



TECHNICAL REPORT

**Emergency Communications (EMTEL);  
Total Conversation for emergency communications;  
implementation guidelines**

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Reference

DTR/EMTEL-00029

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Keywords

accessibility, disability, emergency, IMS, IP, real-time text, SIP, total conversation

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## Foreword

This Technical Report (TR) has been produced by ETSI Special Committee Emergency Communications (EMTEL).

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## 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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## Introduction

The present document contains recommendations and guidelines on the implementation of Total Conversation for emergency service access and provision. Total Conversation enables conversation in real-time text, video and audio. Subsets are also considered including the combination of real-time text and audio that forms the text telephony service.

Total Conversation services and terminals are deployed in some European countries, and have been adopted by people with disabilities who, for example, need video for sign language, or real-time text for a text based conversation or as complement to a voice conversation. The use of Total Conversation for Emergency Communications would enable and/or improve access to emergency services by people with disabilities. However the few deployments of Total Conversation for Emergency Communications that exist, are implemented in different ways in different countries, and are not implemented according to the latest development of ETSI, IETF and 3GPP standards for Emergency Communications. This non-harmonized deployment means that there are no interoperable solutions for emergency service access across the EU countries, which is contrary to EU policy.

The present document is intended to assist ETSI SC EMTEL to coordinate with other standards bodies and relevant stakeholders so that the recommendations of ETSI TS 101 470 [i.2] and ETSI TR 103 170 [i.3] can be implemented. It can also be used to assess if Total Conversation requirements are fulfilled by other necessary standards, and to ensure that there are no contradictions.

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# 1 Scope

The present document:

- Assesses the support of Total Conversation for emergency communications by existing specifications, in particular those from 3GPP and IETF.
- Identifies any changes that might be needed to those specifications to support Total Conversation for emergency communications.
- Provides guidance for developers and PSAPs planning to implement Total Conversation for emergency communications, and for users of the Total Conversation service.

The present document covers emergency calls with the full media set of Total Conversation as well as subsets of the media, except voice calls in which no assisting service is needed.

Although the focus of the present document is Total Conversation for emergency communications, no Total Conversation user can be expected to use Total Conversation only for contacting emergency services. Therefore and where applicable, some aspects of use of Total Conversation for non-emergency communications are also covered.

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## 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 reference document (including any amendments) applies.

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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] EENA NG1-1-2 Long Term Definition.
- [i.2] ETSI TS 101 470: "Emergency Communications (EMTEL); Total Conversation Access to Emergency Services".
- [i.3] ETSI TR 103 170: "Emergency Communications (EMTEL); Total Conversation Access to Emergency Services".
- [i.4] ETSI TR 102 180: "Emergency Communications (EMTEL); Basis of requirements for communication of individuals with authorities/organizations in case of distress (emergency call handling)".



- [i.5] ETSI ES 202 975: "Human Factors (HF); Requirements for relay services".
- [i.6] ETSI TS 122 173: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; IP Multimedia Core Network Subsystem (IMS) Multimedia Telephony Service and supplementary services; Stage 1 (3GPP TS 22.173)".
- [i.7] ETSI TS 126 114: "Universal Mobile Telecommunications System (UMTS); LTE; IP Multimedia Subsystem (IMS); Multimedia telephony; Media handling and interaction (3GPP TS 26.114)".
- [i.8] ETSI TS 123 167: "Universal Mobile Telecommunications System (UMTS); LTE; IP Multimedia Subsystem (IMS) emergency sessions (3GPP TS 23.167)".
- [i.9] ETSI TS 122 101: "Universal Mobile Telecommunications System (UMTS); LTE; Service aspects; Service principles (3GPP TS 22.101)".
- [i.10] ETSI TS 122 228: "Digital cellular telecommunications system (Phase 2+); 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.11] IETF RFC 6881: "Best Current Practice for Communications Services in Support of Emergency Calling (BCP 181)".
- [i.12] IETF RFC 4596: "Guidelines for Usage of the Session Initiation Protocol (SIP) Caller Preferences Extension".
- [i.13] IETF RFC 3841: "Caller Preferences for the Session Initiation Protocol (SIP)".
- [i.14] IETF RFC 6443: "Framework for Emergency Calling Using Internet Multimedia".
- [i.15] IETF RFC 4190: "Framework for Supporting Emergency Telecommunications Service (ETS) in IP Telephony".
- [i.16] IETF RFC 3261: "Session Initiation Protocol".
- [i.17] Recommendation ITU-T F.742: "Service description and requirements for videotelephony services over IP networks".
- [i.18] Recommendation ITU-T F.703: "Multimedia Conversational Services".
- [i.19] Recommendation ITU-T V.18: "Operational and interworking requirements for DCEs operating in the text telephone mode".
- [i.20] ETSI TS 122 071: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Services (LCS); Service description (3GPP TS 22.071)".
- [i.21] ETSI TS 123 271: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Functional stage 2 description of Location Services (LCS) (3GPP TS 23.271)".
- [i.22] ETSI TS 124 229: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP) (3GPP TS 24.229)".
- [i.23] IETF RFC 4353: "A Framework for Conferencing with the Session Initiation Protocol (SIP)".
- [i.24] Directive 2002/22/EC of the European Parliament and the Council of 7 March 2002 on universal service and users' rights relating to electronic communications networks and services (Universal Service Directive).
- [i.25] ETSI TS 133 106: "Universal Mobile Telecommunications System (UMTS); LTE; 3G security; Lawful interception requirements (3GPP TS 33.106)".
- [i.26] ETSI ES 201 158 (V1.2.1): "Telecommunications security; Lawful Interception (LI); Requirements for network functions".

- [i.27] ETSI TS 123 237: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; IP Multimedia Subsystem (IMS) Service Continuity; Stage 2 (3GPP TS 23.237)".
  - [i.28] IETF RFC 5631: "Session Initiation Protocol (SIP) Session Mobility".
  - [i.29] IETF RFC 5589: "Session Initiation Protocol (SIP) Call Control - Transfer".
  - [i.30] IETF RFC 4579: "Session Initiation Protocol (SIP) Call Control - Conferencing for User Agents".
  - [i.31] IETF RFC 3515: "The Session Initiation Protocol (SIP) Refer Method".
  - [i.32] IETF RFC 3725: "Best Current Practices for Third Party Call Control (3pcc) in the Session Initiation Protocol (SIP)".
  - [i.33] IETF RFC 5646: "Tags for Identifying Languages".
  - [i.34] ETSI TS 122 226: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Global text telephony (GTT); Stage 1 (3GPP TS 22.226)".
  - [i.35] ETSI TS 123 226: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Global text telephony (GTT); Stage 2 (3GPP TS 23.226)".
  - [i.36] ETSI TS 129 332: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); Media Gateway Control Function (MGCF) - IM Media Gateway; Mn interface".
  - [i.37] Errata 1203 to IETF RFC 4103: "RTP Payload for Text Conversation".
- NOTE: Available at [https://www.rfc-editor.org/errata\\_search.php?rfc=4103](https://www.rfc-editor.org/errata_search.php?rfc=4103).
- [i.38] IETF Draft: "Additional Data Related to an Emergency Call, draft-ietf-ecrit-additional-data-37".
  - [i.39] IETF RFC 4566: "SDP: Session Description Protocol".
  - [i.40] IETF Draft: "Negotiating Human Language in Real-Time Communications, draft-ietf-slim-negotiating-human-language-00".
  - [i.41] IETF RFC 3264: "An Offer/Answer Model with the Session Description Protocol (SDP)".
  - [i.42] IETF RFC 6157: "IPv6 Transition in the Session Initiation Protocol (SIP)".
  - [i.43] IETF RFC 7090: "Public Safety Answering Point (PSAP) Callback".
  - [i.44] IETF RFC 6189: "ZRTP: Media Path Key Agreement for Secure RTP".
  - [i.45] GSMA IR.92: "IMS Profile for Voice and SMS".
  - [i.46] GSMA IR.94: "IMS Profile for Conversational Video Service".
  - [i.47] IETF RFC 3840: "Indicating User Agent Capabilities in the Session Initiation Protocol (SIP)".
  - [i.48] EENA 3.5.4.1: "Transnational Emergency Calls version 1.0".
  - [i.49] IETF RFC 5012: "Requirements for Emergency Context Resolution with Internet Technologies".
  - [i.50] IETF RFC 5764: "Datagram Transport Layer Security (DTLS) Extension to Establish Keys for the Secure Real-time Transport Protocol (SRTP)".
  - [i.51] IETF RFC 6497: "BCP 47 Extension T - Transformed Content".
  - [i.52] ETSI TS 127 007: "Technical Specification Group Core Network and Terminals; AT command set for User Equipment (UE)".

- [i.53] ETSI ES 203 178: "Functional architecture to support European requirements on emergency caller location determination and transport".
- [i.54] IETF Draft: "A Routing Request Extension for the HELD Protocol, draft-ietf-ecrit-held-routing".
- [i.55] ETSI draft ETSI ES 203 283: "Protocol specifications for emergency service caller location determination and transport".
- [i.56] BoR(15)135: "Update of the report on equivalent access and choice for disabled end-users". BEREC, 2015. Retrieved 29 November 2015.
- NOTE: Available at: [http://berec.europa.eu/eng/document\\_register/subject\\_matter/berec/download/0/5418-update-of-the-report-on-equivalent-access\\_0.pdf](http://berec.europa.eu/eng/document_register/subject_matter/berec/download/0/5418-update-of-the-report-on-equivalent-access_0.pdf).
- [i.57] RERC-TA "Real-time text interoperability; status and field trial", Gallaudet University 2015, Retrieved 8 December 2015.
- NOTE: Available at <http://tap.gallaudet.edu/IPTransition/TTYTrial/>.
- [i.58] ETSI TR 102 974: "Human Factors (HF); Telecommunications relay services".
- [i.59] REACH112 European project 238940 in the European Commission PSP programme 2009-2012.
- NOTE: Available at <http://www.reach112.eu>.
- [i.60] NEXES, Next Generation Emergency Services, Horizon 2020 project under grant agreement No 653337. Retrieved 29 November 2015.
- NOTE: Available at <http://nexes.eu>.
- [i.61] EMYNOS, nExt generation eMergencY CommuNicatiOnS, Horizon 2020 project under grant agreement no 663762. Retrieved 06 January 2016.
- NOTE: Available at <http://www.emynos.eu>.
- [i.62] ETSI NG112 Emergency Communications Plugtest.
- NOTE: Available at <http://www.etsi.org/news-events/events/977>.
- [i.63] Hellström, G., (2011): "Mixing multi-party real-time text for presentation in conference-unaware user agents", R3TF, Retrieved 6 December 2015.
- NOTE: Available at [http://www.realtimetext.org/sites/default/files/Files\\_and\\_Documents/Specifications/multiparty-real-time-text-mixer-2011-04-30.pdf](http://www.realtimetext.org/sites/default/files/Files_and_Documents/Specifications/multiparty-real-time-text-mixer-2011-04-30.pdf).
- [i.64] IMTC1013, 2014: "2SIP Video profile best practices", IMTC, Retrieved 6 December 2015.
- NOTE: Available at [http://portal.imtc.org/DesktopModules/Inventures\\_Document/FileDownload.aspx?ContentID=21794](http://portal.imtc.org/DesktopModules/Inventures_Document/FileDownload.aspx?ContentID=21794).
- [i.65] Recommendation ITU-T H.264: "Advanced video coding for generic audiovisual services".
- [i.66] Recommendation ITU-T G.711 (11/1988): "Pulse code modulation (PCM) of voice frequencies".
- [i.67] Recommendation ITU-T G.722 (09/2012): "7 kHz audio-coding within 64 kbit/s".
- [i.68] Recommendation ITU-T T.140 (02/2000): "Protocol for multimedia application text conversation - Addendum 1".
- [i.69] IETF RFC 4103 (2005): "RTP Payload for Text Conversation", G. Hellstrom and P. Jones.
- [i.70] FCC EAAC: "Emergency Access Advisory Committee (EAAC) Report on TTY Transition". March 2013, retrieved January 05, 2016.
- NOTE: Available at <https://www.fcc.gov/document/emergency-access-advisory-committee-eaac-report-tty-transition>.

## 3 Definitions and abbreviations

### 3.1 Definitions

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

Many of the definitions have been copied from ETSI TS 101 470 [i.2]. If there are any discrepancies between the definitions that follow and those in ETSI TS 101 470 [i.2], then the definitions that follow apply to the present document.

**additional data:** additional call related information provided by various entities in the path of the call in accordance with the data structures and mechanisms described in Draft IETF Additional Data Related to an Emergency Call [i.38]

**address:** identifier of the destination of a call containing only numbers, or a wider range of characters depending on the rules established by the application service provider

**application service provider:** organization or entity that, via a serving network, provides application-layer services, which may include voice, video and text communication

**assisting services:** services invoked during a call, assisting the Total Conversation user or the call-taker with specific tasks in the call

NOTE: Such tasks can for example be language translations, relay service or expert advice.

**call-taker:** agent at the PSAP that accepts calls and may dispatch emergency help

NOTE: Sometimes the functions of call taking and dispatching are handled by different groups of people, but these divisions of labour are not generally visible to the caller and thus do not concern us here (definition is copied from IETF RFC 5012 [i.49])

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

NOTE: An ESInet delivers emergency requests and corresponding data to emergency services providers and facilitates communication between emergency service providers and other supporting entities. An ESInet is typically deployed to support a set of PSAPs and other public safety agencies on a geographic basis. A given PSAP, or other appropriate entity, may connect to one or more ESInets. ESInets may be interconnected to facilitate emergency event handling and other related interactions (from EENA NG112 LTD [i.1]).

**home environment:** environment responsible for overall provision and control of the Personal Service Environment containing personalized information defining how subscribed services are provided and presented towards the user

NOTE: Each subscriber of the Home Environment has her own Personal Service Environment. The Personal Service Environment is defined in terms of one or more User Profiles.

**IETF SIP:** session control environment for calls, using the IETF RFC 3261 [i.16] and related protocols in the IP networks

NOTE 1: The above refers to an environment outside the scope of IMS.

NOTE 2: In ETSI TS 101 470 [i.2], the term "basic SIP" is used.

**IP Multimedia Subsystem (IMS):** standardized architectural framework for delivering Internet Protocol (IP) multimedia services, as described in ETSI TS 122 228 [i.10]

**modalities:** methods for human expression and perception of communication

NOTE: Examples are written, signed and spoken languages, pictures, gestures, etc.

**multi-party call / conference:** real-time communication session with more than two participants where media sent from participants are distributed for presentation among the participants in the call

**NG112:** next generation 112 emergency services provided via the Emergency Services IP network (ESInet)

**non-Total Conversation emergency session:** voice-only IP based emergency session that is not a Total Conversation session

**Public Safety Answering Point (PSAP):** physical location where emergency calls are received under the responsibility of a public authority

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

**relay node:** functional entity providing a conference bridge to multiple parties, including and not limited to the Total Conversation user and the relay service, engaged in a total conversation session

NOTE: In the context of Total Conversation emergency session, the PSAP is also considered as a party using the relay Node.

**relay service:** telecommunications service that enables users of different modes of communication e.g. text, sign, speech, to interact by providing conversion between the modes of communication, normally by a human operator

NOTE: A type of assisting service (definition from ETSI ES 202 975 [i.5]).

**serving network:** entity that provides the user with access to the services of the home environment

**Total Conversation:** audiovisual conversation service providing bidirectional symmetric real-time transfer of motion video, Real-Time Text and voice between users in two or more locations

NOTE: Definition from Recommendation ITU-T F.703 [i.18].

**Total Conversation emergency service:** emergency service capable of handling total conversation emergency sessions

**Total Conversation terminal:** user terminal capable of being used for total conversation

**Total Conversation user:** individual taking advantage of the total conversation service

**UICC:** physically secure device, a Universal Integrated Circuit Card (or 'smart card'), that can be inserted and removed from the terminal

NOTE: It may contain one or more applications. One of the applications may be a USIM.

**user profile:** set of information necessary to provide a user with a consistent, personalized service environment, irrespective of the user's location or the terminal used (within the limitations of the terminal and the serving network)

## 3.2 Abbreviations

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

3GPP	3 <sup>rd</sup> Generation Partnership Project
3PCC	3 <sup>rd</sup> Party Call Control
AMR	Adaptive Multi Rate
AT command	Attention Command
AVP	Audio Video Profile
AVPF	Audio Video Profile with Feedback
BOM	Byte Order Mark
CEPT	European Conference of Postal and Telecommunications Administrations
CN	Core Network
CS	Circuit Switched
CSG	Closed Subscriber Group
DTLS	Datagram Transport Layer Security
DTMF	Dual Tone Multi Frequency
EC	European Commission
ECC	Electronic Communication Committee (of the CEPT)
ECRIT	Emergency Context Resolution with Internet Technologies
EENA	European Emergency Number Association
ESInet	Emergency Services IP Network
ESRP	Emergency Service Routing Proxy
GSMA	GSM Association

GTT	Global Text Telephony
GTT-IP	Text Telephony in IMS via the Real-Time Text protocol over RTP
HELD	HTTP Enabled Location Delivery
HF	Human Factors
ICS	IMS Centralised Services
ICT	Information and Communications Technology
IM	IP Multimedia
IMS	IP Multimedia Subsystem
IP	Internet Protocols
JPEG	Joint Photographic Experts Group
LA	Location Area
LOST	Location to Service Translation protocol
MES	Multimedia Emergency Session
MTSI	Multimedia Telephony Service for IMS
NAT	Network Address Translation
NENA	National Emergency Number Association
OTT	Over The Top
PLMN	Public Land Mobile Network
PS	Packet Switched
PSAP	Public Safety Answering Point
PSTN	Public Switched Telephone Network
RTCP	Real Time Control Protocol
RTP	Real Time Protocol
RTT	Real Time Text
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SLIM	Selection of Language for Internet Media
SRTP	Secure Real Time Protocol
TC	Total Conversation
UA	User Agent
UE	User Equipment
UICC	Universal Integrated Circuit Card
URI	Universal Resource Identifier
URN	Uniform Resource Name
USIM	Universal Subscriber Identity Module
WebRTC	Web real-time communications
XML	Extensible Markup Language
ZRTP	Z Real Time Protocol

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## 4 Background

Total Conversation is a conversational service providing real-time communication in video, real-time text and audio. For the purpose of the current document, also subsets of these media are included in the concept.

Descriptions of the general usefulness of Total Conversation can be found in the use cases of Recommendation ITU-T F.742 [i.17].

The basic functionality of the Total Conversation emergency service is to provide its Total Conversation user with a way to make emergency calls and also to receive a call-back from the PSAP with Total Conversation if needed. This allows the Total Conversation user to communicate in a conversational way using combinations of video, real-time text and audio. Each media type is conveyed in real-time bi-directional manner between the participants in the call.

The use of Total Conversation in emergency service sessions is described in ETSI TR 103 170 [i.3], and specified in ETSI TS 101 470 [i.2].

The main motivation for use of Total Conversation for emergency communications is that the availability of three media provides improved access for all in all environments, and in particular those Total Conversation users with disabilities. Some examples are given in table 1.

**Table 1: Examples of supported media in Total Conversation**

Media	Description
Video	<ul style="list-style-type: none"> <li>• useful for visual assessment of conditions at site of emergency</li> <li>• can carry sign language between call-taker or interpreter and hard-of-hearing or deaf user</li> <li>• can improve intelligibility of/to caller in noisy environment</li> <li>• view of call-taker can convey calming impression to Total Conversation user in distress</li> </ul>
Real-Time Text	<ul style="list-style-type: none"> <li>• useful for short items that need to be remembered or for rapid provision of exact spelling for e.g. addresses</li> <li>• can provide part or all of dialogue to/from hard-of-hearing or deaf user</li> <li>• useful for clarification of key words from Total Conversation user calling from noisy environment</li> </ul>
Audio	<ul style="list-style-type: none"> <li>• spoken dialogue between Total Conversation user and call-taker</li> <li>• spoken interpreted dialogue between call-taker and interpreter</li> <li>• audio information from the emergency-site - for assessment</li> <li>• partial spoken dialogue when used in combination with real-time text or sign language</li> </ul>

A brief overview of the situation in Europe at the time of writing for Total Conversation and emergency service access for persons with disabilities can be found in Annex D.

## 5 Existing requirements, recommendations, and assumptions

### 5.1 General

Specific requirements and recommendations for Total Conversation access to emergency services can be found in ETSI TS 101 470 [i.2] and ETSI TR 103 170 [i.3] respectively. The following clause provides a brief overview of those requirements and recommendations. They form the basis for analysis of the state of standards and specifications supporting these requirements and recommendations. For more details of the requirements see clause 5 of ETSI TS 101 470 [i.2] and the related outstanding issues in Annex B of the present document.

In addition, requirements for support of Multimedia Emergency Sessions and for IMS Multimedia Telephony which fulfil the service requirement for Total Conversation can be found in clause 10 of ETSI TS 122 101 [i.9], in clause 4 of ETSI TS 122 173 [i.6], and in ETSI TS 126 114 [i.7]. Further requirements relating to the negotiation of language at session invocation can be found in ETSI TS 122 228 [i.10]. More general architectural requirements for IMS emergency sessions can be found in ETSI TS 123 167 [i.8]. The IMS requirements to support Total Conversation for Emergency Communications are summarized in Annex C of the present document.

### 5.2 Summary of existing requirements

The emergency service capable of handling Total Conversation emergency calls will provide its Total Conversation users with a way to make and receive emergency calls using combinations of video, Real-Time Text and audio. The requirements applicable to non-Total Conversation emergency (e.g. voice calls) also apply to Total Conversation emergency calls.

The following Total Conversation specific requirements have also been identified in ETSI TS 101 470 [i.2]:

- **Call initiation.** A Total Conversation user terminal intended to support emergency calls should recognize an emergency call and perform the specific steps necessary for establishing the call with the desired media.
- **Call addressing.** A consistent address should be assigned for Total Conversation Emergency sessions. Using just the emergency number (e.g. 112) in the Total Conversation user terminal as the only address should result in an emergency call. If the Total Conversation user is provided with other methods for entering a destination address and other conditions for the call, e.g. the need to include a relay service, then that form should also be possible to use with the emergency number (according to the destination address).

- **PSAP assignation.** Total Conversation Emergency sessions should follow the principle of ETSI TR 102 180 [i.4] that the session should be connected to the most appropriate PSAP, bearing in mind that the PSAP may be assigned based on the location, media, and language capabilities of the call-taker.
- **Media Support.** Support for full Total Conversation comprising three media should be implemented in the PSAP call-taker workstations used for Total Conversation access as per EENA NG112 LTD [i.1]. Total Conversation user terminals may make emergency sessions using the full media set or a subset thereof.
- **Assisting services.** The inclusion of assisting services such as those providing language interpretation should be supported. These may include relay services that translate between modalities of human communication. Mechanisms should be provided for the Total Conversation user and the PSAP to be able to invoke such services at the time of session establishment, or to add them during the session.
- **Caller identity preservation.** For caller identity, the principle of ETSI ES 202 975 [i.5] should be followed, i.e. the inclusion of a relay service in the call should not modify the caller identity information for the Total Conversation user. Rather the caller identity should be forwarded transparently by the relay service to PSAP.
- **Location information preservation.** The inclusion of a relay service in the session should not modify the location information for the Total Conversation user. Rather, the user location information should be forwarded transparently by the relay services to the PSAP.
- **Call-back.** In the case of call-back, the PSAP should ensure that the same media, and if possible and applicable the same relay service, are used for the call-back as in the original session.
- **Involvement of other entities.** A PSAP may involve a separate emergency control centre or a third party to take the necessary action on the emergency case. Means should be provided to transfer the Total Conversation emergency session to such an entity, maintaining all the media supported by the entity to which the session is transferred.
- **Call logging.** Means should be provided at the PSAP to log information about Total Conversation emergency sessions including media used, identity and address of any assisting services included, etc. This requirement is further elaborated in EENA NG112 LTD [i.1] and is out of scope of the present document.
- **Call recording.** Means should be provided at the PSAP to record all media used in a session without degradation, for later retrieval and play-back. This requirement is further elaborated in EENA NG112 LTD [i.1] and is out of scope of the present document.
- **Security.** Security and privacy requirements are also applicable to any relay service invoked during the session. In particular, data protection and privacy requirements should be respected. Both technical means and contractual obligations are needed to fulfil the requirements.

## 5.3 Study and Analysis Assumptions

### 5.3.1 General

The following clauses list the assumptions on which the analysis of the present document are based.

### 5.3.2 Assumptions related to Total Conversation for Emergency Communication

The following general assumptions are made:

- Total Conversation for emergency communication is also referred to as "Total Conversation for Emergency services" in the present document.
- Total Conversation as a concept by itself exists already in standards. It is not further studied in the present document unless impacts on the Total Conversation concept are found to be necessary in order to support its use for emergency communications.
- Basic requirements and implementations for Total Conversation used as a basis for Total Conversation for emergency services are not repeated in the present document.



### 5.3.3 Assumptions about the PSAP and Relay node networks

The following assumptions are made:

- Future deployments will comprise NG112 PSAPs. Each NG112 PSAP is connected to a SIP based network, called the ESInet, see EENA NG 112 LTD [i.1]. This means that IMS based PSAPs are not considered in the present document. Interworking between IMS and PSAPs using IETF SIP is discussed in ETSI TS 123 167 [i.8].
- NG112 PSAPs will support all the media needed for Total Conversation for Emergency Communication, see clause 5.2. However, during migration to NG112 deployments some PSAPs might remain that do not provide full media support.
- A non NG112 PSAP is connected to CS based network: A gateway is needed to perform interworking between the SIP signalling of the IP based access network and the CS based signalling of the non NG112 PSAP network, as well as media interworking which might reduce the quality of service. Video is not expected to be supported in non NG112 PSAPs, which will also reduce the quality of experience.
- Relay nodes are connected to SIP based networks.

NOTE: The legacy PSAP, supporting PSTN, and how it connects to a multimedia relay service node is not dealt with in detail in the present document.

### 5.3.4 Assumptions about the Total Conversation user terminal and access network

The following assumptions are made:

- The Total Conversation user terminal supports multimedia conversational services (see Recommendation ITU-T F.703 [i.18]) and is connected to an IP based access network (example IETF SIP or IMS).

NOTE: The Total Conversation user terminal that is connected to an access network interfacing with a CS based PSAP network through an interworking unit is likely to experience service limitations, e.g. Video is not likely to be supported. This type of access is covered in ETSI TS 101 470 [i.2], clause 5.2.4.

### 5.3.5 Assumptions on the routing of Total Conversation for emergency communication calls

The following assumptions are made:

- Total Conversation emergency sessions are routed to the most appropriate PSAP, i.e. one that is in the jurisdiction of the incident. This is also valid for users roaming outside their home countries; and
- routing within the ESInet and all additional services provided by the ESInet, such as providing Location information, are not studied in the present document. This is because the present document looks mainly at access networks and service provider networks, and their interfaces to ESInets.

NOTE: The ESInet and the services it provides is based on SIP and IETF recommendations and EENA specifications.

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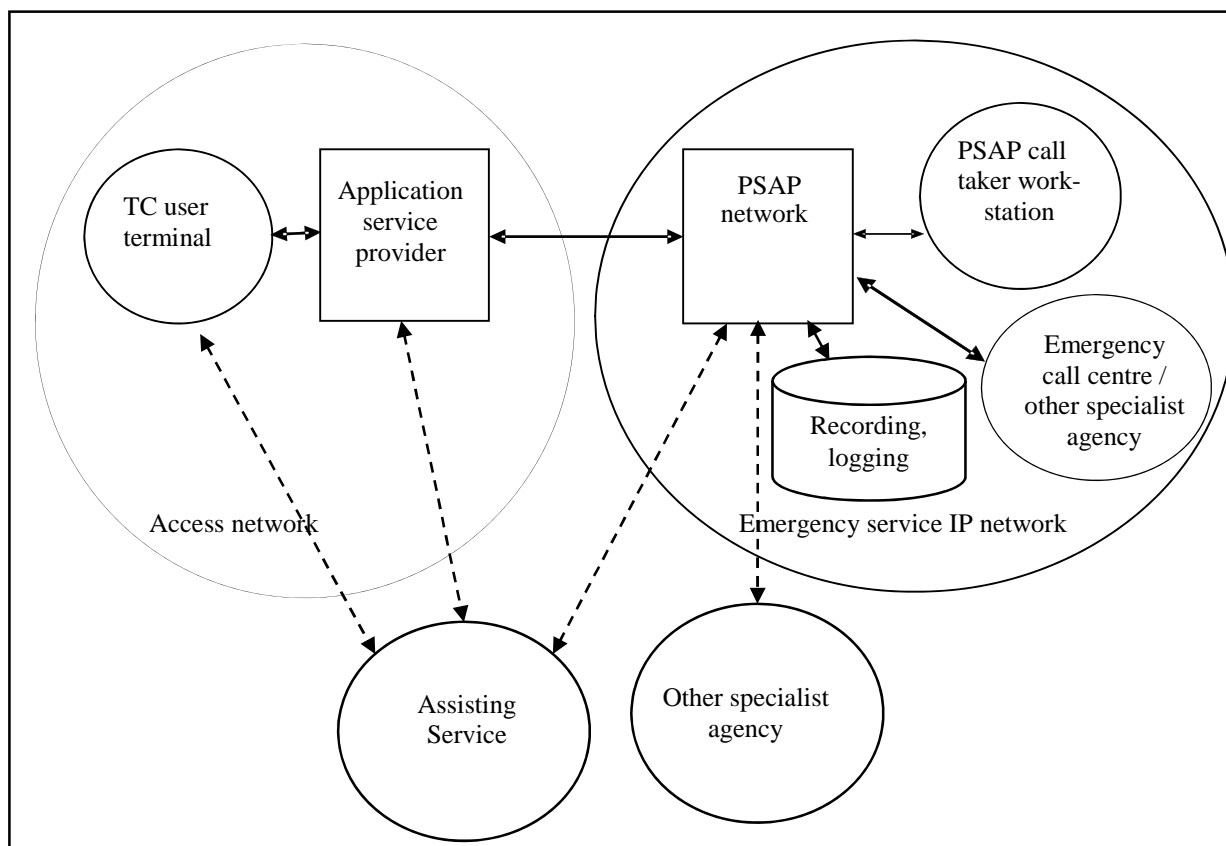
## 6 Architecture principles

### 6.1 General

The requirements and recommendations for Total Conversation for Emergency Communications span over a wide range of entities between which the technical systems for emergency service session handling need to interoperate according to the technical and operational standards and specifications.

### 6.2 Functional Architecture

The functional architecture is shown in Figure 1.



**Figure 1: Functional Architecture**

The functional architecture comprises the following elements:

- 1) **The Total Conversation user terminal:**  
The Total Conversation terminal that the Total Conversation user uses to contact the emergency service centre (PSAP).

NOTE 1: This is usually also the terminal that is used for everyday calling.

- 2) **Access network:**  
The network handling the Total Conversation call between the Total Conversation user, the Application Service Provider and the emergency service network.

NOTE 2: This is also the network handling the everyday calling for conversational services by the Total Conversation user.

- 3) **Emergency service IP network:**  
An IP based network connecting one or more NG112 PSAP networks for conveying emergency service calls and enabling collaboration between PSAPs and other functional units in emergency services. This type of network is called ESInet.
- 4) **PSAP:**  
The point within the emergency service network initially answering and handling an emergency service call. The PSAP collects information and decides on the next step in handling of the call. That can involve linking in external expert services, translation services and first responders for the call, or transferring the call to another party.
- 5) **PSAP network:**  
The network connecting a number of PSAPs in the same location or administration area together.
- 6) **Emergency service recording and logging unit:**  
The functional unit within an emergency service network that handles recording of emergency service calls and retrievals of the calls.

- 7) **Assisting service.**  
Assisting services including translation services, relay services and relay nodes which can be invoked when needed during a Total Conversation emergency session. These could be invoked by the Total Conversation user terminal, by the Application Service Provider in the access network, in the ESInet or by the PSAP.
- 8) **Other specialist agency.**  
Other specialist agencies may be emergency specific (e.g. hostage negotiation) and reside in the ESInet or provide specialist advice not necessarily specific to emergency situations (e.g. chemical spillage) and therefore reside outside of the ESInet.

## 6.3 Call scenarios

### 6.3.1 General

At the top level, clause 5.6 of ETSI TS 101 470 [i.2] considers two main call scenarios as follows:

- 1) TC calls with no assisting service invoked.
- 2) TC calls with assisting service(s) invoked.

Where assisting services are included they can be invoked manually or automatically at the time of Total Conversation emergency session establishment by either the Total Conversation user, the Total Conversation application service provider, or by the PSAP. If the need for an assisting service arises whilst the Total Conversation emergency session is ongoing, then the assisting service could potentially be invoked by the PSAP, by the Total Conversation application service provider, or by the Total Conversation user depending on the availability of a suitable/optimum implementation.

Clauses 6.3.2 to 6.3.9 describe aspects of Total Conversation sessions using the architecture of clause 6.2, which might need special handling over and above those for VoIP emergency sessions.

Possible routing scenarios of Total Conversation emergency sessions toward the PSAP are also considered in clauses 6.3.2 to 6.3.9.

### 6.3.2 Baseline session (without any assisting service)

The baseline session is a Total Conversation session comprising the media audio, video, and real-time text, or a subset thereof, which is established between the Total Conversation user and the most appropriate PSAP. The Total Conversation user should be able to request to add or subtract media from the session once it is in progress.

It is assumed that routing towards the PSAP is based on the Total Conversation user's location, that all PSAPs in the emergency services network are capable of supporting the full media set associated with Total Conversation, and that no assisting services are needed.

**Supporting functionality specific to Total Conversation for Emergency Communications: None.**

### 6.3.3 Routing based on media capability of PSAP

In the case where not all PSAPs in the emergency services network support the full Total Conversation media set, it is desirable to route Total Conversation sessions towards PSAPs that support Total Conversation. In this case, it will be necessary to be able to differentiate between a Total Conversation emergency session and a voice only emergency session.

**Supporting functionality specific to Total Conversation for Emergency Communications: Total Conversation emergency session identification.**

### 6.3.4 Routing based on language / modality

If the Total Conversation user is not in their home network, and/or prefers to communicate with the PSAP in a language / modality other than the default language / modality of the country, then the Total Conversation user should be able to indicate the preferred language or modality at the time of session establishment. The session should be routed to the most appropriate PSAP taking that preference into consideration.

This could include the case where the Total Conversation user prefers to send information using one modality (e.g. speech), and receive information using another (e.g. real-time text, or sign-language). In this case, the Total Conversation user should be able to indicate the preferred method of communication for each direction of the session, so that the call-taker can be selected appropriately or an appropriate assisting service be invoked.

Routing of a Total Conversation session could potentially be directly to a PSAP supporting the Total Conversation user's preferred language / modality, or via an assisting relay service.

**Supporting functionality specific to Total Conversation for Emergency Communications: Language / modality indication for each direction of communication.**

### 6.3.5 Routing via an assisting service

The Total Conversation user could include an assisting service manually, e.g. by adding the assisting service address to the session routing path at the time of the session initiation, or automatically through pre-configuration of the Total Conversation terminal (e.g. including the necessary relay service in the routing for all sessions). In this case, the assisting service should be indicated in the signalling from the Total Conversation user's terminal.

The assisting service could also be included by the application service provider on behalf of the Total Conversation user at the time the session is established. In this case, the application service provider will need to know in advance the address of the assisting service to be included. This information might be configured for each user and stored as part of the User Profile for Total Conversation users. The assisting service may originate the emergency session on the Total Conversation user's behalf.

In the case where the emergency call is routed to the PSAP via an assisting service, the Total Conversation user identity and location information should be relayed transparently to the PSAP, together with any Additional Data as per the IETF draft on Additional Data [i.38] included by the Total Conversation user terminal.

**Supporting functionality specific to Total Conversation for Emergency Communications: Assisting service invocation, location information preservation, Total Conversation user identity preservation.**

### 6.3.6 Conferencing with third party including assisting relay services

The Total Conversation user should be able to invoke an assisting relay service adding that service to a multi-party conference at the time of the session initiation. This could be achieved either manually, or automatically through pre-configuration of the Total Conversation terminal (e.g. invoking the necessary relay service for all sessions).

The assisting service could also be invoked by the application service provider establishing a multi-party conference on behalf of the Total Conversation user at the time the session is established. In this case, the application service provider will need to know in advance the nature of the assisting service to be invoked. This information may be configured for each user and stored as part of the User Profile for Total Conversation users.

The PSAP should be able to enlist the help of a specialist third party agency or assisting relay service in order to resolve the emergency either from the start of an emergency session or during the ongoing session (e.g. specialist service providing specific incident related expertise, or a relay service for language / modality reasons). In this case, the PSAP should be able to establish a conference session including the third party, sharing either all or a subset of the available media, depending on the third party media support capability and the preferences of the PSAP, as well as the regulatory requirements pertaining to the PSAP and the third party.

In the case where an assisting service is already included in the session, that service should if still needed, remain in the session after the conference has been established.

**Supporting functionality specific to Total Conversation for Emergency Communications: Interworking of media, preservation of existing assisting service.**

### 6.3.7 Transfer of session from PSAP

During the course of a Total Conversation emergency session, the PSAP may wish to transfer responsibility for resolving the emergency to a third party (e.g. another PSAP or external agency). For a Total Conversation emergency session all established media should be transferred. If the third party does not support all established media, then a subset of the established media may be transferred, with an appropriate notification sent to the Total Conversation user and any other participants in the call.

In the case where an assisting service is already included in the session, it should if still needed, remain in the session after the session has been transferred. The PSAP should be able to add an assisting service to the session as part of the to the transfer procedure.

**Supporting functionality specific to Total Conversation for Emergency Communications: Interworking of media, preservation of existing assisting service, assisting service invocation.**

### 6.3.8 Transfer of session to more appropriate Total Conversation user terminal

A Total Conversation user may establish an emergency session using a terminal that does not support, or does not adequately support, all media (e.g. audio only, or terminal with small / low resolution video screen). Later in the call the Total Conversation user may be in a position to access a more capable terminal. In that case, it could be useful to transfer certain media (e.g. video) or the whole session to a more media capable terminal in order to better communicate with the PSAP (if necessary adding media in the process). The reason could for example be to show the scene of the emergency via video media, or to add the possibility for the Total Conversation user to lip-read the call-taker, or to get access to a more convenient keyboard for typing real-time text during the call. In the case where an assisting service is already included in the session, it should if still needed, remain in the session after the transfer.

NOTE: This is no different in principle from a non-emergency TC session, excepting the additional problems that might exist for an IMS terminal when it is IMS emergency registered only (see ETSI TS 123 167 [i.8]).

**Supporting functionality specific to Total Conversation for Emergency Communications: Interworking of media, preservation of existing assisting service.**

### 6.3.9 Call-back

The PSAP should be able to call-back to the Total Conversation user. In this case, the session should by default: be established using the same media as used for the initial emergency session; include any assisting services previously present in the session; and be routed to the terminal used by the Total Conversation user at the time of session establishment.

The PSAP should also be able to add media to the established call-back with variations from the default set of media.

If the Total Conversation user has access to more than one terminal, they may inform the PSAP of the identity of the other SIP registered terminal in order to direct the PSAP to call-back towards that terminal rather than that from which the Total Conversation emergency session was initially established. In that case, the PSAP should be able to establish the call-back with additional media depending on the Total Conversation user's indicated preferences and the terminal capability.

PSAP call-back is considered as a normal call and can only be differentiated from other non-emergency SIP calls through the "SIP PSAP Callback Indicator" as described in IETF RFC 7090 [i.43]. This indicator is used accordingly in the SIP based PSAP call-back. In the IMS core, the "SIP PSAP Callback Indicator" may be removed by IMS nodes or by the Total Conversation user terminal as no trust domain for the indicator exists in the network (see ETSI TS 124 229 [i.22]).

As call-back is a normal call, the Total Conversation user terminal needs to be registered with the service provider to receive the service/call. Therefore in IMS case normal IMS registration is needed to receive call-back as described in ETSI TS 124 229 [i.22].

It would be useful for user agents and proxies to deactivate any features that interfere with success of call-back for a predefined period of time (example 30 minutes), e.g. as described in IETF RFC 6881 [i.11] and IETF RFC 6443 [i.14] except for actions with the goal to invoke relay services without interruption of the session with the user.

**Supporting functionality specific to Total Conversation for Emergency Communications: Total Conversation identity preservation, interworking of media, preservation of existing assisting service.**

## 6.4 Implementation technologies

### 6.4.1 General

Total Conversation is implemented using multimedia session control and media protocols. Therefore the implementation of multimedia protocols is needed in all parts of the emergency service session chain.

The technologies most often used or discussed for conversational communication and for emergency service implementation are briefly presented here. They are further analysed in the later clauses of the present document.

### 6.4.2 IETF SIP

IETF SIP is used for Total Conversation user terminals, for SIP communication services, for NG112 emergency service access and for NG112 PSAPs.

### 6.4.3 IMS

IMS is specified by 3GPP and used for terminals and network entities. It is specified for both mobile, fixed and WiFi network access.

### 6.4.4 Web based technologies

Web based technologies are emerging as suitable for emergency service access by addition of standardized real-time communication features in web browsers.

WebRTC is one such technology for web based real-time communication. Work in standardization and implementation is going on in W3C, IETF and 3GPP for this technology. If eventually supported by the main web browser and operating systems manufacturers, this technology could enable additional application specific ways to communicate.

WebRTC use in IMS environment is specified in ETSI TS 122 228 [i.10], but WebRTC use for emergency sessions is not currently specified. It is therefore not further considered in the present document.

### 6.4.5 PSTN

In PSTN, only voice and text telephony is feasible, thus allowing for a service with some similarities to a limited subset of Total Conversation.

The use of text telephony in PSTN is in general decline (see e.g. FCC EAAC Report on TTY Transition [i.70]), but it remains operational in a few countries, where it can provide access to emergency services.

For general use in PSTN, text telephony makes use of various sub-modes of Recommendation ITU-T V.18. [i.19]. The access to PSTN based emergency services is described in the same standard.

EENA NG 112 LTD [i.1] describes the conversion between PSTN text telephony and SIP based emergency service networks and PSAPs.

If a Total Conversation user calls emergency services and the most appropriate emergency service PSAP is PSTN based, and the Total Conversation user in that situation is depending on video, e.g. for sign language use, then a mechanism is needed to provide sign language access via an assisting service in IMS or IETF SIP. This mechanism should convey the call between the signing Total Conversation user and the talking PSAP call-taker. ETSI TS 101 470 [i.2] discusses such situations.

## 6.5 Supported functions in Total Conversation for Emergency Communications

### 6.5.1 General

This clause describes the functions outline in the call scenarios of clause 6.3 in more detail, and refers to some of the documents in which they are specified. It also clarifies the functional aspects that are unique for Total Conversation as opposed to speech only emergency sessions.

Total Conversation for emergency communications can for the most part be implemented using existing standardized means specified by IETF and 3GPP respectively for emergency calls and for multimedia calls - see IETF RFC 6881 [i.11], IETF RFC 6443 [i.14], IETF RFC 4190 [i.15], and ETSI TS 122 173 [i.6], ETSI TS 126 114 [i.7], ETSI TS 123 167 [i.8] and ETSI TS 122 101 [i.9].

Some specific functions needed to support Total Conversation for emergency communications are identified in clause 6.3. The underlying architectural aspects that facilitate these functions are further discussed the clauses that follow. Potential solutions for implementation of the functions to support Total Conversation for emergency communications are discussed under clause 7.2 of the present document.

## 6.5.2 Identification of the Total Conversation emergency session

Total Conversation emergency session requests should be differentiated from other emergency session requests. This could be achieved through either explicit or implicit indication of the Total Conversation nature of the emergency session.

The Total Conversation emergency session indication can be used by the Application Service Provider and ESInet for identifying the Total Conversation emergency session and for routing it to the most appropriate PSAP. The Total Conversation emergency session indication can also help the receiving PSAP to differentiate the emergency session as a Total Conversation emergency session. The most appropriate PSAP needs to be capable of receiving and handling the Total Conversation emergency session appropriately, i.e. either supporting the Total Conversation users' preferred language and modality directly or being able to request assistance from relay service, other PSAP or third party.

The Total Conversation emergency service indication should be included in the signalling by the Total Conversation user terminal or by the Application Service Provider. The Application Service Provider should if necessary be able to differentiate and route the Total Conversation emergency session based on this indication towards the most appropriate PSAP that supports Total Conversation. Total Conversation by definition involves conversational real-time text, audio and video. The SIP signalling for a Total Conversation session therefore has to contain information (in the SDP) about the codecs necessary to support those media. Given the presence of such an implicit indication in SIP signalling for a Total Conversation session, there appears no necessity to include an additional explicit indication in the signalling.

## 6.5.3 Naming and addressing

### 6.5.3.1 Caller / Originator Identity

The Total Conversation user terminal should provide its identity to the PSAP to facilitate PSAP Call-Back as described in IETF RFC 7090 [i.43], IETF RFC 6881 [i.11] and ETSI TS 124 229 [i.22]. The caller identity is needed to allow the PSAP to call-back to the Total Conversation user terminal. The Application Service Provider (IMS or IETF SIP) should provide the caller identity (Directory Number/ SIP URI) to the PSAP in a similar way as for a non-Total Conversation emergency session. The caller identity for a Total Conversation session should be presented to the PSAP in the same way as for a non-Total Conversation emergency session.

NOTE: Tel-URI will be provided for the case where the PSAP is CS/PSTN based.

### 6.5.3.2 Relay Service identification and addressing

Relay services to be included in emergency sessions should be identified by a SIP URI. A known relay service identity can be used directly by a Total Conversation user, e.g. whose terminal is configured to send all outgoing sessions to a specific relay service, or by an Application Service Provider acting on the identity included in the Total Conversation User Profile. When a relay service is included in a session by the Total Conversation user or Application Service Provider, its identity also needs to be provided to the PSAP to allow its inclusion in any session established as the result of call-back.

NOTE: The relay service needs to provide its address/identity to the PSAP for a number of reasons, as described in the present document, as well as to enable call-back. In case of call-back it may not be possible to engage the same relay communications agent or interpreter, however the same relay-service should be contacted.

Consideration also needs to be given for how to address both the relay service and the destination (in this case the emergency service.) A common convention as per ETSI ES 202 975 [i.5] that works when the destination can be addressed by a number is to use only a domain name for addressing the relay service and have the destination number as the user part. But in the emergency service case, the destination address is first on the form urn:service:sos, and after routing resolution, it is in the full SIP:user@domain form. A possibility might be to include the relay service identity in the Route field.

If a specific relay service is to be invoked by the PSAP, the SIP URI of that relay service will need to be either communicated directly to the PSAP, e.g. by the Total Conversation user, or obtainable by the PSAP through some other means, e.g. through interrogation of a database using previously acquired knowledge about the Total Conversation user's communication needs.

It is likely that in most countries, deployed relay services will handle a mixture of emergency and non-emergency sessions. However, in some countries it is possible that emergency only relay services will be deployed. In that case, it would be possible for them to be addressed using emergency specific URIs.

General requirements for relay service deployed in ICT networks can be found in ETSI ES 202 975 [i.5]. However, that document does not include provision for how relay services can be identified by PSAPs for inclusion in emergency sessions. Clause 7.2.5 of the present document discusses potential solutions for obtaining the identity of a specific relay service.

## 6.5.4 Language considerations

The preferred communication language of the Total Conversation user should be identifiable for the emergency session to be established.

The language / modality indication can be used by the network operator/service provider for identifying the languages / modalities by which the Total Conversation user is able to communicate. It can be used in the network for routing sessions to PSAPs operators with whom the Total Conversation user will be able to converse. It could also be used in combination with other session attributes by the Application Service Provider or the PSAP to identify when it may be necessary to invoke an assisting service to facilitate communication between the Total Conversation user and the call-taker. E.g. during migration to NG112 when the Total Conversation user SDP offers video media, and the Total Conversation user indicates a preference for sign-language, but the PSAP cannot support video.

Total Conversation users may also have different modality preferences for receiving and sending information. E.g. a hard of hearing user may prefer to receive information using sign-language, but to send information using speech. For this reason, the Total Conversation user should also be able to indicate modality preference for each direction of communication.

ETSI TS 122 228, clause 7.5.2 [i.10] currently specifies that user equipment should be able to negotiate the user's desired languages and modalities, and that the service provider 'shall' be able to pass this information between end points. There are currently no requirements in ETSI TS 122 228 [i.10] for the service provider to be able to route calls based on language / modality indications.

IETF RFC 4596 [i.12] provides guidelines for usage of the SIP caller preferences extension specified in IETF RFC 3841 [i.13] and IETF RFC 3840 [i.47]. This includes the ability to indicate the languages / modalities supported by the caller, the caller's proficiency in those languages / modalities, and how this information can be used to route within a network to called parties supporting those languages / modalities.

**NOTE:** There is currently no solution specified to allow users to express language / modality preference for sending/receiving information. IETF SLIM has been looking at this issue in draft-ietf-slim-negotiating-human-language [i.40]. However, the work is not completed at the time of writing the present document and there are no requirements for this in current 3GPP or IETF specifications.

## 6.5.5 Location and routing

### 6.5.5.1 Location information

Location information is needed in the ESI-net to be able to route the Total Conversation emergency session to the most appropriate PSAP. It is also needed at the PSAP and by the emergency services first responder to know the location of the caller in order to provide assistance in the most efficient manner. The location information for a Total Conversation session should be presented to the PSAP in the same way as for a non-Total Conversation emergency session.



The requirements and mechanisms for obtaining and providing location information for emergency sessions in IP based systems are specified for IETF SIP and IMS respectively in IETF RFC 6881 [i.11], EENA NG112 LTD [i.1], and ETSI TS 203 178 [i.53], and in ETSI TS 122 101 [i.9], ETSI TS 122 071 [i.20], ETSI TS 123 167 [i.8], ETSI TS 123 271 [i.21] and ETSI TS 124 229 [i.22].

### 6.5.5.2 Routing information

Routing information is needed by the Application Service Provider to identify the entry point of the ESInet that connects to the most appropriate PSAP. When the caller, the Application Service Provider and the ESInet are all in the same jurisdiction, this routing information can be hard coded into routing proxies or made easily obtainable through proprietary means. However, it may still be necessary to verify the location of the Total Conversation user, e.g. through interrogation of a location server. When a caller is roaming in an IETF SIP network, their Application Service Provider may not be in the same jurisdiction as the ESInet to which the session should be routed. If that is the case, it will be necessary to also obtain reliable routing information in a secure way from a location server.

In the case of routing of an IETF SIP emergency session via an Application Service Provider that is not in the same country as the calling Total Conversation user, routing information should be obtained by the Application Service Provider as described for the Voice Service Provider in ETSI ES 203 178 [i.53] using the HELD routing extension of draft-ietf-ecrit-held-routing [i.54] as specified in ETSI draft ETSI ES 203 283 [i.55].

In the case of Total Conversation user is IMS roaming, emergency calls are always connected via the visited network, see ETSI TS 123 167 [i.8].

### 6.5.6 Assisting services

The invocation of assisting services can help Total Conversation users in different ways. It can help by providing instant translation between spoken languages (e.g. English to French) as well as support for persons with disabilities wishing to contact emergency services (e.g. deaf/hard-of-hearing using other or complementary modalities than speech).

Assisting services can be implemented in different ways in different countries. In some countries, a dedicated PSAP may be assigned for Total Conversation users requiring spoken language translation or sign-language interpretation. However, in many countries a separate assisting service may need to be included in the session in order to enable full communication with the emergency services. The requirements for support of assisting services as described in ETSI ES 202 975 [i.5] are specified in ETSI TS 101 470 [i.2].

The goal of including an assisting service is that the Total Conversation user should be efficiently served in a language and modality that is convenient and manageable, and that a PSAP with authority to act on the emergency situation (i.e. one that is in the same country as the Total Conversation user) is able to handle the call in a supported language and modality. The need for assisting services could be indicated in a number of ways such as:

- in the signalling from the user equipment;
- in the Total Conversation user profile; or
- explicitly by the Total Conversation user to the call-taker once the session has been established.

The indication may be acted upon in the access network, the ESInet, or at the PSAP. The invocation of an assisting service may occur at the time of session establishment and/or once the emergency session has been established.

Assisting services for Total Conversation users can be invoked in different ways in different networks for non-emergency sessions, e.g. by the Application Service Provider. However, for IMS roaming users and IMS emergency registered users the Application Service Provider might not be included in the session routing and would therefore be unable to invoke the assisting service. There is a need for consistent implementation across different networks to ensure successful inclusion of assisting services in those cases.

In the case when a Total Conversation user is in the habit of undertaking explicit actions in order to include a specific assisting relay service in their day to day calls, it is highly likely that the Total Conversation users will repeat those actions when making an emergency call. Depending on implementation, the actions might or might not result in the inclusion of the specified assisting relay service in the emergency session. However, regardless of the result, it is necessary to ensure that an appropriate assisting relay service is still included when needed, using one of the solutions described in clause 7.2.2.

Any emergency sessions that are routed via an assisting service (e.g. in the home network) and not identified as emergency sessions by the network, would be subject to network charges that could be billable to the Total Conversation user. This is contrary to the EU Universal Service Directive [i.24] requirement that all citizens should be able to call emergency services free of charge. Lawful Interception would also need to be supported for such calls (see ETSI TS 133 106 [i.25] and ETSI ES 201 158 [i.26]).

#### IMS:

- In IMS, a user terminal detecting an emergency session when it is not IMS registered in its home network or is IMS registered but roaming outside of its home network, has to initiate IMS emergency registration as per clause 7.2 of ETSI TS 123 167 [i.8]. This means that emergency session establishment is handled differently to non-emergency session establishment, and as a consequence, the usual method used for indicating the need for an assisting services for non-emergency sessions might not be supported.

#### IETF SIP:

- In IETF SIP, Total Conversation users connecting outside of their home network will always be SIP registered in the home network. This is regardless of whether an emergency session is to be established or not. In that case, there could be a problem if both the Application Service Provider in the home network, and the PSAP in the visited country try to invoke the assisting services for the session, based on a signalled indication from the Total Conversation user.

The obstacles to identifying and invoking assisting services are related to how to assess what modality(s) and language(s) the Total Conversation user and the PSAP are able to support, and how to invoke an appropriate assisting service in an interoperable way.

Given the potential compatibility problems outlined above, it is highly desirable to ensure that there are common solutions across all networks for indicating/identifying the need for, and invoking, assisting services when it comes to Total Conversation for emergency communications. Further discussions of solutions and recommendations are described under clause 7.2 of the present document.

Regardless of how the need for assisting services is indicated/identified, or by whom and when the assisting services are included in the session, they will for the most part be invoked as one participant in a multi-party conference between the Total Conversation user, the PSAP, and the assisting service. When applying functions for multi-party sessions, the specific requirements for caller location and identity information provision in the emergency service session case should be respected.

The requirements for the IMS conference service are specified in ETSI TS 122 173 [i.6]. The IETF SIP conference framework is specified in IETF RFC 4353 [i.23].

## 6.5.7 Security and Privacy

Security requirements are described in ETSI TS 101 470 v1.1.1 [i.2]. The interface between the Total Conversation terminal and the relay node is covered by IMS and IETF SIP specifications. The interface between the PSAP and the relay node is covered by the ESInet solution. Security functionality for emergency calling is specified in ETSI TS 123 167 [i.8] for IMS, and described in IETF RFC 6443 [i.14].

Media encryption and key management as per ETSI TS 101 470 [i.2] are described for IETF SIP including the security methods recommended by EENA and IETF. However ETSI TS 101 470 [i.2] also refers to the use of ZRTP [i.44] based key exchange that is not mentioned by EENA and IETF specifications. Therefore at the time of writing, ZRTP implementation is not a requirement for the implementing Total Conversation for emergency communications.

Requirements for privacy and data protection of user information such as location, identity, etc. are subject to local regulations. They are applicable for the different entities that receive these data such as the access network, service provider, assisting service, PSAP, and others.

Authentication and trust of the PSAP towards the used assisting service is essential. If the PSAP invokes a trusted assisting service into its multi-party conference then no further authentication is needed. Otherwise if the relay service is provided or invoked by the Total Conversation user or network service provider, then a mechanism should be defined to authenticate the assisting service. This may need some work to identify how key exchange is achieved and how to protect the functional entities, e.g. PSAP, ESInet from security and Denial of Service attacks.

NOTE: Key exchange and the protection against DoS attacks are not within the scope of the present document.

## 6.5.8 Transfer of session to another Total Conversation user terminal

Clause 6.3.8 of the present document discusses the potential need to transfer specific media from an ongoing session or the entire Total Conversation session, to a more appropriate Total Conversation terminal, e.g. one that has a higher resolution screen. Requirements for the transfer of media between user terminals is specified for IMS in ETSI TS 122 228 [i.10] with procedures specified in ETSI TS 123 237 [i.27]. For IETF, SIP Session Mobility is described in IETF RFC 5631 [i.28]. Requirements for Explicit Communication Transfer are described in ETSI TS 122 173 [i.6] for IMS and for IETF SIP Best Current Practice for Call Control - Transfer is specified in IETF RFC 5589 [i.29].

If an attempted session transfer fails, then the Total Conversation user may be disconnected until such time that a call-back from the PSAP is successfully established. This would cause a delay in service. If the attempted transfer of media to another terminal fails, this might not result in the session being disconnected, but the attempt would cause distraction for the Total Conversation user and the call-taker, and could interrupt the service resulting in a delayed response. In addition, the transference of media between an emergency registered terminal in IMS and a SIP registered terminal might not be straightforward and would need further standardization in order to work. For IMS, it is therefore specified that the terminal 'shall not' invoke Inter UE Transfer (see ETSI TS 123 237 [i.27]) during an emergency call. ETSI TS 122 173 [i.6] further specifies that Explicit Communication Transfer by any entity other than a PSAP is prohibited during an emergency call. For IETF SIP, IETF RFC 6443 [i.14] and IETF RFC 6881 [i.11] specify that the terminal should disable all features that will interrupt an ongoing emergency call. Although the scenario presented in clause 6.3.8 could occur, it should be seen as an exception and something that could potentially be worked around, e.g. by the PSAP operator agreeing with the Total Conversation user to establish an outgoing call to the other terminal, or by inviting the other terminal to a conference in which both the source and the target Total Conversation user terminal will be participants.

NOTE: This would assume that the other terminal was addressable by the PSAP, discussion of which is beyond the scope of the present document.

As a conclusion, given the changes that would be needed to existing standards to support session or media transfer, the exceptional nature of the scenario presented, and the potential for a work around solution, the present document does not further consider transference of an ongoing session between Total Conversation user terminals.

## 6.5.9 Testing

The requirements for automatic functional testing of Total Conversation emergency communications from Total Conversation user terminals in production are described in ETSI TS 101 470 v1.1.1 [i.2]. The URN used in this case should start with "test", for example "urn:service:test.sos" is defined specifically for testing as described in IETF RFC 6881 [i.11]. The PSAP used for test purposes should be able to test the media combinations that are required for Total Conversation, see IETF RFC 6881 [i.11]. Also including a relay service in the Total Conversation emergency session should be tested.

The Test for Total Conversation for emergency communications uses the same ESInet as the regular emergency calls, it is therefore recommended if the PSAP is aware of the present ESInet load, to consider the number of terminals performing the test at any moment in time in order to prevent congestion.

NOTE: It is out of scope of the present document to consider how the PSAP determines the load on the ESInet.

## 6.5.10 Media

As the three media of Total Conversation are primarily used for language content, it is important that all functional elements in the Total Conversation emergency call support the quality levels necessary for adequate language perception of sign language and lip-reading in video, spoken language perception in audio, and written language reading in real-time text.

The available media can also be used to communicate information other than language between the Total Conversation user and PSAP. For example, video can be used for medical assessment, audio for environmental information, and real-time text for rapid exact spelling information. It is however not expected that the functional components in the Total Conversation emergency call need to support any higher quality of the media than that necessary for conversational language support.

ETSI TS 101 470 [i.2] provides the basic quality requirements, as well as the codecs and transports that all functional elements need to support. There are also options within each media coding and transport that should be supported and used to ensure necessary levels of quality and interoperability. The acceptance and use of such options tend to evolve over time. The codecs are commonly specified in high level requirement specifications, while the options selected for implementation are defined in the technical specifications for the emergency services and for the application service providers.

Clause 7 of the present document provides discussion of media options. Clause 8 of the present document contains recommendations for media aspects.

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## 7 Key Issues for implementing Total Conversation for Emergency services

### 7.1 General

The key implementation issues for Total Conversation for Emergency Communications are described in this clause. For each key issue one or several solutions with analysis and then a conclusion is provided. The analysis is the key factor to identify if changes are needed in 3GPP and/or IETF specifications.

In all cases different solutions are evaluated while maintaining the need to meet existing requirements for non-Total Conversation emergency sessions.

### 7.2 Identification and invocation of assisting relay service

#### 7.2.1 General

If the Total Conversation user has a communication need that might warrant the inclusion of an assisting relay service in the emergency session, this can be indicated in different ways (see clause 7.2.3). The indication can be used by functional entities at different locations within the network to trigger the invocation of a multi-party conference that includes the assisting service as one of the participants if needed and as outlined in clause 6.5.6.

The need for a specific assisting relay service could be explicitly indicated in the Total Conversation user's User Profile or in signalling from the Total Conversation user's terminal. Alternatively, it could be inferred by entities in the network based on the Total Conversation user's indicated language / modality preferences and the known capabilities of operators and/or equipment at the receiving PSAP in the case of emergency invocation.

The two main issues that need to be considered with respect to assisting services for emergency communications are therefore:

- 1) Where and by which entity in the networks should the multi-party conference be invoked.
- 2) How to indicate/identify the need for the assisting service.

Other issues include: the handling of emergency calls by assisting relay services; how to obtain details for a specific assisting relay service; and how to cater for the default behaviour of the Total Conversation user when making a call.

#### 7.2.2 Issue #1: Invoking the conference for inclusion of an assisting relay service

##### 7.2.2.1 General

A multi-party conference could be invoked for inclusion of an assisting service by the following entities:

- 1) the Total Conversation user terminal;
- 2) the access network by the Application Service Provider (which could provide the assisting relay service itself);  
or
- 3) the ESInet, e.g. by the Public Service Access Point (PSAP).

NOTE: There are many different ways in which a conference can be invoked and it is not considered practical to describe and analyse all of those in the clauses that follow. However, sufficient examples are given to allow identification of the advantages and disadvantages associated with having the conference invoked by each of the three entities listed above.

## 7.2.2.2 Solution #1: Conference invocation by the Total Conversation terminal

### 7.2.2.2.1 Description

This describes the case where the conference is hosted by a multi-party conference bridge external to the Total Conversation terminal.

A three way conference session is invoked and controlled by the Total Conversation terminal, e.g. by using the following steps:

- 1) the Total Conversation user terminal requests creation of a conference focus (host) using SIP ad hoc methods as per IETF RFC 4579 [i.30];
- 2) the Total Conversation user terminal then sends a REFER request as per IETF RFC 3515 [i.31] to the assisting service referring it to a conference URI;
- 3) the Total Conversation user terminal then uses 3<sup>rd</sup> Party Call Control (3PCC) as per IETF RFC 3725 [i.32] to connect the media from the PSAP to the conference focus.

There are other ways in which PSAP may be connected to the multi-party conference bridge, e.g. by sending a REFER request. One advantage of using 3PCC however, is that the INVITE message from the Total Conversation user terminal appears no different than any other INVITE that might be sent by the Total Conversation user for a Total Conversation emergency session, i.e. there is no special indication that the session will involve a conference with a 3<sup>rd</sup> party relay service.

This method assumes that the address of the assisting service to be invoked is configured in the Total Conversation terminal. The address may be obtained by the Total Conversation user subscribing to the assisting service, or configured by the application service provider, e.g. via service provisioning or an application downloaded to the terminal.

### 7.2.2.2.2 Evaluation

#### Advantages:

- There is no need for the Total Conversation terminal to include an explicit indication in signalling of the Total Conversation user's communication needs. Hence, there should be no impact to existing standards on the user to network signalling.
- The PSAP will receive the necessary location and caller identity information in the INVITE directly from the Total Conversation user terminal, as within the existing solutions for emergency services.
- The Total Conversation user usually has the best knowledge about which assisting relay service is needed in a session.
- It is possible to include an assisting relay service for the purposes of translation between a Total Conversation user requiring sign-language or text support and a voice only PSAP located in the CS domain.

#### Disadvantages:

- Not all terminals making emergency calls will be conference aware or support SIP conferencing call control conventions as per IETF RFC 4579 [i.30].
- Not all terminals making emergency calls support the SIP REFER mechanism.
- IMS in ETSI TS 122 173 [i.6] currently prohibits that any entity other than the PSAP can invoke the CONF service for an emergency session. For IETF SIP IETF RFC 6443 [i.14] recommends and IETF RFC 6881 [i.11] mandates that three-way-calling be disabled in terminals.
- The Total Conversation user terminal would need to have specific application functionality to make the invocation of assisting relay services convenient for the user.

- An IMS emergency registered Total Conversation user will not be able to establish a conference between the PSAP and the assisting service, because the registered contact field can only be used for emergency sessions, and because media associated with the emergency sessions have to be handled independently from media associated with any other sessions as per ETSI TS 124 229 [i.22].
- The PSAP would not be aware of the of the contact information for the assisting service for the purposes of including that service again in the event of a call-back to the Total Conversation user. This means that the Total Conversation user terminal would have to establish the multi-party conference again at the time of the call-back.
- The PSAP will not be aware of the multi-party conference so the conference party details will not be included in session logging and recording.
- If this is not the only implemented method, there may be a risk for invocation of an assisting service even when there is no need for it, e.g. when the PSAP has capability for the communication modality of the Total Conversation user.

### Conclusion:

This solution is not recommended for IMS, because of the modifications that would be needed to the 3GPP standard in order to allow the Total Conversation terminal to invoke a conference for the emergency session.

This solution could be acceptable for IETF SIP if a variant were selected that made the connection with the assisting service simultaneously with the connection to the PSAP, because this would seem unlikely to interrupt the session (IETF specifications IETF RFC 6443 [i.14] and IETF RFC 6881 [i.11] only discourage such operations in the event that they will interrupt an ongoing emergency session). In this case all Total Conversation user terminals would need to implement IETF RFC 4579 [i.30] and SIP REFER mechanisms accordingly.

## 7.2.2.3 Solution #2: Conference invocation initiated in the Access Network

### 7.2.2.3.1 Description

This describes the case where the conference is hosted by a multi-party conference bridge in the Application Service Provider/access network.

A three way conference session is invoked and controlled by the Application Service Provider, e.g. as follows:

- 1) The Total Conversation user initiates a Total Conversation emergency session by sending an INVITE indicating an emergency session.
- 2) The need for an assisting relay service to be included in the session is identified by the Application Service Provider e.g. through the User Profile indicating need of the assisting service for all calls to voice-only terminals, i.e. those that do not support video and text, and for all calls with emergency services.
- 3) An Application Server invokes a multi-party conference by dialling into a dedicated conference URI as per IETF RFC 4579 [i.30].
- 4) The Application Server refers the Total Conversation user and the needed assisting relay service to the conference as per IETF RFC 3515 [i.31].
- 5) The Application Server initiates an emergency session on behalf of the Total Conversation user, passing the caller identity and location information transparently to the PSAP, and using 3<sup>rd</sup> Party Call Control to connect the media from the PSAP to the conference, as per IETF RFC 3725 [i.32].

This method assumes that the Application Service Provider knows the address of the assisting relay service to be invoked and that it is configured in the Total Conversation user profile. This may be because it is the Application Service Provider that offers the assisting relay service to the Total Conversation user as part of their subscription.

Other mechanisms than those above are available to the Application Service Provider for connecting the Total Conversation user, assisting service, and PSAP to the conference.

### 7.2.2.3.2 Evaluation

#### Advantages:

- If the address of the assisting service were stored as part of the User Profile, there would be no need for the Total Conversation user terminal to include any indication in signalling of the need for an assisting relay service to be invoked by 3<sup>rd</sup> party in either the Access Network or the ESI<sup>n</sup>et. Hence there would be no impact on existing terminal to network signalling.
- The PSAP can receive the necessary location and caller identity information in the INVITE from the Total Conversation user terminal if relayed transparently by the Application Server initiating the emergency session.
- The Application Service may be set up to invoke the same suitable assisting relay service as is done in everyday calls with the Total Conversation user.
- It is possible to include an assisting relay service for the purposes of translation between Total Conversation user requiring sign-language support and a voice only PSAP located in the CS domain.

#### Disadvantages:

- IMS in ETSI TS 122 173 [i.6] currently prohibits that any entity other than the PSAP can invoke the CONF service for an emergency session. For IETF SIP, IETF RFC 6443 [i.14] recommends and IETF RFC 6881 [i.11] mandates that three-way-calling be disabled in the network during emergency sessions.
- Total Conversation users' sessions from IMS Emergency registered subscribers whether roaming or in their home networks are not routed via the home Application Service Provider. As such, the User Profile is not accessed, in which case no conference can be invoked (see ETSI TS 123 167 [i.8]).
- If the need for a specific assisting relay service were not stored as part of the user profile, there would be a need for the Total Conversation user terminal to include an explicit indication for the specific relay service in signalling.
- The PSAP will not be aware of the contact information for the assisting service for the purposes of including that service again in the event of a call-back to the Total Conversation user. However, the service could potentially be invoked automatically by the Application Service Provider on behalf of the Total Conversation user as part of the terminating call procedures.
- The PSAP will not be aware of the multi-party conference so the details might not be included in session logging and will only be included in the session recording if the PSAP supports all media of the user and the assisting service together.

NOTE: If Additional Data is implemented (see Additional Data Related to an Emergency Call [i.38]) then information about the assisting service will be available to the PSAP.

- Not all Total Conversation user terminals support the SIP REFER mechanism.

#### Conclusion:

This solution is not recommended for IMS, because of the changes that would be needed to the 3GPP standard in order to allow the Application Service provider to invoke a conference for the emergency session.

This solution could be acceptable for IETF SIP if a variant were selected that made the connection with the assisting service simultaneously with the connection to the PSAP, because this would seem unlikely to interrupt the session (IETF RFC 6443 [i.14] and IETF RFC 6881 [i.11] only discourage such operations in the event that they will interrupt an ongoing emergency session).

## 7.2.2.4 Solution #3: Conference invocation in the ESInet

### 7.2.2.4.1 Description

This describes the case where the conference is hosted by the PSAP that controls the multi-party conference bridge.

A three way conference session can be invoked and controlled by the PSAP, e.g. as follows:

- 1) The Total Conversation user terminal sends an INVITE to the PSAP requesting an emergency session, and the PSAP answers the call using a bridge capable of supporting a multi-party conference.
- 2) The need for an assisting service is identified by The PSAP. e.g. by checking the Total Conversation user's indicated language / modality preferences and recognizing that those are not supported by the PSAP, or through reception of a REFER request with the Refer To: header containing the SIP URI of a specific relay service.
- 3) The PSAP sends an INVITE to the assisting relay service to add it to the conference.

In the case where the PSAP does not answer the call using a bridge capable of supporting a multi-party conference, then it needs to invoke one (e.g. as per IETF RFC 4579 [i.30]), and to direct the Total Conversation user's media towards that bridge (e.g. using 3PCC as per IETF RFC 3725 [i.32]).

### 7.2.2.4.2 Evaluation

#### Advantages:

- The PSAP will in any case have to support conference functionality in order to be able to conference in and transfer to other agencies and to first responder assisting services for cases where the need is discovered by the call-taker during the emergency session.
- The PSAP will have contact information for the assisting relay service so that it can be included again in the event of a call-back to the Total Conversation user.
- The PSAP will be fully aware of and in control of the emergency conference. This would facilitate re-establishment of the conference in the case of call-back.
- The Total Conversation user terminal need not be conference aware, nor support SIP REFER mechanism.

#### Disadvantages:

- This will not work during the migration phase in the event of a Total Conversation session being routed to a PSAP in the CS domain, e.g. PSTN, where an assisting service e.g. could be used to provide sign-language translation via video to Total Conversation user and audio only to the PSAP. In this case media gateways are used and fall-back to text service may be used if available.

NOTE 1: Since the assumption in the present document is that the PSAP and relay service are connected to IP based network ESInet, then this disadvantage should not appear in full deployments of Total Conversation for emergency communications by member states.

#### Conclusion:

This appears to be a good solution for IMS and IETF SIP, because it places no special requirement on the Total Conversation user terminal, and would have limited impact on existing standards.

NOTE 2: During migration to NG112, if the need for an assisting service could be identified at the gateway between the access network and the CS based PSAP network, then it may be possible to invoke the conference at that point. The details of this solution are out of scope for the present document.



## 7.2.3 Issue #2: Indicating/identifying the communication needs of the Total Conversation user

### 7.2.3.1 General

It would be possible in some circumstances for the relay service to be invoked in the same way for an emergency session as it is for a non-emergency session. This could have benefits for the Total Conversation user, e.g. being able to use a relay service provider that has been tried and is trusted due to its everyday use by the Total Conversation user. However, guidance received from some stakeholders has indicated a preference for a single common solution controlled by the PSAP which would work for all Total Conversation users regardless of their registration or roaming status. Also the analysis undertaken in clause 7.2.2 indicates that this would appear less controversial from the point of view of the impact to existing standards. The present clause therefore focusses on the way in which the communication needs of the Total Conversation user, or a specific relay service to be used, can be indicated to the PSAP. A description of a solution that allows relay services to be added by different parties in the emergency session depending on the registration and roaming status of the Total Conversation user is provided in Annex A.

NOTE: If the PSAP answers every call directly using a conference bridge, then the steps below involving PSAP invocation of the conference are not necessary.

### 7.2.3.2 Solution #1: Preferred Language / Modality indicated by Total Conversation terminal in Accept-Contact header

#### 7.2.3.2.1 Description

The indication of the Total Conversation user's communication needs are signalled by the Total Conversation user terminal using the user preference mechanism extensions described in IETF RFC 3840 [i.47] and IETF RFC 3841 [i.13]. They use the language / modality tags defined in the IANA language sub-tag registry described in IETF RFC 5646 [i.33], and the media tags of IETF RFC 3840 [i.47]. Examples of use can be found in IETF RFC 4596 [i.12].

The method builds upon the premise that called parties have registered their language/modality/media capabilities by listing their associated tags when registering to the SIP service. The caller expresses their needs and preferences with similar tags in the Accept-Contact field. The addressed service provider matches the preferred capabilities against those indicated as available, and routes the call to the best match. The resulting capabilities of the PSAP in the session are expressed in the Contact field of the response so that a negotiation is completed.

IETF RFC 6443 [i.14], clause 9.2 and EENA NG112 LTD [i.1], clause 4.3.1.5 briefly describe that it is possible to route emergency calls in the ESInet based on headers in the INVITE message:

- 1) The Total Conversation user terminal is configured by the Total Conversation user, or by the Application Service provider to include an indication of the needed language / modality in an Accept-contact header in the INVITE message.
- 2) When the INVITE arrives at the ESInet the Accept-contact header is cross checked with the registered capabilities of the PSAPs.
- 3) If one or more of the PSAPs is able to support the needed language / modality combination, then the call is routed to the most appropriate of those PSAP for handling the call.
- 4) If the indicated language / modality requirement of the Total Conversation user cannot be accommodated by any of the PSAP operators, then a conference is invoked by the most appropriate PSAP as per clause 7.2.2.3 and the assisting service is added to the session.
- 5) The supported modalities and languages in the session are presented in the response by SIP media and language tags according to IETF RFC 3840 [i.47].

### 7.2.3.2.2 Evaluation

#### Advantages:

- The method for inclusion of a user's language / modality preferences in SIP messages is already specified in IETF RFC 4596 [i.12] for the languages / modalities covered in IETF RFC 5646 [i.33], as is the ability to route locally based on those preferences. No further standardization work would therefore be needed in IETF to cover a requirement for a specific spoken/written or signed language. However, there is no commonly agreed way to differentiate between spoken and written language.
- The ability to negotiate a user's desired language and modality, and the ability to pass this information between end points are requirements of IMS. Only limited changes to the SIP 3GPP profile might be needed to guarantee this functionality for IMS.

#### Disadvantages:

- This method is limited to indicating the Total Conversation user's need for a specific language / modality combination, i.e. spoken/written, or signed. It cannot currently be used to indicate e.g. the communication needs for someone requiring translation due to a severe speech impediment who might usually use a speech-to-speech relay service.
- This method would not be suitable to indicate a need for rapid captioning of speech.
- This method would, with the currently specified language and media tags, not be able to clearly discriminate between spoken and written language. Further, it would not allow for the preferred directionality of communication to be indicated. E.g. a user wishing to use speech for outward communication but preferring text for incoming communication.

NOTE: The IETF SLIM work [i.40] ongoing at the time of writing is focussed on inclusion of enhanced language/modality tags only in the SDP part of relevant SIP messages and not in SIP headers. It would therefore not contribute to this solution, based on SIP headers.

- This method would not work if the Total Conversation user were to attempt an emergency call using a Total Conversation terminal other than their own which had therefore not been configured with the Total Conversation user's communication preference settings.

#### Conclusion:

Routing based on SIP headers is already specified and supported in some implementations, however, it is not recommended, because there would be additional value from indicating language / modality in SDP. The IETF SLIM group working on solutions for directional and media level indication in draft-ietf-slim-negotiating-human-language [i.40] is focussed on inclusion of the tags in the SDP part of relevant SIP messages, and not on their inclusion in SIP headers (see also clause 7.2.3.3).

### 7.2.3.3 Solution #2: Preferred Language / Modality indicated by Total Conversation terminal in SDP language attribute

#### 7.2.3.3.1 Description

The indication of the Total Conversation user's communication needs are signalled by the Total Conversation terminal using the language attribute of the SDP (IETF RFC 4566 [i.39]), and the languages / modalities defined in the IANA language sub-tag registry described in IETF RFC 5646 [i.33].

IETF SLIM is currently working to specify a human interactive language tag to indicate which language should be used for each direction of an interactive media stream. See draft-ietf-slim-negotiating-human-language [i.40]:

- 1) The Total Conversation user terminal is configured by the Total Conversation user, or by the Application Service provider to include an indication of the needed language / modality in a language attribute in the SDP offer included with the INVITE message.
- 2) When the SDP offer arrives at the ESInet the language attribute is cross checked with the registered capabilities of the PSAPs. This function is specified in IETF RFC 6443 [i.14] clause 9.2.
- 3) If at least one of the PSAPs is able to support the needed language / modality combination, then the call is routed to the PSAP most appropriate for handling the call.

NOTE: Use of an SDP language attribute could also allow for negotiation of the language to be used via the SDP offer-answer mechanism. See IETF RFC 3264 [i.41] updated by IETF RFC 6157 [i.42].

- 4) If the indicated language / modality requirement of the Total Conversation user cannot be accommodated by any of the PSAP operators, then a conference is invoked by the PSAP as per clause 7.2.2.3 and the assisting service is added to the session.

### 7.2.3.3.2 Evaluation

#### Advantages:

- The method for inclusion of a user's language / modality preferences in the SDP body of a SIP message is already specified in IETF RFC 4566 [i.39] for the languages / modalities covered in IETF RFC 5646 [i.33]. IETF RFC 6443 [i.14] states that Emergency Service Routing Proxies (ESRP) are expected to be able to provide routing based on media described in SDP and this could equally apply to language tags in SDP.
- Whilst some further standardization work is needed in IETF to allow the preferred language / modality to be indicated in the SDP part of the INVITE for each media direction rather than for the entire media session, this work is already underway in IETF SLIM. Therefore no further standardization work than that would be needed in IETF.
- The ability to negotiate a user's desired language and modality, and the ability to pass this information between end points are existing requirements of IMS, as is the inclusion of SDP in SIP messages. Therefore, potentially no changes to the SIP 3GPP profile would be needed to add this functionality to IMS.

#### Disadvantages:

- This method is limited to indicating the Total Conversation user's need for a specific language / modality combination, i.e. spoken, written or signed. It cannot currently be used to indicate e.g. the communication needs for someone requiring translation due to a severe speech impediment who might usually use a speech-to-speech relay service.
- This method would not be suitable to indicate a need for rapid captioning of speech.
- This method would not allow for the preferred directionality of communication to be indicated. E.g. a user wishing to use speech for outward communication but preferring text for incoming communication.

NOTE 1: The IETF SLIM work [i.40] ongoing at the time of writing, covers the most common cases with respect to directionality, and the aim is to continue the standardization to cover more complicated use cases.

- There is currently no specification in EENA NG112 LTD [i.1] to route sessions based on the contents of SDP in NG112 solutions, because SDP attributes are described using text and not in machine parsable XML.

NOTE 2: The expectation among IETF experts is that future NENA and EENA specifications and deployments will support routing based on SDP.

- This method would not work if the Total Conversation user were to attempt an emergency call using a Total Conversation terminal other than their own which had therefore not been configured for the communications needs of the calling user.

#### Conclusion:

This method of indication is recommended for the case when the Total Conversation user has language / modality needs that can be indicated using this mechanism, and on the assumption that future NG112 implementations will support routing based on SDP.

### 7.2.3.4 Solution #3: Preferred Language / Modality tag added by Application Service Provider

#### 7.2.3.4.1 Description

This method is similar to that for solutions #1 and #2 except that the Total Conversation user's communication needs are added by the Application Service provider to the signalling.

NOTE: This method will only work if the Total Conversation user has asked their Application Service Provider to set the preferences in a personal profile.

- 1) The Total Conversation user sends an INVITE to establish the emergency session.
- 2) The communication needs of the Total Conversation user are stored in the User Profile with the Application Service provider. Service logic in the network at the routing proxy, triggers an Application Server which adds the needed language / modality indication to the Accept-Contact header or to the SDP body of the INVITE and forwards the message on towards the PSAP.
- 3) The session signalling continues as per steps 2-4 of clauses 7.2.3.2 and 7.2.3.3.

#### 7.2.3.4.2 Evaluation

##### Advantages:

- The advantages for solution #1 in clause 7.2.3.2.2 or solution #2 in clause 7.2.3.2.2 apply respectively depending on whether the language / modality indication is added to the Accept-Contact header or to the SDP.
- The TC user can easily switch Total Conversation user terminals without losing the language/modality settings providing the same subscription is used.

##### Disadvantages:

- Total Conversation user's sessions from IMS Emergency registered or roaming subscribers (see ETSI TS 123 167 [i.8]) are not routed via their home Application Service provider, so it cannot be guaranteed that all emergency session request messages will have the language / modality preferences added.
- This method would in all practical cases not work if the Total Conversation were to attempt an emergency call in IMS using a Total Conversation terminal other than their own.

NOTE: This could potentially work if the UICC hosting the subscription for the Total Conversation user were transferred into that terminal, but that would seem very unlikely to happen in an emergency situation.

- Depending on how the language / modality indication is added, the disadvantages of solution #1 in clause 7.2.3.2.2 or solution #2 in clause 7.2.3.3.2 also apply respectively.

##### Conclusion:

This method of indication is not recommended due to the fact that it cannot be guaranteed to include the language / modality information for all emergency sessions.

#### 7.2.3.5 Solution #4: Use of Additional Data to indicate communication needs and / or a specific relay service

Chapter 9 of the EENA Long Term Development specification [i.1] talks speculatively about the potential use of "additional data" by the user terminal to send information about a specific relay service to the PSAP for inclusion in the ensuing emergency conference. Since [i.1] was finalized, work has progressed in relation to the "additional data" in the IETF ECRIT group. The latest draft version of the specification is approved and can be found in IETF draft-ietf-ecrit-additional-data [i.38].

Looking at the IETF Additional Data draft [i.38] however, it does not appear that any of the defined data structures readily lend themselves to inclusion of information relating to a relay service (unless included by a relay service itself if present in the routing of the session).

Firstly, the draft specification does not include any structures for data associated with a caller, only for data associated with a call. Caller related data perhaps related to communication needs and/or a specific relay service may be the subject of future specification, but that will not happen within the timeframe of the elaboration of the present document.

Secondly, there does not appear to be an appropriate parameter in the vCard structure describing subscriber information other than "lang" for including information about communication needs of the user in subscriber information for the call. Whilst this could be used to provide an indication of the user's preferred language for the session and could potentially be used at the PSAP for invoking an appropriate relay service, as confidential user data it is subject to more stringent privacy requirements and so cannot be used for routing or negotiation during session establishment.

Finally, there appears to be no means to include a URI associated with a relay service, or indeed any indication that such a URI would be associated with a relay service that needed to be included in the call.

However, given the solutions described in the present document, it appears likely that the majority of cases where a relay service needs to be included could be handled without having to use Additional Data for identification of the service. As such its use for providing information about a communication need and/or relay service to be included, could be considered purely as a fall-back mechanism of last resort. In that case, it would be possible for the Total Conversation user terminal to include in the "Comment" block of the Additional Data [i.38], an indication of the needed relay service and its URI to be used in the event that the PSAP cannot identify one based on the language / modality indication provided elsewhere in the signalling.

The "Comment" block is not intended to be machine readable so could not be used to trigger automatic inclusion of the relay service by a PSAP local implementation. However, it would be available to the PSAP operator at the time the session was established so could be used to manually add the necessary relay service before the call-taker starts communication with the Total Conversation user. The "Comment" block could also potentially be used to indicate communication needs that are not covered by existing language / modality tags, as well as the URI of the relay service.

## 7.2.4 Issue #3: Emergency session handling at the assisting relay service

### 7.2.4.1 General

It is likely that the assisting relay services used to enhance communication for Total Conversation for emergency sessions will in many cases be the same ones as those used for non-emergency sessions. In that case, there are a number of reasons why it is necessary for the relay service to be able to differentiate between emergency sessions and non-emergency sessions.

One reason is that it is desirable to treat incoming emergency sessions to the assisting relay service with higher priority than non-emergency sessions. Another reason is that the interpreters and call assistants working in the relay service need to know and accept that they will be handling emergency calls, because of the extra stress that can cause. It is desirable, and in some jurisdictions it may also be required, that interpreters have been specially trained to handle emergency calls in the relay service.

The assisting relay service could be invoked for an emergency session by different entities in the network. In the case where the assisting service is able to be invoked by the Total Conversation user terminal or by the Application Service Provider including the URI of the needed assisting service in the signalling of the emergency session, then the assisting relay service could act on the "urn:service:sos" indication included in the INVITE and initiate an emergency session towards the PSAP. However, in the case where the assisting relay service is being invoked by the PSAP itself, or by a relay node setting up the multi-party call, the INVITE will appear as "normal" (non-emergency) session establishment, and no such indication will by default be available. It is therefore necessary to include a specific indication.

The following clauses consider different solutions for how to indicate the emergency nature of the incoming session to the assisting relay service.

### 7.2.4.2 Solution #1: Use of "SIP PSAP callback indicator"

#### 7.2.4.2.1 Description

In this case, the PSAP would include the "SIP PSAP callback indicator" as specified in IETF RFC 7090 [i.43]. This indication was specified in order to be used in the event of the PSAP having to call back to the SIP registered user, so that the session would be recognized in a user's terminal as relating to a previously established emergency session.

#### 7.2.4.2.2 Evaluation

Whilst the "SIP PSAP callback indicator" could potentially be used for the purposes of indicating an incoming emergency session from a PSAP to the assisting relay service, such use is not in line with the original intention of the indicator and would not necessarily result in priority handling of the INVITE by the network. Also the SIP PSAP callback indicator is not considered "trusted" according to the IMS standard (e.g. it is to be ignored by IMS registered User Equipment). As such, it might be completely ignored or even removed by intermediate network entities. (See clause 6.3.10).

#### **Conclusion:**

This solution is not recommended because it could not provide a reliable indication to the assisting relay service of the nature of the emergency nature of the incoming session.

### 7.2.4.3 Solution #2: Use of assisting relay service emergency specific URI

#### 7.2.4.3.1 Description

In this case, an emergency call specific URI is used by the invoking entity to address the assisting relay service in order to enable priority handling and other emergency service specific actions.

#### 7.2.4.3.2 Evaluation

An emergency specific URI used to address the assisting relay service should not be subject to any lack of trust by the system or network entities, so would appear reliably at the assisting relay service allowing priority treatment of the incoming emergency session and preparedness of the personnel involved.

The use of such a URI is currently not specified for assisting relay services, though it is assumed to exist according to the specifications in EENA NG112 LTD [i.1] and in ETSI TS 101 470 [i.2]. Methods for making the URIs of suitable relay services known to the functional entities that will invoke them need to be established. Such methods are discussed in clause 7.2.5.

#### **Conclusion:**

This solution is not recommended. It can provide a reliable indication of the emergency nature of an incoming session. But separate entry addresses need to be agreed and communicated to the parties doing the invocation.

### 7.2.4.4 Solution #3: Trusted relay services recognizing PSAP specific originating URIs

#### 7.2.4.4.1 Description

A database of PSAP URIs is made available to dedicated and therefore trusted relay services. If the URI of the originating party of an incoming session to the relay service is recognized as the URIs of a PSAP, then it will be identified as an emergency session and handled appropriately by the assisting relay service.

#### 7.2.4.4.2 Evaluation

Use of a dedicated SIP Contact URI by the PSAP for outgoing emergency sessions should provide a secure and reliable way to indicate the emergency nature of the incoming session to a relay service, thus enabling the session to be handled accordingly.

However, the identification of PSAP URI would need to be collected in a database that would also have to be kept up-to-date. Ownership of this data-base would need to be identified and agreed.

#### **Conclusion:**

This is a possible solution, however it would need political consensus and a decision on who would be responsible for creating and maintaining the database. It is not recommended as a short term solution, but could be considered in the longer term.

### 7.2.4.5 Solution #4: include a SIP field in the call to use for assisting relay service invocation

#### 7.2.4.5.1 Description

The party invoking the assisting relay service includes a SIP header or parameter in the session establishment signalling indicating that the session is related to an emergency. A suitable mechanism for such an indication would be the addition of a Call-Info field with a purpose parameter indicating relay service request for an emergency call.

The Call-Info field is specified in IETF RFC 3261 [i.16]. It contains a URI and a purpose parameter. The URI would be the URI used to call the relay service from a multi-party bridge to invoke it to the relay service call. The "purpose" parameter would be registered with IANA to indicate emergency related assisting relay service call, e.g. "emergency--relay".

#### 7.2.4.5.2 Evaluation

This proposed method has promising properties. It would however need IETF standardization and IANA parameter registration. Such operations have a tendency to take a long time.

**Conclusion:**

This is a possible solution, but not one that can be recommended in the short term due to the additional standardization work needed. It has benefits that may be explored for other reasons but is not recommended for discriminating emergency calls from normal call.

#### 7.2.4.6 Solution #5: include a SIP priority tag in the call to the assisting relay service to indicate its emergency status

##### 7.2.4.6.1 Description

As an indication to the assisting service that the call is related to an emergency service call, a priority indication with value "emergency" is included in the INVITE to the relay service. The assisting service can recognize this indication and handle the session appropriately.

There are currently two standardized ways to do this. One is to include the SIP header field "Priority" with value "emergency" in the Invite. This field is specified in IETF RFC 3261 [i.16]. Another is to use the feature tag "Priority" with the value "emergency" in the Contact and Accept-Contact fields as specified in IETF RFC 3840 [i.47] and IETF RFC 3841 [i.13]. The assisting service could also make its SIP registration for the emergency queue with the same value in order to support automatic internal routing of the calls. The following discussion is valid for the approach with the priority feature tag.

##### 7.2.4.6.2 Evaluation

The indication of the feature tag "Priority: emergency" offers an already standardized way to indicate to the assisting service that the call is about an emergency. This is a straightforward and apparent way that can be easily implemented. There are also standardized methods to make automatic routing based on existence of this feature tag that may be used by the assisting service.

When this tag is used, the URI used to call an assisting service can be the same for everyday calls as for emergency calls.

There are risks for fraud and Denial of Service attacks, but no more so than for solutions #1, #2 and #4.

NOTE: The mitigation of these risks is outside the scope of the present report.

**Conclusion:**

This solution is recommended as it has many positive aspects. In addition, it should be included in the interface specification that is in any case needed between the invoking entity and the assisting relay service.

## 7.2.5 Issue #4: Obtaining the identity of a specific assisting service

### 7.2.5.1 General

Clause 6.5.3.3 outlines the need for identification of the necessary relay services by the Total Conversation user, the Application Service Provider and the PSAP.

If the Total Conversation user is a regular user of a relay service, then it is likely that they will have preferred and trusted provider of that service whose identity is known to them. In that case, the identity of the preferred relay service could be used directly when making a call, e.g. by the Total Conversation user terminal being configured to address all outgoing sessions to the relay service, and the relay service making the call to the third party on their behalf. A known relay service identity included in the Total Conversation user's User Profile could also be used by an Application Service Provider to divert a call towards the preferred relay service for outgoing and incoming calls.

In the case where it is the PSAP responsible for invoking a specific relay service, a solution needs to be found for how to identify the correct relay service. The following clauses assess potential solutions.

#### 7.2.5.2 Solution #1: PSAP interrogation of a database

##### 7.2.5.2.1 Description

In this case, the PSAP has access to a database of trusted relay services providing translation for a range of languages and modality.

When the PSAP receives an incoming call containing an indication of Total Conversation user communication needs that it is not able to handle, e.g. sign-language, it will interrogate the database using the indicated communication needs of the caller, and once it has identified an appropriate relay service to handle the translation, it will include it in the session. The relay service can be included in the session as described in clause 7.2.2.4. The communication needs of the Total Conversation user can be indicated as described in clause 7.2.3.3.

#### 7.2.5.2.2 Evaluation

##### Advantages:

- The relay service invoked by the PSAP would be a trusted service capable of handling emergency calls.
- The process of identifying the needed relay service based on the Total Conversation user's indicated communication needs could be automated, thereby expediting the establishment of the emergency session including the relay service.
- The database could be used by PSAPs to cater for users of Total Conversation services having communication modality needs identifiable by modality tags as in clause 7.2.3, but also to identify language translation services for roaming users not able to speak the local language.
- The database would be scalable and could evolve over time to cater for an increasing number of identifiable communication needs and languages.
- The database could be used in combination with other mechanisms as a cross check to see if a relay service explicitly identified by a user with communication needs not identifiable using existing language / modality tags.

##### Disadvantages:

- The identities of trusted Relay Services would need to be collected in a database that would also have to be kept up-to-date. Ownership of this data-base would need to be identified and agreed.

##### Conclusion:

This solution is recommended as a longer term solutions to provide trusted assisting relay services catering for users with communication needs that can be indicated using language / modality tags as described in clause 7.2.3.

#### 7.2.5.3 Solution #2: Identity communicated by the Total Conversation user terminal

##### 7.2.5.3.1 Description

In this case, the contact information for the preferred relay service is communicated directly to the PSAP the Total Conversation user as follows:

- 1) the Total Conversation user terminal establishes a session with the PSAP by sending an INVITE as per IETF RFC 6881 [i.11];
- 2) during the session establishment dialog, the Total Conversation user terminal sends a REFER to the PSAP including the SIP URI of the preferred relay service in the Refer To header using a method=INVITE URI parameter as described in IETF RFC 3515 [i.31];
- 3) the call-taker at the PSAP receives an indication of the referred relay service name and URI and approves its addition to the session;

NOTE: This could be satisfied through accessing configured policy, or by interactive query.

- 4) a NOTIFY is sent to the Total Conversation user terminal that the request has been approved; and
- 5) the UA at the PSAP sends an INVITE to the relay service addressed by the referred to URI and includes the relay service in a multiparty conference with the Total Conversation user.



This may be used in combination with a database of trusted relay services described in clause 7.2.5.2.1. When the PSAP receives an incoming call containing a referral to a preferred relay service provider, it can interrogate the database using the referred to SIP URI. Once it has authenticated the relay service as trusted, it will include it in the session.

### 7.2.5.3.2 Evaluation

#### Advantages:

- The necessary relay service will be unequivocally identified, and will be that known and trusted by the Total Conversation user.
- As the solution does not rely on identification of the relay service based on the communicated Total Conversation user's communication needs, it will work for communication needs that cannot be identified using existing language / modality tags, e.g. speech to speech relay for persons with a speech impediment, or interpretation for those with cognitive impairment.

#### Disadvantages:

- The Total Conversation user terminal will have to support the REFER method.
- The terminal receiving the REFER should seek approval from the call-taker before proceeding with the INVITE to the referred to party. If this needs manual intervention, i.e. it is obtained by interactive query, it will increase the time taken to add the relay service to the session.
- There might be no trust relationship between the relay service identified in the Refer-To header and the PSAP unless, additional some additional authentication takes place. As such the referral might not be approved for all relay services.

NOTE: Authentication could be achieved if the identity of the referred to relay service were cross checked with the database of trusted relay service providers described in clause 7.2.5.2.

#### Conclusion:

This method would be a good one for general identification of the Total Conversation users' preferred relay service if mechanisms were available to authenticate relay services, e.g. a database of trusted relay services. As such it is recommended.

## 7.2.6 Issue #5: Total Conversation user default behaviour

If a Total Conversation user is in the habit of undertaking explicit actions in order to include a specific assisting relay service in their day to day calls, it is highly likely that the Total Conversation user will repeat those actions when making an emergency call. It is necessary to ensure that an appropriate assisting relay service is included when needed, regardless of results of the Total Conversation user's actions at the time of session initiation.

In the case where the user is roaming in IMS and presses '112', the Total Conversation user terminal will become emergency registered (this may also happen in the case that the Total Conversation user is in their IMS home network). When the Total Conversation user terminal is IMS emergency registered, it is unlikely that the Total Conversation user's explicit actions to include a relay service will have any impact on the call handling, because IMS emergency registered terminals are only able to connected to the emergency set of numbers. As such, assuming the Total Conversation user terminal has been configured to identify the user's communication needs, e.g. as described in clause 7.2.3, the PSAP will include an appropriate assisting relay service as described in clause 7.2.2.

In the case where the Total Conversation user is connecting via an IETF SIP network or is in their home IMS network and not required to be IMS emergency registered to establish an IMS emergency session, the emergency session will be handled by their usual Application Service Provider and the Total Conversation user's explicit actions could result in the inclusion of their usual relay service for the emergency session. However, a Total Conversation user terminal configured according to clause 7.2.3 would also still identify the Total Conversation user's communication needs, in which case the PSAP would also act to include an appropriate relay service. If the session is identifiable as an emergency session, the home Application Service provider may, unless prohibited by local regulation, include the assisting relay service identified by the Total Conversation user in the same way as for non-emergency sessions. The Application Service Provider may take the Total Conversation user's User Profile into consideration, e.g. to identify whether the Total Conversation user's communication need will be adequately identified to the PSAP for the most appropriate assisting relay service to be invoked.

In the case where the relay service is not to be included by the PSAP the following actions should be taken by the Application Service Provider:

- The indicated communication needs should be changed appropriately (e.g. to no longer specify a need for sign-language support). This will ensure that the PSAP does not attempt to invoke assisting relay service in addition to that already present in the session.
- The presence of the assisting relay service should be indicated in Additional Data if used (see [i.38]). This will allow the PSAP to still be aware that the caller has special communication needs which could be relevant in the choice of responder to dispatch.
- The connection of the PSAP should proceed regardless of whether the assisting relay service has been connected. I.e. connection of the PSAP and the assisting relay service should take place in parallel.

In the case where the relay service is to be included by the PSAP based on the indicated language/modality communication need:

- The Application Service Provider should override any parameters that would result in the assisting relay service being included in the session in a way that is not controlled by the PSAP.

NOTE: How this would be implemented by the Application Service Provider is outside the scope of the present document.

- The Application service provider should interrogate the answer from the PSAP for availability of the proper media, modality and language support for a functioning call, and include an appropriate relay service if needed.

The Total Conversation user terminal may do a similar comparison between desired and available support in the call, and include an appropriate relay service to cover gaps.

## 7.3 Automatic translation

### 7.3.1 General

The following clauses contain description of the automatic language translation that could be performed by "off the shelf" applications. These applications are available in the market and are widely used in today's communications, examples are described in [https://en.wikipedia.org/wiki/Comparison\\_of\\_machine\\_translation](https://en.wikipedia.org/wiki/Comparison_of_machine_translation). However sign-language automatic translation is not available yet in a reliable quality that can be used for every-day communications and emergency communications.

NOTE: Potential solutions for next generation emergency communications including automatic translation are being considered as part of the European H2020 project EMYNOS [i.61].

### 7.3.2 Solution #1: Automatic translation at the PSAP

#### 7.3.2.1 Description

The indication of the Total Conversation user's communication needs are signalled by the Total Conversation user terminal using the user preference extension mechanism described in clause 7.2.3.2.

This solution assumes use of an "off the shelf" or specifically developed application providing automatic spoken/text language translation, i.e. spoken to spoken language, spoken to text, text to spoken and text to text language translation.

- 1) The PSAP is configured to use an application for automatic spoken/text language translation.
- 2) The PSAP receiving the incoming Total Conversation emergency session checks the indication of needed language / modality to see if automatic translation between the requested Total Conversation user language/modality and the language/modality understood by the call-taker is available.
- 3) If automatic translation is available, then it takes place at the PSAP between the Total Conversation user and the PSAP call-taker. The Total Conversation user terminal should receive information about the PSAP's spoken language/modality supported in the response messages from the PSAP.

### 7.3.2.2 Evaluation

#### Advantages:

- The method can re-use the indication of the Total Conversation user's communication needs as described in clause 7.2.3.2, but avoids the additional challenges presented by having to identify and invoke a separate relay service.
- Translation applications are generally easy to install and link to the SIP and IMS software (some may already be included) and can be available to all. This can also make it a cost effective solution.
- This method can be convenient for roaming scenarios when the Total Conversation user contacts a local PSAP for the emergency session.
- No additional standardization work is needed to support this method, as it is performed at the PSAP.

#### Disadvantages:

- Automatic translation is limited to support of translation between spoken languages, between spoken and written languages and between written languages only. It cannot currently be used to support the communication needs for someone requiring translation between signed and spoken language.

NOTE: It is possible that reliable automatic translation will be developed also for signed languages in the future. It is not within the scope of the present document to consider this further.

- Automatic translation applications for use in an emergency context would need rigorous testing and in some jurisdictions potentially certification in order to provide sufficient confidence in PSAP operators and Total Conversation users of their utility and acceptability.

#### Conclusion:

Whilst automatic translation applications would be easy to deploy and be in-line with developing trends the ICT area, their accuracy and acceptability by stakeholders should be carefully considered.

In the longer term as the technology evolves, automatic translation could potentially be used to reduce the need for translation relay services offering translation between spoken languages, between spoken and written languages and between written languages. It is therefore recommended that the usage of such applications are further explored by stakeholders.

## 7.3.3 Solution #2: Automatic translation at the Total Conversation user terminal

### 7.3.3.1 Description

This solution assumes the use of an "off the shelf" or specifically developed application providing automatic spoken / text language translation, i.e. spoken to spoken language, spoken to text, text to spoken and text to text language translation.

The Total Conversation user may activate the language translation application on his terminal to automatically translate the communication received to the language understood by the Total Conversation user and the communication sent to the language spoken or written at the PSAP. E.g. while roaming in France, a UK Total Conversation user prefers to listen and speak in English, so the received speech from the PSAP is translated to English, and the spoken English from the Total Conversation user is translated to French before sending. This could also work for written language requirements.

This needs an indication that a local translation is made, to be communicated to the PSAP in the signalling messages as described in IETF RFC 6497 [i.51], by adding the transformed content in subtag -t (e.g. French translated from English is declared as fr-t-en). Also the preferred languages of the Total Conversation user need to be provided to the PSAP as defined in IETF draft-ietf-slim-negotiating-human-language [i.40] for the fall back case where the local translation may fail. The present document does not cover this method in more detail due to its complexity.

The method discussed in the present document is simple and easy to implement, where the Total Conversation user receives the communication in the language sent by the PSAP and local translation is performed at the Total Conversation user terminal. This needs the Total Conversation user to activate the translation program installed on the terminal. The received communication should be shown in both the received and translated language in written text, if text is available. This method can enable the Total Conversation user to use other devices or terminals on which the automatic translation application is installed, but that are not configured for Total Conversation for contacting emergency services, especially if the PSAP supports such feature as well.

### 7.3.3.2 Evaluation

#### Advantages:

- Translation applications are generally easy to install and link to the SIP and IMS software (some maybe installed as part of the software delivered within a purchased device) and can be available to all. This can also make it a cost effective solution.
- This method can be convenient for roaming scenarios when the Total Conversation user contacts a local PSAP for the emergency session.
- It is possible to use this method without additional standardization work.
- Support of automatic translation in the Total Conversation user terminal as the general solution would reduce the need to indicate the Total Conversation user's communications needs to the PSAP.

#### Disadvantages:

- Automatic translation is limited to support of translation between spoken languages, between spoken and written languages and between written languages only. It cannot currently be used to support the communication needs for someone needing translation between signed and spoken language.

NOTE: It is possible that reliable automatic translation will be developed also for signed languages in the future. It is not within the scope of the present document to consider this further.

- Automatic translation applications for use in an emergency context would need rigorous testing and in some jurisdictions potentially certification in order to provide sufficient confidence in PSAP operators and Total Conversation users of their utility and acceptability.

#### Conclusion:

Whilst automatic translation applications would be easy to deploy and be in-line with developing trends the ICT area, their accuracy and acceptability by stakeholders should be carefully considered.

In the longer term as the technology evolves, automatic translation could potentially be used to reduce the need for translation relay services offering translation between spoken languages, between spoken and written languages and between written languages. It is therefore recommended that the usage of such applications are further explored by stakeholders.

## 7.4 Media

### 7.4.1 General

The following clauses present options of interest for the media components of Total Conversation.

In general, all aspects of media and media options described herein are valid for the PSAP connection, the Application Service Provider connection, and the Assisting Service connection. They are also valid for the Total Conversation user terminal connection for the cases when the Application Service Provider does not invoke transcoding in Total Conversation emergency calls.

A common topic for all media is security. Media secured by SRTP-DTLS [i.50] is recommended by IMS specifications, by IETF standards, by EENA NG112 LTD [i.1] and by ETSI TS 101 470 [i.2]. In addition, ZRTP [i.44] is recommended as an alternative by ETSI TS 101 470 [i.2], because of its more convenient method to achieve end-to-end security, but because that method is not mentioned in the other specifications, it is not realistic to expect that ZRTP will be commonly used in Total Conversation emergency calls and therefore the recommendation of SRTP-DTLS is seen as sufficient.

## 7.4.2 Video

In order to establish and maintain good quality for language communication, a suitable balance should be set between available bandwidth, frame-rate and picture resolution, so that the quality goals documented in ETSI TS 101 470 [i.2] are achieved. This goal usually means to put more emphasis on high frame-rate than on high picture resolution.

For good performance of video in the varying conditions that often prevail in IP networks, methods for adaptive rate control should be applied. This implies for example sending proper RTCP reports about video performance and acting on those reports. It also implies keeping the amount of video bit stream buffered for transmission at the source to a minimum so that excessive delays are not created.

For the rapid actions necessary to maintain good quality video, it is preferred to use RTCP feedback according to the AVPF profile. However, many older devices support only the AVP profile, and in order to achieve a workable video interoperability with such devices, it is recommendable to support both AVPF and AVP. The disadvantage of using this method is that proper standards compliant interoperability between devices with and without support for AVPF and SRTP creates long and complex SDP bodies with contents that can create interoperability issues with earlier devices. Potential, but in some cases non-standard, means to reduce these risks and the complexity are found in IMTC 1013 [i.64].

At the time of writing, the common video codec that all functional entities should implement is Recommendation ITU-T H.264 [i.65].

## 7.4.3 Audio

Wide band audio is of great value for spoken language perception by many Total Conversation users.

For IMS, ETSI TS 126 114 [i.7] specifies a range of audio codecs to support. A selection of them should be supported by Total Conversation user terminals for emergency calling, including wide-band alternatives. At the time of writing the same codecs are specified for use in NG 112 ESInets and PSAPs in EENA NG112 LTD [i.1], thereby assuring interoperability.

For IETF SIP, the implementation Recommendation ITU-T G.711 [i.66], and Recommendation ITU-T G.722 [i.67] should be followed as per IETF RFC 6881 [i.11].

## 7.4.4 Real-time text

### 7.4.4.1 General

The real-time text codec T.140 [i.68], transported by RTP as specified in IETF RFC 4103 [i.69] should be supported by all functional components involved in the Total Conversation emergency sessions. Details on its use can be found in ETSI TS 101 470 [i.2], clause 5.3.2. The details for implementation in IETF SIP environment specified are valid also for the IMS case.

Many implementations of Total Conversation are located behind NAT routers and firewalls. Keep-alive actions needed for maintaining connectivity through such network components are described in ETSI TS 101 470 [i.2]. These actions should be initiated and start with transmission of a Unicode BOM character as soon as the call is established with real-time text, or when real-time text is added to the call. This should be done in order to prepare the media path for reception.

Some implementers have commented that the actions after an idle period specified in the fourth paragraph of IETF RFC 4103 [i.69] clause 5.2 are not strictly followed by all implementations. The specification states that any empty T140block sent as primary data 'must' be included as redundant T140blocks in subsequent data packets, in the same way that normal text T140blocks would be, unless the empty T140block is too old to be transmitted.

However, some implementations start after an idle period, regardless of its length by sending all generations of redundant empty T140blocks with random timer offsets, while other implementations start after a long idle period by sending just primary text in the first packet, and then building on this with generated redundancy in the following transmission intervals. Receivers should be prepared to receive both coding variants. The sending side may select any of these ways to code transmission after an idle period.

#### 7.4.4.2 Multi-party handling of real-time text

Most current implementations of Total Conversation user terminals are not conference-aware and can only handle one media-stream of each type in a call. As such, they need specific consideration when connecting to a multi-party bridge in order to adequately handle presentation of multi-party calls. Multi-party bridges should therefore interrogate the multi-party awareness of the terminals involved in multi-party calls. For IETF SIP and IMS systems, this is done according to the framework for SIP conferences in IETF RFC 4353 [i.23].

In the case of conference-unaware terminals capable of handling only one call at a time, the bridge needs to do the mixing. The mixing of real-time text into one stream to a conference-unaware terminal should be done in a way that makes the mixer of the bridge create easily distinguishable text from each participant, while also maintaining the conversational real-time impression of the presentation. An example of a possible mixing method is documented in "Multi-party real-time text for presentation in conference-unaware user agents" [i.63].

Presentation of real-time text to a conference-aware terminal should be done through any of the methods specified in IETF RFC 4353 [i.23]. Possible arrangements of text are specified in Recommendation ITU-T T.140 [i.68]. When used in Total Conversation emergency calls, one possible position for presentation of real-time text in conference-aware terminals is under each video picture of the same participant. Other layouts can be imagined. A default layout should be available that has the opportunity to provide a view of the text from the other participants without detailed planning of the display.

## 8 Recommendations

### 8.1 General

The following clauses contain recommendations for an implementation based on solutions evaluated under clause 7 of the present document.

### 8.2 Assisting services

Looking at existing 3GPP and IETF specifications to see how Total Conversation for Emergency Communications can be supported, as documented in the earlier clauses of the present document, there is one major difference compared to voice only emergency sessions. This is the fact that the additional media capabilities of Total Conversation session enable the inclusion of an assisting relay service between the Total Conversation user and the call-taker at the PSAP to provide interpretation between one modality of communication via one medium and another modality via another medium, e.g. between sign-language via video and spoken language via audio.

Whilst it is possible to support assisting relay services for non-emergency Total Conversation sessions, the technical interfaces are not clearly specified by existing standards. Current standards are also sufficiently flexible to allow for different implementations in different networks. However, for Total Conversation for Emergency Communications to be in accordance with European Policy [i.24] is it necessary for such assisting relay services to be available free of charge to users having additional communication needs, regardless of whether they are calling in their home networks or travelling to another member state. This means that a well specified and general solution is desirable.

The ability to invoke an assisting relay service in a range of different operating environments presents a number of challenges, solutions for which have been described and evaluated under clause 7 of the present document. Based on the discussions and conclusions in that clause, the following recommendations are made with respect to implementation of Total Conversation for emergency communications with assisting services.

#### **Recommendation #1 - Invocation of assisting services:**

- Invocation and control of the multi-party conference to include an assisting relay service should be the responsibility of the PSAP. See clause 7.2.2.4.
- The NG112 PSAP should answer all Total Conversation emergency sessions directly into a bridge in order to facilitate subsequent inclusion of an assisting relay service or other assisting party when needed.

NOTE 1: This constitutes a narrowing of the alternatives described in clause 4.8 of the EENA Ltd specification [i.1].

- When the Total Conversation user terminal is IETF SIP registered, or IMS registered in their home network, and the Total Conversation user terminal or the Application Service Provider in the home network usually invokes an assisting relay service for non-emergency Total Conversation sessions, they may invoke the assisting relay service in the same way for Total Conversation emergency sessions.
- In the case that an assisting relay service is invoked by an entity other than the PSAP:
  - Additional Data should be used to notify the PSAP that the relay service is included in the session.
  - The indication of the Total Conversation user's communication needs should be updated to indicate support of spoken language in both directions (if applicable) so as to avoid unnecessary additional invocation of a relay service by the PSAP.
  - The invoking entity should not wait for the connection of the assisting relay service before attempting to connect the PSAP, i.e. the connection should take place in parallel.
- In the case that an assisting relay service is to be invoked by the PSAP:
  - The Application Service Provider should override any parameters included by Total Conversation user explicit action that would result in the assisting relay service being included in the session in a way that is not controlled by the PSAP.

NOTE 2: How this would be implemented by the Application Service Provider is outside the scope of the present document.

- The Application service provider should interrogate the answer from the PSAP for availability of the proper media, modality and language support for a functioning call, and include an appropriate relay service if needed.
- The Total Conversation user terminal may do a similar comparison between desired and available support in the call, and include an appropriate relay service to cover gaps.
- In the event, that the assisting relay service is not invoked by the Total Conversation user terminal or by the Application Service Provider including the URI of the needed assisting service in the routing path of the emergency session, i.e. the relay service does not receive the "urn:service:sos" indication which would be included in the INVITE towards the PSAP, the invoking entity should include SIP priority tag in the call to the assisting relay service to indicate its emergency status as per clause 7.2.4.5.
- In the event of roaming where the Total Conversation emergency session is routed via the Total Conversation user's home network to the home PSAP, an assisting relay service may, if needed, be invoked by the home PSAP before it connects a PSAP in the visited network.
- If a Total Conversation user makes an emergency call and needs support of another modality than plain speech in both directions, but a PSAP with only audio answers, then the calling Total Conversation user or its Application Service Provider may in the case of an IETF SIP emergency session invoke a suitable assisting relay service as a third party in the call in a way that does not interrupt the call if that functionality is supported. This situation would occur if the addressed PSAP were not NG112 enabled.

#### **Recommendation #2 - Indication of communication needs:**

- If the Total Conversation user has particular communication needs that can be indicated using language / modality tags, then their communication needs should be signalled by inclusion of the needed language / modality tag per media line in the SDP offer sent with the INVITE by the Total Communication user terminal or the communication service provider. See clause 7.2.3.2.

NOTE 3: The language / modality tags could be either the existing "lang" tags [i.33] or the "humintlang" tag being developed in IETF SLIM [i.38].

#### **Recommendation #3 - Identification of assisting service:**

- If the Total Conversation user prefers a specific assisting relay service to be included in the session, then the identity of the preferred assisting relay service should be signalled by the Total Conversation user terminal sending a REFER message to the PSAP indicating the preferred relay service URI in the Refer-To header as described in clause 7.2.5.3.

- A Europe wide database of trusted assisting relay services for inclusion by PSAP in Total Conversation emergency sessions, similar to that for PSAPs described in [i.48] should be developed, e.g. by EENA to enable PSAPs in any country to identify an appropriate relay service to meet the Total Conversation user's identified communication needs. See clause 7.2.3.2.
  - This database should contain as many signed, spoken, and written languages as possible, including all the national languages of European member states.
  - The database should be a machine readable online resource so as to facilitate automatic invocation of the necessary assisting relay service during session establishment.
  - The database should also be accessible by the PSAP call-takers to allow manual invocation of an assisting relay service if identified as needed during an ongoing call.
  - The services listed in the database should provide a SIP interface suitable for invocation as a third party in a multi-party conference, see IETF RFC 4353 [i.23], e.g. as specified in EENA NG112 LTD [i.1].

#### **Recommendation #4 - PSAP support of text communication:**

- If the user needs text communication, or text one way and spoken communication the other way, and the PSAP is NG112 enabled (implying that it supports and is interoperable with the user terminal in real-time text), then the call should be connected to the PSAP without any assisting relay service. It is assumed that the PSAP and the user will then be able to communicate in text, possibly combined with speech.

## 8.3 Roaming sessions routed via the home network

A challenge not directly related to the inclusion of assisting relay services, or even especially unique to Total Conversation emergency sessions, is how to deal with sessions from Total Conversation users roaming in visited networks whose emergency sessions are routed via their home networks. This will be the case for roaming IETF SIP registered users.

#### **Recommendation #1: Session routing**

- An open standardized system, e.g. using the HELD protocol as described in clause 6.5.5.2, should be implemented for routing of emergency calls from access networks covering all of Europe, even for countries who are not yet NG112 enabled.
- If no response is received from the use of this mechanism, the call should be routed to a default ESInet address provided by the Application Service Provider.

#### **Recommendation #2: PSAP contact database**

- To cater for the for the case when the call cannot be routed automatically to a PSAP in the visited network e.g. which might occur during migration, the existing EENA/ECC PSAP database [i.48] should be extended to include URIs for NG112 based PSAPs in order to allow the home PSAP to contact an appropriate PSAP in the visited network.
- The database should be developed into a machine readable online resource for PSAPs to enable automatic identification of the appropriate PSAP in the visited network based on the location of the calling Total Conversation user.
- The database should also be accessible to the call-taker for manual calling between PSAPs.

## 8.4 Media

#### **Recommendation 1: General media support**

- All media necessary for Total Conversation should be supported and implemented in all PSAP call-taker workstations used for Total Conversation access. This is to facilitate providing the service to the Total Conversation user at call initiation and if additional media is needed during the lifetime of the call.



NOTE: The cost associated with providing NG112 PSAP with technology that supports all the media needed for Total Conversation (Voice, Video and RTT) is not considered prohibitive. This is because the cost of multimedia hardware and software, as well as that for the necessary bandwidth, has fallen significantly during the last decade.

- All media should be secured by use of SRTP-DTLS, see IETF RFC 5764 [i.50].
- All supported media in the Total Conversation user terminal should be provided during the session initiation by inclusion of their associated media feature tags in the SIP Contact header. This is 'required' by IMS specification ETSI TS 124 229 [i.22] and will enable policy based routing on media support in the ESInet and PSAP.
- Users who may depend on some media during the Total Conversation emergency communication session should initiate all the necessary media from the beginning of the call. This ensures the routing to the most appropriate ESInet and PSAP supporting the Total Conversation emergency session for the lifetime of the call.

#### **Recommendation #2: Video quality and performance**

- Methods for adaptive rate control should be applied to ensure consistent levels of quality in varying transmission conditions. E.g. RTCP reports should be promptly sent and rapidly acted on. Transmission buffer content amount should be kept low at the source.
- In order to ensure interoperability between older Total Conversation user terminals and NG112 PSAPs, both AVP and AVPF should be supported at the PSAP.
- H.264 [i.65] baseline profile levels 1-3 should be supported by all parties in a Total Conversation emergency call.

#### **Recommendation #3: Audio codec selection and end to end interoperability.**

- In order to provide good speech perception, the Total Conversation user terminal should support a wideband audio codec for use in emergency communications.
- Support of the codecs specified in EENA NG 112 LTD [i.1] should be provided by the ESInets and PSAPs, and a selection of the ones specified for IMS Multimedia Telephony in ETSI TS 126 114 [i.7]. This should be sufficient for interoperability without transcoding.
- If the statement above does not result in compatible codecs, the Application Service Provider should provide transcoding at the border of the access network and the ESInet to ensure interoperability between Total Conversation user terminal and the PSAP.

#### **Recommendation #4: Real-time text details**

- The details about real-time text communication specified in ETSI TS 101 470 [i.2] for IETF SIP should also be applied in IMS.
- Real-time text transmission should start with sending the Unicode BOM character as soon as a call with real-time text is completed, and as soon as real-time text is added to a call.
- Receivers of real-time text according to IETF RFC 4103 [i.69] should accept receiving packets after an idle period both with and without redundant text consisting of empty T140blocks regardless of the length of the idle period.

#### **Recommendation #5: Multiparty handling of real-time text**

- The multi-party bridge used for a Total Conversation emergency session should interrogate the conference awareness of a Total Conversation user terminal involved in a multi-party session.
- If the Total Conversation user terminal is conference unaware, then the multi-party bridge should mix the real-time text in such a way as to make the text from each party clearly distinguishable.
- If the Total Conversation user terminal is conference aware, then presentation of the real-time text may be done through any of the methods specified in IETF RFC 4353 [i.23].

## 8.5 Security

Requirements for privacy and data protection of user information such as location, identity, and also indication of a Total Conversation user's communication needs, are subject to local regulations. This will be applicable for all entities that receive such data including any third party relay service. Authentication and trust between the Total Conversation user, Application Service Provide, PSAP, and assisting relay service is therefore essential.

### Recommendation #1: Inclusion of trusted relay service

- Where possible the PSAP should invoke trusted assisting relay services into any multi-party conference, thus avoiding the need for additional authentication mechanisms.

### Recommendation #2: Inclusion of non-trusted relay service

- Where an assisting relay service is provided or invoked by the Total Conversation user or Application Service Provider, then an appropriate mechanism should be used to authenticate the relay service.

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## 9 Recommended updates and new specification work

### 9.1 Gap analysis and remedial actions

#### 9.1.1 Use of terms GTT and RTT in 3GPP specifications

##### 9.1.1.1 Description of issue

References in ETSI TS 122 101 [i.9] to GTT (Global Text Telephony) steer the reader of the specification towards specifications ETSI TS 122 226 [i.34] and ETSI TS 123 226 [i.35] for requirements and implementation respectively. However, RTT as part of a Multimedia Emergency Session (MES) supporting Total Conversation is covered in ETSI TS 122 173 [i.6], with codec implementation specified in ETSI TS 126 114 [i.7]. It is not clear from the existing references which path should be followed for implementation, and it is possible that the resulting confusion could lead to incompatible implementations.

ETSI TS 126 114 [i.7] is the most appropriate and up to date specification for media handling and interaction for Total Conversation in IMS, so the references in ETSI TS 122 101 [i.9] should guide the reader towards that specification for implementation. In addition, the text in the specification should make it clear that GTT is a feature that enables real-time text conversation, and that text media is part of the multimedia service enabled by IMS.

##### 9.1.1.2 Remedial action

A Liaison Statement outlining the issue was sent from ETSI TC EMTEL (June 2015) to 3GPP SA1 #71 (August 2015). Two draft Change Requests to ETSI TS 122 101 [i.9] and to ETSI TS 122 173 [i.6] were produced respectively in S1-154233 and S1-154236 for consideration by 3GPP SA1.

##### 9.1.1.3 Status

The Change Requests "Clarification of relationship between GTT and Real Time Text" to ETSI TS 122 101 [i.9] and ETSI TS 122 173 [i.6] were approved in SP-1500749 at 3GPP SA #70 (December 2015).

#### 9.1.2 Order of payload types for real-time text in 3GPP specifications

##### 9.1.2.1 Description of issue

In ETSI TS 129 332 [i.36], clause 10.2.3.5, an example SDP is provided for a session including real-time text with redundancy. The order of the payload types in the m-line of the SDP is however incorrect. The payload type number for T140 occurs first, followed by the payload type number for the payload including redundancy. In order to prioritize use of redundancy the order should be reversed. If not, then there is a risk that the real-time text medium is transmitted without redundancy, which could result in an unacceptable rate of text loss.

The correct order is shown in ETSI TS 126 114 [i.7], so there is also an inconsistency in the 3GPP standard. This order is also shown by Errata 1203 to IETF RFC 4103 [i.37]. Errata 1203 to IETF RFC 4103 [i.37].

### 9.1.2.2 Remedial action

The issue was communicated to 3GPP CT4 by email and a Change Request was produced by a 3GPP member company in C4-151346 CR0196 to ETSI TS 129 332 [i.36] Correction on SDP for Real-Time text.

### 9.1.2.3 Status

The Change "Correction on SDP for Real-Time Text" is approved in CP-150430.

## 9.1.3 Provision of feedback and comments on draft version ETSI ES 202 975

### 9.1.3.1 Description of issue

Feedback and comments on the "draft version 1.4.101" of the relay service requirements in ETSI ES 202 975 [i.5] were requested by ETSI TC HF. The following gaps were found:

- missing condition in clause 6.3.1 that imply the requirement of 24 hour service has to be implemented for the case of Emergency services;
- missing requirement in clause 6.5, where emergency services need to be given high priority in the queue;
- missing requirement in clause 6.15, where the establishment of an emergency call by a relay service on behalf of a user, shall provide the location information for that user; and
- other minor editorial corrections.

### 9.1.3.2 Remedial action

Resolutions were provided to insert the missing requirement listed above. This was shared with the rapporteur of the specification by email.

### 9.1.3.3 Status

The proposed changes listed above (clause 9.1.3.1) except the first one are accepted and included in the published version ETSI ES 202 975 [i.5], in the following manner:

- Missing condition in ETSI ES 202 975 [i.5] clause 6.3.1 was not accepted, since the existing sentence "*A service claiming to be a 24-hour service shall be open 24 hours a day, every single day of the year.*" was seen to cover the condition for emergency services.
- Missing requirement in ETSI ES 202 975 [i.5], clauses 6.5 and 6.15, the changes have been incorporated in clause 6.15 as follows:  
*"When the relay service receives information related to the emergency call, e.g. location information, such information shall be made available to the emergency service.  
 Emergency calls shall be given priority in getting a communications assistant assigned to them."*

## 9.1.4 Inclusion of Total Conversation for emergency services as a regional requirement

### 9.1.4.1 Description of issue

The GSMA Interworking and Roaming Experts Group (IREG) produces a number of Permanent Reference Documents (PRDs) related to implementation of services using IMS. In particular, the Next Generation Roaming in LTE (NGRiLTE) activity has produced PRDs IR.92 [i.45] and IR.94 [i.46] that respectively specify IMS profiles for Voice and SMS, and for Conversational Video Service. These GSMA PRDs are interdependent, and IR.92 also specifies support of Global Text Telephony as a regional requirement.

GSMA PRDs IR.92 and IR.94 include good provision for support of IMS Multimedia Emergency Sessions as specified in ETSI TS 122 101 [i.9]. They also support Total Conversation as PRD IR.94 is intended to be combined with PRD IR.92 and all three media of video, audio and real-time text refer to ETSI TS 126 114 [i.7]. However, there is no explicit mention of Total Conversation for emergency communications as fulfilled by ETSI TS 122 173 [i.6] and ETSI TS 126 114 [i.7]. Moreover, the latest version of PRD IR.92 does not reflect European policy requiring the support of Total Conversation for emergency communications.

#### 9.1.4.2 Remedial action

A Liaison Statement was sent from EMTEL #33 in EMTEL(15)000023r2 to officially inform GSMA NGRiLTE of the activities relating to production of the present document, and to request the inclusion of support for Total Conversation for emergency communications as a regional requirement in order to comply with European policy as per [i.24].

#### 9.1.4.3 Status

The GSMA NGRiLTE group updates PRD IR.92 on an annual basis. At the time of writing GSMA NGRiLTE is discussing a proposed change that is expected to be approved for inclusion in the next version of IR.92.

## 9.2 New and revised specifications

### 9.2.1 General

This clause recommends changes to existing specifications and/or the introduction of new specifications to achieve the implementation and deployment of Total Conversation for Emergency Communication based on the present document.

### 9.2.2 3GPP/ETSI specifications

#### 9.2.2.1 Relay service interface for emergency communications

Work undertaken for the present document did not find a technical interface description for relay services used in emergency communications. As such, there appears at present to be no means to ensure interoperability between PSAPs and the relay services that will be used by Total Conversation users. A standard describing the interface for connecting relay services for emergency communication is therefore needed.

#### 9.2.2.2 Test Specifications for Total Conversation

Test specification owned by 3GPP RAN5 might be updated if found necessary to cover for the implementations of Total Conversation for Emergency Communications, in particular while invoking the relay services. This could for example include testing the inclusion of the additional data (see Additional Data Related to an Emergency Call [i.38]), using RTT as a media by the Total Conversation user, etc.

#### 9.2.2.3 AT commands

AT commands, see ETSI TS 127 007 [i.52], need to be defined for Total Conversation, especially for the identification of the media/modality and for providing the assisting relay service address.

#### 9.2.2.4 ETSI TS 101 470

Annex B identifies some inconsistencies between the present document and ETSI TS 101 470 [i.2]. It is recommended to revise ETSI TS 101 470 [i.2] in order to resolve these, and to include other recommendations from the present document where appropriate.

### 9.2.3 IETF specifications

#### 9.2.3.1 Best practices in real-time text

The present document makes recommendations in clause 8 related to real-time text, which at the time of writing are not covered in any other standards document. Therefore, as the IETF is the source of the real-time text transmission specification referenced for use in Total Conversation, an IETF best practice document on this topic would be beneficial. The document should include the issues covered in Recommendation #4 and #5 of clause 8.4, which relate to the use of IETF RFC 4103 [i.69] and to multi-party handling of real-time text.

## 9.2.4 EENA Specifications

### 9.2.4.1 SDP based routing

IETF RFC 6443 [i.14] recommends that the emergency call INVITE should include a SDP offer in order to enable routing based on media. In addition, the IETF SLIM group working on solutions for directional and media level indication of preferred languages / modalities in draft-ietf-slim-negotiating-human-language [i.40], are focussed on inclusion of the tags in the SDP part of relevant SIP messages. However, the current EENA NG112 LTD specification [i.1] does not include provision for routing based on SDP because SDP attributes are not described using machine parsable XML. Changes are therefore needed to the EENA NG112 LTD specification [i.1] in order to support routing based on the contents of SDP and to bring it into line with the IETF specifications on which NG112 implementations will be built.

## 9.3 Open issues

### 9.3.1 Use of non-usual terminals

One issue that arises for all potential solutions to the issue of the Total Conversation user indicating their particular communication needs, is that this will only work if they are registered in the network using their usual subscription (e.g. are IETF SIP registered via their usual account), or are using their own Total Conversation terminal that is configured to include an indication of their communication needs in the emergency session signalling. In the event that the Total Conversation user has to use a third party terminal that it is not associated with their subscription, or in which they have not configured their preferences, then it will not be possible to communicate their needed language/modality to the PSAP that would in turn enable automatic invocation of an appropriate assisting service.

This is one instance where it does not appear possible to ensure that accessibility for disabled users is equal to that of non-disabled users who benefit from being able to use the default speech mode of emergency services. However, whilst this is a gap that cannot be filled, it would seem to be one likely to occur with little probability. Also in the event that it were to happen, it would seem possible in most instances to communicate using the media that was available to the Total Conversation user and the PSAP, in order to identify and manually invoke the appropriate relay service. The means for invocation of appropriate assisting relay services manually by the PSAP during a call have been discussed and proposed solutions appear in other clauses of the present document. It is therefore noted here as an open issue, but one for which there appears no need to develop a solution.

### 9.3.2 Configuration of Total Conversation settings

#### 9.3.2.1 General

To achieve wide implementation of Total Conversation, there need to be commercial incentives for Service Providers to provide the service and enabling configuration of Total Conversation user preferences and terminal settings in Application Servers. As long as the dominating deployment of Total Conversation is in the accessibility area, the commercial incentives are created by public procurement of the services.

#### 9.3.2.2 Setting by Application Service Provider

In this case, the Application Service Provider provides the Total Conversation service to its subscribers. This needs User Profile settings that include the user's subscription preferences. E.g. if a Total Conversation user prefers to have a specific relay service invoked while initiating a Total Conversation session then this is configured in the User Profile by the Application Service Provider. In the case, where the Total Conversation user is roaming outside of their home network, and the session signalling is routed via their home network, the local service provider would normally expect to charge the Total Conversation user for the service and in particular for the leg towards the relay service.

However, as Emergency services should be provided to the user free of charge, this kind of service cost would not be paid to the service provider by the Total Conversation user.

#### 9.3.2.3 Setting of the Total Conversation Terminal

In this case, the Total Conversation user terminal needs to include a Man Machine Interface that allows the Total Conversation user to configure their terminal to include the desired parameters, e.g. the language and modality, the URI of the relay service of their choice, in signalling. AT commands can be used for this purpose, and are described for IMS devices in ETSI TS 127.007 [i.52]. This would need extension to cover for the Total Conversation requirements.

### 9.3.3 Rapid captioning of speech

Rapid captioning of speech appears to need a service need indication rather than a modality need indication. Such indications are as yet to be defined and are not within the scope of the present document.

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## 10 Conclusions

The present document considers the implementation of Total Conversation for emergency communications. It makes recommendations in clause 8 that will allow the requirements of ETSI TS 101 470 [i.2] to be met whilst complying with European Union policy as per the EU Universal Service Directive [i.24].

In an ideal world, next generation PSAP call-takers would be able to communicate in a range of languages and modalities including sign-languages in order to serve the needs of any potential caller. That is not practical, but call-takers will in the future be able to handle calls using a combination of speech and real-time text while also having video in the call. For interpretation of sign-language and other modalities, complementing assisting relay services may need to be employed.

For the most part, support for Total Conversation for emergency communications is already provided by existing ETSI IMS and IETF SIP standards as also specified in ETSI TS 101 470 [i.2]. The Next Generation 112 system based on these standards, and specified by EENA in [i.1], will also support Total Conversation for emergency communications. However, in order to adequately cater for disabled users who rely on communication modalities other than the ones managed by NG112 PSAPs, the NG112 system should support call processing able to take signalled language / modality attributes in SDP into consideration, so that suitable assisting relay services can be identified and included in emergency calls. System implementation would also benefit from other standardization activities identified in clause 9.2 and in particular specification of the interface between NG112 PSAPs and assisting relay services.

From a Total Conversation user perspective, they should be able to make emergency calls in the same way as non-emergency calls. This includes the case where assisting relay services are needed. However, that is not always possible because of the way in which systems and Total Conversation emergency services are implemented, and in particular because of the need to cater for the roaming case of the Total Conversation users. An implementation that could serve for all Total Conversation users regardless of their roaming status, should therefore at least allow a Total Conversation user to identify an assisting relay service that is known and trusted, and that can be then added to a multi-party conference controlled by the PSAP. If that is not possible, then language / modality tags in session establishment signalling can be used to identify the specific communication needs of the caller so that an appropriate assisting relay service can be identified by the PSAP for inclusion in the session. The recommendations in clause 8 outline how this could be achieved.

The present document focuses on refinement of existing standards and specifications. To ensure harmonized implementation and interoperability of Total Conversation emergency systems across Europe, the recommendations from clause 8 should be followed. Further regulation may however be needed to enforce the recommended solutions in order to ensure that in the future, adequate service levels are available for all citizens in line with European policy.

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# Annex A: Alternative solution for relay service identification and invocation

## A.1 General

The assumption in the present document is based on guidance provided by stakeholders that it should always be the PSAP which identifies and includes a relay service if needed by the Total Conversation user (see clause 7.2.3.4). However, in some cases it would also be possible for relay service to be added to the emergency session in the same way that they are for a non-emergency session. E.g. when the Total Conversation user is IETF SIP registered or SIP registered in the IMS home network in which case the session will be routed via the home service provider able to interrogate the User Profile.

This annex describes a hybrid solution that would allow a mix between PSAP and other entity invocation and inclusion of relay services depending on the registration status of the Total Conversation user terminal.

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## A.2 Hybrid solution rationale

### A.2.1 Sessions where the Total Conversation user terminal is IETF or IMS SIP registered

#### A.2.1.1 Sessions where the Total Conversation user terminal is not roaming

For emergency sessions where the Total Conversation user terminal is IETF SIP or IMS registered, i.e. when the terminal is not IMS emergency registered, call routing for the session will be via the home service provider. In that case, it should be possible to include relay services in a call using the same mechanism that is used for non-emergency sessions, e.g. the relay service could be included automatically by the service provider, or the necessary routing information could be included directly by the Total Conversation user terminal in the INVITE message. There would be no need for the Total Conversation user terminal to provide any indication to the PSAP about the user's communication needs, or a specific relay service. Such emergency sessions would appear likely to account for the majority of emergency sessions so it could be considered beneficial that the same relay services that are normally used by Total Conversation users when making calls, can be included in a similar way. Indeed, in the case where a "PSAP only" invocation of relay services solution was adopted, it would be necessary to specify behaviour for Total Conversation user terminals and/or home network service providers such that any normally used relay services were not invoked, though the same service might later be invoked by the PSAP.

Any relay service that was added to and present in the session when it was answered at the PSAP should use the Additional Data mechanism described in [i.38] to insert information about itself in the signalling of the session for the information of the call-taker. This would facilitate inclusion of the same relay service in the event of any call-back to the Total Conversation user. As call-back is treated as a "normal session" it would be useful also to consider interaction with the mechanisms used for invocation of relay services in terminating sessions. This mechanism could be a terminating filter in the session path.

#### A.2.1.2 Sessions where the Total Conversation user terminal is roaming

When the user is roaming outside of their home country and is IETF SIP registered, the user will still be SIP registered as normal and the session signalling will be routed via the home service provider. This could potentially present a problem until location support using the HELD routing extension as specified in ETSI ES 203 178 [i.53] and in draft-ietf-ecrit-held-routing [i.55] is deployed, because in that case, the home service provider may not be able to find routing information for the ESInet entry point in the roamed-to country.

NOTE 1: This problem is not unique to Total Conversation for emergency communication, but to all emergency sessions.

In this case, when the access network is not able to identify an appropriate ESInet entry point based on the present location of the Total Conversation user, it should route the session to a designated PSAP in the home network. The designated PSAP should answer the session from the roaming Total Conversation user, and should then conference in a PSAP from the country to which the caller was roaming for the dispatch of the necessary emergency service(s).

NOTE 2: It is beyond the scope of the present document to specify how the designated PSAP can obtain information about the PSAP in the roamed to country. However, the cross border PSAP contact information as collected by EENA as described in [i.48] and maintained by ECC could be used for this purpose.

## A.2.2 Sessions where the Total Conversation user terminal is IMS emergency registered

When the user is IMS emergency registered, as will be the case for all roaming IMS subscribers and for some non-roaming IMS subscribers as described in ETSI TS 123 167 [i.8], the session will not be routed via the home Application Service Provider. Also in that case, only calls to specific emergency numbers are permitted. As such, it will not be possible to include the necessary relay service by the "normal" means outlined above. This is where the solutions described in clause 7.2.3 can be used, including a language/modality or service preference indication in the call signalling.

The Total Conversation user terminal will be aware of when it is emergency registered, and in that case should be configured to include a needed language / modality indication in the session signalling, using the solution recommended for issue #2 in clause 7.2.3. That indication can be used in the ESInet for routing to a PSAP / call taker that supports the indicated language/modality. This might not be the one local to the Total Conversation user or the incident being reported.

NOTE 1: It is not in the scope for the present document to specify how the ESInet routes based on SDP.

If a PSAP operator cannot be found that supports the language / modality needs of the Total Conversation user, then it will be necessary to invoke a suitable relay service. If the Total Conversation user is calling in their home network, the PSAP should be able to identify a suitable relay service to meet the user's communication needs. If the PSAP were not able to identify a suitable relay service, then another mechanism would be needed to assist the PSAP. This is where the Additional Data mechanism as defined in the IETF Draft of [i.38] could be used.

When the PSAP is not able to handle the indicated communication needs of the Total Conversation user, the call-taker can use information provided by the Total Conversation user terminal in the "Comment" block of the Additional Data as described in clause 7.2.3.4. If the Total Conversation user terminal were configured to include an indication of the Total Conversation user's communication needs and the URI of an associated relay service in this way, the PSAP operator would be able to manually invoke a conference and add the necessary relay service near the start of the emergency session.

NOTE 2: It is outside of the scope of the present document to consider further whether information in the "Comment" block of Additional Data relating to a specific relay service including an associated URI would be trusted by the PSAP.

## A.2.3 Example call-flow

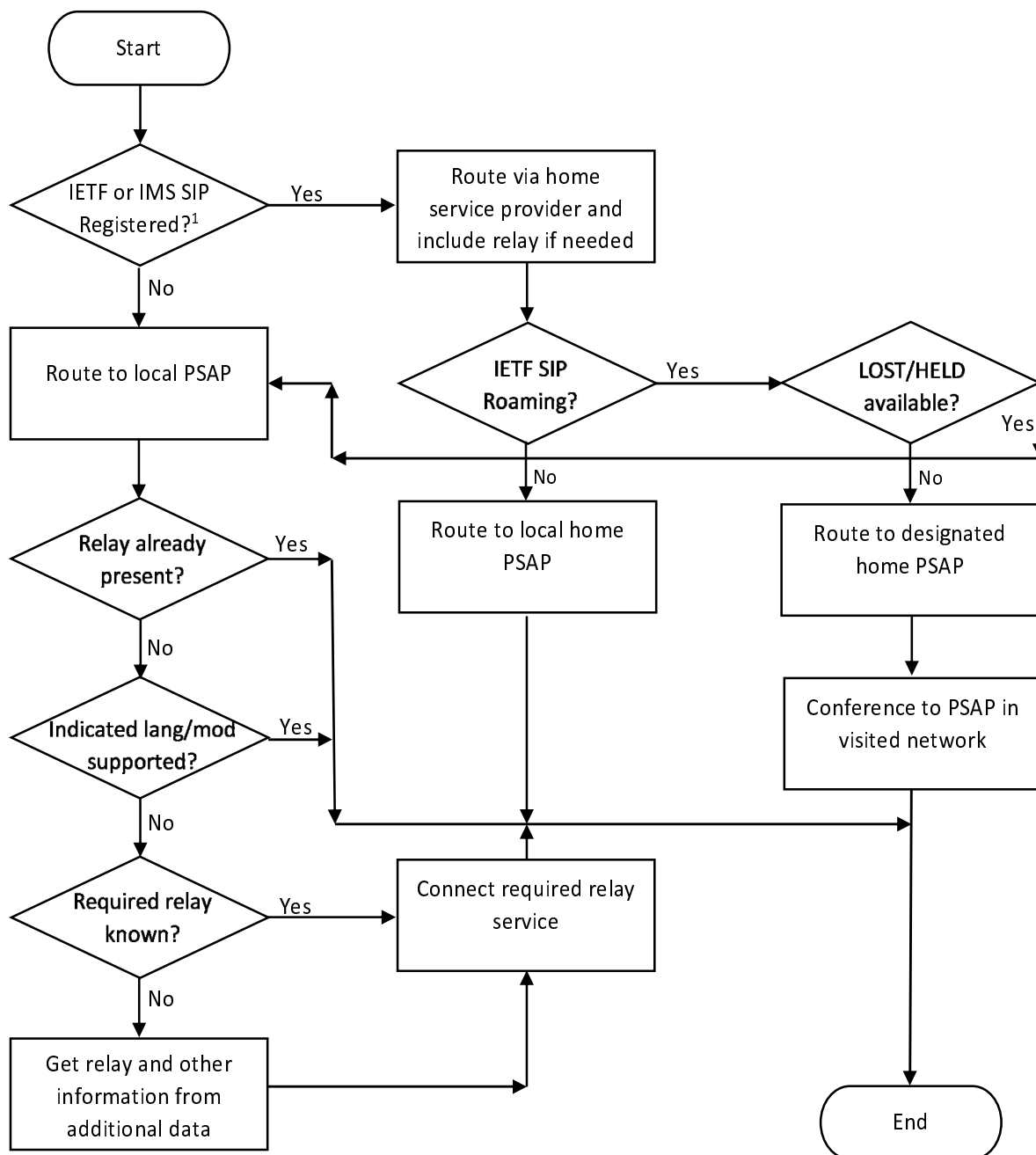
The following call-flow provides an example based on the rational in clause A.2 and shows how the different mechanisms can be used in combination with each other to ensure that appropriate relay services are included when needed, in a range of scenarios. This is not an ideal solution, and could almost certainly be improved through further specification. However, it would seem to represent what might be achievable in the short term using currently specified means and those in the process of being specified. The call flow is shown in figure A.1.

At the start of session establishment:

- If the Total Conversation user is not IMS emergency registered, i.e. is SIP IETF registered or SIP IMS registered in home network:
  - The call is routed via the home service provider and include the relay service if needed as per a non-emergency session. The Total Conversation user terminal need not include a language/modality indication in the signalling.



- If the Total Conversation user is not roaming:
  - The home network routes the session to the appropriate home PSAP.
  - The home PSAP answers the call and the session ensues with the relay service included as needed.
- If the Total Conversation user is roaming:
  - If an appropriate PSAP in the roamed to country cannot be identified, e.g. using LOST or HELD protocols:
    - The home network routes the session to a designated PSAP in the home country.
    - The home PSAP answers the call and the conferences in an appropriate PSAP in the roamed-to country using the EENA / ECC maintained PSAP contact database.
    - The session ensues with the relay service included as needed.
  - If an appropriate PSAP in the roamed-to country can be identified:
    - The home network routes the session to an appropriate PSAP in the roamed-to country.
    - As the relay is already present (if needed), the session ensues without further action necessary.
- If the Total Conversation is emergency registered:
  - The network (home or visited) routes the session to the appropriate PSAP with a language/modality indication included in the signalling by the Total Conversation user terminal, and an indication of any preferred relay service URI plus any non-specified interpretation requirement in the "Comment" block of the Additional Data.
  - As there is no relay present in the session when it arrives at the PSAP:
    - If the PSAP supports the indicated language/modality:
      - The PSAP answers the call and the session ensues without the need for a relay service.
    - If the PSAP does not support the indicated language/modality:
      - If the PSAP is able to identify an appropriate relay service for the language/modality information:
        - The local PSAP connects the appropriate relay service to the session.
        - The session ensues with an appropriate relay service included.
      - If the PSAP is not able to identify an appropriate relay service for the language/modality information:
        - The PSAP operators checks the "Comment" block in the Additional Data provided by the Total Conversation user terminal.
        - The PSAP operator notes the communication needs of the of the Total Conversation user and connects the appropriate relay service as identified in the Additional Data.
        - The session ensues with the appropriate relay service included.



<sup>1</sup> I.e. Not IMS Emergency Registered

**Figure A.1: Call-flow for Total Conversation emergency session potentially involving a relay service**

There is no need to explicitly direct the PSAP call-taker to the Additional Data in the event that the communication needs of the Total Conversation user are not included, because the PSAP call-taker should in any case check the Additional Data sent with an emergency session signalling, regardless of what information is included in the signalling. In this case, the PSAP call-taker acts as the final check point to make the sure communication needs of the Total Conversation user will be satisfied.

This call flow should ensure that in most cases, an appropriate relay service can be identified and connected to an emergency session, albeit with a slight delay in the case where the PSAP operator has to interrogate the Additional Data in order to identify the needed relay service and add it manually.

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## Annex B: Outstanding issues from ETSI TS 101 470 (V1.1.1)

### B.1 General

This annex provides analysis of issues of ETSI TS 101 470 (V1.1.1) [i.2] that are either not in-line with the existing standards of 3GPP and IETF or not covered by the present document.

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### B.2 Analysis of issues regarding ETSI TS 101 470 (V1.1.1)

#### B.2.1 Text related to clause "5.4.2.1 Relay service"

- With regards to the following Note; the need of means for reducing the noise is not studied in the present document.

*"NOTE: Because loud noise from the user site can disturb voice communication between the relay service and the emergency service, some means of temporarily reducing such external noise may help increase communication between the user and the PSAP operator. Further details related to this topic may be considered in an implementation guide."*

#### B.2.2 Text related to clause "5.4.2.1.2 IMS support"

- The requirement; *"Means shall be provided by the serving network provider to invoke relay services in an IMS Total Conversation Emergency service at the interface between the serving network and the ESIInet."*, is not specified in IMS existing specifications. If this were agreed to serve some scenarios Total Conversation for emergency communications as an optional solution, then it would be a gap that needs a technical specification to be developed. The solution is further discussed in clause 7.2.2.3.

#### B.2.3 Text related to clause "5.4.3 Multi-party multi-media call"

- The requirement; *"Means to establish and perform multi-party Total Conversation emergency calls shall be provided by the serving network, in which all the enabled media in the call are shared between the call participants."*, points towards a possible solution, but is not one that is recommended in the present document. This requirement only applies in the case that this solution is followed.

#### B.2.4 Text related to clause "5.4.4 Transfer and Forward supplementary services"

- In regard to the paragraph; *"Transfer and forward supplementary services shall be available for Total Conversation emergency calls for invocation from the PSAP and the application service provider."*, ETSI TR 103 170 [i.3], clause 5.2 describes just one case when it would be desirable for a Total Conversation user to make a call transfer during a Total Conversation emergency session. Others exist, also for the Application Service provider. However, once a Total Conversation emergency session is established then it is recommended to ignore call forwarding and call transfer on the user side for a predefined period of time, see IETF RFC 6881 [i.11], IETF RFC 6443 [i.14], and ETSI TS 122 173 [i.6]. Changes in that policy would need much more preparation than simply stating it in the TS.

#### B.2.5 Text related to clause "5.11.1 Basic SIP support"

- The Note; *"NOTE: It is for further study how a Total Conversation user terminal can prevent PSAP impersonation call-backs."* Is out of scope of the present document as it is general for emergency services and not specific for Total Conversation for Emergency Communications.

## B.2.6 Text related to clause "5.17.1 Basic SIP support"

- The security solution: "For key management, PSAPs and external services should support both DTLS-SRTP according to IETF RFC 5764 [i.50] and ZRTP according to IETF RFC 6189 [i.44]. Total Conversation user terminals may use either of these methods for security", is seemingly unnecessarily needing more than the IMS and IETF solutions where ZRTP is not a requirement (see clause 6.5.7).

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## Annex C: Existing IMS requirements for support of Total Conversation

### C.1 General

This annex captures the main requirements for support of Total Conversation in IMS.

Existing requirements for support of Total Conversation are largely implicit and are spread across several 3GPP specifications. More explicit requirements can be found in three documents in particular: ETSI TS 122 101 [i.9], ETSI TS 122 173 [i.6], and ETSI TS 126 114 [i.7]. In addition, requirements relating to the negotiation of language at session invocation can be found in ETSI TS 122 228 [i.10]. More general architectural requirements for IMS emergency sessions can be found in ETSI TS 123 167 [i.8].

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### C.2 Requirements from ETSI TS 122 101

#### C.2.1 General

The following requirements from ETSI TS 122 101 [i.9] cover the support of Total Conversation:

#### C.2.2 Requirements in clause 7.2 of ETSI TS 122 101

##### "7.2 Multimedia

3GPP specifications shall support development of multimedia services and provide the necessary capabilities.

*Multimedia services combine two or more media components (e.g. voice, audio, data, video, pictures, text) within one call. A multimedia service may involve several parties and connections (different parties may provide different media components) and therefore flexibility is required in order to add and delete both resources and parties.*

*Multimedia services are typically classified as interactive or distribution services.*

*Interactive services are typically subdivided into conversational, messaging and retrieval services:*

*Conversational services are real time (no store and forward), usually bi-directional where low end to end delays (< 100 ms) and a high degree of synchronisation between media components (implying low delay variation) are required. Video telephony and video conferencing are typical conversational services."*

#### C.2.3 Requirements in clause 7.2.4 of ETSI TS 122 101

##### "7.2.4 Real-Time Text Conversation

*Real-Time Text (RTT) conversation is a service enabled in 3GPP networks by the Global Text Telephony (GTT) feature [26].*

- *GTT enables real time, character by character, text conversation to be included in any conversational service, Circuit Switched as well as IP based.*
- *It is possible to use the text component in a session together with other media components, especially video and voice.*
- *Interworking with existing text telephony in PSTN as well as emerging forms of standardised text conversation in all networks is within the scope of this feature.*
- *The text media component can be included initially in the session, or added at any stage during the session.*
- *The text component is intended for human input and reading, and therefore supports human capabilities in text input speed. The character set support is suitable for the languages the users communicate in.*

- *GTT specifies limited interoperation with Multimedia Messaging Services including a possibility to divert to messaging in case of call failure and sharing user interface equipment and external UE interfaces."*

## C.2.4 Requirements in clause 10.4 of ETSI TS 122 101

### "10.4 Emergency calls in the IM CN subsystem

#### 10.4.1 General

*The IM CN subsystem shall support IMS emergency calls. It shall be possible to set up emergency calls initiated by an emergency call number.*

*If a UE supports IMS Multimedia Telephony service with speech media as specified in TS 22.173 [40] via an access network, then it shall also support IMS emergency calls via that access network.*

*Subject to the regulatory requirement, the IM CN subsystem shall be able to unambiguously identify each emergency service defined in the national numbering plan for the country in which the UE is located.*

*In accordance with national regulations for where the subscriber is located, if the UE does not recognize a dialled number as an emergency call number but the IM CN where the subscriber is located does recognise the dialled number as an emergency call number (e.g. a number used in the local emergency numbering plan) then the call shall be routed as an emergency call indicating the type of emergency service to the correct PSAP. Subject to operator setting the call may be prioritized.*

*Note 1: The above does not preclude the network rejecting the call and requesting the UE to setup a new emergency call to the same emergency service.*

*Emergency calls may be initiated using a private numbering plan [49].*

*Note 2: There can be an overlap between the private numbering plan of a hosted enterprise and the public numbering plan, which makes translation of emergency numbers necessary.*

*Emergency calls may be initiated by a service when requested by the user.*

*Note 3: It is not intended to enable automatic setup of emergency calls.*

*Note 4: Only speech and GTT-IP [47] media are supported, when required per subclause 10.1, for emergency services towards a CS PSAP.*

*An emergency call shall take precedence over any other services a UE may be engaged in, if required by local regulation.*

*Emergency calls from an unauthenticated UE (as far as the IM CN is concerned) shall be supported by the IM CN subsystem, if required by local regulation.*

*Subject to regulatory requirements, when UEs must be authenticated, both the network and the UE shall support the same authentication and security methods that are used for non-emergency sessions.*

#### 10.4.2 IMS Multimedia Emergency Sessions

##### 10.4.2.1 General

*For IMS emergency calls towards IP PSAPs, other media types may be supported by the UE and the IMS, subject to regulatory requirements.*

*The media types that may be supported during an IMS MES include:*

- *Real time video (simplex, full duplex), synchronized with speech if present;*
- *Session mode text-based instant messaging;*
- *File transfer;*

- Video clip sharing, picture sharing, audio clip sharing;
- Voice; and
- Real-Time Text.

*Note 1: An IMS MES need not contain voice or Real-Time Text.*

*To avoid interworking issues, a UE and IMS that supports text based instant messaging shall support a common session mode text-based instant messaging protocol.*

*IMS MES does not include support for legacy store and forward messaging such as the Short Messaging Service (SMS).*

*Calls from non-human associated devices (e.g. fire alarms) are outside the scope of this specification.*

*Adding, removing and modifying individual media to/from an IMS MES shall be supported.*

*An IMS MES is not a subscription service. A UE capable of IMS emergency calls and capable of supporting the other media types should also be able to support initiation of an IMSMES.*

*An IMS MES from an unauthenticated UE (as far as the IM CN is concerned) shall be supported by the IM CN subsystem, if required by local regulation.*

*IMS MES shall be supported by UEs that are subject to service restrictions, e.g. for UEs camping on a cell in a forbidden PLMN or in a forbidden LA (see 3GPP TS 22.011 [11]), or on a CSG cell without the subscriber being a member of that CSG (see 3GPP TS 22.220 [48]). Such IMS MES shall be accepted by the network if required by local regulation.*

*An IMS MES shall support providing the location of the UE, in a manner similar to IMS emergency voice calls.*

*An IMS UE that supports IMS MES shall identify an emergency number dialled by the end user as a valid emergency number utilizing the same mechanisms as used for IMS emergency voice calls as defined in subclause 10.1.1.*

*Note 2: This capability supports the general public, including facilitating emergency communications by individuals with disabilities (e.g. persons who are deaf, deaf-blind, hard of hearing, or have a speech disability).*

*An originating network and UE may support some or all of these other media types, and support of any specific media by an originating network or UE may be subject to regulatory requirements.*

*Voice call continuity per clause 21 shall be supported when a UE with an active IMS MES with voice and other media moves out of IMS voice coverage and voice call continuity is supported by the UE and network. The remaining media (i.e. voice call) then becomes a CS emergency call e.g. TS12 call for 3GPP systems as defined in 3GPP TS 22.003 [14].*

*Other media shall be dropped when a UE with an active IMS MES moves out of IMS voice coverage, irrespective of whether or not there is an active voice session."*

## C.3 Requirements from ETSI TS 122 173

The following requirements from ETSI TS 122 173 [i.6] cover the support of Total Conversation:

### "4.2 Default media handling capabilities of IMS Multimedia Telephony service

*IMS Multimedia Telephony can support many different types of media.*

*IMS Multimedia Telephony service includes the following standardized media capabilities:*

- Full duplex speech;
- Real time video (simplex, full duplex), synchronized with speech if present;
- Real-Time Text communication

- *File transfer;*
- *Video clip sharing, picture sharing, audio clip sharing. Transferred files may be displayed/replayed on receiving terminal for specified file formats*
- *Fax;*
- *Data (CS).*

*The support of each of these media capabilities is optional for a UE.*

*At least one common standardized format (e.g. JPEG, AMR) shall be supported per media type.*

*Note: IMS Multimedia Telephony service fulfils the service requirement for the Total Conversation in Recommendation ITU-T F 703 [11].*

*The IMS Multimedia Telephony service shall support the following handling of media*

- *Adding, removing and modifying individual media to/from an IMS Multimedia Telephony communication"*

## C.4 Requirements from ETSI TS 126 114

ETSI TS 126 114 [i.7] specifies a client for the Multimedia Telephony Service for IMS (MTSI) supporting conversational speech (including DTMF), video and text transported over RTP. As such, the entire specification is relevant to the support of Total Conversation.

## C.5 Requirements from ETSI TS 122 228

The following requirements from ETSI TS 122 228 [i.10] cover the negotiation of language at the time of session establishment:

### "7.5.2 Negotiation at IM session invocation

*It shall be possible for the capability negotiation to take place at the time of the IP multimedia session invocation. Refer to clause 7.3 for further details on capability negotiation on IP multimedia session invocation.*

*A UE should support negotiation of the user's desired language(s) (as defined in IANA [29]) and modalities for spoken, signed and written languages.*

*The system should be able to negotiate the user's desired language(s) and modalities, per media stream and/or session, in order of preference.*

*A service provider shall be able to pass language and modality information between the endpoints. With respect to the language and modality information, there are no other service provider actions required."*

## C.6 Requirements from ETSI TS 123 167

The following architectural requirements from ETSI TS 123 167 [i.8] cover the support of Total Conversation media during an emergency session:

### "4.1 Architectural Principles

...

12. *The architecture shall enable emergency centres and PSAPs to request a PSAP call back to a UE with which the Emergency centres or PSAPs had an emergency session. The serving network of the UE shall use the appropriate call termination procedures e.g. IMS if the UE is available for voice over PS, or ICS if the user is available over CS. PSAP call back is subject to local regulation.*

*NOTE 2: PSAP call back sessions are treated as normal calls.*



*NOTE 3: Subject to local regulation, any supported media can be used during a call back attempt from a PSAP.*

...

27. *When a call is established with a PSAP that supports voice and other media, voice, GTT and other media according to TS 22.101 [8] (e.g. video, session mode text-based instant messaging) can be used during an IMS emergency session if required by local regulation. This media may be used in addition to or instead of voice and/or GTT.*

... / ...

## 4.5 Media

- *When the call is established with a PSAP that supports voice only, voice and subject to local regulation, GTT media is allowed during the IMS emergency session.*
- *When the call is established with a PSAP that supports voice and other media, subject to UE and network support for the other media and local regulation, voice, GTT and other media according to TS 22.101 [8] can be used during the IMS emergency session.*
- *For sessions with a PSAP that supports voice and other media, media can be added, modified or removed during the IMS emergency session (e.g. adding video to a voice call) per media negotiation in TS 23.228 [1].*
- *When a PSAP that supports voice and other media attempts to add media, the media shall be added if "accepted by the UE."*

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## Annex D: Current situation in Europe for Total Conversation and emergency service access for persons with disabilities

### D.1 General

This annex provides a brief overview over the implementation situation for Total Conversation, and for emergency service access for persons with disabilities in Europe. The information is partially based on a report from BEREC: (15)135 [i.56] "Update of the report on equivalent access and choice for disabled end-users". Information is also retrieved from generally available information in the autumn of 2015.

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### D.2 Total Conversation, RTT and subsets for everyday use

Implementation of standards based Total Conversation for everyday calls is currently only known from the area of accessibility for persons with disabilities. In that area, implementations of standards based Total Conversation exist in France, The Netherlands, Norway, Sweden and UK. They are all IETF SIP based. Implementations of Total Conversation with non-standard implementation of the real-time text medium is known to exist in Belgium, Germany and Finland.

Standard IETF SIP based RTT services are known to exist in Sweden. A proprietary system for RTT by IP and audio by circuit switching has been recently introduced in UK.

Proprietary OTT systems for calling with video, audio and text messaging together are in widespread use by persons with disabilities as well as the general population in most countries in Europe.

The reason to use real-time text in Total Conversation is because it provides more direct communication and a greater sense of contact between calling and called parties than instant messaging. It is therefore preferred in conversational situations. This is reported in the report "Real-time text interoperability; status and field trial" [i.57].

Videophone systems in use by persons with disabilities are known to have been deployed in Switzerland and Spain.

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### D.3 Relay service availability

Relay services are available in a few countries in Europe. Of the 25 respondents to this question in the survey of BEREC [i.56], 14 countries have text relay services, 10 have a video relay service for sign language users, one country has a speech-to-speech relay service, and 10 countries responded that they had no available relay services. There are 37 member countries in BEREC, so there may be some provision of relay services European countries that does not show up in the survey [i.56].

Of the available relay services, only those in four countries are known to use the standards for IETF SIP based Total Conversation.

Many of the relay services, especially the video relay services have limited opening hours.

An earlier survey of relay service availability from 2009 can be found in ETSI TR 102 974 [i.58].

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### D.4 Emergency service access

Only a very low number of European countries provide direct access to emergency services by any form of real-time text. The Netherlands is the only country known to provide it for terminals using IETF SIP and the standards for real-time text according to the present document. A few other countries provide direct access for PSTN text telephones or non-standard forms of real-time text or text messages.

Most relay services are prepared to provide communication with emergency services. Note though that this implies that the same limited opening hours for the relay services also apply for emergency service access by their support.

A number of European projects are, and have been, working on the topic of IP based access to emergency services including Total Conversation. REACH112 [i.59] had pilot implementations in five countries in 2012, using the IETF SIP protocols considered in the present document. NEXES [i.60], EMYNOS [i.61] and ETSI NG112 Emergency Communications Plugtest [i.62] are three current projects with EC funding that have plans to use the standards considered in the present document for Total Conversation access by IETF SIP and possibly also IMS.

Also in USA the current access to emergency services are mainly through relay services. The US form of PSTN text telephony, called TTY, has direct access to 9-1-1, while IP based forms of text and video communication go through relay services.

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## D.5 Conclusion

Even if the EU Universal Service Directive [i.24], Article 26.4 already from 2009 includes the requirement for harmonized and equivalent access to emergency services for persons with disabilities, there is very limited availability of such services in reality.

The availability of specifications and standardized protocols suitable for access to emergency services for persons with disabilities integrated with the mainstream protocols, form a basis for the eventual fulfilment of the requirements in ETSI TS 101 470 [i.2]. Completed and ongoing European projects in this area provide the most apparent indication that the development in this area is heading in the right direction. The present document provides guidance in that process.

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## History

<b>Document history</b>		
V1.1.1	March 2016	Publication