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TECHNICAL SPECIFICATION

Emergency Communications (EMTEL); Accessibility and interoperability of emergency communications and for the answering of emergency communications by the public safety answering points (PSAPs) (including to the single European Emergency number 112)

Reference

DTS/EMTEL-00068

Keywords

accessibility, emergency communications, language, location information, PSAP, real-time text, relay services, roaming, sign language, SIP, total conversation

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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

Accessible emergency communications is needed to provide equal opportunities for all persons. The present document specifies technical, and accessibility means to provide accessible emergency communications. The communications environment is based on packet switched communications technologies, and ETSI TS 103 479 [3] specifies the details for emergency communications. The present document requires a set of functions to ensure access to other media than voice, namely real time text and video, additional functions such as text messaging and access to supporting services are also specified.

When voice, video and real time text are provided together, the communication is called total conversation and is regarded to make communication more accessible than voice communication.

Video is commonly used for sign language communication. The provision of total conversation in emergency communications enables rapid and fluent emergency communications for sign language users, when competence in the sign language favoured by the user in emergency can be provided. In many cases, interpreters need to be invoked in the communication in a three-party fashion. The present document tells which technical solutions are to be implemented to support such interaction.

Technical details are provided for SIP based technologies, in particular communication services known as IMS and SIP based VoIP. Since the intention is that users shall be able to request emergency assistance through efficient emergency communications anywhere in Europe, it is essential that communication interoperability is established for travelling users. That requires few and well specified interfaces for the communication. Openings for emergency apps and other technologies are briefly mentioned. The present document describes these interfaces, which are consistent with the default interoperability solutions prescribed in ETSI EN 301 549 [1], clause 6, enhancing them with the specific requirements needed for accessible emergency communications.

The present document serves as a basis for a future harmonised standard on the same topic.

General accessibility aspects are presented in ETSI EN 301 549 [1].

Introduction

When communication is made accessible for persons they are provided with accessible complements in communications. This may be in the communication media, where real time text and video are accessible complements for voice in various situations. Larger fonts and a spoken user interface are also used as complements to regular visual user interfaces. The emergency communications related requirements on accessible electronic communications are specified in the present document, while the general accessibility requirements on emergency communications are specified in ETSI EN 301 549 [1].

The present document presents in clause 4 the operational profile concept for expressing a scope for the requirements. In clause 5 the functional accessibility and interoperability requirements, usually on the whole chain involved in an emergency communication. Clause 6 provides a symbolic technical architecture with the division in the components involved in emergency communications from user terminal to PSAP.

Clauses 7-10 shows how these different parts of the emergency communications chain fulfil their requirements when providing accessible and interoperable emergency communications, referring to the functional requirements of clause 5 by a set of labels for requirements established in clause 5, and cross referenced in Annex D.

The division between clauses 7-10 is:

- Clause 7: user equipment requirements.
- Clause 8: originating service requirements.
- Clause 9: emergency communications network and PSAP.
- Clause 10: supporting services.

1 Scope

The present document specifies interoperable and accessible emergency communications, which incorporates voice, video, and real time text. Involvement of originating devices, originating service provider, packet switched emergency communications infrastructure, PSAPs, and supporting services in the emergency communications chain is specified, and the use of the different media.

The present document also addresses technical aspects of interoperability between communication services and emergency communication networks, and the interoperability and functionality required to be able to route emergency communications to the most appropriate PSAP.

Focus is on SIP and IMS technologies for originating services and SIP technology for the emergency communications systems, while also other technologies are briefly touched.

For conversion of modality between sign language and spoken language, the possibility to invoke sign language interpreters in the communication is specified.

The emergency communications related requirements on accessible electronic communications are specified in the present document, while the general accessibility requirements on emergency communications are specified in ETSI EN 301 549 [1].

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.

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

NOTE: While any hyperlinks included in this clause were valid at the time of publication, ETSI cannot guarantee their long term validity.

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

[1] ETSI/CEN/CENELEC EN 301 549 (V3.2.1): "Accessibility requirements for ICT products and services".

[2] ETSI ES 202 975 (V2.1.1): "Human Factors (HF); Requirements for relay services".

[3] ETSI TS 103 479 (V1.2.1): "Emergency Communications (EMTEL); Core elements for network independent access to emergency services".

[4] ETSI TS 103 698 (V1.1.1): "Emergency Communications (EMTEL); Lightweight Messaging Protocol for Emergency Service Accessibility (LMPE)".

[5] ETSI TS 122 101 (V18.6.0): "Universal Mobile Telecommunications System (UMTS); LTE; 5G; Service aspects; Service principles (3GPP TS 22.101)".

[6] ETSI TS 122 173 (V18.0.1): "Digital cellular telecommunications system (Phase 2+) (GSM); Universal Mobile Telecommunications System (UMTS); LTE; IP Multimedia Core Network Subsystem (IMS) Multimedia Telephony Service and supplementary services; Stage 1 (3GPP TS 22.173)".

[7] ETSI TS 123 167 (V17.2.0): "Universal Mobile Telecommunications System (UMTS); LTE; IP Multimedia Subsystem (IMS) emergency sessions (3GPP TS 23.167)".

[8] ETSI TS 124 229 (V18.5.0): "Digital cellular telecommunications system (Phase 2+) (GSM); Universal Mobile Telecommunications System (UMTS); LTE; 5G; IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3 (3GPP TS 24.229)".

[9] ETSI TS 126 114 (V18.6.0): "Universal Mobile Telecommunications System (UMTS); LTE; 5G; IP Multimedia Subsystem (IMS); Multimedia telephony; Media handling and interaction (3GPP TS 26.114)".

[10] IETF RFC 3261 (2002): "Session Initiation Protocol (SIP)", Rosenberg J., et al.

[11] IETF RFC 3550 (2003): "RTP: A Transport Protocol for Real-Time Applications", H. Schulzrinne et al.

[12] IETF RFC 3840 (2004): "Indicating User Agent Capabilities in the Session Initiation Protocol (SIP)", Rosenberg J., et al.

[13] IETF RFC 3841 (2004): "Caller Preferences for the Session Initiation Protocol (SIP)", Rosenberg J., et al.

[14] IETF RFC 4579 (2006): "Session Initiation Protocol (SIP) Call Control - Conferencing for User Agents", Johnston A., Levin O.

[15] IETF RFC 7852 (2016): "Additional Data Related to an Emergency Call", Gellens R., Tschofenig H., Rosen B., Marschall R., Winterbottom J.

[16] IETF RFC 8373 (2018): "Negotiating Human Language in Real-Time Communications", Gellens R.

[17] IETF RFC 8866 (2021): "SDP Session Description Protocol", Began A., et al.

[18] IETF RFC 7090 (2014): "Public Safety Answering Point (PSAP) Callback", Schulzrinne H., et al.

2.2 Informative 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.

NOTE: While any hyperlinks included in this clause were valid at the time of publication, ETSI cannot guarantee their long term validity.

The following referenced documents are not necessary for the application of the present document but they assist the user with regard to a particular subject area.

[i.1] ETSI TS 101 470 (V1.1.1): "Emergency Communications (EMTEL); Total Conversation Access to Emergency Services".

[i.2] ETSI TS 122 228 (V18.0.1): "Digital cellular telecommunications system (Phase 2+) (GSM); Universal Mobile Telecommunications System (UMTS); LTE; Service requirements for the Internet Protocol (IP) multimedia core network subsystem (IMS); Stage 1 (3GPP TS 22.228)".

[i.3] IETF RFC 4596 (2006): "Guidelines for Usage of the Session Initiation Protocol (SIP) Caller Preferences Extension", Rosenberg J., Kyzivat P.

[i.4] ETSI TS 103 478 (V1.2.1): "Emergency Communications (EMTEL); Pan-European Mobile Emergency Application".

[i.5] ETSI TS 103 755 (V1.1.1): "Emergency Communications (EMTEL); PEMEA ESInet Shared Services".

[i.6] ETSI TS 103 871 (V1.1.1): "Emergency Communications (EMTEL); PEMEA Real-Time Text Extension".

[i.7] ETSI TS 103 872 (V1.1.1): "Emergency Communications (EMTEL); PEMEA Service Discovery Extension".

[i.8] ETSI TS 103 945 (V1.1.1): "Emergency Communications (EMTEL); PEMEA Audio Video Extension".

[i.9] ISO 9241-11:2018: "Ergonomics of human-system interaction; Part 11: Usability: Definitions and concepts".

[i.10] Directive (EU) 2019/882 of the European Parliament and of the Council of 17 April 2019 on the accessibility requirements for products and services.

3 Definition of terms, symbols and abbreviations

3.1 Terms

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

accessibility: extent to which products, systems, services, environments, and facilities can be used by people from a population with the widest range of user needs, characteristics, and capabilities, to achieve identified goals in identified contexts of use (from ISO 9241-11:2018 [i.9])

NOTE 1: Context of use includes direct use or use supported by assistive technologies.

NOTE 2: The context in which the ICT is used may affect its overall accessibility. This context could include other products and services with which the ICT may interact.

assistive technology any item, piece of equipment, service or product system including software that is used to increase, maintain, substitute, or improve functional capabilities of persons with disabilities or for, alleviation and compensation of impairments, activity limitations or participation restrictions

emergency communication: communication by means of interpersonal communications services between an end-user and the PSAP with the goal to request and receive emergency relief from emergency services

emergency communications system: ESInet and PSAPs together, including technology for both emergency communication handling and additional functions such as emergency communication distribution, emergency communication recording, logging, and connection to the emergency communication queue

Emergency Services IP network (ESInet): Internet Protocol (IP) based communications network dedicated for emergency communications for public safety use

NOTE: An ESInet is a managed IP network that is used for emergency services communications. It provides the IP transport infrastructure upon which independent application platforms and core services can be deployed. ESInets may be interconnected at local, regional, state, federal, national, and international levels to form an IP-based network. The term ESInet designates the network, not the services that are conveyed on the network.

interpersonal communications service: service normally provided for remuneration that enables direct interpersonal and interactive exchange of information via electronic communications networks between a finite number of persons, whereby the persons initiating or participating in the communication determine its recipient(s) and does not include services which enable interpersonal and interactive communication merely as a minor ancillary feature that is intrinsically linked to another service

most appropriate PSAP: PSAP established by responsible authorities to cover emergency communications from a certain area or for emergency communications of a certain type

Real Time Text (RTT): form of text conversation in point-to-point situations or in multipoint conferencing where the text being entered is sent in such a way that the communication is perceived by the user as being continuous on a character-by-character basis

total conversation: multimedia real time conversation service that provides bidirectional symmetric real time transfer of motion video, real time text and voice between users in two or more locations

3.2 Symbols

Void.

3.3 Abbreviations

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

3GPP	Third Generation Partnership Project
AEC	Accessible Emergency Communication
ASR	Automatic Speech Recognition
EAA	European Accessibility Act
ESInet	Emergency Services IP network
GTT	Global Text Telephony = RTT
GTT-IP	Global Text Telephony - Internet Protocols = RTT
ICT	Information and Communications Technology
IMS	IP-Multimedia Subsystem
LoST	Location to Service Translation
MTSI	Multimedia Telephony Service for IMS
PEMEA	Pan-European Mobile Emergency Application
PLMN	Public Land Mobile Network
PSAP	Public Safety Answering Point
RTP	Real Time Protocol
RTT	Real Time Text
SIP	Session Initiation Protocol
UE	User Equipment

4 Operational profile

The technical requirements of the present document apply under the operational profile for the equipment and services, which shall be in accordance with its intended use, but as a minimum, shall be that specified in the applicable operational scenarios for testing contained in clause B.4.2 of the present document. The equipment and services shall comply with all the technical requirements of the present document when operating within the boundary limits of the operational profile defined by its intended use.

5 Accessible Emergency Communication

5.1 General

Making sure that emergency communications are accessible to persons with a range of capabilities implies considerations on many levels. The continuation of clause 5 covers specific interoperability and accessibility aspects and functional requirements on the complete chain of emergency communication technology including accessible interpersonal communication services and emergency communications, emergency service IP networks and PSAPs, as well as relay services and other supporting services.

All requirements or recommendations in this clause are clearly labelled to enable reference to them in the present document. Thereby it is possible to follow up how a functional requirement can be fulfilled. The label consists of three parts: three letters, being AEC for the main area: accessible emergency communication; two letters indicating a sub-area (example ML for modality and language) and a sequence number. The labels are repeated where the requirements are fulfilled by a technical requirement.

A cross reference table is provided in Annex D, providing the clause numbers where requirements are provided corresponding to the functional requirements labels.

Many aspects of accessible emergency communication are described in ETSI TS 101 470 [i.1], where the total conversation access to emergency communications is specified. The use of media for different purposes is described and implementations in various technical environments specified.

5.2 Modality and Language

Proper support for the modalities and languages in communication managed by the user in emergency is a prerequisite for efficient emergency communications. The means for providing of such support is by the user or user equipment providing information about the preferences and capabilities and the emergency services to collect this information and assessing how to best provide the support needed, by routing and possible invocation of supporting services. Modality competence has two directions and therefore, can be different for the two directions of expression and perception.

To support modalities other than voice, accessible emergency communications:

AEC-ML-01: shall enable users to set their modality and language preferences in their equipment or service.

NOTE: The modalities include signed, written and spoken expression carried in video, real time text and audio media together called total conversation enabling language use in all three modalities.

AEC-ML-02: shall enable the users to set their modality and language preferences separately for expression and perception.

AEC-ML-03: shall use the language selected for the user equipment as default language preference, if not set by the user.

AEC-ML-04: shall provide a means of transmitting information on modality and language unchanged to the PSAP at initiation of emergency communication.

AEC-ML-05: shall extract the language and modality preference information and include it in the routing decision.

AEC-ML-06: should upon answering emergency communication, send a greeting and a brief standardized question expressed in the preferred modality and language.

AEC-ML-07: shall upon answering an emergency communications test session, send a text response containing the received location, modality and language preferences and other additional data provided by the user.

AEC-ML-08: shall provide a means of transmitting information on support services desired in the communications unchanged to the PSAP at initiation of emergency communication.

AEC-ML-09: shall extract any information on the preferred support service for modality or language translation to be used and include the information in the emergency communication.

5.3 User Interface and general accessibility

Emergency communications users are best provided with emergency communications when such communications are supported by the end user equipment and electronic communications used in everyday communication.

The user interface and communication features used for accessible emergency communications shall:

AEC-UI-01: comply with the applicable requirements of ETSI EN 301 549 [1], clauses 5, 6, 8, 9, 11 and 13.

AEC-UI-02: provide a means to initiate a general emergency communication recognized as 112.

NOTE: An emergency call button can be more accessible than a numeric keypad, but with this solution follows a risk for communication initiation by mistake. Other ways to initiate the emergency communication without exactly relating it to a number could be considered e.g. because of accessibility reasons.

AEC-UI-03: support the input and output of voice and real time text.

AEC-UI-04: support in PSAP the input and output of video in emergency communications.

AEC-UI-05: support in user equipment the input and output of video in emergency communications if video is supported in other bidirectional communications.

The user interface used for accessible emergency communications may:

AEC-UI-06: provide means to include services or specific parts thereof specialized in different communication modalities and languages or motivated by accessibility reasons.

AEC-UI-07: support the input and output of text messaging.

5.4 Communication Features

5.4.1 General

The present clause specifies general communications operations used in the processing of emergency communications.

5.4.2 Session Control and emergency contextual information

Session setup in accessible emergency communications shall:

AEC-SC-01: include information describing the desired media.

AEC-SC-02: include accurate location information.

AEC-SC-03: include an identity or address to be used for emergency call back.

AEC-SC-04: support the technical possibility to handle the media, modality and language preferred by the user.

Session setup in accessible emergency communications should:

AEC-SC-05: include other contextual information about the user and the emergency available to enable the communication to be routed to the most appropriate PSAP, considering both geographical and accessibility factors.

AEC-SC-06: while in a waiting state, provide information via all activated media indicating to the user that the user is already connected but in a queue.

AEC-SC-07: when the communication is connected to a PSAP, answer the incoming communication with a suitable greeting phrase, if possible, in the preferred modality and language of the user when that can be determined.

5.4.3 Routing

Several factors influence decisions on routing the communication to a region and to a PSAP. Factors are responsibility over the location of the emergency and the load of the PSAP. In relation to accessibility, additional factors are of concern in decision of which is the most appropriate PSAP. The additional factors are capability of handling the modalities that the user in emergency manages (spoken, written and signed), the competence of the call taker in handling the preferred sign language or spoken or written language, experience of the call taker to assess the needs of persons with disabilities and invoke appropriate support in the communication.

The procedures for routing to the most appropriate PSAP in accessible emergency communications shall have the possibility to retrieve the following information for use in the routing decision process:

AEC-RO-01: location information and corresponding regions of PSAPs.

AEC-RO-02: the emergency service or subtype of emergency service (if any) being requested.

NOTE 1: Emergency service subtype e.g. Mountain rescue, coast guard etc. which may be coded into the initiation of the emergency communications.

AEC-RO-03: preferred modalities and language preferences.

NOTE 2: Within the concept of modalities and languages are sign language use, real time text use and speech use, with possibility to express different preferences in different directions.

AEC-RO-04: knowledge of the region of the emergency and how to request actions of first responders.

NOTE 3: This requirement reflects the national organization and if there is a preference for routing to a regional PSAP close to the location of the emergency, or if no geographical preference is assigned within the country.

AEC-RO-05: knowledge of call takers of how to communicate with a specific need for communication.

AEC-RO-06: immediately available location information so as not to delay the communication setup.

AEC-RO-07: mechanisms to support users travelling to other countries than home country (e.g. roaming).

NOTE 4: For some cases combinations of PSAPs or PSAP and supporting service may best correspond to the term "most appropriate PSAP".

AEC-RO-08: media information provided in the connection setup to select the PSAP with the highest match.

AEC-RO-09: an international packet-switching infrastructure to connect regional PSAPs in combination with a mapping hierarchy to determine access to PSAPs in remote service areas.

5.4.4 Communication Transfer

When transfer of accessible emergency communications is available, the system performing the transfer shall:

AEC-CT-01: have means to make attended transfer of the communication to other parties within the emergency service organization, where the:

a) original party may stay as long as suitable in the transferred communication;

b) transfer shall affect any media used in the communication before the transfer.

AEC-CT-02: have the means to make attended transfer to enable international cooperation if coordination between emergency services in different countries is required to resolve an emergency, where the:

a) original party may stay as long as suitable in the transferred communication;

b) transfer shall affect all media in the communication.

NOTE: Examples related to accessibility is when a person with a disability visiting a country gets into an emergency and only relies on their fluent communication with a relative in the home country, who gets asked to initiate an emergency communication about the emergency case. That communication will end up in the home country and a need for cooperation appears.

5.4.5 Conferencing

When conferencing in accessible emergency communications is available, the conferencing functions shall:

AEC-CO-01: have means to invoke and support multiparty communications in an emergency communication (e.g. in case of a communication transfer or when including relay services), where the multiparty communication shall be able to support all media.

AEC-CO-02: support a common media type for each combination of parties, if there is a difference in media support by different parties.

AEC-CO-03: have the means to invoke multiparty communications to enable international cooperation if coordination between emergency services in different countries is required to resolve an emergency.

NOTE: Examples related to accessibility is when a person with a disability visiting a country gets into an emergency and only relies on their fluent communication with a relative in the home country, who gets asked to initiate an emergency communication about the emergency case. That communication will end up in the home country and a need for cooperation appears.

5.4.6 Call back

Accessible emergency communications shall:

AEC-CB-01: be enabled to call back to the user in emergency.

The call back feature shall have the following characteristics:

AEC-CB-02: By default, enabling the same media as used in the incoming communication.

AEC-CB-03: Having the option to vary the media composition compared to the incoming communication.

AEC-CB-04: By default, if a supporting service was included during the initial communication, include the same supporting service in the call back.

AEC-CB-05: Having the option to vary the inclusion of a supporting service.

AEC-CB-06: Use globally routable addressing achieved from the incoming communication.

5.4.7 Charging

AEC-CG-01: The user shall not be charged specifically for accessible emergency communications.

5.5 Communication Media

5.5.1 General

Many of the accessibility considerations are related to the availability of media for the communication suitable for the capabilities of the user in emergency. The electronic communication service used by the user in an emergency shall support multiple communication participants for all media used in emergency communication.

5.5.2 Audio

Audio in emergency communications is needed for spoken communication, and for providing sounds from the emergency scene for assessing the emergency.

Accessible emergency communications shall:

AEC-CM-01: include bidirectional audio media in the communication from the beginning of the communication.

AEC-CM-02: fulfil the requirements on audio as in ETSI EN 301 549 [1], clause 6.1.

5.5.3 Video

Video in emergency communications is needed for sign language communication, for enhancing understanding of spoken language, for conveying calming influence and for providing views from the scene of the emergency for efficient assessment of the emergency.

Accessible emergency communications shall:

AEC-CM-03; provide the possibility to use bidirectional video communications with the PSAP.

AEC-CM-04: provide the user to include bidirectional video media in the communication from the beginning of the communication as well as by addition during the communication if video is available for other communications in the user equipment.

AEC-CM-05: provide the possibility to disconnect video once used in the communication.

AEC-CM-06: when supported, fulfil the requirements on video as specified in ETSI EN 301 549 [1], clause 6.5.

5.5.4 Real Time Text

Accessible emergency communications shall:

AEC-CM-07: provide the user to include real time text media in the communication from the beginning of the communication as well as by addition during the communication.

AEC-CM-08: fulfil the requirements on real time text as in ETSI EN 301 549 [1], clause 6.2.

5.5.5 Text Messaging

Accessible emergency communications may:

AEC-CM-09: provide the user to include the text messaging in the communication as well as by addition during the communication.

AEC-CM-10: support emergency communications that started with messaging to be upgraded to a communication with real time media by adding audio and any other real time media.

NOTE: Users of sentence-wise text messaging could have a preference to use text messaging also in an emergency because of their familiarity with the user interface even if it is slower than real time text. When the electronic communications service used by the user in emergency offers messaging for regular user-to-user communication, the service can be used also during and outside of an emergency communication.

5.5.6 Total Conversation

Total conversation is audio, video and real time text enabled and synchronized in a single bidirectional communication and fulfilling performance requirements on these media. The quality requirements are found in ETSI EN 301 549 [1], clause 6.

NOTE: The synchronization is assumed when all three media fulfil their individual performance requirements because the latency requirements are stricter than the synchronization requirements.

Accessible emergency communications in a PSAP shall:

AEC-CM-11: provide total conversation.

Accessible emergency communications in user equipment shall:

AEC-CM-12: when video is provided in user equipment and originating service, provide total conversation.

5.6 Relay-Service Invocation by the User in Emergency

Although it is preferable to directly contact emergency services, some users reach emergency services by first initiating communication with a relay service and then ask the relay service to initiate an emergency communication. The location of the user is then interrogated by manual means from the user in emergency by the relay service.

Accessible emergency communications shall:

AEC-RS-01: when a relay service is available to the user, enable a relay service to insert the location of the user in the emergency communications and get it properly routed.

NOTE 1: This is independent on where the communication is initiated.

AEC-RS-02: support to provide the location of the user from the relay service to the PSAP once connected as a fallback option.

NOTE 2: The latter case though can easily cause contact with a PSAP less appropriate to handle the emergency communication leaving to the PSAP to set up a connection with a more appropriate PSAP to handle the situation.

5.7 Supporting Services

5.7.1 General

Some situations require support to PSAPs from other organizations or services. Such supporting services may be relay services for modality translation, language support services for language translation and expert support services for assistance in the emergency situations. See ETSI EN 301 549 [1], clause 13, and ETSI ES 202 975 [2].

Emergency communication networks and PSAPs shall:

AEC-SS-01: have means to invoke supporting services in a multiparty fashion in the emergency communication.

AEC-SS-02: have means to automatically and manually include, extract and use addresses to such supporting services provided in user data related to the emergency communication.

AEC-SS-03: have means to find addresses to such supporting services for manual invocation.

AEC-SS-04: support a preference for invoking these services in a multiparty fashion, so that the user in emergency can perceive the communication with any party involved.

AEC-SS-05: have means to establish chains of supporting services for situations where a modality translating service needs to be combined with a spoken language translation service.

NOTE: This situation can e.g. appear when a sign language user causes invocation of a sign language interpreter, but there is no interpreter available for translating any language supported by the emergency service in the country of the emergency. Then also a spoken language interpreter needs to be invoked in the communication.

5.7.2 Relay Services

Relay services convert between different real time communication modalities.

During accessible emergency communications:

AEC-SS-06: when a relay service is available to the user, the PSAP shall have the possibility to invoke the relay service in the emergency communications by a single multimedia connection. See ETSI ES 202 975 [2].

NOTE 1: Relay services often operate between one incoming and one outgoing connection. For ease of setting up the communication, the connection in this case will be a single one, and the relay service acting on the media in this single connection.

Relay services may be:

AEC-SS-07: video relay services between sign language and speech providing means for occasional interaction in real time text.

AEC-SS-08: text relay services between real time text and speech.

AEC-SS-09: speech-to-speech relay services between speech that is hard to understand because of a disability and clear speech.

AEC-SS-10: relay services supporting a person's memory or other cognitive capabilities.

AEC-SS-11: captioned relay services adding real time text captions to speech in one direction of a communication.

NOTE 2: Relay services are often asked to operate only in one direction of the communication. That is e.g. when a person is hard-of-hearing, but talks well. Then modality conversion is only needed from speech to real time text.

5.7.3 Translation Service

Translation services translate mainly between two spoken languages.

Accessible emergency communications shall:

AEC-SS-12: have the technical means to invoke a translation service supporting the media used in the emergency communication.

NOTE: This inclusion can be important to bridge a language gap between a PSAP and a relay service in a foreign country.

5.7.4 Expert service

An expert service gives advice to the user or the PSAP during an emergency communication.

NOTE: This may be internal or external to the emergency service organization and handle poison expertise, medical advice, expert in specific disabilities, or any other similar advice function found needed.

Accessible emergency communications shall:

AEC-SS-13: when needed, have the technical means to invoke an expert service in the emergency communications supporting any media used.

5.8 Documentation

Accessible documentation about products and services involved in emergency communication shall be provided for the following:

AEC-AD-01: consumer terminal equipment with interactive computing capability, used for electronic communications services and emergency communications.

AEC-AD-02: electronic communications services providing emergency communications.

AEC-AD-03: emergency communications networks and PSAP services.

AEC-AD-04: supporting services used to support emergency communications.

The documentation shall be provided according to the requirements in ETSI EN 301 549 [1], clause 12.

NOTE: This requirement is only presented here.

6 Technical Architecture

Figure 1 presents a symbolic figure over the functional entities participating in the emergency communication and the interfaces between them.

The user equipment resides in an originating service domain, with an originating communication service provider.

The communication service provider initiates emergency communications with an emergency communications network of the PSAP domain.

The use of a Forest Guide results in routing of the communication to the emergency communications network in the region responsible for handling emergencies in the target area (illustrated as other PSAP domain).

Routing based on location and other factors take place so that the most appropriate PSAP receives the communication setup and starts handling it.

Information in the communication establishment, either automatically provided, or manually with the call taker and the user concludes further details in the handling of the emergency communication, e.g. if any supporting service will be invoked in the communication.

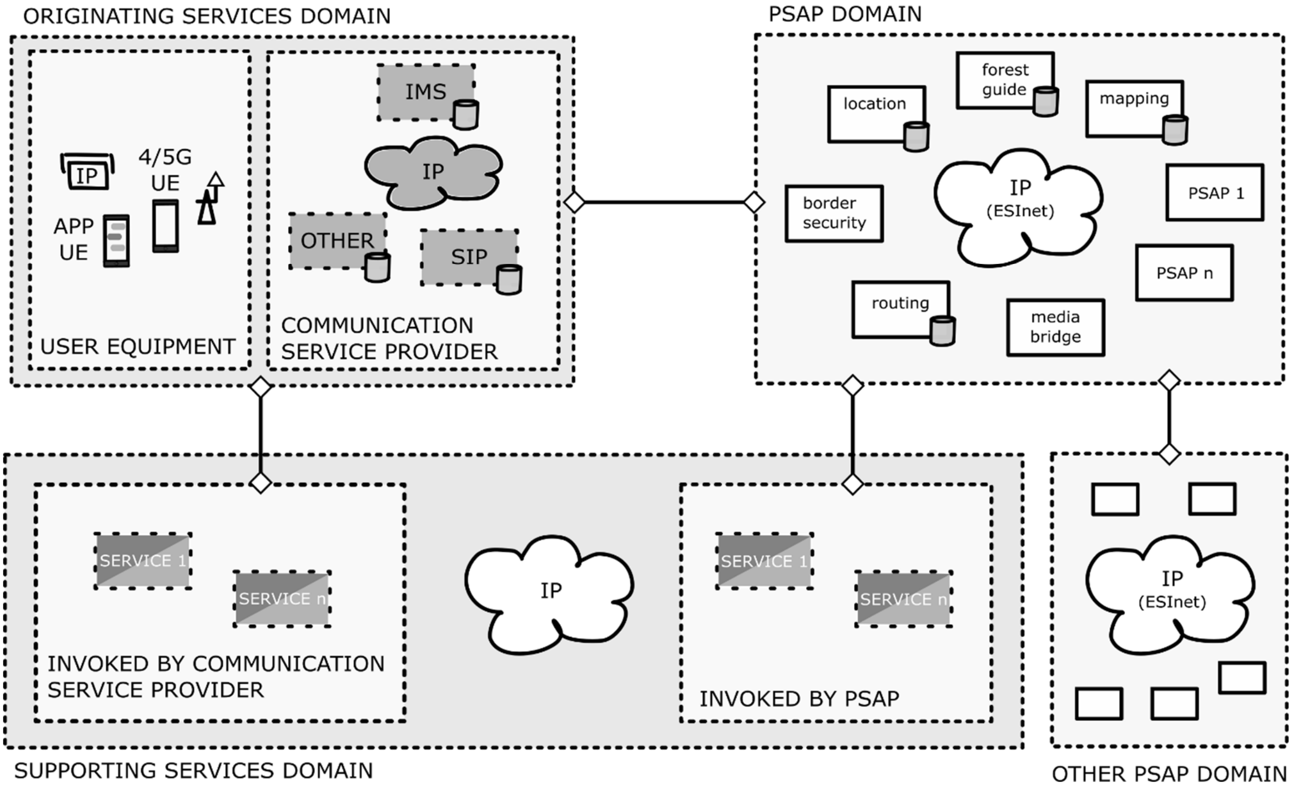

Figure 1: Conceptual architecture of emergency communications domains

IP Multimedia Subsystem (IMS) and Session Initiation Protocol (SIP) are both integral components in current communication systems of the originating services domain, but they serve different purposes and have distinct features. IMS works with multiple access networks, including Wi-Fi®, LTE, and 5G, and provides a standardized platform for mobile network operators to deliver multimedia services and SIP is used in traditional Voice over IP services.

- Although SIP is a component of IMS, there are aspects of accessible emergency communication that need to be considered differently for IMS and SIP based communication system. For this reason, both solutions required for accessible emergency communication are described in detail in the present document. For the user equipment and the communication service provider within the originating service domain, the range of technologies to be implemented are: IMS based, using packet switched technologies in mobile or fixed networks, using the specifications being developed and maintained by 3GPP, available from ETSI and introduced in ETSI TS 122 228 [i.2].

- SIP based, meaning equipment and services using the session initiation protocol SIP (IETF RFC 3261 [10]) in a general way, commonly on the Internet. This is the technology specified for the packet switched emergency communications networks and interfaces to PSAPs, and often used by voice and multimedia over IP (VoIP) services.

Any other technology than the above for communication establishment and media communication with potential to implement the requirements on accessible emergency communication is introduced as other technologies for emergency communication. One reason to introduce this category is to indicate a potential for evolution of emergency communications without causing fragmentation and degradation of service level.

The following clauses 7-10 contain technical requirements divided in separate clauses for each component or functional entity of the introduced architecture taking part in the emergency communication. The labels assigned to functional requirements in the previous clauses of the present document are repeated here in parenthesis to indicate which technical requirement resolves the functional requirement.

7 User equipment requirements

7.1 General

The User Equipment (UE) handles the emergency communications with the user in emergency together with the electronic communications service it resides in. The result is that the emergency communications in the interface between the service and the emergency service network are aligned with the conventions in that interface as specified in ETSI TS 103 479 [3], clause 6 except 6.1.3, 6.2, 6.3, 6.4 and 6.8. The part of the User Equipment for these procedures is specified in this clause for two closely related technologies, IMS and SIP. In general, other technologies may distribute the procedures in other ways to accomplish the same result in the emergency service network interface.

This clause has focus on factors related to accessibility of emergency communications. For general aspects of emergency communications, see ETSI TS 103 479 [3].

For general accessible functionality of user equipment. the applicable requirements of ETSI EN 301 549 [1], clauses 5, 6, 8, 9, 11 and 13 shall be fulfilled. (AEC-UI-01).

7.2 IMS Based User Equipment

7.2.1 General

The User Equipment (UE) handles the emergency communications with the user in emergency together with the IMS service it resides in. The result is that the emergency communications in the interface between the IMS service and the emergency service network are aligned with the conventions in that interface as specified in ETSI TS 103 479 [3], clause 6 except 6.1.3, 6.2, 6.3, 6.4 and 6.8. The part of the User Equipment for these procedures is specified in the present clause.

The present clause has focus on factors related to accessibility. For general aspects, see ETSI TS 103 479 [3].

For accessible functionality, see clause 5 of the present document.

IMS based user equipment is user equipment using the IP multimedia subsystem concept (IMS) ETSI TS 122 228 [i.2].

Real Time Text (RTT) shall be supported as specified in ETSI TS 126 114 [9], in the clauses about real time text and text. (AEC-CM-07), (AEC-UI-03).

NOTE: Real time text is varyingly called text, real time text, GTT, GTT-IP and RTT in the ETSI IMS documents.

Wide band audio shall be supported as specified in ETSI TS 126 114 [9] in the clauses about audio. (AEC-CM-01), (AEC-CM-02), (AEC-UI-03).

When video is supported, it shall be supported as specified in ETSI TS 126 114 [9] in the clauses about video with quality as required by ETSI TS 103 479 [3], clause 6.6.2.3. (AEC-CM-03), (AEC-CM-06), (AEC-UI-05).

When video is supported, it shall be possible to include video from the beginning of the communication and added or deleted during the communication. (AEC-CM-04).

When video is supported, total conversation shall be supported. (AEC-CM-12).

Multiparty handling shall be supported as specified in IETF RFC 4579 [14] and the specific procedures required for multiparty handling of each supported media. (AEC-CT-01).

7.2.2 Settings

The user equipment shall enable the user to set accessibility related preferences of importance for emergency communications. The settings are stored in the communications service.

These settings contain but are not limited to the following:

- Preference for always including RTT initially in both outgoing and incoming communications. (AEC-CM-08).

- Data about Subscriber/Owner (user) including data of interest from accessibility point of view to be provided in the Additional information in emergency communications according to IETF RFC 7852 [15].
- Preferred modality and language. See clause 9.3 of the present document. (AEC-ML-01), (AEC-ML-02).
- Address to any preferred support service. (AEC-SS-02).

7.2.3 Initiation of emergency communications

The emergency communications shall be initiated according to the procedures specified in ETSI TS 126 114 [9], ETSI TS 122 101 [5], clause 10 except clauses 10.2 and 10.3, and ETSI TS 124 229 [8], clauses 4.7 and 5.1.6 for emergency communications in IMS. (AEC-SC-01), (AEC-SC-02), (AEC-UI-02).

Dialling number 112 shall be among the supported ways to initiate emergency communications. The same way shall be available regardless of desired modality, language and media support. (AEC-UI-02).

Location information shall be included as specified in ETSI TS 103 479 [3], clauses 6.1.2.2 and 6.1.4. (AEC-SC-02).

Any media preferences indicated by the user shall influence the initiation by including the preferred media in the initiation and include indications of preferred modalities and language if provided by the user. (AEC-SC-01), (AEC-SC-04).

The initiation of emergency communication in limited-service state (i.e. the home PLMN is not available, but another PLMN) shall be supported according to the procedures defined in ETSI TS 123 167 [7].

A routable address to the user equipment shall be included by the user equipment if the user equipment has access to that information. (AEC-SC-03).

Additional data including its settings for the Subscriber/Owner (user) shall be included in the Call-Info header according to IETF RFC 7852 [15]. (AEC-SC-05).

Any indication of preferred modality or language shall be provided in the initiation according to IETF RFC 8373 [16]. If the user has not made any settings about preferred modality and language, then the language of the user equipment platform shall be included as the language preference. (AEC-SC-04), (AEC-ML-03).

Preferred support service address shall be included in the RELATED property of the vCard element of the SubscriberData according to IETF RFC 7852 [15] if specified by the user. (AEC-ML-08).

7.2.4 Call Back

A call back is indicated by the SIP Priority header with value "psap-callback" from the PSAP to the user equipment, as defined in IETF RFC 7090 [18].

Any call back from the emergency service shall be handled by the user equipment and indicated to the user as emergency communication. (AEC-CB-01).

The media shall be handled as in any incoming communication. If the user has preference for including RTT, and RTT is not included in the call back, then the user equipment shall modify the SIP dialog to add RTT as specified in ETSI TS 103 479 [3], clause 6.6.2.4. (AEC-CB-02).

If video is available in the service and the user responds with an indication to add video to the call, and video is not included initially in the call back, then the user equipment shall modify the SIP dialog to add video as specified in ETSI TS 103 479 [3], clause 6.6.2.3. (AEC-CB-03).

NOTE: The mechanism described allows the PSAP to reject media offered by the user equipment for operational reasons in any case.

7.2.5 Visiting regions and networks

The IMS system includes roaming functionality, which means that user equipment in a visited network performs emergency communications through the functional entities in the visited network. That causes specific considerations in roaming conditions, specified in ETSI TS 123 167 [7] and ETSI TS 124 229 [8], clauses 4.7 and 5.1.6 which user equipment in roaming conditions shall follow. (AEC-RO-06).

The UE shall be able to use the media supported by the UE in this situation and provide the required functionality. (AEC-CM01), (AEC-CM-02), (AEC-CM-03), (AEC-CM-07), (AEC-CM-10), (AEC-CM-11).

7.2.6 Invocation of supporting services by the user

Users who find themselves in an emergency may not remember that their accessibility needs will be met efficiently if they initiate the emergency communication in the recommended way. This could especially be the case if they are used to calling support services in their daily communication. When a user invokes a support service in an emergency, a fallback method is required. In this fallback method, the user initiates communication with the preferred support service and asks the support service personnel to initiate emergency communication. (AEC-UI-06).

For this fallback case, there is no requirement to be equally efficient as for direct communication with a PSAP.

The user or user equipment shall be enabled to provide location information in the connection with the supporting service. (AEC-RS-02).

When the user equipment is in a roaming situation, it is of extra importance that the support service is able to use the location information of the user for the emergency communication and operates a connection with a local ESInet, as this will enable the communication to reach the emergency service in the region where the user in emergency is. (AEC-RS-01), (AEC-RS-02).

The actions by the support service for this case shall be as specified in clause 7.2.3 of the present document with information about the support service replacing that of the user device where appropriate. (AEC-SC-03), (AEC-SC-05).

7.3 SIP Based User Equipment

7.3.1 General

User equipment called "SIP based" in the present document is user equipment using IETF RFC 3261 [10] and IETF RFC 8866 [17] for session and media control, and not included within the IP multimedia subsystem concept (IMS) ETSI TS 122 228 [i.2].

Real time text (RTT) shall be supported as specified in ETSI TS 103 479 [3], clause 6.6.2.4.(AEC-CM-01), (AEC-CM-02), (AEC-CM-07), (AEC-UI-03).

Wide band audio shall be supported as specified in the wide band parts of ETSI TS 103 479 [3], clause 6.6.2.2 (AEC-CM-02), (AEC-UI-03).

When video is supported by the user equipment in general bidirectional communication, it shall be supported in emergency communication as specified in ETSI TS 103 479 [3], clause 6.6.2.3.(AEC-CM-03), (AEC-CM-05), (AEC-UI-05).

When video is supported, it shall be possible to include video from the beginning of the communication and added or deleted during the communication. (AEC-CM-04).

When video is supported, total conversation shall be supported. (AEC-CM-12).

Multiparty handling shall be supported as specified in in IETF RFC 4579 [14] and the specific procedures required for multiparty handling of each supported media. (AEC-CT-01).

Text messaging may be supported as defined in ETSI TS 103 698 [4] (AEC-UI-07), (AEC-CM-09), (AEC-CM-10).

7.3.2 Settings

The user equipment shall enable the user to set accessibility related preferences of importance for emergency communications. The settings may be stored in the user equipment or in the service as decided by the service provider.

These settings comply but are not limited to the following:

- Preference for always including RTT initially in both outgoing and incoming communications. (AEC-CM-08).

- Data about Subscriber/Owner including data of interest from accessibility point of view to be provided in the Additional information in emergency communications according to IETF RFC 7852 [15]. (AEC-SC-05).
- Preferred modality and language. See clause 9.3 of the present document. (AEC-ML-01), (AEC-ML-02).
- Address to preferred support service. (AEC-SS-02).

7.3.3 Initiation of emergency communications

The emergency communications shall be initiated according to the procedures specified in ETSI TS 103 479 [3], clause 6 except 6.1.3, 6.2, 6.3, 6.4 and 6.8. (AEC-SC-01), (AEC-UI-02).

Dialling number 112 shall be among the supported ways to initiate emergency communications. The same way shall be available regardless of desired modality, language and media support. (AEC-UI-02).

Location information shall be included as specified in ETSI TS 103 479 [3], clauses 6.1.2.2 and 6.1.4. (AEC-SC-02).

The media preferences indicated by the user shall influence the initiation by including the preferred media in the initiation and include indications of preferred modalities and language. (AEC-SC-01), (AEC-SC-04).

A routable address to the UE shall be included by the user equipment if the UE has access to that information. (AEC-SC-03).

Additional data including the settings for the user shall be included according to IETF RFC 7852 [15]. (AEC-SC-05).

Preferred modality and language shall be indicated in the initiation according to IETF RFC 8373 [16]. If the user has not made any settings about preferred modality and language, then the language of the user equipment platform shall be included as the language preference. (AEC-ML-03), (AEC-SC-04).

Preferred support service address shall be included in the RELATED property of the vCard element of the SubscriberData according to IETF RFC 7852 [15] if specified by the user. (AEC-ML-08).

7.3.4 Call back

A call back is indicated by the SIP Priority header with value "psap-callback" from the PSAP to the user equipment, as defined in IETF RFC 7090 [18].

Any call back from the emergency service shall be handled by the user equipment and indicated to the user. (AEC-CB-01).

The media shall be handled as in any incoming communication. If the user has preference for including RTT, and RTT is not included in the call back, then the user equipment shall modify the SIP dialog to add RTT as specified in ETSI TS 103 479 [3], clause 6.6.2.4. (AEC-CB-02).

If the user responds with an indication to add video to the call, and video is not included initially in the call back, then the user equipment shall modify the SIP dialog to add video as specified in ETSI TS 103 479 [3], clause 6.6.2.3. (AEC-CB-03).

NOTE: The mechanism described allows the PSAP to reject media offered by the user equipment for operational reasons in any case.

7.3.5 Invocation of support services by the user

Users who find themselves in an emergency may not remember that their accessibility needs will be met efficiently if they initiate the emergency communication in the recommended way. This could especially be the case if they are used to calling support services in their daily communication. When a user invokes a support service in an emergency, a fallback method is required. In this fallback method, the user initiates communication with the preferred support service and asks the support service personnel to initiate emergency communication. (AEC-RS-01), (AEC-UI-06).

For this fallback case, there is no requirement to be equally efficient as for direct communication with PSAPs.

The user or user equipment shall be enabled to provide location information in the connection with the supporting service. (AEC-RS-02).

All media present in the path between the user and the relay service shall be present in the path between the relay service and the emergency communications network. (AEC-SS-04).

The actions by the support service for this case shall be as specified in clause 7.3.3 of the present document with information about the support service replacing that of the user device where appropriate. (AEC-SC-03), (AEC-SC-05).

7.4 Other technologies for communication services

If other technologies in addition to IMS and SIP are used for session and media control by the user equipment, the underlying communication service is responsible for performing emergency communication according to the procedures described in clause 8.4 of present document. In that case, the service involves the user equipment in the emergency communications procedures as specified for the interpersonal communications service in clause 8.4 of the present document. The same accessibility functional considerations as for the IMS and SIP shall be fulfilled. (AEC-CM-01), (AEC-CM-02), (AEC-CM-03), (AEC-CM-06), (AEC-CM-07), (AEC-CM-12), (AEC-CT-01), (AEC-SC-01), (AEC-SC-02), (AEC-SC-03), (AEC-SC-04) (AEC-SC-05), (AEC-SS-02), (AEC-ML-03), (AEC-UI-01), (AEC-UI-02), (AEC-UI.-03), (AEC-UI-05), (AEC-UI-06).

8 Originating Service Requirements

8.1 General

The originating service handles the emergency communications with the PSAP as requested by the user equipment. The result is that the emergency communications in the interface between the service and the emergency communications network (ESInet) is aligned with the conventions in that interface as specified in ETSI TS 103 479 [3], clause 6 except 6.1.3, 6.2, 6.3, 6.4 and 6.8. The part of the originating service for these procedures is specified in the present clause for IMS, SIP, and other technologies in addition to IMS and SIP.

The present clause has focus on factors related to accessibility of emergency communications. For general aspects of emergency communications, see ETSI TS 103 479 [3].

For general accessible functionality of originating services. the applicable requirements of ETSI EN 301 549 [1], clauses 5,6,8, 9, 11 and 13 shall be fulfilled. (AEC-UI-01).

The user shall not be charged for resources being part of an accessible emergency communication and under control of the electronic communications service. (AEC-CG-01).

For accessible functionality, see also clause 5 of the present document.

8.2 IMS Based Originating Service

8.2.1 General

The IMS multimedia telephony service, where the user equipment (UE) is when the emergency communications is established, handles the emergency communications between the user in emergency and the PSAPs. The result shall be that the emergency communications in the interface between the IMS service and the emergency communications network are aligned with the conventions in that interface as specified in ETSI TS 103 479 [3], clause 6 except 6.1.3, 6.2, 6.3, 6.4 and 6.8. The part of the originating service for these procedures is specified in the present clause.

The present clause has focus on factors related to accessibility. For general aspects, see ETSI TS 103 479 [3].

For accessible functionality, see clause 5 of the present document.

IMS based multimedia telephony services implement the IP multimedia subsystem concept (IMS) ETSI TS 122 228 [i.2] and the IMS Multimedia Telephony services as specified in ETSI TS 122 173 [6].

Audio and real time text (RTT) shall be supported as specified in ETSI TS 126 114 [9], clauses about real time text and text. (AEC-CM-01), (AEC-CM-07).

NOTE: Real time text is varyingly called text, real time text, GTT, GTT-IP and RTT in the ETSI IMS documents.

Wide band audio shall be supported as specified in ETSI TS 126 114 [9] audio related clauses. (AEC-CM-02).

When video is supported, it shall be supported as specified in ETSI TS 126 114 [9] with quality as required by ETSI TS 103 479 [3], clause 6.6.2.3. (AEC-CM-03), (AEC-CM-06).

When video is supported, total conversation shall be supported. (AEC-CM-12).

Multiparty handling shall be supported as specified in IETF RFC 4579 [14] and the specific procedures required for multiparty handling of each supported media.

8.2.2 Settings

The IMS service shall enable the user to set accessibility related preferences of importance for emergency communications. The settings are stored in the service.

These settings comply but are not limited to the following:

- Preference for always including RTT initially in both outgoing and incoming communications. (AEC-CM-07).
- Data about Subscriber/Owner including data of interest from accessibility point of view to be provided in the Additional information in emergency communications according to IETF RFC 7852 [15]. (AEC-SC-05).
- Preferred modality and language. See clause 9.3 of the present document. (AEC-ML-01), (AEC-ML-02).
- Address to preferred support service. (AEC-SS-02).

8.2.3 Initiation of emergency communications

The emergency communications shall be initiated according to the procedures specified in ETSI TS 126 114 [9], ETSI TS 122 101 [5], clause 10 except clauses 10.2 and 10.3, and ETSI TS 124 229 [8], clauses 4.7 and 5.1.6 for emergency communications in IMS. The exchange with the PSAP shall be performed as specified in ETSI TS 123 167 [7]. (AEC-SC-01), (AEC-SC-02).

Location information shall be included as specified in ETSI TS 123 167 [7] so that it can be retrieved as specified in ETSI TS 103 479 [3], clauses 6.1.2.2 and 6.1.4.

The media preferences indicated by the user shall, when available, be conveyed in the initiation. (AEC-SC-01).

The initiation of emergency communication in limited-service state (i.e. the home PLMN is not available, but another PLMN) shall be supported according to the procedures defined in ETSI TS 123 167 [7].

A routable address to the user equipment shall be included in the initiation. (AEC-SC-03).

Additional data including service information and settings for the user shall, when available, be included according to IETF RFC 7852 [15]. (AEC-SC-05).

Preferred modality and language shall, when available, be conveyed in the initiation. (AEC-SC-01), (AEC-SC-05).

Preferred support service address shall be included in the RELATED property of the vCard element of the SubscriberData according to IETF RFC 7852 [15] if specified by the user. (AEC-ML-08).

8.2.4 Call back

Any call back from the PSAP shall be handled by the IMS service and conveyed to the user equipment. (AEC-CB-01), (AEC-CB-02), (AEC-CB-03).

The IMS service shall convey the SIP Priority header with value "psap-callback" from the PSAP to the user equipment, as defined in IETF RFC 7090 [18].

8.2.5 Visiting regions and networks

Support for roaming user equipment to have emergency communications shall be provided by the visited service. (AEC-RO-07).

That causes specific considerations in roaming conditions, specified in ETSI TS 123 167 [7] and ETSI TS 124 229 [8], clauses 4.7 and 5.1.6 which services and user equipment in roaming conditions shall follow. (AEC-RO-07). See also clause 9.7 of the present document.

The UE shall be supported to have communication in all three media of total conversation in the visiting scenario. (AEC-CM01), (AEC-CM-02), (AEC-CM-03), (AEC-CM-07), (AEC-CM-10), (AEC-CM-11).

8.3 SIP Based Originating Service

8.3.1 General

Originating services here called "SIP based" use IETF RFC 3261 [10] and IETF RFC 8866 [17] for session and media control, and are not included within the IP multimedia subsystem concept (IMS) ETSI TS 122 228 [i.2].

Real time text (RTT) shall be supported as specified in ETSI TS 103 479 [3], clause 6.6.2.4. (AEC-CM01), (AEC-CM-02).

Wide band audio shall be supported as specified in the wide band parts of ETSI TS 103 479 [3], clause 6.6.2.2. (AEC-CM-02).

When video is supported, it shall be supported as specified in ETSI TS 103 479 [3], clause 6.6.2.3. (AEC-CM-03).

When video is supported, total conversation shall be supported. (AEC-CM-12).

Multiparty handling shall be supported as specified in in IETF RFC 4579 [14] and the specific procedures required for multiparty handling of each supported media. (AEC-CT-01).

8.3.2 Settings

The originating service shall enable the user to set accessibility related preferences of importance for emergency communications. The settings may be stored in the user equipment or in the service as decided by the service provider.

These settings comply but are not limited to the following:

- Preference for always including RTT initially in both outgoing and incoming communications. (AEC-CM-07).
- Data about Subscriber/Owner including data of interest from accessibility point of view to be provided in the Additional information in emergency communications according to IETF RFC 7852 [15].
- Preferred modality and language. See clause 9.3 of the present document. (AEC-ML-01), (AEC-ML-02).
- Address to preferred support service.

8.3.3 Initiation of emergency communications

The emergency communications shall be initiated according to the procedures specified in ETSI TS 103 479 [3], clause 6 except 6.1.3, 6.2, 6.3, 6.4 and 6.8. (AEC-SC-01), (AEC-SC-02).

Location information shall be included as specified in ETSI TS 103 479 [3], clauses 6.1.2.2 and 6.1.4.

Any media preferences indicated by the user shall be conveyed to the PSAP and influence the initiation so that the communication is routed to the most appropriate call taker to handle the communication. (AEC-SC-01), (AEC-SC-04).

A routable address to the user equipment shall be included by the originating service. (AEC-SC-03).

Additional data for the service and the settings for the user shall when available be included according to IETF RFC 7852 [15].

Preferred modality and language shall when available be conveyed in the initiation. (AEC-SC-01), (AEC-SC-04).

Preferred support service address shall when available be conveyed if specified by the user. included in the RELATED property of the vCard element of the SubscriberData according to IETF RFC 7852 [15] if specified by the user. (AEC-ML-08).

8.3.4 Call back

Any call back from the PSAP shall be handled by the SIP service and conveyed to the user equipment. (AEC-CB-01), (AEC-CB-02), (AEC-CB-03).

The SIP service shall convey the SIP Priority header with value "psap-callback" from the PSAP to the user equipment, as defined in IETF RFC 7090 [18].

8.3.5 Visiting regions and networks

When an emergency communication is initiated from a region other than the home region of the user in emergency, some conditions, which always shall be considered, stand out requiring extra considerations specified in clause 9.8 of the present document.

International peering or interconnect of PSAP domains is prerequisite to route remotely connecting VoIP subscribers to the most appropriate PSAP in visiting regions. PSAP domains shall interconnect their regional network infrastructure and support the mandatory interfaces according to clause 4.3 of ETSI TS 103 479 [3] via this interconnection and may implement optional interfaces according to clause 4.4 of ETSI TS 103 479 [3] via this interconnection.

Mapping procedures that are carried out for the forwarding of emergency communications are described in ETSI TS 103 479 [3]. This is based on regional mapping functions that are used in a hierarchical structure (tree-like) with a forest guide as an element for navigation between these trees. Regional forest guides shall maintain a LoST interface, as described in ETSI TS 103 479 [3], clause 5.3.

8.4 Other technologies for emergency communication

8.4.1 General

If other technologies in addition to IMS and SIP are used for session and media control by the user equipment, the underlying communication service is responsible for performing emergency communication according to the procedures described in clause 8.4 of present document.

If other technologies in addition to IMS and SIP are used for session and media control within the originating communications service, including with its user equipment, the originating service is responsible to perform the emergency communications as required in the interface to the emergency communications network. Regardless of how these services are implemented, they shall provide an accurate location, shall make routing decisions that are consistent with the principles of the present document, and shall provide a globally routable identity or address used for the emergency call-back.

All functional requirements specified in clause 5 of the present document that are applicable to user communication devices and interpersonal communications services, shall be fulfilled by such other technologies in addition to IMS and SIP if used for emergency communications. (AEC-**-**).

Options for interworking with ETSI TS 103 479 [3] are described in the following clauses.

8.4.2 Conversion to ETSI TS 103 479 emergency communication interfaces

To establish the required pan-European interoperability for emergency communications when using other technologies in addition to IMS and SIP, the following shall apply:

- The communication between the communication service provider and the emergency communications network in the region of the emergency shall be as specified in ETSI TS 103 479 [3], clause 6 except 6.1.3, 6.2, 6.3, 6.4 and 6.8. In that case, any conversions needed to adapt to the procedures in the emergency communications network interface shall be done by the originating communications service.
- The procedure shall use the location information of the user in emergency to route the communication to the appropriate emergency communications network. (AEC-SC-02).
- The media preferences indicated by the user shall if available be conveyed in the initiation. (AEC-SC-04).
- A routable address to the user equipment shall be included in the initiation for use in call back situations. (AEC-SC-03).
- Additional data including service information and settings for the user shall if available be included according to IETF RFC 7852 [15]. (AEC-SC-05).
- Preferred modality and language shall if available be conveyed in the initiation. (AEC-SC-04).
- Preferred support service address shall be included if specified by the user.

8.4.3 Use of ETSI TS 103 479 together with PEMEA

8.4.3.1 General

To IMS and native SIP technologies, another complementary method for establishing emergency communications is by using PEMEA specifications according to ETSI TS 103 478 [i.4].

The PEMEA specification provides standardized interfaces for multimedia emergency communications and emergency context information. Refer to ETSI TS 103 478 [i.4] for the general aspects, ETSI TS 103 871 [i.6] for RTT, ETSI TS 103 945 [i.8] for video and audio and ETSI TS 103 755 [i.5] for integration with ETSI TS 103 479 [3].

The use of ETSI TS 103 479 [3] together with PEMEA (refer to ETSI TS 103 478 [i.4], clause 8 of ETSI TS 103 478 [i.4]), and ETSI TS 103 755 [i.5]) shall be under conditions described in the present clause.

Further guidelines on interoperability, re-use, and enhancement of PEMEA can be found in ETSI TS 103 755 [i.5].

8.4.3.2 Initiating emergency communication with PEMEA support

If a user of mobile application initiates emergency communications from a user equipment (UE) as described in ETSI TS 103 478 [i.4], the application should establish the emergency communication according to the procedures defined in ETSI TS 126 114 [9], ETSI TS 122 101 [5], clause 10 except clauses 10.2 and 10.3, and ETSI TS 124 229 [8], clauses 4.7 and 5.1.6 for emergency communications in IMS. The exchange with the emergency communications network and PSAP shall in the latter case be performed as specified in ETSI TS 123 167 [7]. (AEC-SC-01), (AEC-SC-02).

8.4.3.3 Emergency communications by IMS in regions without PEMEA support

If a mobile application discovers that a PEMEA service is not available as described in ETSI TS 103 872 [i.7], and the UE supports IMS emergency sessions as specified in ETSI TS 123 167 [7], it shall immediately establish an emergency communication according to the procedures defined in ETSI TS 126 114 [9], ETSI TS 122 101 [5], clause 10 except clauses 10.2 and 10.3, and ETSI TS 124 229 [8], clauses 4.7 and 5.1.6 for emergency communications in IMS. The exchange with the emergency communications network and PSAP shall be performed as specified in ETSI TS 123 167 [7]. (AEC-SC-01), (AEC-SC-02).

8.4.3.4 Emergency communications by SIP in regions without PEMEA support

If an application discovers that a PEMEA service is not available as described in ETSI TS 103 872 [i.7], and the UE supports SIP based emergency sessions as specified in clause 8.3.3 of the present document, it should immediately establish an emergency communication according to the procedures defined in clause 8.3.3 of the present document.

8.4.3.5 Accessibility related details in use of PEMEA

The *emergencyDataSend* information as described in ETSI TS 103 478 [i.4], clause 6.2 includes an identity or address to be used for emergency call back. (AEC-SC-03).

To utilize the multimedia capabilities of PEMEA or to enrich emergency communications with additional media capabilities, the media and modality preferences, if indicated by the user, are conveyed in the initiation of emergency communications by the means described in ETSI TS 103 478 [i.4] in the Additional Data element in clause 13.6.2.4 "SubscriberData :- language for both ways".

If language preference both ways are not specified, a general language and modality preference can be specified in the Additional Data element in clause 13.6.2.1 "SubscriberData :- vcard profile", the language field. (AEC-SC-01), (AEC-SC-04).

Preferred support service address shall be included if specified by the user. (AEC-SS-01).

8.4.4 Use of ETSI TS 103 479 with future technologies

If future technologies in addition to IMS and SIP are used for emergency communications with full accessibility functions according to clause 5 of the present document (AEC-**-**), then the communication between the communication service provider and the emergency communications network and PSAP in the region of the emergency may use that interface for the emergency communication. Procedures of future technologies should be in accordance with the procedures described in clause 8.4 of the present document.

9 Emergency communications network and PSAP

9.1 General

The emergency communications network and PSAPs shall support the mandatory interfaces listed in clause 4.3 and defined in clause 6 of ETSI TS 103 479 [3] and may support optional interfaces listed in clause 4.4 and defined in clause 6 of ETSI TS 103 479 [3] to provide an interface to the originating communication services. The services provided by the emergency communications network and PSAPs shall fulfil the applicable accessibility requirements of ETSI EN 301 549 [1], clauses 5, 6 and 8, 9, 11, 12, 13 as well as the specific requirements of the present document. (AEC-UI-01).

9.2 Media

The media: video, audio and real time text (RTT) shall be supported as specified in ETSI TS 103 479 [3], clauses 6.6.2.4, 6.6.2.2, and 6.6.2.3. (AEC-CM-01), (AEC-CM-02), (AEC-CM-03), (AEC-CM-04), (AEC-CM-05), (AEC-CM-06), (AEC-UI-04).

Total conversation shall be supported. (AEC-CM-11).

Audio and RTT shall be supported in all media handling elements. (AEC-CM-01), (AEC-CM-02).

Multiparty handling according to IETF RFC 4579 [14] shall be supported by media handling elements and in the external interfaces. (AEC-CO-01), (AEC-CO-02).

9.3 Modality and Language indication

The following indications shall be supported by emergency communications networks and PSAPs and be used to convey the need for support of efficient communication with users with disabilities involving specific modality and language needs with call takers. (AEC-ML-03), (AEC-ML-04), (AEC-ML-05), (AEC-ML-06).

IETF RFC 8373 [16], a=hlang-send and a=hlang-recv sdp media attributes including negotiation of modality and language support. (AEC-SC-04), (AEC-ML-05).

IETF RFC 7852 [15] Additional data in an emergency call. (AEC-SC-05).

The following indications should be supported by emergency communications networks and PSAPs and may be used to convey the need for support of efficient connection with users with disabilities involving specific modality and language needs with call takers:

IETF RFC 3841 [13] and IETF RFC 3840 [12] Languages attribute used in Contact header and Language attribute in Accept-Contact header in SIP according to examples in IETF RFC 4596 [i.3]. (AEC-ML-05).

IETF RFC 7852 [15] lang element in xCard for Subscriber's data in additional data in an emergency call. (AEC-ML-05).

IETF RFC 7852 [15] Data provider element can be from a Relay Provider or an Emergency Modality Translation or a Relay Provider including lang elements indicating languages used in the communication. (AEC-ML-05).

IETF RFC 8866 [17] a=lang sdp attribute on session and media level (AEC-ML-05).

9.4 Routing

The emergency communications enter the emergency communications network in an interface where the communication can be routed to a PSAP responsible for handling the communication.

Then routing to the most appropriate PSAP shall be performed using location information (AEC-RO-01), any type or subtype of emergency service indicated (AEC-RO-02), and relevant parts of the following accessibility related characteristics shall be enabled to influence the routing logic:

- Is video offered in the communication? (AEC-RO-08).
- Is preference for a specific sign language indicated? (AEC-RO-03).
- Is preference for a specific spoken or written language indicated? (AEC-RO-03).
- Is preference for RTT indicated? (AEC-RO-03).
- Is preference for RTT in a specific language indicated? (AEC-RO-03).
- Is a favourite specific support service indicated? (AEC-SS-01), (AEC-SS-02), (AEC-ML-08).
- Is competence for the preferred modality and language available in a PSAP in the region? (AEC-SS-05).
- Is a support service for the preferred modality and language known and available? ((AEC-SS-03), (AEC-ML-08).
- Is a call taker available with competence in the same spoken language as the support service? (AEC-RO-07).

Then route the communication to a suitable call taker and include if needed a support service and if needed a spoken language interpreter. (AEC-RO-01, (AEC-RO-02), (AEC-RO-03), (AEC-RO-04), (AEC-RO-05), (AEC-RO-06), (AEC-RO-07), (AEC-RO-08).

Else route to a call taker being the best match regarding media capability and language capability and have the call taker sort out best communications support in communication with the user in emergency. (AEC-RO-01), (AEC-RO-02), (AEC-RO-03), (AEC-RO-04), (AEC-RO-05), (AEC-RO-06), (AEC-RO-07), (AEC-RO-08).

Mapping procedures that are performed to route emergency communications are described in ETSI TS 103 479 [3]. To ensure international routing of emergency communications, a packet-switched infrastructure shall be operated to connect regions with each other. (AEC-RO-09).

Regional mapping functions shall be deployed in a hierarchical structure (tree-like) with a Forest Guide as the element to navigate between those trees. Regional Forest Guides shall maintain a LoST interface, as described in in ETSI TS 103 479 [3], clause 5.3. (AEC-RO-09).

9.5 Bridging

When the PSAP has access to multiparty bridging for audio, the bridge operations shall also include video and RTT when these media are used in the emergency communication. The multiparty bridging shall be used when needed for attended call transfer between call takers where the first call taker can stay as long as needed in the communication. The multiparty bridging shall also be possible to be used when needed for call taker training and monitoring, for including experts, for including first responders, support services, relay services and interpreters. (AEC-CT-01), (AEC-CT-02), (AEC-CO-01), (AEC-CO-02), (AEC-CO-03).

When a bridge is available, the bridge shall use IETF RFC 4579 [14] procedures for the multiparty operations and be able to mix media in ways suitable for presentation.

9.6 Call back

A PSAP shall have the possibility to call back to the equipment of the user in emergency, using the routable uri provided in the original emergency communications and the call control and media control procedures specified in ETSI TS 103 479 [3]. (AEC-SC-03), (AEC-CB-01), (AEC-CB-06).

The default action shall be to include the same media as in the original communication. (AEC-CB-02).

If a support service was included in the original communication, the default action shall be to include the same support service in the call back. (AEC-CB-04).

The call taker shall be provided with means to select other media in the call back than in the original communication. (AEC-CB-03).

The call taker shall be provided with means to exclude a support service which was included in the original communication and also to include other support services in the call back. (AEC-CB-05).

The PSAP shall include the SIP Priority header with value "psap-callback" in the callback, as defined in IETF RFC 7090 [18].

9.7 Communications handling

ETSI TS 103 479 [3] specifies communications handling for letting incoming emergency communications wait for handling by appropriate call takers. The functional accessibility requirements on the PSAP are specified in clause 5 of the present document.

In order to arrange for efficient answering on emergency communications from persons with various needs and capabilities, the following functionality should be supported:

Call takers to be assigned emergency communications with matching language preferences and capabilities. (AEC-RO-01), (AEC-RO-02), (AEC-RO-03), (AEC-RO-04), (AEC-RO-05), (AEC-RO-06), (AEC-RO-07), (AEC-RO-08), (AEC-RO-09).

Call takers to be assigned emergency communications with matching modality preferences and capabilities. (AEC-RO-01), (AEC-RO-02), (AEC-RO-03), (AEC-RO-04), (AEC-RO-05), (AEC-RO-06), (AEC-RO-07), (AEC-RO-08), (AEC-RO-09).

Call takers to identify when support services are needed in the communication and invoke such services. (AEC-CT-01), (AEC-CT-02), (AEC-SS-02), (AEC-SS-03), (AEC-ML-08).

Call takers to select a brief standard phrase to answer emergency communications. If a preferred language and modality is expressed in the incoming emergency communication, it is preferred that this greeting phrase is expressed in that language and modality. Otherwise, the phrase should be expressed in all media enabled in the established emergency communication and in any suitable language. (AEC-SC-07).

When an emergency communication is in a wait state, information about the wait state shall be sent in all media enabled in the communication. (AEC-SC-06).

9.8 Considerations for PSAP domains

9.8.1 General

When an emergency communication or an emergency callback communication is initiated, the actions specified in clause 9.8 of the present document shall be taken by the emergency communication networks and PSAPs handing the emergency communication.

NOTE: Some of the considerations are of extra importance when the user in emergency is visiting another region than the home region.

9.8.2 Forest Guide for routing to the responsible PSAP domain

When an emergency communication enters an emergency communication network, from a communication service using SIP, the emergency communication shall request information regarding routing to the region responsible for the location of the emergency from a Forest Guide. (AEC-RO-09).

NOTE: For emergency communications from IMS systems, this step is not needed but causes no error.

9.8.3 Spoken or written language competence

For efficient handling of the emergency communication, when the user and the call taker have no spoken or written language competence in common, the PSAP and call taker shall be provided with the ability to transfer the communication to a better matching call taker or to invoke a text relay service or language translation service to enable communication. (AEC-SS-01), (AEC-SS-02), (AEC-SS-03), (AEC-SS-04), (AEC-SS-05), (AEC-SS-06), (AEC-SS-08), (AEC-SS-09), (AEC-SS-10), (AEC-SS-11).

NOTE: The communication, which may involve the PSAP in a region and a text relay service in a user's home country, requires strict adherence to specified interface standards for successful interoperation.

9.8.4 Sign language handling

All PSAPs shall have capability to support video communication for sign language users and invoke proper sign language competence in the emergency communication. (AEC-SS-01), (AEC-SS-02), (AEC-SS-03), (AEC-SS-04), (AEC-SS-05), (AEC-SS-06), (AEC-SS-07).

NOTE: The user may have needs to use one specific sign language which requires video media support in the answering PSAP and language competence by either the call taker or a suitable sign language interpreter. Competence in sign languages among call takers is expected to be continued to be low, especially in foreign sign languages. Therefore, it is expected that most communications with sign language users will need support by a video relay service assigned to the user, and possibly also by a spoken language interpreter to match any gap in spoken language competence between the sign language interpreter and the call taker. For sign language emergency communications in visited countries, it should be preferred to route this kind of communication to an English speaking call taker to make language matching easier. The communication, involving the PSAP in the visited region and the video relay service in the user's home country requires strict adherence to specified interface standards for successful interoperation.

10 Supporting Services

Supporting services include relay services, language translating services and expert services, which may be invoked during emergency communications to support resolving the emergency.

The main method to invoke support services in the communication is to include it by automatic or manual means by the PSAP through one multimedia communication in a multiparty bridge. (AEC-SS-01), (AEC-SS-04), (AEC-SS-06), (AEC-SS-07), (AEC-SS-08), (AEC-SS-09), (AEC-SS-10), (AEC-SS-11), (AEC-SS-12), (AEC-SS-13), (AEC-CO-01), (AEC-ML-08).

When video is supported, total conversation shall be supported. (AEC-CM-12).

Any supporting service should include IETF RFC 7852 [15] Data provider element which can be coded as from a Relay Provider or an Emergency Modality Translation service including language elements indicating languages and modalities used in the communication. (AEC-SS-05).

The methods to invoke supporting services shall be by bridge operations as specified in ETSI TS 103 479 [3], clause 5.6. (AEC-SS-01), (AEC-SS-06), (AEC-ML-08).

A fallback method to invoke supporting services by the user is specified in clauses 7.2.6 and 7.3.5 in the present document. In that case the supporting service shall convey location information and other contextual data about the user and the emergency to the PSAP. (AEC-RS-01), (AEC-RS-02), (AEC-SC-05).

Annex A (informative): Relationship between the present document and the essential requirements of Directive (EU) 2019/882

Table A.1 presents the relationship between requirements of Directive (EU) 219/882 [i.10] and clauses in the present document.

Table A.1: Relationship between the present document and the essential requirements of Directive (EU) 2019/882

ETSI TS 103 919					
Requirement				**Requirement Conditionality**	
No	**Description**	**Essential requirements of Directive**	**Clause(s) of the present document**	**U/C**	**Condition**
1	Accessibility of user interface and functionality design of **products in general used for providing** emergency communications	Annex I; Section III; (a) for services of Article 2(2)(a) Referring to Annex I; Section I; (2) a-n	5.3, 7.1	U	
2	Accessibility of user interface and functionality of **terminals used for electronic communications services for emergency communication**	Annex I; Section III; (a) for services of Article 2(2)(a) Referring to Annex I; Section I; (2) o (iii)	5.3, 5.4, 7	U	
3	Providing **accessible information** about the services and products	Annex I; Section III; (b) for services of Article 2(2)(a)	5.8	U	
4	**support services** (help desks, call centres, technical support, **relay services** and training services) providing information on the accessibility of the service and its compatibility with assistive technologies, in accessible modes of communication.	Annex I; Section III; (c) for services of Article 2(2)(a)	5.8	U	
5	Electronic communications **services** including emergency communications **shall provide real time text in addition to voice**	Annex I; Section IV (a)(i)	5.5.2, 5.5.4, 8.2.1, 8.3.1	U	
6	Electronic communications **services** including emergency communications shall **provide total conversation where video is provided in addition to voice**	Annex I; Section IV (a)(ii)	5.5.6, 8.2.1	U	
7	Electronic communications **services** including emergency communications **shall synchronize media.**	Annex I; Section IV (a)(iii)	5.5, 8.2.1, 8.3.1	U	

ETSI TS 103 919					
Requirement				Requirement Conditionality	
No	**Description**	**Essential requirements of Directive**	**Clause(s) of the present document**	**U/C**	**Condition**
8	Electronic communications **services** including emergency communications **shall provide the emergency communications to the most appropriate PSAP**.	Annex I;Section IV (a)(iii)	5.4.3, 8, 9.4, 9.8.2	U	
9	Electronic communications **services** including emergency communications **shall ensure interoperability with assistive technologies.**	Annex I, Section IV (a)	5.3, 5.6, 5.7, 7.1, 10	U	
10	The emergency communications enables **two-way interactive communication** between the **end-user with disabilities and the PSAP**.	Annex I, Section IV (a) referring to Article 109 of Directive (EU) 2018/1972, amended by the Commission delegated regulation (EU) 2023/444 Article 4 (a)	5.4.2, 5.5, 5.6, 5.7, 7.2, 7.3, 8.2, 8.3, 8.4, 9.2, 9.3, 9.4, 9.5, 9.6, 9.7, 9.8, 10	U	
11	The emergency communications is **available in a seamless way,** without pre-registration, to end-users with disabilities **travelling in another Member State**.	Annex I, Section IV (a) referring to Article 109 of Directive (EU) 2018/1972, amended by the Commission delegated regulation (EU) 2023/444 Article 4 (b)	5.4.2, 5.4.3, 7.3.5, 8.2.5, 8.3.5, 9.4, 9.8.2	U	
12	The emergency communications is provided to end-users with disabilities **free of charge**.	Annex I, Section IV (a) referring to Article 109 of Directive (EU) 2018/1972, amended by the Commission delegated regulation (EU) 2023/444 Article 4 (c)	5.4.7, 8.1	U	
13	The emergency communications is routed without delay to the **most appropriate PSAP that is qualified and equipped to appropriately answer** and process the emergency communications from end-users with disabilities	Annex I, Section IV (a) referring to Article 109 of Directive (EU) 2018/1972, amended by the Commission delegated regulation (EU) 2023/444 Article 4 (d)	5.4.2, 5.4.3, 5.4.2, 5.4.9, 9	U	
14	Equivalent levels of accuracy and reliability of **caller location information** are ensured for the emergency communications for end-users with disabilities as for emergency calls by other end-users.	Annex I, Section IV (a) referring to Article 109 of Directive (EU) 2018/1972, amended by the Commission delegated regulation (EU) 2023/444 Article 4 (e)	5.4.2, 5.4.3, 7.2.3, 7.3.3, 8.2.3, 8.3.3, 8.4,	U	
15	Member states shall ensure that emergency communications and caller location information are **routed without delay to the most appropriate PSAP**	Annex I, Section IV (a) referring to Article 109 of Directive (EU) 2018/1972, amended by the Commission delegated regulation (EU) 2023/444 Article 5	5.4.3, 9.4, 9.8.2	U	

ETSI TS 103 919					
Requirement				Requirement Conditionality	
No	**Description**	**Essential requirements of Directive**	**Clause(s) of the present document**	**U/C**	**Condition**
16	**Answering emergency communications** shall include functions and **procedures for persons with disabilities**	Annex I; Section V, first paragraph	9.3, 9.4, 10	U	
17	**Answering** emergency communications shall use **the same means as in the received** communications.	Annex I; Section V, second paragraph	5.4.2, 9.2, 9.4	U	
18	Emergency communications to the single European emergency number '112' shall be **appropriately answered.**	Annex I; Section V, second paragraph	5.2, 9.4, 9.7	U	
19	Emergency communications to the single European emergency number '112' shall be appropriately answered, **by the most appropriate PSAP**	Annex I; Section V, second paragraph	5.4.3, 9.7, 9.8	U	
20	Emergency communications to the single European emergency number '112' shall be appropriately answered**, by using synchronized voice and text (including real time text.**	Annex I; Section V, second paragraph	5.5.2, 5.5.4, 5.5.5, 9.2	U	
21	Emergency communications to the single European emergency number '112' shall be appropriately answered, **where video is provided, voice, text (including real time text) and video**	Annex I; Section V, second paragraph	5.5.3, 9.2	U	
22	Emergency communications to the single European emergency number '112' shall be appropriately answered, where video is provided, voice, text (including real time text) and **video synchronized as total conversation.**	Annex I; Section V, second paragraph	5.5.6, 9.2	U	

Key to columns:

Requirement:

No A unique identifier for one row of the table which may be used to identify a requirement.

Description A textual reference to the requirement.

Essential requirements of Directive

Identification of article(s) defining the requirement in the Directive.

Clause(s) of the present document

Identification of clause(s) defining the requirement in the present document unless another document is referenced explicitly.

Requirement Conditionality:

U/C Indicates whether the requirement is unconditionally applicable (U) or is conditional upon the manufacturer's claimed functionality of the equipment (C).

Condition Explains the conditions when the requirement is or is not applicable for a requirement which is classified "conditional".

Annex B (normative): Testing for compliance with technical requirements

B.1 Introduction

This annex contains text cases corresponding to the requirement clauses in the main body of the present document.

For ease of reference, the clause numbering is in line with the clause numbering for the corresponding requirements. Therefore there are empty clauses in this annex corresponding to clauses where no requirements are expressed.

B.2 Specific concerns when testing emergency communications

When testing emergency communications, it is of importance that the tests are done in a way that they do not cause excessive load on the PSAP. Whenever possible, the test method described in ETSI TS 103 479 [3], clause 6.1.2.10 should be used. When test communications which engage personnel in an operational PSAPs are planned, this should be in agreement with the target PSAP.

B.3 Testing with and without technical interface observations

Where appropriate, the tests are divided in two sections, one with pure human actions and observations in user interfaces, called "interoperability section", and another adding observations in technical interfaces in the systems, called "conformance section". The intention with this division is that it shall be possible to vary the ambition level of the tests and thereby the complexity to perform them.

The additional test in the conformance sections can be omitted during testing when so found suitable.

To ease handling a test protocol, the numbering of the preconditions and the procedure steps continue from the interoperability sections into the conformance sections.

The tests are intended to be possible to perform both in test environments and in implemented services.

B.4 Operational scenarios for testing

B.4.1 Introduction of operational scenarios for testing

Tests defined in the present document shall be carried out at representative points within the boundary limits of the operational profile of the products and services under test defined by its intended use, which, as a minimum, shall be that specified in the applicable operational scenarios for testing contained in clause B.4.2 of the present document.

Where technical performance of the products or services under test can be expected to vary depending on the selected operational scenario, tests shall be carried out under a sufficient variety of operational scenarios as specified in the present clause, to give confidence of compliance with the applicable technical requirements. The variations for video support are included in the test specifications, while the variations in other aspects of operational scenarios are covered by explicit test clauses.

Testing may be performed in variations of the provided test scenarios as motivated by any reason appearing during testing.

B.4.2 Operational scenarios for testing

B.4.2.1 Scenarios for testing of user terminals

When the product under test is or contains a user terminal, the tests shall be performed at least in the applicable of the following operational scenarios:

1) In home country: The user terminal is in the home country of the user or originating service, whichever scenario provides.
2) In visited country.
3) Including video.
4) Not including video.
5) Initiating communication first with relay service.

B.4.2.2 Scenarios for testing of originating services

When the ICT under test is or contains an electronic communications service, the tests shall be performed in the applicable of the following operational scenarios:

1) User is in home country.
2) User is visiting in the service.
3) User is visiting other country.
4) Video included.
5) Video not included.

B.4.2.3 Scenarios for testing of emergency communication networks and PSAPs

When the ICT under test is or contains a PSAP, the tests shall be performed in the applicable of the following operational scenarios:

1) User is in home country.
2) User is visiting other country.
3) Video included.
4) Video not included.

B.4.2.4 Scenarios for testing of relay services

When the product under test is or contains a relay service, the tests shall be performed in the applicable of the following operational scenarios:

1) User initiates communication with relay services first.
2) PSAP includes relay service automatically.
3) PSAP includes relay service manually.
4) Video included.
5) Video not included.

B.4.2.5 Scenarios for testing of other supporting services

When the product under test is or contains another supporting service, the tests shall be performed in the applicable of the following operational scenarios:

1) PSAP includes supporting service automatically.
2) PSAP includes supporting service manually.
3) Video included.
4) Video not included.

B.5 Accessible Emergency Communication

B.5.1 General

There are no tests for this clause.

B.5.2 Modality and Language

There are no tests for this clause.

B.5.3 User Interface and General Accessibility

There are no tests for this clause.

B.5.4 Communication Features

There are no tests for this clause.

B.5.5 Communication Media

There are no tests for this clause.

B.5.6 Relay Service Invocation by the User in Emergency

There are no tests for this clause.

B.5.7 Supporting Services

There are no tests for this clause.

B.5.8 Documentation

Interoperability section	
Precondition	1. The ICT under test has a role in provision of accessible emergency communication.
Procedure	Check 1: that documentation about the accessibility of the ICT under test is provided. Check 2: that the provided documentation about the ICT under test is accessible.
Result	Pass: Checks 1 and 2 are true. Fail: Check 1 or 2 is false. Not applicable: Pre-condition 1 is not met.

B.6 Void

There are no tests for this clause.

B.7 User equipment

B.7.1 General

The clause has no testable requirements.

B.7.2 IMS Based User Equipment

B.7.2.1 General

The test is identical with clause B.7.2.3.

B.7.2.2 Settings

Interoperability section	
Precondition	1. The ICT under test is user equipment for IMS MTSI electronic communication including SIP for call control and RTP [11] for voice and RTT. 2. Video is included if it is supported by the user equipment. 3. An IMS communication service is available where the equipment under test resides.
Procedure	1. Check the user interface for a setting of preference to include RTT in all calls or not. 2. Set this setting on. 3. Check the user interface for a setting of preferred language in emergency calls. 4. Check that a list of spoken/written languages is available for selection. 5. Select a language. 6. Initiate a test emergency communication with a PSAP. 7. Check that the emergency communication is received and answered and a communication is set up, and that RTT is included from the beginning. 8. Check that video is included if it is supported by the user equipment. 9. Check that basic communication is possible in all supported media. 10. Check with the PSAP that the language preference is visible in the PSAP user interface.
Result	Pass: Checks 1, 3, 4, 7, 8, 9, 10 are true. Fail: Check 1 or 3 or 4 or 7 or 8 or 9 is false. Not applicable: Pre-condition 1 or 3 is not met.

Conformance section	
Preconditions	4. The equipment under test or the communication service is set in a mode where it records traces of the communication in SIP session control and RTP [11] media and so that it can be analysed.
Procedure	11. Check the communication trace and analyse the SIP and RTP [11] parts to verify the following: a) That the INVITE is formed according to ETSI TS 126 114 [9] requirements, and specifically that IETF RFC 7852 [15] additional data with subscriber info language indication is included and matches the setting. b) That RTT is offered with a media specification similar to this example (but ports and payload type numbers may vary): m=text 11000 RTP/AVP 100 98 a=rtpmap:98 t140/1000 a=fmtp:98 cps=90 a=rtpmap:100 red/1000 a=fmtp:100 98/98/98 a=rtt-mixer indicating multiparty support and use of two-fold redundancy. c) Video is offered, if supported. d) A SIP level media feature tag "text" is included in the Contact header. e) That audio is offered with at least one wide-band codec supported according to ETSI TS 103 479 [3]. Check that the OK from the PSAP confirms the codecs supported by the user equipment.
Result	Pass: Checks 10, 11 are true. Fail: Check 10 or 11 is false.

B.7.2.3 Initiation of emergency communications

Interoperability section	
Precondition	1. The ICT under test is user equipment for electronic communication in the IMS mobile environment SIP for call control and voice and RTT. 2. Video media is enabled in the user equipment, if it is supported by the user equipment. 3. An IMS MTSI communication service is available where the equipment under test resides.
Procedure	1. Initiate an emergency communication with RTT voice and also video, if video is supported by the user equipment. 2. Check that the IMS MTSI electronic communications service initiates communication with a PSAP. 3. Check that the emergency communication is received and answered and a communication is set up. 4. Check that basic communication is possible in all media supported by the user equipment.
Result	Pass: Checks 2, 3 and 4 are true. Fail: Check 2 or 3 or 4 is false. Not applicable: Pre-condition 1 or 3 is not met.

Conformance section	
Preconditions	4. The equipment under test or the communication service is set in a mode where it records traces of the communication in SIP session control and RTP [11] media and so that it can be analysed.
Procedure	5. Check the communication trace and analyse the SIP and RTP [11] parts to verify the following: a) That the INVITE is formed according to ETSI TS 126 114 [9] requirements, and specifically that the SIP request URI contains a service URN (e.g. urn:service:sos or any subservice). b) That RTT is offered with a media specification similar to this example (but ports and payload type numbers may vary): m=text 11000 RTP/AVP 100 98 a=rtpmap:98 t140/1000 a=fmtp:98 cps=90 a=rtpmap:100 red/1000 a=fmtp:100 98/98/98 a=rtt-mixer indicating multiparty support and use of two-fold redundancy. c) That video is offered, if it is supported by the user equipment. d) That audio is offered with at least one wide-band codec supported. 6. Check that the OK from the PSAP confirms the codecs supported by the user equipment. 7. Check that RTT text packets contain original and two redundant transmissions of text.
Result	Pass: Checks 5, 6, 7 are true. Fail: Check 5 or 6 or 7 is false.

B.7.2.4 Call back

Interoperability section	
Precondition	1. The ICT under test is user equipment for electronic communication in IMS MTSI service including SIP for call control and voice and RTT media. 2 Video is included, if supported by the communications service. 3. An IMS MTSI communication service is available where the equipment under test resides.
Procedure	1. Initiate an emergency communication with RTT and voice. Include video, if supported by the user equipment. 2. Check that the IMS electronic communications service initiates communication with a PSAP. 3. Check that the emergency communication is received and answered and a communication is set up. 4. Check that basic communication is possible in voice and RTT. 5. Check that video communication is included and working if it is supported by the user equipment. 6. Ask the call taker to disconnect and call back. 7. Check that the call back is received and can be answered. 8. Check that the same media can be used as what was available in the initial emergency communication.
Result	Pass: Checks 2-5, 7-8 are true. Fail: Check 2 or 3 or 4 or 5 or 7 or 8 is false. Not applicable: Pre-condition 1 or 3 is not met.

Conformance section	
Preconditions	4. The equipment under test or the communication service is set in a mode where it records traces of the communication in SIP session control and RTP [11] media and so that it can be analysed.
Procedure	9. Check the communication trace and analyse the SIP and RTP [11] parts to verify the following: a) That the INVITE of the call back is formed according to ETSI TS 126 114 [9] requirements, and specifically that the mark "psap call back" for emergency call back is set in the priority SIP header. b) That RTT is offered with a media specification similar to this example (but ports and payload type numbers may vary): m=text 11000 RTP/AVP 100 98 a=rtpmap:98 t140/1000 a=fmtp:98 cps=90 a=rtpmap:100 red/1000 a=fmtp:100 98/98/98 a=rtt-mixer indicating multiparty support and use of two-fold redundancy. c) That audio is offered with at least one wide-band codec supported according to ETSI TS 103 479 [3]. d) That video is offered, if it was included in the initial communication with at least one video codec supported according to ETSI TS 103 479 [3]. 10. Check that the OK from the device under test confirms the audio and RTT codecs. 11. Check that the OK from the device under test confirms the video codec, if video was included in the initial emergency communication. 12. Check that RTT text packets contain original and two redundant transmissions of text.
Result	Pass: Checks 9, 10, 11, 12 are true. Fail: Check 9 or 10 or 11 or 12 is false.

B.7.2.5 Visiting regions and networks

Interoperability section	
Precondition	1. The ICT under test is user equipment for electronic communication in the IMS mobile environment SIP for call control and voice, RTT and video media. 2. An IMS MTSI communication service is available where the equipment under test resides. 3. The equipment under test is located where it is roaming in another IMS service.
Procedure	1. Initiate an emergency communication with RTT and voice. Include video, if supported by the ICT under test. 2. Check that the emergency communication is conveyed by the visited IMS MTSI electronic communications service and initiates communication with a PSAP. 3. Check that the emergency communication is received and answered and a communication is set up. 4. Check that basic communication is possible in RTT and video, and in video, if video is supported by the user equipment.
Result	Pass: Checks 2, 3 and 4 are true. Fail: Check 2 or 3 or 4 is false. Not applicable: Pre-condition 1 or 2 or 3 is not met.

Conformance section	
Preconditions	4. The equipment under test or the communication service is set in a mode where it records traces of the communication in SIP session control and RTP [11] media and so that it can be analysed.
Procedure	5. Check the communication trace and analyse the SIP and RTP [11] parts to verify the following: a) That the INVITE is formed according to ETSI TS 126 114 [9] and ETSI TS 124 229 [8], clauses 4.7 and 5.1.6 requirements, and specifically that the SIP request URI contains a service URN (e.g. urn:service:sos or any subservice). b) That RTT is offered with a media specification similar to this example (but ports and payload type numbers may vary): m=text 11000 RTP/AVP 100 98 a=rtpmap:98 t140/1000 a=fmtp:98 cps=90 a=rtpmap:100 red/1000 a=fmtp:100 98/98/98 a=rtt-mixer indicating multiparty support and use of two-fold redundancy. c) That audio is offered with at least one wide-band codec supported. d) That if video is supported by the user equipment, video is offered with at least one of the codecs supported by ETSI TS 103 479 [3]. 6. Check that the OK from the PSAP confirms these codecs. 7. Check that RTT text packets contain original and two redundant transmissions of text.
Result	Pass: Checks 5, 6, 7 are true. Fail: Check 5 or 6 or 7 is false.

B.7.2.6 Invocation of support services by the user

Interoperability section	
Precondition	1. The ICT under test is user equipment in IMS for electronic communication including SIP for call control and voice and RTT media. 2. An IMS communication service is available where the equipment under test resides.
Procedure	1. Initiate an emergency communication with RTT and voice to a text relay service. 2. Ask the relay service to initiate emergency communication. 3. Check that the relay service asks the user for location or gets location information automatically. 4. Check that the relay service initiates communication with a PSAP. 5. Check that the emergency communication is answered and a communication is set up. 6. Check that basic communication is possible in voice and RTT media. 7. Check by asking, that the call taker has got location information.
Result	Pass: Checks 1-7 are true. Fail: Check 2 or 3 or 4 or 5 or 6 or 7 is false. Not applicable: Pre-condition 1 or 2 is not met.

Conformance section	
Preconditions	3. The equipment under test or the communication service is set in a mode where it records traces of the communication in SIP session control and RTP [11] media and so that it can be analysed.
Procedure	8. Check the communication trace and analyse the SIP and RTP [11] parts to verify the following in the user- to -relay service communication. a) That RTT is offered with a media specification similar to this example (but ports and payload type numbers may vary): m=text 11000 RTP/AVP 100 98 a=rtpmap:98 t140/1000 a=fmtp:98 cps=90 a=rtpmap:100 red/1000 a=fmtp:100 98/98/98 a=rtt-mixer indicating multiparty support and use of two-fold redundancy. b) That audio is offered with at least one wide-band codec supported according to ETSI TS 103 479 [3] 9. Check that the OK from the relay service confirms these codecs. 10. Check that RTT text packets contain original and two redundant transmissions of text.
Result	Pass: Checks 8, 9, 10 are true. Fail: Check 8 or 9 or 10 is false.

B.7.3 SIP Based User Equipment

B.7.3.1 General

The test is identical to clause B.7.3.3

B.7.3.2 Settings

Interoperability section	
Precondition	1. The ICT under test is user equipment for electronic communication including SIP for call control and RTP [11] for voice, and RTT media. Video is included if it is supported by the user equipment. 2. A communication service is available where the equipment under test resides.
Procedure	1. Check the user interface for a setting of preference to include RTT in all calls or not. 2. Set this setting on. 3. Check the user interface for a setting of preferred language in emergency calls. 4. Check that a list spoken/written languages are available for selection. 5. Select and set one language. 6. Initiate a test emergency communication with a PSAP. 7. Check that the emergency communication is received and answered and a communication is set up and that RTT is included from the beginning. 8. Check that basic communication is possible in RTT and voice. 9. Check with the PSAP that the language preference is visible in the PSAP user interface.
Result	Pass: Checks 1, 3, 4, 7, 8, 9 are true. Fail: Check 1 or 3 or 4 or 7 or 8 or 9 is false. Not applicable: Pre-condition 1 or 2 is not met.

Conformance section	
Precondition	3. The equipment under test or the communication service is set in a mode where it records traces of the communication in SIP session control and RTP [11] media and so that it can be analysed.
Procedure	10. Check the communication trace and analyse the SIP and RTP [11] parts to verify the following: a) That the INVITE is formed according to ETSI TS 103 479 [3] requirements, and specifically that IETF RFC 7852 [15] additional data with subscriber info language indication is included and matches the setting. b) That RTT is offered with a media specification similar to this example (but ports and payload type numbers may vary): m=text 11000 RTP/AVP 100 98 a=rtpmap:98 t140/1000 a=fmtp:98 cps=90 a=rtpmap:100 red/1000 a=fmtp:100 98/98/98 a=rtt-mixer indicating multiparty support and use of two-fold redundancy. c) Text is included as a Media tag in the Accept-contact SIP field. d) That audio is offered with at least one wide-band codec supported according to ETSI TS 103 479 [3]. 11. Check that the OK from the PSAP confirms these codecs.
Conformance result	Pass: Checks 10, 11 are true. Fail: Check 10 or 11 is false. Not applicable: Precondition 3 was not met.

B.7.3.3 Initiation of emergency communications

Interoperability section	
Precondition	1. The ICT under test is user equipment for electronic communication including SIP for call control and voice and RTT. 2. Video is included if supported by the user equipment. 3. A communication service is available where the equipment under test resides.
Procedure	1. Initiate an emergency communication with RTT and voice. Include video if video is supported by the user equipment. 2. Check that the electronic communications service initiates communication with a PSAP. 3. Check that the emergency communication is received and answered and a communication is set up. 4. Check that basic communication is possible in voice and audio and in video if video is supported by the user equipment.
Result	Pass: Checks 2, 3 and 4 are true. Fail: Check 2 or 3 or 4 is false. Not applicable: Pre-condition 1 or 3 is not met.

Conformance section	
Precondition	4. The equipment under test or the communication service is set in a mode where it records traces of the communication in SIP session control and RTP [11] media and so that it can be analysed.
Procedure	5. Check the communication trace and analyse the SIP and RTP [11] parts to verify the following: a) That the INVITE is formed according to ETSI TS 103 479 [3] requirements, and specifically that the SIP request URI contains a service URN (e.g. urn:service:sos or any subservice). b) That video is offered with at least one codec according to ETSI TS 103 479 [3] requirements if video is supported by the user equipment. c) That RTT is offered with a media specification similar to this example (but ports and payload type numbers may vary): m=text 11000 RTP/AVP 100 98 a=rtpmap:98 t140/1000 a=fmtp:98 cps=90 a=rtpmap:100 red/1000 a=fmtp:100 98/98/98 a=rtt-mixer indicating multiparty support and use of two-fold redundancy. d) That audio is offered with at least one wide-band codec supported according to ETSI TS 103 479 [3]. 6. Check that the OK from the PSAP confirms the voice and RTT codecs and also the video codec if the user equipment supports video. 7. Check that RTT text packets contain original and two redundant transmissions of text.
Result	Pass: 5 and 6 and 7 are true. Fail: 5 or 6 or 7 is false.

B.7.3.4 Call back

Interoperability section	
Precondition	1. The ICT under test is user equipment for electronic communication including SIP for call control and voice and RTT. 2. Video is included in the test if supported by the user equipment. 3. A communication service is available where the equipment under test resides.
Procedure	1. Initiate an emergency communication with RTT and voice. Video is included if supported by the user equipment. 2. Check that the electronic communications service initiates communication with a PSAP. 3. Check that the emergency communication is received and answered and a communication is set up. 4. Check that basic communication is possible in voice and RTT media, and in video if video is supported by the user equipment. 5. Ask the call taker to disconnect and call back. 6. Check that the call back is received and can be answered. 7. Check that the same media can be used in the call back as in the initial emergency communication.
Result	Pass: Checks 1-7 are true. Fail: Check 2 or 3 or 4 or 5 or 6 or 7 is false. Not applicable: Pre-condition 1 or 3 is not met.

Conformance section	
Preconditions	3. The equipment under test or the communication service is set in a mode where it records traces of the communication in SIP session control and RTP [11] media and so that it can be analysed.
Procedure	8. Check the communication trace and analyse the SIP and RTP [11] parts to verify the following: a) That the INVITE of the call back is formed according to ETSI TS 103 479 [3] requirements, and specifically that the mark "psap-callback" for emergency call back is set in the priority SIP header. b) That if video is supported by the user equipment, video is offered with at least one codec according to ETSI TS 103 479 [3] requirements. c) That RTT is offered with a media specification similar to this example (but ports and payload type numbers may vary) m=text 11000 RTP/AVP 100 98 a=rtpmap:98 t140/1000 a=fmtp:98 cps=90 a=rtpmap:100 red/1000 a=fmtp:100 98/98/98 a=rtt-mixer indicating multiparty support and use of two-fold redundancy. d) That audio is offered with at least one wide-band codec supported according to ETSI TS 103 479 [3]. 9. Check that the OK from the device under test confirms the supported codecs. 10. Check that RTT text packets contain original and two redundant transmissions of text.
Result	Pass: Checks 8, 9, 10 are true. Fail: Check 8 or 9 or 10 is false.

B.7.3.5 Invocation of support services by the user

This is the non-preferred method to initiate emergency communications via a relay service. The preferred way is that the user initiates emergency communications and the PSAP invokes a relay service if needed.

Interoperability section	
Precondition	1. The ICT under test is user equipment for electronic communication including SIP for call control and voice, RTT and video media. 2. A communication service is available where the equipment under test resides.
Procedure	1. Initiate an emergency communication with video RTT and voice to a video relay service. 2. Ask the relay service to initiate emergency communication. 3. Check that the relay service asks the user for location. 4. Check that the relay service initiates communication with a PSAP. 5. Check that the emergency communication is answered and a communication is set up. 6. Check that basic communication is possible in all three media. 7. Check by asking, that the call taker has got location information.
Result	Pass: Checks 1-7 are true. Fail: Check 2 or 3 or 4 or 5 or 6 or 7 is false. Not applicable: Pre-condition 1 or 2 is not met.

Conformance section	
Preconditions	3. The equipment under test or the communication service is set in a mode where it records traces of the communication in SIP session control and RTP [11] media and so that it can be analysed.
Procedure	8. Check the communication trace and analyse the SIP and RTP [11] parts to verify the following in the user-to-relay service communication. a) That video is offered with at least one codec according to ETSI TS 103 479 [3] requirements. b) That RTT is offered with a media specification similar to this example (but ports and payload type numbers may vary): m=text 11000 RTP/AVP 100 98 a=rtpmap:98 t140/1000 a=fmtp:98 cps=90 a=rtpmap:100 red/1000 a=fmtp:100 98/98/98 a=rtt-mixer indicating multiparty support and use of two-fold redundancy. c) That audio is offered with at least one wide-band codec supported according to ETSI TS 103 479 [3]. 9. Check that the OK from the relay service confirms these codecs. 10. Check that RTT text packets contain original and two redundant transmissions of text.
Result	Pass: Checks 8, 9, 10 are true. Complete test fail: Check 8 or 9 or 10 is false.

B.7.4 Other technologies for emergency communication

If other technologies than IMS or SIP are used within the electronic communication services, the communication shall be converted to ETSI TS 103 479 [3] format for the interface to the emergency communication network. Technical test procedures for the end-user equipment are out of scope for the present document.

B.8 Originating Service

B.8.1 General

Requirement for no charging for the accessible emergency communication.

Interoperability section	
Precondition	1. The ICT under test is an electronic communication service providing emergency communications.
Procedure	1. Agree with a PSAP to initiate an emergency communication. 2. Perform an emergency communication with accessibility features via the ICT under test. 3. Check the coming monthly charge for the subscription used for the emergency communication that no charge for the emergency communication is specified.
Result	Pass: Checks 1 is true. Fail: Check 1 is false. Not applicable: Pre-condition 1 is not met.

B.8.2 IMS Based Originating Service

B.8.2.1 General

The clause has no testable requirements.

B.8.2.2 Settings

Interoperability section	
Precondition	1. The ICT under test is an IMS electronic communication service using SIP for call control and voice and RTT media. 2. If the service supports test, video is included in the test. 3. An IMS MTSI equipment for test is available registered in the service.
Procedure	1. Find the setting in the end user equipment for a preference to include RTT in all calls. 2. Set this setting on. 3. Find in the user interface of the end user equipment a setting of preferred language in emergency calls. 4. Find a list of spoken/written languages available for selection. 5. Select one language. 6. Initiate a test emergency communication with a PSAP. 7. Check that the emergency communication is received and answered and a communication is set up and that RTT is included from the beginning. 8. Check that basic communication is possible in RTT and voice media. 9. Check that, if video is supported, video is also included in the emergency communication. 10. Check with the PSAP that the language preference is visible in the PSAP user interface.
Result	Pass: Checks 1, 3, 4, 7, 8, 9, 10 are true. Fail: Check 1 or 3 or 4 or 7 or 8 or 9 or 10 is false. Not applicable: Pre-condition 1 or 3 is not met.

Conformance section	
Preconditions	4. The communication service is set in a mode where it records traces of the communication in the interface with PSAP in SIP session control and RTP [11] media and so that it can be analysed.
Procedure	11. Check the communication trace of the interface to the emergency network or PSAP and analyse the SIP and RTP [11] parts to verify the following: a) That the INVITE is formed according to ETSI TS 103 479 [3] requirements, and specifically that IETF RFC 7852 [15] additional data with subscriber info language indication is included and matches the setting. b) That RTT is offered with a media specification similar to this example (but ports and payload type numbers may vary): m=text 11000 RTP/AVP 100 98 a=rtpmap:98 t140/1000 a=fmtp:98 cps=90 a=rtpmap:100 red/1000 a=fmtp:100 98/98/98 a=rtt-mixer indicating multiparty support and use of two-fold redundancy. c) Text is included as a SIP level Media tag. d) That audio is offered with at least one wide-band codec supported according to ETSI TS 103 479 [3]. e) That, if video is supported by the communications service, video is offered according to ETSI TS 103 479 [3]. 12. Check that the OK from the PSAP confirms audio and RTT codecs, and video codec if video is supported by the communication service.
Result	Pass: Checks 11, 12 are true. Fail: Check 11 or 12 is false.

B.8.2.3 Initiation of emergency communications

Interoperability section	
Precondition	1. The ICT under test is an IMS electronic communication service using SIP for call control and voice and RTT. 2. Video is included in the test if video is supported by the communications service. 3. An IMS MTSI equipment for test is available registered in the service.
Procedure	1. Initiate an emergency communication with RTT and voice from the test equipment. 2. Check that the IMS MTSI electronic communications service initiates communication with a PSAP. 3. Check that the emergency communication is received and answered and a communication is set up. 4. Check that basic communication is possible in voice and RTT, and also in video if video is supported by the communications service.
Result	Pass: Checks 2, 3 and 4 are true. Fail: Check 2 or 3 or 4 is false. Not applicable: Pre-condition 1 or 3 is not met.

Conformance section	
Preconditions	3. The communication service is set in a mode where it records traces of the communication in the interface with PSAPs in SIP session control and RTP [11] media and so that it can be analysed.
Procedure	5. Check the communication trace and analyse the SIP and RTP [11] parts to verify the following: a) That the INVITE is formed according to ETSI TS 103 479 [3] requirements (by conversion as specified in ETSI TS 123 167 [7]), and specifically that the SIP request URI contains a service URN (e.g. urn:service:sos or any subservice). b) That RTT is offered with a media specification similar to this example (but ports and payload type numbers may vary): m=text 11000 RTP/AVP 100 98 a=rtpmap:98 t140/1000 a=fmtp:98 cps=90 a=rtpmap:100 red/1000 a=fmtp:100 98/98/98 a=rtt-mixer indicating multiparty support and use of two-fold redundancy. c) That audio is offered with at least one wide-band codec supported. d) That if video is supported by the communications service, video is offered with at least one codec. 6. Check that the OK from the PSAP confirms voice and RTT, and also video if video is supported by the communications service. 7. Check that RTT text packets contain original and two redundant transmissions of text.
Result	Pass: Checks 5, 6, 7 are true. Fail: Check 5 or 6 or 7 is false.

B.8.2.4 Call back

Interoperability section	
Precondition	1. The ICT under test is an IMS electronic communication service using SIP for call control and voice and RTT media. 2. Video is included in the test if video is supported by the communications service. 3. An IMS MTSI equipment for test is available registered in the service.
Procedure	1. Initiate an emergency communication with RTT and voice from the IMS test user equipment, and include also video if video is supported by the communications service. 2. Check that the IMS electronic communications service initiates communication with the expected PSAP. 3. Check that the emergency communication is received and answered and a communication is set up. 4. Check that basic communication is possible in voice, and RTT media and also in video if video is supported by the communications service. 5. Ask the call taker to disconnect and call back. 6. Check that the call back is received and can be answered. 7. Check that the same media as in the initial communication can be used.
Result	Pass: Checks 2, 3, 4, 6, 7 are true. Fail: Check 2 or 3 or 4 or 5 or 6 or 7 is false. Not applicable: Pre-condition 1 or 3 is not met.

Conformance section	
Preconditions	3. The communication service is set in a mode where it records traces of the communication in the interface with PSAP in SIP session control and RTP [11] media and so that it can be analysed.
Procedure	8. Check the communication trace and analyse the SIP and RTP [11] parts to verify the following: a) That the INVITE of the call back is formed according to ETSI TS 103 479 [3] requirements, and specifically that the mark "psap-callback" for emergency call back is set in the priority SIP header. b) That RTT is offered with a media specification similar to this example (but ports and payload type numbers may vary): m=text 11000 RTP/AVP 100 98 a=rtpmap:98 t140/1000 a=fmtp:98 cps=90 a=rtpmap:100 red/1000 a=fmtp:100 98/98/98 a=rtt-mixer indicating multiparty support and use of two-fold redundancy. c) That audio is offered with at least one wide-band codec supported according to ETSI TS 103 479 [3]. d) That if video is supported in the communications service, video is offered with at least one video codec supported according to ETSI TS 103 479 [3]. 9. Check that the OK from the device under test confirms the audio and RTT codecs, and also the video codec if video is supported by the communications service. 10. Check that RTT text packets contain original and two redundant transmissions of text.
Result	Pass: Checks 8, 9, 10 are true. Fail: Check 8 or 9 or 10 is false.

B.8.2.5 Visiting regions and networks

Interoperability section	
Precondition	1. The ICT under test is an IMS electronic communication service using SIP for call control and voice and RTT media. 2. Video is also included in the test if video is supported by the communications service. 3. An IMS MTSI equipment for test from another IMS service is available roaming in the service.
Procedure	1. Initiate an emergency communication with RTT and voice. Include also video if video is supported by the home communications service. 2. Check that the emergency communication is conveyed by the visited IMS electronic communications service and initiates communication with the expected PSAP. 3. Check that the emergency communication is received and answered and a communication is set up. 4. Check that basic communication is possible in voice and RTT, and also in video if video is supported by the home communication service.
Result	Pass: Checks 2, 3 and 4 are true. Fail: Check 2 or 3 or 4 is false. Not applicable: Pre-condition 1 or 3 is not met.

Conformance section	
Preconditions	4. The IMS communication service is set in a mode where it records traces of the communication in the interface with the PSAP in SIP session control and RTP [11] media and so that it can be analysed.
Procedure	5. Check the communication trace and analyse the SIP and RTP [11] parts to verify the following: a) That the INVITE is formed according to ETSI TS 103 479 [3] (by conversion from ETSI TS 126 114 [9] and ETSI TS 124 229 [8], clauses 4.7 and 5.1.6 requirements converted as specified in ETSI TS 123 167 [7]), and specifically that the SIP URI is a service urn using the top-level service label 'sos'. b) That RTT is offered with a media specification similar to this example (but ports and payload type numbers may vary): m=text 11000 RTP/AVP 100 98 a=rtpmap:98 t140/1000 a=fmtp:98 cps=90 a=rtpmap:100 red/1000 a=fmtp:100 98/98/98 a=rtt-mixer indicating multiparty support and use of two-fold redundancy. c) That audio is offered with at least one wide-band codec supported. d) That if video is supported by the home communications service, video is offered with at least one video codec. 6. Check that the OK from the PSAP confirms the voice and RTT codecs and also the video codec, if supported by the home communication service. 7. Check that RTT text packets contain original and two redundant transmissions of text.
Result	Pass: Checks 5, 6, 7 are true. Fail: Check 5 or 6 or 7 is false.

B.8.3 SIP Based Originating Service

B.8.3.1 General

The clause has no testable requirements.

B.8.3.2 Settings

Interoperability section	
Precondition	1. The ICT under test is an electronic communication service using SIP for call control and voice and RTT media. 2. Video is also included in the test, if video is supported by the communications service. 3. An end user equipment for test is available registered in the service.
Procedure	1. Find the setting in the end user equipment for a preference to include RTT in all calls. 2. Set this setting on. 3. Find in the user interface of the end user equipment a setting of preferred language in emergency calls. 4. Find a list of sign languages and spoken/written languages available for selection. 5. Select a language. 6. Initiate a test emergency communication with a PSAP. 7. Check that the emergency communication is received and answered and a communication is set up and that RTT is included from the beginning. 8. Check that basic communication is possible in voice and RTT media and that video communication is possible, if video is supported by the communications service. 9. Check with the PSAP that the language preference is visible in the PSAP user interface.
Result	Pass: Checks 1, 3, 4, 7, 8, 9 are true. Fail: Check 1 or 3 or 4 or 7 or 8 or 9 is false. Not applicable: Pre-condition 1 or 3 is not met.

Conformance section	
Preconditions	4. The communication service is set in a mode where it records traces of the communication in the interface with PSAPs in SIP session control and RTP [11] media and so that it can be analysed.
Procedure	10. Check the communication trace of the interface to the PSAP and analyse the SIP and RTP [11] parts to verify the following: a) That the INVITE is formed according to ETSI TS 103 479 [3] requirements, and specifically that IETF RFC 7852 [15] additional data with subscriber info language indication is included and matches the setting. b) That if video is supported by the communications service, video is offered with at least one codec according to ETSI TS 103 479 [3] requirements. c) That RTT is offered with a media specification similar to this example (but ports and payload type numbers may vary): m=text 11000 RTP/AVP 100 98 a=rtpmap:98 t140/1000 a=fmtp:98 cps=90 a=rtpmap:100 red/1000 a=fmtp:100 98/98/98 a=rtt-mixer indicating multiparty support and use of two-fold redundancy. d) Text is included as a SIP level Media tag. e) That audio is offered with at least one wide-band codec supported according to ETSI TS 103 479 [3]. 11. Check that the OK from the PSAP confirms voice and RTT codecs and also video codec, if video is supported by the communications service.
Result	Pass: Checks 10, 11 are true. Fail: Check 10 or 11 is false.

B.8.3.3 Initiation of emergency communications

Interoperability section	
Precondition	1. The ICT under test is an electronic communication service using SIP for call control and voice and RTT media. 2. Video is included in the test, if video is supported by the communications service. 3. An end user equipment for test is available registered in the service.
Procedure	1. Initiate an emergency communication with RTT and voice from the test equipment. Include video media if video is supported by the communications service. 2. Check that the SIP electronic communications service initiates communication with a PSAP. 3. Check that the emergency communication is received and answered and a communication is set up. 4. Check that basic communication is possible in voice and RTT media, and that video communication is possible, if video is supported by the communications service.
Result	Pass: Checks 2, 3 and 4 are true. Fail: Check 2 or 3 or 4 is false. Not applicable: Pre-condition 1 or 3 is not met.

Conformance section	
Preconditions	4. The communication service is set in a mode where it records traces of the communication in the interface PSAPs in SIP session control and RTP [11] media and so that it can be analysed.
Procedure	5. Check the communication trace and analyse the SIP and RTP [11] parts to verify the following: a) That the INVITE is formed according to ETSI TS 103 479 [3] requirements, and specifically that the SIP request URI contains a service URN (e.g. urn:service:sos or any subservice). b) That RTT is offered with a media specification similar to this example (but ports and payload type numbers may vary): m=text 11000 RTP/AVP 100 98 a=rtpmap:98 t140/1000 a=fmtp:98 cps=90 a=rtpmap:100 red/1000 a=fmtp:100 98/98/98 a=rtt-mixer indicating multiparty support and use of two-fold redundancy. c) That audio is offered with at least one wide-band codec supported. d) That video is offered with a video codec as specified in ETSI TS 103 479 [3], if video is supported in the communication service. 6. Check that the OK from the PSAP confirms the voice and RTT codecs and that also the video codec is confirmed, if video is supported in the communications service. 7. Check that RTT text packets contain original and two redundant transmissions of text.
Result	Pass: Checks 5, 6, 7 are true. Fail: Check 5 or 6 or 7 is false.

B.8.3.4 Call back

Interoperability section	
Precondition	1. The ICT under test is an electronic communication service using SIP for call control and RTP [11] for voice and RTT. 2. Video is included in the test if video is supported by the communications service. 3. End user equipment is available registered in the service.
Procedure	1. Initiate an emergency communication with video RTT and voice. Include also video, if video is supported by the communications service. 2. Check that the electronic communications service initiates communication with a PSAP. 3. Check that the emergency communication is received and answered and a communication is set up. 4. Check that basic communication is possible in all voice and RTT and also in video, if video is supported by the communications service. 5. Ask the call taker to disconnect and call back. 6. Check that the call back is received and can be answered. 7. Check that the media which were included in the communication establishment can be used.
Result	Pass: Checks 1-7 are true. Fail: Check 2 or 3 or 4 or 5 or 6 or 7 is false. Not applicable: Pre-condition 1 or 3 is not met.

Conformance section	
Preconditions	4. The communication service is set in a mode where it records traces of the communication in SIP session control and RTP [11] media in the interface with PSAPs and so that it can be analysed.
Procedure	8. Check the communication trace and analyse the SIP and RTP [11] parts to verify the following: a) That the INVITE of the call back is formed according to ETSI TS 103 479 [3] requirements, and specifically that the mark "psap-callback" for emergency call back is set in the priority SIP header. b) That video is offered with at least one codec according to ETSI TS 103 479 [3] requirements. c) That RTT is offered with a media specification similar to this example (but ports and payload type numbers may vary): m=text 11000 RTP/AVP 100 98 a=rtpmap:98 t140/1000 a=fmtp:98 cps=90 a=rtpmap:100 red/1000 a=fmtp:100 98/98/98 a=rtt-mixer indicating multiparty support and use of two-fold redundancy. d) That audio is offered with at least one wide-band codec supported according to ETSI TS 103 479 [3]. e) That video is offered with at least one video codec supported according to ETSI TS 103 479 [3], if video is supported by the communication service. 9. Check that the OK from the device under test confirms these codecs. 10. Check that RTT text packets contain original and two redundant transmissions of text.
Result	Pass: Checks 8, 9, 10 are true. Fail: Check 8 or 9 or 10 is false.

B.8.4 Other technologies for emergency communication

B.8.4.1 General

Void.

B.8.4.2 Conversion to ETSI TS 103 479 emergency communication interfaces

Interoperability section	
Precondition	1. The ICT under test is an electronic communication service other technologies than SIP or IMS for call control and voice and RTT. Video media is included in the test, if video is supported by the communication service. 2. An end user equipment for test is available registered in the service.
Procedure	1. Initiate an emergency communication with RTT and voice from the test equipment. Include also video media, if the communication service supports video. 2. Check that the electronic communications service initiates communication with a PSAP. 3. Check that the emergency communication is received and answered and a communication is set up. 4. Check that basic communication is possible in video and RTT media, and also in video, if video is supported by the communication service
Result	Pass: Checks 2, 3 and 4 are true. Fail: Check 2 or 3 or 4 is false. Not applicable: Pre-condition 1 or 2 is not met.

Conformance section	
Preconditions	3. The communication service is set in a mode where it records traces of the communication in the interface with PSAPs in SIP session control and RTP [11] media and so that it can be analysed.
Procedure	5. Check the communication trace and analyse the SIP and RTP [11] parts to verify the following: a) That the INVITE is formed according to ETSI TS 103 479 [3] requirements, and specifically that the SIP request URI contains a service URN (e.g. urn:service:sos or any subservice). b) That RTT is offered with a media specification similar to this example indicating multiparty support and use of two-fold redundancy (but ports and payload type numbers may vary): m=text 11000 RTP/AVP 100 98 a=rtpmap:98 t140/1000 a=fmtp:98 cps=90 a=rtpmap:100 red/1000 a=fmtp:100 98/98/98 a=rtt-mixer c) That audio is offered with at least one wide-band codec supported. d) That video is offered, if video is supported by the communications service. 6. Check that the OK from the PSAP confirms these codecs. 7. Check that RTT text packets contain original and two redundant transmissions of text.
Result	Pass: Checks 5, 6, 7 are true. Fail: Check 5 or 6 or 7 is false.

B.8.4.3 Use of ETSI TS 103 479 together with PEMEA

B.8.4.3.1 General

Interoperability section	
Precondition	1. The ICT under test is a system for emergency communication based on the PEMEA standards. 2. The user equipment of the ICT under test is located in a region with PEMEA support. 3. A PSAP is available with support for PEMEA and PEMEA-based communication in voice, RTT and video.
Procedure	1. Initiate a PEMEA based emergency communications with RTT and voice and video from the user equipment. 2. Check that an emergency communication based on PEMEA is initiated. 3. Check that the emergency communication is received and answered and a communication is set up. 4. Check that basic communication is possible in the media supported by the PEMEA system in the appropriate PSAP. 5. Check that emergency context information is made available to the PSAP.
Result	Pass: Checks 2, 3, 4 and 5 are true. Fail: Check 2 or 3 or 4 or 5 is false. Not applicable: Pre-condition 1 or 2 or 3 is not met.

Conformance section	
Preconditions	4. The PSAP domain is set in a mode where it records traces of the PEMEA-based communication in the interface with the PSAP so that it can be analysed.
Procedure	5. Check the communication trace and analyse the PEMEA communication to verify the following: a) That the communication is formed according to PEMEA standards. b) That context information is formatted according to PEMEA standards.
Result	Pass: Check 5 is true. Fail: Check 5 is false.

B.8.4.3.2 Initiating emergency communication with PEMEA support

Interoperability section	
Precondition	1. The ICT under test is a system used to convey emergency context information based on the PEMEA standards. 2. The user equipment of the ICT under test is located in a region with PEMEA support. 3. The user equipment is also capable of making IMS emergency calls.
Procedure	1. Initiate an emergency communication with RTT and voice so that it makes use of PEMEA system for emergency context information, but makes use of IMS for the real-time emergency communication. 2. Check that an emergency communication based on IMS is initiated. 3. Check that the emergency communication is received and answered and a communication is set up. 4. Check that basic communication is possible in the media supported by the IMS MTSI service. 5. Check that the PEMEA based emergency context information is available to the PSAP.
Result	Pass: Checks 2, 3, 4 and 5 are true. Fail: Check 2 or 3 or 4 or 5 is false. Not applicable: Pre-condition 1 or 2 or 3 is not met.

Conformance section	
Preconditions	4. The PSAP domain is set in a mode where it records traces of the communication in the interface with PSAP in SIP session control and RTP [11] media, and so that it can be analysed.
Procedure	5. Check the communication trace and analyse the SIP and RTP [11] parts to verify the following: a) That the INVITE is formed according to ETSI TS 103 479 [3] requirements, and specifically that the SIP request URI contains a service URN (e.g. urn:service:sos or any subservice). b) That RTT is offered with a media specification similar to this example indicating multiparty support and use of two-fold redundancy (but ports and payload type numbers may vary): m=text 11000 RTP/AVP 100 98 a=rtpmap:98 t140/1000 a=fmtp:98 cps=90 a=rtpmap:100 red/1000 a=fmtp:100 98/98/98 a=rtt-mixer c) That voice is offered with at least one wide-band codec supported. d) That if video is supported by the IMS service it is offered as required by ETSI TS 103 479 [3]. 6. Check that the OK from the PSAP domain confirms these codecs. 7. Check that RTT text packets contain original and two redundant transmissions of text.
Result	Pass: Checks 5, 6, 7 are true. Fail: Check 5 or 6 or 7 is false.

B.8.4.3.3 Emergency communications by IMS in regions without PEMEA support

Interoperability section	
Precondition	1. The ICT under test is a system for emergency communication based on the PEMEA standards. 2. The user equipment of the ICT under test is located in a region without PEMEA support. 3. The user equipment is also capable of making IMS emergency calls.
Procedure	1. Initiate an emergency communication with RTT and voice and video from the PEMEA based user equipment. 2. Check that an emergency communication based on IMS is initiated as a fall-back where PEMEA is not supported. 3. Check that the emergency communication is received and answered and a communication is set up. 4. Check that basic communication is possible in the media supported by the IMS MTSI service.
Result	Pass: Checks 2, 3 and 4 are true. Fail: Check 2 or 3 or 4 is false. Not applicable: Pre-condition 1 or 2 or 3 is not met.

Conformance section	
Preconditions	4. The PSAP domain is set in a mode where it records traces of the communication in the interface with PSAPs in SIP session control and RTP [11] media and so that it can be analysed.
Procedure	5. Check the communication trace and analyse the SIP and RTP [11] parts to verify the following: a) That the INVITE is formed according to ETSI TS 103 479 [3] requirements, and specifically that the SIP request URI contains a service URN (e.g. urn:service:sos or any subservice). b) That RTT is offered with a media specification similar to this example indicating multiparty support and use of two-fold redundancy (but ports and payload type numbers may vary): m=text 11000 RTP/AVP 100 98 a=rtpmap:98 t140/1000 a=fmtp:98 cps=90 a=rtpmap:100 red/1000 a=fmtp:100 98/98/98 a=rtt-mixer c) That voice is offered with at least one wide-band codec supported. d) That if video is supported by the IMS service it is offered as required by ETSI TS 103 479 [3] 6. Check that the OK from the PSAP domain confirms these codecs. 7. Check that RTT text packets contain original and two redundant transmissions of text.
Result	Pass: Checks 5, 6, 7 are true. Fail: Check 5 or 6 or 7 is false.

B.8.4.3.4 Emergency communications by SIP in regions without PEMEA support

Interoperability section	
Precondition	1. The ICT under test is a system for emergency communication based on the PEMEA standards. 2. The user equipment of the ICT under test is located in a region without PEMEA support. 3. The user equipment is also capable of making SIP based emergency calls.
Procedure	1. Initiate an emergency communication with RTT and voice and video from the PEMEA based user equipment. 2. Check that an emergency communication based on SIP is initiated as a fall-back where PEMEA is not supported. 3. Check that the emergency communication is received and answered and a communication is set up. 4. Check that basic communication is possible in the media supported by the SIP service.
Result	Pass: Checks 2, 3 and 4 are true. Fail: Check 2 or 3 or 4 is false. Not applicable: Pre-condition 1 or 2 or 3 is not met.

Conformance section	
Preconditions	4. The PSAP domain is set in a mode where it records traces of the communication in the interface with PSAPs in SIP session control and RTP [11] media and so that it can be analysed.
Procedure	5. Check the communication trace and analyse the SIP and RTP [11] parts to verify the following: a) That the INVITE is formed according to ETSI TS 103 479 [3] requirements, and specifically that the SIP request URI contains a service URN (e.g. urn:service:sos or any subservice). b) That RTT is offered with a media specification similar to this example indicating multiparty support and use of two-fold redundancy (but ports and payload type numbers may vary): m=text 11000 RTP/AVP 100 98 a=rtpmap:98 t140/1000 a=fmtp:98 cps=90 a=rtpmap:100 red/1000 a=fmtp:100 98/98/98 a=rtt-mixer c) That voice is offered with at least one wide-band codec supported. d) That if video is supported by the SIP service it is offered as required by ETSI TS 103 479 [3] 6. Check that the OK from the PSAP domain confirms these codecs. 7. Check that RTT text packets contain original and two redundant transmissions of text.
Result	Pass: Checks 5, 6, 7 are true. Fail: Check 5 or 6 or 7 is false.

B.8.4.3.5 Accessibility related details in use of PEMEA

Interoperability section	
Precondition	1. The ICT under test is a system for emergency communication based on the PEMEA standards. 2. The user equipment of the ICT under test makes use of PEMEA and is located in a region with PEMEA support. 3. A PSAP is available with support for PEMEA and PEMEA-based communication in voice, RTT and video. 4. A relay service is available with interface suitable for connection to the PSAP with PEMEA implementation.
Procedure	1. Make settings for preferred modality and language in both directions in the user interface of the PEMEA based app. 2. Make settings for the address of the relay service available to the user. 3. Initiate a PEMEA based emergency communication with RTT and voice and video from the user equipment. 4. Check that an emergency communication based on PEMEA is initiated. 5. Check that the emergency communication is received and answered and a communication is set up. 6. Check that the PSAP got information on the preferred languages and modalities of the user in both directions. 7. Check that the PSAP can invoke the preferred relay service in the communication and get the conversation translated.
Result	Pass: Checks 4, 5, 6 and 7 are true. Fail: Check 4 or 5 or 6 or 7 is false. Not applicable: Pre-condition 1 or 2 or 3 or 4 is not met.

Conformance section	
Preconditions	5. The PSAP domain is set in a mode where it records traces of the PEMEA-based communication in the interface with the PSAP so that it can be analysed.
Procedure	8. Check the communication trace and analyse the PEMEA communication to verify the following: a) That the language and modality preference is formed according to PEMEA standards. b) That the preferred relay service address is formatted according to PEMEA standards.
Result	Pass: Check 8 is true. Fail: Check 8 is false.

B.8.4.4 Use of ETSI TS 103 479 with future technologies

The clause has no testable requirements.

B.9 Emergency communications network including PSAP

B.9.1 General

The clause has no testable requirements.

B.9.2 Media

Interoperability section	
Precondition	1. The ICT under test is an emergency communication network and PSAP using ETSI TS 103 479 [3] with SIP for call control and RTP [11] for video, voice and RTT media. 2. A SIP based communication service with end user equipment to be used as a counterpart is available capable of handling RTT, video and voice. NOTE: Since IMS services also use SIP according to ETSI TS 103 479 [3] in their emergency communications interface to the PSAP domain, this test can be performed with an IMS MTSI service as counterpart.
Procedure	1. Initiate an emergency communication with RTT, video and voice from the test equipment. 2. Check that the expected PSAP receives the initiation. 3. Check that the emergency communication is answered and a communication is set up. 4. Check that basic communication is possible in all three media.
Result	Pass: Checks 2, 3 and 4 are true. Fail: Check 2 or 3 or 4 is false. Not applicable: Pre-condition 1 or 2 is not met.

Conformance section	
Preconditions	3. The emergency communications network interface is set in a mode where it records traces of the communication in the interface with emergency services in SIP session control and RTP [11] media and so that it can be analysed.
Procedure	5. Check the communication trace and analyse the SIP and RTP [11] parts to verify the following: a) That the INVITE is formed according to ETSI TS 103 479 [3] requirements, and specifically that the SIP request URI contains a service URN (e.g. urn:service:sos or any subservice). b) That RTT is offered with a media specification similar to this example (but ports and payload type numbers may vary): m=text 11000 RTP/AVP 100 98 a=rtpmap:98 t140/1000 a=fmtp:98 cps=90 a=rtpmap:100 red/1000 a=fmtp:100 98/98/98 a=rtt-mixer indicating multiparty support and use of two-fold redundancy. c) That audio is offered with at least one wide-band codec supported. d) That video is supported according to ETSI TS 103 479 [3]. 6. Check that the OK from the PSAP confirms these codecs. 7. Check that RTT text packets contain original and two redundant transmissions of text.
Result	Pass: Checks 5, 6, 7 are true. Fail: Check 5 or 6 or 7 is false.

B.9.3 Modality and Language indication

Interoperability section	
Precondition	1. The ICT under test is an emergency communication network and PSAP using ETSI TS 103 479 [3] with SIP for call control and RTP [11] for video, voice and RTT media. 2. A SIP based communication service with end user equipment for test is available capable of handling RTT, video and voice. NOTE: Since IMS services also use SIP according to ETSI TS 103 479 [3] in their emergency communications interface to the PSAP domain, this test can be performed with an IMS MTSI service as counterpart. 3. The end user equipment is provided with a setting of language preference to be set into the base for subscriber additional data according to IETF RFC 7852 [15] and the language is set to English. 4. The ICT under test is set in a mode where only some PSAPs are set to handle English spoken and written language.
Procedure	1. Initiate an emergency communication with RTT, video and voice from the test equipment. 2. Check that a PSAP with setting to handle English spoken or written language receives the emergency communication and that it is answered, and a communication is set up. 3. Check that the PSAP can begin communication in RTT, voice and video.
Result	Pass: Checks 2, 3 are true. Fail: Check 2 or 3 is false. Not applicable: Pre-condition 1 or 2 or 3 or 4 is not met.

Conformance section	
Preconditions	5. The emergency communications network interface is set in a mode where it records traces of the communication in the interface with the PSAP in SIP session control and RTP [11] media and so that it can be analysed.
Procedure	4. Check the communication trace and analyse the SIP and RTP [11] parts to verify the following: a) That the initiation is formed according to ETSI TS 103 479 [3] requirements and that IETF RFC 7852 [15] additional data is provided and contains English subscriber language indication. b) That the media availability is properly provided in the initiation and answered.
Result	Pass: Check 4 is true. Fail: Check 4 is false.

B.9.4 Routing

Interoperability section	
Precondition	1. The ICT under test is an emergency communications network and PSAP using ETSI TS 103 479 [3] with SIP for call control and RTP [11] for video, voice and RTT media. 2. A SIP based communication service with end user equipment to be used as a counterpart, is available capable of handling RTT, video and voice. NOTE: Since IMS services also use SIP according to ETSI TS 103 479 [3] in their emergency communications interface to the PSAP domain, this test can be performed with an IMS MTSI service as counterpart. 3. The ICT under test is set in a mode where only some PSAPs have video capability.
Procedure	1. Initiate an emergency communication with RTT, video and voice from the test equipment. 2. Check that the expected regional emergency communications network receives the initiation. 3. Check that a video capable PSAP receives the emergency communications and that it is answered, and a communication is set up. 4. Check that the PSAP can begin communication in all media.
Result	Pass: Checks 2, 3, 4 are true. Fail: Check 2 or 3 or 4 is false. Not applicable: Pre-condition 1 or 2 or 3 is not met.

Conformance section	
Preconditions	4. The PSAP interface is set in a mode where it records traces of the communication in the interface with PSAPs in SIP session control and RTP [11] media and so that it can be analysed.
Procedure	5. Check the communication trace and analyse the SIP and RTP [11] parts to verify the following: a) That the initiation is formed according to ETSI TS 103 479 [3] requirements and that proper location information is provided with the initiation. b) That the media availability is properly provided in the initiation and answered.
Result	Pass: Check 5 is true. Fail: Check 5 is false.

B.9.5 Bridging

Interoperability section	
Precondition	1. The ICT under test is an emergency communications network and PSAP using ETSI TS 103 479 [3] with SIP for call control and RTP [11] for video, voice and RTT media, having at least two call takers and a multiparty bridge for RTT, video and voice. 2. A SIP based communication service with end user equipment is available capable of handling RTT, and voice. If video is supported by the communication service, video is also used if supported by the communications service. NOTE: Since IMS services also use SIP according to ETSI TS 103 479 [3] in their emergency communications interface to the PSAP domain, this test can be performed with an IMS MTSI service as counterpart.
Procedure	Initiate a RTT + voice emergency communications. Include video if video is supported by the communications service. 1. Check that the communication is answered by the PSAP. 2. Check that RTT and voice is enabled and functional and also video, if video is supported by the communication service. 3. Let the call taker include another call taker in the call to create a three-party call. 4. Let the call takers send RTT text simultaneously. 5. Check that RTT is presented in real time in a readable way on the user equipment indicating an approximate time order of the received text. 6. Check that RTT from the user equipment is presented in real time to both PSAP call takers. 7. Check that voice is mixed. 8. Check that if video was supported by the communication service, video is included in the communication, it is presented to all three participants.
Result	Pass: Checks 1, 2, 5, 6, 7 and 8 are true. Fail: Check 1 or 2 or 5 or 6 or 7 or 8 is false. Not applicable: Pre-condition 1 or 2 is not met.

Conformance section	
Preconditions	3. The emergency communications network interface is set in a mode where it records traces of the communication in the interface with the communication service in SIP session control and RTP [11] media and so that it can be analysed.
Procedure	9. Check the communication trace and analyse the SIP and RTP [11] parts to verify the following: a) That the transfer to the bridge and invocation of the second call taker is visible in the trace and formed according to ETSI TS 103 479 [3] requirements. b) That RTT is offered with a media specification similar to this example (but ports and payload type numbers may vary): m=text 11000 RTP/AVP 100 98 a=rtpmap:98 t140/1000 a=fmtp:98 cps=90 a=rtpmap:100 red/1000 a=fmtp:100 98/98/98 a=rtt-mixer indicating multiparty support and use of two-fold redundancy.
Result	Pass: Check 9 is true. Fail: Check 9 is false.

B.9.6 Call back

Interoperability section	
Precondition	1. The ICT under test is an emergency communications network and PSAP using ETSI TS 103 479 [3] with SIP for call control and RTP [11] for video, voice and RTT media. 2. A SIP based communication service with end user equipment is available to be used as a counterpart, capable of handling RTT, video and voice. NOTE: Since IMS services also use SIP according to ETSI TS 103 479 [3] in their emergency communications interface to the PSAP domain, this test can be performed with an IMS MTSI service as counterpart.
Procedure	1. Initiate an emergency communication with RTT and voice from the equipment for test. Include also video, if video is supported by the communications service. 2. Check that the electronic communications service initiates communication with the emergency communications network. 3. Check that the emergency communication is received and answered and a communication is set up. 4. Check that basic communication is possible in voice and RTT, and also in video if video is supported by the communications service. 5. Ask the call taker to disconnect and call back. 6. Check that the call back is initiated and answered. 7. Check that voice and RTT media can be used, and also video media if supported by the communication service.
Result	Pass: Checks 2, 3, 4, 6, 7 are true. Fail: Check 2 or 3 or 4 or 6 or 7 is false. Not applicable: Pre-condition 1 or 2 is not met.

Conformance section	
Preconditions	3. The PSAP domain emergency communications interface is set in a mode where it records traces of the communication in the interface with the communications service in SIP session control and RTP [11] media and so that it can be analysed.
Procedure	8. Check the communication trace and analyse the SIP and RTP [11] parts to verify the following: a) That the INVITE of the call back is formed according to ETSI TS 103 479 [3] requirements, and specifically that the mark "psap-callback" for emergency call back is set in the priority SIP header. b) That if the communications service supports video, video is offered with at least one codec according to ETSI TS 103 479 [3] requirements. c) That RTT is offered with a media specification similar to this example (but ports and payload type numbers may vary): m=text 11000 RTP/AVP 100 98 a=rtpmap:98 t140/1000 a=fmtp:98 cps=90 a=rtpmap:100 red/1000 a=fmtp:100 98/98/98 a=rtt-mixer indicating multiparty support and use of two-fold redundancy. d) That audio is offered with at least one wide-band codec supported according to ETSI TS 103 479 [3]. 9. Check that the OK from the device under test confirms these codecs. 10. Check that RTT text packets contain original and two redundant transmissions of text.
Result	Pass: Checks 8, 9, 10 are true. Fail: Check 8 or 9 or 10 is false.

B.9.7 Communications handling

Interoperability section	
Precondition	1. The ICT under test is an emergency communications network and PSAP using ETSI TS 103 479 [3] with SIP for call control and RTP [11] for video, voice and RTT media. 2. A SIP based communication service with end user equipment for test is available capable of handling RTT, video and voice. NOTE: Since IMS services also use SIP according to ETSI TS 103 479 [3] in their emergency communications interface to the PSAP domain, this test can be performed with an IMS MTSI service as counterpart. 3. The ICT under test is set in a mode where an incoming communication for test is placed in queue. 4. The ICT under test is set in a mode where only some PSAPs have video capability.
Procedure	1. Initiate an emergency communication with RTT, video and voice from the test equipment. 2. Check that the expected emergency communications network receives the initiation. 3. Check that the emergency communication is answered and a communication is set up. 4. Check that queue information is sent in all three media and received by the test equipment. 5. Enable the PSAPs to take communication. 6. Check that a PSAP with video capability is offered the test communication and can begin communication in all media.
Result	Pass: Checks 2, 3, 4, 5, 6 are true. Fail: Check 2 or 3 or 4 or 5 or 6 is false. Not applicable: Pre-condition 1 or 2 or 3 or 4 is not met.

Conformance section	
Preconditions	5. The emergency communication network interface is set in a mode where it records traces of the communication in the interface with PSAPs in SIP session control and RTP [11] media and so that it can be analysed.
Procedure	7. Check the communication trace and analyse the SIP and RTP [11] parts to verify the following: a) That the initiation is formed according to ETSI TS 103 479 [3] requirements and that waiting information in all media is sent. b) That the move to an active PSAP is done as a reINVITE with proper media enabled.
Result	Complete test pass: Check 7 is true. Complete test fail: Check 7 is false.

B.10 Supporting Services

Interoperability section	
Precondition	1. The ICT under test is a supporting service to PSAPs, (e.g. a relay service or an expert advice service). 2. A SIP based communication service with end user equipment for test is available capable of handling RTT, and voice. NOTE: Since IMS services also use SIP according to ETSI TS 103 479 [3] in their emergency communications interface to the PSAP domain, this test can be performed with an IMS MTSI service as counterpart. 3. An emergency communications network and PSAP for test is available.
Procedure	Initiate a RTT + voice emergency communications from the IMS user equipment: 1. Check that the communication is answered by the PSAP. 2. Check that RTT and voice is enabled and functional. 3. Assume that the call taker finds a need to invoke a relay service or expert advice or other supporting service. 4. Let the call taker include the supporting service in the call to create a three-party call in both media. 5. Check that the inclusion of the supporting service is successful. 6. Check that RTT is presented in real time in a readable way on the user equipment indicating an approximate time order of the received text. 7. Check that the RTT from the user is presented in real time to the call taker and support service. 8. Check that voice is mixed.
Result	Pass: Checks 1, 2, 5, 6, 7 and 8 are true. Fail: Check 1 or 2 or 5 or 6 or 7 or 8 is false. Not applicable: Pre-condition 1 or 2 or 3 is not met.

Conformance section	
Preconditions	4. The supporting service interface with the PSAP is set in a mode where it records traces of the communication in the interface with the PSAP in SIP session control and RTP [11] media and so that it can be analysed.
Procedure	8. Check the communication trace and analyse the SIP and RTP [11] parts to verify the following: a) That the inclusion of the supporting service is formed according to ETSI TS 103 479 [3] requirements. b) That RTT is offered with a media specification similar to this example (but ports and payload type numbers may vary): m=text 11000 RTP/AVP 100 98 a=rtpmap:98 t140/1000 a=fmtp:98 cps=90 a=rtpmap:100 red/1000 a=fmtp:100 98/98/98 a=rtt-mixer indicating multiparty support and use of two-fold redundancy. c) That voice is included with a wide-band codec. d) Check that RTT text packets contain original and two redundant transmissions of text.
Result	Pass: Check 8 is true. Fail: Check 8 is false.

Annex C (informative): User Level Use Cases

C.1 Introduction

The present informative annex contains a set of use case descriptions for emergency communication situations involving users with disabilities in emergency situations. The use cases can be used as examples and clarifications for situations which may appear in emergency cases and be handled through emergency communications.

The collection of use cases has no ambition to cover all possible cases.

C.2 Users with mobile phones with voice and RTT

C.2.1 General

The following use cases are examples of use of mobile phones with only voice or with voice and real time text.

C.2.2 User with low vision calling emergency with voice in home country

C.2.3 Blind user calling in emergency calling with voice in foreign country

C.2.4 Hard-of-hearing user in emergency calling with RTT and voice in home country

C.2.5 Call back to hard of hearing user with voice and RTT

C.2.6 Deaf user in emergency calling with RTT and being transferred to mountain guard using RTT, alternatively needing support for modality conversion between RTT and speech.

C.2.7 Deaf user in emergency in foreign country calling with RTT and getting language support in RTT.

C.2.8 Call back to deaf user in foreign country with voice and RTT and including same language support as in original call

C.2.9 Deaf user in emergency not trusting the PSAP to be able to use text, calling text relay service and requesting emergency communication

C.2.10 Hard-of-hearing user in emergency visiting other country, not believing she will be able to communicate with the local emergency centre, therefore calling relative in home country asking the relative to make emergency call.

C.2.11 Deaf-blind user in emergency having mobile phone with built-in assistive technology providing Braille display and QWERTY keyboard, making emergency call and being served with RTT.

C.2.12 Hard-of-hearing user in emergency calling with RTT and voice in home country. Being able to talk, but not hear. The answering call taker attaches automatic speech-to-text to the call in a way that displays the call takers transmitted text also to the call taker for catching any mistakes. The call handling continues mainly by speech, vs speech-to-text occasionally complemented by RTT or ASR corrections.

C.2.13 User with mental disability in emergency calling emergency call with mobile phone including assistive technology software enabling the user to communicate by sending pictograms translated to text and voice by the assistive technology. The call taker transfers the call by attended transfer to a call taker with competence in communicating with persons with disabilities. The caller understands speech, so the call taker can answer by speech. Alternatively the user sends Blisscode, and the call taker answers in rtt which the caller's assistive technology software converts to Blissymbols.

C.2.14 User with mental disability in emergency calling emergency call with mobile phone including assistive technology software enabling the user to communicate by sending Blissymbols translated to text and voice by the assistive technology. The call taker transfers the call by attended transfer to a call taker with competence in communicating with persons with disabilities. The call taker answers in rtt which the caller's assistive technology software converts to Blissymbols.

C.2.15 A user in medical emergency initiates an emergency communication with a mobile phone with voice. The call taker includes a medical expert in a three-party fashion. They discuss pharmaceuticals which the user has. A medicine name "phlyxophentine" is hard to pronounce so that the medical expert understands, so the user adds RTT to the call and types the name. The medical expert reads the name and the call can continue to solve the issue.

C.2.16 A deaf user in emergency calls for emergency communication. There is a catastrophic situation, so all call takers are busy for a moment. The calling user gets calming queue messages in both voice and RTT until a call taker is free and can take the call.

C.3 Users with mobile phones with total conversation

C.3.1 General

The following use cases are examples of use of mobile phones with users who have use of total conversation.

C.3.2 Hard-of-hearing user in emergency calling with total conversation in mobile phone, wanting to use voice and support understanding by seeing the call taker and having RTT as fallback when something gets hard to understand, reaching call taker using voice and having corresponding features.

C.3.3 Deaf sign language user in emergency making emergency call with total conversation in mobile phone, including preferred language indication in the call, reaching call taker with sign language competence. Using RTT for fallback for phrases requiring exact spelling.

C.3.4 Call-back to deaf sign language user in emergency with total conversation in mobile phone, using video for sign language and RTT for fallback for phrases requiring exact spelling.

C.3.5 Deaf sign language user in emergency making emergency call with total conversation in mobile phone, including preferred language indication in the call, reaching call taker without sign language competence. The call taker seeing the language preference invokes sign language interpreter as third party in the call. Using RTT to begin handling the case before the interpreter is included, then continuing in translated sign language with RTT fallback when needed.

C.3.6 Deaf sign language user in emergency making emergency call with total conversation in mobile phone, not including any preferred language indication in the call, reaching call taker without sign language competence, communicating in RTT and gestures about language preferences and invoking sign language interpreter as third party in the call. But the call taker and the interpreter have no common competence in spoken languages, so the call taker transfers the call via attended call transfer to a call taker with competence in the same spoken language as the interpreter. Then continuing in translated sign language with RTT fallback when needed.

C.3.7 Deaf sign language user in emergency not trusting that the emergency centre has competence in the user's sign language, therefore calling his usual video relay service from his total conversation in a mobile device, and asks for emergency communication. The relay service redirects the call to emergency communication in a way that makes the user terminal make a proper emergency call, and causing the relay service to be connected in tree-party mode in the call. Reaching call taker without sign language competence, communicating in translated sign language with RTT fallback when needed.

C.4 Users with app or web based Total Conversation

C.4.1 General

The following use cases are examples of use of app or web-based communication with users who have use of total conversation

C.4.2 Hard-of-hearing user in emergency calling with app or web based total conversation, wanting to use voice and support understanding by seeing the call taker and having RTT as fallback when something gets hard to understand, reaching call taker using voice and having corresponding features.

C.4.3 Deaf sign language user in emergency making emergency call with app or web based total conversation, including preferred language indication in the call, reaching call taker with sign language competence. Using RTT for fallback for phrases requiring exact spelling.

C.4.4 Call-back to deaf sign language user in emergency with app or web based total conversation, using video for sign language and RTT for fallback for phrases requiring exact spelling.

C.4.5 Deaf sign language user in emergency making emergency call with app or web based total conversation, including preferred language indication in the call, reaching call taker without sign language competence. The call taker seeing the language preference invokes sign language interpreter as third party in the call. Using RTT to begin handling the case before the interpreter is included, then continuing in translated sign language with RTT fallback when needed.

C.4.6 Deaf sign language user in emergency making emergency call with app or web based total conversation, not including any preferred language indication in the call, reaching call taker without sign language competence, communicating in RTT and gestures about language preferences and invoking sign language interpreter as third party in the call. But the call taker and the interpreter have no common competence in spoken languages, so the call taker transfers the call via attended call transfer to a call taker with competence in the same spoken language as the interpreter. Then continuing in translated sign language with RTT fallback when needed.

C.4.7 Deaf sign language user in emergency not trusting that the emergency centre has competence in the user's sign language, therefore calling their usual video relay service from his app or web based total conversation account, and asks for emergency communication. The relay service redirects the call to emergency communications in a way that makes the user terminal make a proper emergency call, and causing the relay service to be connected in tree-party mode in the call. Reaching call taker without sign language competence, communicating in translated sign language with RTT fallback when needed.

Annex D (informative): Cross References Between Requirements Labels and Clauses

This annex contains table D.1 presenting where the labels representing functional requirements appear in the technical requirements clauses, and table D.2 presenting the labels in alphabetical order with information on their main topic and where in clause 5 they are defined.

Table D.1: Cross reference table between functional requirement labels and clauses where they are used

Clause of functional requirement	Main topic	Label of functional requirement	Clause with technical requirement
5.2	Modality and Language		
		AEC-ML-01	7.2.2, 7.3.2, 8.2.2, 8.3.2, 8.4.1, 8.4.4
		AEC-ML-02	7.2.2, 7.3.2, 8.2.2, 8.3.2, 8.4.1, 8.4.4
		AEC-ML-03	7.2.3, 8.4.1, 8.4.4
		AEC-ML-04	8.4.1, 8.4.4, 9.3
		AEC-ML-05	8.4.1, 8.4.4, 9.3
		AEC-ML-06	8.4.1, 8.4.4, 9.3
		AEC-ML-07	8.4.1, 8.4.4
		AEC-ML-08	7.2.3, 7.3.3, 8.2.3, 8.3.3, 8.4.1, 8.4.4, 8.4.1, 8.4.4, 9.4, 9.7, 10
		AEC-ML-09	8.4.1, 8.4.4
5.3	User Interface and general accessibility		
		AEC-UI-01	5.3, 7.1, 7.4, 8.1, 8.4.1, 8.4.4, 9.1
		AEC-UI-02	7.2.3, 7.3.3, 7.4, 8.4.1, 8.4.4
		AEC-UI-03	7.2.1, 7.3.1, 7.4, 8.4.1, 8.4.4
		AEC-UI-04	8.4.1, 8.4.4, 9.2
		AEC-UI-05	7.2.1, 7.3.1, 7.4, 8.4.1, 8.4.4
		AEC-UI-06	7.2.6, 7.3.5, 7.4, 8.4.1, 8.4.4
		AEC-UI-07	7.3.1, 8.4.1, 8.4.4
5.4.2	Session Control and emergency contextual information		
		AEC-SC-01	7.2.3, 7.3.3, 7.4, 8.2.3, 8.3.3, 8.4.1, 8.4.3.2, 8.4.3.3, 8.4.3.5, 8.4.4
		AEC-SC-02	7.3.3, 7.4, 8.2.3, 8.3.3, 8.4.1, 8.4.2, 8.4.3.2, 8.4.3.3, 8.4.4
		AEC-SC-03	7.2.3, 7.2.6, 7.3.3, 7.3.5, 7.4, 8.2.3, 8.3.3, 8.4.1, 8.4.2, 8.4.3, 8.4.4, 9.6
		AEC-SC-04	7.2.3, 7.3.3, 7.4, 8.3.3, 8.4.1, 8.4.2, 8.4.3, 8.4.4
		AEC-SC-05	7.2.63, 7.2.6, 7.3.2, 7.3.3, 7.3.5, 7.4, 8.2, 8.2.3, 8.4.1, 8.4.2, 8.4.4, 9.3, 10
		AEC-SC-06	8.4.1, 8.4.4, 9.7
		AEC-SC-07	8.4.1, 8.4.4, 9.7
5.4.3	Routing		
		AEC-RO-01	8.4.1, 8.4.4, 9.4, 9.7
		AEC-RO-02	8.4.1, 8.4.4, 9.4, 9.7
		AEC-RO-03	8.4.1, 8.4.4, 9.4, 9.7
		AEC-RO-04	8.4.1, 8.4.4, 9.4, 9.7
		AEC-RO-05	8.4.1, 8.4.4, 9.4, 9.7
		AEC-RO-06	7.2.5, 8.4.1, 8.4.4, 9.4, 9.7
		AEC-RO-07	8.2.5, 8.4.1, 8.4.4, 9.4, 9.7
		AEC-RO-08	8.4.1, 8.4.4, 9.4, 9.7
		AEC-RO-09	8.4.1, 8.4.4, 9.4, 9.7, 9.8.2

Clause of functional requirement	Main topic	Label of functional requirement	Clause with technical requirement
5.4.4	Communication Transfer		
		AEC-CT-01	7.2.1, 7.3.1, 7.4, 8.3.1, 8.4.1, 8.4.4, 9.5, 9.7
		AEC-CT-02	8.4.1, 8.4.4, 9.5, 9.7
5.4.5	Conferencing		
		AEC-CO-01	8.4.1, 8.4.4, 9.2, 9.5, 10
		AEC-CO-02	8.4.1, 8.4.4, 9.2, 9.5
		AEC-CO-03	8.4.1, 8.4.4, 9.5
5.4.6	Call Back		
		AEC-CB-01	7.2.4, 7.3.4, 8.2.4, 8.3.4, 8.4.1, 8.4.4, 9.5
		AEC-CB-02	7.2.4, 7.3.4, 8.2.4, 8.3.4, 8.4.1, 8.4.4, 9.6
		AEC-CB-03	7.2.4, 7.3.4, 8.2.4, 8.3.4, 8.4.1, 8.4.4, 9.6
		AEC-CB-04	8.4.1, 8.4.4, 9.6
		AEC-CB-05	8.4.1, 8.4.4, 9.6
		AEC-CB-06	8.4.1, 8.4.4, 9.6
5.4.7	Charging		
		AEC-CG-01	8.1, 8.4.1, 8.4.4
5.5.2	Audio		
		AEC-CM-01	7.2.1, 7.3.1, 7.4, 8.2.1, 8.2.5, 8.3.1, 8.4.1, 8.4.4, 9.2
		AEC-CM-02	7.2.1, 7.2.5, 7.3.1, 7.4, 8.2.1, 8.2.5, 8.3.1, 8.4.1, 8.4.4, 9.2
5.5.3	Video		
		AEC-CM-03	7.2.1, 7.2.5, 7.3.1, 7.4, 8.2.1, 8.2.5, 8.3.1, 8.4.1, 8.4.4, 9.2
		AEC-CM-04	7.2.1, 7.3.1, 8.4.1, 8.4.4, 9.2
		AEC-CM-05	7.3.1, 8.4.1, 8.4.4, 9.2
		AEC-CM-06	7.2.1, 7.4, 8.2.1, 8.4.1, 8.4.4, 9.2
5.5.4	Real Time Text		
		AEC-CM-07	7.2.1, 7.2.5, 7.3.1, 7.4, 8.2.1, 8.2.2, 8.2.5, 8.3.2, 8.4.1, 8.4.4
		AEC-CM-08	7.2.2, 7.3.2, 8.4.1, 8.4.4
5.5.5	Text Messaging		
		AEC-CM-09	7.3.1, 8.4.1, 8.4.4
		AEC-CM-10	7.2.5, 7.3.1, 8.2.5, 8.4.1, 8.4.4
5.5.6	Total Conversation		
		AEC-CM-11	7.2.5, 8.2.5, 8.4.1, 8.4.4, 9.2
5.6	Relay Service Invocation by the User in Emergency		
		AEC-RS-01	7.2.6, 7.3.5, 8.4.1, 8.4.4, 10
		AEC-RS-02	7.2.6, 7.3.5, 8.4.1, 8.4.4, 10
5.7.1	Supporting Services; General		
		AEC-SS-01	8.4.1, 8.4.3, 8.4.4, 9.4, 9.8.3, 9.8.4, 10
		AEC-SS-02	7.2.2, 7.3.2, 7.4, 8.2.2, 8.4.1, 8.4.4, 9,4, 9.7, 9.8.3, 9.8.4
		AEC-SS-03	8.4.1, 8.4.4, 9,4, 9.7, 9.8.3, 9.8.4
		AEC-SS-04	7.3.5, 8.4.1, 8.4.4, 9.8,3, 9.8.4, 10
		AEC-SS-05	8.4.1, 8.4.4, 9.4, 9.8,3, 9.8.4, 10
5.7.2	Relay Services		
		AEC-SS-06	8.4.1, 8.4.2, 9.8.3, 9.8.4, 10
		AEC-SS-07	8.4.1, 8.4.2, 9.8.4, 10
		AEC-SS-08	8.4.1, 8.4.2, 9.8.3, 10
		AEC-SS-09	8.4.1, 8.4.2, 9.8.3, 10
		AEC-SS-10	8.4.1, 8.4.2, 9.8.3, 10
		AEC-SS-11	8.4.1, 8.4.2, 9.8.3, 10
5.7.3	Translation Services		
		AEC-SS-12	8.4.1, 8.4.2, 10
5.7.4	Expert Services		
		AEC-SS-13	8.4.1, 8.4.2, 10

Clause of functional requirement	Main topic	Label of functional requirement	Clause with technical requirement
5.8	Documentation		
		AEC-AD-01	5.8
		AEC-AD-02	5.8
		AEC-AD-03	5.8
		AEC-AD-04	5.8

Table D.2: List of functional requirements labels and their location in clause 5

Functional Requirements Label	Defined in clause	Clause header
AEC-AD-01	5.8	Documentation
AEC-AD-02	5.8	Documentation
AEC-AD-03	5.8	Documentation
AEC-AD-04	5.8	Documentation
AEC-CB-01	5.4.6	Call Back
AEC-CB-02	5.4.6	Call Back
AEC-CB-03	5.4.6	Call Back
AEC-CB-04	5.4.6	Call Back
AEC-CB-05	5.4.6	Call Back
AEC-CB-06	5.4.6	Call Back
AEC-CG-01	5.4.7	Charging
AEC-CM-01	5.5.2	Audio
AEC-CM-02	5.5.2	Audio
AEC-CM-03	5.5.3	Video
AEC-CM-04	5.5.3	Video
AEC-CM-05	5.5.3	Video
AEC-CM-06	5.5.3	Video
AEC-CM-07	5.5.4	Real Time Text
AEC-CM-08	5.5.4	Real Time Text
AEC-CM-09	5.5.5	Text Messaging
AEC-CM-10	5.5.5	Text Messaging
AEC-CM-11	5.5.6	Total Conversation
AEC-CO-01	5.4.5	Conferencing
AEC-CO-02	5.4.5	Conferencing
AEC-CO-03	5.4.5	Conferencing
AEC-CT-01	5.4.4	Communication Transfer
AEC-CT-02	5.4.4	Communication Transfer
AEC-ML-01	5.2	Modality and Language
AEC-ML-02	5.2	Modality and Language
AEC-ML-03	5.2	Modality and Language
AEC-ML-04	5.2	Modality and Language
AEC-ML-05	5.2	Modality and Language
AEC-ML-06	5.2	Modality and Language
AEC-ML-07	5.2	Modality and Language
AEC-ML-08	5.2	Modality and Language
AEC-ML-09	5.2	Modality and Language
AEC-RO-01	5.4.3	Routing
AEC-RO-02	5.4.3	Routing
AEC-RO-03	5.4.3	Routing
AEC-RO-04	5.4.3	Routing
AEC-RO-05	5.4.3	Routing
AEC-RO-06	5.4.3	Routing
AEC-RO-07	5.4.3	Routing
AEC-RO-08	5.4.3	Routing
AEC-RO-09	5.4.3	Routing
AEC-RS-01	5.7.2	Relay Services
AEC-RS-02	5.7.2	Relay Services
AEC-SC-01	5.4.2	Session Control and emergency contextual information
AEC-SC-02	5.4.2	Session Control and emergency contextual information
AEC-SC-03	5.4.2	Session Control and emergency contextual information

Functional Requirements Label	Defined in clause	Clause header
AEC-SC-04	5.4.2	Session Control and emergency contextual information
AEC-SC-05	5.4.2	Session Control and emergency contextual information
AEC-SC-06	5.4.2	Session Control and emergency contextual information
AEC-SC-07	5.4.2	Session Control and emergency contextual information
AEC-SS-01	5.7.1	Supporting Services; General
AEC-SS-02	5.7.1	Supporting Services; General
AEC-SS-03	5.7.1	Supporting Services; General
AEC-SS-04	5.7.1	Supporting Services; General
AEC-SS-05	5.7.1	Supporting Services; General
AEC-SS-06	5.7.2	Relay Services
AEC-SS-07	5.7.2	Relay Services
AEC-SS-08	5.7.2	Relay Services
AEC-SS-09	5.7.2	Relay Services
AEC-SS-10	5.7.2	Relay Services
AEC-SS-11	5.7.2	Relay Services
AEC-SS-12	5.7.3	Translation Service
AEC-SS-13	5.7.4	Expert Service
AEC-UI-01	5.3	User Interface and general accessibility
AEC-UI-02	5.3	User Interface and general accessibility
AEC-UI-03	5.3	User Interface and general accessibility
AEC-UI-04	5.3	User Interface and general accessibility
AEC-UI-05	5.3	User Interface and general accessibility
AEC-UI-06	5.3	User Interface and general accessibility
AEC-UI-07	5.3	User Interface and general accessibility

History

Document history		
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