GALLAUDET UNIVERSITY, OMNITOR, AND UNIVERSITY OF WISCONSIN-MADISON

Real-time text Interoperability

Status and field trial

http://tap.gallaudet.edu/IPTransition/TTYTrial/

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TECHNOLOGY ACCESS PROGRAM

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1 Executive Summary

This is a report on user experience with real-time text, and the status of its deployment and standardization. Real-time text (RTT) is a mode of text communication, where the text is sent immediately while it is being created. Real-time text is often combined with audio and also often with video in a multimedia call offering the benefit of using all media that are of value for the real-time communication situations. Real-time text is both an accessibility feature for communication with persons who are deaf, hard-of-hearing, deaf-blind or speech-disabled, and also a feature with advantages for all users on voice or conversational text calls.

One of the main components of this report is a field trial on the interoperability of real-time text in three different calling environments: TTY on PSTN, RFC 4103 on SIP, and experimental RTT on WebRTC. The field trial was performed in 2015 by the Rehabilitation Engineering Research Center on Telecommunication Access (RERC-TA) partners in Sweden and the United States. 49 participants were involved including people who were deaf, people who were hard of hearing, and hearing friends and relatives of the deaf or hard-of-hearing participants. Participants made RTT-only and RTT+voice calls across these three environments.

Key findings are:

- Participants reported high satisfaction scores on the tested RTT technology.
- Among those participants who tested RTT+voice, a majority deemed this feature to be important.
- Participants preferred sending and receiving real-time text over IM-style turn-based text
- Participants preferred being able to type at the same time as their partners.
- Participants overwhelmingly judged interoperability across TTY and RTT, and interoperability across different calling environments, to be critical.

Participants most frequently asked for these additional features, not covered in the trial:

- Addition of video to the conversation.
- Better mobility through implementation of RTT solution on smartphones.
- Alerting devices for accessible indication of incoming calls.
- More control over fonts, colors, etc.
- Improved text conversation handling by splitting up long text from TTYs.

The findings of this trial are consistent with was what reported in earlier research, which are reviewed in this report. The current state of RTT standards and how the findings relate to these standards are also discussed.

The main conclusion from the field trial is that RTT is preferred over messaging for conversational situations. The main conclusion from the standards discussion is that RFC 4103 is the most widely cited standard for RTT, and should be used in SIP and IMS technologies. For environments where RFC 4103 does not fit, conversion to RFC 4103 should be supported wherever they interface with SIP or IMS.

2 Terms

2.1 Abbreviations

3GPP	Third Generation Partnership Program; consortium for specification of wireless systems
AEGIS	Open Accessibility Everywhere: Groundwork, Infrastructure, Standards; (European accessibility project)
EAAC	Emergency Access Advisory Committee
EENA	European Emergency Number Association
EG	ETSI Guide
EN	European Norm
ES	European Standard
ETSI	European Telecommunications Standardization Institute
FCC	Federal Communications Commission; the electronic communications authority in USA
GA	Go Ahead; convention used for giving a turn in TTY communication
GSMA	GSM Association
GTT	Global Text Telephony
HTML5	HyperText Mark-up Language version 5 (language for web content)
IETF	Internet Engineering Task Force, standards organization for Internet standards
IM	Instant Messaging; general term for text messaging
IMS	IP Multimedia Subsystem
IP	Internet Protocol
ITU-T	International Telecommunications Union -
NENA	National Emergency Number Association

NIDILRR	National Institute on Disability, Independent Living, and Rehabilitation Research
PBX	Public Branch Exchange; telecommunications switching system
PSTN	Public Switched Telephone Network
RFC	Request For Comments, the term for standards from IETF
RERC-TA	Rehabilitation Engineering Research Center on Telecommunications Access, a collaborative research project under the University of Wisconsin-Madison, Gallaudet University, and Omnitor with partial funding from NIDILRR
RIM	Research In Motion; (telecommunications company)
RTP	Real-Time Protocol
RTT	Real-time text
SIP	Session Initiation Protocol
SMS	Short Message Service (text messaging in wireless technology)
TC	Total Conversation – simultaneous video, audio and text in real-time
TDI	Telecommunications for the Deaf Inc. (a US consumer advocacy group)
TIA	Telecommunications Industry Association
ToIP	Text over IP. an earlier term for RTT in IP technology
TS	Technical Specification
TTY	Text Telephone for PSTN of the type used in USA
VoIP	Voice over IP
VRS	Video Relay Service
WebRTC	Web Real-Time Communication
XEP	XMPP Extension Protocol
XMPP	Extensible Messaging and Presence Protocol

2.2 Definitions

real-time text	text transmitted instantly while it is being typed or created
WebSocket	data communication protocol specified in RFC 6455 [18]
web-based	technology using web pages, web browsers and web servers

3 Introduction

This is a report on user experience with real-time text, and the status of its deployment and standardization in electronic communications.

This report covers a field trial of real-time text and its interoperability. This trial was performed by the Rehabilitation Engineering Research Center on Telecommunications Access (RERC-TA) across three different calling environments: TTY on PSTN, RFC 4103 on SIP, and an experimental RTT on WebRTC. The trial was performed during 2015.

The report also covers overviews of the current deployment situation and an overview of standards specifying and including real-time text.

Real-time text is a mode of text communication, where the text is sent immediately while it is typed, so that the receiver gets an opportunity to follow the thoughts of the sender as they are formed into words. Functionally, this is akin to voice calls where the receiver also follows the words of the sender the moment they are formed and spoken. The RTT mode contrasts with the messaging mode, where text is collected by the sender in messages, and a special action (e.g., hitting Enter or pressing Send) is used to transmit the message only after the sender has finished composing it in its entirety.

Real-time text is often discussed as an accessibility feature for communication with persons who are deaf, hard-of-hearing, deaf-blind or speech-disabled, but may have wider applications in the mainstream.

Real-time text is often combined with audio and also often with video in a multimedia call offering the benefit to use all media that are of value for the actual communication situation.

Real-time text has previously been implemented in a functionally limited form in TTYs in the USA using the traditional telephone network (PSTN). The TTY is the text telephone type used in USA. There is an ongoing transition from PSTN to Internet Protocol (IP)-based communication. The emergence of different IP-based implementation environments, such as SIP [33] and WebRTC [37], makes seamless interworking critically important in order to avoid fragmenting the communication options for persons with disabilities.

Voice telephony interworks across different telephone systems and carriers. Information is needed as to what degree users of TTY and different forms of IP-based real-time text desire having similar levels of seamless interworking. Additionally, information is needed whether existing standards and implementations for real-time text and for interoperability between different environments are suitable and sufficient.

TTYs do not just offer a limited form of text communication, but also offer the ability to mix text and voice in the same call. However, the topic of being able to use voice and text together in the same call has received very limited research, and more information is needed on the perceived benefits, especially in light of the proliferation of IP-based text communication solutions.

In order to gather information on the questions mentioned above, a field trial was performed by the RERC-TA during 2015, by the RERC's three main collaborating organizations: Trace Center at the University of Wisconsin-Madison, the Technology Access Program at Gallaudet University and Omnitor in Sweden.

4 RTT Field Trial

Today there are several options for text communication that are widely used, both over the traditional PSTN network, the wireless voice network, and over IP networks. Communication tools are available both as pre-installed or installable applications in devices and as web pages using communication features in web browsers. The various services are easily accessed by anyone who has TTY, mobile phones, tablets and computers along with telephone or Internet connections. The available environments were chosen to reflect the following range of options: TTY on PSTN, RFC 4103 on SIP, and an experimental RTT on WebRTC using the Google Chrome browser. The trial also offered, depending on participant choice and preferred communication methods, the option to use only RTT, or a mix of RTT and voice. No video communications were offered as part of the trial.

The field trial was performed between March and October 2015, and involved both American and Swedish participants.

4.1 Goals

The overall goals were to collect information on how people with hearing loss are impacted by technologies for the TTY to IP transition, and to evaluate ways to use text conversation along with speech in peer-to-peer conversations. The study dealt with two main areas:

1. Evaluation of whether the ability to have simultaneous RTT and voice is desired and used by people who are deaf or hard of hearing and their friends and relatives.

2. Evaluation of user preferences for text conversation in direct peer-to-peer communication, by using the different technologies offered in the trial and evaluating how these relate to the individual communication habits and preferences outside of the trial.

The high level questions that guided the trial were:

1. Would the users value wider deployment of the ability to make calls between TTY and IP based text communication?
2. Would the users value wider deployment of web-based and app based text communication with possibility to call between these environments?
3. What mode for reception of text do the users prefer? Real-time-text or messages?
4. What mode for typing text during calls do the users prefer? Real-time-text or messages?
5. Do the users prefer that both parties are allowed to type simultaneously or rather use strict turn-taking?
6. Do users use the ability to use both voice and text during a call?
7. What features are important in a real-time text service?

- Would the users prefer to continue using the services under evaluation in the field trial?

4.2 Participants

There were three groups representing 3 probes that were done.

Group 1 consisted of 13 participants who were recruited from number of regions in Sweden. Participants in the study included hearing, hard-of-hearing and deaf individuals.The 10 Swedish participants composed pairs that had one person with a hearing loss who was unable to make voice-only calls, and at one hearing family member or close friend. There was one group with three participants, one of whom was hard of hearing and two of which were hearing. The six groups tested simultaneous RTT and voice telecommunications between SIP-RFC 4103 and WebRTC (websocket based T.140 RTT) for a period of two months.

Group 2 consisted of 34 participants who were recruited from the United States in the Washington DC metro area. The 34 participants were either deaf or hard of hearing individuals. Amongst them were also TTY users who were new to SIP-based technologies. The 34 participants in this group attended individual lab-based sessions where they evaluated ways of using simultaneous voice and text conversation between 1) PSTN (TTY) +voice and RTT-RFC 4103 + voice and 2) RTT-RFC 4103 RTT + Voice over IP on both sides.

Group 3 consisted of 2 participants recruited from the United States in the Washington DC metro area. The two participants evaluated interworking between three formats of real-time text; PSTN (TTY), WebRTC (websocket based T.140 RTT), and SIP (RFC 4103) over a period of 2 months.

In total there were 49 participants, with 36 in the United States and 13 in Sweden.

4.3 Technologies Used

Total Conversation (TC) clients and gateways have been developed in the RERC-TA project to enable accessible telecommunications, mainly by addition and interoperability of real-time text (RTT) in SIP-based communications. This communication technology also includes the HTML5 [38] web-based WebRTC [37] technology.

The gateways allow communication between legacy TTYs in the PSTN network and SIP-clients as well as SIP-clients and WebRTC-clients. The gateways can be chained so that TTY and WebRTC are interoperable.

More specifically, the following technologies were used in the field trial. The gateways and clients (except for TTY client devices) were developed within the RERC-TA project.

Clients:

- TTY – Ultratec (TIA 825-A [35]– text and voice)
- SIP-devices for real-time text and voice: Omnitor eCtouch in Android tablets (Traditional SIP for call control, RFC4103 [20] for text, G.711 [25] for voice)
- WebRTC [37] technology for real-time text and audio: Omnitor eCweb using Google Chrome web browser (WebSocket based SIP, RTT over WebSocket [10], G.711 for voice)

Gateways:

- TTY-SIP gateway for conversion between TTY in PSTN network and SIP solutions in the IP network, developed by Omnitor for the RERC-TA project based on the Asterisk open source switching system [9] (between TIA 825-A+analog voice and RFC 4103+G.711 voice)
- SIP-WebRTC gateway[10] for conversion between SIP protocol and HTML5 based WebRTC protocol. developed by Ivés for the RERC-TA project (WebSocket , G.711, RFC 4103)

Figure 1 below illustrates the technologies presented above and their relationships in the system.

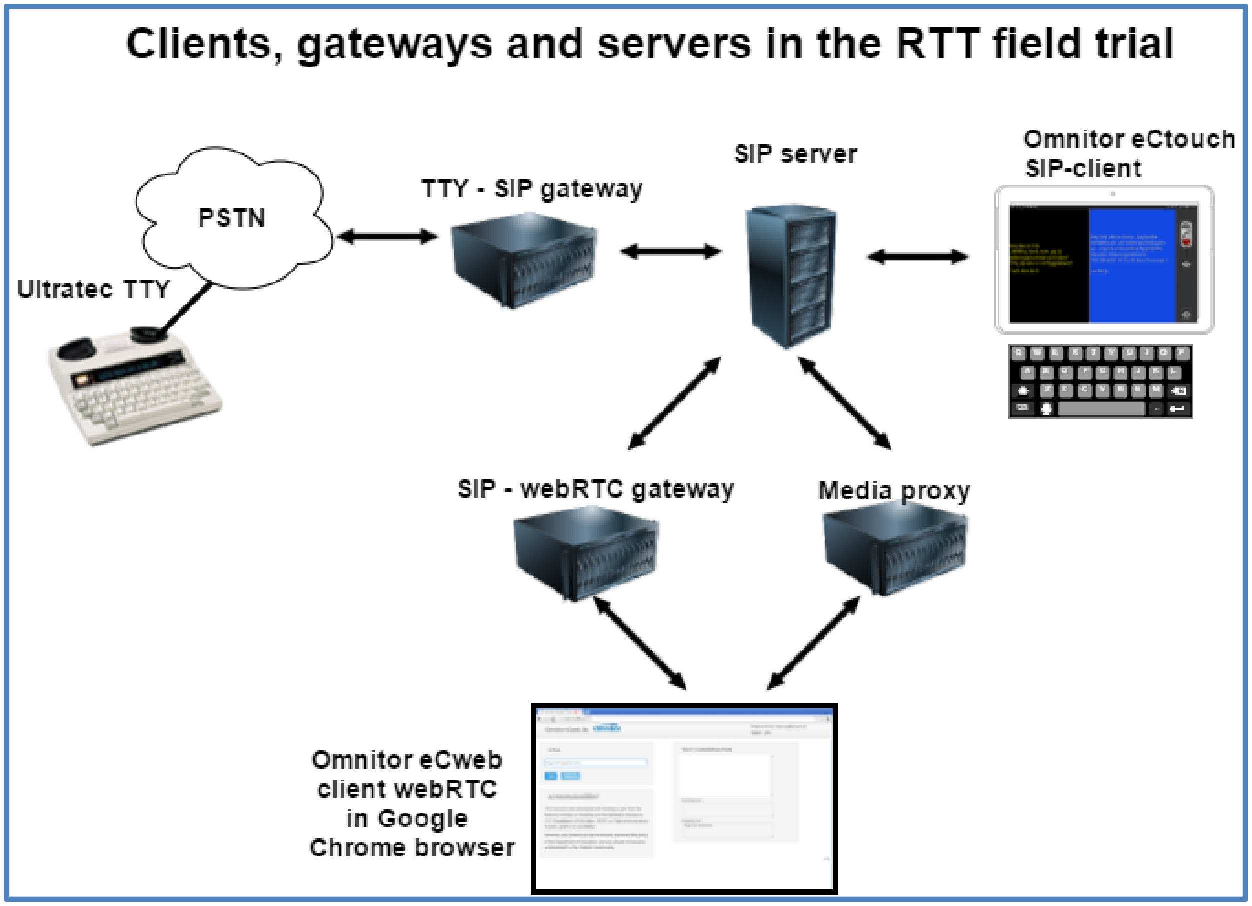


Figure 1: Overview of the technologies used in the field trial

The SIP-based installed app eCtouch for tablets looked like this when using an on-screen keyboard. Usually the users used an external keyboard for more convenient rapid typing.

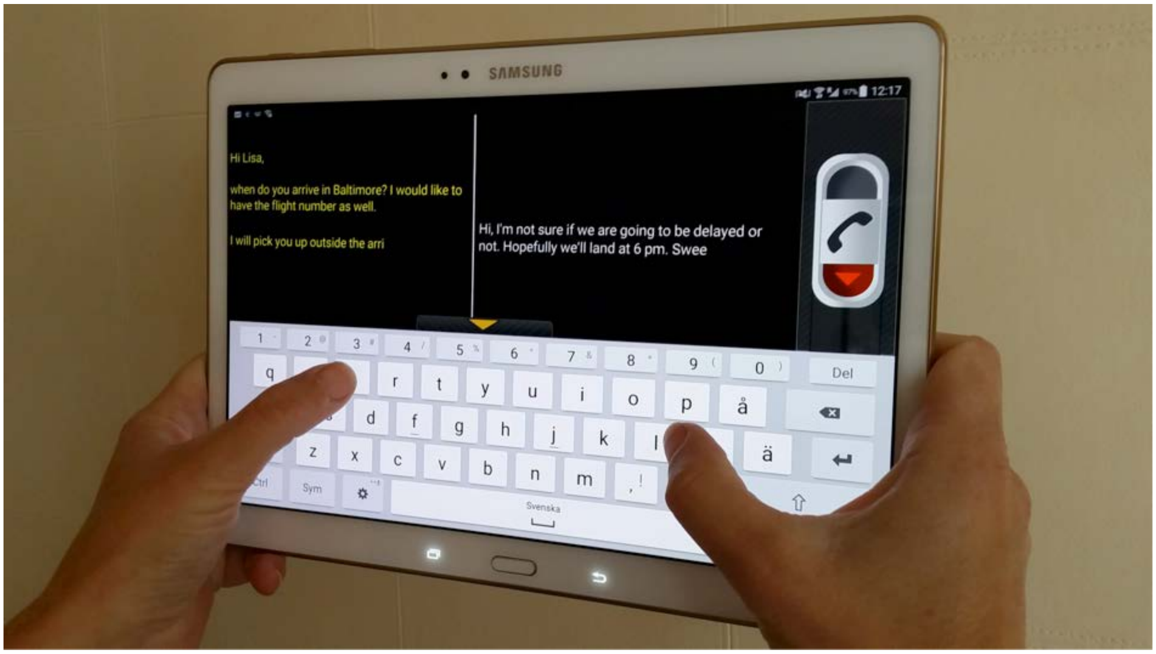


Figure 2: The RTT user interface of the eCtouch installed app

The WebRTC-based user interface looked like this when used in a web browser on a laptop computer.

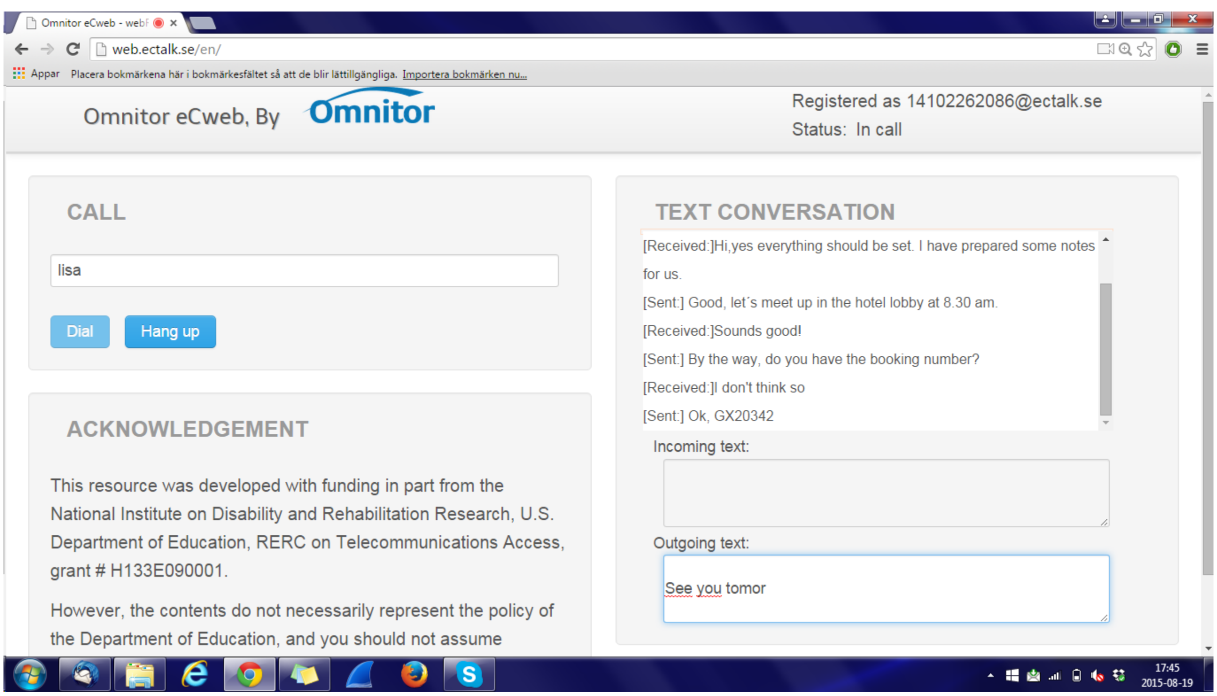


Figure 3: The RTT user interface of the eCweb WebRTC based solution.

4.4 Methods

The study was performed in Sweden and the USA, with slightly different approaches in each country.

The study in Sweden was a field trial that lasted for two months. This group looked at calls between SIP-RFC 4103 and web-based RTT (using WebRTC) solutions through an interworking function in the network. In this group there were 5 pairs and 1 triplet with a deaf person using SIP-RFC 4103 and one (or two) hearing persons using web-based RTT (using WebRTC) and communicating through an interworking function (gateway) in the network.

In the USA the main study focused on testing calls between PSTN-TTY and SIP-RTT (RFC 4103) through the use of an interworking function in the network as well as between different SIP-RFC 4103 solutions. In this study people were invited to individual one-hour testing sessions.

The side study in the US (2 subjects) provided the ability to have calls between TTY and web-based RTT (using WebRTC) solutions through an interworking function in the network that translated TTY to RFC 4103 and then RFC 4103 to WebRTC. In this study one a deaf TTY user and a deaf SIP user tested the technology in the field over two months

The participants in all groups (both countries) were asked the same questions on communication habits and also the same questions on their experiences and preferences after the trial or testing sessions were concluded.

4.4.1 Functional Elements Tested

The following functional elements were tested in the overall field trial and testing sessions:

- TTY text conversation versus simultaneous bidirectional real-time text
- The usage of text and audio together in direct telecommunication (in Sweden)
- Interoperability between legacy TTY on the PSTN network and real-time text over IP (SIP-based and WebRTC based)
- Using real-time text and voice over IP, both SIP-based and also interoperability between SIP- and WebRTC-based technologies

4.4.2 Trial Procedures in Sweden

The participants in the Swedish study were recruited by the local authorities responsible for prescribing communication aids. The group of interest was individuals who prefer to speak but are unable to use regular voice telephony because of hearing loss. The six participants with hearing loss were paired with one or more hearing partners of choice (typically family members or close friends).

The participants were equipped with tablets and software for simultaneous voice and real-time text, and the partners had interoperable technology in tablet or web-based format.

In the individual training sessions, the pairs and one triplet learned how to use the communication tool; how to make a call and to answer. The training also covered usage of real-time text along with speech for direct peer-to-peer communication, where the choice of combination is based on the user's needs and preferences.

After the training session the participants were asked to use the technology and make calls during a period of two months.

Each participant began by completing a questionnaire about their communication preferences. After the trials, participants were given a questionnaire about their experiences using the technology.

4.4.3 Testing Session and Trial Procedures for the Main study in the USA

Faculty, staff, and students from Gallaudet University were recruited through email blasts and notices posted around the campus. The 34 participants for the Main study at Gallaudet were recruited this way.

Participants were introduced to the equipment they would be using, and how to use it. For those who had never used a TTY, they were given a quick explanation of TTY communication conventions, such as the necessity of using GA (go ahead) to indicate that it was the communication partner's turn to type, and SK (stop keying) to indicate the end of a conversation.

Each participant began by completing a questionnaire about their communication preferences. Next, after explaining that during the one-hour trial the conversation pairs would be both making and receiving a call lasting approximately 10 minutes each, participants were given a list of conversation topics. These included topics such as favorite places to travel, questions about their pets, favorite foods, etc. Conversations were not limited to the list of topics, but served as a convenient starter for participants who were shy.

The original testing plan was to have half of the participants use WebRTC and half use a tablet with SIP based RFC 4103 communication and a wireless keyboard. However, technical issues prevented the use of WebRTC during the period of the intensive one-hour testing, so all 34 participants in that period used the tablet with the SIP based communication and wireless keyboard.

The participants alternated between initiating a call to either the TTY or a laptop with SIP communications software, and receiving a call from the TTY or the laptop. During the trial, each participant participated in two 10 minute conversations. The test leader was the conversation partner in these communication trial sessions.

After the two conversations, participants were given a questionnaire about their experiences using the technology.

4.4.4 Testing Session and Trial Procedures for the 2 person case study in the USA

Deaf and hard of hearing individuals were also recruited to participate in a study that would allow TTY users to communicate with friends and family members who do not use TTYs. Email advertisements were sent out to members of local deaf and hard of hearing advocacy groups such as the Hearing Loss Association of America, TDI, the Northern Virginia Resource Center for the Deaf and Hard of Hearing, and the Alexander Graham Bell Association for the Deaf and Hard of Hearing. This resulted in just two participants.

This pair used the communication setup for a two-month period. Both participants were deaf. One of the two participants was a long-time TTY user, and preferred that as her method for telecommunication. Her communication partner used a TTY before the trial only for the purpose of communicating directly with her. During the trial her partner used WebRTC based technology on a mainstream device using an interworking function in the network that translated TTY to RFC 4103 and then RFC 4103 to WebRTC.

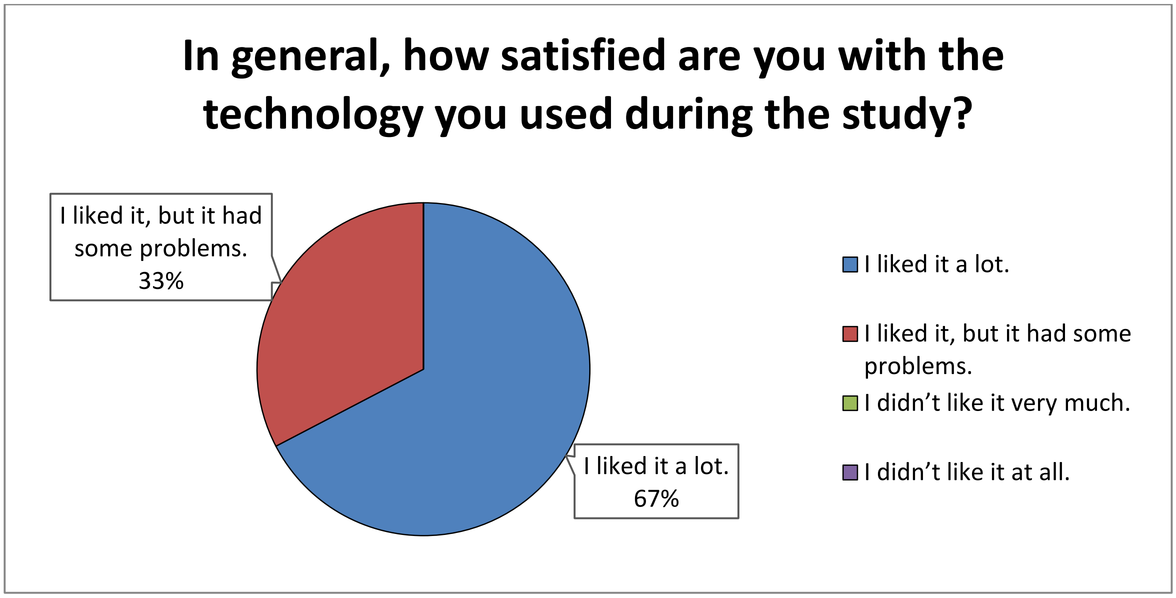
5 Results

Our main findings are presented below, and in detail in the following subsection.

5.1 Main Findings

5.1.1 Positive Stance on the Trialed Solution

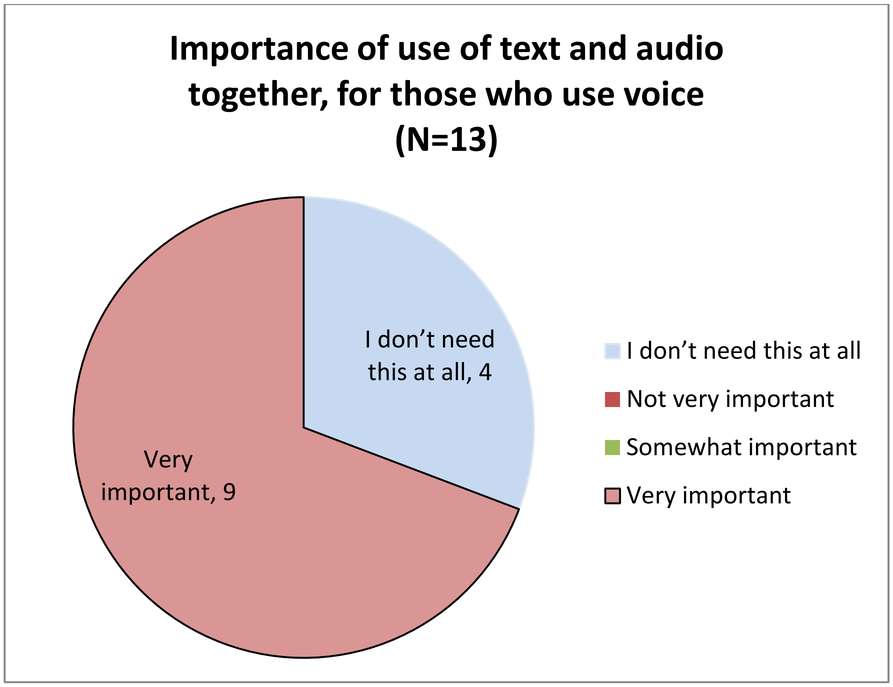
On the matter of overall satisfaction with the technologies that were offered in the trial, the participants were positive; 67% chose “I like it a lot”, 33% chose “I like it but it had some problems”, 0% chose “I didn’t like it very much” or “I didn’t like it at all”. The results were consistent across the groups. The most common comments from the 33% were that they wanted a mobile solution rather than a tablet or computer based solution.



Group	Liked a Lot	Liked w/Prob	Didn't much	Didn't at all
Sweden 13	6	7	0	0
USA 34	27	7	0	0
USA 2	0	2	0	0
Total	**33**	**16**	**0**	**0**
%	**67%**	**33%**	**0 %**	**0%**

5.1.2 Use of Voice and Text Together

The thirteen Swedish participants used speech in their calls during the trial. Of those, 9 out of the 13 answering the question reported that it was very important to being able to use speech in a call where simultaneous audio and real-time text was provided.



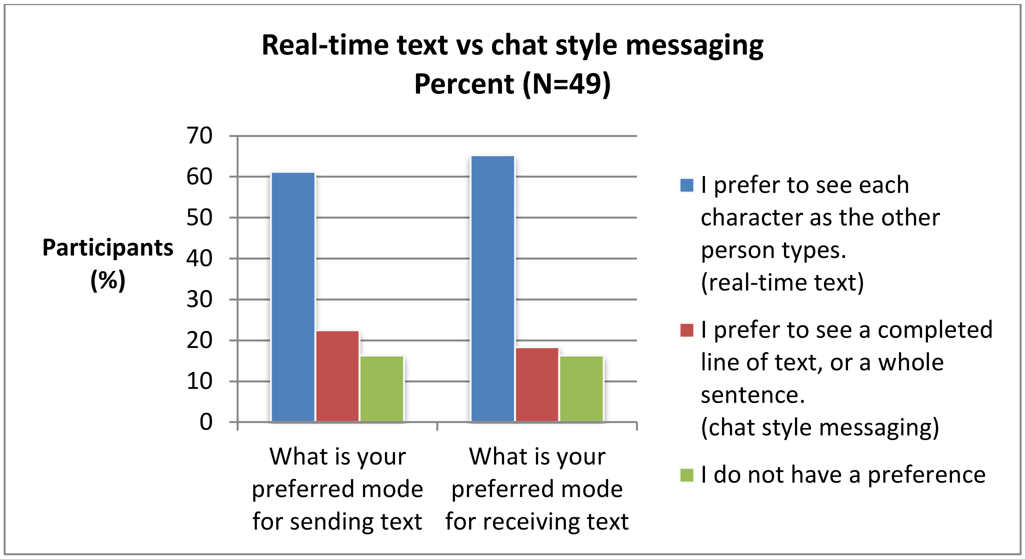
The comments were on the ability to speak one way and type the other direction, also by using text-supported speech. The conversation progressed more fluidly and naturally - compared to being unable to hear one party or both using text only.

5.1.3 Preference of Real-Time Text over Chat-Style

Regarding the ways to use text in direct communication, the users shared their experiences from using real-time text and chat style conversation.

Two-thirds (65.3%) of the participants in the study preferred real-time text for receiving text. (18.4% preferred chat style and 16.3% did not have any preference).

For sending text, slightly less than two-thirds (61.2%) preferred to send real-time text. (22.4 % preferred chat style, and 16.3% did not have any preference).



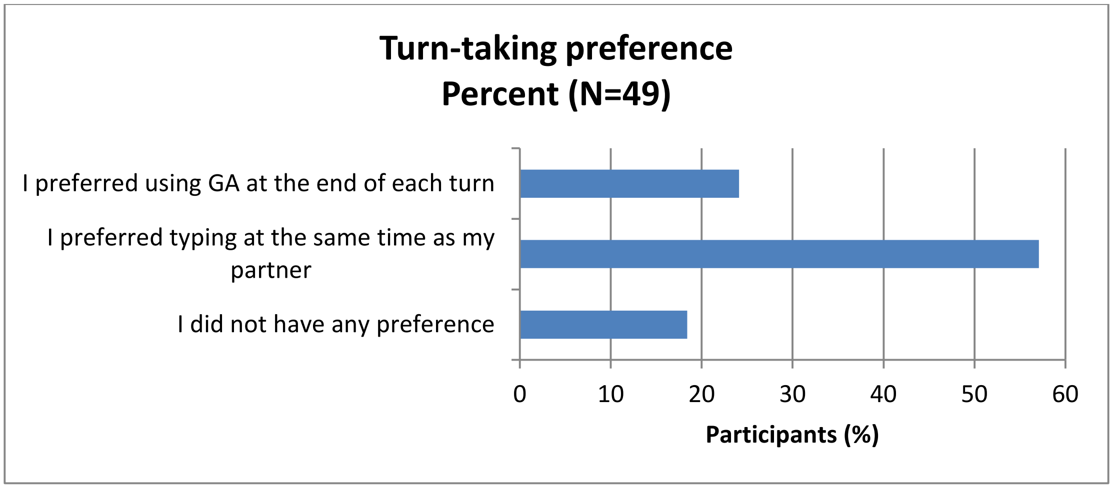
Sending - Receiving

Group	Char by Char	Line/Sentence	No Preference
Sweden 13	8 – 7	1 – 3	4 – 3
USA 34	20 – 23	10 – 6	4 – 5
USA 2	2 – 2	0 – 0	0 – 0
Total	**30 – 32**	**11 - 9**	**8 – 8**
%	**61.2% - 65.3%**	**22.4 – 18.4 %**	**16.3% - 16.3%**

5.1.4 Turn-taking in Text Conversation vs Simultaneous Typing

On the question on turn-taking according to the TTY convention using GA at the end of each turn or being allowed to type at the same time as the conversation partner, over half (57.1%) chose simultaneous typing over strict turn-taking. A quarter (24.5%) preferred TTY-style and 18.4% had no preference. This question has historical background in that the TTY is by its technical design limited to communicate in text only in one direction at a time. The turns must therefore be signaled by turn-taking tokens in the text. "GA" (for "go ahead") is used for that purpose. The question is intended to investigate if the participants see value in maintaining this method even with RTT, when no technical limitation prevents simultaneous typing. On another question about experienced difficulties with turn-taking, 49% had no difficulties and 51% had difficulties sometimes or rarely. No one reported continuous difficulties with turn-taking.

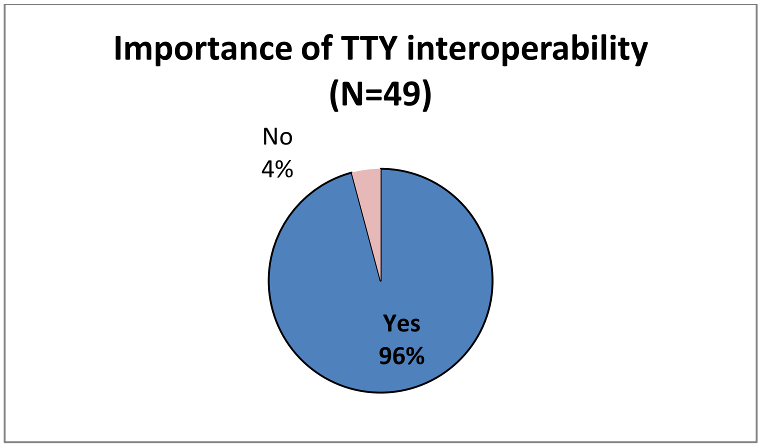
The majority of participants either preferred not to use traditional turn-taking conventions or had no opinion. This indicates that there is no clear need to maintain turn taking conventions in IP technologies.



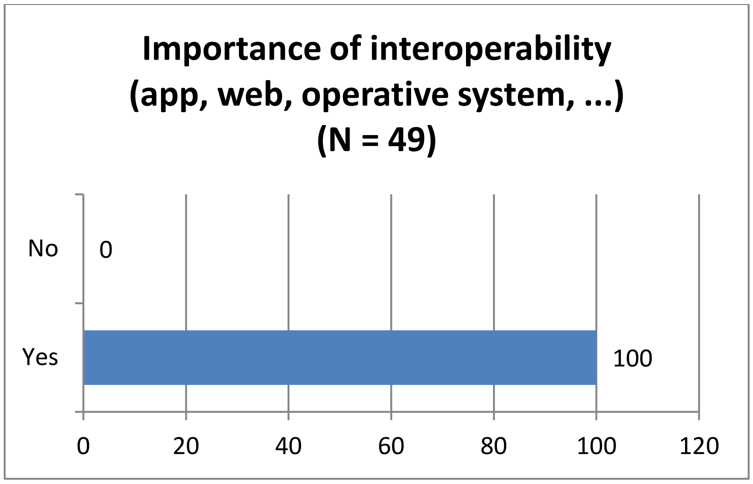
Group	GA at end of Line	Same Time	No Preference
Sweden 13	2	6	5
USA 34	9	21	4
USA 2	1	1	0
Total	**12**	**28**	**9**
%	**24.5%**	**57.1%**	**18.4%**

5.1.5 Interoperability and Access

Ninety-six percent agreed on the importance of being able to interconnect TTY and the IP-based real-time text conversation. The most common rationales offered by the participants were (a) not to leave TTY users behind, and (b) to allow accessibility and inclusion of people who are deaf or hard of hearing without high-speed Internet access in rural areas.



All of the participants (100%) affirmed the importance of being able to call between different technologies (TTY, IP, WebRTC) and environments.



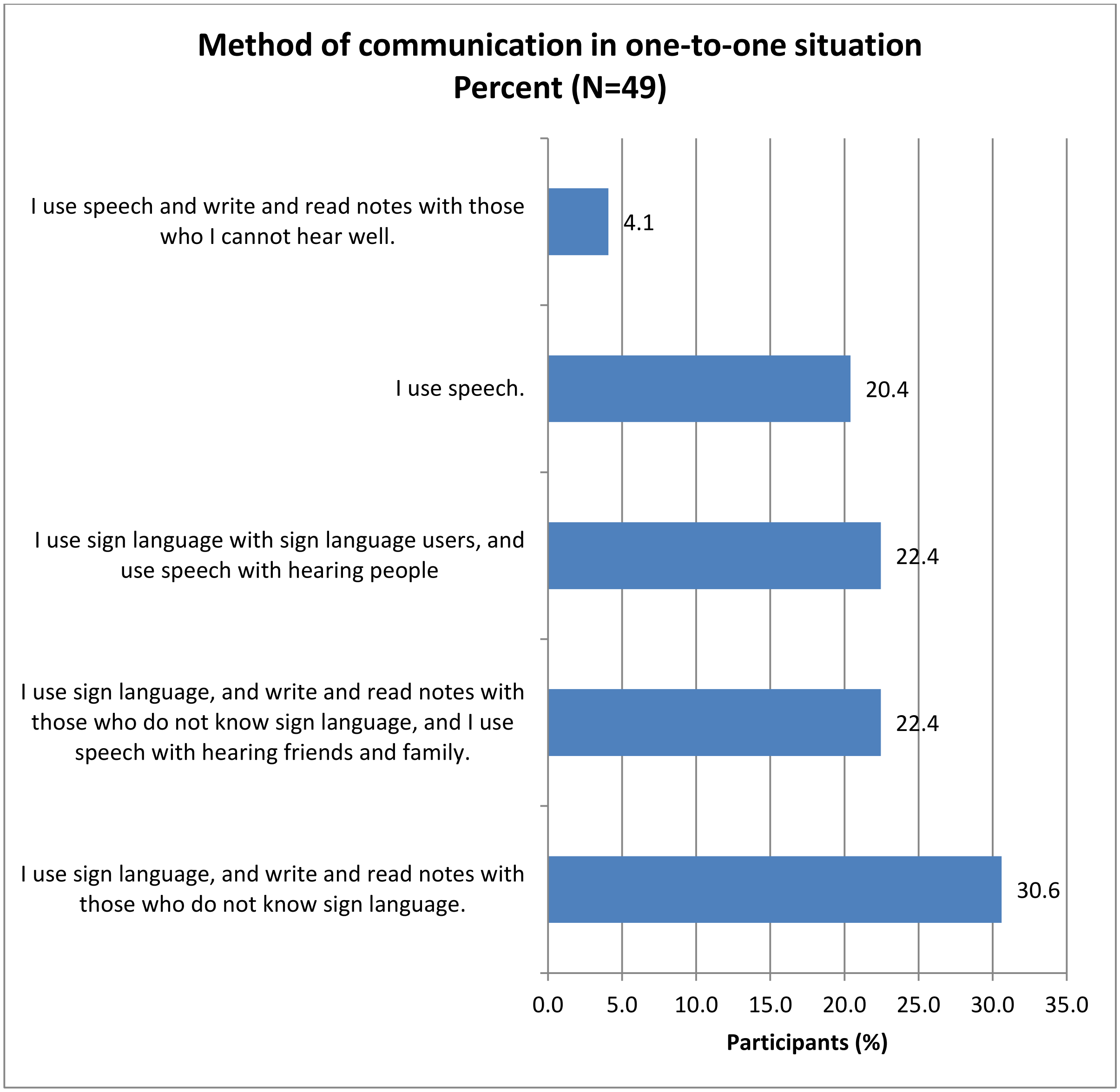
Typical comments were that every person is unique with different needs and preferences; and that expansion of the technologies should be allowed rather than limiting choices to a few. It should be natural to provide users with more flexibility as well as functional equivalence.

5.2 Detailed Responses

Below are the responses from the 49 participants in the trial based on two questionnaires; first the background questions on communication habits, abilities, and preferences; second the questions after the trial on the participants' experiences and technology preferences.

5.3 Background Questions before the Trial

Q1. What is your typical method of communication in a one-to-one situation?



Forty-nine participants answered. About 30% of the participants use sign language complemented with reading and writing. About 22% also use speech with hearing friends and family. About 22% answered that they use sign language and speech. About 20% are mainly speech users and 4% use mainly speech but writing and reading when it is not possible to use speech.

Group	I use speech and write and read notes with those who I cannot hear well	I use speech	I use ASL with ASL users, and use speech with hearing people	I use ASL, and write and read notes with those who do not know ASL, and I use speech with hearing friends and family	I use ASL, and write and read notes with those who do not know ASL
Sweden 13	2	10	1	0	0
USA 34	0	0	10	11	13
USA 2	0	0	0	0	2
Total	**2**	**10**	**11**	**11**	**15**
%	**4.1%**	**20.4%**	**22.4%**	**22.4%**	**30.6%**

Comments:

Typical comments from the users:

- The level of success depends on who I am speaking with.
- Sign-supported speech helps.
- Using speech only with family members.
- I am avoiding speech due to risk for misunderstanding.

Analysis:

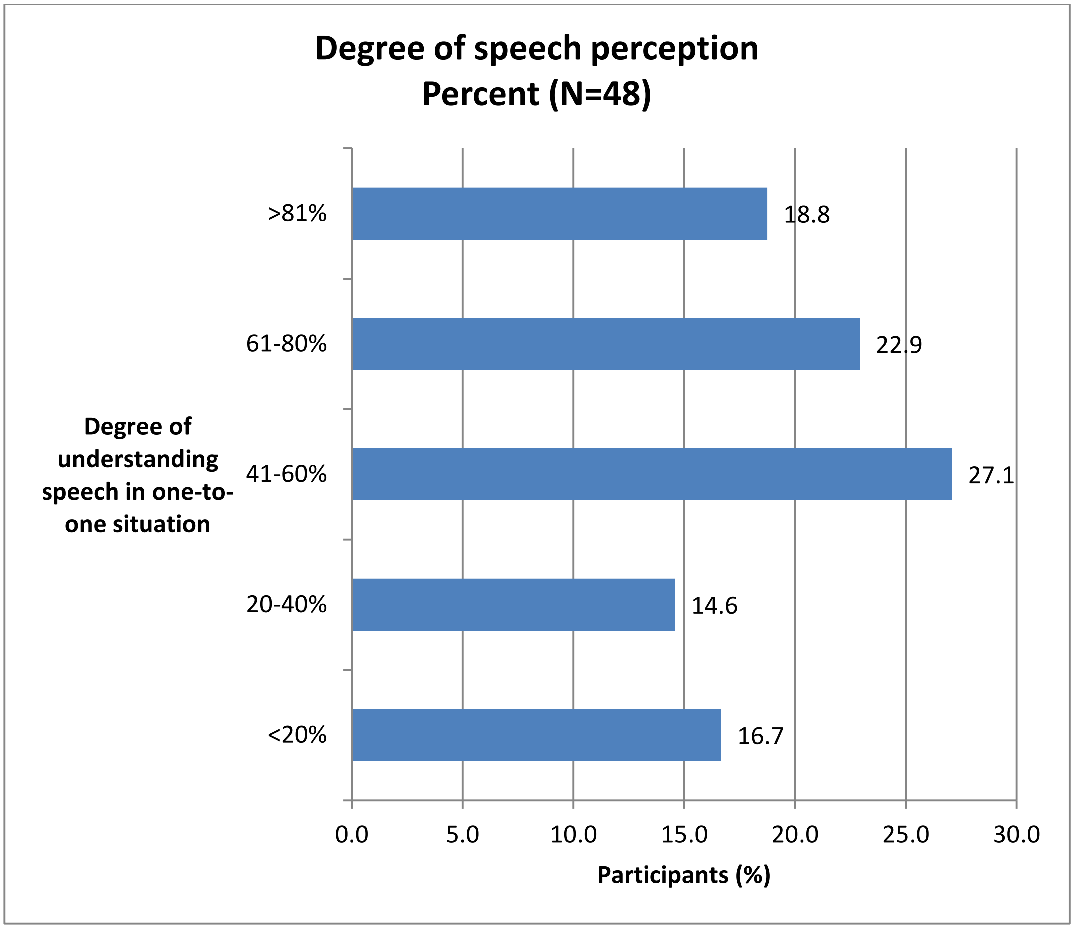
Most of the participants are referring to themselves as deaf or hard-of-hearing as 75% are using sign language with signing peers.

About 53% of the participants are deaf or hard-of-hearing and use speech with hearing people, in particular with hearing friends and family.

The group also consists of 7 hearing participants (15%).

A clear majority are using multiple ways to communicate, selecting the method that suits best for each situation.

Q2. In a face-to-face conversation, in the best possible setting, (a quiet, well-lit room) using hearing devices that you typically use if any (hearing aids and/or cochlear implants), approximately what percentage of the other person's speech do you understand? (Maybe add the descriptions here for each response?)



The answers are quite evenly distributed between 5 alternative ranges of understanding from 0-20% of the speech understood to 81-100 % speech understood. The 41-60% percent range has however collected noticeably more answers: 27.1%, while the other classes had between 14 and 23% of the answers each.

Group	>81%	61-80%	41-60%	21-40%	Less than 20%
Sweden 13	6	4	2	1	0
USA 34	3	7	11	6	6
USA 2	0	2	10	0	0
Total	**9**	**13**	**23**	**7**	**6**
%	**18.8%**	**22.9%**	**27.1%**	**14.6%**	**16.7%**

Comments:

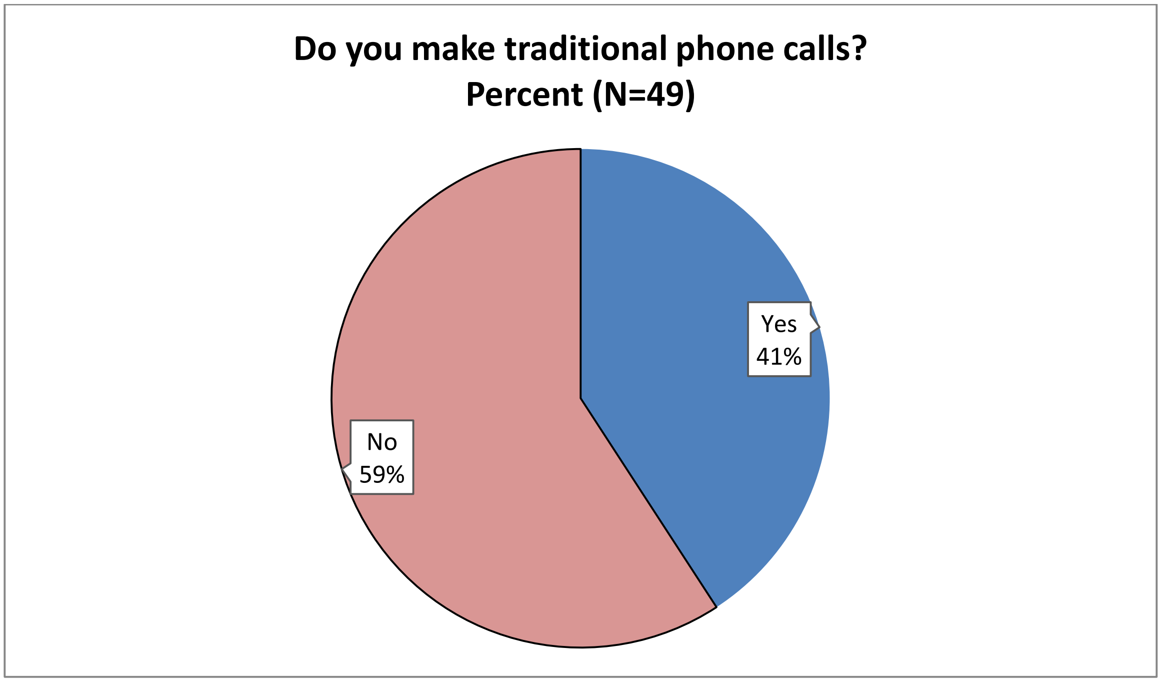
A majority of the participants commented and stressed that the ability to understand the other party is variable; it depends on whether the person is familiar, the topic, existence of facial hair, articulation, accent, and also the surrounding.

Analysis:

Seven of the participants were fully hearing, and 2 participants with hearing loss felt that they could communicate by voice without problem.

The question was a bit hard for the non-hearing participants; most of them added comments on the situation dependence. In general most of the participants felt that they understand approximately half of a regular everyday conversation in favorable conditions.

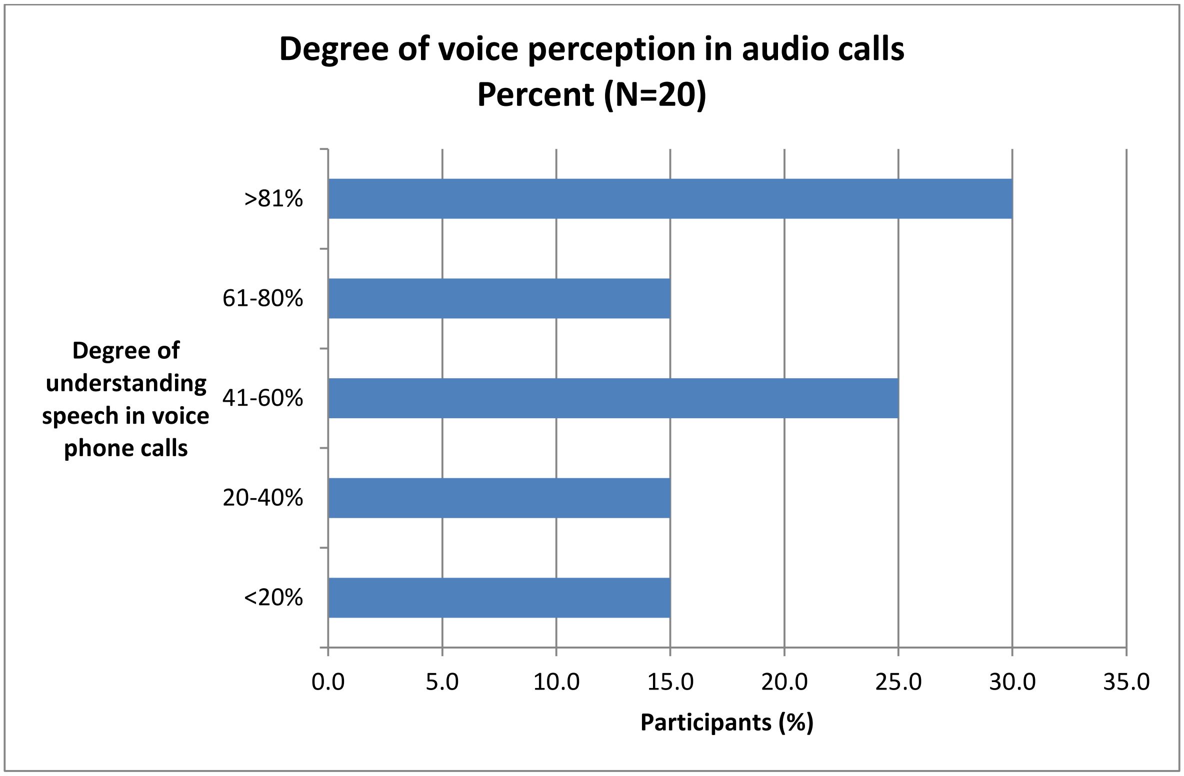
Q3. Do you ever use a cell phone or a landline phone to make voice calls to friends and family?



Of the participants, 41% responded that they make traditional phone calls, while 59% do not.

Analysis: The 15% hearing participants clearly are expected to use traditional phone calls. An additional 26% among the participants also answered that they make traditional voice phone calls. This implies that regular voice telephony is not an accessible alternative for a dominating part of the deaf and hard-of-hearing participants.

Q4. If yes, approximately what percentage of the person's speech do you understand through listening only (without the help of captions or lip-reading through video?)



About half of the participants use voice telephony. One third of them were hearing conversation partners.

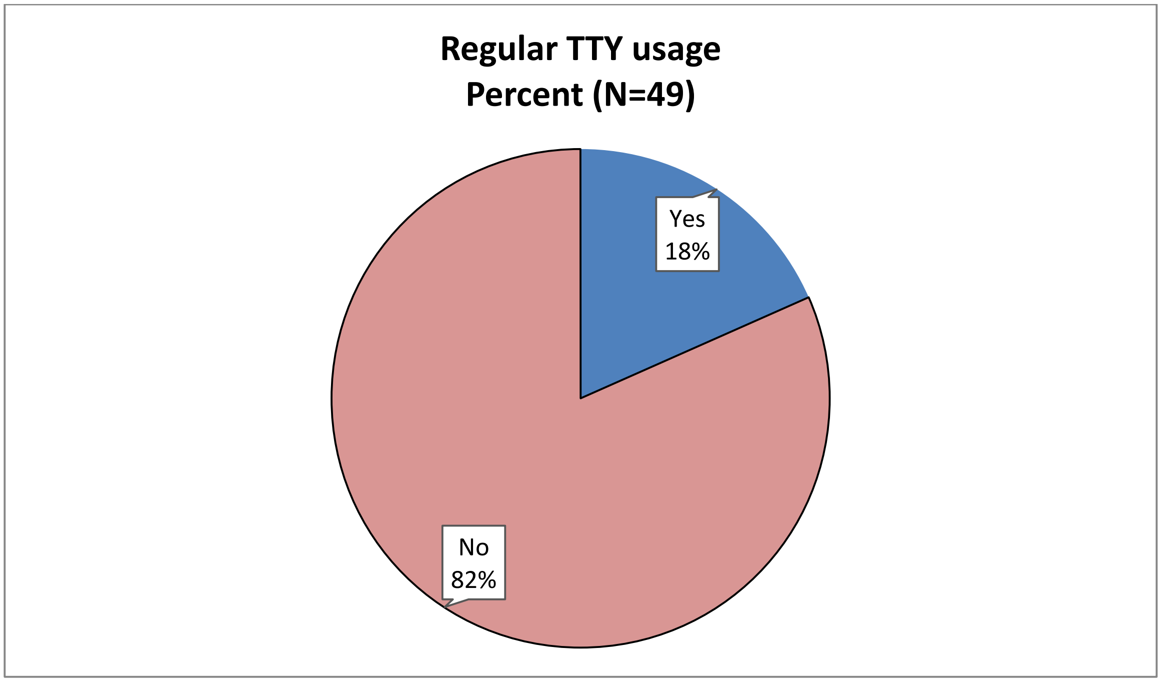
Comments:
Those with hearing loss who use voice telephony, remarked that they only had conversations with the immediate family and in an emergency. They also made very short calls, some only containing yes-no questions.

Group	>81%	61-80%	41-60%	21-40%	Less than 20%
Sweden 13	6	1	1	1	3
USA 34	0	2	4	2	0
USA 2	0	0	0	0	0
Total	**6**	**3**	**5**	**3**	**3**
%	**30%**	**15%**	**25%**	**15%**	**15%**

Analysis:

For many persons with hearing loss, voice telephony is generally not an option unless in an emergency situation or with close friends or family.

Q5. Do you regularly use a TTY to communicate with friends, relatives, and businesses?



Eighteen percent of the participants use a TTY regularly (or corresponding textphone in Sweden, also classified as TTY for the purpose of this study).

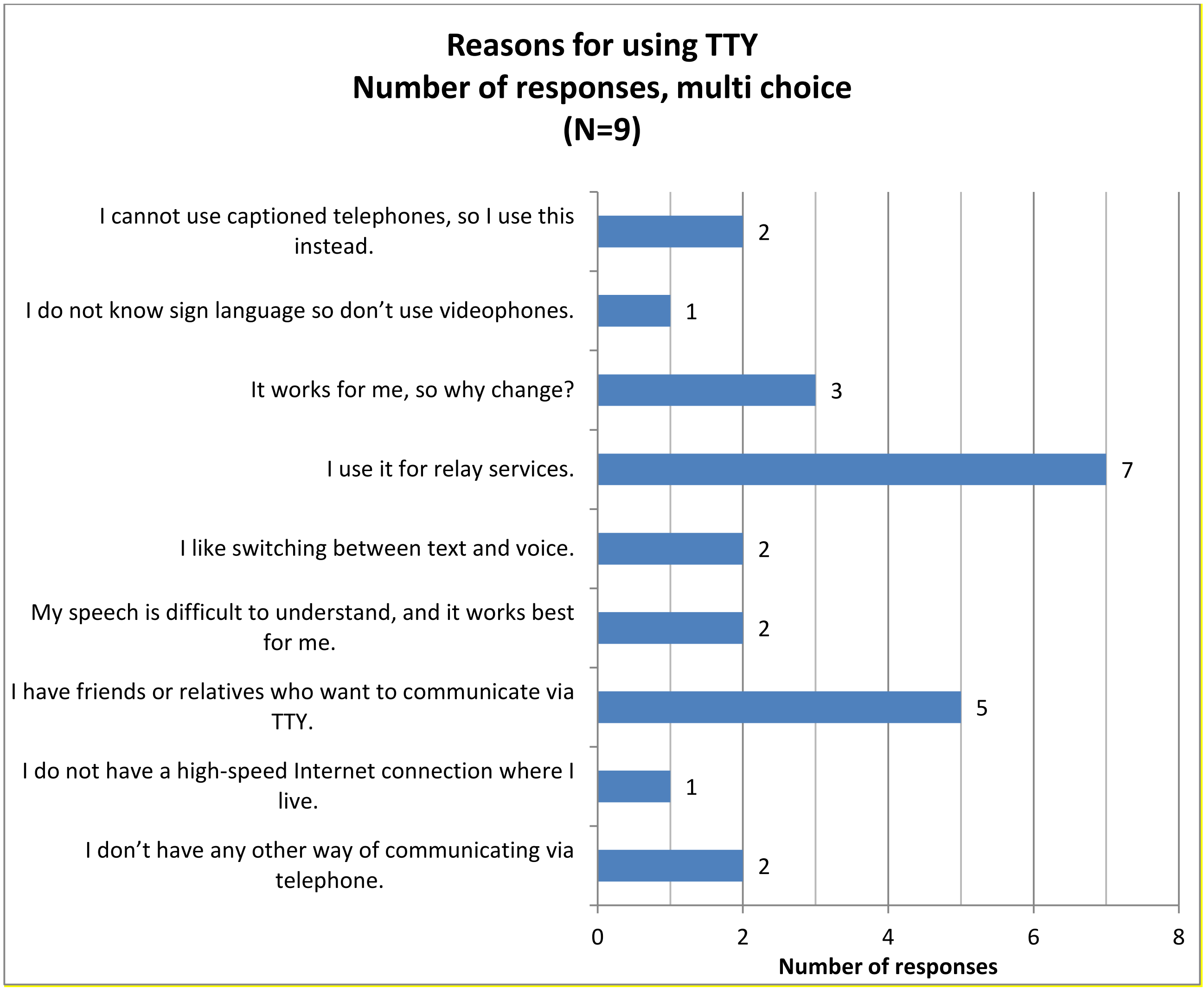
Comments:

Some participants commented inline that they are keeping the TTY for communication with older generations or when in rural areas without high speed Internet access and therefore don't use it regularly. Yet more participants remarked that they were using TTYs in the past before the existence of IP based communication technologies but not nowadays.

Analysis:

Nine participants out of 49 use a TTY regularly. The number of people using a TTY for their main phone conversation is decreasing in favor of newer technologies. TTYs are used when the users are left with few options such as no access to high-speed Internet or when keeping in touch with TTY-only-users.

Q6. If yes, please explain why. (Check all that apply.)



This is a multi-choice question. The few TTY users had many reasons to keep using them.

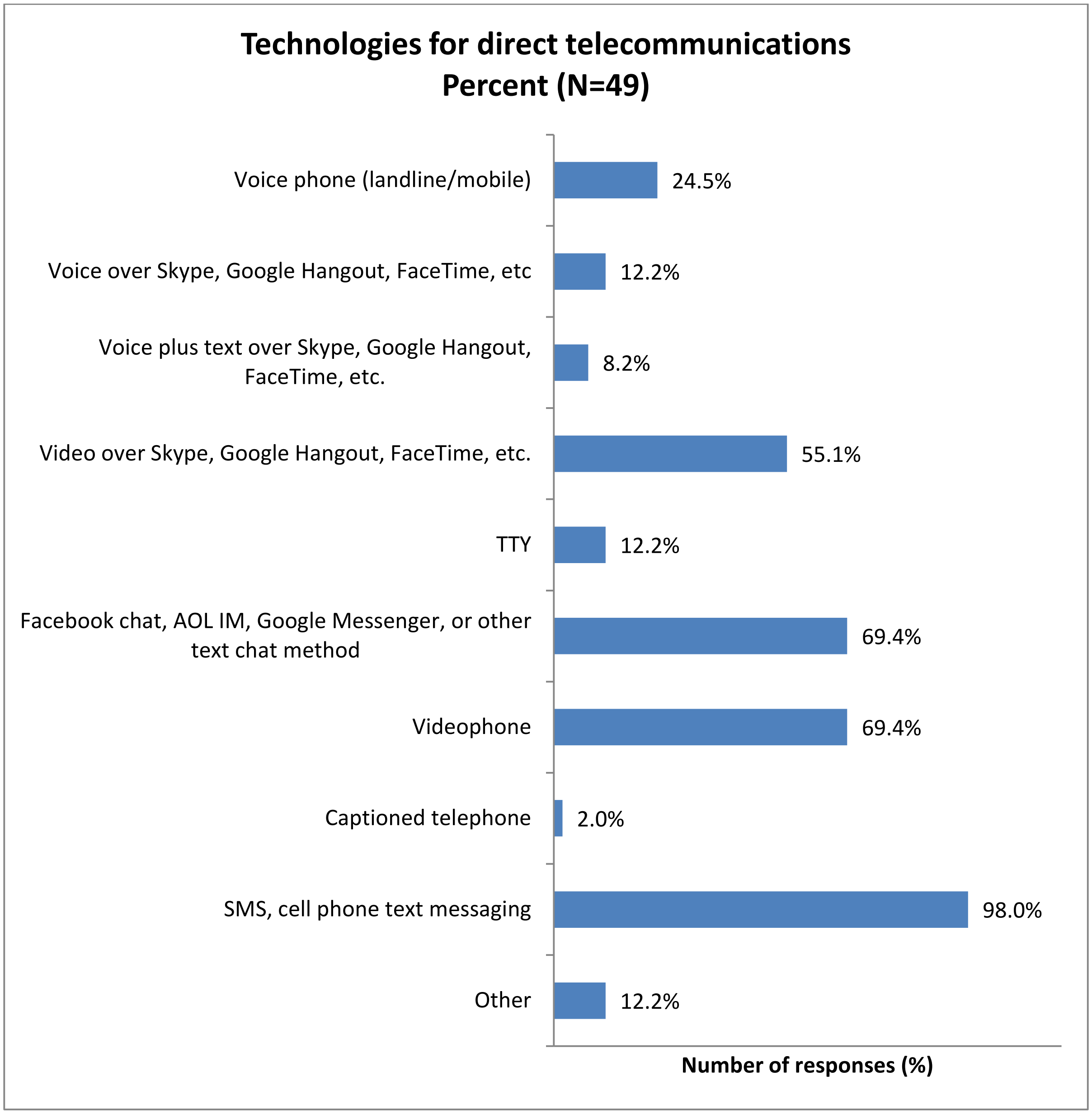
Comment:

Two comments were on the use of TTY to communicate with older friends so that they would not be left behind. Two users added that they wanted to use captioned telephone.

Analysis:

There were very few responses on TTY usage as there were only nine reported TTY users, as per the previous question. The most frequent reason for keeping a TTY was using relay services. Some other participants were using it with specific people or when they did not have Internet access but PSTN. In Sweden there is no captioned telephone service; two of the participants who use a TTY wish to have access to such service.

Q7. What do you typically use for direct telecommunications to communicate with friends, family, or businesses?



The answers were spread among a number of different communication technologies. Multiple answers were allowed.

Comment and observation:

One of the participants was computer illiterate and also unable to use voice telephony, thus the spouse handled the phone calls (interpreting by voice/lip-reading). One other participant also usually asks someone to handle the calls.

Analysis:

Practically everyone used SMS; it is ubiquitous and available in both feature phones and smartphones. SMS is also interoperable and “built-in” to the devices contrary to other texting apps. Yet nearly 70% regularly use text chat through various services.

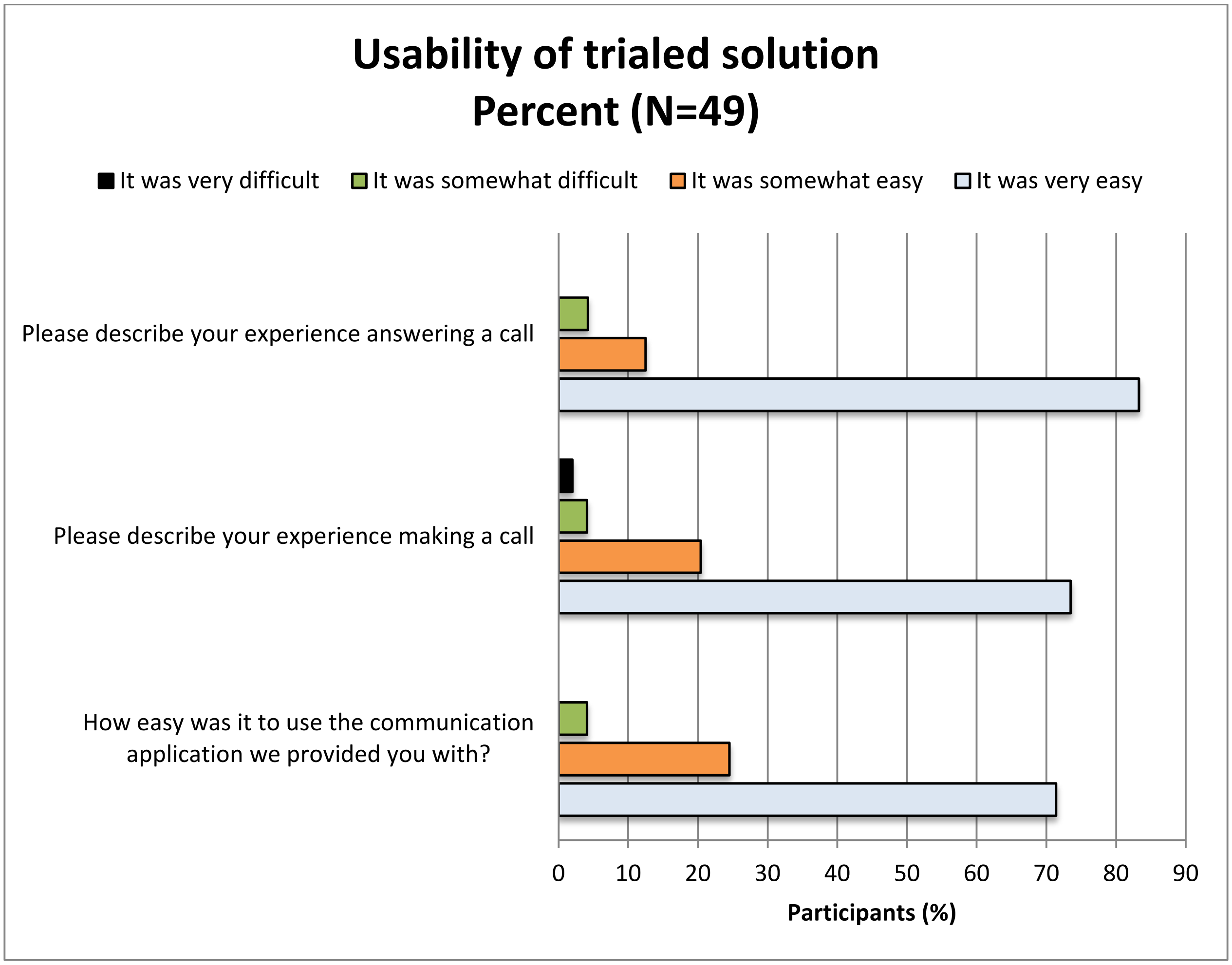
In addition two thirds of the participants also used videophones. This is not surprising since 75% of the participants identified themselves as sign language users.

5.4 Questions Asked after the Trial/Testing Sessions

Q8. How easy was it to use the communication application we provided you with?

Q9. Please describe your experience making a call

Q10. Please describe your experience answering a call



Three usability factors are collected in this chart: How easy it was to answer a call, to make a call and how easy the solution was to use in general. For all, the dominating answer was that it was very easy (83%, 73% and 71%). Somewhat easy got 13-24% responses, while the other alternatives expressing lower usability only got a few percent of the answers.

Comments:

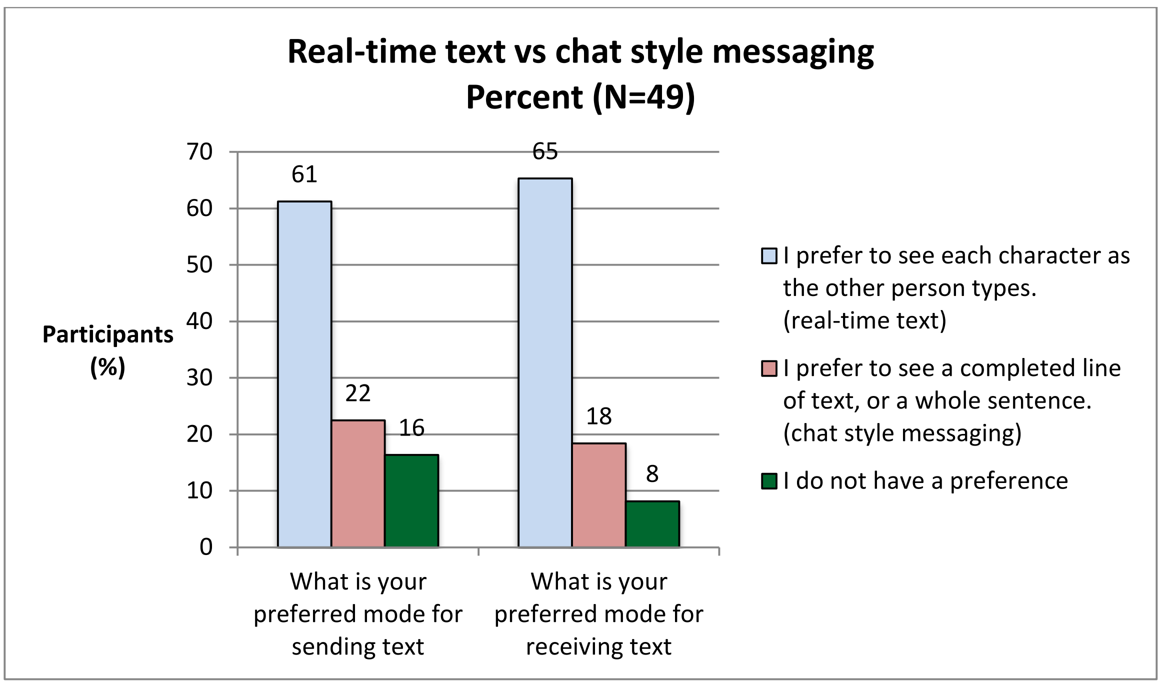
Participants commented positively on the integrated phone book in the eCtouch app and the convenient way of making calls compared to TTYs.

Other comments included the desire of more mobile solutions (smartphones) as it would have increased the frequency and possibility to perform calls, in particular in the field trial that lasted two months. This also included the need of some alerting devices to be reachable. The web-based solution was not very user friendly with its small text windows.

Analysis:

The overall experience on the usability of the trialed solution is that it was easy or very easy to use. 5% thought it was somewhat difficult, related to the lack of alerting devices when receiving calls and the other side not responding.

Q11. What is your preference for sending/receiving text?



The preferences for real-time text versus messaging style text were asked divided in the typing situation and the reading situation. The responses were very similar for these situations, with 61-65% preferring real-time text mode, 18-22% preferring the message mode and 8-16% having no preference.

Comments:

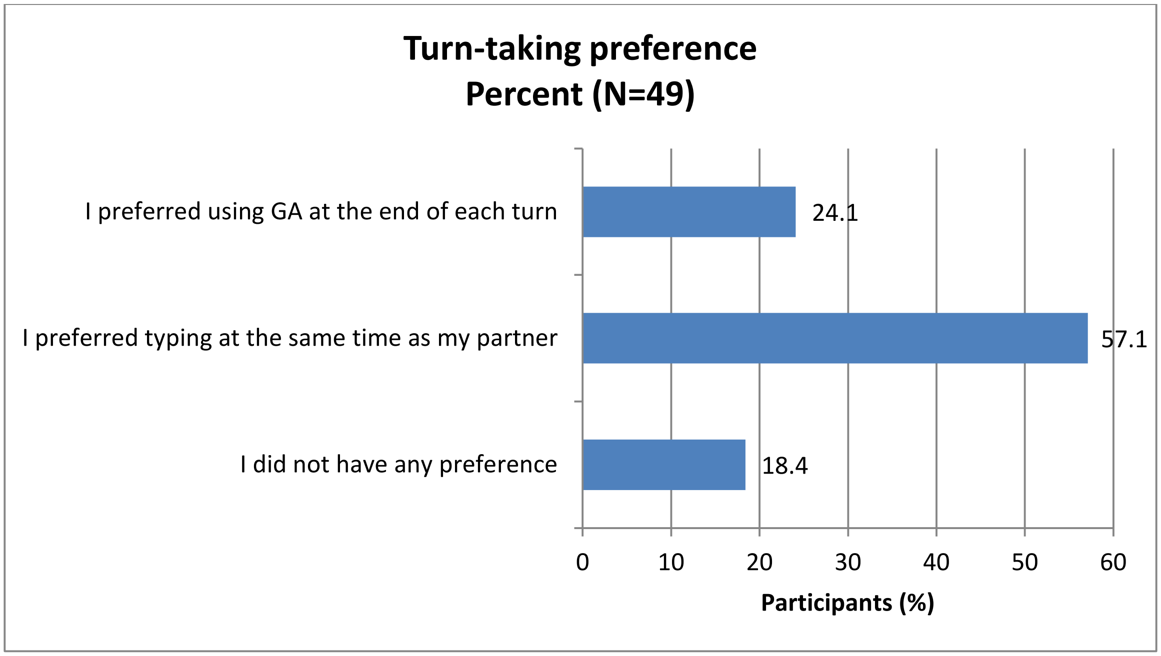
Participants expressed the following opinions:

- Real-time is easier and more natural.
- RTT is faster as one is able to interrupt with corrections and comments without waiting for a complete sentence.
- It is great that both can see what is typed right away.
- But sometimes I get unsure when the other side will be finished without indication.

Analysis:

A majority of the participants preferred real-time text both for sending and receiving text in a text conversation as it speeded up the conversation and allowed interruption. One-fifth wanted some indication of when the other side was done with the texting.

Q12. If you did NOT use TTYs to make your calls, during text-based calls, did you prefer using TTY conventions (using GA at the end of each turn) or did you prefer more natural conversation where you could type at the same time as your partner?



Nearly twenty-five percent of the participants preferred using the strict turn-taking habit with "GA" from TTY usage, while 57.1% preferred to type at the same time without specific turn-taking conventions, and 18.4% had no preference. On another question about experienced difficulties with turn-taking, 49% had no difficulties and 51% had difficulties sometimes or rarely. No one reported continuous difficulties with turn-taking.

Comments:

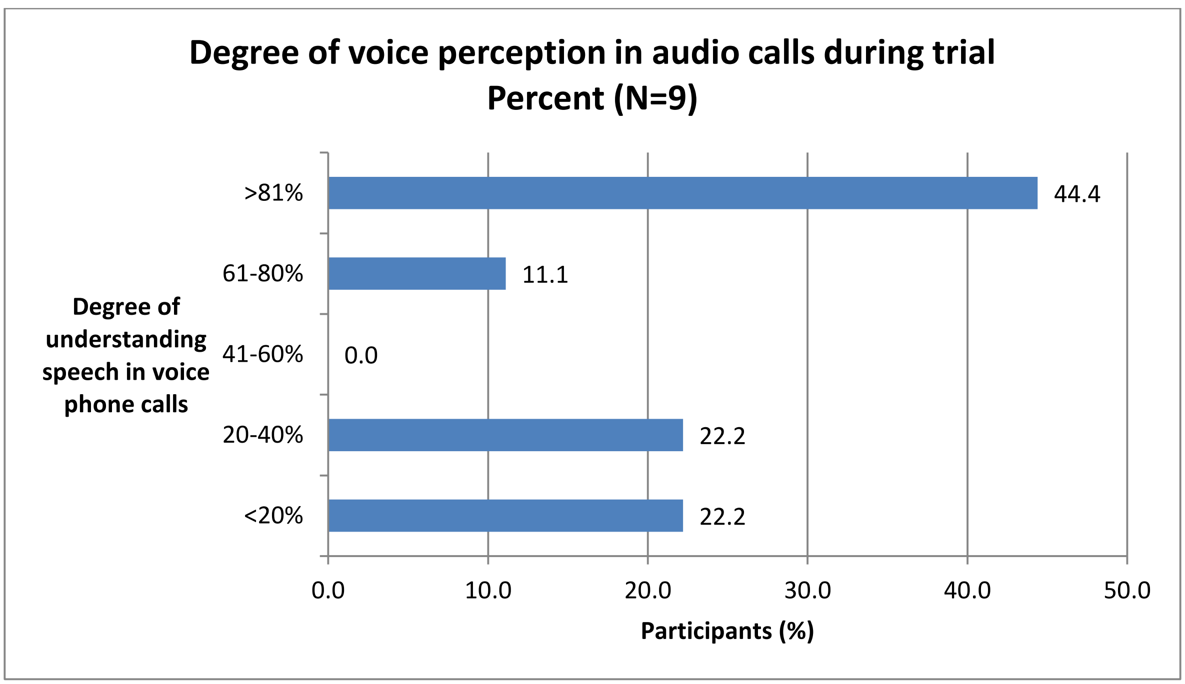
Some of the participants were used to waiting for their turn using TTY convention with GA at the end of a turn; a part of them later changed preference. A few still preferred TTY style as they felt awkward if they could go ahead typing. Others appreciated the possibility to type at the same time as it speeded up the conversation.

Yet another group was only using text one way as their partners were using speech the other way, thus having no preference.

Analysis:

The preference is closely connected to whether the user is still using a TTY or not. The hearing partners who typed while listening tend to have no preference as they did not use turn-taking in text. The dominance of preference for not using traditional turn-taking conventions indicate that there is no clear need to maintain these conventions in IP technologies.

Q13. Did you use speech during the calls? If yes, approximately what percentage of the other person's speech did you understand?



The judgment for degree of understanding speech was divided in five ranges 0-19, 20-40, 41-60, 61-80 and over 81%. Only 9 participants answered this question. Of them, 44.4% answered that they understood over 81%, and the remaining few responded with lower understanding rates.

Comments:

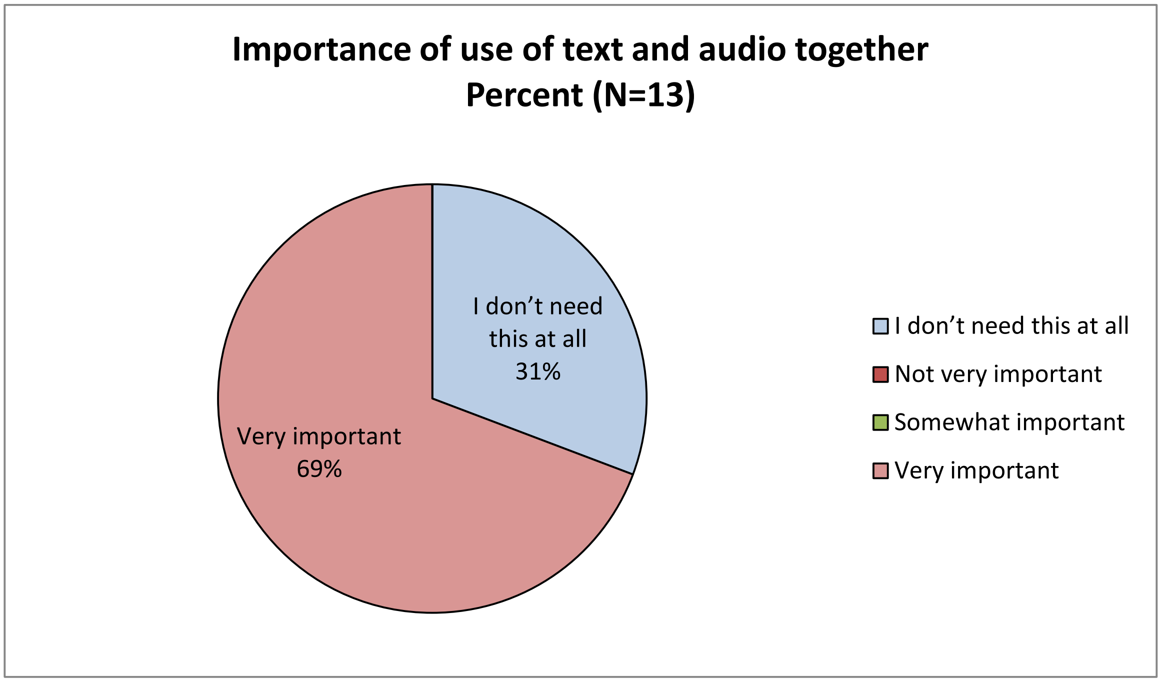
Participants expressed the following:

- Synergy effect if text was added to voice, then the level of perception increased to almost wholly intelligible.
- I appreciated that my partner was able to speak; the conversation became more fluent as he didn't have to type.

Analysis:

Seven of the participants, who used speech in the call, are hearing and typed back to their partners with hearing loss. In some cases they also used voice along with text. Two of the participants remarked that text in addition to voice increased the quality in the communication, according to the level of perception as well as the flow in the conversation.

Q14. If you used speech during the call, how important was this feature to you?

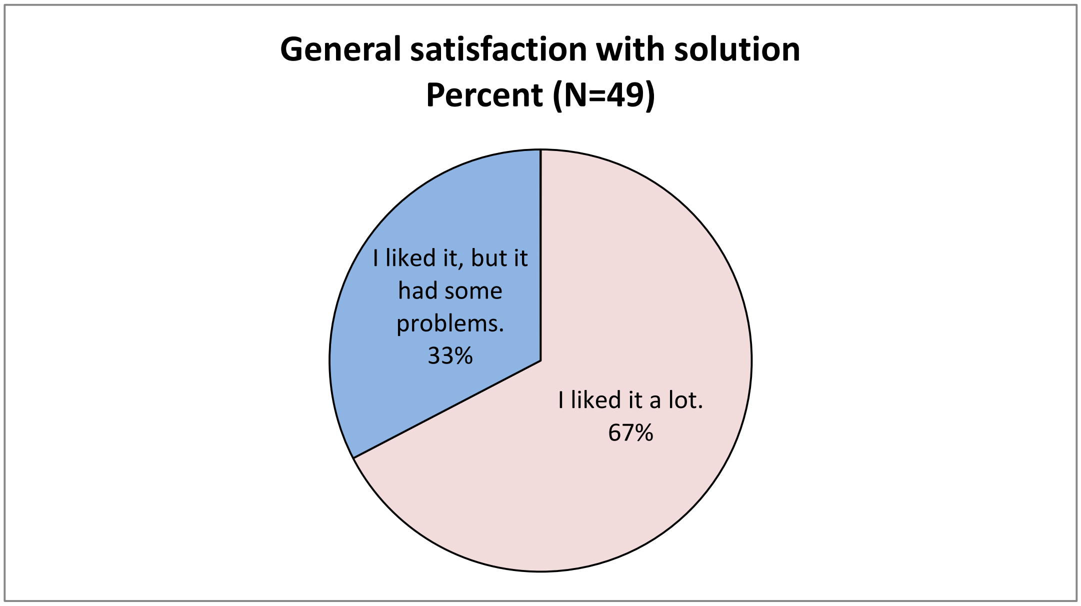


Sixty-nine percent of the participants answering this question saw it as very important to be able to use text and audio together in the call, while 31 % answered that they do not need the feature at all.

Analysis:

The possibility to use text combined with audio is experienced as a very important function in real-time communication.

Q15. In general, how satisfied are you with the technology you used during the study?



Sixty-seven percent of the participants answered that they liked the solution a lot, while 33% liked it but experienced some problems. No lower grades of satisfaction were reported.

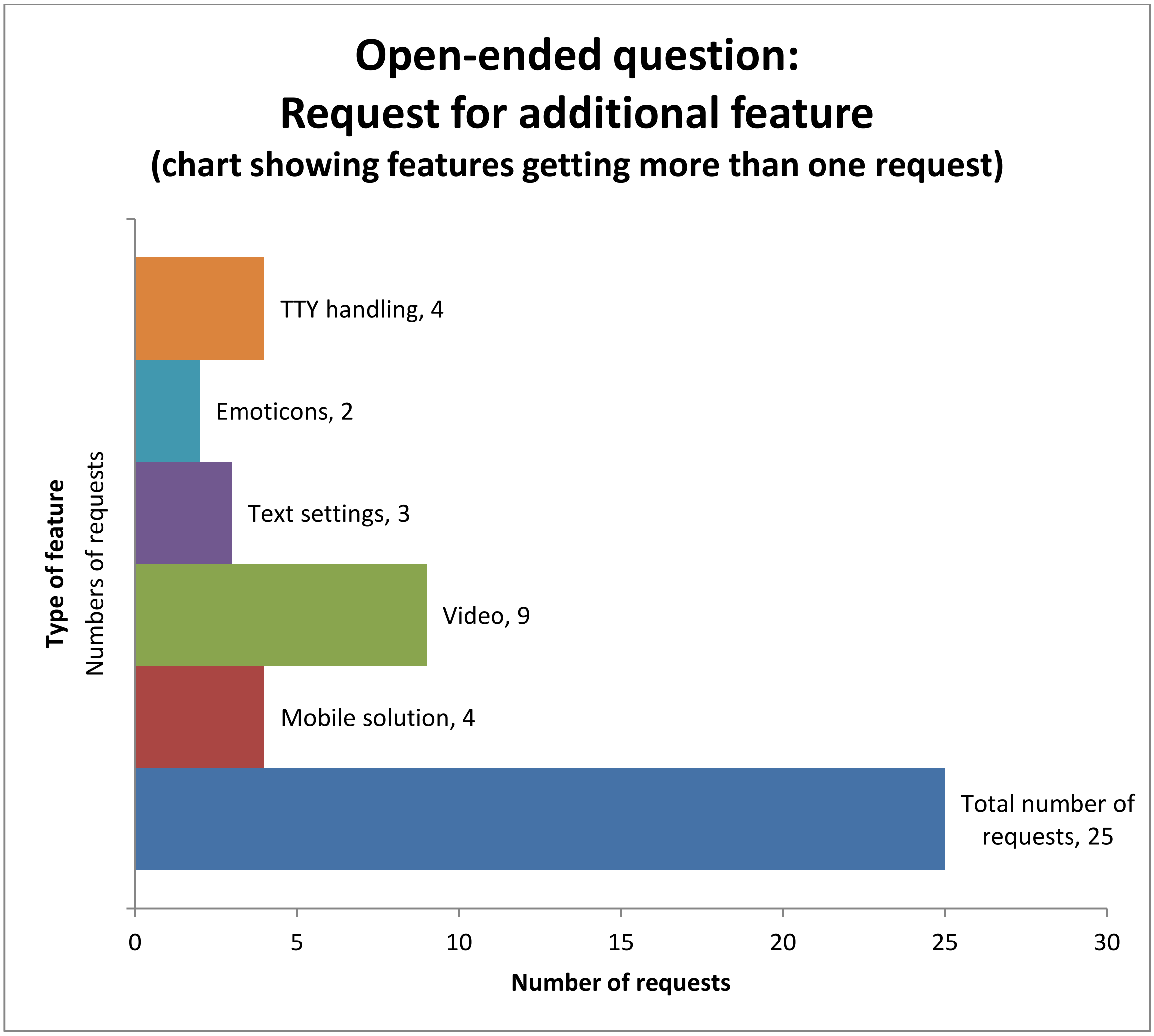
Comments:

Many of the comments were regarding the mobility and accessibility; if the solution was in a smartphone, they would have used it more. One wished to have text-to-voice in real-time.

Analysis:

Overall, the participants were satisfied with the solution; none of the participants reported that they disliked the solution.

Q16. Were there any features that you would have liked to see in the technology, but they were not included? If yes, please describe.



The responses collecting more than one respondents are shown here. Four wanted better TTY handling, two wanted emoticons in the text, three wanted more detailed text settings, nine wanted video communication feature and 4 wanted a mobile solution in a smartphone. Overall, 25 responses were received on this open-ended question.

Comments:

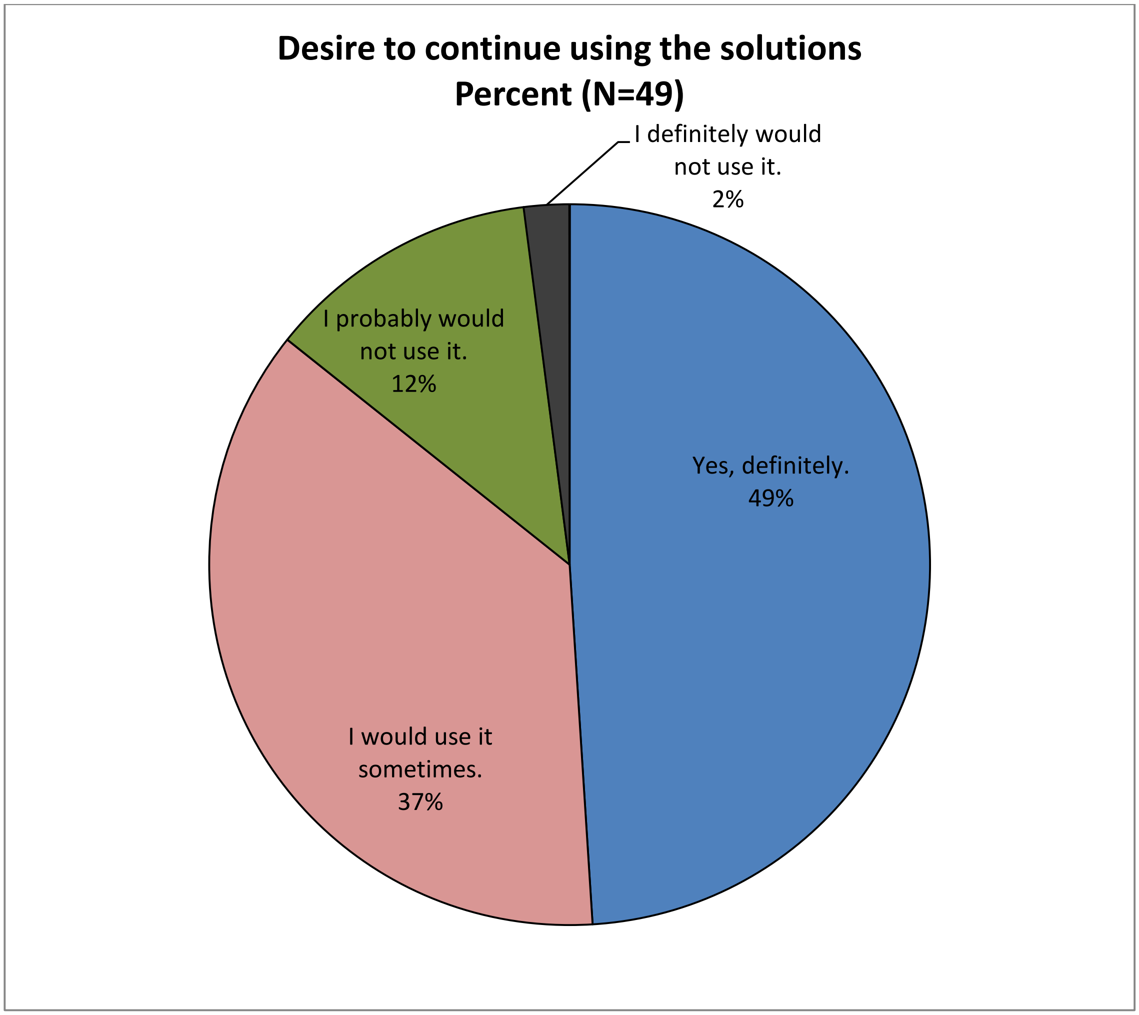
The responses were varied. Addition of video was commonly desired. Addition of emoticons was another expressed desire. Yet other comments on desirable features were: Adjustable text conversation windows, with the option to have two columns for IP-based RTT to RTT, one per conversation partner; and a single window for conversations with a TTY user. Split up the presentation of text from the TTY in sentences via automatic insertion of a new line. Many TTYs lack the ability to send a new line, and therefore text from the TTY is presented in a compact chunk on a RTT terminal expecting new lines as separators. This makes the timing of the text from the TTY hard to follow and inconvenient to read.

Most of the participants who answered the open-ended question wished for the addition of video in the call, as they are using this feature in other communication solutions.

Analysis:

The desire for extra features was on video in the communication solution to express feelings, and to provide for more natural communication with sign language and the possibility of lip reading. A mobile solution was also among the desired features. Many also thought it was important to be able to adjust text-settings as well as text conversation windows and the format of the text display.

Q17. If you have the opportunity to continue using the technology you used during the trial period, and your peers also had this technology available to them, do you think you would use it for your everyday communication?



A dominating 86% of the participants thought they would use the solution, with 49% judging that they would definitively use it and 37% would use it sometimes. Twelve percent did not think that they would use it while 2% were sure that they would not use it.

Comments:

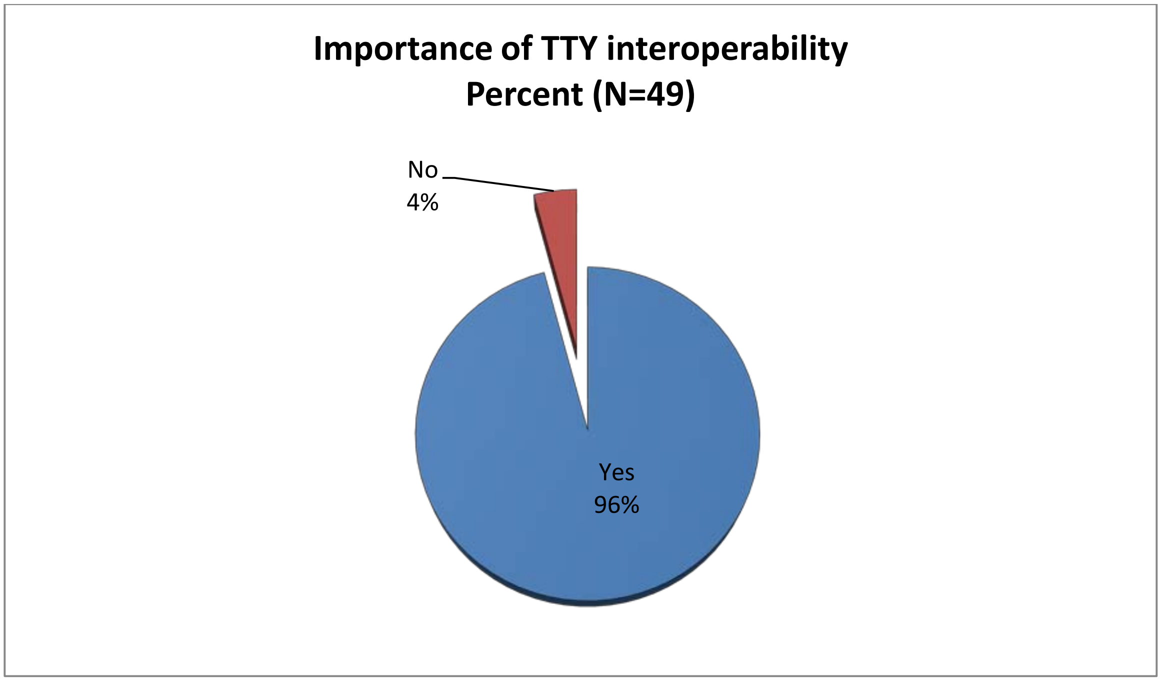
Participants expressed the following:

- Increase of the accessibility and possibility of spontaneous use via a mobile solution would lead them to use it more often.
- My conversation partner had huge benefit from the text support in our conversations.
- Adding video would make the solution more attractive.

Analysis:

In general the participants would like to use the communication solution, in particular if more features were applicable; video, functionality for TTY/RTT handling, text settings, and most important mobile solutions for smartphones for communication on the go.

Q18. Do you think it is important to be able to make calls between TTY and IP based communication technologies?



A dominating part (96%) of the participants responded that it is important to be able to make calls between TTYs and IP based communication technologies.

Comments:

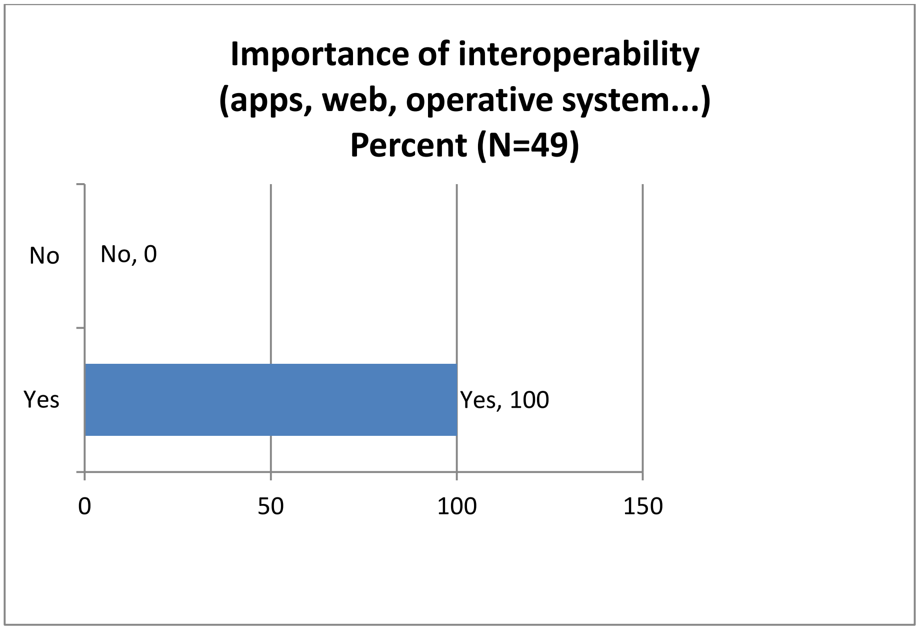
The following two quotes summarize the content of the other comments:

- Inclusion of deaf people who do not have videophone or other than TTY.
- This is the key to ensure TTY callers are not left behind with the inevitable uptake of RTT.

Analysis:

A huge majority agreed on the importance of access between older technologies and the new solutions as there are still groups of people having no options other than TTYs for reasons such as affordability, no access to high speed Internet, or being used to it and see no reason to change.

Q19. Do you think it is important to be able to make calls between different technologies and environment (web-based, app based, etc.)?



All participants answered that it is important to have interoperability in calls across different technologies.

Comments

Participants expressed the following:

- Expand more technologies available rather than limit to few.
- Hearing people can make these types of calls, so why shouldn't text callers be able to do so?
- To provide users with more flexibility and functional equivalency.
- There are many people who use mobile devices.
- Yes, support in several environments makes the communication flexible.
- The more options and environments the better as everyone is unique with different needs and preferences.

Analysis:

There was total agreement on the importance of interoperability between different solutions such as web-based solutions, installed apps, different devices and environments.

6 Earlier research on Real-time text user experience

This chapter presents brief summaries of earlier research in Real-time text user experience.

In a number of studies, with varying participation by both persons who are deaf or hard of hearing, or who have speech disabilities as well as participants without disabilities, the majority of users prefer the real-time text style, rather than handling completed messages turn-by-turn. The real-time flow of text in RTT is especially appreciated by the receiving party. (In one study there was no preference for professional discussions, but RTT was preferred for emergency and casual conversations.)

The studies are outlined below. According to the research results, RTT would be suitable as a service designed for all, but with specific considerations for its use as an accessible communications service.

6.1 "Real-time text and IM"

In a study by Solomon, at the University of Michigan [43],[44], Real-time text is compared to instant messaging. Twenty-four hearing students participated, performing conversational tasks in pairs using real-time text and Instant Messaging (IM). No audio was used.

The only significant differences that were found between real-time text and IM showed an advantage for real-time text.

- Less typing out of turn. 29% typed out of turn for RTT and 39% for IM.
- Less editing of own text. 38% edited their own text for RTT and 51% for IM.
- Less simultaneous typing. 11% typed simultaneously for RTT and 17% for IM.

This seemed to show a more conversational flow with RTT. Fewer cases are created where confusion can appear. It was also found that real-time text was preferred among the participants who read more but typed less, while the IM style was preferred among those who typed and sent much more text than receiving and reading text.

6.2 Text-Based Mobile Communication

In a small usability study of three different text-based communications modalities (Push Email, IM and RTT) conducted by RIM [32] with 5 deaf and 3 hard-of-hearing participants, they found the following:

100% of participants preferred RTT in emergency situations.

88% of participants reported that they would use IM over RTT to contact someone they were not familiar with, because they would not want the other person to see their mistakes and corrections.

50% of participants would rather use RTT over push email and IM to check in with the office.

50% of participants raised concerns about what the person would see (i.e., mistakes, corrections, immediate thoughts).

RTT Preference - participants preferred RTT over Email and IM initially because it felt like a text message, but was as quick as TTY.

Selective RTT Usage – participants' preference for RTT was limited to casual short conversations, whereas, professional and/or long communications would be sent using Email or IM.

Expected RTT Ubiquity – participants expected/hoped RTT would be available as an option in the future, and expect the same level of quality as found using other communication methods.

6.3 Comparison between Real-time text Conversation and Message Oriented Text Communication

This study of Hörlin, S[22] investigated how users perceive Real-time text and message oriented text communication) for two-way text communication over data- or telecommunications networks.

Eleven persons participated in the trial. Four identified as deaf or hard-of-hearing. All participants had experience using message-oriented text communication. Ten participants also had experience using real-time text communication. Nine persons indicated a preference for the real-time text method. One of the two persons preferring message-oriented text communication had never experienced real-time text communication before.

During the study, two persons communicated with each other via computer using the text communication software SipCon1 [42]. This program can be set to communicate in real-time text mode or sentence by sentence. It can also be set to present the dialogue in two columns or in a one column chat style with real-time preview. The real-time preview mode was used in the trial. Each pair in the trial made two trial sessions, one in each mode.

The persons who preferred real-time text conversation reported that they thought this mode was more efficient because they could see what the other person was typing in real time. This was perceived to be more interactive and giving a better indication of the other person's feelings and presence.

The persons who preferred to communicate with message-style text thought the conversation became easier to follow and understand with this mode and communicating with this method made it easier to focus on the task.

Several persons stated that a disadvantage with communicating with the message-oriented text method was that it was slow and inefficient. Several persons responded that the IM method made the conversation less natural and fluid.

No difference could be found in the answers between hearing, hard-of-hearing and deaf participants.

Most of the people participating in the trial preferred communicating with the real-time text method. This test also indicated an interest in using real-time text communication, both among deaf and hard of hearing people and hearing people. The extent of this interest is hard to assess exactly based on this trial because there were so few participants.

7 Summary of the State of RTT

The results from the trial and testing sessions indicate that RTT should be expeditiously deployed in electronic communications. Interoperability is critical, as also indicated by participant responses in the trial.

In the following, the current state of RTT deployments worldwide is reviewed, as are the availability and status of technical standards. Given currently existing implementations and selections of standards, using RFC 4103[20] as the basis for interoperability is an obvious choice for SIP-based implementations. This is also the standard that was used throughout the trials and testing sessions in this report, as indicated in Section 4.3.

7.1 Real-Time Text in Products and Networks around the World

7.1.1 Implementations including the RFC 4103 standard for Real-time text

The IP-based RFC 4103[20] RTT standard has widespread implementations in services and products in the US and abroad. Text relay services in France, the Netherlands and Sweden provide access for communication providers using SIP and RTT using RFC 4103. Video relay services in France, the Netherlands, Norway and Sweden have been procured to provide real-time text during the video call also using SIP with RFC 4103 for RTT. At least four communication technology providers and a number of communication service providers in Europe are providing terminals, terminal software, communication services, interoperability with other providers, interoperability with legacy PSTN text telephones, answering machine services, relay service access and emergency service access all using RFC 4103. The first of these services has provided SIP-based calls with RFC 4103 RTT since 2003.

In most cases the real-time text is provided together with video and audio in the calls, forming the total conversation service, defined by ITU-T in year 2000 in the standard ITU-T F.703[24]. There are also users who use only audio and RFC 4103 real-time text (without video).

In the US, the systems provided by Star VRS (formerly CAAG), and used for RTT by deaf-blind user communication and as accessible telephony to enterprises, includes support for RFC 4103 as well as another real-time text transmission method. Providers of NG9-1-1 emergency service technology provide RTT access to the systems using SIP and RFC 4103 as specified by NENA and have been performing interoperability test events with that functionality.

The technologies used by AT&T for a series of demonstrations in May 2015 used RFC 4103 for real-time text, and a RFC-4103-RTT to TTY-gateway for conversion of text communication in calls between RTT users and TTY users. Furthermore, AT&T has proposed RFC 4103 to be used in IP-based wireless services in a petition for rulemaking to the Federal Communications Commission [8] and has pledged to deploy RFC 4103 throughout its networks by 2017.

A proof of concept of the technologies that implement the RTT standards and interwork with the TTY standards, developed by the RERC-TA, was demonstrated at the 2015 Biennial Telecommunications for the Deaf Conference in Baltimore, MD. A call was made from a web-based client for simultaneous real-time text and audio in the IP network. The call went through two gateways for transition of protocols and media to a traditional TTY in the PSTN network. Maryland TTY Relay received the call and relayed the call by voice to a mobile phone belonging to a member of the Technology Access Program at Gallaudet University. This demonstrated the feasibility of interoperability between TTY and SIP based communication using the RFC 4103 standard.

7.1.2 Implementations of Real-time text standards in other modern technologies

7.1.2.1 XMPP extension XEP-0301 for Real-time text in the XMPP environment

Real-time text can be implemented in other modern IP environments than SIP and IMS. For SIP and IMS RFC 4103 is the most natural technologies for RTT, because RFC 4103 makes use of the same basic media transport protocol as other real-time media in these technologies.

However for XMPP [39],networks, the XEP—301 standard might be used with a RFC 4103 gateway where the XMPP network interfaces with a SIP or IMS network. The XEP-0301[31] standard is published for the purpose of implementing real-time text to XMPP. The standard was created in 2013. It is reported to be implemented in about four implementations as improvements to XMPP text messaging so that XMPP messaging can be experienced with real-time flow.

XMPP is a text chat standard, so by implementing XEP-0301 it is possible to achieve real-time text communication in a number of text chat services. It is also possible to combine it with voice on XMPP.

7.1.2.2 WebRTC implementation of Real-time text

Implementation of Real-time text in WebRTC[37] technology was provided for the trials described in this report. The specification of the data channel for WebRTC was not mature in 2014 when the implementation started, so an implementation based on WebSocket[18] transport of ITU-T T.140 [28] Real-time text was created and used instead for the trials. The implementation included a gateway to SIP with RFC 4103, for interoperability with other forms of Real-time text.

Now, late in 2015 the data channel specifications for WebRTC are more mature, so it would be feasible to move the implementation to the WebRTC data channel [30] and make an early implementation of the draft specification for Real-time text in WebRTC in draft-schwarz-mmusic-t140-usage-data-channel [11]. Again, the WebRTC data channel real-time text should be converted to RFC 4103 where the WebRTC network connects to SIP or IMS networks.

7.1.2.3 Non-standard methods for real-time text

It is possible for any developer to create and use a Real-time text method within a limited application area without standardization. There are such implementations in operation in some products and services. They can coexist with the standardized methods as long as their providers convert to the RTT standard for any other networks they connect to (for example RFC 4103 where it connects to SIP or IMS networks). More about interoperability is described in the next section.

7.1.3 The Need for Interoperability in RTT and Voice

The benefits of a communications system increase greatly as more users and services can be reached by it. And, a national communication system that is not interoperable is a contradiction in terms. Voice telephony has had a long tradition of providing interoperability between different users, providers, devices and services so that anyone can call anyone regardless of the devices or services they subscribe to. This is accomplished though a system of requirements that all software and hardware on a network are required to implement.

The functional equivalence requirements require a similar scope of interoperability for RTT services – and a similar set of (at least one) RTT standard that MUST be supported by ALL software and hardware on the network.

The traditional way of achieving interoperability is to agree on a common specification for technologies used on a network and specification of the exchange on the border between providers, equipment or technologies, and then verify that interoperability has been achieved by verification tests when the technologies are implemented.

Interoperability is not only about users of RTT to be able to call each other. It is about a complete eco-system that needs to provide interoperability. That means interoperability so that users of different providers using different terminal equipment can reach and be reached by other users. It includes users of the same and other providers. It also includes the possibility to have calls with relay service support. 9-1-1 emergency service calls also need to be interoperable with RTT so that users of RTT can reach all important society functions in real-time text and voice.

Interoperability not only means that the endpoints need to be able to communicate, but also that connectivity is provided through all of the networks and all equipment from one end of the call to the other. Routers must not put obstacles for the calls consciously or by not being specified to carry the intended traffic. Firewalls must not block the transmission as long as it is in agreement with the communication policy of the organization having the firewall.

When deciding on how to implement Real-time text, extreme care shall thus be taken so that the interoperability and functionality goals are not compromised. It shall also be considered that no extra complexity is forced upon the implementers who only want to implement the initially standardized solution based on RFC 4103. A feasible way to achieve both interoperability, good functionality and flexibility is to require interoperability with the standards already specified by NENA, 3GPP, IETF and GSMA in interfaces between services using technologies where these standards are relevant. New standards may be needed when real-time text is introduced in other than the already standardized technologies. For such cases, real-time text connection, transmission and interoperability with current systems need to be specified. That is the approach taken by the draft section 255/508 refresh [7].

It is especially important for interoperability that the interface between services follow agreed interoperability standards, so that users are not locked into a single network. In addition, terminals also need to meet interoperability standards so as to avoid locking the user into a single device or carrier and to ensure that people with disabilities have the same kinds of choices in a competitive market as the population in general.

The interest of the users are best served if providers compete by providing the solution with the best functionality within the agreed interoperability standards, rather than inventing new ways to communicate for the same basic function.

7.2 Status of RTT Standards

RTT is mature, has been standardized for many years by the electronic communications industries in international standards organizations, and is used in the field. This can be seen both from all of the standards that specify its use, and all of the examples of technology and systems that use it today.

Note that proper references to many of the documents mentioned in this section can be found in the EAAC TTY transition report[12].

It has become a practice in making standards for real-time communication divided into service level requirements, call-control standards, media coding and presentation standards, and media transport standards.

Communication features that are to be implemented with interoperability between many implementing organizations need to be standardized. The communication industry recognized the need to standardize real-time text in digital networks in the 1990s. At that time the feature was called "text conversation".

The *service level description* was made in ITU-T in year 2000, and resulted in inclusion of real-time text media component description in ITU-T F.700[23] and its use in services in ITU-T F.703 [24] (Multimedia conversational services; service description).

The coding and presentation of RTT was specified in ITU-T T.140 [28] in 1998 and used in a number of multimedia standards. The first standard for transport of real-time text in the IP environment was developed under IETF in 2000, in RFC 2793 "RTP Payload for text conversation" [21] and applied first in the multimedia protocol environment ITU-T H.323 [26], in ITU-T H.323 Annex G [27], and soon thereafter applied in the now-dominant IP based multimedia protocol RFC 3261 (SIP)[33]. In 2005, the transport standard RFC 2793 was revised and was assigned the number RFC 4103 [20], also titled "RTP payload for text conversation".

While the development of electronic communications today is focused on IP communication, there were some standards for real-time text developed for circuit switched environments before IP technologies became dominant. These standards for real-time text in circuit switched multimedia technologies are described in ETSI EG 202 320 [16], and are not further mentioned here since the circuit switched network (PSTN) is being phased out.

As noted above there is an instant messaging technology for XMPP environments [39]. A standard has been established for improving the user experience of XMPP messaging by adding real-time text transmission using the standard called XEP-0301 [31] In-Band Real time text, published in 2013. While this standard can contribute to an important improvement of the messaging concept by reducing the waiting times and increasing the feeling of contact between text chat participants, it has not yet been discussed for inclusion in any infra-structures such as interfaces to emergency services, relay services or interworking between carriers since these are all SIP and IMS based.

Real-time communication (voice, video and real-time text) based on web browser technology is in very active development under the name of WebRTC [37]. The standardization of this environment is split between W3C and IETF. W3C produces application program interface standards and IETF produces communication protocol standards for this area. It has been decided that Real-time text shall use the data channel of the WebRTC concept for its standard implementation in the WebRTC technology. There is a draft under development for real-time text transport in WebRTC, named draft-schwarz-mmusic-t140-usage-data-channel [11]. This draft also describes how an implementation of real-time text in WebRTC shall be able to interoperate with SIP based technologies and RFC 4103.

7.2.1 Citations of RFC 4103 in Standards

3GPP, the global organization for standardization of wireless communication, standardized use of RTT in SIP calls in 2001, called GTT-IP, in specifications 3GPP TS 22.226 [2] and 3GPP TS 23.226 [4]. These specifications use the same transport for text; RFC 4103 [20] as specified by IETF.

A more general multimedia communications system called IMS Multimedia Telephony was introduced a few years later in 3GPP, and Real-time Text was included as standardized medium together with audio and video. The first version of this standard was completed in 2007 in 3GPP TS 26.114 "IMS Multimedia Telephony, Media handling and interaction"[5]. As in the other above-mentioned standards, IETF RFC 4103 [20] is the base for transport of text and ITU-T T.140 [28] the base for coding and presentation of conversational text in this environment.

Later on, the use of Real-time text as specified in these 3GPP specifications TS 26.114 (which specifies RFC 4103), and TS 23.226 (which also specifies RFC 4103) has been picked up and its use clarified in further detailed 3GPP specifications for different situations, such as for emergency services in TS 23.167 [3] and TS 22.101 [1](thus specifying RFC 4103), and in general service specifications in 3GPP TS 22.101 (thus specifying RFC 4103) and interoperation with TTY in 3GPP TS 29.163 Annex I [6](thus specifying RFC 4103).

GSMA is the organization picking up 3GPP specifications to be used in wireless products. Real-time text as specified in 3GPP TS 26.114 (which specifies RFC 4103) was picked up and described for inclusion in wireless products in GSMA IR.92 IMS Profile for voice and SMS [19] (thus specifying RFC 4103 for real-time text).

In IETF, the use of Real-time text in the SIP environment was described in 2008 in RFC 5194 "Framework for Real-time text over IP using the Session Initiation Protocol (SIP)" [36]. This specification clarifies the use of RFC 4103 for transport of text.

The use of RFC 4103 in emergency services is also specified by IETF, in RFC 6881, "Best current practices for communications services in support of emergency calling" [34]. RFC 4103 is also specified for the US by NENA in NENA NG9-1-1 08-003, "Detailed functional and interface specification for the NENA i3 solution."[41], and for Europe in EENA NG112 LTD [13].

The European Standards Institute ETSI has specified RFC 4103 for Real-time text in a number of standards, including EG 202 320 Duplex Universal Speech and Text [16]. ES 202 975 [15] Relay Service Requirements has an indirect reference via ETSI EN 301 549 [17]. ETSI TS 101 470 Total Conversation Access to Emergency Services [14] specifies the use in emergency service access.

SIPFORUM, in their profile for US VRS providers interoperability in SIP [40] specifies use of RFC 4103 for real-time text.

Accessibility requirements standards are also including Real-time text. In the US, the Access Board Section 255/508 draft refresh [7] includes requirements for interoperable real-time text wherever there is voice communication, and refers to RFC 4103 for SIP technologies.

In Europe, the accessible procurement standard EN 301 549 [17] requires interoperability of Real-time text, and points out what to support in three named environments, the telephone network PSTN, SIP in IP networks, and 3GPP IMS. For PSTN, the European standard references ITU-T V.18 [29] or a nationally used subset of it for text telephony. That corresponds to the TTY requirement in the US. TTY is a sub-mode of ITU-T V.18. In SIP, RFC 4103 is referenced as the transport for real-time text. In IMS, the real-time text part of 3GPP TS 26.114 (which requires RFC 4103) is referenced for transport of real-time text.

For the 255/508 refresh, TIA-825a is specified for PSTN networks. RFC 4103 is specified for SIP based networks. IMS is not mentioned in 255/508 refresh, because IMS is a SIP network (which requires RFC 4103), and the IMS specification TS 26.114 referenced by EN 301 549 requires RFC 4103 so it would be redundant.

Therefore, the IP based standards named in EN 301 549 [17] and 255/508, all refer to RFC 4103 for real-time text.

In summary, RFC 4103 has been specified for RTT in standards set by major standards organizations around the world - standards that have been adopted by telecommunications carriers in the United States and by governments and service providers in Europe.

8 Path for evolution of RTT to new standards in the future

As with any communication technology, it is important that there be a path for adoption of new technologies overtime as they evolve. The path for real-time text would be the same as is used for updating, or evolving/ incorporating new voice technologies. With voice systems, there are minimum required formats that all network elements (software and hardware) (network and terminal) must support. New formats/technologies can be introduced, but the old required formats must continue to be supported until there are no systems or software left in the network that do not support the new technology (as well as the old). Once this occurs, the old technology can be retired since all technology on the network supports the newer technology.

For real-time text, the same approach can be used. In this case however only a single, rather than multiple, real-time text formats would be required initially for each network type (e.g. PSTN, SIP, XMPP, etc). If a new real-time text format is created that it is believed to be superior for a network type, it can be introduced alongside of the old real-time text format and required as a second required real-time text format (that must be supported by all new software and hardware, network and terminal, technologies on the network). Over time, as all of the old software and hardware is replaced, there will eventually come a time when all of the software and hardware on the network will support the new (as well as the old of course) required real-time text format. At this point, the old real-time text format can be retired (or no longer required) since all of the elements of the network support the new format.

Thus adoption of any format as a required format does not inhibit the introduction of any new format(s) in the future. The requirement of at least one format however, that must be supported by all elements of the network, is required for interoperability to be achievable.

9 Conclusions

9.1 Conclusions from the trial and RTT research

The technologies for electronic communications that have been developed, trialed and disseminated in the RERC-TA project were appreciated by deaf and hard-of-hearing users, as well as hearing persons having reason to communicate with their deaf and hard-of-hearing peers.

Real-time text for both receiving and sending text in direct communication is the main choice for 34 individuals in the field trial with 49 participants. Thirty-four participants also preferred to be allowed to type at the same time as their partners for a more rapid text communication and to be able to interrupt where needed.

In the longer field trial that lasted two months, 8 of the 15 participants added voluntary comments that they appreciated to be able to communicate directly with the communication partner. Before the trial the users had to select other ways such as asking someone to call, send text messages or only communicate face-to-face.

Among those who use voice for their telecommunication, regardless of hearing, they thought it was important to have audio and text together.

The following additional features were desired in the communication solutions:

- addition of video in the conversation
- mobility and accessibility (smart phones)
- alerting devices to be reachable
- ability to change text settings (font size, colors, etc.)
- text conversation handling (improved, less compact layout of text from TTY with newlines inserted at appropriate locations)

The request for video in the call is natural as a majority of the participants use sign language in one-to-one situations and are videophone users. Several conversation apps, in smartphones, tablets, and similar devices, already allow video chat.

In the field trial, the participants felt that they were stuck with tablets and computers that were not very mobile, leaving them to be inaccessible and unreachable when not nearby. This is also connected to the request for alerting devices.

The web-based communication solution built on WebRTC technology was a simple version without text settings or the possibility to adjust the text conversation windows. The participants commented on the need of improvement.

In the TTY-RTT conversations there were problems with the lack of ability to send a new line from the TTY. This made the text inconvenient to read after a while.

The users who normally were using TTY for their telecommunication had some trouble with the turn-taking in RTT calls as they were waiting for GA or some other indication to proceed with the conversation not realizing that it was not required when TTYs were not used on the call.

A conclusion from the participant responses is that real-time text together with audio and video in smartphones with additional alerting features is the way to go for accessible electronic communication. The solutions also should include settings and adaptations for both text style and text presentation layout.

There was an overwhelming agreement among the trial participants that interoperability in calls with RTT and voice between different devices and technologies as well as interoperability with legacy TTYs is important. The motivation is to ensure accessibility for individuals with hearing loss to the world of electronic communication.

The conclusion of preference of RTT over messaging for conversational situations is also supported by the included overview of earlier research. Less simultaneous typing and less risk for confusion is also reported by these studies.

9.2 Conclusions from the standardization and implementation overviews

The conclusions from the implementations and standardization work were that the most widespread and most referenced standard for Real-time text is RFC 4103, referenced and used in SIP and IMS technologies. The fact that this standard is required in IP-based emergency service standards and specifications and in procurement standards makes it recommended for use in these technologies and as a default for interoperability between these technologies and other technologies.

There are technologies where RFC 4103 cannot be used. In such cases other standards should be defined – with conversion to RFC 4103 where these technologies interface with SIP or IMS networks.

One such other network technology is XMPP. Here there is a standard for Real-time text in XMPP messaging called XEP-0301. It is not in wide spread use yet, but it is ready for deployment in XMPP networks.

Another technology of current high interest is WebRTC, where Real-time text is currently under standardization, and early implementations for trials can be done. Again, for interoperability the Real-time text for each of these would need to be converted into the Real-time text of the other network environments where they interface with them.

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