REACH112

REsponding to **A**ll **C**itizens Needing **H**elp

D3.2 REACH112 Platform Specification	
Project Acronym	REACH112
Grant Agreement Number	238940
Project Title	REACH112 REsponding to All Citizens Needing Help
Deliverable number	D3.2
Deliverable title	REACH112 Platform Specification
Revision	Final version
Due date of deliverable	31 March 2012
Actual submission date of the deliverable	28 October 2011
Start date of project	01 July 2009
Duration of the project	36 months
Work Package	WP3
Task	T3.3
Leader for this deliverable	AuPix Limited
Other contributing partners	RNID, IVES, FT, SIS, NOKIA, IES, OMNITOR
Author	Gunnar Hellström (Omnitor), John Martin (AuPix), Emmanuel Buu (Ives), Massimo Cristaldi (IES), Regis Nevo (FT-Orange), James Hamlin (AuPix)
Deliverable reviewer	EENA
Deliverable abstract	Specification of interoperation and calling methods in a total conversation network and for access to emergency services.

Project co-funded by the ICT Policy Support Programme		
DISSEMINATION LEVEL		
PU	**Public**	**X**

Revision History

Revision	Date	Description
2.0.02	28 October 2011	First public release.

PROPRIETARY RIGHTS STATEMENT

This is public information from the REACH112 consortium. The consortium has copyright on the contents. The document and its contents may be used freely. This restriction legend shall not be altered or obliterated on or from this document.

STATEMENT OF ORIGINALITY

This deliverable contains original unpublished work except where clearly indicated otherwise. Acknowledgement of previously published material and of the work of others has been made through appropriate citation, quotation or both.

Executive Summary

The REACH112 project, supported by the European Commission, aims to radically improve access to emergency services for all, as well as person-to-person communication, with a particular focus on people with disabilities. To demonstrate the advances in telecommunications required, the project has piloted services in five European countries. The project consortium here presents the specification for such services.

We discuss the user requirements for multimedia telecommunications including the benefits of real-time-text and relay interpreting services. We specify a complete service including user account management, peer to peer calling as well as access to emergency services. For the benefit of potential service providers, we describe a complete architecture based on Session Initiation Protocol (SIP) which we mandate for interworking between REACH112 services. A guide for service providers describes the roles that different organisations and providers will have in a service. We also describe the approach to data sharing based on the non-proprietary standards Common Alerting Protocol (CAP).

Definitions and Abbreviations

Common Alerting Protocol (CAP)	An XML based protocol for transferring information about emergencies and disasters.
E.164 NUmber Mapping (ENUM)	A method of mapping ITU-T E.164 telephone numbers to URIs using DNS. There is a public ENUM tree in which telephone numbers are given DNS names.
e164.arpa	The DNS postfix to public ENUM telephone number entries.
Internet Engineering Task Force (IETF)	An organisation whose mission is to make the Internet work better by producing high quality, relevant technical documents that influence the way people design, use, and manage the Internet.
Public Safety Answering Point (PSAP)	A PSAP is a facility where emergency calls are received under the responsibility of a public authority. This terminology is used by both the European Telecommunications Standards Institute (ETSI) (ETSI SR 002 180 [1]) and the National Emergency Number Association (NENA). In the United Kingdom, PSAPs are called Operator Assistance Centres; in New Zealand, Communications Centres. Within this document, it is assumed, that PSAPs may or may not support the receipt of emergency calls over IP.
Public Switched Telephone Network (PSTN)	The network of the world's public circuit-switched telephone networks.
Real Time Text (RTT)	The transport of text characters between two points on a network in real time. That is, the impression by the receiver is that the characters flow as they were typed.
Relay Service	A service that allows a user to call another person or service and get support in the call by an interpreter. For instance the interpreter may translate between sign language and spoken language or between real-time-text and spoken-language.
Total Conversation (TC)	A multimedia service offering real-time conversation in video, real-time text and voice according to interoperable standards. All media streams flow in real time.
Uniform Resource Identifier (URI)	A string of characters used to identify a name or a resource on the Internet. Such identification enables interaction with representations of the resource over a network (typically the World Wide Web) using specific protocols. Schemes specifying a concrete syntax and associated protocols define each URI.

Table of Contents

1 Introduction

While voice telecommunication has had massive benefits to society as a whole, it has created barriers to communication for some, especially deaf, hard of hearing and speech impaired people. Video communication can mitigate these barriers for sign language users. For sign language users to be able to call non sign language users, an interpreting relay service is required. Real-time-text (RTT) in which text characters are displayed as they are typed, has been used for many years by deaf, hard of hearing and speech impaired people. Again relay services are available in some countries to allow a real-time-text user to call a voice phone user.

The core concept that underlies REACH112 and the services that it provides is, Total Conversation (TC). Total conversation was probably best described by the Internet Engineering Task Force (IETF) as follows:

> *A multimedia service offering real-time conversation in video, real-time text and voice according to interoperable standards. All media streams flow in real time.*

So total conversation always enables its users to use voice, video and real time text in any conversation. Users may however choose not to use all three modes by turning some of them off at the start of a call, or by ceasing to send one or more media during the call.

Where there are language or communication barriers, for example between a Deaf sign language user and a hearing user who doesn't use sign language, a relay interpreter may be necessary. Note that a sign language user may be able to sign many times faster than they could type. Other situations where a relay service may be required include calls, from deaf people, to voice only systems such as Public Switched Telephone Network (PSTN) telephones. Section 3.1 has more information about where relay services may be required.

The core functionality of a REACH112 service is therefore to facilitate total conversation calls:-

- Between its own service users
- Between users and the emergency services
- Between its users and users of other REACH112 services
- Between its users and certain other networks like the PSTN (Section 3.1 lists these)

To facilitate these calls, relay services will be required in some cases.

REACH112 defines an architecture of interconnected REACH112 services (section 2). The interconnections between REACH112 services are strongly defined, based on the Session Initiation Protocol (SIP) [2]. We have deliberately defined a REACH112 service as including and having control over:-

- its own user's terminal equipment and network access.
- its interface with the emergency services in its region
- its interface with relay services.

This means that a REACH112 service has great flexibility to deal with local circumstances such as emergency services and relay services that might only be accessible via the PSTN. However services are expected to use SIP according to this specification, within their own network, so far as reasonably practicable.

Section 3 covers functionality of a REACH112 service, from a user's point of view, to provide the reader with background and understanding. We do not attempt here to document the process of design from user requirements to the specification.

2 The Logical Model of REACH112

As described in the introduction, REACH112 defines an architecture of interconnected REACH112 services. In this document, the term *REACH112 service* refers to a service that manages users and provides them with total conversation and access to the emergency services as described in section 3. This is illustrated in Figure 1. This definition does not require a REACH112 service to be provided by only one legal entity. The owner of a REACH112 service is the legal entity that has a contract with users to provide a total conversation service. It is envisioned that user's contracts with their REACH112 service will include the

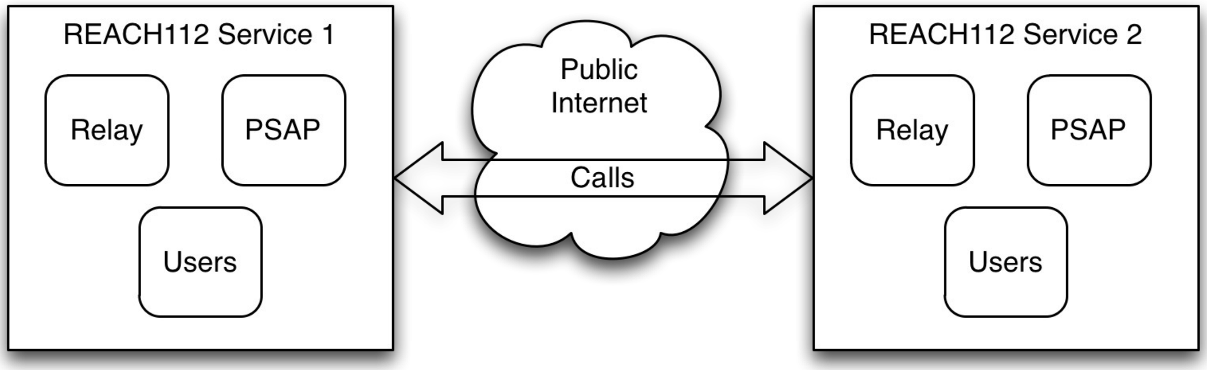

Figure 1: Simplified Architecture – Two REACH112 Services Providing Relay and Emergency Access for their Users

whole service including access to relay services even if the REACH112 service owner arranges access to a relay service provided by another party. This model does not preclude users from using relay services from outside their own REACH112 service. However it is intended that a service provider would take the responsibility for providing users with any relay service as may be necessary to provide for their communication needs.

A REACH112 service is likely to cover a whole country or administrative region but there is no requirement about the way territory is divided up between REACH112 services and there is no restriction on REACH112 regions overlapping. Where access to emergency or relay services is shared between two or more REACH112 services, it becomes much more important for tight standardisation of those service interfaces.

This document provides specification for all the interfaces shown in Figure 2. These are, Inter-service, Relay, User and PSAP. The core protocol for REACH112 is SIP [2] according to section 4 of this document.

The proscriptions in section 4 are mandatory for inter-service links and strongly recommended for relay, user and PSAP links. Sections 5 and 6 provide extra information about relay service invocation and providing access to emergency services respectively.

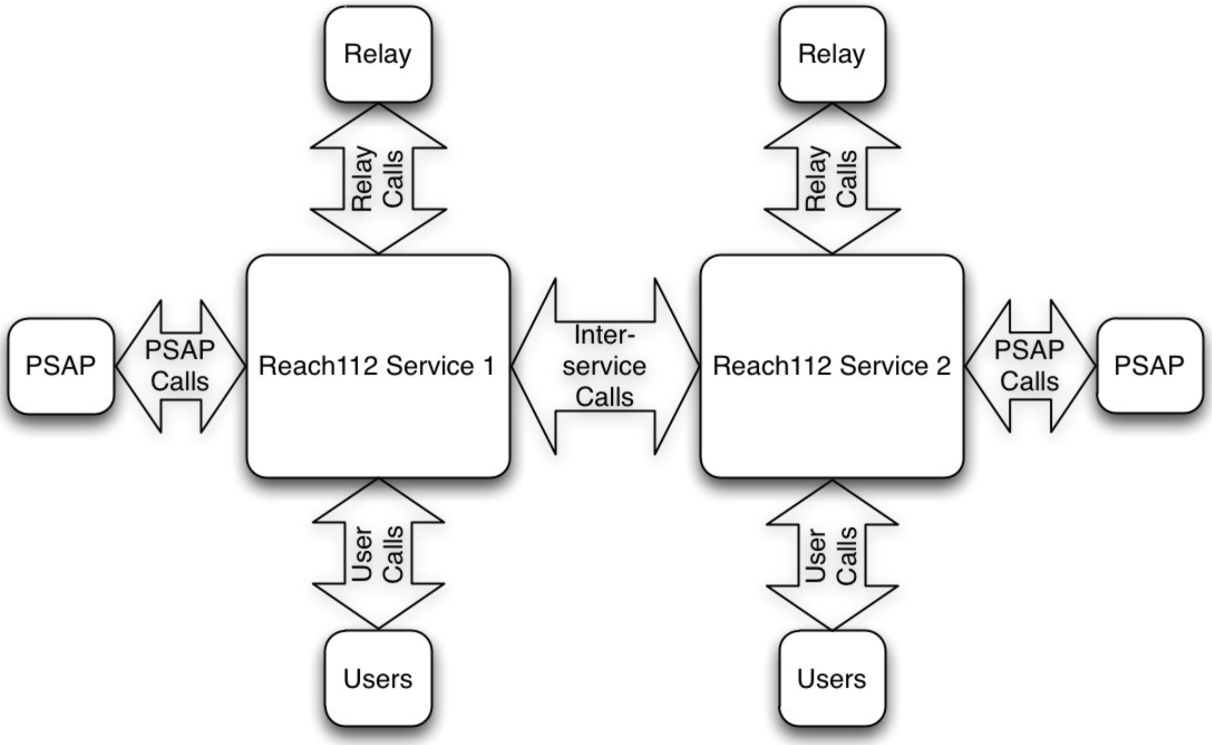

Figure 2: Two REACH112 Services Using Only Standard Total Conversation Interfaces

Usually the underlying network between services will be the public Internet. However services may route calls over private Internet Protocol links by mutual agreement.

Most REACH112 networks will have an interconnection with the PSTN to allow their users to communicate with PSTN based telephones and in some cases textphones. Two users on different REACH112 services would communicate via the main IP based interconnection rather than via the PSTN. Figure 1 deliberately omits PSTN interconnections for simplicity but they should be regarded as local to each REACH112 service.

3 Functionality Provided to Users

As mentioned in the introduction, this section aims to give background information to service providers about the service to users but we do not document here the process of development of the platform specification from user requirements.

At a minimum, a REACH112 service must provide its users with:-

- Total conversation communication between its own service users
- Communication between its users and the emergency services
- Total conversation communication between its users and other REACH112 services
- Access to a voice/video/text mail system to take messages when they cannot answer calls

While this bare minimum is useful to many people, it is accepted that such a service would be completely unsatisfactory for other users. In particular users who cannot use voice to communicate. A Reach112 service without interconnection with the PSTN would not be of use to many users. Legislators and regulators are strongly urged to ensure that REACH112 services are provided with a much higher level of service as follows:-

- Access to emergency services in sign language
- Access to emergency services in real-time-text and voice
- Provision of sign relay for non-signing and signing users to communicate
- Access to voice telephony in voice
- Access to voice telephony by sign language and text via a sign relay service
- Access to voice telephony by text via a text relay service
- Access to PSTN based text telephony in text and voice where applicable

These features should be available to all citizens who might reasonable need them, 24 hours a day all year round. Assistive features like relay services should be free at the point of use and call charges should be no more than equivalent voice telephony charges.

As the access network, between the core of a REACH112 network and its user's terminal may not be under the control of the REACH112 network, intra and inter service links should be designed so as to provide where possible a good quality call experience. A user should be able to communicate in their preferred mode or modes of communication comfortably for a minimum of 20 minutes.

3.1 Call Cases

The range of call cases supported by a REACH112 service, includes: calls between its own users; and calls between its users and other networks and services. Table 1 describes some of the networks with which a REACH112 service might interconnect.

Service	Description	Media
Voice-only emergency services	Traditionally emergency services are available only in voice and are usually available over the PSTN.	Voice
Total conversation emergency services	Although traditionally emergency services are available only by voice, REACH112 aims for them to ultimately be available using total conversation. This means that the call takers for all emergency services would have total conversation terminals.	Voice/video/text
Public Switched Telephone Network (PSTN)	Principally a voice network but textphones use modems to carry text over the voice channel. The PSTN connects the vast majority of the world's telecommunications users at present.	Voice Text in some cases
3G	3G networks mainly carry voice but there are many 3G videophones.	Voice/Video
H.323	IP based protocol used by many videophones	Voice/Video
Internet based RTT	There are a few Internet based Real Time Text (RTT) systems that allow users to use real time text but without voice and video.	Text
Internet based instant messaging	Instant messaging (IM) allows users to use text to communicate but in message chunks rather than character-by-character. In many cases, instant messaging systems have been complimented with video and voice.	Text messaging (but not RTT) and often real-time video/audio

Table 1: Services with which REACH112 Services might Interoperate

Table 2 lists the range of relay services that might be provided. Note that users benefit from each media type to different degrees but overall the provision of total conversation allows for the best possible communication across the whole group. For more information on relay services see ETSI ES 202 975 Harmonized Relay Services [3].

Relay	Description
Text	Uses human operators who speak what a deaf person types, to a hearing person and type back what the hearing person says back to the deaf person. When provided over the PSTN, the deaf person must use a textphone. Most textphones do not fully support simultaneous voice and text but it is generally possible to switch between voice and text so the deaf person can speak to the hearing person directly. This is often referred to as voice carry over. In total conversation, simultaneous voice and text is possible, which means voice carry over can be very effective when relaying between total conversation and a voice phone on the PSTN. In most cases, there would be little need for text to speech relay where both ends of the call have total conversation as both sides could type. Exceptions would be were either side was unable to type or read.
Captioning	This is similar to text relay except that there is a rapid way of creating text from voice from the hearing person's side, and audio is always transmitted through the service between the two call participants.
Sign	Sign relay uses human operators to translate between sign language and speech. Real-time text should be supported as a side channel for clarifications and items requiring exact spelling. Voice carry over is also possible with sign relay. The speed of communication, in sign, is roughly the same as that of spoken language and so sign relay is generally much faster than text relay. Sign languages vary around the world and it is important to note that they are not visual versions of spoken languages. They have different grammar and sentence structure and so translation, between sign and spoken language, is not a word-for-word process. For many Deaf people, sign language is their first language and so they will be much more comfortable using sign language than using text.

Table 2: Types of Relay

3.2 Relay Services

The type of relay services that are made available by a REACH112 service, to its users will vary, depending largely on the legislation and policy support in the respective region. The basic ways to invoke relay services are described here. It is at the discretion of the REACH112 service provider to provide more advanced variations. Users may require a relay service in any call, not just calls to PSTN voice telephones. For example a sign language user may use an interpreter in a call to a hearing total conversation user. Total conversation in this case has the advantage that the hearing user can see the conversation between the sign language user and the interpreter and will understand why they are having to wait.

Relay services may be invoked in the following ways:-

1. At the request of the calling user
2. At the request of the called user
3. According to the pre-set preferences of the calling user
4. According to the pre-set preferences of the called user
5. According to the media types supported by each side of the call

6. Some combination of 1 to 5

In cases 1 and 2 above providing invocation of relay services on manual request, the user may activate relay by pressing a button or some other control on their user interface or they may alter the number or URI that they dial. In general it is better to allow the user to use the normal address or number of the person they wish to call and to separately specify that they want a relay service but in some areas, users may already have used a relay service with a particular dial prefix and the service provider may wish to allow the use of this prefix to continue. In cases 3 and 4 providing invocation of relay services controlled by registered user preferences, the user might have specified a particular relay service in the configuration of their terminal software or it might be a setting managed centrally as part of their user account.

A typical example of case 5, where an automatic decision to invoke relay service is based on the media supported by the terminals in the call, would be a call from a total conversation terminal with the voice channel disabled, to a PSTN phone number. As there is no common media capability, a relay must be required. The relay service invoked could be a text only service or a combined text and sign language relay service and this choice could be made based on service availability and user preference.

Case 6 is any combination of choice factors. It is for the service provider to provide the best service to users, in the local circumstances. If the service provider has created its own user terminal systems, then it will be able to add buttons and settings to allow the user to control relay invocation whereas standard terminals may require relay to be invoked with a number prefix.

Whichever method or combination of methods is used to select relay invocation the invocation of relay must be clear to at least the party on whose part the relay is being invoked. So if a caller has a preferred relay service configured in their account preferences, then it should be clear to the caller that a relay service is being invoked when they make a call. As mentioned in the user profile section of IETF RFC 3351 [4], a deaf person may not wish the party they are calling to know that they are deaf. This may be to avoid potential discrimination by the party they are calling. So far as reasonably practicable, REACH112 services should respect and support a user's desire to keep their disabilities confidential. This may require that for some calls, the other party is not made aware that a relay service is being used.

3.3 Emergency Services Access

The European Universal Service Directive 2002-22-EC, amended in 2009 [5], requires that all end-users of communication services shall be provided means to call 112 to reach emergency services. Therefore REACH112 users should be able to use the 112 dial string to access emergency services.

Wherever possible, the calling user's location must be passed to the emergency services. Section 6.8 discusses methods for achieving this.

In the long term, it is envisioned that all emergency services call takers will have total conversation terminals. There is great value in having video in an emergency call, to help the call taker to assess the emergency situation and in helping to calm the caller. Text communication can also be useful in confirming detailed information. These are advantages for all users, not only people with disabilities.

In the short and medium term, variations in handling of the emergency calls makes the user experience different in different REACH112 services. Emergency access may be provided through relay operators to PSTN or total conversation emergency operators and emergency services may themselves provide signing call takers. Section 6 provides both background on existing emergency services and models for providing REACH112 emergency access.

As calling emergency services with different media and relay services may be more complicated than existing voice service, from a user's point of view, it is important that:-

- It is clear to them how to access emergency services using their REACH112 service; and
- It is clear how to interact with the emergency services. For example it must be clear to a user if they are expected to use text.

3.4 Call Queueing Requirements

Where a user initiates a call that requires a human operator to help complete the call (for instance within a relay or emergency call centre), a REACH112 service shall provide a queueing system. The queueing system will provide the user with feedback about the availability of an agent and allow the platform to select the best agent based on information such as the user's profile and other pertinent information. As a minimum functionality:

- The queueing function should be able to play audio / video and text messages to inform the user that they are queued and optionally provide their position in the queue;
- An agent shall be able to be associated with one or more queues and a priority level should be associated to each queue and/or agent;
- Agents shall be able to log-on or log-off from a queue and to indicate that they are temporarily unavailable; and
- The queue mechanisms should be able to limit the number of simultaneous calls waiting to be answered and set a maximum time before a call should be answered.

4 Call Signalling

As described in section 2, REACH112 makes a clear distinction between signalling within a REACH112 service and signalling between different REACH112 services. The following is to be considered mandatory between services and should be used so far as reasonably practicable inside REACH112 services. Sections 4.4 and 4.6 have specific information about emergency service, intra-REACH112 service and inter-REACH112 service signalling respectively.

4.1 SIP Signalling

Call signalling should use Session Initiation Protocol (SIP) as defined in [2]. SIP can also be used in IP Multimedia Subsystem IMS networks, with some modifications as described in ETSI 122 173 Multimedia Telephony Stage 1 [6].

REACH112 adds some extra specifications in addition to those in [2] which are described in the next paragraphs. Section 4.5 outlines circumstances where these network requirements may not be applied. For links between REACH112 services, these requirements and those in the inter service requirements section must be rigorously applied.

All SIP proxies must precisely conform to the loose routing function as described in [2] (Page 21). Proxies must insert a Record-Route header that ensures subsequent transactions will flow through them. The URI's must contain the "lr" parameter and the domain part of the URI must be an IP address, not a host name. Proxies must support the use of pre-set routes. When forwarding a packet according to a route set, proxies must leave the request URI unchanged.

REACH112 requires the SIP method functionality in Table 3 in SIP interfaces.

INVITE	Mandatory in order to initiate a call. This request must be constructed and handled in accordance with RFC 3261 [2].
RE-INVITE	This request must be constructed and handled in accordance with RFC 3261 [2].
CANCEL	Mandatory in order to Cancel a call before its completion. This request must be constructed and handled in accordance with RFC 3261 [2].
ACK	Mandatory in order to acknowledge completion of a request. This request must be constructed and handled in accordance with RFC 3261 [2].
BYE	Mandatory in order to terminate a session to terminate a call. This request must be constructed and handled in accordance with RFC 3261 [2].
OPTIONS	Must be supported in order to determine the capabilities of a user agent server or to determine the basic state of a user agent server. Option messages may be part of a SIP dialogue or outside any SIP dialogue. This request must be constructed and handled in accordance with RFC 3261 [2].

Table 3: SIP Methods and Method usage

Table 4 gives the SIP Headers that are mandated for a REACH112 service.

Header name	***Reference***	***Request***	***Response***
Call-ID	*[2]*	*Mandatory*	*Mandatory*
Contact	*[2]*	*Mandatory*	*Mandatory in 2XX*
Content-Type	*[2]*	*Mandatory if the body is not empty*	*Mandatory if the body is not empty*
CSeq	*[2]*	*Mandatory*	*Mandatory*
From	*[2]*	*Mandatory*	*Mandatory*
Max-Forwards	*[2]*	*Mandatory*	
To	*[2]*	*Mandatory*	*Mandatory*
Via	*[2]*	*Mandatory*	*Mandatory*
P-Asserted-Identity	*[2]*	*Mandatory in Invite*	
Geolocation	*[7]*	*Mandatory in Invite for emergency calls*	
Route	*[2]*	*Supported*	
Record-	*[2]*	*Supported*	*supported*

Header name	Reference	Request	Response
Route			
Allow	[2]		Mandatory in 405
Unsupported	[2]		Mandatory in 420

Table 4: SIP Headers

4.2 Call Media Signalling

Session Description Protocol (SDP) must be used to negotiate call media sessions. The offer-answer exchange shall be performed according to RFC3261 [2], RFC3264 [8] and RFC4566 [9]. SDP may only be used in the body of SIP INVITE, re-INVITE, UPDATE, ACK, 200 OK (INVITE, re-INVITE) and 18x (INVITE) messages.

In an SDP offer, codecs shall be listed under their respective media stream "m=" lines, in order of preference, the first codec format listed being the preferred one.

The "a=ptime" is a media attribute which indicates the desired packetization interval that the end point would like to consider in reception for a specific audio media stream (but not for a specific codec). The "ptime" parameter is recommended where the information is available. The recommended packetisation times for audio codecs are described in section 4.3.

If there are no media formats in common in the SDP offer received in an initial INVITE or re-INVITE, it shall be rejected by a 488 "Not acceptable here" response with a warning header code "304 Media type not available" or "305 Incompatible media format".

4.3 Media Transport and Codecs

The transport of media must use the Real Time Protocol as defined by RFC 3550 [10].

The media used are video, real-time text and audio. The following characteristics apply for the media in the interoperability interface.

4.3.1 Voice

When audio is used, ITU-T G.711 A-law audio shall be among the supported coding forms. The recommended packetization delay for G.711 is 20ms. It is advised to support also G.729 - with and without the annexe A - with packetization intervals of 20 ms and 40 ms.

4.3.2 Video

ITU-T H.263 video decoding shall be supported for resolutions of at least QCIF and CIF at frame rates of at least 20 fps for QCIF and 20 fps for CIF. Encoding must be at resolutions of at least QCIF and frame rates of at least 20 fps.

It is strongly recommended that ITU-T H.264 video decoding be supported for resolutions of at least QCIF and QVGA at frame rates of at least 20 fps. Where used, encoding must be at a resolution of at least QCIF and frame rates must be at least 20 fps for QCIF resolution and 20 fps for QVGA resolution.

For H.263, the packetisation and SDP negotiation shall be done as specified in RFC 4629 [11]. For H.264, the packetisation and SDP negotiation shall be done as specified in RFC 3984 [12]. Full Intra Requests (FIRs) shall be sent when needed and are facilitated by way of RFC 5168 [13].

4.3.3 Real-Time Text

Real-time-text communication shall be coded and presented according to ITU-T T.140. Packetisation and SDP negotiation shall be performed as specified in RFC 4103 [14]. The IETF document draft-hellstrom-textpreview [15] contains advice on best practises for real-time text.

The strong recommendations of 300 ms transmission interval and 2 generations redundancy are mandatory in a REACH112 service. SIP clients must maintain at least a 2K character buffer for erasure and scrolling.

Note that two errata are available in IETF for RFC 4103 [14]: ordering of payload types in m=text lines should be RED first and T140 second as in the second example of RFC 4103 section 7.2; the second picture of the packet with redundancy in 7.1 should indicate that the contents of redundant data is R2, and the text should say that R1 is empty.

Text must be encoded in UTF-8 and each participant in a session must transmit the UTF-8 Byte order mark (BOM) (\UEFBBBF) at the start of a media session. Session keep-alive should be applied as described in IETF RFC 6263 [16] although regularly transmitting the UTF-8 BOM character is an alternative keep-alive transmission. BOM must not be interpreted as user-generated text.

On reception, any character codes that cannot be presented shall be ignored. This may be because of graphical rendering limitations or for special signalling character codes such as SOS-ST strings specified in ITU-T T.140. When deleting, each backspace character should delete exactly one Unicode character, regardless of the byte-length of that character in UTF-8. The only exception to this is that CRLF combinations sent by non-compliant systems, instead of the the proper Unicode line-end character, should be treated as a single character when deleting.

4.4 Emergency Call Signalling

Where the emergency services are accessed in total conversation, call signalling should be implemented according to the previous parts of this section. It is important that the URI or number presented to the emergency services can be used to call the user back.

The network to user interface used to connect the PSAP to the total conversation platform will be subject to standardization at European level and national authorities may add additional technical or functional requirement. In that respect, it it therefore strongly recommended to use the SIP and RTP protocols as suggested at the start of section 4 Call Signalling and 4.2 Call Media Signalling.
Where available caller location information should be conveyed for emergency calls, according to draft-ietf-ecrit-phonebcp [17].

4.5 Intra-service Call Signalling

To provide the required range of total conversation services to users in a flexible way, REACH112 mandates the use of Session Initiation Protocol (SIP) as the core call signalling protocol as described in the preceding sections. However inside a REACH112 service, there may be reasons to ignore some requirements or to put other protocols on the call path in some cases. Examples include: linking to the PSTN; linking to proprietary networks based on IP; providing service on very unreliable IP networks where users may benefit from more limited functionality over say HTTP; using a third party relay service that has an old or poorly implemented SIP interface. Gateways should separate a REACH112 service's core SIP network from these proprietary or non-compliant networks.

4.6 Inter-service Call Signalling

To ensure interoperability between REACH112 networks, the interfaces between them must comply with the requirements in the preceding sections. If a REACH112 service introduces a relay service then this must be done transparently to the other REACH112 service in the call. This may mean mixing media streams from the relay and the caller, so that the call recipient on another REACH112 service will see the relay operator and the caller. This mixing must be done on one of the services so that the other network does not see any renegotiation of call media.

4.6.1 Inter-service Call Routing

ENUM will be used for inter-service telephone number discovery. Network operators must publish their user's telephone numbers to an ENUM service in line with the following requirements:

- Each ENUM record shall match an allocated telephone number;
- Each ENUM number shall be resolved in one SIP URI; and
- The domain of this SIP URI must be matched against the list of authorized domains by the calling network.

The use of the public ENUM tree, e164.arpa, is encouraged. However, if the public ENUM tree cannot be used, the platform shall provide a private ENUM tree and a set of telephone prefixes associated. The private ENUM tree may be accessible from public DNS or located on a specific server and shall only be queried if the E.164 number matches one of the associated prefix. The prefix may be a country prefix.

The query of ENUM trees for inter-service number discovery will be carried in the following order:

1. Query to e164.arpa;
2. Prefix matching then query to other nationally or internationally recognized ENUM trees; then
3. Platform prefix matching then query to private ENUM trees.

5 Relay Service Invocation

This section describes protocol mechanisms for controlling the invocation of relay services for the most basic call establishment mechanisms.

Refinements are possible and may be done by the REACH112 service providers. The basic set of relay service invocation methods documented below is however important for harmonization and interoperability.

5.1 Manual Actions by the Calling User

The user calls the destination by sip-address or number and makes an additional manual action to indicate that a relay service is wanted in the call. That manual action results in one of the following procedures.

5.1.1 Call to number@relay-domain

If the destination can be called by number, the called address is formed as number@relay-domain. The server at relay-domain controls a multimedia conference bridge and invokes the relay service and the destination in a relay three-party call. This mechanism is mandatory.

5.1.2 Referred-By pointing to relay service

The user terminal sends an INVITE with the address of the destination, including the Referred-By header pointing to the relay service. A pre-set route is also included pointing to the SIP server of the user's REACH112 service. The Sip server uses the Referred-By information (RFC 3892 [18]) to make the third leg of the call to the relay service. This mechanism is optional.

5.2 Manual actions by the destination

If the answering party of a call realizes that a relay service is desired in the call, a manual action can be made to invoke a relay service when the incoming call is offered.

The basic solution is the following, which however is optional depending on terminal capabilities:

5.2.1 Two calls

The user accepts the incoming call, and by user request an INVITE is issued to the relay service for a second call, and the media of the two calls are connected in a local mixer according to RFC 4117 [19]. Terminals having support for two simultaneous calls and local mixing can enable this operation.

5.3 Manual actions during a call

If, during a call, a user decides to invoke relay then the same action as in section 5.2.1 can be used.

5.4 Manual actions by the relay service agent

It should be possible to request a relayed call to be set up by prearrangement by a relay service.

The following should be the basic procedure:

Conference

The relay service is enabled to do conference establishment according to RFC 4579 [20], RFC 5366 [21] and RFC 5370 [22].

5.5 Automatic actions

It is possible and recommended to establish automatic methods that control the decision to invoke relay services in a call, based on criteria and information about the call participants. The methods may provide a good appropriate result for many call situations. They may need to be complemented with manual actions to allow the user to override the automatic procedure.

The most basic invocations by automatic actions are described here:

5.5.1 Automatic relay service invocation in calls from voice phone users

Many voice phones can only dial numbers. In order to be able to be called from all voice phones, total conversation users and real time text users need to have a phone number each.

For users who are not interested in direct voice-only calls, this number can lead to automatic relay service invocation together with the call being set up to the destination.

A suitable procedure is the following:

1. The voice phone user makes the call to the phone number of the user.
2. If the call was from the PSTN, the call arrives through a VoIP gateway to the IP environment.
3. The number is converted to a sip address by ENUM or by an internal list.
4. The call is routed according to the SIP address.
5. That SIP-address is not the final destination SIP address, but one that takes the call through the relay service invocation server, but is still unique for the called user.
6. The desired type of relay service is connected.
7. The destination terminal SIP address is deduced from ENUM or user data.
8. The destination is connected.
9. The call is completed.

In countries where PSTN based textphones prevail, a step may need to be added to the procedure for selection between textphone gateway invocation and relay service invocation. This is possible in countries where the textphone type used has a carrier tone. Detection of that carrier tone in any step of the procedure above can be allowed to invoke the textphone gateway and cancel invocation of relay service.

5.5.2 Automatic Relay Service Invocation in Calls to the PSTN and Voice only End Points

When a total conversation or real-time text user calls, a relay service can be invoked automatically if the user has indicated an interest to get relay included in all calls where the destination has only support for voice.

(The user is called total conversation user here for simplicity)

This is a suitable procedure for that simple automation.

1. The total conversation user makes the call to a phone number or SIP address of the destination.
2. The call is routed through the relay service invocation function of the REACH112 service provider.
3. The destination is interrogated by an OPTIONS transaction for support of media.
4. The caller's registered interest for getting relay services invoked automatically is interrogated and if true, the capabilities of the destination is compared with the capabilities of the calling terminal.
5. If the destination supports audio, but not any other common media, the decision is taken to invoke relay services.
6. The desired type of relay service is connected in a three-party bridge with support of video, real-time text and audio.
7. The destination is connected in the third leg of the tree-party call, and its voice channel is connected to the relay and the calling user.
8. The call is completed.

In countries where PSTN textphones prevail, it will not be possible to detect if a PSTN number destination supports text through a textphone. PSTN textphones make use of the voice channel for their text communication. If a user has registered interest in conducting textphone calls with PSTN textphones, and the call out to PSTN is met by textphone tones, the invocation of relay service should be dropped, and the textphone gateway activated instead.

This procedure will match the desired functionality in the majority of calls.

The user should be provided with means to do manual invocation of relay services, and to do manual actions to avoid unwanted invocation of relay services for the cases when the automatic action does not provide the desired result. The procedures outlined in the section above on manual actions can be used for such cases.

5.6 Media Control During Calls Including Relay Services

The relay service operator should be provided with means to control media mixing in relayed calls.

5.7 Location Information Handling in Relayed Calls

In emergency calls, it is important to maintain all information related to the calling user's location that may be provided with the call from the user, and make sure that it is conveyed to the destination.

6 Providing Emergency Services Access

The requirements for emergency services access, from a user's point of view, are described in section 3.3. Calls will be routed back to the caller's home REACH112 service when they roam outside their home region.

Call signalling shall include a URN in accordance with RFC 5031 [23] and in accordance with the recommendations of RFC 5012 and other specifications from the IETF ECRIT working group.

Call signalling should include location information in accordance with the recommendations of the IETF ECRIT working group.

The National Emergency Number Association (NENA), in the United States, has published a comprehensive specification for next generation emergency services [24]. NENA's document focuses only on Emergency services, whereas this document covers access networks as well. The REACH112 project focused on current deployment and so this document considers the short term to a greater degree than NENA's specification.

6.1 Background to Existing Emergency Services

The organisation of emergency services differs greatly across Europe. REACH112 cannot completely change the way emergency services are run and so REACH112 service providers must deal with emergency services access according to local circumstances. REACH112 accepts that in creating a new REACH112 service, there may or the may not be scope to alter the working of the existing emergency services that it works with.

To describe the range of possibilities we must first establish some terms and definitions. In this document we shall use the term PSAP (Public Safety Answering Point) to refer to the centres that answer emergency calls. In some countries, there are two levels of PSAP. The first – often referred to as a stage 1 PSAP - answers emergency calls to establish the location of the caller and which service is required. Once this information is established, the call is forwarded to an appropriate PSAP – referred to as stage 2 – that handles calls for the region from which the call is made. The stage 2 PSAP manages the dispatch of emergency service resources to help the caller. In general existing PSAPs are PSTN based and answer in voice only.

In some countries, calls are answered by a PSAP in the region in which the call is made, and this PSAP dispatches emergency resources for all types of emergency in their area and can communicate with other regions if the caller's network has connected them to the wrong location.

During the project we identify three main models for introducing total conversation to emergency services:

1. If a REACH112 service is able to integrate closely with its emergency services, it may be able to install total conversation equipment so that emergency call takers can answer calls in total conversation. A relay service may be needed for sign language translation. Alternatively, PSAPS may have staff who use sign language allowing them to take calls directly in sign, without the need for a relay service.
2. Where integration is not possible, there may be no alternative but to simply invoke the necessary relay service and call the emergency service in voice via the PSTN.

6.2 The Relay Model

In this model (see Figure 3), users primarily communicate with a relay operator who then relays information to the emergency services. In its simplest form, the emergency services are accessed via the PSTN. This model can work with any arrangement of existing emergency services.

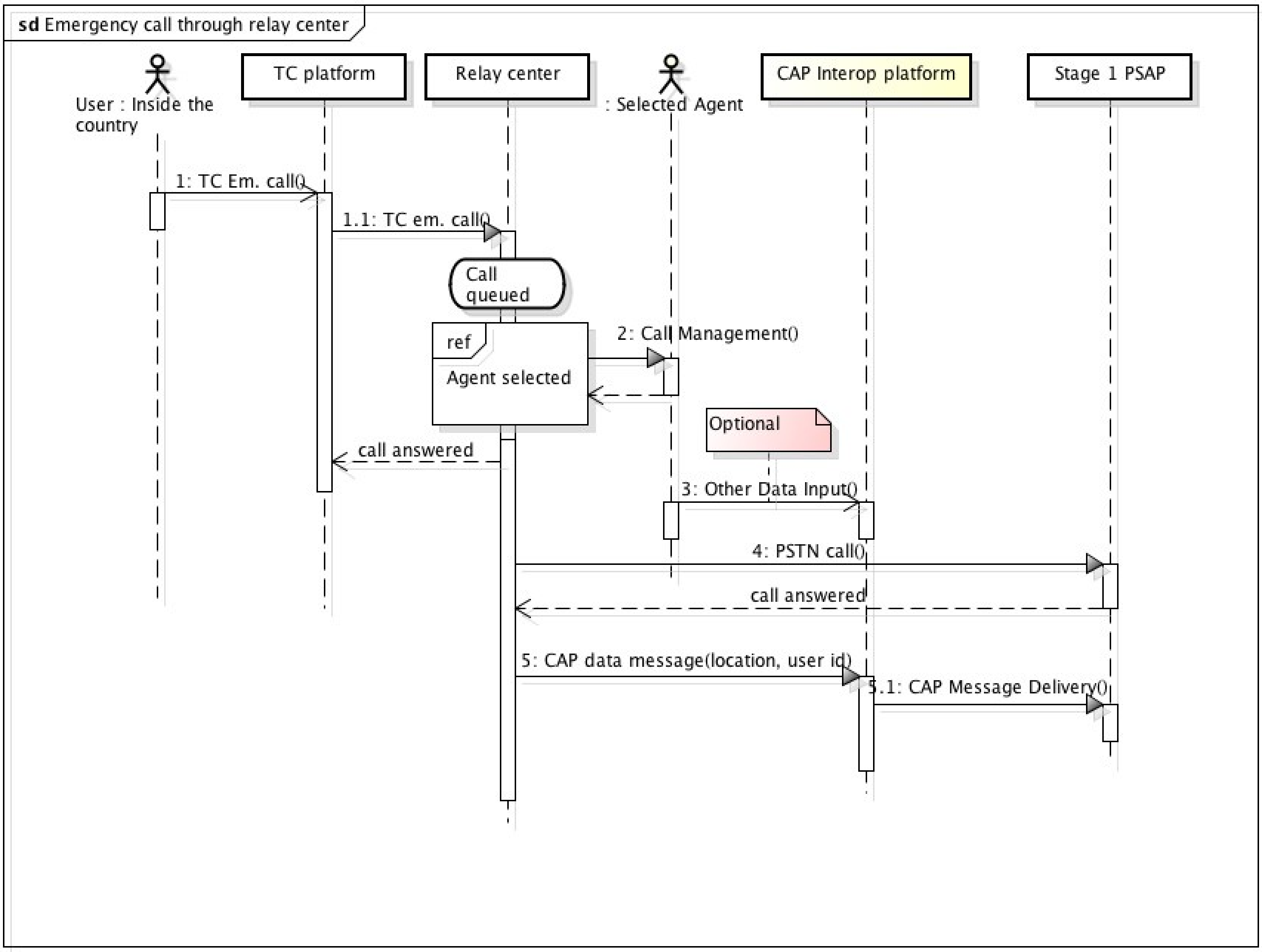

Figure 3: Emergency Call Relayed Directly to a Voice Stage 1 PSAP

However where emergency services are able to answer in total conversation, the user may be able to interact with the emergency services operator as well. For example when a sign language relay operator is listening to the emergency service operator, the Deaf caller, can see the emergency operator's lips moving and understands that they have to wait for the relay operator to start signing.

6.3 The Total Conversation Enabled Stage 1 Public Safety Answering Point Model

Here (see Figure 4) a stage 1 PSAP is provided, equipped with total conversation equipment and trained personnel. Calls can be taken in sign language, text or any other communication means. In this model, the stage 2 PSAP may be connected via the PSTN or it may available as a total conversation service.

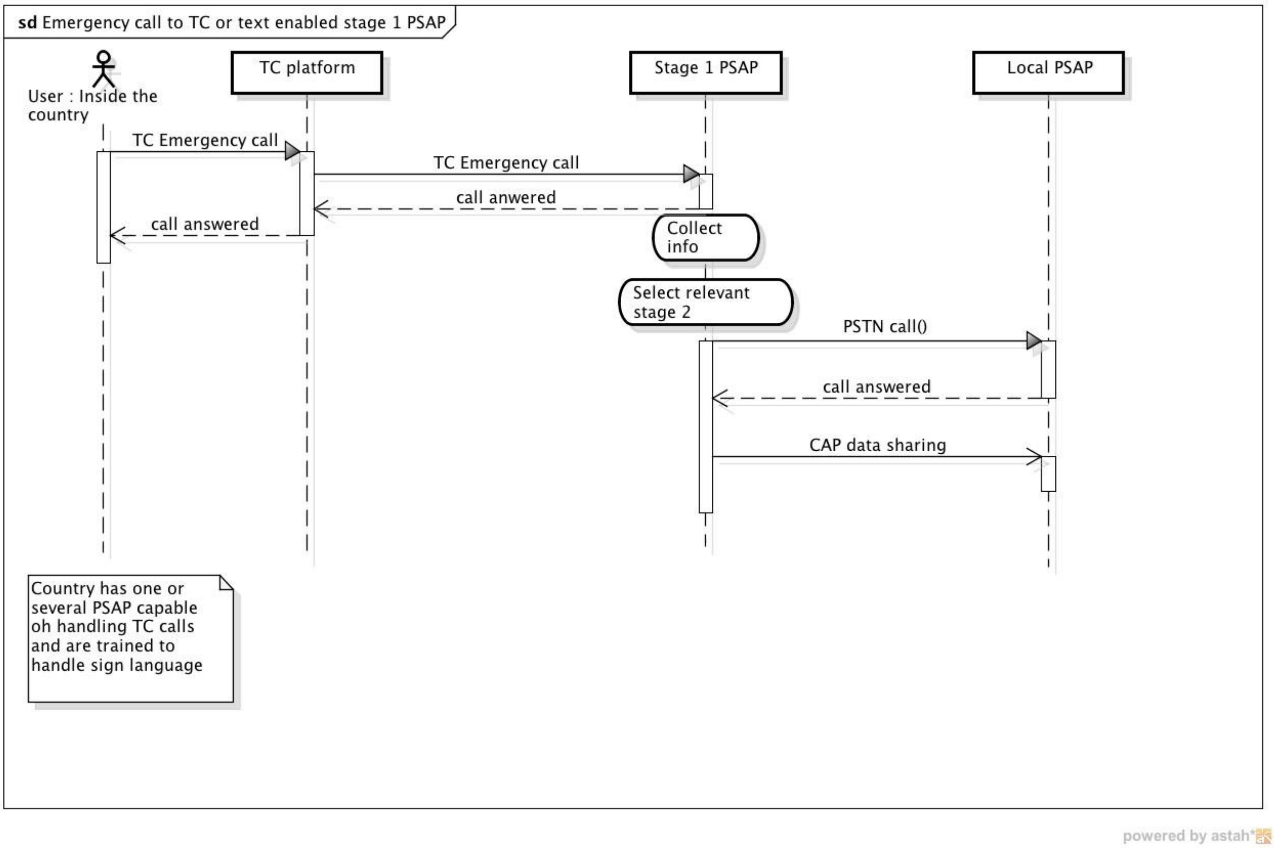

Figure 4: Emergency Call Handled by Total Conversation Enabled Stage 1 PSAP

6.4 The Model Combining Total Conversation Enabled Stage 1 Public Services Answering Point with a Relay Service

Here (see Figure 5) a stage 1 PSAP is provided, equipped with total conversation equipment and trained but not sign language competent personnel. A relay service is therefore invoked in a three-party fashion in the call. The call goes to both the PSAP and the relay simultaneously because total conversation provides some means of communication even if the relay service introduction is delayed. Calls can be taken in sign language, text or any other communication means.

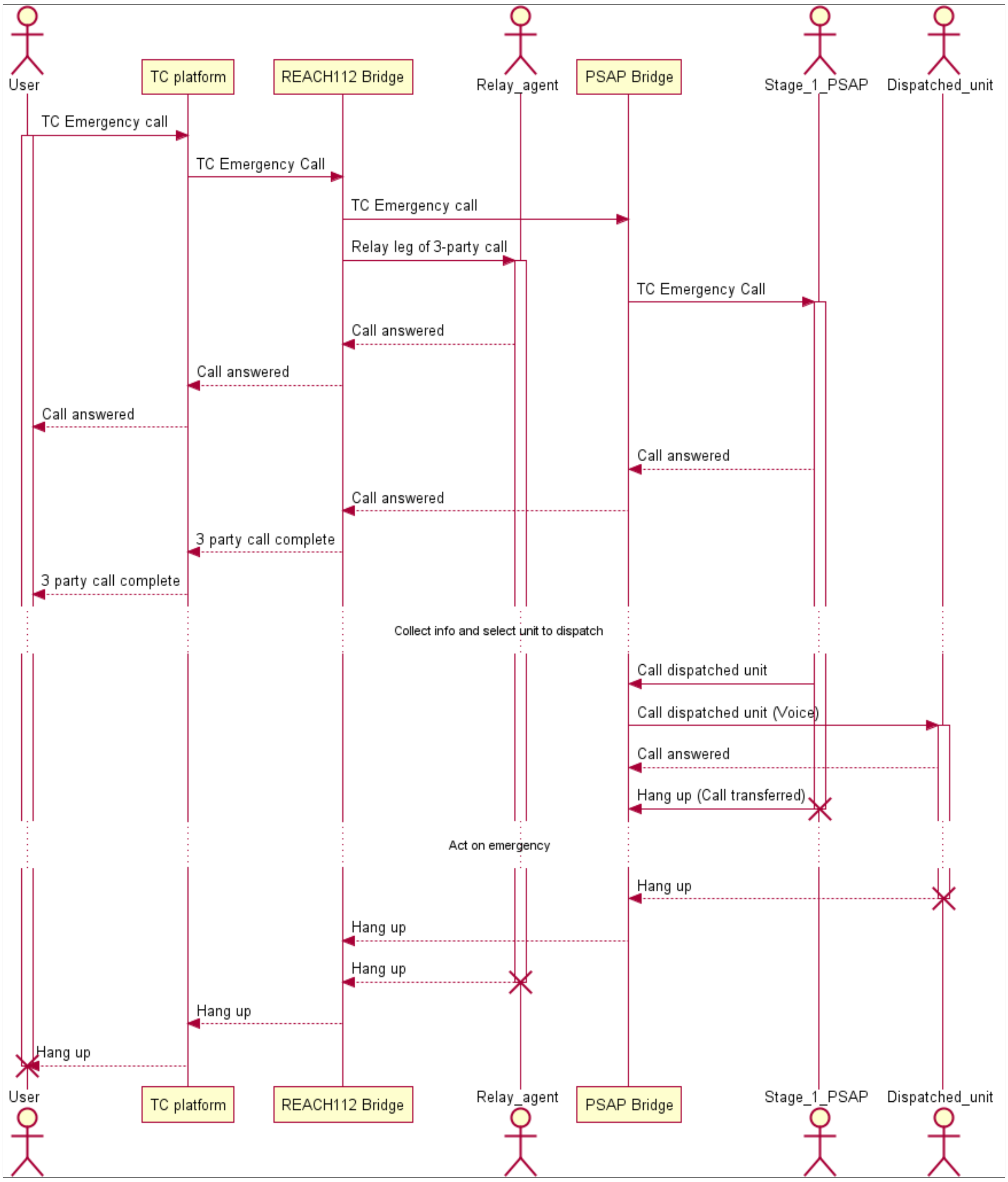

Figure 5: Emergency call handled by relay and Total Conversation enabled PSAP

In this model, the dispatched unit may be connected via the PSTN or it may be available as a total conversation service. When the dispatched unit has taken over the dialogue, the stage1 PSAP may disconnect, because the relay service continues relaying the conversation in sign and text.

6.5 Emergency Services Invocation

When a user places an emergency call, the total conversation platform shall recognize it according to the called number or URI. It should tag it as a priority call. From that point, a number of platform specific call routing strategies may be adopted:

- The call may be routed to the most appropriate PSAP equipped with total conversation terminals;

- The call may be routed through a relay service so as to be connected to a non-total-conversation equipped PSAP;
- The platform may invoke a relay service so as to be connected to a non-total-conversation equipped PSAP; or
- The platform may invoke a relay service and a PSAP with total conversation capabilities.

In the case of relay invocation, the platform may chose to invoke the relay operator in a series or parallel fashion. Series invocation connects the caller to the relay operator and then the PSAP is called, either manually or automatically by the platform. With parallel invocation, the platform dials the relay operator and the PSAP at the same time. Call handling procedures in the relay call centre and the PSAP must educate the operators to this behaviour where the other operator is not yet connected to the caller.

The platform shall give priority to emergency calls over other waiting calls and prioritize any subsequent PSAP call-back.

6.6 Reliability

The link between the platform and the PSAP terminals should be constructed on a network with controllable quality and backup to ensure the availability of emergency services. Consideration should be made to using a fall-back PSTN connection from the platform to the PSAP in case of IP network failure or congestion. A call from a platform to a PSAP should use at least a three tier fall-back of:

- Primary IP Connection;
- Secondary IP Connection; and
- PSTN Connection.

6.7 Callback

The information provided with the call to the PSAP shall be sufficient, for the PSAP, to call the calling user back. If a relay service was included in the incoming call, the same kind of relay service shall be included in any callback. The callback shall be identified as an emergency call in the ways that the IETF standards specify. This is important as it is sometimes necessary for the PSAP call the caller back to get more information or to inform them of progress in dealing with their situation.

6.8 Location Information

An extensive internal report on call routing, location based information and caller-id has been written within the REACH112 project. This work has not been integrated into this document due to time constraints, but it is expected that location information will be dealt with much more extensively in a further version.

Location information is of great value in emergency calls. In case of such a call the user to network interface shall make provision to transfer location data as additional information to third party applications. The protocol must be able to:

- Transfer civic addresses; and
- Transfer geo-location information (i.e. long, lat, altitude).

Where SIP is used for signalling, the specifications from IETF ECRIT regarding location information should be implemented. The location must be inserted according to Location Conveyance for the Session Initiation Protocol [7].

The format to encode the information above must be LO-PIDF. Some additional information must be added if available:

- The cell ID of caller; and

- The public IP address of the caller's network access.

Location information must be inserted in an emergency call:

- By the terminal if it is equipped with a GPS capability and is subsequently capable of obtaining its location at the time of the call;
- By the infrastructure using the location information entered by the user profile; or
- By the infrastructure itself using exiting or future IP geo-location services.

All mistrusted location information shall be filtered out by the calling network.

Location information must be sent when the call is initiated and may be updated during the call. The location may be sent by value where all the information is included in the signalling packet or by reference. In the later case, the signalling only carries a URI that points to a location server.

The location server will be needed to enable the translation of the location reference URI into an actual location specified in the LO-PIDF format. The access protocol for retrieving the location is restricted to:

- HTTP, in which case, the UA shall use a GET query to obtain the data and may poll the HTTP URI to obtain the data; or
- SIP, in which case, the UA shall SUBSCRIBE to the URI and obtain the data through NOTIFY messages sent by the location server.

6.9 Data sharing function

The data sharing, in the REACH 112 context, is functionality implemented as a platform dedicated to the exchange of CAP messages between total conversation users. Its main use is to enable a total conversation enabled stage 1 PSAP to deliver collected information to a stage 2 PSAP or to a PSAP of a different organisation, in an open and automated manner. A CAP message may include the recording of the call media. The CAP protocol is an open XML-based standard structure, which allows an effective standardisation of what a PSAP typically needs to share with other PSAPs.
The data sharing chain is composed of the following web-based components:

- a CAP generator module, which is the information system that total conversation systems invoke for issuing CAP message or to update a previous message;
- a CAP router module, which make the CAP messages available to the intended recipients (i.e. other PSAP) selected by the sender PSAP; and
- a CAP viewer module which is the information system capable of receiving CAP messages, storing and displaying them.

REACH 112 total conversation emergency platforms should be capable of generating CAP message (i.e. implementing the CAP Generator functions). It is the responsibility of the sender to provide a unique ID for the CAP message. The transport interface to post CAP messages to the CAP Router module must use HTTP to send the message.

If data sharing is enabled on a platform, the use of a CAP Router will allow the sender to:

- check the message structure and confirm if it is a valid CAP message;
- store a copy of the CAP message; and
- provide CAP message reference to the intended recipient(s) in the form of an HTTP URI.

This reference may be conveyed in SIP signalling using the Call-Info header.

A CAP enabled target platform may:

- fetch the CAP incident using the HTTP protocol and a CAP reference URI; or
- subscribe to an RSS feed and scan it to retrieve the incident data.

REACH 112 makes the provision of: (i) a CAP generator application able to receive raw data from a total conversation system and create a correct CAP message to be shared; and (ii) a CAP viewer application that enables stage 2 PSAPs that are not CAP enabled to display the shared information using a simple web browser. The distribution of CAP messages between sender and recipient(s) is also offered by the provision of a simplified CAP routing functionality.

6.10 Total Conversation Call Recording

In many emergency services, all conversations between the public and the emergency responder are recorded. Total conversation compliant emergency platforms shall be able to record the total conversation conversations between the agent and the responder with all media used (audio, video and text). Retrieval and play-out of recorded sessions shall also be provided. All usual privacy and security concerns related to telecommunications and emergency service applies.

7 Implementation Guide

This section is to provide guidance, although publishing interconnect information to other REACH112 services according to section 7.1 is mandatory. Sections 7.3, 7.4 and 7.5 form an example of how to handle calls from another REACH112 service, from the service's own users and how to handle a call from either of these sources, that is directed at one of the service's own users.

7.1 Interconnection Parameters to Circulate to Other REACH112 Peers

In the REACH112 pilot, the information necessary to set up peer links has been published on a website that is accessible to all REACH112 partners. It is important that all REACH112 services establish a peering arrangement with each of the others, but how the information is communicated is to be decided by individual services. It is important that in the current arrangement, some of this information could be of some use to people who might attempt denial of service attacks against REACH112 services and their users and so this information should always remain confidential between partners. Error: Reference source not found shows the parameters that must be communicated between REACH112 services.

Parameter	Comment
List of domains	This is the list of domains that may appear in target URIs for inbound INVITEs and in from addresses and p-asserted identities in outbound INVITEs.
Interconnect points	This is a list of domain names or IP addresses of SIP interconnect points. For any one call, one of these addresses must be used for both SIP traffic and media traffic. Implementers will usually use a back to back SIP user agent as an interconnect point, to achieve this.
ENUM number range	The rang of numbers that might be found in the service's ENUM tree. This is published so to reduce the number of ENUM queries that must be done in processing a call.
ENUM source addresses	A list of source addresses from which ENUM queries to other network's ENUM trees will be made.
ENUM suffix	The suffix to apply to a number when making an ENUM DNS query.
ENUM server address	The server to query when looking up numbers for this service. This can be a host name, or "Public DNS". In the latter case the ENUM lookup should begin with the public route DNS servers or cached information there-from.
Test services	The numbers or URLs of automated test services that can be used to make test calls. It is suggested that at least an echo service should be provided that will accept calls and echo back media.
Terminals	Optionally a REACH112 service may circulate a list of SIP terminals that it uses. This may provide information that is useful in debugging call faults. However REACH112 services may use a back-to-back user agent at their interconnect point which will hide differences between terminal types to a great extent. The name of the SIP stack, SIP server and RTP stack used at the interconnection point could also be listed here.
Virtual Private Network details	This is beyond the scope of the current specification but REACH112 services may, by mutual agreement, use virtual private networks to carry SIP and media data between them. In a future version of this specification, this may be mandatory.

Table 5: Details that a REACH112 Service Must Distribute to its Peers

7.2 Example Architecture for a REACH112 Service

REACH112 does not mandate a particular architecture internally so providers may design their networks as they feel is appropriate to local circumstances. Here we give an example architecture Figure 6. The major components are numbered 1 to 12 and the main communication links are labelled A to H going clockwise around the diagram. These are described in the text. These components may be implemented on single servers or may be clusters of systems for scalability and reliability.

It is important to note that interfaces B, D and E are the inter-service, PSAP and Relay interfaces mentioned in section 2 and must use SIP signalling in accordance with section 4.

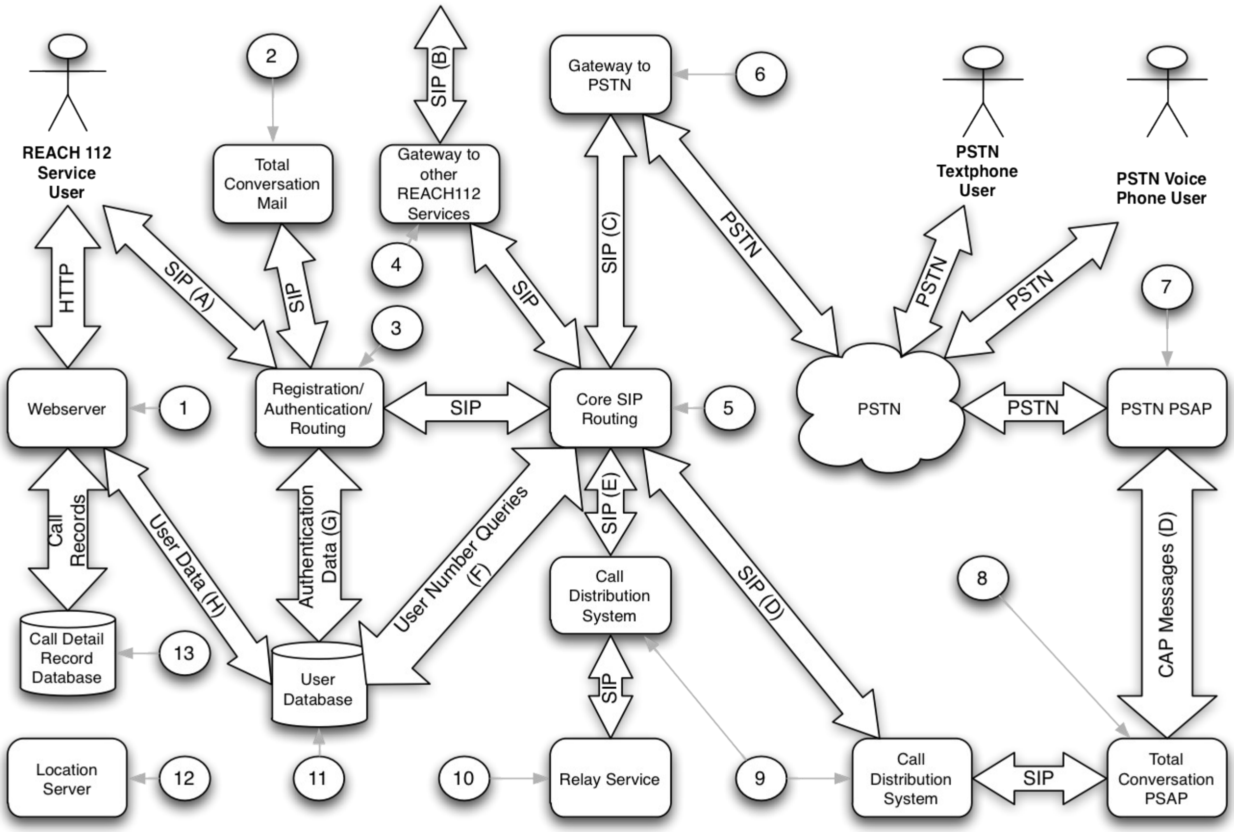

Figure 6: Example REACH112 Architecture

This architecture centres around a core SIP routing system (5) which directs calls to various gateways and systems. It connects to the user database (F) so as to be able to detect which telephone numbers or names refer to registered users and is so able to route calls for them via proxy (3).

Service users have SIP terminals which may be hardware or software based and which register with and route calls through system (3). System (3) is both default proxy and registrar for the user's terminal and it provides user authentication (A). It can also direct calls, for users who are offline, to their mail boxes (2) and may act as a back-to-back user agent, ensuring that all media in peer to peer calls goes through it which can improve reliability for users behind some firewalls. Users can access their personal details and call records and may be able to register and download terminal software via a website provided by a server (1) which has access to the user (11) and call detail record databases (13).

Calls to and from other REACH112 services go via a gateway (4). This may be a back-to-back SIP user agent that can separate internal signalling from inter-service signalling to other services (B).

Calls to and from the PSTN go via a PSTN gateway (6). Here we show only one PSTN gateway although it is likely that there would be separate gateways for voice and textphones. Emergency calls to the PSTN also go through this gateway. This diagram doesn't show how calls are routed but the PSTN PSAP (7) could be a stage 2 PSAP that receives calls that have been initially handled by a total conversation enabled PSAP (8). The stage 1 PSAP can transfer details captured to the stage 2 PSAP using CAP messaging (D).

Both the relay service (10) and total conversation enabled PSAP (8) have call distribution systems (9) that handle call queuing and distribution to agents. These components may be run by the REACH112 service although as already mentioned the protocol for links (E) and (D) are defined in section 4. Two separate REACH112 service could share a the relay or total conversation PSAP by connecting to their respective call distribution systems (9).

For simplicity, most of the links to the call detail record database are not shown. The points that would be expected to generate call detail records are: the inter-service gateway (4); the PSTN gateway (6); the call distribution systems for the relay and PSAPS (9); and the User SIP Proxy (3).

7.3 Receiving a Call from Another REACH112 Service

Calls from other REACH112 services will always be signalled with SIP according to section 4 and in particular section 4.6. Security between REACH112 partners is primarily based, on checking the source IP address of inbound connections so that only other REACH112 peers can connect. This level of security is common in the telecommunications industry at present but the use of virtual private networks is considered to be a likely addition to future REACH112 specifications.

In addition to checking source IP addresses, a REACH112 service may make the following checks on SIP messages from other REACH112 services:-

Check	Details
Check the domain in the P-Asserted identify and from address of inbound INVITE messages.	The domain should be in the list of domains handled by the peer that matches the source address of the SIP INVITE.
Check valid ENUM.	As all REACH112 users must have e164 numbers to make inter-REACH112 calls, a REACH112 service may check that the username in the from address or P-Asserted identify can be found in the ENUM tree of the REACH112 service that matches the source address of the SIP INVITE.

Table 6: Checks on Inbound INVITE messages from other REACH112 Services

These checks are to ensure that the calling number or URI, displayed to the callee is valid for making a return call.

If an INVITE message passes the checks and its target URI is an emergency URI then it should be handled as an emergency call and if it matches a user account then it should be handled as in section 7.5.

7.4 Routing of a User's Call

REACH112 services may route calls in whatever way they choose, within their own service. Error: Reference source not found shows a realistic example of how a REACH112 service might route a call from one of its own users. In this case, the relay service handles both emergency and non-emergency calls. It may simply relay all calls to the PSTN or it may provide a stage 1 PSAP function by asking the caller for their location and the service they require.

In this example, the user terminal routes all calls to a single proxy and so the REACH112 service routes all SIP invite transactions. The terminal may have only a phone key pad or it may allow the user to enter non digits and domain names. As all REACH112 users are required to be allocated e164 numbers, there is no requirement for domains and non numeric names but neither are prohibited. The first decision point (1) in Error: Reference source not found is to add a domain to the destination address if one wasn't already supplied. When a domain is added, it could be the default for the REACH112 service or configured for the user. Decision (2) allows for connections to other domains by name. A connection will only be made to a destination if that domain has a peering relationship with the REACH112 service (3). In decision (4), emergency calls are sent to the relay service which will relay either to the PSTN or to a total conversation enabled PSAP. The identifier for emergency calls might be: 112, 999, 911 or SOS in the name field; or any of these formatted as “tel: identifier” [23]. If the user name in the destination address is a registered user under the domain then the call is routed to that user (5) as in section 7.5. In decision (6), the name is matched against all the peer's ENUM trees and if a match is found, then the call is sent to the peer URI found. If there is no ENUM match at (6) then the call is sent to the relay service which will attempt to relay to the PSTN.

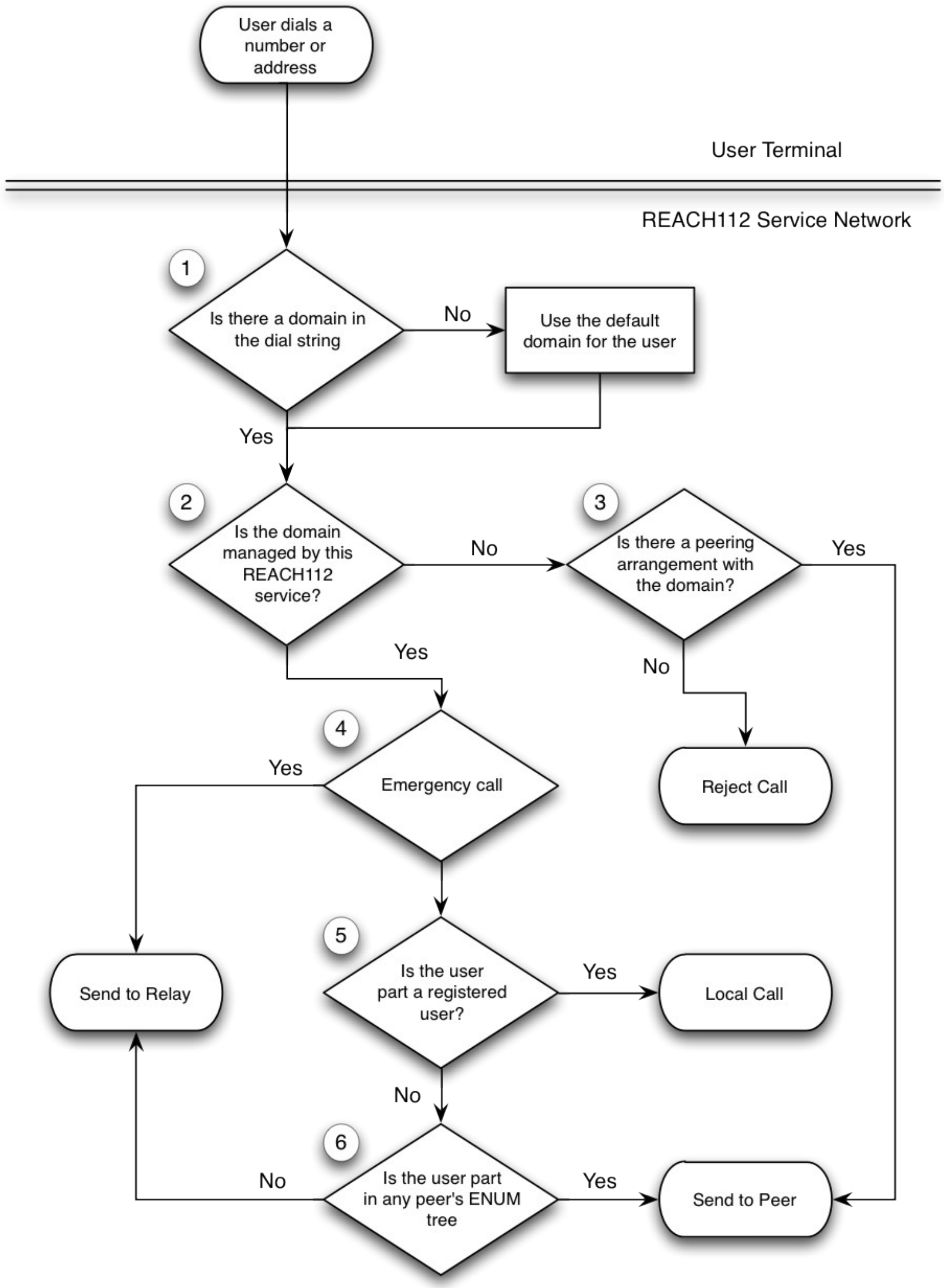

Figure 7: Routing of a User Call

In addition, the relay service may offer assistance when a call is made to an invalid number and there could be number prefixes that can be used to signal that the caller wants to call another total conversation user but with relay involved. In general the outbound call that the relay service makes, could be treated as another call from an ordinary user apart from it never being looped back to the relay service but being rejected if the number cannot be matched at decision (6).

In this example, all call signalling uses SIP according to section 4 except for breakout to the PSTN. The emergency services access could be of the relay model in section 6.2 or the total conversation enabled stage 1 PSAP model in section 6.3 depending upon how the relay service handles emergency calls.

7.5 Handling a call to a Service User

A call can be directed to a user if: it has come from another REACH112 service as in section 7.3; it has come from another user on the same service as in section 7.4; or it has come from the relay service were the caller asked the relay operator to connect them to the callee also in section 7.3.

Once it has been established that a call is to be attempted to a service user, the platform checks whether the callee is online. This is normally done by a SIP Proxy that is also acting as a SIP registrar and therefore holds the details of all users who are registered with the service. If the callee is not currently registered online, then the call is sent to a total conversation mail service. If the callee is online then the call is sent to them but redirected to the mail service if the callee doesn't answer within a predetermined time period.

8 Testing

It is important to verify that communication works correctly from a user's perspective when making calls between REACH112 services. It is the responsibility of REACH112 services to make sure that this happens by mutual cooperation. However the primary way that interoperability is achieved in a sustainable way is for service providers to follow the standards and guidelines in this document. Where problems are found, it may be necessary for the specifications to be further tightened. The first principle in maintaining interoperability is to establish at all times the rules that must be followed by REACH112 services. If the current specifications are inadequately defined then services should agree extra rules between themselves and engage with standardisation work. This is far better than making add hoc fixes to client software and terminals to make them work in current circumstances.

As previously suggested in section 7, REACH112 services may choose to implement a back-to-back user agent as an inter-service gateway so that inter-service communication is isolated from the details of internal communication. We do not suggest that every user terminal type on every REACH112 service be tested against every terminal type on every other service as this is unsustainable as new terminal software is developed and refined. We instead list tests that should be done using representative client software or specific automated test software as may be appropriate. Further testing should be done by service providers when users report problems. Table 7 lists the basic tests that should be done on a new inter-service link or whenever there is a need to verify interoperability between REACH112 services or terminals. Further detailed test specifications for both inter-service and intra-service tests are collected in separate documents.

Test	Comment	Pass Criteria
Inbound call	Make calls. These should be done in total conversation and any combination of media that might get used by clients on the network. If a service does not allow any client software that can switch off media then it is only necessary to test in total conversation. If the other REACH112 in the test has any client terminals that support answering without accepting all media, then all combinations of turning off media should be tried.	Call established and all media negotiated work.
Outbound call	The same as inbound call except initiated by the other REACH112 service in the test.	
Caller hangup	Verify that when the caller hangs up a call, that both ends see the call indicated as hung up.	Call appears hung up on client terminal. This varies but normally camera light is turned off and text entry box closed or greyed out.
Callee hangup	Same as "caller hangup" except hung up by the callee.	
Long call	Make a call and check that after 20 minutes waiting that media are still flowing. During this 20 minutes wait, audio should be muted at both ends, if client terminals support this and text should not be typed.	Call media flows in both directions after 20 minute wait.
Multiple simultaneous Calls	At a minimum, two simultaneous calls should be tested. Ideally a test should be done with the designed capacity of the interconnect. If there is a hard limit on concurrent calls over the interconnection then this should be tested to check that users get a suitable failure message.	All media flow in both directions on all calls. If a hard limit on concurrent calls is reached then the caller should get a suitable message when they are rejected.
Multiple consecutive calls	At least 10 consecutive calls should be made to verify that results are reliable. If a call fails then more test should be done to establish failure rate. As REACH112 inter service links use the public Internet, we do not set a tight service level but more than one failure in ten attempts should be considered unacceptable.	< 1 failure in 10 attempts
Re-invite send	If a REACH112 service expects to make re-invites in calls then this should be tested. This might happen when a relay service is introduced into a call in which case making a relay call to the other REACH112 service in the test would achieve this.	All media continue to flow after the re-invite has happened.
Re-invite receive	Same as "re-invite" send but done by the other REACH112 service in the test.	
Latency	Media latency should be measured, at least approximately. This can be done manually or using measurement tools.	Text latency < 500ms, Audio latency < 200ms, Video latency

Test	Comment	Pass Criteria
		< 200 ms.
Real-time text quality	Send text and verify text compatibility with at least the following tests, by verifying with the receiver how the display looks at the far end and compare with the transmission side: Send plain text. Send national characters (e.g. åäöüñ etc) Make new line and type on the new line. Erase a few characters. Verify that the displays are in synch. Erase back over a manually inserted new line and a few characters on the line above. Verify that the displays are in synch. Change colour or style of text, if that is supported, check if it is displayed properly if it is supported, or properly ignored if it is not supported. If possible, cause packet loss and verify that loss of one or two packets are recovered, while loss of three or more packets is marked on receiving side with the replacement character or an apostrophe.	Text is equivalent on both sides except for any features not supported, and for marks for lost text if the packet loss and recovery test was performed.
Video quality	This should preferably be done by two sign language users to check that they can adequately use sign language over the call. Statistics gathered by client software or network analysis may also be used. Video size should at least QCIF.	Adequate for sign language. If measured frame rate >= 20fps. Size >= QCIF.
Audio quality	This can be done subjectively. The audio quality should be adequate for understanding without glitches from packet loss or jitter.	Adequate for audibility.

Table 7: Inter Service Tests

9 Conclusion

This document is intended to show how total conversation can be used for the benefit of Deaf, hard of hearing and speech impaired people in communicating with the emergency services and with other people. We have defined the concept of a REACH112 service that uses total conversation for calls:

- between its own service users;
- between users and the emergency services;
- between its users and users of other REACH112 services; and
- between its users and certain other networks like the PSTN.

In all cases, the specified platform can accommodate the presence of relay services where required.

The document introduces the required functionality from a user's point of view and goes on to describe the use of communication protocols to provide such functionality. The Session Initiation Protocol (SIP) is the core protocol used. Pilots in five European countries have verified the feasibility of implementing REACH112 services.

It is our intention that this document will be used as a base specification and implementation guide for further deployment of total conversation services in the European Union. Furthermore we hope to promote political and regulatory support for these most essential services.

10 References

[1] ETSI SR 002 180, "Requirements for communications of citizens with authorities/organisations in cases of distress (emergency call handling)", ETSI, December 2003

[2] J. Rosenberg et al, SIP: Session Initiation Protocol, IETF RFC 3261, June 2002; www.ietf.org/rfc/rfc3261.txt

[3] ETSI ES 202 975, "Human Factors (HF); Harmonized relay services", ETSI, October 2009

[4] N. Charlton et al, User Requirements for the Session Initiation Protocol (SIP) in Support of Deaf, Hard of Hearing and Speech-impaired Individuals, IETF RFC 3351, August 2002; www.ietf.org/rfc/rfc3351.txt

[5] 2009/136/EC, Directive 2009/136/EC of the European Parliament and of the Council, Official Journal of the European Union, November 2009

[6] ETSI TS 122 173, "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); IP Multimedia Core Network Subsystem (IMS) Multimedia Telephony Service and supplementary services; Stage 1 (3GPP TS 22.173 version 7.4.0 Release 7)", ETSI, October 2010

[7] J. Polk, B. Rosen, J Peterson, Location Conveyance for the Session Initiation Protocol, draft-ietf-sipcore-location-conveyance-08, IETF Internet draft, work in progress, May 2011

[8] J. Rosenberg, H. Schulzrinne, An Offer/Answer Model with the Session Description Protocol (SDP), IETF RFC 3264, June 2002; www.ietf.org/rfc/rfc3264.txt

[9] M. Handley, V. Jacobson, C. Perkins, SDP: Session Description Protocol, IETF RFC 4566, July 2006; www.ietf.org/rfc/rfc4566.txt

[10] H. Schulzrinne et al, RTP: A Transport Protocol for Real-Time Applications, IETF RFC 3550, July 2003; www.ietf.org/rfc/rfc3550.txt

[11] J. Ott et al., RTP Payload Format for ITU-T Rec. H.263 Video, IETF RFC 4629, January 2007; www.ietf.org/rfc/rfc4629.txt

[12] S. Wenger et al., RTP Payload Format for H.264 Video, IETF RFC 3984, February 2005; www.ietf.org/rfc/rfc3984.txt

[13] B. Haberman and J. Martin, Internet Group Management Protocol Version 3 (IGMPv3) / Multicast Listener Discovery Version 2 (MLDv2) and Multicast Routing Protocol Interaction, IETF RFC 5168, May 2008; www.ietf.org/rfc/rfc5186.txt

[14] G. Hellstrom et al, RTP Payload for Text Conversation, IETF RFC 4103, 2005; www.rfc-editor.org/rfc/rfc4103.txt

[15] G. Hellström, Presentation of Text Conversation in real-time and en-bloc form, draft-hellstrom-textpreview-08, IETF Internet draft, work in progress, May 2011

[16] X. Marjou, A. Sollaud and France Telecom Orange, Application Mechanism for Keeping Alive the NAT Mappings Associated with RTP/RTP Control Protocol (RTCP) Flows, IETF RFC 6263, June 2011; www.rfc-editor.org/rfc/rfc6263.txt

[17] B. Rosen and J. Polk, Best Current Practice for Communications Services in support of Emergency Calling, draft-ietf-ecrit-phonebcp-18, IETF Internet draft, work in progress, July 2011

[18] R. Sparks, The Session Initiation Protocol (SIP) Referred-By Mechanism, IETF RFC 3892, September 2004; www.ietf.org/rfc/rfc3892.txt

[19] G. Camarillo, Transcoding Services Invocation in the Session Initiation Protocol (SIP) Using Third Party Call Control (3pcc), IETF RFC 4117, June 2005; www.rfc-editor.org/rfc/rfc4117.txt

[20] A. Johnston and O. Levin, Session Initiation Protocol (SIP) Call Control - Conferencing for User Agents, IETF RFC 4579, August 2006; www.ietf.org/rfc/rfc4579.txt

[21] G. Camarillo and A. Johnston, Conference Establishment Using Request-Contained Lists in the Session Initiation Protocol (SIP), IETF RFC 5366, October 2008; www.ietf.org/rfc/rfc5366.txt

[22] G. Camarillo, The Session Initiation Protocol (SIP) Conference Bridge Transcoding Model, IETF RFC 5370, October 2008; www.ietf.org/rfc/rfc5370.txt

[23] H. Schulzrinne, A Uniform Resource Name (URN) for Emergency and Other Well-Known Services, IETF RFC 5031, January 2008; www.rfc-editor.org/rfc/rfc5031.txt

[24] NENA 08-003, Detailed Functional and Interface Specification for the NENA i3 Solution – Stage 3, NENA, June 2011