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Title: Global Text Telephony Architecture Discussion

Document for: Discussion

Agenda Item: 5

Summary

This document describes why it is proposed to introduce a CTM-SRF service node in the core network to provide the interworking functions between PSTN Text Telephony and GTT-Voice.

Another location mentioned has been to place the conversion in the transcoders.

A good reason for the core network location is the good opportunity to rapidly develop the text conversation feature according to evolving needs when it is located in a separate environment not restricted by many other concerns.

Another important factor is that the proposed solution is the only one that can provide roaming support for text telephone users, allowing an equal level of service to be accomplished for these users even if the uptake of the feature is uneven.

The document is written with a goal to cover the open issues mentioned in previous 3GPP meetings and documents.

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Introduction

Real time, character by character text conversation is valuable in distant conversation. The GTT feature specifies how text conversation is introduced in a set of environments. The trend is towards the IP Multimedia environment, where the users can benefit from simultaneous communication in video, text and voice. This combination offers vastly enhanced usability compared to any of its single components.

In IP Multimedia, text conversation can be included from the beginning, and be treated as any other media stream with mainstream protocols. This is the important future conversation environment for people who have little or no use of voice telephony today. The trend in this direction is already evident in the fixed networks, encouraged by governments and telecom authorities.

It is important to make sure that the introduced solutions for GTT are extendable in this direction.

For interworking between today's PSTN Text Telephony and GTT-Voice, the CTM-SRF service node is proposed to be introduced and located in the core network. Calls between the two environments are routed through the service node with standardised routing mechanisms.

This architecture was proposed in 3GPP TS 23.226 GTT Stage 2 draft in September 2000.

This document lists a number of questions that has been placed on GTT in general and on the Core Network location of the CTM conversion in special.

The proposed urgent conclusion for the Cicuit Switched Voice path case is documented in two CR:s to 3GPP TS 23.002 Architecture.

1 Location of CTM for text telephone interworking

When satisfying the requirements for PSTN Text Telephone interworking with GTT-Voice, a conversion must be placed in the network between the PSTN terminal and the radio interface. The conversion is between CTM, and PSTN type of text telephone protocols. The PSTN textphone protocols are based on different low speed modem technologies and are defined in ITU-T V.18. CTM is a kind of robust, error tolerant modem technology suitable for voice channel transmission of real time text and is defined in 3GPP TS 26.226. The text is coded as specified in ITU-T T.140.

CTM and the conversion to PSTN text telephony was created with the view that it should be placed in a service node in the core network. This was reflected in the draft of 3GPP GTT Stage 2. TS 23.226, version 0.0.4.

A discussion has appeared about instead locating the CTM-PSTN conversion in the transcoders in the BSS. At first view, this looks possible.

However, there are a series of conditions that make the core network placement more favourable, or even the only feasible.

Therefore, the now proposed addition of descriptions to the network architecture specification 3GPP TS 23.002 recommends a core network location. It also proposes routing of potential text telephone calls to the conversion facility by means of CAMEL procedures for general user calls. For emergency calls, a simple routing action of the network switching functions is proposed.

Central to the proposal is that only calls that may contain text telephony are treated in the service node. Therefore, the discussion is to a large degree about how selection and routing of these calls shall be made.

A proposed Annex to 3GPP TS 23.002 describes an example of methods for routing. It is complete with mechanisms in ISUP and CAMEL Phase 1.

2 Requirements

The requirements for GTT are documented in 3GPP TS 22.226 GTT Stage 1. The original requirements are in the annexes to that document.

3 Characteristics of the solution

The main characteristics of the Core Network located solution is specified here. A detailed example for the selection and routing is found in the proposed Annex to 23.002.

- If it is decided that emergency calls shall pass the Service Node, routing of them, based on an indication that they are emergency calls shall take place.
- If a mobile originated user to user text call shall be established, it shall use the CAMEL procedures to route the calls.
- If a mobile terminated user to user text call shall be established, it shall use the CAMEL procedures to route the calls.

3.1 Functionality of the CTM-SRF service node

A call through the CTM-SRF service node shall go through a text telephone termination and a CTM termination that together form the call path.

Furthermore, some routing actions shall be . That is described in the proposed 23.002 Annex.

The default action of the call path in the CTM-SRF is to transfer audio transparently while monitoring for text telephone signals. When valid text telephone signals are detected, the converting action of the node takes effect. The pathc converts between the detected PSTN text telephone method and the text presentation code ITU-T T.140, and between ITU-T T.140 and CTM. This mode of operation continues until text signaling ceases. Then transparent audio transport is re-established, again monitoring for text signals. This way of action allows alternating use of text and voice during the call according to established conventions in text telephony.

3GPP TS 26.226 describes the details of CTM and an indication of how CTM can be combined with a text telephone modem to compose a conversion function in the call path..

ITU-T H.248 Annex F describes the principles of conversion between PSTN Text telephony as in the text telephone Recommendation V.18 and any general real time text conversation feature. So, even if CTM is not mentioned in that Recommendation, its general descriptions are valid for this case. On the PSTN end it is valid also for specific sub-modes of V.18 (Including the US method Baudot 45). The handling is slightly different depending on if the selected V.18 sub-mode is carrier-based or carrier-less, and if the call is known to be with a textphone or being general.
ITU-T H.248 Annex F is also available as an Internet draft.

The descriptions in H.248 Annex F can be taken as functional descriptions of the call path without full implementation in a H.248 environment.

It should be judged if CTM should be added to the transports in H.248 Annex F or its IETF counterpart.

4 Issues with GTT and CTM

The following issues have been brought up in the last 3GPP SA meeting in SP-010174, and by other parties in other situations.

4.1 Service Loop prevention

Conclusion: The method for loop prevention was a concern in an earlier revision of the routing spec. Now solved

with a specific internal CPC value.

In the Mobile Terminating routing case with CAMEL Phase 1 there is a loop going two turns through the CTM-SCP. It is prevented to continue by marking the call with a specific CPC value. The original CPC value is saved and restored in the routing process.

4.2 MMI for emergency text calls is not specified

Conclusion: Differs over the world. For the user it is best to have the same number as anybody. FFS in other fora.

It is asked if all emergency centres accept text calls on the national or regional emergency number.

It is true that the usage varies a lot between how text emergency calls are handled. Some take them to the regional emergency number, some to a specific text emergency number, some through the national text relay service.

It is of course a lengthy process to decide if the countries want any change when text emergency opportunities go mobile.

The draft GTT Stage 1 22.226, leaves it to the operator to configure for emergency calls or not. The routing procedures in the proposed annex to 23.002 are cofigurable.

A proposal is to let this level of standardisation do for the moment. Accept it, becuse it reflects the real situation, and then bring up the issue in any regional fora if there is an interest to harmonize text emergency handling.

4.3 Emergency call recognition.

Conclusion: Emergency calls have info for routing decision, but e.g.Teleservice=Emergency is not available.

When emergency calls come to the MSC for routing, they are in USA marked with CPC H'E0 for "emergency service call". In other regions the real destination number of the actually selected emergency centre is the only reliable indication of emergency.

4.4 Can emergency routing procedures filter out only text user calls?

Conclusion: In regions where the same emergency number is used for text and voice there is no convenient mechanism to select only text emergency calls for treatment in the CTM-SRF.

A selection of only text calls for handling by CTM would reduce the load on the CTM-SRF and reduce concerns about influence on voice call performance and general emergency service performance.

If further reduction of number of calls to handle should be done, it would be important to be able to route only emergency calls from text telephone users to the node.

With current MSC routing capabilities there are no indications to do such routing on. Only if text users have different numbers for emergency it would be possible with existing routing criteria. That is on the other hand a complicated information task to maintain knowledge of different emergency access numbers.

An indication from the terminal when the text user interface is activated would also enable selection of the text emergency calls but that solution also requires further standardisation and intruduction of mechanisms that affects terminals and the core network.

In 3GPP R-4, a series of emergency categories are introduced for submission from the terminals. The required base for selection could be established by extension of this series of categories.

When procedures for selection are studied, it should be kept in mind that a US requirement is that emergency text calls shall be possible to perform from a phone borrowed for the moment.

This item is proposed to be left for further study.

d that is not used by text telephone subscribers.

4.5 Emergency call back

Conclusion: The provided support of text telephone users is sufficient.

A requirement is mentioned from FCC in US to support calls from the emergency services.

For text users – this will work as ordinary text calls to users.
For SIM-less – this is impossible and not a requirement.

For calls to non-subscribed users , that is to phones borrowed to make the emergency call, the procedures do not support selection of these calls to route them to the service node. However, they get limited service anyway by direct transmission of the text telephone signaling in the voice channel. This works as long as the user has the text interface through a PSTN textphone connected to the phone. It is only for these cases that it is realistic to expect to hook the equipment on a borrowed phone. If the user has a built-in user interface or a digital accessory for text, it would be in their own terminal, and therefore supported by the first case. There is no guarantee that when the call comes back the user still has the textphone connected. It can be a follow up the next day, when the text user is far away from the once borrowed phone. Thus the service level provided by the solution seems sufficient.

4.6 Charging

Conclusion: Different tarriffs is a national UK tradition. Possible to support as extra add on programming in CTM-SCP and CTM-SRF.

The issue of different tariffs for text calls is brought up. In UK there is a tradition to give rebate to text users because the calls are much less information dense than than voice calls.

Such operations MAY be supported by the CTM-SCP, in connection with an indication from the CTM-SRF that text was really used in the call.

Such additional functionality is easily achievable as extra service application programming in the rapid development environments that can be used for the CTM-SCP and CTM-SRF.

Since it is just one country who has the tradition today, and it is possible as add on development it is proposed to not do any specific action in standardisation for it.

(for a BSS solution, there is less chance to offer this functionality.)

4.7 Can text telephone users abroad use the service.

Conclusion: The Core Network CAMEL based solution can offer roaming support for text telephone users.

Among the benefits of the CAMEL based service node solution is the opportunity to offer international roaming support.

For Europe, with small countries and largely varying social support for text telephony, it seems critical to be able to offer international roaming. The same for USA where regulations require services to give equal service level to all. (Telecom Act, Section 251). Emergency service support still need local support in the visited network, but that is according to the nature of emergency services.

4.8 Subscriber management

Conclusion: Enabling user subscriptions to the CAMEL service for text telephony can be done with regular user maintenance procedures, and convenient methods for self-registration can be established.

Users must be subscribers of CAMEL with the correct Service Keys and other user data in HLR. Convenient methods to set up this user data are needed.

The user groups have expressed concerns about the need to go through such procedures. A reason is that the information about the procedure must be easily available and maintained. This must be kept in mind when introducing the text telephone support.

For general user registration all HLR vendors have procedures for customer management. Assigning CAMEL parameters is a normal task in such procedures.

A self registration application may be possible, causing transactions to the HLR. It can not be standardised because the HLR management interface is not standard.

4.9 Interface between phone and textphone

Conclusion: A US standard for connector is on the way. For 3GPP, co-ordination around user if with MMS is described in GTT WI. No action yet.

Both physical and higher layer interfaces can be specified between components of the user equipment needed for GTT functionality.

T2 has not yet been active in GTT. The draft Stage 1 suggests that text calls shall be possible from the same user interface equipment as the text part of MMS. There may be a case for a loose connection between the MMS and GTT services in that if a GTT call fails, it can be of interest to offer the opportunity to send a MMS message. Nothing is done here, and it can be discussed what approach to take with the proposals.

In USA there is a discussion about physical interface specifications between handsets and modified PSTN textphones, aiming at an ANSI standard for a 2.5 mm jack interface with specified connections, levels and impedances. (IS-112)

A Unicode (ITU-T T.140) based interface using a Bluetooth profile or a reference interface point could be of interest to enable producers of equipment for people with further disabilities to attach their special devices to form accessible mobile solutions. This could for example be for deaf-blind people or people with severe mobility disabilities. The same interface could normally be used for PDA:s or other digital accessories.

4.10 Can CTM be integrated in a mobile

Conclusion: Yes it is possible to integrate CTM in the Mobile Terminal. DTMF generation from number keys should have priority as ususal.

Most manufacturers have in the US announced their intention to provide handsets with both Baudot45 and CTM functionality so that it can connect to external US textphones.

It is not recommended to change the actions of the number keys to produce DTMF during calls. The CTM implementation would not interfere with DTMF.

When no one is typing, the path is free for using DTMF. If CTM is transmitted, the DTMF may be generated and suppress CTM sound in the network. That is perfectly OK with normal text telephone handling. No conflict seen.

4.11 How is the functional split of the A interface impacted?

Conclusion: No influence for the CN solution.

It is proposed that the CR to 23.002 is sufficient for R-4 and that it can be accompanied by a completed Stage 2 in 23.226 for Rel-5.

For the BSS location, a need for a CR to 08.02 seems logical.

4.12 How do call hold, call wait and MPTY work?

Conclusion: The users will appreciate text supported supplementary services. The CTM-SRF node can provide that, but it is out of scope for the current goals.

Supplementary services are cumbersome or impossible for most deaf people and many hard-of-hearing. That is when they are prompted with voice messages or tones. Composition and timing of answers is impossible when you cannot hear the prompts.

The service node architecture offers an opportunity to develop text support in such services.

The supplementary services procedures are very little influenced by introduction of the CN based CTM conversion.

Call Hold would not be impacted.

Call waiting would require a detector for the signal in the terminal and a visual indication that there is a call waiting, but otherwise would be possible to use in the CTM environment.

MPTY would be tricky with CTM. A one-to-many session would be feasible, while for example a three to three session is harder to perform problem free. Since the network does not know when text is used, no specific action should be taken.

The lack of a modem carrier and the fact that all transmission is in the same transmission channel offers some interesting opportunities for MPTY.

4.13 What are the impacts on TFO?

TBD

4.14 Can TrFO work?

TBD

4.15 Half rate quality not investigated

Conclusion: An investigation of CTM performance in Half Rate should be done. A brief report is expected to the GTT Workshop.

There has been no evaluation of the performance of CTM on GSM Half rate connections. For global application it may be required to have a report about the performance in that condition. The main parameter to verify is the goal to have less than 1% character errors in conditions where voice start to become unpleasant to use. We might even need to ask ITU and 3GPP S4 to include CTM performance in the conditions for testing of future voice coders for CS use.

4.16 How does GTT work in UMTS?

Conclusion: The service node can work as before.

In UMTS, the transcoding moves to the Core Network. Therefore the Core Network location of CTM is the only viable. The emergency service routing, and the CAMEL procedure for selection and routing of calls can be used with the precaution that the routing shall take place on the PSTN side of the transcoding function.

4.17 How are tones and announcements supported?

Conclusion: Important with visual display of tones and network messages in text. Can be provided by the CTM-SRF. ffs.

The CTM-SRF makes it possible to intervene in a call and inject text coded network information or service information.

However hearing people may be text subscribers, but make voice calls without turning on the text user interface. Therefore, before replacing a piece of audible network info or voice announcement with text, the application should check for CTM availability in the terminal. CTM has mechanisms for that.

This is an important item for further study within the capabilities of the service node concept but beyond the ambition level of the current GTT Stage 1. It can be implemented by an operator as an extra service enhancement without further standardisation.

4.18 What is the globally standardised MMI for text to voice and voice to text calls?

Conclusion: Interesting question that can lead to better procedures to invoke relay services. Work on the way in IETF for general IP Multimedia invocation of relay services.

So far, GTT handles only straight transmission of the text and voice media.

Services offering text-to-voice and voice-to-text are of course of interest for the same users as text telephones. Manned such services are in production, called text relay services. They are mentioned as valuable text telephone services in GTT Stage 1.

Automatic translation services have been tried and text-to-voice services are offered in a few areas. These services are not mature to replace the manned services.

Today, most text relay service act through a two stage dialogue. You call the service in one mode, they ask you for the destination and make the call out in the other mode.

For the user, two improvements would be important.

1. It would be convenient to dial the destination and add an indication that you want a relay service included in the call.

2. An opportunity to invoke a relay service in an established call that was addressed directly but either user discovers that they are not compatible in modes and need a relay service in between.

The MMI is TBD, and it would not hurt to start an investigation about it. Some work has been done on this line in IETF for IP text calls and IP Multimedia conversation.