Brief on Disability Issues in the Transition from PSTN to VoIP

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Introduction

This is a purposefully brief overview of issues and considerations related to disability access to telecommunication – as they relate to transition from PSTN to VoIP.

Bullets and short phrases are used where possible to keep the presentation concise. Additional information is available on request.

Mostly it is all Good (potentially)

If done properly, the transition to IP based communication can provide benefits on all fronts with regard to disability access

- Better accessibility
- New access opportunities not available with Analog Technologies
- Much lower cost to manufacturers to include access
- Lower costs to government for same level of service
- A way to address the problems that are on the horizon due to aging population

It is only potential though

The potential for greatly increased access at lower cost is there – but the potential to squander the opportunity is also there – leading to much higher costs later. We have a once-in-a-lifetime opportunity to build access into the emerging telecom structure and, like curbcuts, it is easy to do now before all the concrete is poured – and much more expensive and disruptive to do later. And, like curbcuts and captions, built-in access will see major use by and benefit to mainstream users as well.

We need to ensure that a few key things are done – like ensuring that the emerging telecom system supports ***voice***, ***real-time text*** and ***video everywhere*** and in a ***compatible interoperable*** fashion. If we do – then emergency calls work better for elders and people with disabilities, more people can live independently longer, and people will have to rely less on government-funded services (e.g. relay services) to communicate, and will be able to communicate directly more often.

Disability Issues during the Transition

As with any system change, it would be much easier if we could just instantly dump the old and implement the new. The ugly transition period where all of the legacy equipment must be accommodated creates many issues. This is exacerbated because of the existence of different systems and the slower rollout of IP communication rurally and in remote locations.

There are essentially 3 issues that need to be addressed to avoid loss of access in the PSTN to VoIP transition.

1) We need to ensure that consumers <u>with impaired speech or hearing</u> have, or can invoke, ***call quality on VoIP that is at least as good as in the PSTN***.
   (Or people with both speech and hearing disabilities will have decreased access on VoIP from what they had on PSTN)

2) We need to ensure that consumers that <u>cannot use speech (in one or both directions)</u> have the ability to use ***real-time text intermixed with Voice*** on VoIP ***as they did with the TTY on PSTN calls.***
   (Or people who rely on text or text intermixed with speech will have decreased access on VoIP from what they had on PSTN- rather than the increased access for less cost that should come with IP)

3) Until the last PSTN branch is closed down…
   We need to ensure that consumers that <u>cannot use speech</u> (and must use TTYs because they still only have analog phone service) ***have the ability to use their TTYs reliably to call everyone they could before – including people now on VoIP networks***.
   (Or people who rely on text or text intermixed with speech will be cut off from the majority of the population on the VoIP networks)

All three of these affect people with impaired speech (due to physical or neurological causes), reduced hearing, deafness, and deaf-blindness. There are other disabilities and combinations of disabilities that can be positively improved by the move to VoIP if done correctly – but **<u>these are the three that directly relate to a loss in accessibility from PSTN to VoIP if it is not done correctly</u>**.

The needs of those who CAN use speech reliably (1) are fairly straightforward. The problems for those who CANNOT use speech reliably (2 and 3) are a bit more complicated but have defined solutions.

For people with disabilities <u>who can use speech</u>

For people who can speak and hear well enough to use traditional phones, the only real issue is the introduction of less than PSTN quality audio.

- VoIP holds the promise of much higher fidelity audio, which can positively increase intelligibility of voice communication BOTH for people with speech disabilities, and those with hearing disabilities.
- However digital telecom (including cellular digital telecom) **sometimes uses codecs with lower fidelity**. Calls originating on either PSTN or VoIP that pass through such codecs pose barriers to people with impaired speech and with reduced hearing - which is very common in the growing aging population.

RECOMMENDATION (for those who can use speech): As we transition to VoIP, ensure that consumers have (or can invoke on any phone) call quality that is at least as good as they currently have on the landline PSTN. Higher is of course better.

For people who <u>cannot use speech reliably</u>:

People who cannot speak clearly enough, or cannot hear well enough, to rely on speech alone to communicate, currently use a TTY or the TTY intermixed with speech, to communicate. Some use the real-time text of the TTY in both directions while others, especially late deafened adults, will use speech in one direction and the TTYs real-time text in the other.

In the transition it is essential that

A) after the transition they still have the ability to use ***real-time text intermixed with speech*** on basically any phone they encounter (like they can today on the PSTN by simply connecting their TTY in parallel with any phone using a Y connector)

B) during the transition – those that still are on PSTN networks (no IP available) where TTYs are the only options for text on a phone call, can call those that are not on the PSTN and are no longer using TTYs.

To understand the problem –and the recommendations – some quick facts are useful.

- **BACKGROUND**
  - **TTYs on PSTN**
    - For those on the PSTN landlines (and where no IP connection is possible), TTYs are the ONLY device they can use to communicate with text or text mixed with voice.
      - TTYs allow intermixing of speech and text on the same call – an important capability especially for elders who often cannot type well or at all due to arthritis.
  - **TTYs on VoIP**
    - TTYs **DO work reliably <u>on IP Core Networks</u> of the largest carriers**
    - But TTYs **<u>do NOT work</u> on all parts of IP networks/services** (even the ones of some major providers), nor on some smaller networks, nor on private IP networks that use lower bandwidth codecs
    - And TTYs **<u>do NOT work well</u> on <u>wireless</u> IP networks <u>including Fixed Wireless</u>** networks that are being deployed for rural VoIP and IP.
  - **Real-time text (RTT) on VoIP**
    - Because TTYs have low functionality (limited character set, slow rate of communication, don't allow interruption without data corruption, only one way at a time, can't be used for caption telephony, etc.) better real-time text methods been developed by industry.
    - Most notable is RFC 4103, which was originally specified for SIP and also now cited as the RTT format for IMS, 3GPP, GSMA, emergency services and most SIP-related (and many other) systems. (see R1 in **References**)
    - The FCC Emergency Access Advisory Committee, in a recent study on TTY Transition, also recommended RFC 4103 as the real-time text method to be used on SIP based VoIP systems. (see **References**)
  - **Transition Notes**
    - Even when 80 to 90% of people are on VoIP, there will still be rural and remote users who are still on PSTN only lines.

- These will include people who are speech impaired, deaf, or hard of hearing and who need access to telecommunication – (perhaps MORE than those not in rural or remote locations).
  - **<u>Therefore TTYs will need to be supported until the last PSTN lines disappear unless</u>** we figure out how to do IP voice+text communication over ANALOG lines (not realistic, especially for incoming calls), <u>or</u> we provide IP in some other affordable way to these remote locations (in which case those users do not need the PSTN)**.**

RECOMMENDATIONS (for those who cannot use speech/hearing well to communicate):

1) **<u>That we get as many onto IP as possible – as soon as possible</u>**
   a. This reduces the number of people with disabilities who are stuck on PSTN and allows them to move to IP-text and eliminates their need to stay on TTYs. (This does not solve the problem for those who are still on the PSTN however, hence #3.)

2) **<u>For those who CAN move to VoIP:</u> ensure that there is an IP (data based) <u>*real-time text intermixed with speech on a call*</u> option for them that is (at least) as interoperable and universal as TTY was on the PSTN:**
   a. Works on any phone
   b. One standard that is supported by all phones on the network
      i. For IMS and SIP systems, the format specified for **rtt+voice** calls, by the industry bodies that create the specifications for IMS and SIP, is RFC 4103.
      ii. (other formats can be used on private networks as long as they are transcoded to the public network format where they connect)
   c. The real-time text and voice are implemented on phones (and in network equipment) such that either real-time text or voice can be used individually, or both can be used interchangeably and intermixed on a single call (as was true with TTYs on PSTN).

   NOTE: Moving to IP (data-based) RTT methods also overcomes many of the weakness of TTYs. The data nature of IP RTT allows bi-directionality, interruption, the simultaneous flow of voice and text that is required for non-proprietary captioned telephony, full character sets that support Spanish and other languages, and transmission that doesn't lag fast typing.

3) **<u>For those we CAN'T get onto IP</u> :** (and must use TTYs - their only option) **ensure that they can reliably use TTYs to communicate with others on VoIP or another PSTN segment by:**
   a. **Establishing a single real-time text standard (that works with voice) for the new VoIP Backbone** (as described above) with everyone on the backbone using the **RTT+voice** standards.
      - Again, if the backbone technology is SIP or IMS, the standard is RFC 4103. This would also meet the total conversation (Voice+RTT+Video) requirements that are proposed for emergency and other next generation telecommunication.
   b. **Creating a national TTY to RFC 4103 gateway** (funded by Relay fund) to allow those using TTY to communicate with IP-Text users.
      - Relay funds are used because this is a (non-human) relay and because it will reduce overall use of the relay service from what it would be if humans needed to form the relay bridge rather than the gateway.
      - TTY users reach that gateway by using a phone number (10 digit or preferably a short dial prefix) followed by the target number.
      - VoIP users with special interest in communicating with TTY users can get VoIP subscriptions that route their PSTN directed calls through a TTY enabled gateway, simplifying the dialing procedure between the new (VoIP) and the old (TTYs on PSTN).

c. **Ensuring that high quality Core IP networks** (that are good enough to transport the TTY reliably) **are extended out the places that PSTN networks connect to the IP backbone.**

- In order for this to work, TTY signals coming off of the PSTN trunks need to reach the National TTY to RFC 4103 gateway located within the VoIP Backbone without distortion. The only way to ensure this is to either a) put a transcoding gateway at the connection point of every PSTN branch (very expensive and difficult to do) or b) ensure that the VoIP path from each PSTN trunk back to the transcoding gateway is of sufficient quality to pass the TTY tones reliably.
- Reliably means 99% character accuracy - or 95% word accuracy (no more than 1 out of every 20 words has an error in it)
- If the backbone is not IMS or SIP-based, then substitute the RTT + voice standards for that backbone for RFC 4103 in the above statement.
- This also ensures that TTY users on separate PSTN networks (connected only by the VoIP backbone) can reliably communicate with each other on TTYs

4). Ensure that the number plan includes reachability for calls with all media in IP.

- Today, some carriers route their VoIP calls through the PSTN to connect to VoIP users on other carriers. This will strip off any IP Real-time text or video that is used by people with disabilities to communicate.
- Both during the transition – and afterward – people need to be able to use phone numbers to call from VoIP to VoIP phone without leaving the core VoIP network. The numbers must also allow calls from PSTN phone to VoIP phone (and vice versa) with routing through the RTT transcoding gateway (by pre-number dialing – or request for default routing).

As noted above, there are other things that should be kept in mind when moving from PSTN to the new VoIP standards to ensure cross-disability access to the new infrastructure. Most of these deal with the design of the terminal equipment, and are covered in the guidelines being promulgated by the Access Board and FCC as part of 508 and 255 refresh.

The recommendations above however, deal with things that must be done in the networks, and things that if they are not done will not just miss an opportunity to greatly increase accessibility for little cost, but actually decrease accessibility from what was available on the PSTN that is being closed down.

More information on any aspect is available on request.

References

Readers are also directed to two other related submissions to the FCC,

1. The FCC EAAC TTY Transition report which can be found at http://www.fcc.gov/encyclopedia/emergency-access-advisory-committee-eaac

2. The revised Proposal R1 report from RERC Telecommunications Access which can be found in the FCC filing by the RERC in GN Docket 13-5, 12/6/2013. Online: http://apps.fcc.gov/ecfs/document/view?id=7520960695

For those that find ".doc" versions more accessible, .doc versions of these documents (and others related to IP transition, can be found at http://tap.gallaudet.edu/IPTransition/

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