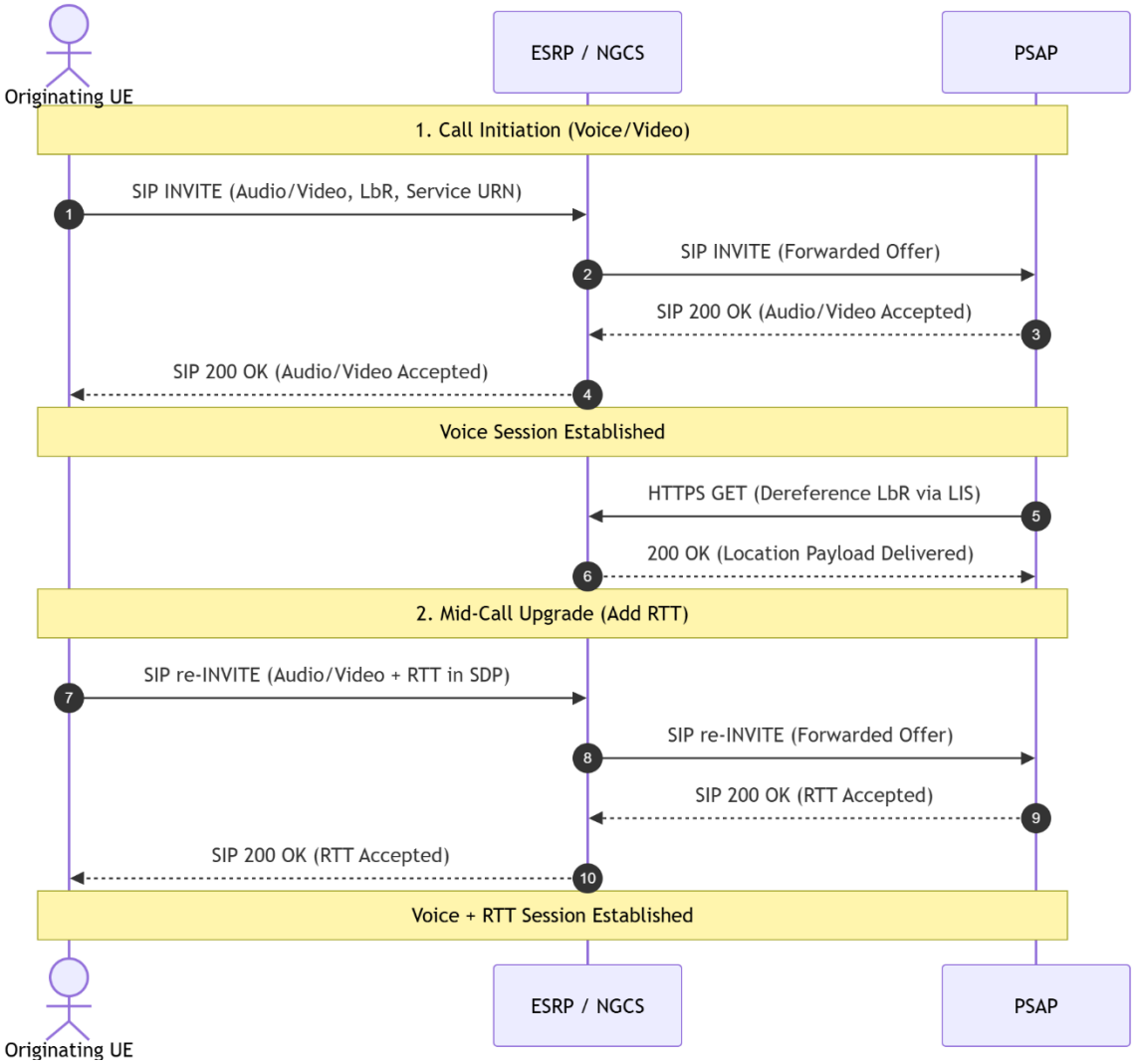

Figure 7: Voice/Video Call with RTT Added Mid-Call

Expected Pass Criteria

- The call routes properly to the correct jurisdiction based on the LbR data.
- The PSAP dereferences the LbR data successfully upon call arrival.
- Audio/video stream remains completely uninterrupted during the re-INVITE handshake.
- RTT stream activates successfully; characters typed on the UE appear instantly at the PSAP and vice-versa.

5.3.6 Emergency Callback Test Scenarios

5.3.6.1 Callback Started with Voice/Video and RTT Simultaneously

Scenarios of this kind are designed to assess the ability of the network and the destination User Equipment (UE) to accept different media types (including "Total Conversation") from the first signalling packet of an emergency callback without needing an in-call modification. A brief overview is provided below; further details on the individual scenarios can be found in TS 103 480 (Version 1.1.1).

Test Setup & Preconditions

- **PSAP**: Configured to initiate a mixed-media emergency callback session containing both voice/video and RTT simultaneously targeting the destination device's MSISDN (SIP URI).
- **SIP Headers**: INVITE must maintain the Resource-Priority and Priority header status.
- **NGCS / ESInet & Core Network**: Operational and configured to recognize the priority of the callback and route standard multi-stream SDP profiles to the subscriber.
- **Destination UE**: Power on, registered to the network, and capable of receiving multi-stream (Voice/Video + RTT) inbound sessions.

Sequence of Steps & Signalling (simplified)

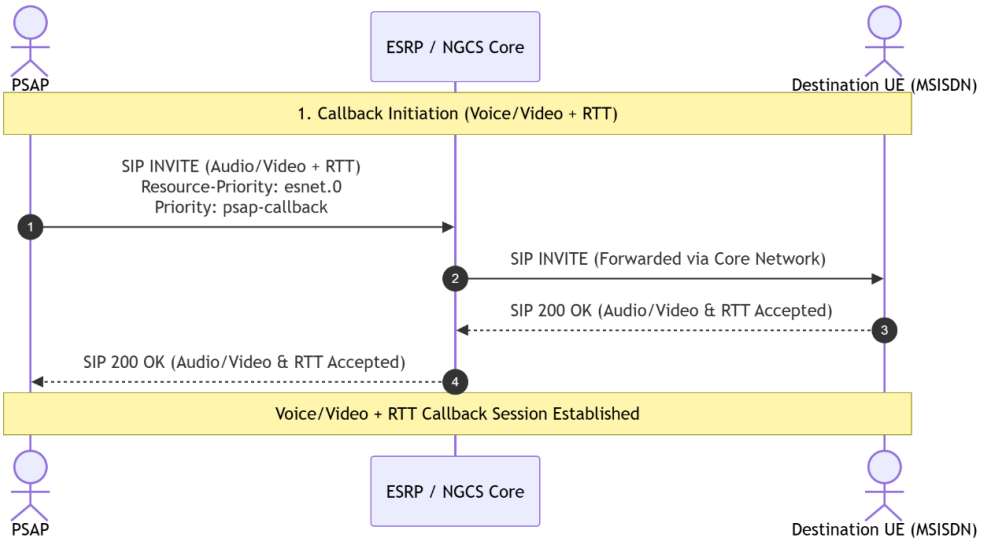

Figure 8: Callback Started with Voice/Video and RTT Simultaneously

Expected Pass Criteria

- The callback routes successfully through the network to the specific destination UE using the provided SIP URI (P-A-I received with emergency communications).
- The network honours the Resource-Priority header, bypassing normal subscriber restrictions (e.g., Do Not Disturb, blocked unknown numbers).

- The destination UE indicates an incoming emergency callback, the call is manually accepted, and both the voice/video and RTT media streams are active immediately.
- No signalling fallback occurs (e.g., the UE does not reject the text stream with port 0 in the SDP answer).

5.3.6.2 Voice/Video Callback with RTT Added Mid-Call

Scenarios of this kind are designed to assess the ability of th e network's and the device's capability to dynamically renegotiate session parameters (SIP re-INVITE) to introduce a Real-Time Text (RTT) media stream into an active emergency callback voice session. A brief overview is provided below; further details on the individual scenarios can be found in TS 103 480 (Version 1.1.1).

Test Setup & Preconditions

- **PSAP**: Configured to initiate a mixed-media emergency callback targeting the destination device's MSISDN (SIP URI). It must be capable of dynamically initiating an RTT renegotiation mid-call.
- **SIP Headers**: INVITE must maintain the Resource-Priority and Priority header status.
- **NGCS / ESInet & Core Network**: Operational and configured to handle high-priority and mid-call session updates.
- **Destination UE**: Registered to the network, ready to accept a voice/video callback, and capable of handling mid-call upgrades to add RTT.

Sequence of Steps & Signalling (simplified)

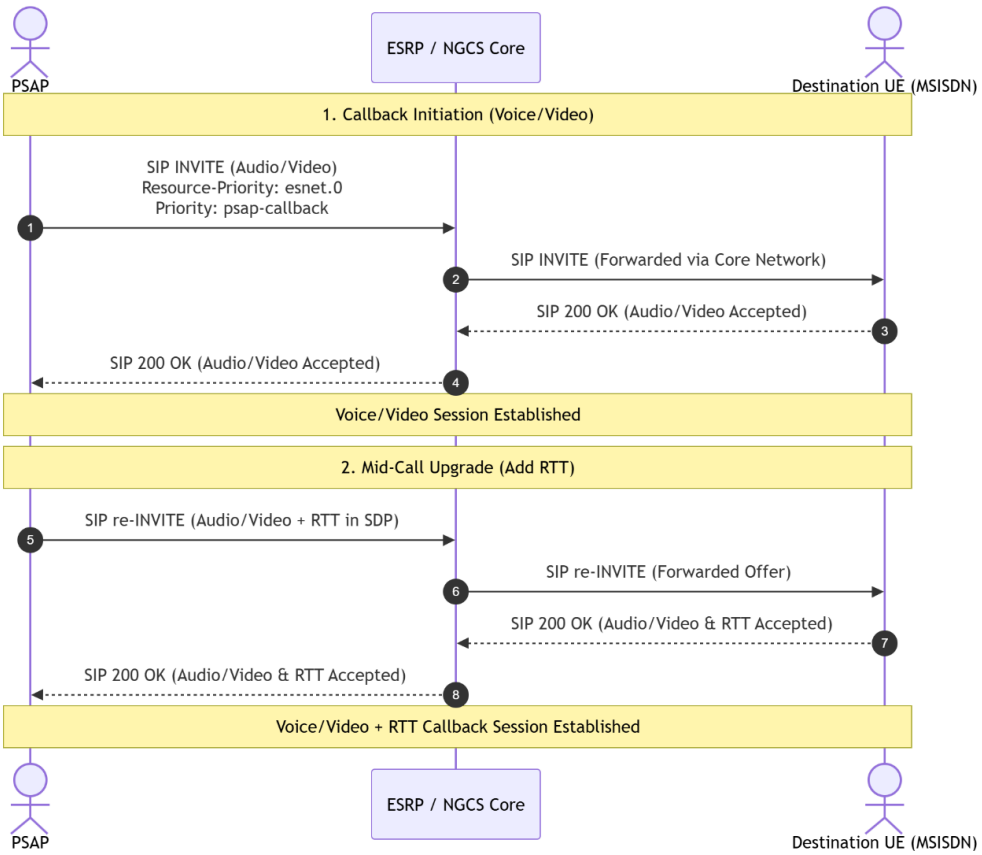

Figure 9: Voice/Video Callback with RTT Added Mid-Call

Expected Pass Criteria

- The callback routes successfully through the network to the specific destination UE using the provided SIP URI (P-A-I received with emergency communications).
- The network honours the Resource-Priority header, bypassing normal subscriber restrictions (e.g., Do Not Disturb, blocked unknown numbers).

- The destination UE indicates an incoming emergency callback, the call is manually accepted, and voice/video media streams are active immediately.
- The RTT stream activates successfully mid-call; characters typed on either the PSAP or the UE appear instantly on the opposing screen.
- No signalling fallback occurs (e.g., the UE does not reject the text stream with port 0 in the SDP answer).

5.3.7 NG eCall Test Scenarios

5.3.7.1 NG eCall Emergency Communications & Callback

This scenario evaluates the end-to-end progression of an active Next Generation eCall. It validates the capability of the network and PSAP to accept the initial eCall with an embedded MSD, request an updated MSD mid-call, and successfully initiate acallback to the IVS after call disconnect, including a further MSD request during the callback session.

Test Setup & Preconditions

Originating IVS: Configured to initiate an NG eCall using the appropriate emergency URN (e.g. `urn:service:sos.ecall.manual` or `automatic`), including the initial MSD payload in the SIP INVITE.

- **IMS / NGCS & PSAP**: Configured to receive multipart MIME bodies (SDP + MSD), establish the audio session, and trigger SIP INFO messages for subsequent MSD requests.

Sequence of Steps & Signalling (simplified)

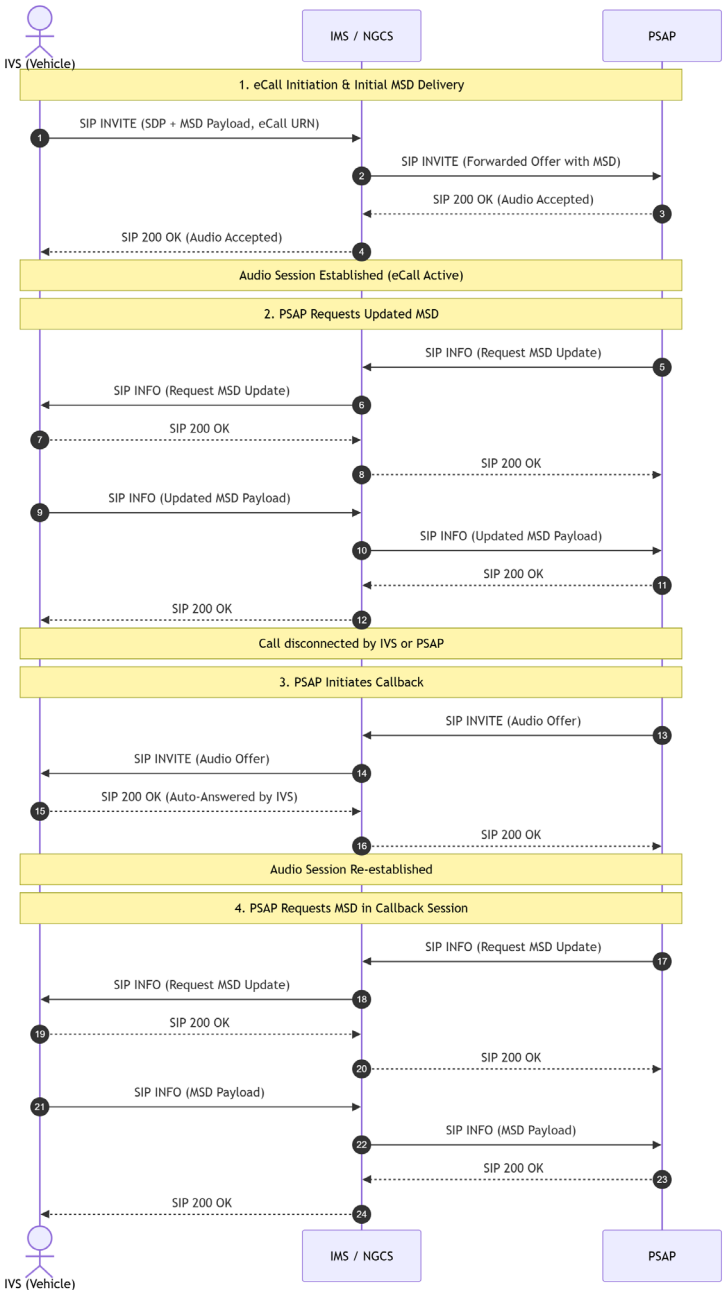

Figure 10: NG eCall Emergency Communications & Callback message flow

Expected Pass Criteria

- The eCall routes properly to the correct PSAP.
- PSAP successfully extracts and parses the initial MSD.
- SIP INFO requests for MSD updates are successfully acknowledged and the new MSD is received without interrupting the audio.
- The emergency callback successfully reaches the IVS, automatically answers, and allows for another successful MSD update.

5.3.7.2 NG eCall Test Call

This scenario validates the network's and PSAP's capability to correctly identify and process a test eCall. The procedure mirrors a real eCall but uses a distinct service URN to ensure test calls use a regular IMS call and do not interfere with live emergency operations.

Test Setup & Preconditions

- **Originating IVS**: Configured to initiate a test eCall using the test URN (`urn:service:test.sos.ecall`), including the test MSD payload.
- **IMS / NGCS & PSAP**: Configured to route and identify the call as a test procedure, handling MSD updates and callbacks appropriately.

Sequence of Steps & Signalling (simplified)

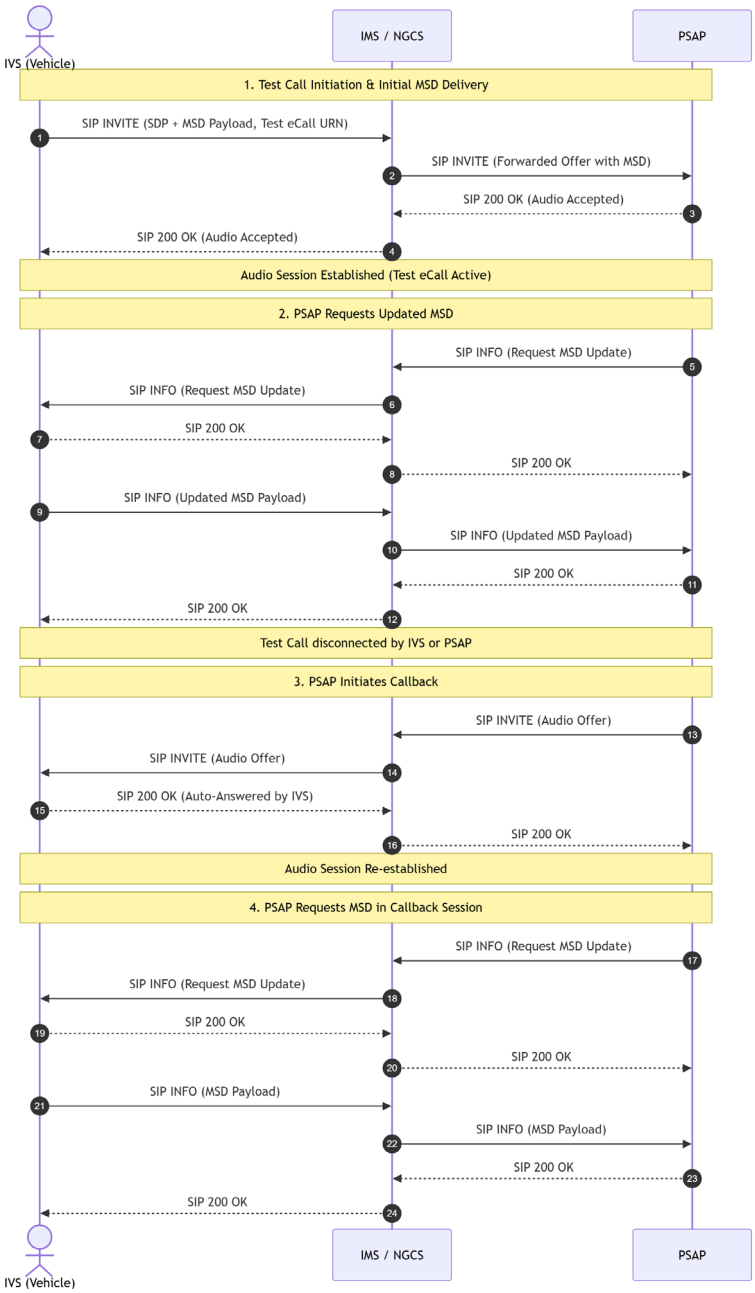

Figure 11: NG eCall test call message flow

Expected Pass Criteria

- The test eCall routes properly to the designated test PSAP or test endpoint.
- PSAP/endpoint successfully identifies the call as a test and extracts the initial MSD.
- SIP INFO requests for MSD updates function correctly.
- The callback reaches the IVS, is auto-answered, and MSD can be requested again.

6 Achieved Results

6.1 NG112 Interoperability Testing Results

6.2.1 Observations

6.2.1.1 General

The test reporting tool is a great help when it comes to planning test sessions and creating possible test combinations. Nevertheless, Test Groups shall be adapted accordingly to match the specific objectives of the Plugtests event to avoid a high number of N/A (not applicable) test results.

6.2.1.2 Emergency Communications Observations

Initial testing highlighted critical IMS routing configuration issues, necessitating refined setup procedures and isolated VoIP verification. Further findings are:

- Basic features defined in TS 103 479 (Version 1.3.1) and end-to-end 3GPP audio codecs were successfully validated using test cases as defined in TS 103 480 (Version 1.1.1)
- Real-Time Text handling with Apple and Android devices worked well, and PSAP callbacks with reverse audio/RTT routing were verified.
- Future events require precise 4/5G/IMS setup definitions, preliminary isolated VoIP checks, and a default PSAP for every test case.
- Test infrastructure can rely entirely on self-signed certificates for HTTPS, TLS, and mTLS instead of official Certificate Authorities.
- Next iterations should invite more mobile device and RTT VoIP vendors, while introducing advanced IMS roaming, media upgrade/downgrade, and RTT redundancy encoding test cases.
- End-to-end support of 3GPP audio codecs shall be considered in future releases of TS 103 479.

Future events should prioritize diverse vendor participation, expand RTT and roaming test scenarios, and leverage self-signed certificates, as core TS 103 479 standards and Android/Apple RTT interoperability were successfully verified.

6.2.1.3 NG eCall Observations

NG eCall originating devices were successfully tested using the NG112 Emergency Services IP Network (ESInet) infrastructure. During these scenarios, all NG eCall calls were dynamically routed by the IMS to the NG112 Border Control Function (BCF), which subsequently forwarded the communications to an ESRP to route to an appropriate PSAP. The testing demonstrated seamless integration and complete compatibility between the core ESInet infrastructure and NG eCall solutions.

- To fully enable ESInet policy or location-based routing for NG eCall, it is recommended to extend current NG eCall test specifications (i.e. including specific test scenarios for a combined test environment).
- Future versions of Technical Specifications should explicitly require the simultaneous inclusion of both the Minimum Set of Data (MSD) and Presence Information Data Format Location Object (PIDF-LO) data during initial NG eCall setup.

6.2.2 Statistics

The high number of N/A tests is due to many individual security tests (TLS, mTLS) that were not the focus of this event but were listed in the test reporting tool for the sake of simplicity. In addition, only one mobile device with video support was included, meaning that all other combinations with mobile devices were unable to test video and TC.

The overall test results, considering configurations as introduced in 5.3.4, as illustrated in the below figure 9.

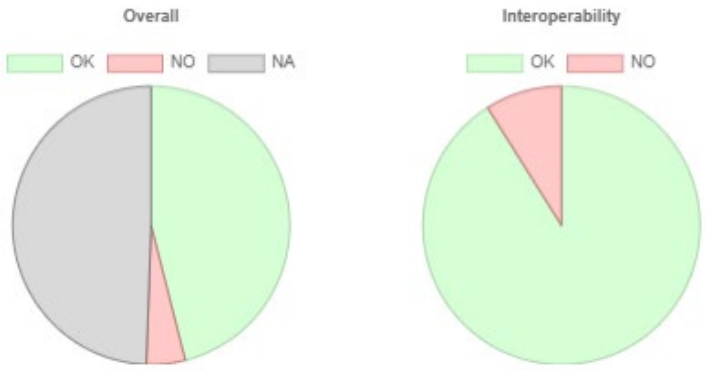

Figure 12: Overall test results diagram

Table 2: Overall test results

Test Group	Total TC count	Run TC count		Not executed
		Passed	Failed	
Overall test results	265	134 (50.6%)		131 (49.4%)
		122 (91.0%)	12 (9.0%)	

Table 3: Group Results

Test Group		Total TC count	Run TC count		Not executed
			Passed	Failed	
Core tests					
Core VoIP tests		47	11 (23.4%)		36 (76.6%)
			11 (100.0%)	0 (0.0%)	
Core IMS tests		40	8 (20.0%)		32 (80.0%)
			7 (87.5%)	1 (12.5%)	
End-2-end tests					
VoIP	General	6	0 (0.0%)		6 (100.0%)
			0 (0.0%)	0 (0.0%)	
	Audio	8	6 (75.0%)		2 (25.0%)
			6 (100.0%)	0 (0.0%)	
	RTT	6	4 (66.7%)		2 (33.3%)
			4 (100.0%)	0 (0.0%)	
IMS	General	33	23 (69.7%)		10 (30.3%)
			21 (91.3%)	2 (8.7%)	
	Audio	49	40 (81.6%)		9 (18.4%)
			38 (95.0%)	2 (5.0%)	
	Video	9	4 (44.4%)		5 (55.6%)
			1 (25.0%)	3 (75.0%)	
	RTT	31	20 (64.5%)		11 (35.5%)
			17 (85.0%)	3 (15.0%)	
	NG eCall	15	9 (60.0%)		6 (40.0%)

			8 (88.9%)	1 (11.1%)	
	mTLS	21	9 (42.9%)		12 (57.1%)
			9 (100.0%)	0 (0.0%)	

6.2.3 Recommendations

- TS 103 479 shall specify error case handling (e.g. rerouting on ESRP when PSAP is not reachable)
- TS 103 479 shall specify the use of Identity Header and explain the STIR/SHAKEN mechanism
- TS 103 480 shall specify callback scenarios and usage of certificates in callback scenarios

History

V0.1	03.07.2026		First draft
V0.2	16.07.2026		Second draft
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