Technical Specification Group Services and System Aspects ***GTT(01)0005***
Global Text Telephony Workshop, Dusseldorf, Germany, 18-19 April 2001

Source: Ericsson

Title: CR to 3GPP TS 23.002. New Annex: Routing of calls through the CTM-SRF for CTM/text telephone conversion.

Document for: Decision

Agenda Item: 5

This Change Request describes possible methods to select and route calls for treatment in a Core Network located function for conversion between legacy PSTN text telephony and GTT voice channel transport with CTM coding.
It is proposed to form a new informative Annex B to 3GPP TS 23.002 Network architecture.
The reason to not make it normative is that the description is detailed for the ISUP type of call control network, and for the CAMEL Phase 1 type of IN service base. This is too restricted to be a normative 3GPP specification. The normative part is proposed for the main body of 23.002 in another CR.

CR-Form-v3

CHANGE REQUEST

⌘	23.002 CR xxx	⌘ rev	-	⌘ Current version:	4.1.1	⌘

*For **HELP** on using this form, see bottom of this page or look at the pop-up text over the ⌘ symbols.*

Proposed change affects: ⌘ (U)SIM [ ] ME/UE [ ] Radio Access Network [ ] Core Network [x]

Title:	⌘	Annex B (informative): Routing of calls through the CTM-SRF for CTM/text telephone conversion.
Source:	⌘	Ericsson
Work item code:	⌘	GTT
Date:	⌘	April 09, 2001
Category:	⌘	**B**
Release:	⌘	REL-4

Use one of the following categories:
F *(essential correction)*
A *(corresponds to a correction in an earlier release)*
B *(Addition of feature),*
C *(Functional modification of feature)*
D *(Editorial modification)*
Detailed explanations of the above categories can be found in 3GPP TR 21.900.

Use one of the following releases:
2 (GSM Phase 2)
R96 (Release 1996)
R97 (Release 1997)
R98 (Release 1998)
R99 (Release 1999)
REL-4 (Release 4)
REL-5 (Release 5)

Reason for change:	⌘	Addition of GTT specific entities
Summary of change:	⌘	Introduction of a new informative Annex detailing possible selection and routing procedures for text telephony calls that need treatment in a CTM/Text telephone conversion function
Consequences if not approved:	⌘	No support for GTT in Release 4

Clauses affected:	⌘	Annex B
Other specs affected:	⌘	[ ] Other core specifications ⌘ [ ] Test specifications [ ] O&M Specifications
Other comments:	⌘	History should be moved to Annex C.

Annex B (informative): Routing of calls through the CTM-SRF for CTM/text telephone conversion.

B.1 Introduction

For text telephone calls in the voice path, a Cellular Text Telephone Modem (CTM) is used to secure text transmission over the radio path. CTM signals are exchanged between the User Equipment and network, where the CTM/Text telephone conversion function can be placed centralized in a CTM-SRF server.
It is a node with routing capabilities and functionality for conversion between fixed network text transmission and CTM.
This document describes a method how to route possible text calls via this CTM-SRF node. It should be taken as an informative example. It is based on CAMEL Phase 1 and call signaling with ISUP. Other call control environments and other IN service platforms can accomplish the same result.

B.2 Scope

The solutions described in this annex is applicable to GSM networks with CAMEL support in GMSC (HPLMN), HLR (in HPLMN), MSC/VLR (in HPLMN/VPLMN). It describes the routing of calls to the CTM-SRF function.
This Annex shows how emergency calls, terminating calls, and originating calls can be routed through the CTM-SRF node without any modification to existing GSM nodes. This covers mobile-to-mobile calls as well.
For cases where emergency call action is wanted, all emergency calls, regardless of whether they are text or speech calls, are routed to the CTM-SRF server. The CTM-SRF server routes the call to the emergency centre. This can be done, as the CTM/textphone conversion itself is transparent to speech.
Originating and terminating calls, from and to possible text telephone users, are handled as CAMEL calls. A CAMEL service assures that the calls are routed through the CTM-SRF node by actions of a CAMEL application.

B.3 Emergency calls

The network has no means to distinguish text emergency calls from voice emergency calls in areas where the same number is used for both types. It is also desired that even a phone borrowed for the purpose of making a text emergency call shall get the text service without any specific text subscription. Therefore, in order to meet these requirements, it shall be possible to configure the network to route all emergency calls through a CTM-SRF server.

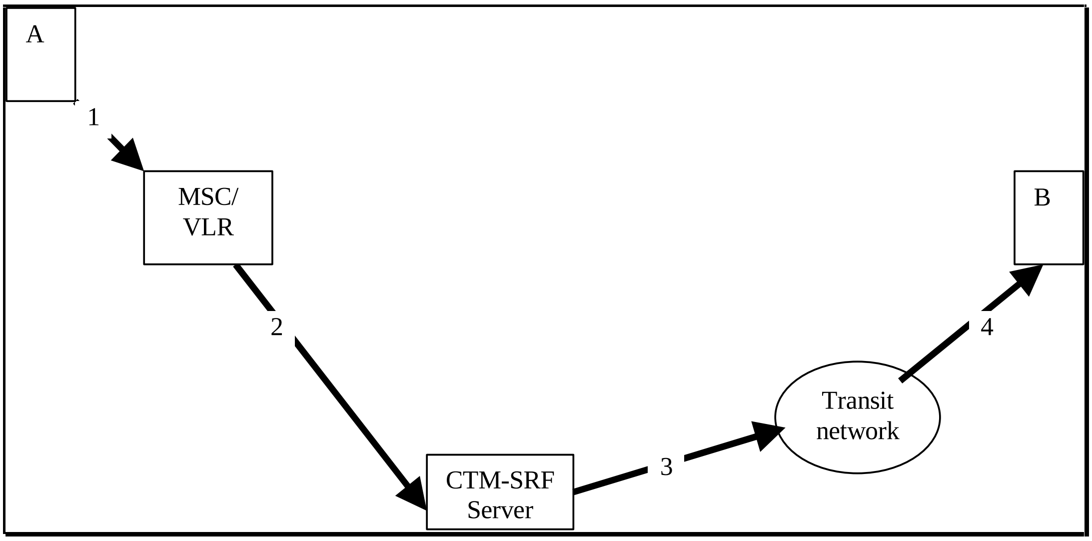

Figure 1 Emergency call routing

If the emergency service use the same number for emergency text calls as for emergency voice calls, routing of all calls, regardless of text or speech, to a CTM-SRF server can be accomplished by configuration in the MSC/VLRs. The MSC/VLR can be configured in a way that, depending on the Emergency Centre addresses, emergency calls are always routed via a CTM-SRF server.

The CTM-SRF links in the CTM/textphone conversion function and routes the call further according to the received IAM.
The CTM-SRF node could recognise emergency calls by knowing all numbers of the emergency centres in question.
For US E-911-calls an additional method is given by looking at the CPC field. A special value is given.

- CPC “H'E0" represents "emergency service call"

By applying one of these methods the CTM-SRF server could also prioritize emergency calls.

In general, the CTM-SRF server handles emergency calls as normal originated calls.

B.4 Routing of regular user calls

Regular user calls that may contain text are routed to the CTM-SRF by a CAMEL service.
Text users are identified by a Text Telephony CAMEL Service Key (SK) and other CAMEL information stored in the HLR. The CTM part of this service modifies the Called Party Address (see following chapters). The call is routed through the CTM-SRF server and is then routed to the original dialed or SCP modified destination. The other service logic then continues as per normal. No further impact is given.

B.5.1 Mobile Terminating calls

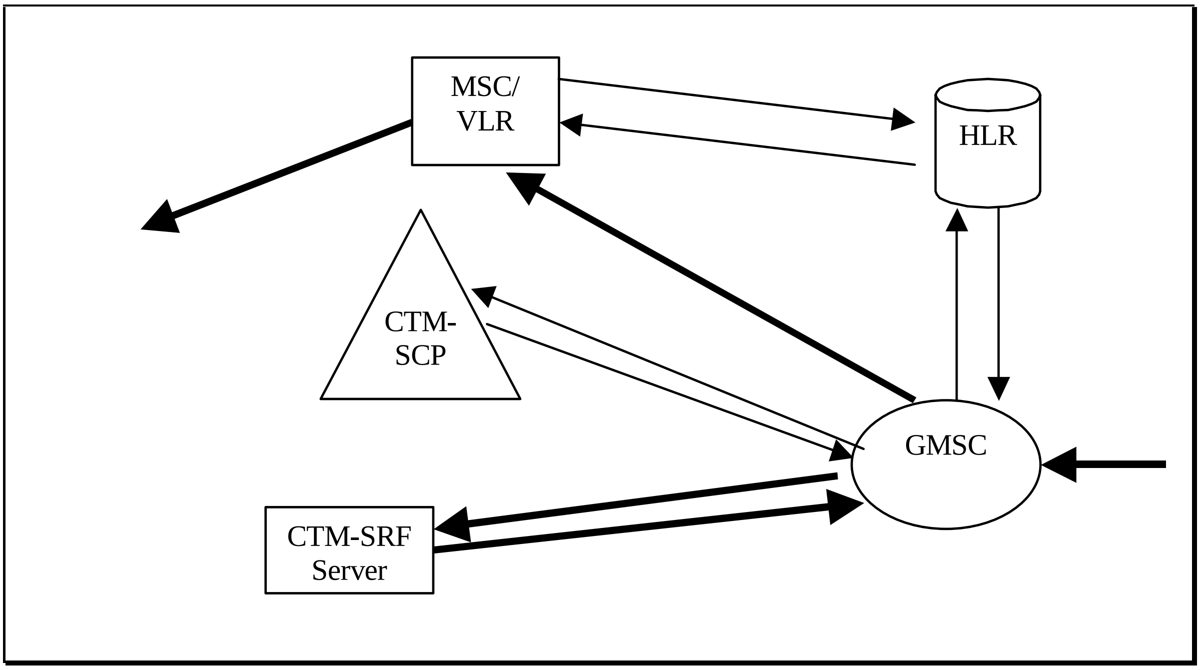

Figure 2 Paths for routing of mobile terminating text calls with call path shown in thick arrows

The GMSC discovers that the user is a subscriber of the text telephony service, by the Camel Subscription Information information received from the HLR. The text Service Key is present and the Detection Point = Terminating_Attempt_Authorized.
The SCP with the text telephony service application is connected and the routing of the call through the CTM-SRF node is performed.

To prevent looping of CAMEL service invocations an indication is sent from the CAMEL service via the CTM-SRF server back to the CAMEL service. The CAMEL service "tunnels" the information that the call has already reached the CAMEL service. This information is used by the CAMEL CTM-SCP service (2nd invocation) to do nothing but just to continue the call.

The indication is carried in the Calling Party Category parameter of ISUP.
This ISUP parameter is supported in CAP V1. InitalDP, and Connect. The CTM-SRF node and the CAMEL CTM-SCP service form an integrated application, and this ISUP parameter handling is regarded internal application signaling.

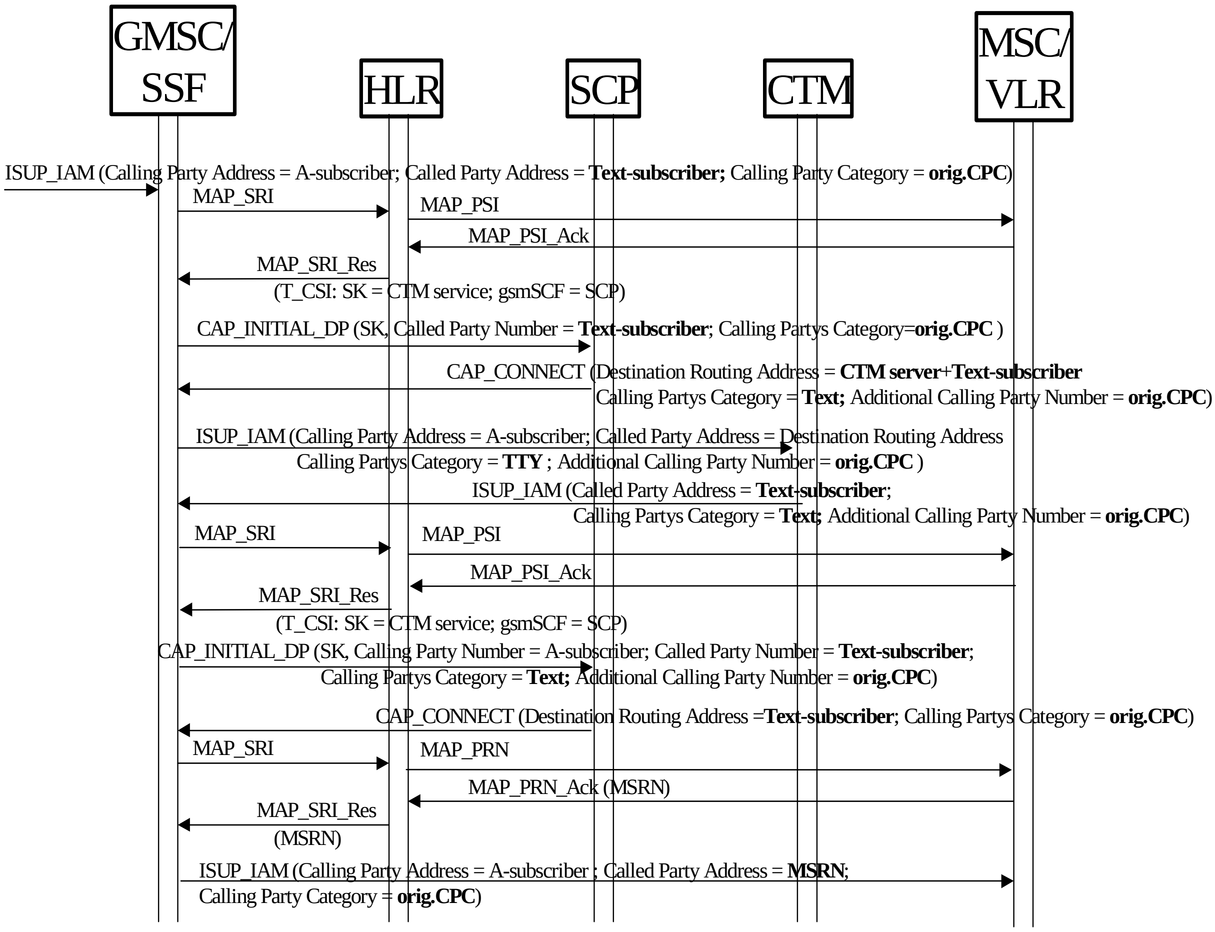

Figure 3 Mobile Terminating Text Call

For the support of text users, the CAMEL CTM-SCP service will return the textphone-number with a CTM-SRF server prefix in CAP_CONNECT. The GMSC creates a new O-BCSM for this CAMEL based forwarding leg. It uses the Destination Routing Address in CAP_CONNECT to do the routing and to send the ISUP_IAM. The routing in based on the CTM-SRF server prefix part.

The CTM-SRF server links in the CTM/textphone conversion function and extracts Text subscriber address, Nature of Address, and the Numbering Plan Indicator from the Destination Routing Address parameter. It stores the original CPC value in Additional Calling Party Number parameter. If Called Party's Category is not received, then this information as such is encoded. It writes this information to the appropriate IAM parameters and routes the call to a GMSC. It could be the same or a different one.

The GMSC regards this incoming IAM as a new terminating call to a text subscriber. A second time a dialogue for the "same text call" to a CAMEL service is invoked. The service realises this fact and does nothing but connecting the call to the Text subscriber (Called Party Number, respectively Destination Routing Address has not been changed by the CTM-SRF server).

B.5.2 Mobile originating calls

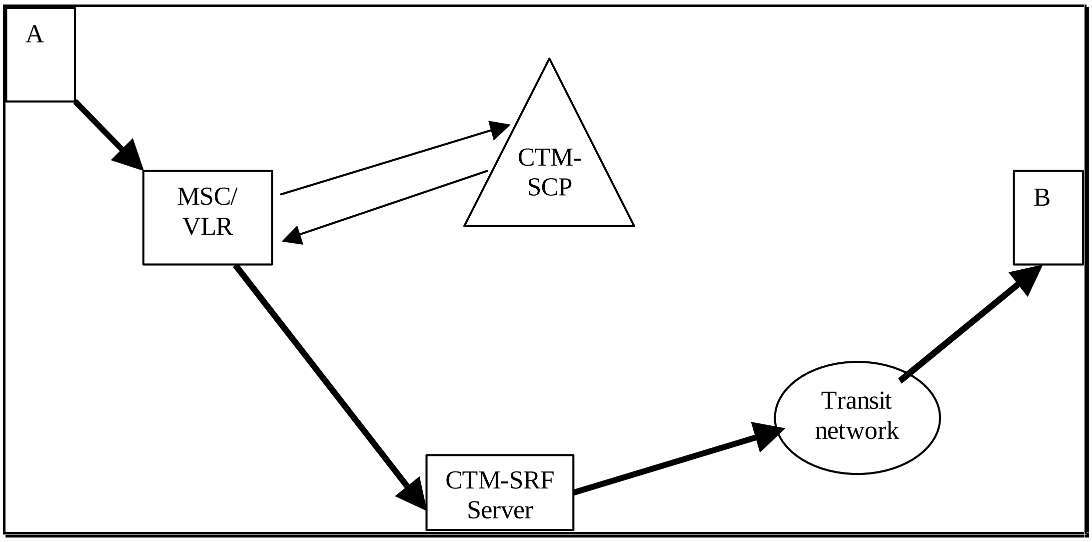

Figure 4 Mobile originating call routing overview

The following sequence chart shows the mobile originating traffic case.

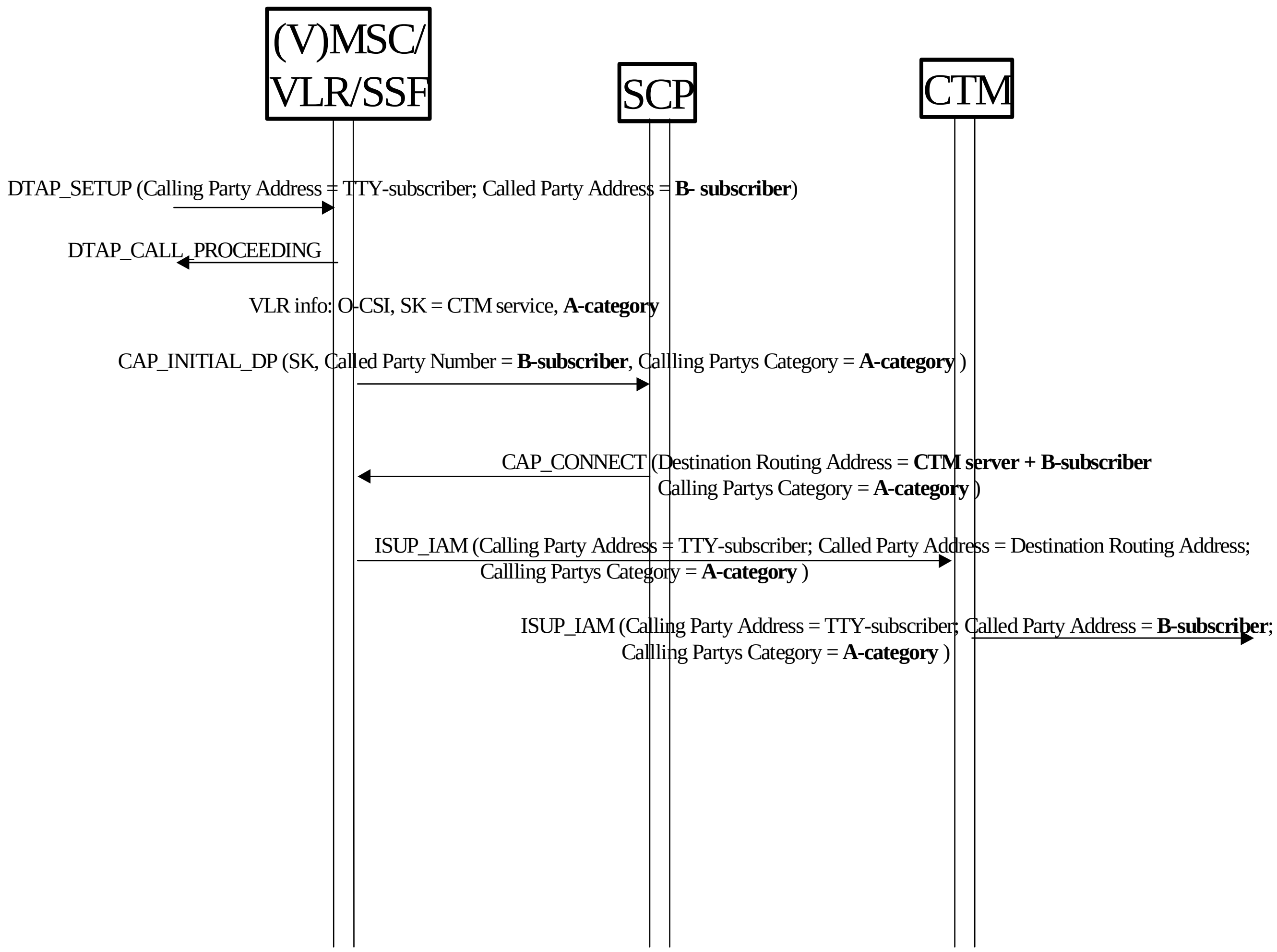

Figure 5 Mobile Originating Text Call

For normal originating calls the MSC finds the CAMEL text telephony Service Key in the VLR, together with information on the Detection Point = Collection of dialled digits. All non-emergency calls for the user will cause a connection to the text telephone service application in the CTM-SCP. The A-category value from the subscriber data in the VLR is used as CPC value. The CAMEL service knows that it is an originating call. In this case it just forwards the Calling Party's Category parameter untouched. This parameter contains either the A-category info.
The CTM-SRF removes its own address digits from the Called Party Address. The CTM/Text telephone conversion function is inserted before the call is routed towards the B-subscriber.

B.5.3 Mobile to Mobile Calls

Two CTM-SRF servers are linked in, one on the originating leg and one on the terminating leg. The originating ISUP_IAM sent from the CTM-SRF server on the outgoing side is received by the GMSC as normal ISUP_IAM.

B.6 Service interactions

B.6.1 Interaction with other CAMEL services

One advantage of the CAMEL based mechanism is that no standardisation is needed. The solution can be installed in the network with minimal impact.

Another big advantage is that the text users can also use other CAMEL services. However, with other CAMEL services using the same Detection Point as the Text Telephone service, the other CAMEL services are impacted when using CAMEL Phase 1 as presented in this document.

As for the impact on existing CAMEL services, some of the services may be impacted anyway to support text users, for example Prepaid for playing announcement in text telephone signals.

Every CAMEL service, which is expected to be used by both voice and text users, and use the same CAMEL detection point as the GTT service, is assigned two SKs, one for non-text users and another for text users. For text users the other service logic has to integrate CTM-SCP routing logic.

B.6.2 Interactions with Supplementary Services

In general, no other interactions are expected than described in the CAMEL standard.

Call Forwarding in GMSC is invoked after terminating CAMEL service invocation. This means the CTM-SRF service node is already linked in the call path. A further invocation of a mobile originated CAMEL based service will cause an additional routing to a CTM-SRF service node. This is needed to convert the text back to text telephone coding, e.g. Baudot, which is required for the following routing towards the forwarded-to-party (C-party).

B.6.3 Emergency Call interaction

Emergency call category is a new 3GPP Release 4 feature (introduced as emergency-call enhancement). It has no impact on routing of emergency calls via CTM-SRF server.

New categories may be specified in order to detect text emergency calls.

B.6.4 Usage of Destination Routing Address

The Destination Routing Address CAMEL parameter is used by the service logic for two purposes.

- It contains the CTM-SRF server address to let the MSC to route the call to the CTM-SRF server.
- It contains the original B-subscriber address, to let the CTM-SRF server to use it as Called Party Number on the outgoing side.

When the service logic reconnects the CTM-SRF server, the service has to save the Nature of Address and the Numbering Plan Indicator of the original Called Party Number in the Destination Routing Address parameter.

CAMEL Phase 1 Destination Routing Address has up to 12 octets. Two octets are needed to encode Nature of Address and the Numbering Plan Indicator. This means that 10 octets are left for CTM-SRF server address digits plus Called Party Address digits. All together 20 digits are given.

A national call has up to 12 digits and an international one up to 15 digits according E.164. This results in having 8 digits, respectively 5 digits left for CTM-SRF server addressing. This seems to be sufficient. In case a pool of CMT servers are shared between several PLMN operators, the National Destination Code (NDC) may be part of the CTM-SRF server address as well. NDC is usually an up to 3 digits code.

B.6.5 Inserting CTM/Text telephone conversion function

The CTM/Text telephone conversion function is direction oriented. A CTM modem should be connected towards the radio side and a modem for V.18 or any of its text telephone submodes towards the other direction. Therefore the CTM-SRF server has to distinguish between mobile terminating and mobile originating calls.

This is achieved by looking the CPC value. For terminating calls the CPC parameter is equal to the special “text telephone”-value set by the CAMEL CTM-SCP routing service. For originating calls the value is different.

B.6.6 Lawful interception

Depending on where on the route the intercept is made, and also depending on the support for CTM in the terminal, the coding in the intercept point will be transmitted in CTM modulation and coding, or native PSTN text telephone coding and modulation. Both are possible to decode, and a combined decoder can be designed.

B.7 Abbreviations

BCSM	Basic Call State Model
CAMEL	Customised Application for Mobile Enhanced Logic
CAP	CAMEL Application Protocol
CLIP	Calling Line Presentation

CPC	Calling Party Category
CTM	Cellular Text Modem
GMSC	Gateway MSC
HPLMN	Home PLMN
ISUP	ISDN User Part
MSC	Mobile Switching Centre
MSRN	Mobile Station Roaming Number
NDC	National Destination Code
PLMN	Public Land Mobile Network
SK	Service Key
TRAU	TRAnscoder Unit
UMTS	Universal Mobile Telecommunication System
VLR	Visited Location Register