3GPP TSG_CN#7 **NP-000104**
ETSI SMG3 Plenary Meeting #7,
Madrid, Spain
13th – 15th March 2000

Agenda item: **5.1.3**

Source: **TSG_N WG1**

Title: **Liaison Statements agreed in CN1 since CN#5**

Introduction:

This document contains liaison statements**,** that have been agreed and sent by **TSG_N WG1,** and are forwarded to **TSG_N Plenary** meeting #7 for information.

LS out from CN1#10 11-14/Jan.2000

Tdoc 3GPP N1-00	Title	Source/ Name	Type/ CR	To	Cc
0135	LS on reply to LS on usage of NSAPI, RB identity, RAB ID and TEID	TSG N1	LS out + N1-000210 + N1-000211	TSG RAN WG3, RAN WG2	SA WG2, CN WG2
0136	Response to Liaison statement concerning HSCSD specifications	TSG N1	LS out	3GPP TSG SA WG1	3GPP TSG SA WG2, 3GPP TSG CN WG3
0138	Response Liaison Statement on Usage of RANAP over MAP/E i/f for UMTS to UMTS Inter-MSC SRNS Relocation	TSG N1	LS out + N1-000199 + N1-000111 + S2-99F02	TSG SA WG2, SMG2	TSG CN WG2, TSG RAN WG 3
0149	LS on questions on the CR 10r1 to TS 23.107	TSG N1	LS out	TSG-S2	TSG S4
0152	Response to LS on RAB linking, Response to LS on RAB pre-emption	TSG N1	LS out	TSG-SA WG1, TSG-RAN WG3	TSG-SA WG2
0153	Response to LS on Capability configuration parameters	TSG N1	LS out	3GPP TSG-T WG3, ETSI SMG9	-
0164	LS to RAN 2, RAN 3, on the Transport of Codec Information during the Codec Negotiation between MS and MSC	TSG N1	LS out + N1-000111 + N1-000163 + N1-000140 + N1-000141 + N1-000033	RAN WG2, RAN WG3, CN WG2	-
0170	LS on removal of Anonymous Access from Release 97 and 98	TSG N1	LS out	TSG-S1	TSG-S2
0182	SAT Handover notification and termination of call	TSG N1	LS out	TSGS1	TSGS3, TSGT3, SMG9 SMG1/9 SAT ad hoc

Tdoc 3GPP N1-00	Title	Source/ Name	Type/ CR	To	Cc
0193	LS on Iu Userplane Initialization at Inter MSC-HO	TSG N1	LS out	SA WG4, RAN WG3, CN WG2	-
0201	Reply to LS on Requirements for Network Selection	TSG N1	LS out + N1-000151	TSG-S1	TSG-S2, TSG-T2, TSG-T1
0202	LS on Enhanced User Identity Confidentiality – open questions	TSG N1	LS out	TSG S3 and TSG N2	-
0205	Response LS on partial SRNS relocation	TSG N1	LS out + N1-99E22	TSG RAN WG3, RAN WG2	-
0209	Reply to Liaison Statement on CR to 23.122 after split in SMG2 and CN1	TSG N1	LS out	SMG2	-
0212	Liaisons for emergency calls	TSG N1	LS out + N1-000038 + N1-000115	S1	-

LS out from CN1 #11 28.Feb - 3.march.2000

Document number	Title	WI	Content	Sent during the meeting	To:	Cc
N1-000403	LS on MS initiated signaling connection release	GSM-UMTS Interworking	LS out	No	3GPP TSG-RAN WG2, 3GPP TSG-RAN WG3	
N1-000443	Response to LS on GPRS Terminal Support of Network Operation Modes I and II	GPRS	LS out	No	GSM Association – IREG GPRSWP V. Chair Scott Bailey	3G W "F inf
N1-000445	LS on reply to Liaison Statement on access signalling and mobile station behaviour for Multicall	Multicall	LS out	Sent during the meeting	TSG CN WG2	TS ho
N1-000446	Question about Idle-mode DRX control		LS out	No	TSG-RAN WG2	TS
N1-000447	2nd LS on the Transport of Codec Information during the Codec Negotiation between MS and MSC	OoBTC	LS out	Sent during the meeting	RAN2, RAN3, CN2B	
N1-000449	USIM triggered authentication and key setting during PS connections	Security	LS out	No	3GPP TSG SA WG3	3G W T W
N1-000450	Response to LS on 5 or 6 digits IMSI HPLMN	USIM	LS out	No	SMG9, T3, T2 SWG1 (MExE)	
N1-000451	LS on TrFO Break procedure (N1-000264 & N1-000367)	OoBTC	LS out	No	N2B	RA
N1-000452	Response on Liaison statement concerning the change of title of 23.060		LS out	Sent to Alain only	3GPP TSG SA WG2	3G W
N1-000453	LS on N1 Working Status of the working plan on OoBTC in R99	OoBTC	LS out	No	TSG SA2	TS RA
N1-000460	Question about Idle-mode DRX control	GPRS	LS out	No	TSG-RAN WG2	TS
N1-000481	LS on Open Issues on Multicall and Proposed Solutions Toward the open Issues	Multicall	LS out	No	TSG CN, TSG CN WG2, TSG CN SSAdoc	TS
N1-000487	LS on RANAP Transaction Sequence	OBTC	LS out	Sent during the meeting	RAN2, RAN3	

N1-000493	LS on MS initiated signaling connection release	GSM-UMTS Interworking	LS out	No	3GPP TSG-RAN WG2, 3GPP TSG-RAN WG3	
N1-000511	LS to CN WG2B proposing a new Specification “Application Part (RANAP) on the E-interface”; 29.108	GSM/UMTS interw	LS Out+ 435	No	CN WG2B	RA
N1-000515	LS to N2 on QoS IE length	QoS	LS out+ CR in N1-000559	No	3GPP N2	3G
N1-000518	LS on AMR modes & Supported Subflow Combinations	OoBTC	LS out	Sent during the meeting	RAN3, S4	N2
N1-000539	Reply to LS on "Introduction of rejection of non ciphered calls for GPRS"	GPRS	LS out	No	TSG-S3/SMG10	TS
N1-000542	Support of Idle-mode DRX control in GMM	GSM UMTS Interw.	LS out + CR	No	TSG-RAN WG2, TSG-RAN WG3	
N1-000553	3rd LS on the Transport of Codec Information during the Codec Negotiation between MS and MSC	OoBTC	LS out/	No	RAN2, RAN3, CN2B	

3GPP TSG-CN-WG1, Meeting #10
11 - 14.January. 2000
Abiko, Japan

Tdoc N1-000135

To: **TSG RAN WG3, RAN WG2**

CC: **SA WG2, CN WG2**

Source: **TSG CN WG1**

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Title: **Reply to LS on usage of NSAPI, RB identity, RAB ID and TEID**

TSG CN WG1 thanks TSG RAN WG3 for their LS on usage of NSAPI, RB identity, RAB ID and TEID.

In R3-99J88, RAN3 asks three questions to CN1. This response to those questions and informs related TSGs CN1 decisions on these topics.

RAN3 comments and CN1 answers are below:

1. The NAS Binding Information IE is provided to RNC in RAB ASSIGNMENT REQUEST both by the PS and CS CN domains.

   Requirement: In addition of being in RAB ASSIGNMENT the NAS Binding Information has to be also exchanged between UE and CN by NAS protocols within the corresponding domain. **(S2+N1?)**

Answer: CN1 is in line with this requirements. CC protocol in CS domain conveys the value of NAS Binding Information from UE to CN as Stream Identifier IE and SM protocol in PS domain conveys it as NSAPI. For detailed information please see the attached CRs (N1-000210, N1-000211).

2. In RAB ASSIGNMENT REQUEST message, regardless of the amount of RAB subflows, a single NAS Binding Information IE is given by CN to RNC.

   Requirement: The NAS Binding Information IE is unique for the UE independently of where it was allocated (e.g. UE, PS CN domain or CS CN domain) **(S2 + N1?)**

Answer: CN1 assumes NAS Binding Info is unique in each domain for the UE.
We believe that this assumption is in line with TS 25.331v3.1.0. In order to separate CS/PS domain protocols cleanly, we assume those protocols in UE don't share the knowledge related to assignment of NAS Binding info values, so to bind an RAB and an NAS protocol UE must receive the NAS Binding information with a CN domain Indication by RB Setup Request.

CN1 kindly asks RAN2 and RAN3 to confirm that this assumption never causes problems.

For detailed information please see the attached CRs to 24.008 (N1-000210, 0211).

3. RAN WG3 would also like to clarify whether the current format assumed in RAN WG3 for the IE NAS Binding Information is suitable for the equivalent of the NAS Binding Information, which is transmitted between UE and CN by NAS protocols in PS and CS domains. The current format for NAS Binding Information IE in RAN WG3 is Octet String containing two octets (i.e. 16 bits). **(N1?)**

Answer: Please see the attached CR (N1-000211) that shows the NAS Binding Information coding format. CN1 kindly asks RAN2 and RAN3 to refer this format in TS 25.331 and TS 25.413.

3GPP/SMG Meeting #10
Abiko, Japan, 11 - 14 January.2000

Document ***N1-000210***
Revised N1-000156

e.g. for 3GPP use the format TP-99xxx
or for SMG, use the format P-99-xxx

CHANGE REQUEST

Please see embedded help file at the bottom of this page for instructions on how to fill in this form correctly.

24.008 CR 032r8 Current Version: 3.2.1

GSM (AA.BB) or 3G (AA.BBB) specification number ↑ *↑ CR number as allocated by MCC support team*

For submission to: CN#7	for approval	X	strategic	X	*(for SMG use only)*
list expected approval meeting # here ↑	for information		non-strategic		

Form: CR cover sheet, version 2 for 3GPP and SMG The latest version of this form is available from: ftp://ftp.3gpp.org/Information/CR-Form-v2.doc

Proposed change affects: (U)SIM [ ] ME [X] UTRAN / Radio [ ] Core Network [X]
(at least one should be marked with an X)

Source: NTT COMMUNICATIONWARE **Date:** 14/01/00

Subject: Addition of the Stream Identifier Information Element

Work item: multicall

Category:				**Release:**	
	F	Correction		Phase 2	
	A	Corresponds to a correction in an earlier release		Release 96	
(only one category	B	Addition of feature	X	Release 97	
shall be marked	C	Functional modification of feature		Release 98	
with an X)	D	Editorial modification		Release 99	X
				Release 00	

Reason for change: Multicall feature shall allow a mobile station to handle more than one bearer service simultaneously. For this case, it is necessary to identify each bearer in order to control the complete call. The name for this element could be Stream Identifier, abbreviated as SI.

Clauses affected: 5.2.1, 5.2.2, 5.2.3, 9.3.2, 9.3.5, 9.3.8, 9.3.23, 10.5.4

Other specs affected:				
	Other 3G core specifications	X	→ List of CRs:	23.008, 23.016, 23.018, 29.002
	Other GSM core specifications		→ List of CRs:	
	MS test specifications		→ List of CRs:	
	BSS test specifications		→ List of CRs:	
	O&M specifications		→ List of CRs:	

Other comments:

Regarding multicall, some open issues were raised in CN#6. But the call control procedure with SI can be discussed independently of the open issues.

According to the stage1, the multicall functionality is optional. Considering the compatibility with GSM R99, SI should be option even in UMTS. The mobile station in UMTS may or may not include the SI for the first call, i.e. when there are no other ongoing calls.

In order to avoid the complexity of network implementation, followings are proposed.

- Value of SI shall be allocated starting from “1”.
- In the case of receiving CC messages (e.g. SETUP, CALL CONFIRMED) with no SI, MSC supporting multicall shall recognise that the value of SI is indicated as “1”.

According to the assumption related to NAS Binding Info value (see N1-000195), SI is independent of session management.

help.doc

<--------- double-click here for help and instructions on how to create a CR.

5 Elementary procedures for circuit-switched Call Control

5.2 Call establishment procedures

5.2.1 Mobile originating call establishment

The call control entity of the mobile station initiates establishment of a CC connection by requesting the MM sublayer to establish a mobile originating MM connection and entering the "MM connection pending" state. There are two kinds of a mobile originating call: basic call and emergency call. The request to establish an MM connection shall contain a parameter to specify whether the call is a basic or an emergency call. This information may lead to specific qualities of services to be provided by the MM sublayers. Timer T303 is started when the CM SERVICE REQUEST message is sent.

For mobile stations supporting eMLPP basic calls may optionally have an associated priority level as defined in GSM 03.67. This information may also lead to specified qualities of service to be provided by the MM sublayers.

While being in the "MM connection pending" state, the call entity of the mobile station may cancel the call prior to sending the first call control message according to the rules given in section 4.5.1.7.

Having entered the "MM connection pending" state, upon MM connection establishment, the call control entity of the mobile station sends a setup message to its peer entity. This setup message is

- a SETUP message, if the call to be established is a basic call, and
- an EMERGENCY SETUP message, if the call to be established is an emergency call.

It then enters the "call initiated" state. Timer T303 is not stopped.

The setup message shall contain all the information required by the network to process the call. In particular, the SETUP message shall contain the called party address information. If multiple traffic channels are supported., the mobile station shall include the Stream Identifier (SI) information element, For the first call, i.e. when there are no other ongoing calls, the SI value 1 (00000001) shall be used.

If the mobile station doesn’t support multiple traffic channels, as an MS option, the Stream Identifier (SI) information may be included in the setup message using the SI value 1 (00000001).

If timer T303 elapses in the "MM connection pending" state, the MM connection in progress shall be aborted and the user shall be informed about the rejection of the call.

5.2.1.1 Call initiation

The “call initiated” state is supervised by timer T303.For normal MO calls, this timer will have already been started after entering the "MM connection pending" state. For network-initiated MO calls this timer will be started in the recall present state as defined in section 5.2.3.4

When the call control entity of the mobile station is in the "call initiated" state and if it receives:

i) a CALL PROCEEDING message, it shall proceed as described in section 5.2.1.3;

ii) an ALERTING message, it shall proceed as described in section 5.2.1.5;

iii) a CONNECT message, it shall proceed as described in section 5.2.1.6;

iv) a RELEASE COMPLETE message it shall proceed as described in section 5.2.1.2.

Abnormal case:

- If timer T303 elapses in the "call initiated" state before any of the CALL PROCEEDING, ALERTING, CONNECT or RELEASE COMPLETE messages has been received, the clearing procedure described in section 5.4 is performed.

5.2.1.2 Receipt of a setup message

In the "null" or "recall present" states, upon receipt of a setup message (a SETUP message or an EMERGENCY SETUP message, see section 5.2.1.1), the call control entity of the network enters the "call initiated" state. It shall then analyse the call information contained in the setup message.

i) If, following the receipt of the setup message, the call control entity of the network determines that the call information received from the mobile station is invalid (e.g. invalid number), then the network shall initiate call clearing as defined in section 5.4 with one of the following cause values:

1 "unassigned (unallocated) number"

3 "no route to destination"

22 "number changed"

28 "invalid number format (incomplete number)"

ii) If, following the receipt of the setup message, the call control entity of the network determines that a requested service is not authorized or is not available, it shall initiate call clearing in accordance with section 5.4.2 with one of the following cause values:

8 "operator determined barring",

57 "bearer capability not authorized",

58 "bearer capability not presently available",

63 "service or option not available, unspecified", or

65 "bearer service not implemented".

iii) The network receiving the SETUP message with no SI shall regard the SI value as 1.

If the network does not support multiple traffic channels, and receives the SETUP message with any other than SI value 1, the call control entity of the network shall initiate call clearing with cause #69 "requested facility not implemented".

In the following cases, the call control entity of the network shall initiate call clearing with cause #95 " semantically incorrect message ".

- the network receives the SETUP message with any other than SI value 1 for the first call

- or: the network receives the SETUP message with SI information element including invalid content

iv~~ii~~) Otherwise, the call control entity of the network shall either:

- send a CALL PROCEEDING message to its peer entity to indicate that the call is being processed; and enter the "mobile originating call proceeding" state.

- or: send an ALERTING message to its peer entity to indicate that alerting has been started at the called user side; and enter the "call received" state.

- or: send a CONNECT message to its peer entity to indicate that the call has been accepted at the called user side; and enter the "connect request" state.

The call control entity of the network may insert bearer capability information element(s) in the CALL PROCEEDING message to select options presented by the mobile station in the Bearer Capability information element(s) of the SETUP message. The bearer capability information element(s) shall contain the same parameters as received in the SETUP except those presenting a choice. Where choices were offered, appropriate parameters indicating the results of those choices shall be included.

The CALL_PROCEEDING message may also contain the priority of the call in the case where eMLPP is applied and where the network has assigned a different priority to the call than that requested by the user, or where the user has not requested a priority and the network has assigned a default priority. Mobile stations supporting eMLPP shall indicate this priority level to higher sublayers and store this information for the duration of the call for further action. Mobile stations not supporting eMLPP shall ignore this information element if provided in a CALL PROCEEDING message.

The call control entity of the network having entered the "mobile originating call proceeding" state, the network may initiate the assignment of a traffic channel according to section 5.2.1.9 (early assignment).

```
   MS                     Network
 +------------------------------+
 |                              |
 |     (EMERGENCY) SETUP        |
 |   ---------------------->    |
 |                              |
 +------------------------------+

        CALL_PROCEEDING           (i)
   < - - - - - - - - - - - - -
          ALERTING               (ii)
   < - - - - - - - - - - - - -
          CONNECT                (iii)
   < - - - - - - - - - - - - -
        RELEASE COMPLETE          (iv)
   < - - - - - - - - - - - - -
```

Figure 5.2/TS 24.008 Mobile originated call initiation and possible subsequent responses.

****** Next Modified Section ******

5.2.1.9 Traffic channel assignment at mobile originating call establishment

If the mobile station includes the Stream Identifier (SI) in the SETUP message, the SI indicates one of the following.

a) Mobile station generates a new SI value at the initiation of an originating call, then a new traffic channel shall be assigned to the mobile originating call.

b) Mobile station indicates an existing SI value, then the indicated traffic channel shall be used for the mobile originating call.

It is a network dependent decision when to initiate the assignment of an appropriate traffic channel during the mobile originating call establishment phase. Initiation of a suitable RR procedure to assign an appropriate traffic channel does neither change the state of a call control entity nor affect any call control timer.

NOTE: During certain phases of such an RR procedure, transmission of CC and MM messages may be suspended, see GSM 04.18, section 3 and GSM 08.08.

The assignment procedure does not affect any call control timer.

****** Next Modified Section ******

5.2.2 Mobile terminating call establishment

5.2.2.3 Call confirmation

5.2.2.3.1 Response to SETUP

Having entered the "call present state" the call control entity of the mobile station shall - with the exception of the cases described below - acknowledge the SETUP message by a CALL CONFIRMED message, and enter the "mobile terminating call confirmed" state.

If multiple traffic channels are supported., the mobile station shall include the Stream Identifier (SI) information element in the CALL CONFIRMED message, For the first call, i.e. when there are no other ongoing calls, the SI value 1 (00000001) shall be used.

If the mobile station doesn’t support multiple traffic channels, as an MS option, the Stream Identifier (SI) information may be included in the CALL CONFIRMED message using the SI value 1 (00000001).

The call control entity of the mobile station may include in the CALL CONFIRMED message to the network one or two bearer capability information elements to the network, either preselected in the mobile station or corresponding to a service dependent directory number (see TS 29.007). The mobile station may also include one or two bearer capabilities in the CALL CONFIRMED message to define the radio channel requirements. In any case the rules specified in section 9.3.2.2 shall be followed.

NOTE: The possibility of alternative responses (e.g., in connection with supplementary services) is for further study.

A busy MS which satisfies the compatibility requirements indicated in the SETUP message shall respond either with a CALL CONFIRMED message if the call setup is allowed to continue or a RELEASE COMPLETE message if the call setup is not allowed to continue, both with cause #17 "user busy".

If the mobile user wishes to refuse the call, a RELEASE COMPLETE message shall be sent with the cause #21 "call rejected" .

In the cases where the mobile station responds to a SETUP message with RELEASE COMPLETE message the mobile station shall release the MM connection and enter the "null" state after sending the RELEASE COMPLETE message.

The network shall process the RELEASE COMPLETE message in accordance with section 5.4.

5.2.2.3.2 Receipt of CALL CONFIRMED and ALERTING by the network

The call control entity of the network in the "call present" state, shall, upon receipt of a CALL CONFIRMED message: stop timer T303, start timer T310 and enter the "mobile terminating call confirmed" state.

The network receiving the CALL CONFIRMED message with no SI shall regard the SI value as 1. If the network does not support multiple traffic channels, and receives the CALL CONFIRMED message with any other than SI value 1, the call control entity of the network shall initiate call clearing with cause #69 "requested facility not implemented". In the following cases, the call control entity of the network shall initiate call clearing with cause #95 " semantically incorrect message ".

- the network receives the CALL CONFIRMED message with any other than SI value 1 for the first call
- or: the network receives the CALL CONFIRMED message with SI information element including invalid content

The call control entity of the mobile station having entered the "mobile terminating call confirmed" state, if the call is accepted at the called user side, the mobile station proceeds as described in 5.2.2.5. Otherwise, if the signal information element was present in the SETUP message user alerting is initiated at the mobile station side; if the signal information element was not present in the SETUP message, user alerting is initiated when an appropriate channel is available.

Here, initiation of user alerting means:

- the generation of an appropriate tone or indication at the mobile station; and
- sending of an ALERTING message by the call control entity of the MS to its peer entity in the network and entering the "call received" state.

The call control entity of the network in the "mobile terminated call confirmed" state shall, upon receipt of an ALERTING message: send a corresponding ALERTING indication to the calling user; stop timer T310; start timer T301, and enter the "call received" state.

In the "mobile terminating call confirmed" state or the "call received" state, if the user of a mobile station is User Determined User Busy then a DISCONNECT message shall be sent with cause #17 "user busy". In the "mobile terminating call confirmed" state, if the user of a mobile station wishes to reject the call then a DISCONNECT message shall be sent with cause #21 "call rejected".

5.2.2.3.3 Call failure procedures

In case of abnormal behaviour the following call failure procedures apply:

i. If the network does not receive any response to the SETUP message prior to the expiration of timer T303, then the network shall: initiate clearing procedures towards the calling user with cause #18 "no user responding"; and initiate clearing procedures towards the called mobile station in accordance with 5.4.4 using cause #102 "recovery on timer expiry".

ii. If the network has received a CALL CONFIRMED message, but does not receive an ALERTING, CONNECT or DISCONNECT message prior to the expiration of timer T310, then the network shall:

- initiate clearing procedures towards the calling user with cause #18 "no user responding"; and
- initiate clearing procedures towards the called MS in accordance with section 5.4.4 using cause #102 "recovery on timer expiry".

iii. If the network has received an ALERTING message, but does not receive a CONNECT or DISCONNECT message prior to the expiry of timer T301 (or a corresponding internal alerting supervision timing function), then the network shall: initiate clearing procedures towards the calling user with cause #19 "user alerting, no answer"; and initiate clearing procedures towards the called mobile station in accordance with section 5.4.4, using cause #102 "recovery on timer expiry" or using cause #31 "normal, unspecified".

NOTE: The choice between cause #31 and cause #102 may have consequences on indications generated by the mobile station, see GSM 02.40.

5.2.2.3.4 Called mobile station clearing during mobile terminating call establishment

See section 5.4.2.

****** Next Modified Section ******

5.2.2.5 Call accept

In the "mobile terminating call confirmed" state or the "call received" state, the call control entity in the mobile station indicates acceptance of a mobile terminating call by:

- sending a CONNECT message to its peer entity in the network;
- starting Timer T313; and
- entering the "connect request" state.

If the call control entity of the mobile station has indicated “No Bearer” as the SI value in the CALL CONFIRMED message, it shall assign the SI value and include the SI information element in the CONNECT message.

5.2.2.6 Active indication

In the "mobile terminated call confirmed" state or in the "call received" state, the call control entity of the network shall, upon receipt of a CONNECT message: through connect the traffic channel (including the connection of an interworking function, if required), stop timers T310, T303 or T301 (if running); send a CONNECT ACKNOWLEDGE message to its peer entity at the mobile station of the called user; initiate procedures to send a CONNECT message towards the calling user and enter the "active" state.

In the "connect request" state, the call control entity of the mobile station shall, upon receipt of a CONNECT ACKNOWLEDGE message: stop timer T313 and enter the "active" state.

When timer T313 expires prior to the receipt of a CONNECT ACKNOWLEDGE message, the mobile station shall initiate clearing in accordance with section 5.4.3.

If the network has received “No Bearer” as the SI value in the CALL CONFIRMED message, but does not receive the SI information element in the CONNECT message, then the network shall initiate clearing procedures with cause # 96 "invalid mandatory information".

If the network receives the different SI value between in the CALL CONFIRMED message and in the CONNECT message, it shall initiate call clearing with cause #95 " semantically incorrect message ".

```
MS                    Network
+------------------------+
|       CONNECT          |
|  --------------------> |
|   CONNECT ACKNOWLEDGE  |
|  <-------------------- |
+------------------------+
```

Figure 5.7/TS 24.008 Call acceptance and active indication at mobile terminating call establishment

5.2.2.7 Traffic channel assignment at mobile terminating call establishment

The mobile station may either require a new traffic channel or, use an existing traffic channel after receiving the SETUP message. The mobile station identifies that traffic channel as the SI value in the first message sent in response to the SETUP message. In this case, the SI indicates one of the following.

a) Mobile station generates a new SI value, then a new traffic channel shall be assigned to the mobile terminated call.

b) Mobile station indicates an existing SI value, then the indicated traffic channel shall be used for the mobile terminated call.

The mobile station can indicate "No Bearer" in the SI information element in the first message sent in response to the SETUP message. The traffic channel shall be then established when the SI value is indicated in the CONNECT message.

It is a network dependent decision when to initiate the assignment of a traffic channel during the mobile terminating call establishment phase.

Initiation of the assignment phase does not directly change the state of a CC entity nor affect any call control timer, but may have some secondary effects (see e.g. clause 5.2.2.3.2).

5.2.2.8 Call queuing at mobile terminating call establishment

The principles described in section 5.2.1.10 apply accordingly.

NOTE: The interworking to the fixed network has to fulfil the network specific requirements.

5.2.2.9 User connection attachment during a mobile terminating call

For speech calls:

The mobile station shall attach the user connection at latest when sending the connect message.

For data calls:

The mobile station shall attach the user connection when receiving the CONNECT ACKNOWLEDGE message from the network.

****** Next Modified Section ******

5.2.3 Network initiated MO call $(CCBS)$

The procedures of section 5.2.3 are mandatory for mobile stations supporting "Network initiated MO call".

NOTE: The behaviour of a mobile station that does not support "Network initiated MO call" is described in section 4.

5.2.3.1 Initiation

Before call establishment can be initiated in the mobile station, the MM connection shall be established by the network.

After the arrival of an appropriate stimulus (for example a Remote User Free Indication), the corresponding call control entity in the network shall initiate the MM connection establishment according to section 4, enter the "CC connection pending" state and start timer T331. The request to establish the MM connection is passed from the CM sublayer to the MM sublayer. It contains the necessary routing information derived from the received stimulus.

Upon completion of the MM connection, the call control entity of the mobile station shall send a START CC message to its peer entity in the network. The mobile station shall then enter the “Wait for network information” state and start timer T332.

If the network receives a START CC message while in the “CC connection pending” state, the network stops T331, sends the CC-ESTABLISHMENT message, starts timer T333 and enters the “CC-establishment present” state.

The MM connection establishment may be unsuccessful for a variety of reasons, in which case the MM sublayer in the network will inform the CC entity in the network with an indication of the reason for the failure. The CC entity shall then stop all running timers, enter the “Null” state and inform all appropriate entities within the network.

If timer T331 expires, the network shall abort the MM connection establishment attempt, stop all running CC timers, enter the “Null” state and inform all appropriate entities within the network.

5.2.3.2 CC-Establishment present

In the “CC establishment present” state, the mobile station, upon receipt of the CC-ESTABLISHMENT message, shall stop timer T332.

The CC-ESTABLISHMENT message contains information which the mobile station shall use for the subsequent SETUP message (if any) related to this CC-ESTABLISHMENT.

The CC-ESTABLISHMENT message shall contain the *Setup Container IE*.

If no CC-ESTABLISHMENT message is received by the call control entity of the mobile station before the expiry of timer T332, then the mobile station shall initiate clearing procedures towards the network using a RELEASE COMPLETE message with cause #102 "recovery on timer expiry" and proceed in accordance with section 5.4.2.

Upon receipt of a CC-ESTABLISHMENT message the mobile station shall perform checks on the Setup Container IE in order to align the contained information with the mobile’s present capabilities and configuration. The “recall alignment procedure” is defined later on in this section.

If the recall alignment procedure has succeeded, the call control entity of the Mobile Station shall:

- form and store the SETUP message for sending later in the “Recall present” state,
- acknowledge the CC-ESTABLISHMENT message with a CC-ESTABLISHMENT CONFIRMED message,
- start timer T335, and
- enter the “CC-establishment confirmed“ state.

Exception:

> A busy mobile station which has successfully performed the recall alignment procedure shall respond with a CC-ESTABLISHMENT CONFIRMED message with cause #17 "user busy", and proceed as stated above.

A mobile station, for which the recall alignment procedure failed, shall respond with a RELEASE COMPLETE message in accordance with section 5.4.2 with the appropriate cause code as indicated in the description of the recall alignment procedure.

The SETUP message is constructed from the *Setup Container IE* received in the CC ESTABLISHMENT MESSAGE. The mobile station shall assume that the *Setup Container IE* contains an entire SETUP message with the exception of the Protocol Discriminator, Transaction ID and Message Type elements. The mobile station may assume that the contents of the *Setup Container IE* are the same as were sent from the subscriber in a previous SETUP message of the mobile originating call establishment attempt. The mobile station shall copy the *Setup Container* to the SETUP message and not modify the contents except as defined in the recall alignment procedure and as defined in *exceptions* below. The mobile station shall not add other Information Elements to the end of the SETUP message.

Exceptions:

*Bearer Capability IE(s), HLC IE(s) and LLC (s) IE(s) (*including *Repeat Indicator(s)*, if there are 2 bearer capabilities) require handling as described in the recall alignment procedure below.

If the *CC Capabilities* in the *Setup Container IE* is different to that supported by the mobile station, the mobile station shall modify the *CC Capabilities* in the SETUP message to indicate the true capabilities of the mobile station.

Facility IE(s) and SS Version IE(s) require handling as described in the recall alignment procedure.

Stream Identifier IE requires handling as described in the recall alignment procedure.

If no response to the CC-ESTABLISHMENT message is received by the call control entity of the network before the expiry of timer T333, then the network shall initiate clearing procedures towards the called mobile station using a RELEASE COMPLETE message with cause #102 "recovery on timer expiry" and inform all appropriate entities within the network, proceeding in accordance with section 5.4.2.

```
 MS                          Network
+----------------------------+
|                            |
|     CC-ESTABLISHMENT       |
|        <---------------    |
|                            |
|  CC-ESTABLISHMENT_CONFIRMED|    (i)
|  - - - - - - - - - - - - > |
|      RELEASE COMPLETE      |    (ii)
|  - - - - - - - - - - - - > |
+----------------------------+
```

Figure 5.7a/TS 24.008 Call initiation and possible subsequent responses.

5.2.3.2.1 Recall Alignment Procedure

The recall alignment procedure consists of three~~two~~ parts :

- basic service group alignment,~~ and~~
- facility alignment, and
- stream identifier alignment.~~.~~

Basic service group alignment:

The mobile station shall check that the *Bearer Capability*, *HLC* and *LLC* and *Repeat Indicator* fields, which are embedded in the *Setup Container IE,* match a basic service group supported by the mobile station.

If this check fails, then the recall alignment procedure has failed. The mobile station shall use the cause #88 “incompatible destination” afterwards.

Otherwise, the mobile station is allowed to alter the content within the *Bearer Capability*, *HLC* and *LLC* Information Elements (e.g. the speech coder version(s), the data rate, the radio channel requirement) provided that the basic service group is not changed. The result shall be that the mobile station has derived *Bearer Capability, HLC* and *LLC* Information Elements, which it can use for a later call setup according to its configuration and capabilities.

Facility alignment:

This only applies if the *Setup Container* contains 1 or more *Facility IEs.* Each *Facility IE* within the *Setup Container* will be associated with the common *SS Version IE*, if present. The handling for each *Facility IE* is defined below. The mobile station shall align each facility IE contained in the *Setup Container.* The rules defined in GSM 04.10 also apply.

The *Facility IE* is encoded as ‘simple recall alignment’, ‘advanced recall alignment’ or ‘recall alignment not essential’ (see GSM 04.10). If the encoding indicates, that

- a simple recall alignment is required, the mobile station shall copy the Facility IE and the common SS version IE from the *Setup Container* to the SETUP message without modifying the content.

- an advanced recall alignment is required, the mobile station must recognise and support the operation defined in the facility. If the mobile station does not recognise or support the operation, then the recall alignment procedure has failed and the mobile station shall use the cause #29 “facility rejected” in the subsequent rejection of the CC establishment request.
- the recall alignment is not essential, then the facility operation is not an essential part of the SETUP. If the MS does not recognise the operation then the SS Version IE and Facility IE are discarded, and NOT copied into the SETUP message.

NOTE. A mobile station may include a *Facility IE* without an associated *SS Version IE.* This would indicate that the SS operation is encoded using Phase 1 protocols.

Further details on Facility handling are given in GSM 04.10

Stream identifier alignment:

The mobile station shall check whether the *Stream Identifier* field is contained in the *Setup Container* or not.

If the *Stream Identifier* is contained in the *Setup Container*, the mobile station shall behave as one of the following.

- the mobile station re-assign the *Stream Identifier* value, and modify the *Stream Identifier* field.
- the mobile station remove the *Stream Identifier* field.

If the *Stream Identifier* is not contained in the *Setup Container*, the mobile station may behave as follows.

- the mobile station assign the *Stream Identifier* value, and add the *Stream Identifier IE* to the end of the SETUP message.

****** Next Modified Section ******

9 Message functional definitions and contents

9.3 Messages for circuit-switched call control

9.3.2 Call confirmed

This message is sent by the called mobile station to confirm an incoming call request.

See table 9.56/TS 24.008.

Message type: CALL CONFIRMED

Significance: local

Direction: mobile station to network

Table 9.56/TS 24.008: CALL CONFIRMED message content

IEI	Information element	Type / Reference	Presence	Format	Length
	Call control protocol discriminator	Protocol discriminator 10.2	M	V	1/2
	Transaction identifier	Transaction identifier 10.3.2	M	V	1/2
	Call confirmed message type	Message type 10.4	M	V	1
D-	Repeat Indicator	Repeat Indicator 10.5.4.22	C	TV	1
04	Bearer capability 1	Bearer capability 10.5.4.5	O	TLV	3-16
04	Bearer capability 2	Bearer capability 10.5.4.5	O	TLV	3-16
08	Cause	Cause 10.5.4.11	O	TLV	4-32
15	CC Capabilities	Call Control Capabilities 10.5.4.5a	O	TLV	3
2D	Stream Identifier	Stream Identifier 10.5.4.XX	O	TLV	3

9.3.2.1 Repeat indicator

The *repeat indicator* information element shall be included if *bearer capability 1* information element and *bearer capability 2* IE are both included in the message.

9.3.2.2 Bearer capability 1 and bearer capability 2

The *bearer capability 1* information element shall be included if and only if at least one of the following five cases holds:

- the mobile station wishes another bearer capability than that given by the *bearer capability 1* information element of the incoming SETUP message;
- the *bearer capability 1* information element is missing or not fully specified in the SETUP message;
- the *bearer capability 1* information element received in the SETUP message is accepted and the "radio channel requirement" of the mobile station is other than "full rate support only mobile station";
- the *bearer capability 1* information element received in the SETUP message indicates speech and is accepted and the mobile station supports other speech versions than GSM version 1;
- the *bearer capability 1* information element received in the SETUP message included the “fixed network user rate“ parameter.

When the *bearer capability 1* information element is followed by the *bearer capability 2* IE in the SETUP, the above rules apply to both *bearer capability* 1 IE and bearer capability 2 IE. Except those cases identified in TS 27.001, if either *bearer capability* needs to be included, both shall be included.

Furthermore, both *bearer capability* information elements may be present if the mobile station wishes to reverse the order of occurrence of the *bearer capability* information elements (which is referred to in the *repeat indicator* information element, see section 10.5.4.22) in cases identified in TS 27.001.

9.3.2.3 Cause

This information element is included if the mobile station is compatible but the user is busy.

9.3.2.4 CC Capabilities

This information element may be included by the mobile station to indicate its call control capabilities.

9.3.2.5 Stream Identifier

This information element shall be included by the mobile station supporting multiple traffic channels. The purpose of the information element is to indicate whether a new traffic channel shall be assigned to the call.

****** Next Modified Section ******

9.3.5 Connect

9.3.5.2 Connect (mobile station to network direction)

This message is sent by the called mobile station to the network to indicate call acceptance by the called user.

See table 9.59a/TS 24.008.

Message type: CONNECT

Significance: global

Direction: mobile station to network

Table 9.59a/TS 24.008: CONNECT message content (mobile station to network direction)

IEI	Information element	Type / Reference	Presence	Format	Length
	Call control protocol discriminator	Protocol discriminator 10.2	M	V	1/2
	Transaction identifier	Transaction identifier 10.3.2	M	V	1/2
	Connect message type	Message type 10.4	M	V	1
1C	Facility	Facility 10.5.4.15	O	TLV	2-?
4D	Connected subaddress	Connected subaddress 10.5.4.14	O	TLV	2-23
7E	User-user	User-user 10.5.4.25	O	TLV	3-131
7F	SS version	SS version indicator 10.5.4.24	O	TLV	2-3
2D	Stream Identifier	Stream Identifier 10.5.4.XX	O	TLV	3

9.3.5.2.1 Facility

This information element may be used for functional operation of supplementary services.

9.3.5.2.2 User-user

This information element is included when the answering mobile station wants to return user information to the calling remote user.

9.3.5.2.3 SS version

This information element shall not be included if the *facility* information element is not present in this message.

This information element shall be included or excluded as defined in TS 24.010. This information element should not be transmitted unless explicitly required by TS 24.010.

9.3.5.2.4 Stream Identifier

This information element shall be included when a mobile station has indicated “No Bearer” as the SI value in the CALL CONFIRMED message.

****** Next Modified Section ******

9.3.8 Emergency setup

This message is sent from the mobile station to initiate emergency call establishment.

See table 9.62/TS 24.008.

Message type: EMERGENCY SETUP

Significance: global

Direction: mobile station to network

Table 9.62/TS 24.008: EMERGENCY SETUP message content

IEI	Information element	Type / Reference	Presence	Format	Length
	Call control protocol discriminator	Protocol discriminator 10.2	M	V	1/2
	Transaction identifier	Transaction identifier 10.3.2	M	V	1/2
	Emergency setup message type	Message type 10.4	M	V	1
04	Bearer capability	Bearer capability 10.5.4.5	O	TLV	3-9
2D	Stream Identifier	Stream Identifier 10.5.4.XX	O	TLV	3

9.3.8.1 Bearer capability

If the element is not included, the network shall by default assume speech and select full rate speech version 1. If this information element is included, it shall indicate speech, the appropriate speech version(s) and have the appropriate value of radio channel requirement field.

9.3.8.2 Stream Identifier

This information element shall be included by the mobile station supporting multiple traffic channels. The purpose of the information element is to indicate whether a new traffic channel shall be assigned to the call.

****** Next Modified Section ******

9.3.23 Setup

9.3.23.2 Setup (mobile originating call establishment)

This message is sent from the mobile station to the network to initiate a mobile originating call establishment.

See table 9.70a/TS 24.008.

Message type: SETUP

Significance: global

Direction: mobile station to network

Table 9.70a/TS 24.008: SETUP message content (mobile station to network direction)

IEI	Information element	Type / Reference	Presence	Format	Length
	Call control protocol discriminator	Protocol discriminator 10.2	M	V	1/2
	Transaction identifier	Transaction identifier	M	V	1/2

		10.3.2			
	Setup message type	Message type 10.4	M	V	1
D-	BC repeat indicator	Repeat indicator 10.5.4.22	C	TV	1
04	Bearer capability 1	Bearer capability 10.5.4.5	M	TLV	3-16
04	Bearer capability 2	Bearer capability 10.5.4.5	O	TLV	3-16
1C	Facility(simple recall alignment)	Facility 10.5.4.15	O	TLV	2-
5D	Calling party sub- address	Calling party subaddr. 10.5.4.10	O	TLV	2-23
5E	Called party BCD number	Called party BCD num. 10.5.4.7	M	TLV	3-43
6D	Called party sub- address	Called party subaddr. 10.5.4.8	O	TLV	2-23
D-	LLC repeat indicator	Repeat indicator 10.5.4.22	O	TV	1
7C	Low layer compatibility I	Low layer comp. 10.5.4.18	O	TLV	2-18
7C	Low layer compatibility II	Low layer comp. 10.5.4.18	O	TLV	2-18
D-	HLC repeat indicator	Repeat indicator 10.5.4.22	O	TV	1
7D	High layer compatibility i	High layer comp. 10.5.4.16	O	TLV	2-5
7D	High layer compatibility ii	High layer comp. 10.5.4.16	O	TLV	2-5
7E	User-user	User-user 10.5.4.25	O	TLV	3-35
7F	SS version	SS version indicator 10.5.4.24	O	TLV	2-3
A1	CLIR suppression	CLIR suppression 10.5.4.11a	C	T	1
A2	CLIR invocation	CLIR invocation 10.5.4.11b	C	T	1

15	CC capabilities	Call Control Capabilities 10.5.4.5a	O	TLV	3
1D	Facility $(CCBS)$ (advanced recall alignment)	Facility 10.5.4.15	O	TLV	2-?
1B	Facility (recall alignment Not essential) $(CCBS)$	Facility 10.5.4.15	O	TLV	2-?
2D	Stream Identifier	Stream Identifier 10.5.4.XX	O	TLV	3

9.3.23.2.1 BC repeat indicator

The *BC repeat indicator* information element is included if and only if *bearer capability 1* IE and *bearer capability 2* IE are both present in the message.

9.3.23.2.2 Facility

The information element may be included for functional operation of supplementary services.

Three different codings of this IE exist, for further details see 04.10.

9.3.23.2.3 LLC repeat indicator

The *LLC repeat indicator* information element is included if and only if both following conditions hold:

- The *BC repeat indicator* IE is contained in the message.
- The *low layer compatibility I* IE is contained in the message.

If included, the *LLC repeat indicator* shall specify the same repeat indication as the *BC repeat indicator* IE.

9.3.23.2.4 Low layer compatibility I

The information element is included in the MS-to-network direction when the calling MS wants to pass low layer compatibility information to the called user.

9.3.23.2.5 Low layer compatibility II

Included if and only if the *LLC repeat indicator* information element is contained in the message.

9.3.23.2.6 HLC repeat indicator

The *HLC repeat indicator* information element is included if and only if both following conditions hold:

- The *BC repeat indicator* IE is contained in the message.
- The *high layer compatibility i* IE is contained in the message.

If included, the *HLC repeat indicator* shall specify the same repeat indication as the *BC repeat indicator* IE.

9.3.23.2.7 High layer compatibility i

The information element is included when the calling MS wants to pass high layer compatibility information to the called user.

9.3.23.2.8 High layer compatibility ii

Included if and only if the *HLC repeat indicator* information element is contained in the message.

9.3.23.2.9 User-user

The information element is included in the calling mobile station to network direction when the calling mobile station wants to pass user information to the called remote user.

9.3.23.2.10 SS version

This information element shall not be included if the *facility* information element is not present in this message.

This information element shall be included or excluded as defined in TS 24.010. This information element should not be transmitted unless explicitly required by TS 24.010.

9.3.23.2.11 CLIR suppression

The information element may be included by the MS (see TS 24.081). If this information element is included the *CLIR invocation* IE shall not be included.

9.3.23.2.12 CLIR invocation

The information element may be included by the MS (see TS 24.081). If this information element is included the *CLIR suppression* IE shall not be included.

9.3.23.2.13 CC Capabilities

This information element may be included by the mobile station to indicate its call control capabilities.

9.3.23.2.14 Stream Identifier

This information element shall be included by the mobile station supporting multiple traffic channels. The purpose of the information element is to indicate whether a new traffic channel shall be assigned to the call.

****** Next Modified Section ******

10 General message format and information elements coding

10.5 Other information elements

10.5.4 Call control information elements.

10.5.4.XX Stream Identifier

The purpose of the stream identifier (SI) information element is to associate a particular call with a Radio Access Bearer (RAB), and to identify whether a new traffic channel shall be assigned within the interface controlled by these signalling procedures. The SI value indicated in the CC protocol shall be sent in the RAB setup message. And mobile station is informed the relationship between the call and the RAB.

The Stream identifier information element is coded as shown in figure 10.5.XX/TS 24.008 and table 10.5.XX/TS 24.008.

The Stream Identifier is a type 4 information element with 3 octets length.

8	7	6	5	4	3	2	1	
Stream Identifier IEI								octet 1
Length of Stream Identifier contents								octet 2
Stream Identifier Value								octet 3

Figure 10.5.XX/TS 24.008: Stream Identifier information element

Table 10.5.XX/TS 24.008: Stream Identifier information element

Stream Identifier value(octet 3)	
Bit	
87654321	
00000000	No bearer
00000001	1
:	
:	
11111111	255

3GPP/SMG Meeting #10
Abiko, Japan, 11 – 14 January, 2000

***Document* N1-000211**

CHANGE REQUEST

Please see embedded help file at the bottom of this page for instructions on how to fill in this form correctly.

24.008	**CR**	**124r2**	Current Version:	3.2.1

GSM (AA.BB) or 3G (AA.BBB) specification number ↑ *↑ CR number as allocated by MCC support team*

For submission to:	CN#7	for approval	**X**	strategic		*(for SMG use only)*
list expected approval meeting # here ↑		for information		non-strategic		

Form: CR cover sheet, version 2 for 3GPP and SMG The latest version of this form is available from: ftp://ftp.3gpp.org/Information/CR-Form-v2.doc

Proposed change affects: *(at least one should be marked with an X)*	(U)SIM		ME	**X**	UTRAN / Radio		Core Network	**X**

Source:	Fujitsu	**Date:**	7 January, 2000
Subject:	Coding of NAS Binding Info field		
Work item:	GSM/UMTS Interworking		

Category:	F	Correction		**Release:**	Phase 2	
	A	Corresponds to a correction in an earlier release			Release 96	
(only one category	B	Addition of feature	**X**		Release 97	
shall be marked	C	Functional modification of feature			Release 98	
with an X)	D	Editorial modification			Release 99	**X**
					Release 00	

Reason for change:

To correlate NAS service (i.e., call, PDP context …) and Radio Access Bearer in Access Stratum, the NAS entity shall set binding information to NAS Binding Info IE, which is carried by RANAP and RRC messages from CN node to UE.

Currently, either of SI (for CC) or NSAPI (for SM) is set in the IE. This CR proposes coding of content of NAS Binding Info IE.

Clauses affected: 10.5.8

Other specs affected:	Other 3G core specifications		→ List of CRs:	
	Other GSM core specifications		→ List of CRs:	
	MS test specifications		→ List of CRs:	
	BSS test specifications		→ List of CRs:	
	O&M specifications		→ List of CRs:	

Other comments:

help.doc

<--------- double-click here for help and instructions on how to create a CR.

10.5.8 Other information elements.

10.5.8.1 NAS Binding Information (UMTS only)

The purpose of the *NAS Binding Information* information element is to bind data stream from the Non-Access Stratum point of view (e.g. bearer of call or PDP context) and radio access bearer in Access Stratum. The content of this information element is transparently transferred from CN node (i.e., MSC or SGSN) to MS by RANAP (see TS 25.413) and RRC (see TS 25.331) messages. Only the coding of the content is in the scope of this specification.

Either of Stream Identifier (refer to section 10.5.4.xx) for CC protocol or NSAPI (refer to section 10.5.6.2) for SM protocol is included in the information element.

The content of *NAS Binding Information* information element is coded as shown in figure 10.5.x/TS 24.008 and table 10.5.x/TS 24.008.

8	7	6	5	4	3	2	1	
Spare 0	0	0	0	0	0	0	0	octet 1
NAS Binding Information value								octet 2

Figure 10.5.x/TS 24.008 *NAS Binding Information* Information element

Table 10.5.x/TS 24.008: *NAS Binding Information* Information element

NAS Binding Information value(octet 2) The 8th bit of octet 2 is the MSB and the 1st bit of octet 2 is the LSB. This field contains binary representation of Stream Identifier value(refer to section 10.5.4.xx) or NSAPI value(refer to section 10.5.6.2). All bits of octet 1 are spare and shall be coded all zeros.

3GPP TSG-CN-WG1, Meeting #10
11 – 14 January 2000
Abiko, Japan

Tdoc N1-000136

Source: TSG-CN WG1

To: 3GPP TSG SA WG1

Cc: 3GPP TSG SA WG2, 3GPP TSG CN WG3

Subject: Response to Liaison statement concerning HSCSD specifications

Contact Person: Janne Muhonen
E-mail: janne.m.muhonen@nokia.com
Tel: +358 40 555 9627

TSG CN WG1 thank TSG SA WG1 for their LS concerning HSCSD specifications.

TSG CN WG1 do not have any objection to TSG SA WG1's analysis that GBS concept is fully applicable to 3GPP systems, and that multislot is only relevant for GERAN.

Furthermore TSG CN WG1 would like to inform TSG SA WG1 that TSG CN WG1 have not yet analysed in detail what changes needs to be done to HSCSD stage 2 (3G TS 23.034) due to the changes in HSCSD stage 1 (3G TS 23.034).

3GPP TSG CN WG1 Meeting
Abiko, Japan
11-14 January 2000

N1-000138

From:	**TSG CN WG1**
To:	**TSG SA WG2, SMG2**
CC:	**TSG CN WG2, TSG RAN WG 3**
Subject:	**Response Liaison Statement on Usage of RANAP over MAP/E i/f for UMTS to UMTS Inter-MSC SRNS Relocation**
Contact Person:	Rouzbeh Farhoumand
	E-mail: rouzbeh.farhoumand@ericsson.com
	Tel: +1-972-583-8061

At TSG CN WG1#10 we reviewed the Liaison Statement and the attached contribution (Tdoc S2-99F02) listing advantages of using RANAP, sent by TSG SA WG2 on usage of RANAP instead of BSSAP over the E interface at UMTS to UMTS inter MSC SRNS relocation. For some counter arguments given during an offline discussion see attached Tdoc N1-000199, and the original document Tdoc S2-99F02. The outcome of the detailed discussions is as follows:

1. For UMTS – UMTS inter MSC SRNS relocation when ATM (supporting OoBTC) is the underlying transport media, RANAP shall be used.

2. For UMTS – UMTS inter MSC SRNS relocation when STM is the underlying transport media, the two following scenarios were recognized:

    - RANAP may be used here as well, and the work to accomplish it in R99 will be carried out in relevant groups in 3GPP.
    - SMG2 while they are enhancing the BSSMAP extensions for UMTS – GSM, also enhance these extensions to apply for UMTS – UMTS inter MSC SRNS relocation.

      The reasons why BSSMAP may be used are:

      - generally the parameters which are needed for the UMTS->UMTS handover (e.g. RAB Id, NAS Binding Information) have to be included anyway in BSSMAP for the case of GSM->UMTS handover.
      - the use of BSSMAP seems to be more appropriate in situations where the transcoder is located in the target MSC

      For the STM case, if both works are done by the finalization of R99 by TSG S#7, then one protocol shall be mandated as the default protocol over the MAP E interface by TSG S#7.

      Obviously, if the work of one the two proposed approaches for any reason is not finalized by TSG S#7, then the other protocol shall be made mandatory, or valid reasons for requesting an extension to finalize the work have to presented to TSG S#7.

TSG CN WG1 proposes that the working assumption stated in 23.121 version 3.1.0 to be changed to read:

For UMTS to UMTS Inter-MSC Handover the following messages shall be used embedded in MAP at the GSM E i/f:

i) RANAP, or BSSAP (i.e. BSSMAP and DTAP) messages with necessary modifications for GSM to UMTS Handover, if STM is used on the link between the anchor and the target MSC.

ii) RANAP messages, if ATM is used on the link between the anchor and the target MSC.

3GPP TSG-N WG1 **Tdoc 3GPP N1-0000111**
Chiba, JAPAN
10 Jan – 14 Jan 2000

Source: **Ericsson L. M.**

Agenda Item: **(Formerly) Out-of-Band Transcoder Control**
Title: **Transport of Codec Information**

Purpose: **For Information**

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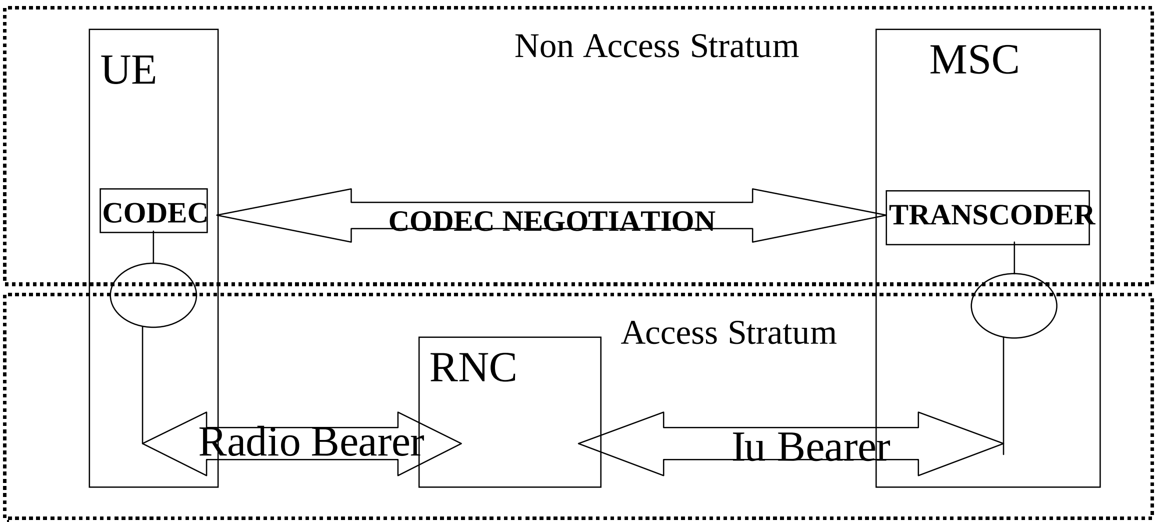
1. Introduction

Information about codecs needs to be exchanged at the access side between UE and MSC. In GSM the Bearer Capability Information Element is used to indicate supported speech codecs from the MS to the MSC. From MSC to BSC to MS the Channel Mode IE in Assignment Command is used to indicate the chosen speech version.

There are three~~two~~ problems discussed in this paper, firstly the issue of distinguishing between UMTS speech versions and GSM speech versions and secondly the issue of informing the UE of the chosen speech version. The latter discussion is a continuation of the discussion introduced to N1 in N1-99632. Thirdly, the handling of speech coding negotiation and control during and after Inter-MSC hnadover is discussed.

2. Principles

It is defined in the 3G TS 23.110 UMTS Access Stratum; Services & Functions, that speech coding is a Non Access Stratum function. This means that negotiation of speech codecs must be performed by Non Access Stratum. Non Access Stratum messages are carried in RANAP in a container – Direct Transfer Message.



~~2.~~3. Current Situation

In the UMTS TS 24.008 the speech versions supported do not differentiate between coding schemes that are to used in UMTS and those that are only to be used in GSM. In UMTS the default codec is AMR (as defined in 26.090). The coding algorithms for the UMTS AMR rates are bit exact to those in the GSM AMR (GSM 06.71) however there are some differences in the behaviour of these codecs (e.g. Rate Control frequency and the DTX framing).

If a UMTS MSC receives a SETUP message containing a list of supported speech versions as it is today in the BC IE then it must make some assumptions on what can be supported over an Iu interface and what can be supported over an A-interface. An obvious statement would be that if Speech Version 3 was included then this would be valid for both GSM (assuming Classmark indicates support of GSM) aswell as UMTS. Then all other speech versions are for GSM only - needed for intersystem handover. This assumption means that all dual system handsets must support AMR for GSM system also and that no handset can introduce GSM speech version encoding for UMTS.

The codec selection should not be based on any assumptions.

Further, in AMR there is the Supported Codec Set which contains the AMR modes which can be handled by a specific Coder/Decoder equipment and the Active Codec Set which contains the negotiated AMR rates that can be used between codecs to perform rate adjustment. In GSM the negotiation of the Active Set is performed between the BSC and the MS, and between TFO codecs using the TFO protocol. In GSM the Active Codec Set is conveyed to the MS via RR message.

For UMTS if a TRAU is required it is located in the CN and so controlled by the MSC and if OoBTC is supported to achieve the goal of ~~to achieve~~ transcoder free operation then OoB Supported Codec Set negotiation is also required. Both of these reasons mean that we cannot rely on control from the radio access or from the TFO protocol (no TFO if no Transcoders) to negotiate these sets. The use of OutOfBand signalling should be deployed as described in N1-99720.

As described in N1-99632 there is no signalling to the UE to indicate the chosen codec. There is no speech version included in the RAB assignment from MSC to RNC and so no speech version can be included in any RRC message from RNC to UE as compared with the handling in GSM.

The MSC requests a Radio Access Bearer providing the required SDU frame sizes for the chosen speech coding rates. The RNC establishes a RAB for these frames and informs the UE of the specific framing required (i.e. the SDU sizes for each Active Codec rate) in the Radio Bearer Setup but not the coding scheme itself, nor the DTX, nor the Rate Adaptation rules (frequency, stepping order). Again in the current situation the assumption that UMTS AMR is the speech version must be made and that all modes are rate adaptable according to UMTS AMR Rate Control Procedures. This prevents the call scenario of UMTS AMR to GSM AMR with TFO being possible for example.

At Inter-System Handover the Anchor MSC sends the preferred speech versions in BSSMAP message carried in MAP Prepare Handover to the Target MSC, which passes this to the BSC where the Transcoder will be located after Handover. This mechanism cannot be used for UMTS Handover as the transcoder will be located in the Target MSC (STM between Target and Anchor). The Target does not have any Call Control entity and so the codec negotiation must be performed between the Anchor MSC and the UE.

~~3.~~4. Proposed Changes

~~3.1~~4.1 UE to MSC Speech version handling

~~3.1.1~~4.1.1 Alternative 1

In UMTS TS 24.008 a new Information Element is introduced to indicate the supported speech versions for UMTS, sent from the UE to the MSC in the SETUP and CALL CONFIRMED messages. The BC IE will be used as for GSM, to indicate the supported speech versions for the GSM system i.e. in UMTS the MSC will use this list for intersystem handover.

It is further proposed that the format of the data in the new IE is taken from the UMTS TS 26.103 currently being proposed for inclusion in the ITU Q.BICC standard for Codec Negotiation. The use of Q.BICC in UMTS has been recommended to N1 by N2 in N1-99720. The introduction of this "list" in the 24.008 would further enhance the handling of the Out-Of-Band Transcoder negotiation, avoiding the need for the MSC to perform any mapping of speech versions and associated parameters from one "list" to another.

This also enables the codec type to be independent of the 24.008 protocol. This is very advantageous as it should be possible to introduce new coding schemes to the UE and CN independently of the protocol that carries the information.

~~3.1.2~~4.1.2 Alternative 2

The Current speech version codepoints are extended to indicate UMTS speech versions and are included in Octet(s) 3a of the Bearer Capability IE. These codepoints would simply indicate the codec type. Then further parameters required to fully describe the codec (Active Codec Set, Supported Codec Set) would be included in a new optional IE sent in SETUP or CALL CONFIRMED.

The coding of the new IE indicates the ACS, MACS, SCS as in 26.103 (but only this data, i.e. 3 Octets per codec type).

The advantage of this alternative is less octets are passed and the second octet is optional. However this means that if the new IE is not included then the MSC is back to the current situation of making assumptions. Secondly if more than one type of codec in the BC for UMTS needs additional parameters then the new IE must indicate the codec type that they are aplicable to - this is then duplicated information. Note this then results in 5 octets per codec type. Further, the list in the BC does not allow for GSM coding schemes to be indicated for UMTS (without creating yet more codepoints e.g. GSM FR used in UMTS). Therefore this solution is really only practical when there is only one UMTS codec, which really defeats the purpose of this WI.

~~3.1.3~~4.1.3 Recommendation

Alternative 1 is recommended because it is future proof and flexible. It is also seen as a clean solution without impacting the Bearer Capability IE. Its disadvantage is the number of octets added to the SETUP or CALL CONFIRMED messages (6 octets per codec type).

Alternative 2 is only an advantage with 1 UMTS coding scheme. One of the motivations of this WI should be to prepare the CC messages to support OoBTC with the goal of achieving optimised transcoding/transcoderless connections. This can only be achieved in the majority of connections if the terminals support a variety of coding schemes (to ensure a compatible match) thus we need a solution that can allow coding schemes to be introduced without impact to the protocol.

~~3.2~~4.2 *MSC to UE Speech Version Handling*

A new CC message should be introduced to indicate the chosen speech version from the MSC to the UE. In a previous paper (N1-99632) the NOTIFY message was proposed, following that it was proposed that this was unsuitable as it should not be sent during call establishment and that PROGRESS massage was more suitable. It is recommended by this paper to use a new message as any existing handling of PROGRESS message could be affected by its use for conveying Speech Codec information and more specifically instructing the UE to select a Codec Type. The sending of this new message could be inhibited for GSM systems by checking the Classmark information although the sending of a new message to a GSM only MS would simply be discarded in unrecognised and so not cause any protocol errors.

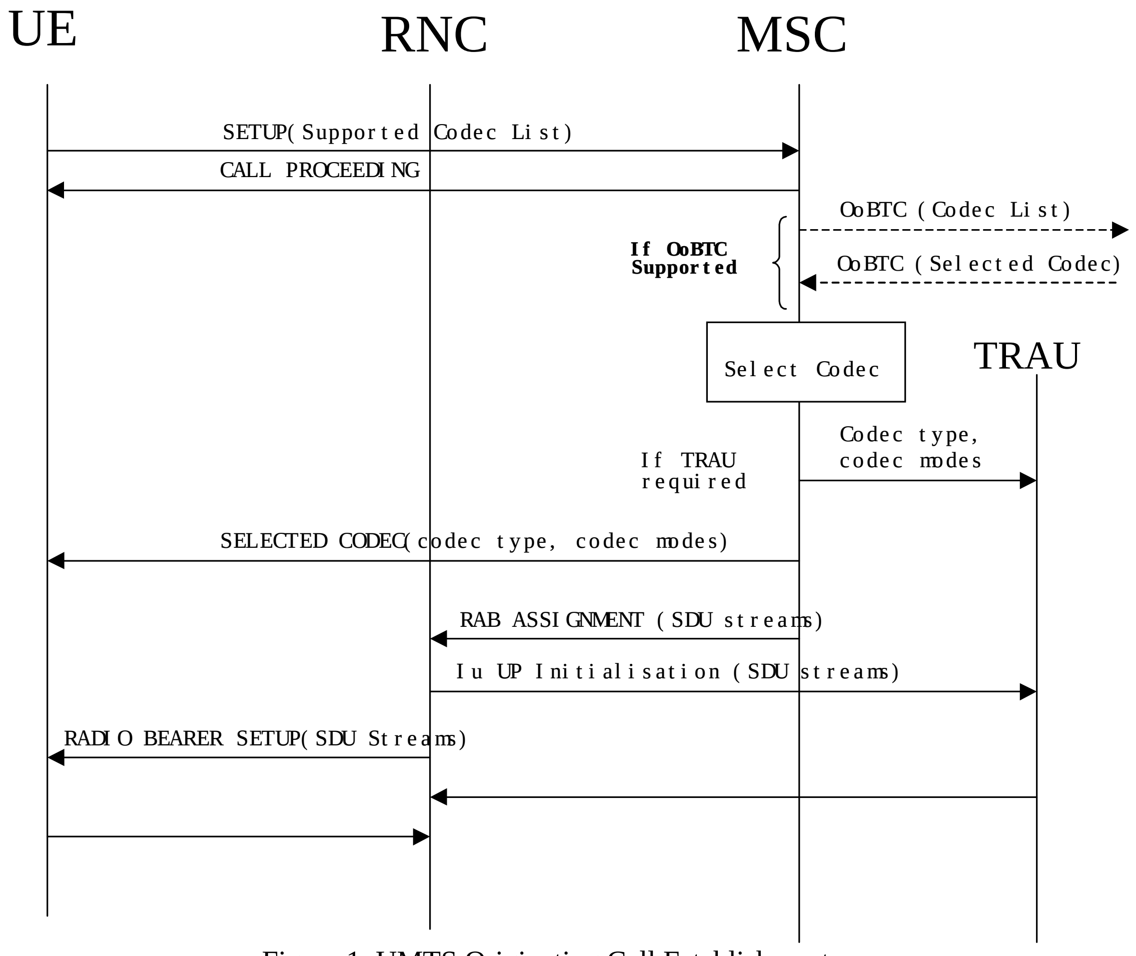


Figure 1. UMTS Originating Call Establishment

Note: It is proposed that this information is added to UMTS TS 23.108

The CR to UMTS TS 24.008 (contained in a companion contribution to this one) provides the specific changes proposed (Alternative 1).

4.3 Handover & Subsequent Supplementary Service Invocation

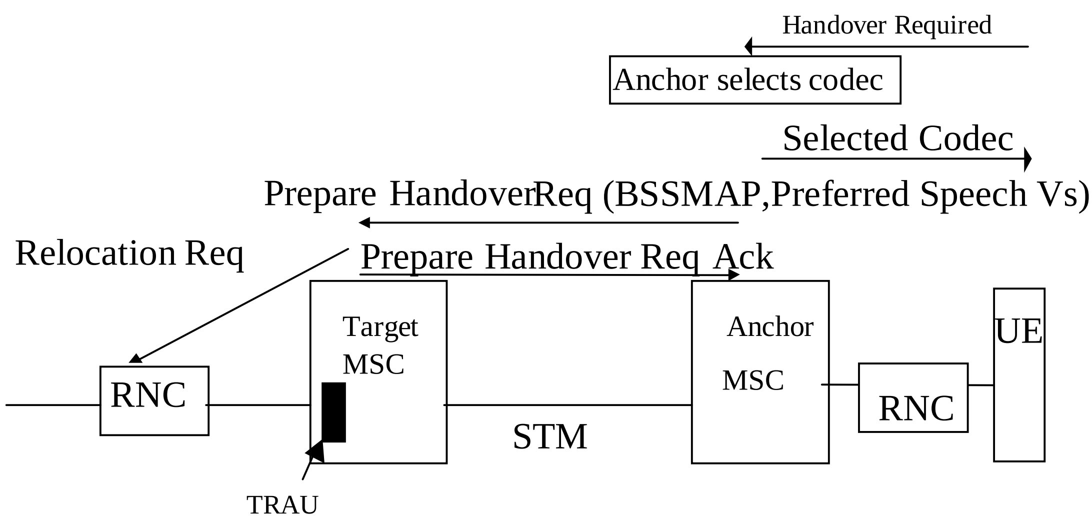
The principle with the handover is that the Anchor MSC is in control of the call. As speech codec handling is defined as part of call control then the Anchor MSC must be in control of the speech codecs. In handover situations where the bearer between Anchor and Target MSC is STM then the Anchor must have knowledge of the supported codecs for the target MSC.

Where ATM exists between MSC's then no transcoding in the target is required. Then the Anchor is in control of the codecs in the UE directly, and the transcoders in the Core Network, either directly (in its own node) or via BICC (in an edge/gateway node).

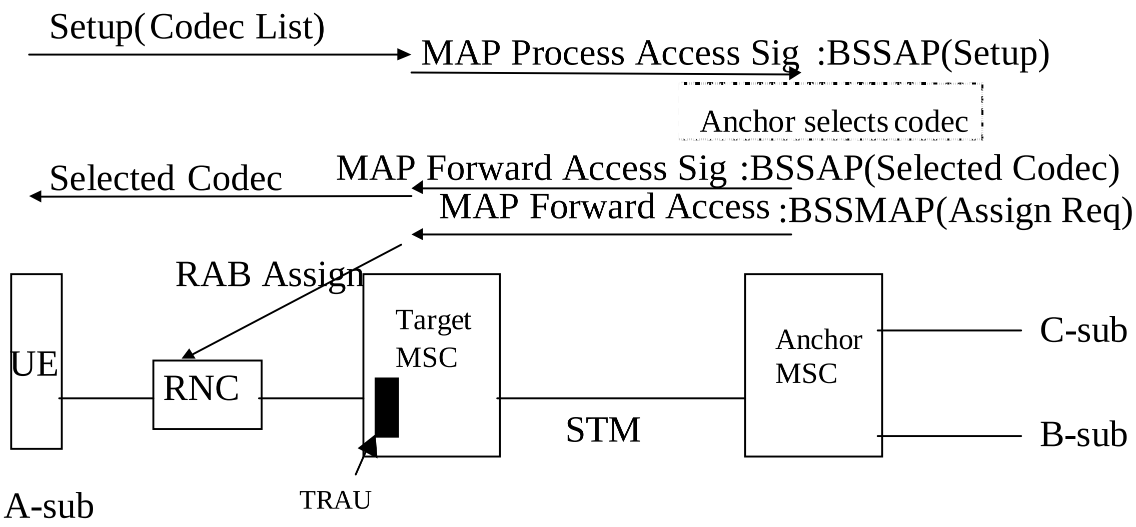
Note the handover from UMTS to GSM does not cause a problem as existing handling must be performed as the BSC is controlling the transcoder after handover.

4.3.1 Proposal 1

The first proposal is that the Anchor has pre-provisioned transcoder information for adjacent MSC's when connected with STM. The Anchor then indicates the selected codec in the Prepare Handover Request to the target MSC. It also sends the Selected Codec CC message to the UE indicating that this is the codec to be used for handover.

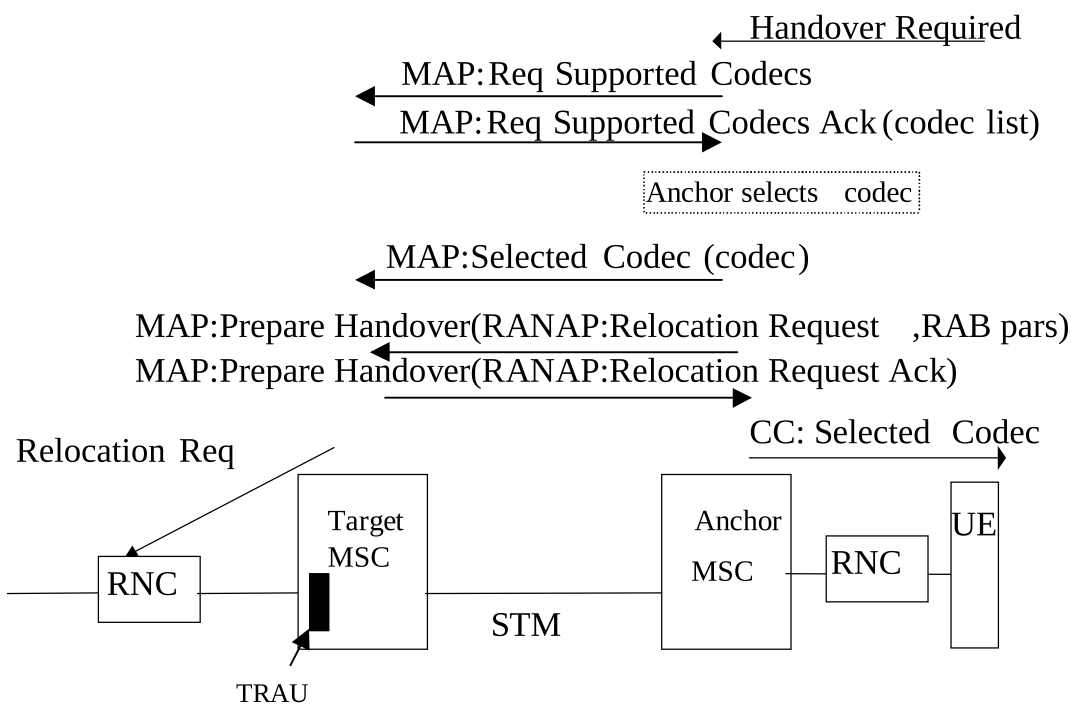


Subsequent supplementary services will be coordinated by the Anchor, if a new Setup message is received due to enquiry call, call waiting etc then the Anchor will select the codec and indicate this back to the UE in the Selected Codec message. This will also be indicated in the BSSMAP assignment to the target MSC so that the target MSC can perform the RAB assignment and also assign the Transcoder in its MSC node.

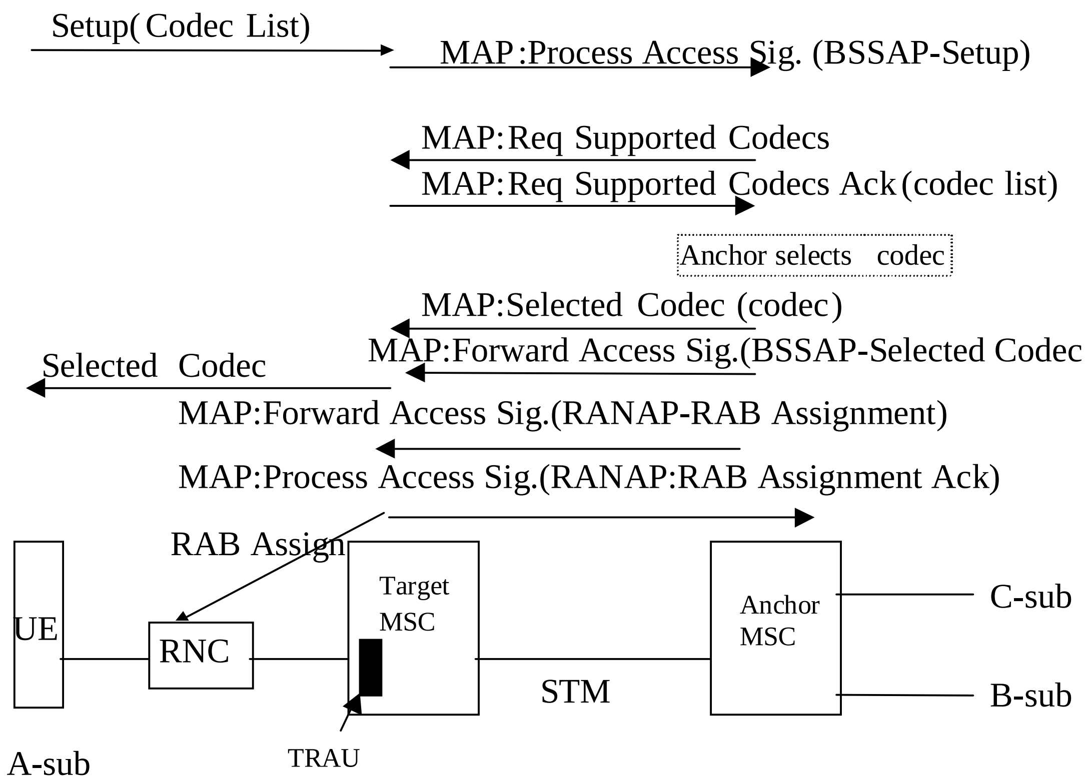


4.3.2 Proposal 2

The second proposal is that the Anchor requests this information from the target. This is proposed to be performed by a new MAP procedure. The MAP procedure will be terminated in the Target MSC and thus can be used to both request the Target MSC's codec capabilities but also to indicate the selected codec to the Target in order that the target can seize the corresponding Transcoder.



Subsequent supplementary service invocation that required any codec negotiation would also use the same mechanism, although it may be unnecessary to request the Targets capabilities if they were negotiated during handover, however if this was not performed due to the handover call be data for example then the full negotiation would be required.



4.3.3 Recommendation

Proposal 2 is recommended as it is most flexible, fits with the architectural principles, and allows for future development in the standards. With the introduction of Q.BICC and Out Of Band Transcoder Negotiation, the Target MSC should be able to negotiate codecs at any time during the call, to the UE and to the CN equipment. At handover this could reside in the Target MSC and then a MAP procedure would be the most logical means to implement this negotiation.

The CR to 3G TS 29.002 (contained in a companion contribution to this one) provides the specific changes for Proposal 2.

3GPP TSG_S2#10 **_Tdoc S2-99F02_**
Abiko, Japan, Nov 29-Dec 3 1999

Source: Ericsson LM

Title: UMTS inter-MSC SRNS Relocation

Agenda item: **23.121**

Document for: Decision

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1 Introduction

23.121 version 3.1.0 states "*For UMTS to UMTS Inter-MSC Handover the GSM E i/f transporting BSSAP messages with necessary modifications for GSM to UMTS Handover shall be used.*".

The Handover messages are in GSM defined in the BSSMAP protocol and it is here assumed that the above statement in 23.121 should be interpreted in the way that some extensions may be needed to BSSMAP to include some UMTS specific information

Rapid deployment of UMTS systems imply decreasing intersystem HO scenarios and at the same time imply increasing 3G HO scenarios. Having this in mind the selection of BSSAP for UMTS to UMTS handover is not the best solution. Instead, the transparent transfer of RANAP information between 3G MSCs has clear advantages that are listed below:

- No redundant Information Elements
- No protocol conversion
- No mapping problems
- No dependency between RANAP and BSSMAP
- Multicall supported
- Future proof

2 Advantages

2.1 No redundant Information Elements

All IEs in RANAP will be relevant.

The mandatory IEs in BSSMAP must always be sent but are not used in pure 3G relocation and are therefore a not wanted overhead. If used they will instead cause mapping problems.

It can be argued that it is not much overhead but on the other hand there is a limited amount of data that can be transferred transparently in the MAP/E messages. Only 200 octets are available and if this limit is exceeded the MAP messages must be segmented and that is not good for a time critical procedure as the handover procedure.

Using BSSMAP instead of RANAP introduces problems how to fill in the mandatory IEs. There is e.g. no obvious mapping between the RANAP QoS parameters and the fields in the BSSMAP Channel Type IE. Using GSM information from MS Classmark and the BC in the Setup message can solve this for UMTS to GSM handover and for dual-mode phones. But this information is not included in pure 3G mode. How should the mandatory Layer 3 Information IE in BSSMAP Handover Request Acknowledge be coded (this can be copied from the transparent container in Relocation Request Acknowledge but it is only an optional element that is not always present)?

2.2 No protocol conversion

A pure 3G_MSC does not need protocol conversion for inter-3G_MSC SRNS relocation if RANAP is used.

Double conversions are needed in the 3G_MSCs for relocation if BSSMAP is used. The first conversion in the anchor 3G_MSC is from RANAP to BSSMAP and in the non-anchor 3G_MSC from BSSMAP to RANAP. These conversion mechanisms must of course be defined anyway to be used for GSM<->UMTS handover but each conversion takes time and the handover procedure is very time critical and therefore conversions should be avoided.

2.3 No mapping problems

No mapping is needed for inter-3G_MSC SRNS relocation if RANAP is used.

Mapping is needed between RANAP and BSSMAP in case BSSMAP is used and any existing BSSMAP IE should be re-used. There is no straightforward solution defined how to convert RANAP QoS to BSSMAP and back again. This may cause information loss at the mapping procedures. One example is that an operator may want to have different transparent data bearers. One for data that is less delay sensitive and one for multimedia that is delay sensitive. At relocation the QoS parameters must in case of BSSMAP be converted to a Channel Type. There is then no way to differ between the different transparent bearers delay parameter in the current BSSMAP version.

2.4 No dependency between RANAP and BSSMAP

To use BSSMAP to transfer RANAP may introduce a dependency between RANAP and BSSMAP, which means that BSSMAP in the worst case must be updated each time RANAP is updated. Currently there are two different forums responsible for these specifications. ETSI (SMG2) is responsible for BSSMAP and 3GPP (R3) for RANAP. This means that changes in BSSMAP due to RANAP changes will take some time. A possible solution to this is to introduce a transparent RANAP container in BSSMAP but this adds extra overhead and possible conflicts between the BSSMAP information and RANAP information in the transparent container.

2.5 Multicall

RANAP supports several independent UE associated calls.

A UE supporting multicall has several associated calls that need to be relocated. There is no support today in BSSMAP to handle multicall. Support for multicall must be added to BSSMAP in case BSSMAP is used for relocation. This should also apply to partial relocation.

2.6 Future proof

New functions may be added in future releases. These functions are simple to transfer via the MAP E interface in case RANAP is used.

3 Disadvantages

One drawback by using RANAP is that MAP version 3 is required for the handover procedure to transfer a transparent RANAP container. However, many other procedures like the new security mechanisms will anyway require that the MSCs are updated with MAP version 3. The necessary change is to add a new protocol id (e.g. RANAP) in the bssAPDU element of some MAP Handover messages.

```
bss-APDU SEQUENCE {
        protocolId ENUMERATED {
          gsm-0408  (1 ),
          gsm-0806  (2 ),
          gsm-BSSMAP  (3 ),
          ets-300102-1  (4 )
          3gpp-RANAP  (5)}
```

4 Conclusion

It is proposed that the decision to use BSSAP for inter-3G_MSC SRNS Relocation should be revised.

3GPP TSG CN WG1 Meeting
Abiko, Japan
11-14 January 2000

N1-000199

Source: **Siemens**

Subject: **Comments on usage of RANAP over MAP/E i/f for UMTS to UMTS inter-MSC SRNS relocation**

TSG CN WG1 has reviewed the Liaison Statement and the attached contribution (Tdoc S2-99F02) sent by TSG SA WG2 on usage of RANAP instead of BSSAP over the E interface at UMTS to UMTS inter-MSC SRNS relocation.

After a detailed discussion of the arguments given in Tdoc S2-99F02 for the usage of RANAP, TSG CN WG1 proposes that the working assumption stated in 23.121 version 3.1.0 to be changed to read:

For UMTS to UMTS Inter-MSC Handover the following messages shall be used embedded in MAP at the GSM E i/f:

i) BSSAP (i.e. BSSMAP and DTAP) messages with necessary modifications for GSM to UMTS Handover, if STM is used on the link between the anchor and the target MSC

ii) RANAP messages, if ATM is used on the link link between the anchor and the target MSC.

With regard to the various issues mentioned in Tdoc S2-99F02, the following arguments were given in CN1:

- *"No redundant Information Elements"*

Note that with regard to the location of the transcoder after MSC-MSC handover the behaviour of ATM and STM links is different. As a consequence, the argument that 'all IEs in RANAP will be relevant' does not apply in the case where after the handover the transcoder is located in the target MSC, i.e. if STM is used on the link between anchor and target MSC. In that case the QoS profile for a radio access bearer at the Iu interface has to be generated by the target MSC, because it is the target MSC which selects the transcoder (and the codec modes), and therefore the necessary information (e.g. concerning the SDU streams/RAB subflow combinations) is available only in the target MSC. If the anchor MSC includes the mandatory information element QoS information in the RANAP Relocation Request, this information will be of no use in the target MSC.

(Note: There was a proposal presented at the CN1 meeting to give the anchor MSC 'direct' control over the transcoders in the target MSC (see Tdoc N1-000111, section 4.3.2). However, this proposal does not work because when the anchor MSC selects the codec, it needs the information which transcoders will be available in the target MSC at the very moment when the command to allocate the codec is received by the target MSC; but this information cannot available in the anchor MSC. Furthermore, by this proposal the time critical handover procedure would be enhanced by three MAP dialogue steps which is in clear contradiction to the argumentation given in Tdoc S2-99F02 itself.)

With regard to the message length, it can be expected that because of the amount of information tranported in the QoS profile, the RANAP messages will be considerably longer than the respective

BSSMAP messages.

- *“No protocol conversion“*

With regard to the protocol conversion in the anchor MSC, the main issue will be the Source RNC to Target RNC transparent container which can be added to the BSSMAP Handover Request message. This has to be done anyway for the GSM to UMTS handover case. Copying this container from Relocation Required to Handover Request does not require a time consuming mapping.

There might be some mapping required from BSSMAP to RANAP in the target MSC, however if we compare the time consumed by this with the time which is required by the additional ASN.1 encoding of a RANAP message in the anchor MSC and the subsequent decoding of the same message in the target MSC, it is questionable whether there is really any advantage for the use of RANAP.

(As mentioned above, in the STM case, if the transcoder is located in the target MSC, the QoS cannot be taken from a Relocation Command received via the E-interface, but has to be generated in the target MSC.)

- *“No mapping problems“*

As N3 stated recently in their LS to S2 on ‘QoS mapping in case of HO from 3G to 2G system’, there is no need to define a mapping from RANAP QoS parameters to BSSMAP Channel Type, as the BSSMAP Channel Type can be derived in the anchor MSC from the GSM Bearer Capability. The mapping from BSSMAP Channel Type to RANAP QoS profile which is needed in the target MSC will have to be defined anyway for the case of UMTS -> GSM handover. (The parameters contained in the RANAP QoS profile allow a much more detailed description of the bearer compared to the BSSMAP Channel Type; therefore it should not be a big problem to define such a mapping.)

- *“No dependency between RANAP and BSSMAP“*

Generally the parameters which are needed for the UMTS->UMTS handover (e.g. RAB Id, NAS Binding Information) have to be included anyway in BSSMAP for the case of GSM->UMTS handover.

SMG2 is planned to be transferred to 3GPP as a new TSG GERAN during this year. This will narrow the institutional gap between BSSMAP and RANAP.

- *“Multicall supported“*

Indeed, for the handover of a multicall, additions to the specifications will be needed (mainly to TS 23.009). The basic idea which can be followed is to include one Handover Request message for each bearer to be handovered. (Inclusion of more than one BSSMAP messag in the MAP Prepare Handover operation is possible already today.)

- *“Future proof“*

The new working assumption takes care of the concerns expressed in Tdoc S2-99F02 by specifying the use of RANAP via ATM links, as ATM is expected to replace STM by and by as transport technology in future networks.

3GPP TSG-CN-WG1, Meeting #10
11 - 14.January. 2000
Abiko, Japan

Tdoc N1-00 0149

To: TSG-S2

cc: TSG S4

Source: TSG N1

Title: LS on questions on the CR 10r1 to TS 23.107

Contact: Roland Gruber, Siemens AG
E-mail: roland.gruber@mch.siemens.de
phone: +49 89 722 46392

N1 has reviewed the CR 10r1 to TS 23.107 in Tdoc S2-99F37 and likes to raise the following question:

Why is it necessary to tear down all active PDP contexts except one in the case of a handover from a Release 99 to a Release 97/98 GPRS network. According to our understanding, parallel active PDP contexts are also supported by Release 97/98 GPRS networks.

Or was the intention of this CR to deal with the situation where several PDP context sharing the same PDP address are active and only one of these could be maintained when changing to a R97/98 SGSN, as the concept of PDP contexts sharing the same address is only available in R99? But if this was the background, there seem to be no need to tear down all PDP context except one, but only all PDP context except one sharing the same address.

3GPP TSG-CN-WG1, Meeting #10
11 - 14.January. 2000
Abiko, Japan

Tdoc N1-000152

Source: TSG-CN WG1

To: TSG-SA WG1, TSG-RAN WG3

Cc: TSG-SA WG2

Contact Person: Richard Brook.
E-mail: rb39@lucent.com
Tel: +44 1793 736185

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Subject: Response to LS on RAB linking

TSG-CN WG1 thank RAN 3 for their LS on “RAB linking” and would like to confirm that N1 do not have any objection to this feature as it could possibly be used to set up a improve data throughput by using Multicall.

CN1 would like to inform RAN 3 that they can therefore proceed with their work on defining the impacted RANAP procedure provided there is no objection from SA1.

Could RAN3 please inform CN1 of any impact on specifications under their control that is found during this work so that the appropriate action can be taken.

Subject: Response to LS on RAB pre-emption.

TSG CN WG1 cannot see any technical reason against this proposal, but would like to ask TSG SA WG1 under what scenarios this would be applied?

One possible scenario put forward during CN1’s meeting was that of radio channel congestion.

As TSG CN WG1 does not see any CN protocol related issues with this question, we would like to leave it up to TSG SA WG1 to decide on the necessity of the service.

3GPP TSG-CN-WG1, Meeting #10
11 - 14.January. 2000
Abiko, Japan

Tdoc N1-000153

Source: 3GPP TSG-CN WG1

To: 3GPP TSG-T WG3, ETSI SMG9

Cc:

Contact Person: Mark Fenton.
E-mail: mailto: mark.fenton@eml.ericsson.se
Tel: +44 1256 864376

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Subject: Response to LS on Capability configuration parameters

TSG-CN WG1 thank T3 for their LS on "Capability configuration parameters" (N1-000117, T3-99420) asking if the configuration and capability parameters to be stored into USIM under EF_{CCP} needs to be extended considering new bearer capabilities have been (or will be) introduced in 3G.

In GSM Phase 2 the bearer capability has a maximum length of 10 octets.
In Phase 2+ Release 95 the maximum length has been increased to 15 octets by the addition of HSCSD.
In Phase 2+ Release 96 the maximum length remains at 15 octets.
In Phase 2+ Release 97 the maximum length remains at 15 octets
In Phase 2+ Release 98 (04.08 version 7.4.0) the maximum length remains at 15 octets.
In Phase 2+ Release 99 (24.008 version 3.2.1) the maximum length has been increased to 16 octets by the addition of EDGE.

Note that the 1st octet represents the IEI, which doesn't need to be stored on the SIM.

For R99 UMTS, some new codepoints have been added for UMTS bearers, but this has not increased the maximum length. It should be noted that a number of R99 work items are still incomplete but it is not expected that the bearer capability maximum length will be increased.

N1 also notes that the useable part of the existing Capability configuration parameters is currently 10 octets. This is not large enough to contain a maximum length bearer capability from R95 or later.

N1 asks SMG9 to consider updating 11.11 for R95 and later releases to correct this error.

It should be noted that the additional octets (above 10) are only used when storing the Capability Configuration parameters relating to HSCSD call (or an EDGE call). So they would only be used by mobiles which support HSCSD (or EDGE). This may explain why this error has not been detected until now. It would hence be sensible to correct this error in as early a release as is possible before HSCSD mobiles appear on the market.

N1 cannot explain how this inconsistency occurred. In future N1 will endeavour to send a LS to SMG9 and T3 whenever the bearer capability maximum length is changed.

3GPP TSG-CN-WG1, Meeting #10
11 - 14.January. 2000
ABIKO, JAPAN

Tdoc N1-000164

Source: **CN WG 1**

Contact Person: Robert Zaus
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To: **RAN WG2, RAN WG3, CN WG2**

Title: **LS on the Transport of Codec Information during the Codec Negotiation between MS and MSC**

During the CN1#10 meeting in Abiko, Japan, CN WG1 discussed different alternatives for transport of codec information during codec negotiation between MS and MSC. For this purpose, CN WG1 considered procedures involving only CC signalling as well as other mechanisms using combination of CC and RR messages. The final agreed working assumption is attached in Tdoc N1-000163.

CN WG1 kindly asks RAN WG2 and WG3 to consider this working assumption and take further actions to implement it in the respective specifications under their responsibility.

We attached the most important input documents (Tdocs N1-000033, N1-000111, N1-000140, N1-000141) as background information.

Codec negotiation between MS and MSC is an essential prerequisite for the finalization of UMTS R99, and therefore CN WG1 kindly asks RAN WG2 and WG3 for response out of their upcoming meetings in January. This will enable CN WG1 to prepare the necessary CRs to its next and last meeting for R99 issues on 28.2.-2.3.2000.

3GPP TSG-N WG1 **Tdoc N1-000033**
Abiko, JAPAN
11 – 14 January 2000

Source: **Siemens AG**

Agenda Item: **Out-of-Band Transcoder Control**
Title: **Comments on Tdoc N1-99F40: Transport of Codec Information**

Purpose: **For Discussion and Decision**

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1. Introduction

During the last CN1 meeting #9, Ericsson and NTT DoCoMo presented a proposal for the transport of codec information between MS and MSC in UMTS (Tdoc N1-99E24, N1-99F40). In contrast to the solution in GSM where the information about the speech codec selected by the network is transported by an RR message, it was suggested to use a new CC message for this purpose. The main reason given for this change was a structural one: as in UMTS the transcoder is controlled by the core network, the codec negotiation is considered as a CC procedure and, therefore, no RR signalling should be involved. Separate signalling would allow an easier introduction of new radio access systems, as the downlink signalling of the selected codec would not have to be included again in every new RR protocol.

There are, however, some functional drawbacks of the proposed solution, which should be studied carefully, before we decide to discard a solution similar to the GSM codec negotiation procedure which is well proven in practise.

Before we start with this study, it is worth while to have a look at the work item description of the work item 'Out-of-Band Transcoder Control'.

2. Requirements for Transcoder Control

The status of the work item description is a bit fuzzy, as its last version (Tdoc NP-99385) which was approved according to the meeting report of TSG CN#5 in Kyongju, Korea, apparently never has been provided by the originators. (It is neither available from the 3GPP server nor from meeting secretary of TSG CN#5.)

The following requirements were therefore taken from the previous version (Tdoc NP-99292). These requirements seem to have been stable now for some time (see also a previous version in Tdoc NP-99275), and no comments concerning these requirements were recorded in the meeting report of TSG CN#5.

x.x.2 Requirements for Transcoder Control

The requirements for Transcoder Control are as follows:

- The negotiation procedure should be applied not only for speech codecs and *multimedia codecs* but also for data communications such as Facsimile, Modem, PPP/PIAFS and so on. The procedure should have flexibility for future enhancements of codec types.

- The negotiation and control procedures for Transcoder Control should be independent of the transcoder location in the network, i.e. Core Network (e.g. MSC) or Radio Access Network (e.g. RNC).
- The negotiation and control procedures for Transcoder Control should be independent of the transport layer (e.g. STM or ATM) of both Core Networks and Radio Access Networks.
- The negotiation and control procedures for Transcoder Control should not cause a significant delay in establishing a through connection in mobile-to-mobile calls. Nor should they cause a significant delay when modifying the communication mode between bypass mode and normal mode (e.g. in support of services such as Multiparty Call).
- Transcoder Control communication should be maintained even if the mobile terminal (MT) executes handover.
- Transcoder control communication should be realized in the case of inter-network connections that have different PCM coding standards (i.e. A/μ-law) in the through connection if possible.
- The mobile terminal (MT) may support multiple codec types. Negotiation procedures between the originating MT (or TRAU) and the terminating MT (or TRAU) are required to select a common codec type for Transcoder Control communication in mobile-to-mobile calls.
- The originating MT (or TRAU) may transmit a list of preferred codec types to the terminating MT (or TRAU) during the negotiation process.
- The terminating MT (or TRAU) should select one codec type from this preferred codec list during Transcoder Control negotiation.

Note that, according to the second bullet point, the negotiation procedure should also be supported for a transcoder location in the radio access network(!), i.e. for a configuration as in the GSM system. Clearly this requirement is not fulfilled by the proposed CC procedure, as the CC message Selected Codec is not interpreted by the RAN.

But let us now study the proposal from a technical point of view.

3. Functional Drawbacks of the CC Procedure

3.1 Control of the Transcoder after MSC-MSC Handover

After an MSC-MSC handover, if STM is used on the trunks between the anchor MSC and the target MSC, the transcoder is located in the target MSC. Call control, however, is still located in the anchor MSC. If the transcoder is to be controlled by call control only, there are several scenarios in which a ‘remote control’ of the transcoder via the E-interface is necessary:

- in case of the MSC-MSC handover itself and of a subsequent MSC-MSC handover,
- in case of an in-call modification involving the activation, deactivation or change of the transcoder,
- in case of invocation of a supplementary service like call hold/call waiting or a service like multicall.

The remote transcoder control can be designed in different ways with different degrees of separation between RR and transcode control in the target MSC.

3.1.1 Signalling via RR Messages

In this case the codec information is included in the RR messages (BSSMAP or RANAP) that are sent via the MAP E-interface. As we have in seen at CN1#9, one of the problems with this approach is that in case of re-assignment (e.g. after call wait) the anchor MSC cannot send the Selected Codec message to the MS before it has been informed with the Assignment Complete message which transcoder has been selected by the target MSC. As a consequence, the radio bearer will be assigned and speech frames may already be received by the MS before it knows which codec is to be applied.

To overcome this problem, Ericsson proposed to give the anchor MSC in principle 'direct' control over the transcoders in the target MSC (Tdoc N1-99F40, section 3.3.1 and 3.3.2). The information which transcoders are available in the target MSC should be provided to the anchor MSC either statically, i.e. by administration in the anchor MSC, or dynamically, during an MSC-MSC handover. Both solutions have the drawback that when the anchor MSC starts the re-assignment and selects the codec, it needs the information which transcoders are currently available in the target MSC at that very moment; but this information is not provided by any of the two proposals. Besides that the administrative solution is not practical and would probably not be accepted by operators.

It has also been objected that the transport of the codec information with an RR message violates the basic idea to separate the transcoder control from the RR layer.

3.1.2 Separate Transcoder Control Protocol

If the remote control of the transcoder is not performed via RR messages, we have to introduce a new protocol allowing the selection and release of a transcoder in the target MSC, with corresponding acknowledgement messages from the target MSC. E.g. before a new radio access bearer can be allocated via the E-interface, an additional dialogue step has to be performed between anchor and target MSC, consisting of the exchange of the messages Select Transcoder Request [Supported Codec List] and Select Transcoder Response [Selected Codec] via MAP.

The same principle would also apply to the handover procedure. However, handover is a very time critical procedure, which should not be prolonged by additional MAP signalling procedures. A more suitable, time-optimized solution would be to include the message Select Transcoder Request [Supported Codec List] together with the Handover Request in the MAP Prepare Handover Request. This would also solve the problem how to establish in the target MSC a correlation between the transcoder control and the handover operation. Another consequence of this would be that the final formatting of the Relocation Request message definitely has to be done by the target MSC, because the format of the SDU streams will depend on the selected transcoder. This would also require some communication between RR and transcoder control in the target MSC. If we look at the unsuccesful cases, e.g. if the requested radio access bearer is not available or the requested transcoder is not available, it would also be beneficial in these cases to have an internal communication between RR and transcoder control in the target MSC. Otherwise the anchor MSC might receive e.g. a (positive) Select Transcoder Response [Selected Codec], but a negative Handover Failure, and it would have to start another MAP operation only to release the transcoder in the target MSC.

So we come to the result that to get an acceptably fast and simple interworking between anchor and target MSC, a tight interworking between RR and transcoder control in the target MSC is required.

3.2 Trigger Event for the Codec Change

As we have seen in the previous section, a handover may require a change of the transcoder. In some cases this may involve also a change of the transcoder type, if the same type is not available in the target MSC. Such a change may also be necessary in case of an intra-MSC handover, if the subscriber moves from a cell with low to a cell with high traffic load and, therefore, a codec consuming less bandwidth has to be assigned.

Fig.1 shows the message flow of such a hard handover with codec change. The Selected Codec message is sent via the serving RNC before the Relocation Command [Handover

Command] message, so that the MS has the information about the new codec available in time. But the actual trigger event for applying the new codec is the receipt of the Handover Command and the change of the radio bearer, because the speech frames of the new codec require a different format of the SDU streams.

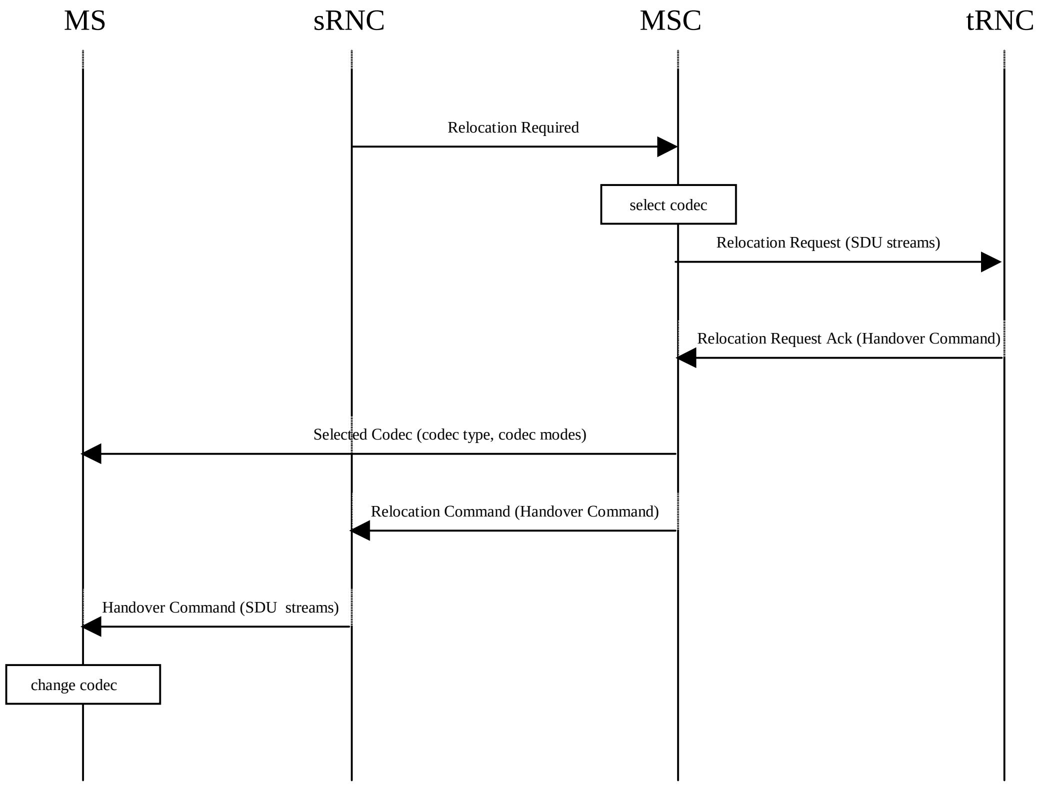

Figure 1: RNC relocation with codec change

Now we consider the following scenario (fig.2): The serving RNS has initiated a relocation procedure. The Selected Codec message has already been delivered to the MS, but the Handover Command which was contained in the Relocation Command is not relayed by the RNC. This can happen in certain glare cases, e.g. if the supervison timer $T_{RELOCprep}$ in the RNC expires between receipt of Direct Transfer [Selected Codec] and Relocation Command [Handover Command], or if the RNC has initiated an RNC-internal handover while the relocation preparation was ongoing and has relayed the Selected Codec message before it has cancelled the relocation procedure.

In this case we have to ensure that the MS does not misinterprete a Handover Command generated by the serving RNC as a trigger to activate the new codec. One possibility would be to send a second Selected Codec message which cancels the effect of the first one, however, the RNC might have already executed an internal handover before the second Selected Coded is relayed to the MS.

The obvious way out is that the MS has to analyse the contents of the Handover Command in detail, and to check whether the change of the radio bearer requires also a change of the

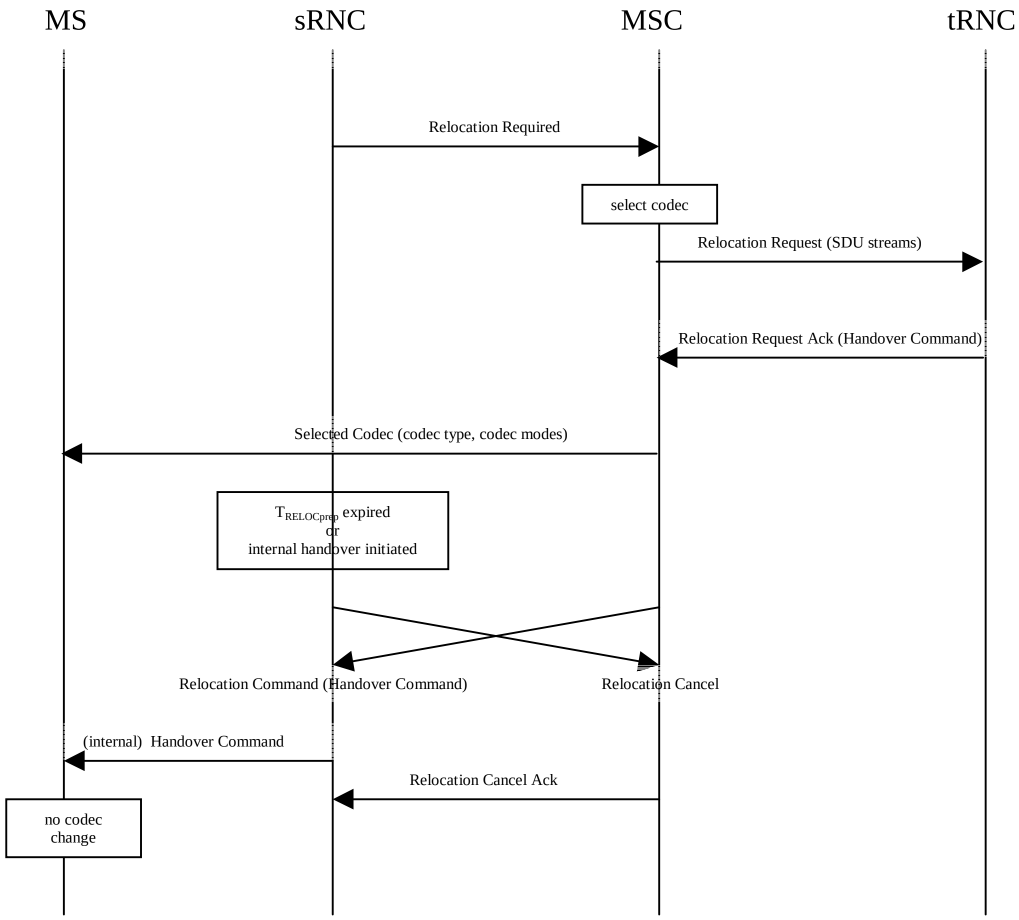

Figure 2: cancelled RNC relocation with codec change and subsequent internal handover

codec. This may be a non-trivial task, as the RNC may use a different channel coding for the same codec in different cells (resulting e.g. in different maximum bit rates in the QoS), and the SDU format can be used for this purpose only if the codec type can be derived from it unambiguously. But if we design the system in such a way, then we are actually relying on an inband signalling mechanism, and the Selected Codec message is redundant.

4. Summary and Conclusion

In section 2 we have seen that the proposed CC procedure is not compliant with the requirements of the work item description. Furthermore, because of the required independency from the transport layer (STM or ATM), a consequent separation of RR signallling and transcoder control makes the introduction of a new protocol between anchor and target MSC necessary (section 3.1). However, for MSC-MSC handover the timing requirements of the procedure probably do not allow such a strict separation. Finally, to be able to support a codec change during handover, the MS has in principle to be able to derive the codec type from the radio bearer description (section 3.2).

In summary, it is our opinion that the proposed CC procedure results in more complex signalling procedures than a combined CC/RR procedure as specified for GSM, without really achieving the postulated separation between transcoder control and RR signalling. Therefore, we propose to send a LS to RAN2, RAN3 and SMG2 WPA, asking them to include the necessary codec information for transport in downlink direction to the MS in the respective RANAP, RRC and BSSMAP messages as described in Annex A.

Annex A: Proposal for the Transport of Codec Information by CC and RR messages

1) The information about the supported codecs (Supported Codec List) is sent by the MS in the CC messages Setup (mobile originating) or Call Confirm (mobile terminating).

2) The information about the Selected Codec is sent by the MSC via the Iu interface in the RANAP messages RAB Assignment Request and Relocation Request. The Selected Codec is added to these messages as an optional information element. The MSC shall include this information element, if the MS has to assign a codec or has to change the codec together with the radio bearer assignment, re-configuration or handover.

3) If the information element is contained in the RANAP message, it has to be included by the RNC in the corresponding RRC message: Radio Bearer Setup, Radio Bearer Reconfiguration, or Handover. (Note: this list may be incomplete.)

4) In case of an MSC-MSC handover (2G->3G), or a radio bearer reconfiguration after such a handover, the Supported Codec List is transported with the BSSMAP messages Handover Request and Assignment Request via the E interface. The target MSC selects a transcoder according to the contents of list and includes the Selected Codec in the Relocation Request or RAB Assignment Request sent to the target RNC. The Selected Codec will also be reported back to the anchor MSC in Handover Request Acknowledge or Assignment Complete, respectively.

   In case of a 3G->3G handover, the anchor MSC should include also the “Selected Codec (serving)”, i.e. the currently active codec, in the Handover Request and Assignment Request message. This information may be used by the target MSC to optimize the RANAP message sent to the target RNC, i.e. to not include the Selected Codec IE in the message unless the codec has to be changed during the radio bearer reconfiguration or handover.

   In case of 3G->3G handover, if it should be decided to use RANAP at the E interface, the Supported Codec List and the information elements Selected Codec and Selected Codec (serving) have to be added also to the respective RANAP messages.

3GPP TSG-N WG1 **Tdoc 3GPP N1-0000111**
Chiba, JAPAN
10 Jan – 14 Jan 2000

Source: **Ericsson L. M.**

Agenda Item: **(Formerly) Out-of-Band Transcoder Control**
Title: **Transport of Codec Information**

Purpose: **For Information**

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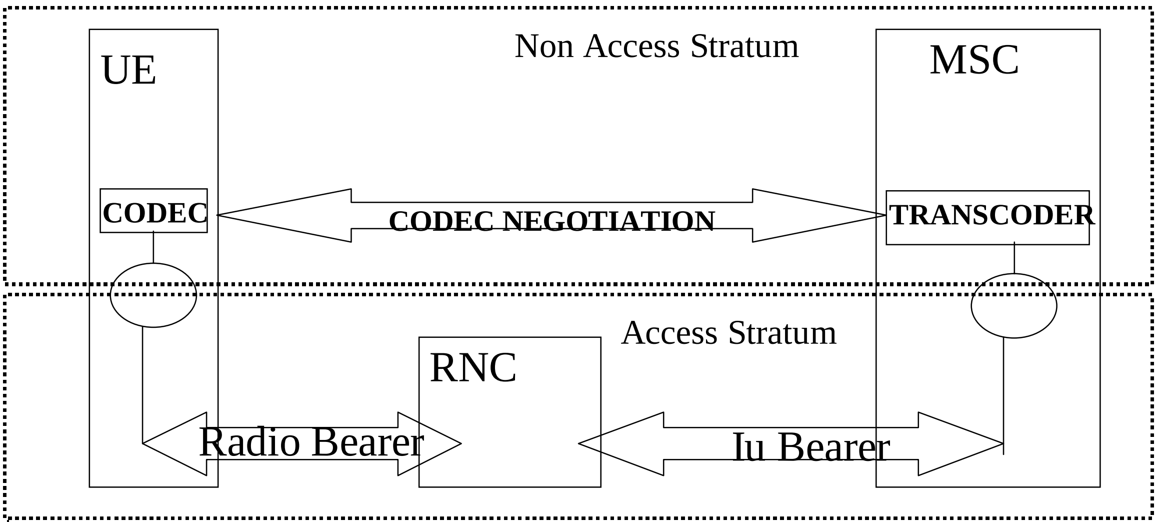
1. Introduction

Information about codecs needs to be exchanged at the access side between UE and MSC. In GSM the Bearer Capability Information Element is used to indicate supported speech codecs from the MS to the MSC. From MSC to BSC to MS the Channel Mode IE in Assignment Command is used to indicate the chosen speech version.

There are three~~two~~ problems discussed in this paper, firstly the issue of distinguishing between UMTS speech versions and GSM speech versions and secondly the issue of informing the UE of the chosen speech version. The latter discussion is a continuation of the discussion introduced to N1 in N1-99632. Thirdly, the handling of speech coding negotiation and control during and after Inter-MSC hnadover is discussed.

2. Principles

It is defined in the 3G TS 23.110 UMTS Access Stratum; Services & Functions, that speech coding is a Non Access Stratum function. This means that negotiation of speech codecs must be performed by Non Access Stratum. Non Access Stratum messages are carried in RANAP in a container – Direct Transfer Message.



~~2.~~3. Current Situation

In the UMTS TS 24.008 the speech versions supported do not differentiate between coding schemes that are to used in UMTS and those that are only to be used in GSM. In UMTS the default codec is AMR (as defined in 26.090). The coding algorithms for the UMTS AMR rates are bit exact to those in the GSM AMR (GSM 06.71) however there are some differences in the behaviour of these codecs (e.g. Rate Control frequency and the DTX framing).

If a UMTS MSC receives a SETUP message containing a list of supported speech versions as it is today in the BC IE then it must make some assumptions on what can be supported over an Iu interface and what can be supported over an A-interface. An obvious statement would be that if Speech Version 3 was included then this would be valid for both GSM (assuming Classmark indicates support of GSM) aswell as UMTS. Then all other speech versions are for GSM only - needed for intersystem handover. This assumption means that all dual system handsets must support AMR for GSM system also and that no handset can introduce GSM speech version encoding for UMTS.

The codec selection should not be based on any assumptions.

Further, in AMR there is the Supported Codec Set which contains the AMR modes which can be handled by a specific Coder/Decoder equipment and the Active Codec Set which contains the negotiated AMR rates that can be used between codecs to perform rate adjustment. In GSM the negotiation of the Active Set is performed between the BSC and the MS, and between TFO codecs using the TFO protocol. In GSM the Active Codec Set is conveyed to the MS via RR message.

For UMTS if a TRAU is required it is located in the CN and so controlled by the MSC and if OoBTC is supported to achieve the goal of ~~to achieve~~ transcoder free operation then OoB Supported Codec Set negotiation is also required. Both of these reasons mean that we cannot rely on control from the radio access or from the TFO protocol (no TFO if no Transcoders) to negotiate these sets. The use of OutOfBand signalling should be deployed as described in N1-99720.

As described in N1-99632 there is no signalling to the UE to indicate the chosen codec. There is no speech version included in the RAB assignment from MSC to RNC and so no speech version can be included in any RRC message from RNC to UE as compared with the handling in GSM.

The MSC requests a Radio Access Bearer providing the required SDU frame sizes for the chosen speech coding rates. The RNC establishes a RAB for these frames and informs the UE of the specific framing required (i.e. the SDU sizes for each Active Codec rate) in the Radio Bearer Setup but not the coding scheme itself, nor the DTX, nor the Rate Adaptation rules (frequency, stepping order). Again in the current situation the assumption that UMTS AMR is the speech version must be made and that all modes are rate adaptable according to UMTS AMR Rate Control Procedures. This prevents the call scenario of UMTS AMR to GSM AMR with TFO being possible for example.

At Inter-System Handover the Anchor MSC sends the preferred speech versions in BSSMAP message carried in MAP Prepare Handover to the Target MSC, which passes this to the BSC where the Transcoder will be located after Handover. This mechanism cannot be used for UMTS Handover as the transcoder will be located in the Target MSC (STM between Target and Anchor). The Target does not have any Call Control entity and so the codec negotiation must be performed between the Anchor MSC and the UE.

~~3.~~4. Proposed Changes

~~3.1~~4.1 UE to MSC Speech version handling

~~3.1.1~~4.1.1 Alternative 1

In UMTS TS 24.008 a new Information Element is introduced to indicate the supported speech versions for UMTS, sent from the UE to the MSC in the SETUP and CALL CONFIRMED messages. The BC IE will be used as for GSM, to indicate the supported speech versions for the GSM system i.e. in UMTS the MSC will use this list for intersystem handover.

It is further proposed that the format of the data in the new IE is taken from the UMTS TS 26.103 currently being proposed for inclusion in the ITU Q.BICC standard for Codec Negotiation. The use of Q.BICC in UMTS has been recommended to N1 by N2 in N1-99720. The introduction of this "list" in the 24.008 would further enhance the handling of the Out-Of-Band Transcoder negotiation, avoiding the need for the MSC to perform any mapping of speech versions and associated parameters from one "list" to another.

This also enables the codec type to be independent of the 24.008 protocol. This is very advantageous as it should be possible to introduce new coding schemes to the UE and CN independently of the protocol that carries the information.

~~3.1.2~~4.1.2 Alternative 2

The Current speech version codepoints are extended to indicate UMTS speech versions and are included in Octet(s) 3a of the Bearer Capability IE. These codepoints would simply indicate the codec type. Then further parameters required to fully describe the codec (Active Codec Set, Supported Codec Set) would be included in a new optional IE sent in SETUP or CALL CONFIRMED.

The coding of the new IE indicates the ACS, MACS, SCS as in 26.103 (but only this data, i.e. 3 Octets per codec type).

The advantage of this alternative is less octets are passed and the second octet is optional. However this means that if the new IE is not included then the MSC is back to the current situation of making assumptions. Secondly if more than one type of codec in the BC for UMTS needs additional parameters then the new IE must indicate the codec type that they are aplicable to - this is then duplicated information. Note this then results in 5 octets per codec type. Further, the list in the BC does not allow for GSM coding schemes to be indicated for UMTS (without creating yet more codepoints e.g. GSM FR used in UMTS). Therefore this solution is really only practical when there is only one UMTS codec, which really defeats the purpose of this WI.

~~3.1.3~~4.1.3 Recommendation

Alternative 1 is recommended because it is future proof and flexible. It is also seen as a clean solution without impacting the Bearer Capability IE. Its disadvantage is the number of octets added to the SETUP or CALL CONFIRMED messages (6 octets per codec type).

Alternative 2 is only an advantage with 1 UMTS coding scheme. One of the motivations of this WI should be to prepare the CC messages to support OoBTC with the goal of achieving optimised transcoding/transcoderless connections. This can only be achieved in the majority of connections if the terminals support a variety of coding schemes (to ensure a compatible match) thus we need a solution that can allow coding schemes to be introduced without impact to the protocol.

~~3.2~~4.2 MSC to UE Speech Version Handling

A new CC message should be introduced to indicate the chosen speech version from the MSC to the UE. In a previous paper (N1-99632) the NOTIFY message was proposed, following that it was proposed that this was unsuitable as it should not be sent during call establishment and that PROGRESS massage was more suitable. It is recommended by this paper to use a new message as any existing handling of PROGRESS message could be affected by its use for conveying Speech Codec information and more specifically instructing the UE to select a Codec Type. The sending of this new message could be inhibited for GSM systems by checking the Classmark information although the sending of a new message to a GSM only MS would simply be discarded in unrecognised and so not cause any protocol errors.

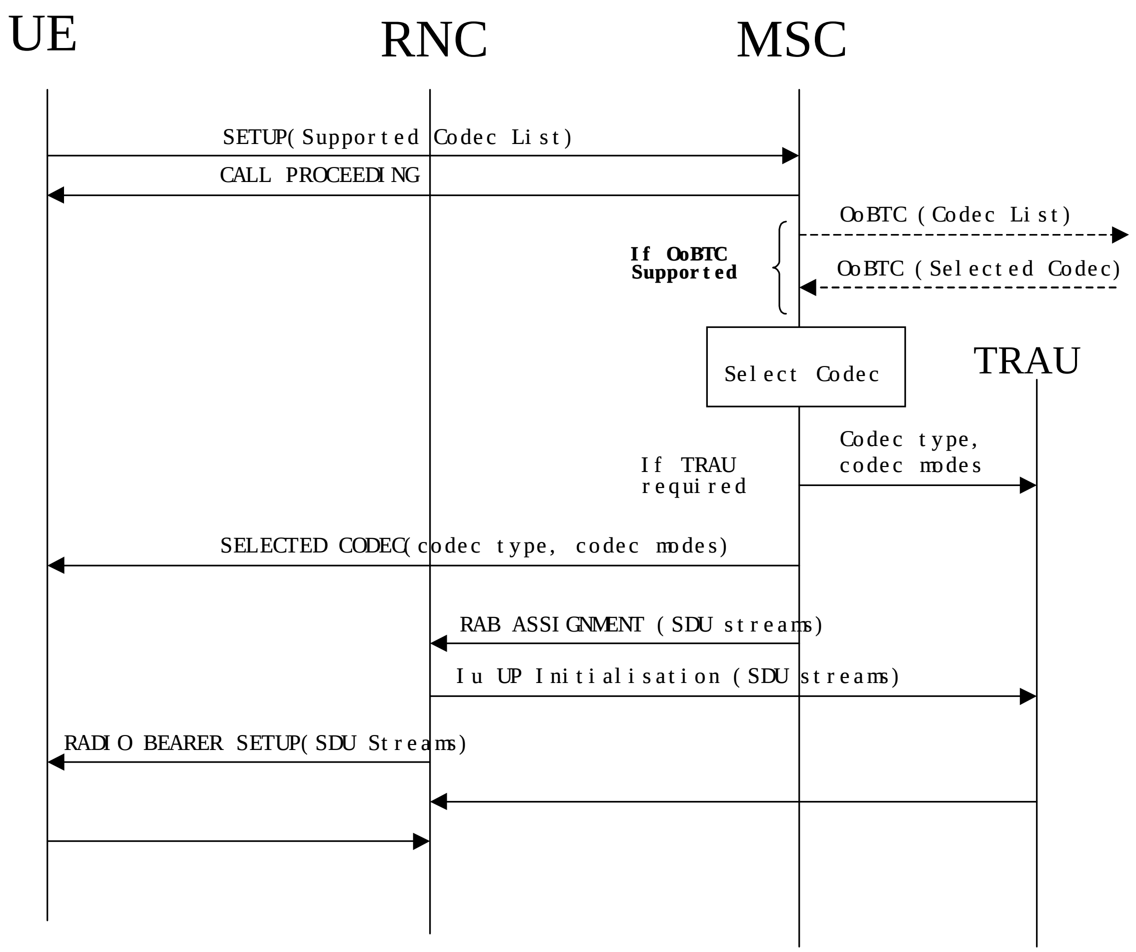


Figure 1. UMTS Originating Call Establishment

Note: It is proposed that this information is added to UMTS TS 23.108

The CR to UMTS TS 24.008 (contained in a companion contribution to this one) provides the specific changes proposed (Alternative 1).

4.3 Handover & Subsequent Supplementary Service Invocation

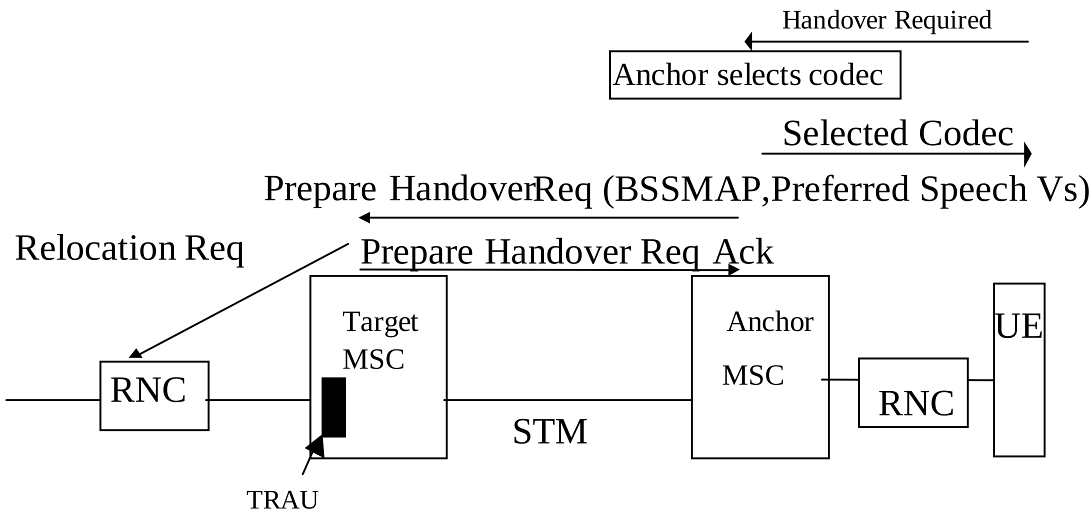
The principle with the handover is that the Anchor MSC is in control of the call. As speech codec handling is defined as part of call control then the Anchor MSC must be in control of the speech codecs. In handover situations where the bearer between Anchor and Target MSC is STM then the Anchor must have knowledge of the supported codecs for the target MSC.

Where ATM exists between MSC's then no transcoding in the target is required. Then the Anchor is in control of the codecs in the UE directly, and the transcoders in the Core Network, either directly (in its own node) or via BICC (in an edge/gateway node).

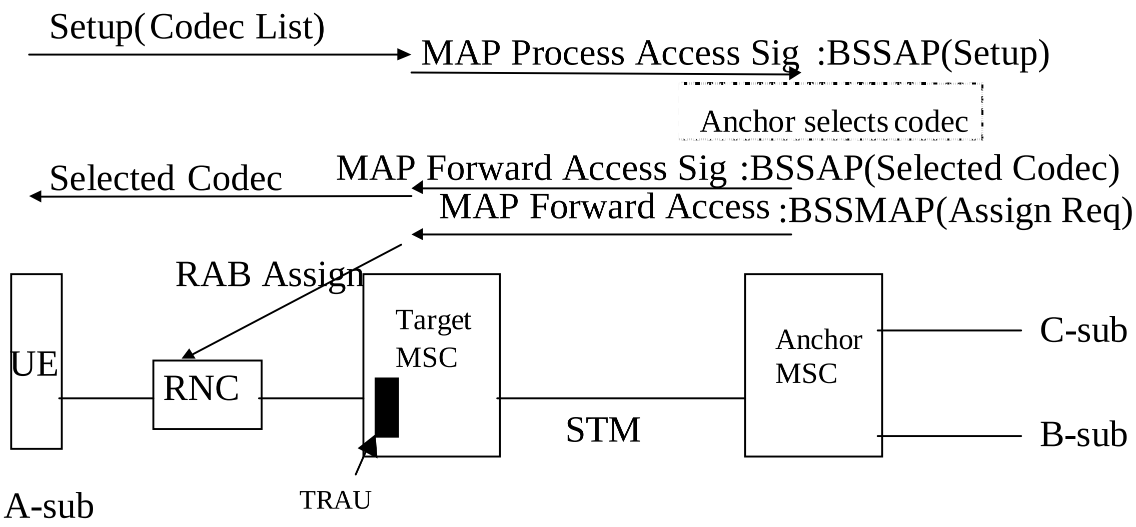
Note the handover from UMTS to GSM does not cause a problem as existing handling must be performed as the BSC is controlling the transcoder after handover.

4.3.1 Proposal 1

The first proposal is that the Anchor has pre-provisioned transcoder information for adjacent MSC's when connected with STM. The Anchor then indicates the selected codec in the Prepare Handover Request to the target MSC. It also sends the Selected Codec CC message to the UE indicating that this is the codec to be used for handover.

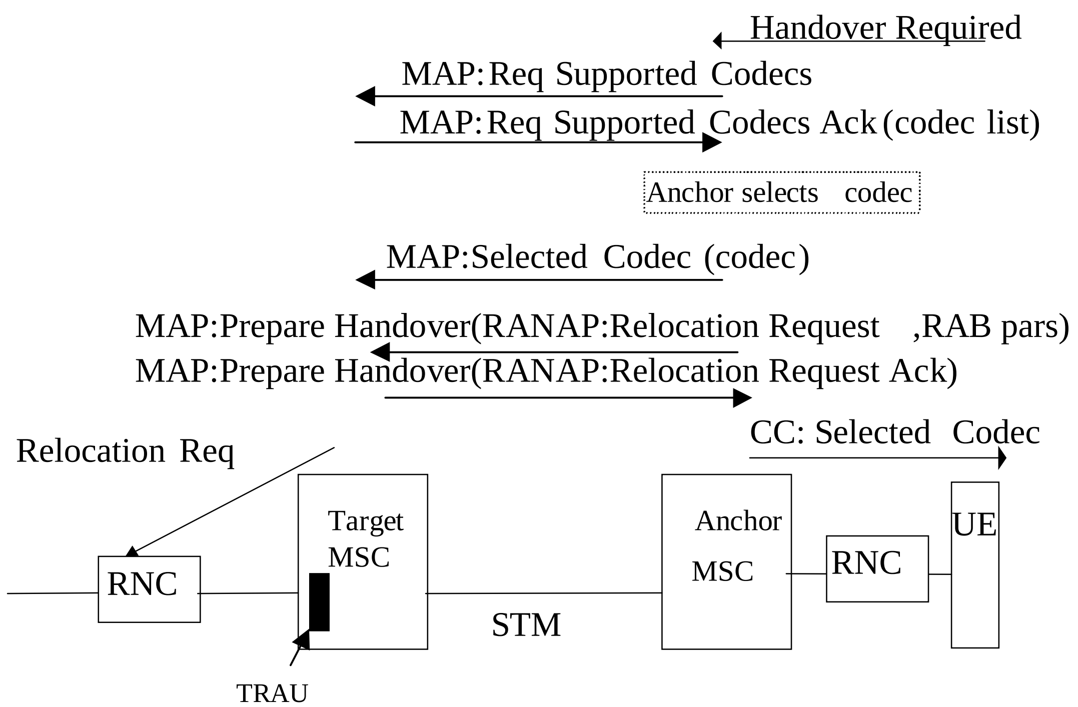


Subsequent supplementary services will be coordinated by the Anchor, if a new Setup message is received due to enquiry call, call waiting etc then the Anchor will select the codec and indicate this back to the UE in the Selected Codec message. This will also be indicated in the BSSMAP assignment to the target MSC so that the target MSC can perform the RAB assignment and also assign the Transcoder in its MSC node.

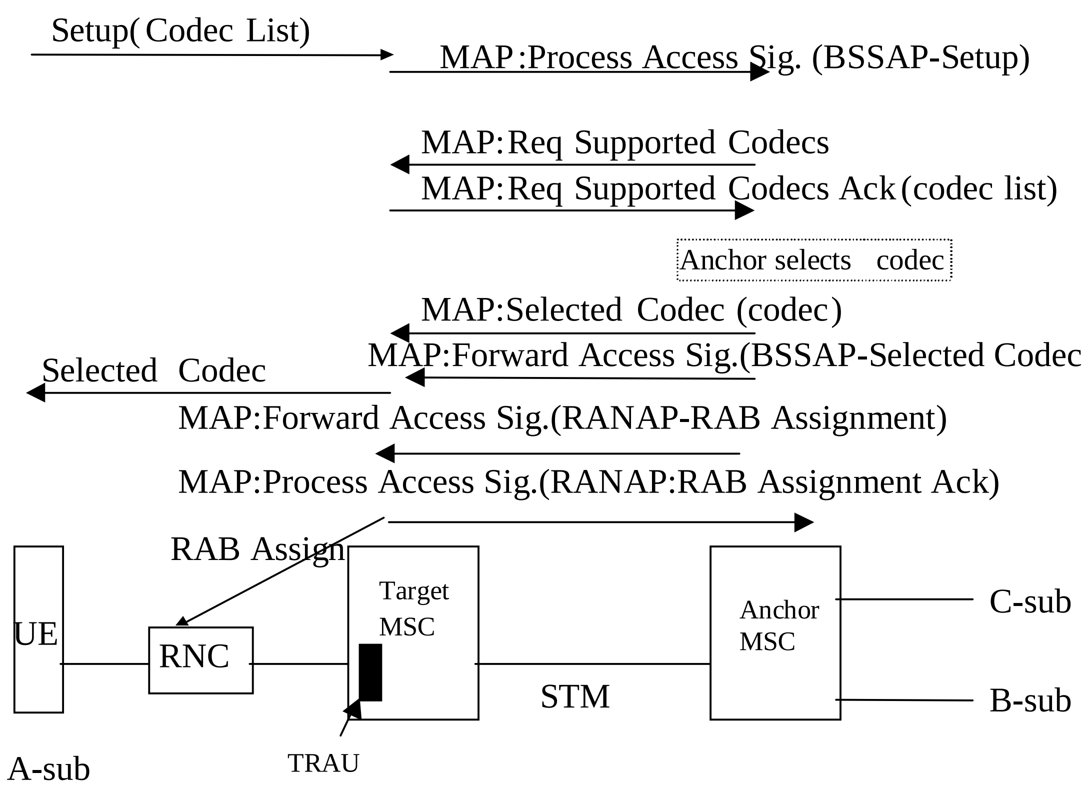


4.3.2 Proposal 2

The second proposal is that the Anchor requests this information from the target. This is proposed to be performed by a new MAP procedure. The MAP procedure will be terminated in the Target MSC and thus can be used to both request the Target MSC's codec capabilities but also to indicate the selected codec to the Target in order that the target can seize the corresponding Transcoder.



Subsequent supplementary service invocation that required any codec negotiation would also use the same mechanism, although it may be unnecessary to request the Targets capabilities if they were negotiated during handover, however if this was not performed due to the handover call be data for example then the full negotiation would be required.



4.3.3 Recommendation

Proposal 2 is recommended as it is most flexible, fits with the architectural principles, and allows for future development in the standards. With the introduction of Q.BICC and Out Of Band Transcoder Negotiation, the Target MSC should be able to negotiate codecs at any time during the call, to the UE and to the CN equipment. At handover this could reside in the Target MSC and then a MAP procedure would be the most logical means to implement this negotiation.

The CR to 3G TS 29.002 (contained in a companion contribution to this one) provides the specific changes for Proposal 2.

3GPP TSG-CN-WG1, Meeting #10
11 - 14.January. 2000
Abiko, Japan

Tdoc N1-000140

Agenda Item: **Out of Band Transcoder Control**
WI / Topic: **Out of band Transocder Control**
Source: **NTT DoCoMo**

Title: Down Link Message for transport codec information (CC vs RRC)

Releases: **R99**
Document for: **Discussion and Approval**

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1. Introduction

During the last N1 meeting #9, Ericsson and NTT DoCoMo proposed that CC message to be used for notificaion of the selected codec(N1-99F40). During the meeting Siemens stated that RRC message better to be used because the CC message has a problem at inter MSC handover.
After the meeting we realized there are two problems that when RRC is used,. These issued by RRC message are discussed in this paper, firstly the issue of functional category of codec negotiation, secondly RRC procedure makes an issue during mutlicall process. Eventually, this papser supportting CC message for the transport of down link codec information.

2. Functional Categorization of Codec Negotiation

2.1 Bearer Independent Call Conrol Protocol (BICC)

New Recommendation (Q.1901 Bearer Independent Call Control Protocol) went into determination[1] in ITU-T SG11 meeting held in December 1999. This Recommendation describes the adaptation of the narrow-band ISDN User Part (ISUP) for the support of narrow-band ISDN services independent from the bearer technology and the signalling mechanism. Additionally, it describes codec negotiation procedure.The scope of the Recommendation is a part of call control signalling. Therefore, codec negotiation function is categorized in Call Control.

2.2 Functional category of codec negotiation in UMTS according to ITU-T

Considering the functions for codec negotiation is categorized in Call Control in Q.1901, the functions for codec negotiation in UMTS also should be categorized in Call Control.
On air interface in UMTS, CC protocol corresponds to Call Control Signalling and RRC protocol correspomds to Bearer Control Signalling respectively. Therefore, functions for Call Contorol in UMTS should be realized using CC protocol, considering the functions for codec negotiation is categorized in Call Control in Q.1901.

2.3 Functional category of handover according to ITU-T

Handover function would be performed by means of RRC(and RANAP, RNSAP) protocol. Therefore, Handover procedure can be categorized in Bearer Control and it is independent from Call Control as a matter of fact.
Consequently, procedures for Codec negotiation is categorized in Call Control therefore, it should not be realized using RRC protocol.

2.4 Conclusion of CC vs RRC message for transport of codec inforamtion from functional category

The functional category point of view, codec negotiation have to be realized by using CC protocol.

3. Conclusion

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[1] „go into determination“ in ITU-T means „stable“ in 3GPP

Codec negotiation have to realized by using CC protocol because the function for codec negotiation is categorized in Call Control.

It is proposed;

- to agree that CC message is used for notification of the modified/selected codec information.
- to recognize that N1 (have to) study and slove the problem of inter MSC handover in the case where CC message is used(Solution is mentioned in N1-00XXXX Solution to the problem of inter MSC handover pointed by Siemens)

3GPP TSG-CN-WG1, Meeting #10 ***Tdoc N1-000141***
11 - 14.January. 2000, Abiko, Japan

Agenda Item: Out of Band Transcoder Control
WI / Topic: **Out of Band Transcoder Control / Down Link message**
Source: **NTT DoCoMo**
Title: **Solution to the problem of inter MSC handover pointed by Siemens**
Effected **R99:**
Document for: **Discussion and Approval**

1. Introduction

Last year Siemens delivered draft discussion paper about the Transport of Codec Information between MS and MSC for CN1#10 meeting in Abiko via N1 mailing list.
This paper explains that there is a problem on the trigger event for the codec change in the case where single CC message is used for notification of selected codec information. This problem is that in the case where anchor RNC cancels an already initiated relocation of serving RNC, ME misinterpret a Handover Command generated by the serving RNC as a trigger to activate the new codec(See, N1-990033 Simens's paper).
This document mentions the solution to the above problems and proposes the using CC message for the notification of selected codec information.

2. Solution to the trigger event for the codec change

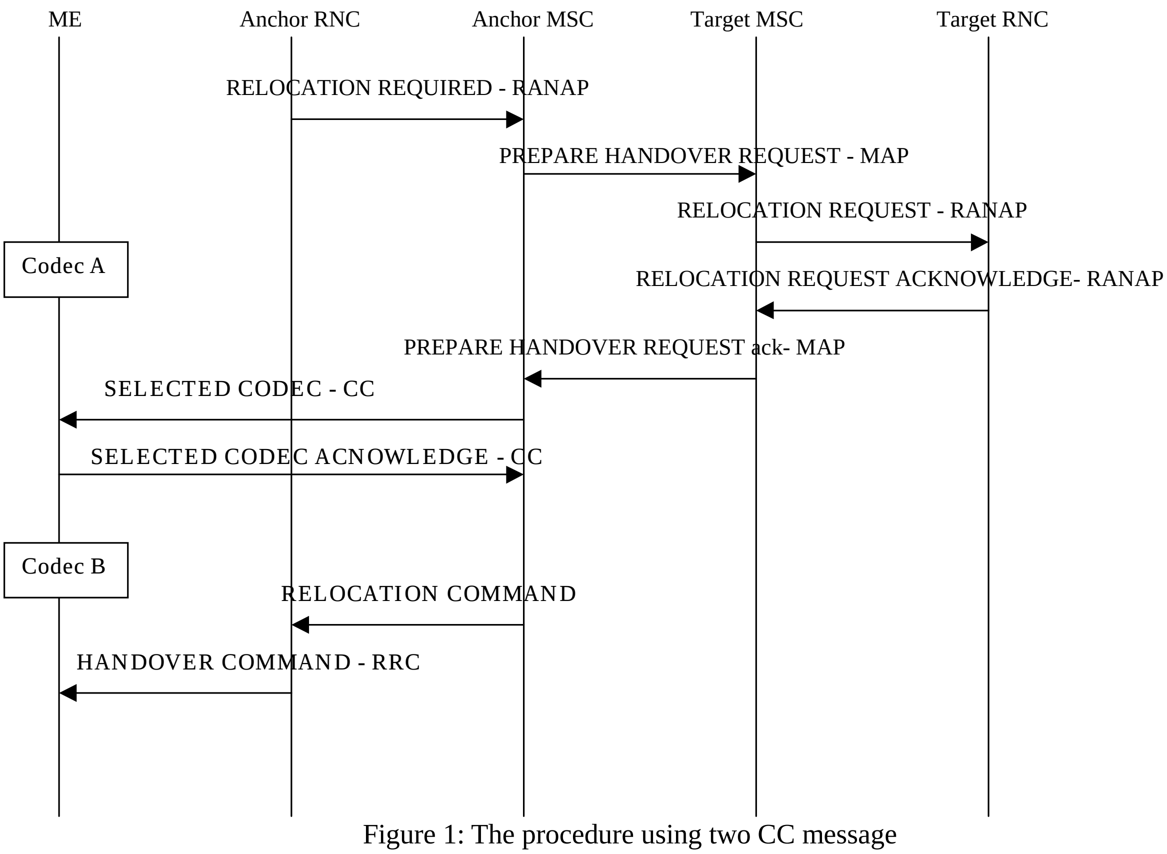

Figure 1: The procedure using two CC message

Previously we proposed an addition of selected codec message to CC was proposed for notification of selected codec information. This paper supports this concept again. Additionally, we propose addition of selected codec acknowledge message to CC in order to solve the trigger event change problem. We have considered the procedure of using selected codec message and selected codec acknowledge message (Fig 1): The anchorRNC initiates a relocation procedure. Anchor MSC knows the used codec after anchor MSC receives MAP prepare_handover_request_ack. in the both cases where anchor MSC selects the codec used in target MSC and target MSC selects the codec used in target MSC. Anchor MSC sends selected codec message to ME. Next, anchor MSC sends relocation_command message to anchor RNC after anchor MSC receives selected_codec_acknowledge message from ME. ME changes the codec as ME receives handover command message.

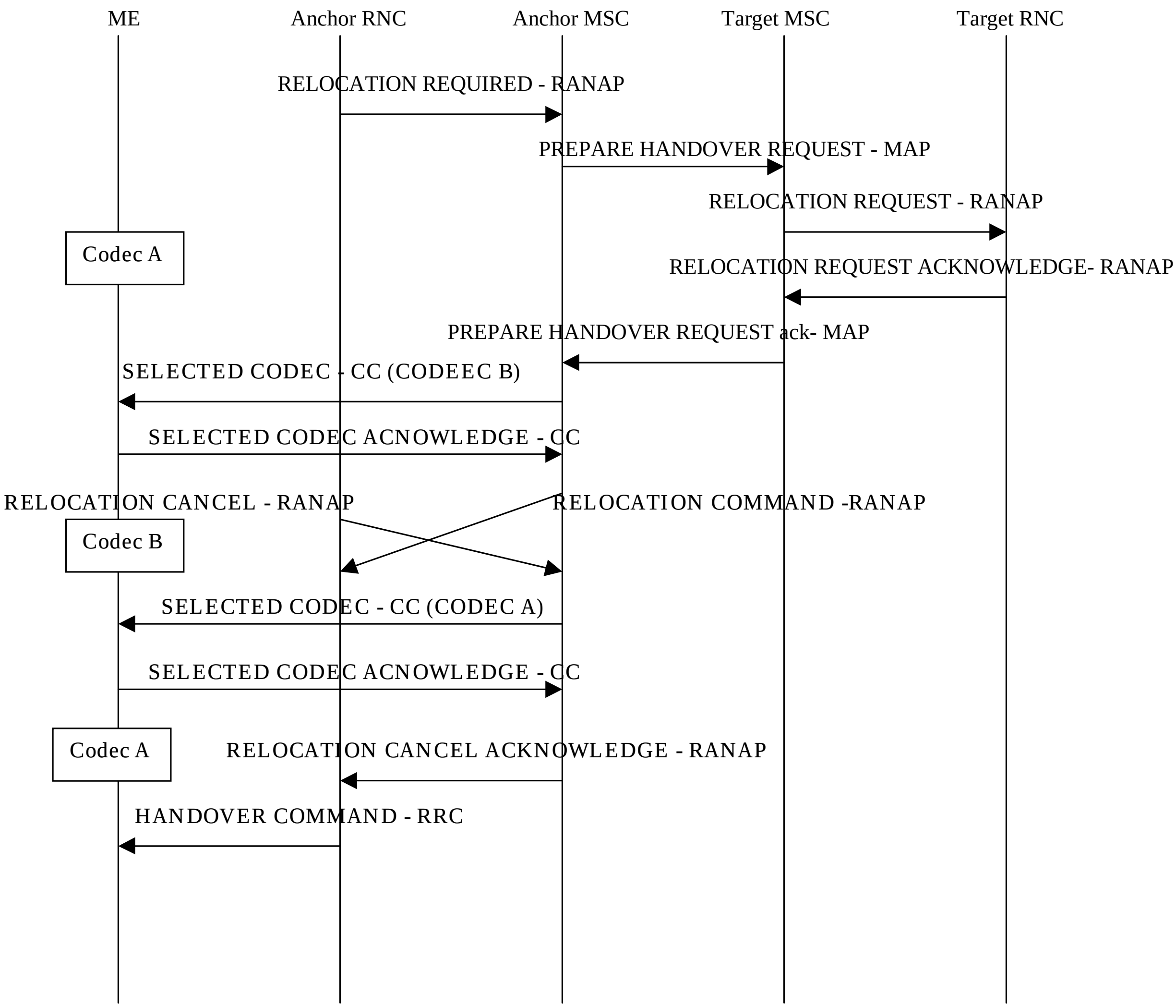

Figure 2: The procedure using two CC messages in the case where an already initiated relocation procedure is canceled

Next, we consider the procedure in the case where anchor RNC cancels an already initiated relocation of serving RNC; The anchor RNC initiates a relocation procedure. The selected codec message and handover command message are sent. But, the handover command is ignored by anchor RNC because , relocation cancel procedure is initiated and intra MSC handover is initiated. Anchor MSC sends the selected codec message contains codec information that has been used to ME after receiving cancel relocation message. Next, anchor MSC sends relocation cancel acknowledge message after receiving selected codec acknowledge message. Finally, anchor RNC sends handover command message after receiving relocation cancel acknowledge message.

The above procedure ensures that ME activates the correct codec.

3. Proposal

We propose;

- addition of a new CC message (selected codec) for the notification of the selected codec information.
- addition of another new CC message (selected codec acknowledge) to ensure that ME activates the correct codec.

3GPP TSG-N WG1 **Tdoc N1-000163**
Abiko, JAPAN
11 – 14 January 2000

Source: **Ericsson, NTTDoCoMo, Siemens AG**

Agenda Item: **Out-of-Band Transcoder Control**

Title: **Outcome of the Informal Ad-hoc Session on Out-of-Band Transcoder Control**

On the first evening of the CN1#10 meeting, an informal ad-hoc session with participants from interested companies took place to discuss the input papers from Ericsson (N1-000111), NTT DoCoMo (N1-00140, 141 and 143) and Siemens (N1-00033) related to the agenda item Out-of-Band Transcoder Control, focusing on transporting downlink codec information.

As an outcome of this ad-hoc session, Ericsson, NTTDoCoMo and Siemens agreed to propose the working assumption given below to N1 for progressing further works in the area of codec negotiation between MS and MSC. This working assumption is based on a modification of the proposal outlined in the Annex of Tdoc N1-000033, reviewing above Ericsson and DoCoMo contributions. The modification consists in including the whole CC message "Selected Codec" (including TI/PD and message type) in the RANAP and RRC messages, instead of including only the information element of Selected Codec. This was done considering the following:

The transcoder is located at the core network; therefore, codec negotiation function should be categorized in Call Control and codec negotiation should be done using only CC message, considering the concept of RAN-CN separation. However, split of RRC message and codec notification jeopardizes the success of handover, and complicates the procedure.

Another issue discussed during the ad-hoc session was the question how the information about the codecs supported by the MS should be encoded in the CC messages (sent by the MS) in the uplink direction (i.e. Setup and Call Confirmed, respectively). The two alternatives proposed are

i) to add a new codepoint for the UMTS AMR codec to the list of speech codecs in octet 3a, etc. of the bearer capability (N1-000143), or

ii) to add a new information element to the messages with a structure according to what has been specified in TS 26.103 (N1-000111)..

The participants could not yet agree on one of these alternatives and will study the matter further. We would like to continue this topic further in the rest of N1#10 meeting.

Working Assumption for the Transport of Codec Information during the Codec Negotiation between MS and MSC:

1) The information about the supported codecs (Supported Codec List) is sent by the MS in the CC messages Setup (mobile originating) or Call Confirm (mobile terminating).

2) The information about the Selected Codec is sent by the MSC via the Iu interface in the RANAP messages RAB Assignment Request and Relocation Request. The CC message Selected Codec is included in these messages as an optional information element. The MSC shall include this information element, if the MS has to assign a codec or has to change the codec together with the radio bearer assignment, re-configuration or handover.

3) If the information element is contained in the RANAP message, it has to be included by the RNC in the corresponding RRC message: Radio Bearer Setup, Radio Bearer Reconfiguration, or Handover message. (Note: this list may be incomplete.)

4) In case of an MSC-MSC handover (2G->3G), or a radio bearer reconfiguration after such a handover, the Supported Codec List is transported with the BSSMAP messages Handover Request and Assignment Request via the E interface. The target MSC selects a transcoder according to the contents of list and includes the Selected Codec message in the Relocation Request or RAB Assignment Request sent to the target RNC. The Selected Codec will also be reported back to the anchor MSC in Handover Request Acknowledge or Assignment Complete, respectively.

3GPP TSG-CN-WG1, Meeting #10
11 - 14.January. 2000
Abiko, Japan

Tdoc N1-000170

To: TSG-S1

cc: TSG-S2

Source: TSG N1

Title: LS on removal of Anonymous Access from Release 97 and 98

Contact: Roland Gruber, Siemens AG
E-mail: roland.gruber@mch.siemens.de
phone: +49 89 722 46392

As the Anonymous Access feature is deleted in the Release 99 N1 does not see any good use of this feature in the older Releases 97/98 of the GPRS specification. It seems not clear how a network that had implemented the AA feature according to R97/98 could be upgraded to R99 or newer. As there seems to be no support for this feature in R99 from any of the delegations, N1 assumes that there is also no support for AA in R97/98.

Because of this N1 like to ask S1 to confirm this assumption. If S1 agrees to delete the Anonymous Access feature also from the older releases 97 and 98 of the specifications N1 would like to ask S1 to provide the needed CRs on the stage1 specifications to N1 for information.

3GPP TSG-CN-WG1, Meeting #10
11 - 14.January. 2000
Abiko, Japan

Tdoc N1-000182

Source: TSGN1
To: TSGS1
Cc: TSGS3, TSGT3, SMG9, SMG1/9 SAT ad hoc
Title: SAT Handover notification and termination of call
Contact Person: Hannu Hietalahti
E-mail: Hannu.Hietalahti@nokia.com
Tel: +358-40-502 1724

N1 thanks S1 for their liaison in tdoc TSG S1 (99) 967 / N1-000018. The LS was discussed in TSGN1 #10 and the following questions and comments were made:

Questions:

- Which details of HO should be given to SAT?
- S1 say that SAT initiated call clearing should work analogically to call termination due to AoC limit having been reached. Aligning the SAT originated call clearing with AoC related call termination means that:
  - All calls are cleared when clearing request is received.
  - Emergency calls are not terminated.
  - MT calls not causing charging will be allowed. How does the MS know if an incoming call is free for the subscriber? This seems to require also AoC support. N1 is not aware of a service requirement which makes AoC support mandatory if SAT feature call clearing is supported.
- AoC does not currently affect packet access so is the intention to ignore PS (VoIP) for handover notification and termination of call also?
- N1 is wondering if the SAT interface already supports indication of cell change in idle mode. If such an IF exists then adding a new one for dedicated mode seems to be redundant?

Comments:

- New concept is being introduced by the proposal and this is against the principle that was agreed in TSGN #6 that we should focus on the completion of the current open issues and not invent new ones. Changing this would risk the R99 schedule.
- Using SAT termination of user initiated calls for limiting subscriber access to e.g. certain cells of the serving network is open for at least two potential sources of problems:
  - Non-supporting mobiles
  - Open SIM – ME interface allows intercepting the call clear command. N1 leaves it for S3 to assess the severity of this problem.
- N1 see that if the proposed call clearing mechanism is to be specified then a new cause value should be defined for it.
- N1 assume that the call clearing should be possible any time during the call, i.e. in all CC states. Due to this all call clearing cases should be studied.

Conclusion:
- As a consequence of these comments N1 does not see the feature feasible for R99.

3GPP TSG-CN-WG1, Meeting #10
11 - 14.January. 2000
Abiko, Japan

Tdoc N1-000193

To: **SA WG4, RAN WG3, CN WG2**

CC:

Source: **TSG CN WG1**

Title: LS on Iu Userplane Initialization at Inter MSC-HO

While N1 were looking into the Inter MSC handover for R99, we detected the following problem concerning the transcoder setting and Iu userplane initialization at Inter MSC handover that requires the attention of SA WG4, RAN WG3 and CN2.

N1 discussed the attached Tdoc N1-000144 during their N1#10 meeting. The contribution outlines and analyzes all combinations of transport layer types, i.e. ATM/STM for transcoder setting at, and after inter MSC handovers.

An issue is that in case of an inter MSC handover it may be necessary to insert a transcoder in the anchor MSC (see Annex A3 of the attached Tdoc). If transcoder free operation was established before the handover, it is not clear how the Iu userplane between the remote RNC (connected to the opposite MSC in Annex A3) and the transcoder in the anchor MSC is initialized.

N1 kindly asks SA WG4 and RAN WG3 to study this issue and take further actions to ensure that Inter MSC handover should work properly in R99.

ATTACHMENT
3GPP TSG-CN-WG1, Meeting #10 ***Tdoc N1-000144***
11 - 14.January. 2000
Abiko, Japan

Agenda Item:	**Out of Band Transcoder Control**
WI / Topic:	**Out of Band Transcoder Control**
Source:	**NTT DoCoMo**
Title:	**Clarification of Transcoder Setting**
Effected	**R99**
Document for:	**Discussion**

1. Introduction

During last N1 #9 meeting, it was clarified that transcoder re-setting is performed as inter MSC handover is performed in the case where transport layer between MSC is STM. But, in the other cases(i.e.,the case where transport layer between MSC is ATM) transcoder setting has not been clarified.
This document aims to clarify that transcoder setting at inter MSC handover and after inter MSC handover in all cases.

2. Clarification of transcoder setting

2.1 Transcoder setting at inter MSC handover

This section studies transcoder setting at inter MSC handover. In the all cases of a combination of AMT and STM transcoder settings are shown in table 3A.

Table 3A Transcoder setting AT inter-MSC HO

	Anchor MSC – Opposite MSC link (Note 1)	Anchor MSC – Target MSC link (Note 1)	Has TrFO been performed between anchor MSC and opposite MSC ?	Transcoder setting	Note
1	ATM	ATM	Yes	Not performed	See, ANNEX A-1
2	ATM	ATM	No(Note2)	Not performed	See, ANNEX A-2
3	ATM	STM	Yes	Performed in anchor MSC and target MSC	See, ANNEX A-3
4	ATM	STM	No(Note 2)	Performed in target MSC	See, ANNEX A-4
5	STM	ATM	No	Not performed	See, ANNEX A-2
6	STM	STM	No	Performed in target MSC	See, ANNEX A-4

Note1: ATM link between “anchor MSC and opposite MSC” or “anchor MSC and target MSC” means that all links between transit MSCs are ATM. STM link between “anchor MSC and opposite MSC” or “anchor MSC and target MSC” means that all links or a part of link between transit MSCs are/is STM. See ANNEX C.
Note2: Codecs supported by each ME are unmatched.

2.2 Transcoder setting AFTER inter MSC handover

In UMTS, additional call setup occurs after the inter MSC handover because UMTS supports mulicall service. Therefore, transcoder setting for another call is performed after inter MSC handover. In all cases of a combination of ATM and STM transcoder settings are shown in table 3B.

Table 3B: Transcoder setting AFTER inter MSC handover

	Anchor MSC – Opposite MSC link (note1)	Anchor MSC – Target MSC link (note1)	TrFO	Transcoder setting	Note
1	ATM	ATM	success	Not performed	See, ANNEX B-1
2	ATM	ATM	Failure (note 2)	Performed in anchor MSC	See, ANNEX B-2

2	ATM	STM	Not applied	Performed in target MSC	See, ANNEX B-3
3	STM	ATM	Failure (note 2) or not applied	Performed in anchor MSC	See, ANNEX B-2
4	STM	STM	Not applied	Performed in target MSC	See, ANNEX B-3

Note1: ATM link between “anchor MSC and opposite MSC” and“anchor MSC and target MSC” means that all links between transit MSCs are ATM. STM link between “anchor MSC and opposite MSC” or “anchor MSC and target MSC” means that all links or a part of link between transit MSCs are/is STM. See ANNEX C.
Note2: Anchor MSC supports ATM but a part of link between MSCs is STM.

3. Proposal

This document proposes;

(1) Transcoder control procedure is considered based on the transcoder setting mentioned in ANNEX A and B

(2) N1 sends LS to R3 and S4 in order that R3 and S4 conform that transcoder setting/re-setting in ANNEX A and B can be performed.

Only ANNEX-A is attached form original paper.

ANNEX A-1

Case:
Link between anchor MSC and target MSC is ATM
Link between anchor MSC and opposite MSC is ATM
Transcoder Free Operation has been performed between anchor MSC and target MSC

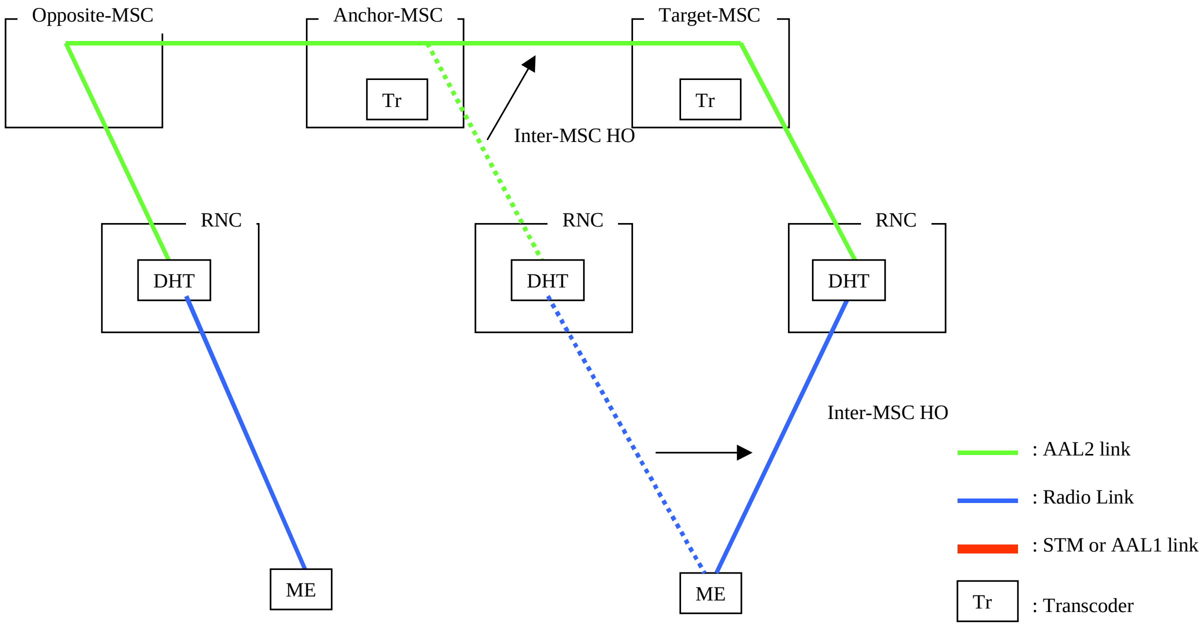

Figure 1: Transcoder setting

ANNEX A-2

Case:

Link between anchor MSC and target MSC is ATM

Link between anchor MSC and opposite MSC is ATM/STM

Transcoder Free Operation has not been performed between anchor MSC and target MSC

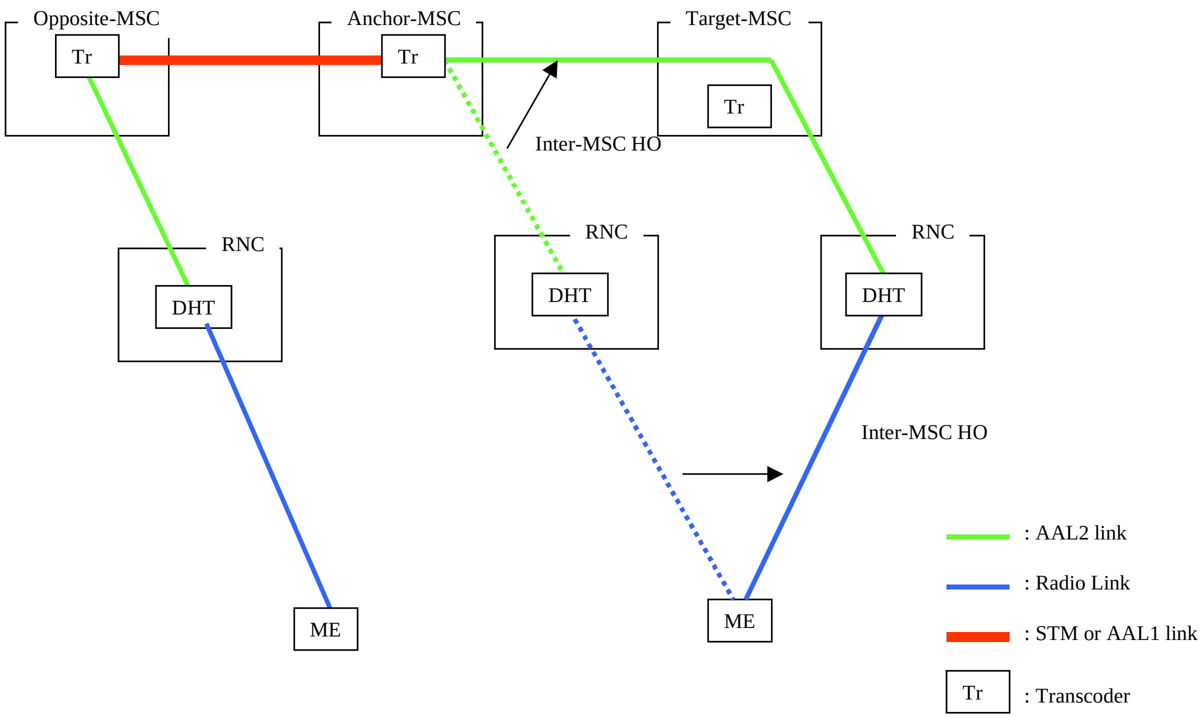

Fig: Transcoder setting

ANNEX A-3

Case:
Link between anchor MSC and target MSC is STM
Link between anchor MSC and opposite MSC is ATM
Transcoder Free Operation has been performed between anchor MSC and target MSC

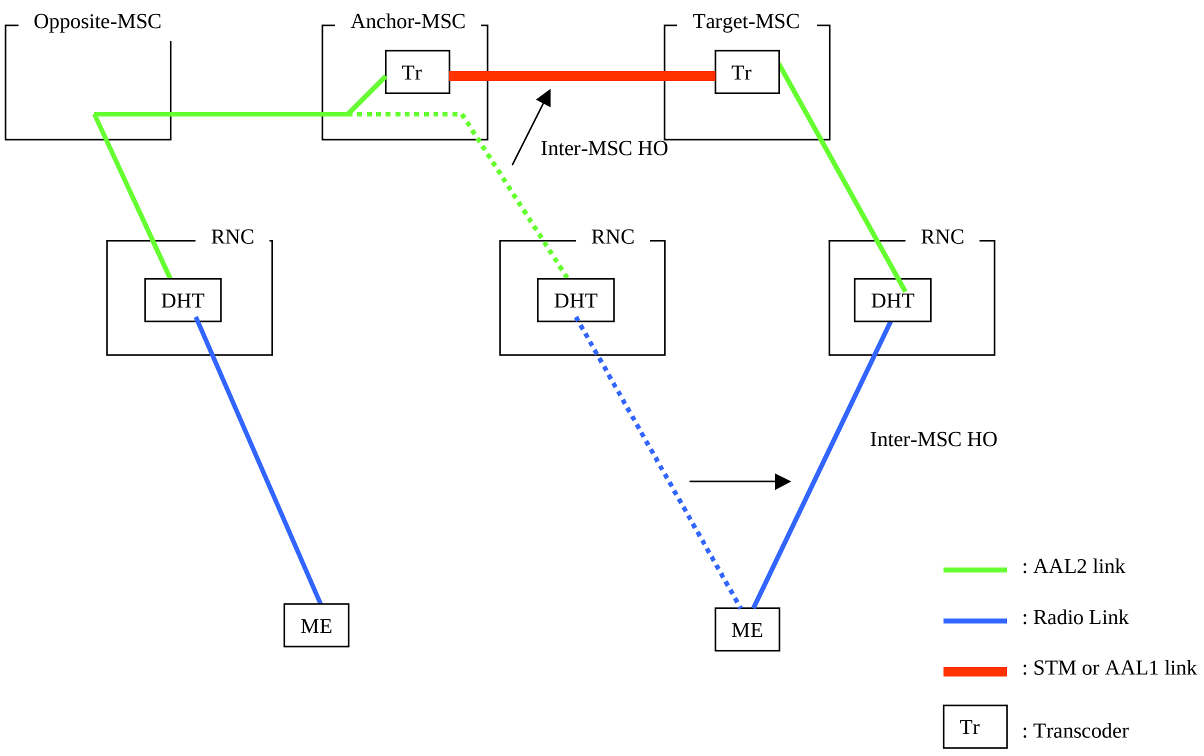

Figure: Transcoder setting

ANNEX A-4

Case:
Link between anchor MSC and target MSC is STM
Link between anchor MSC and opposite MSC is STM
Transcoder Free Operation has been performed between anchor MSC and target MSC

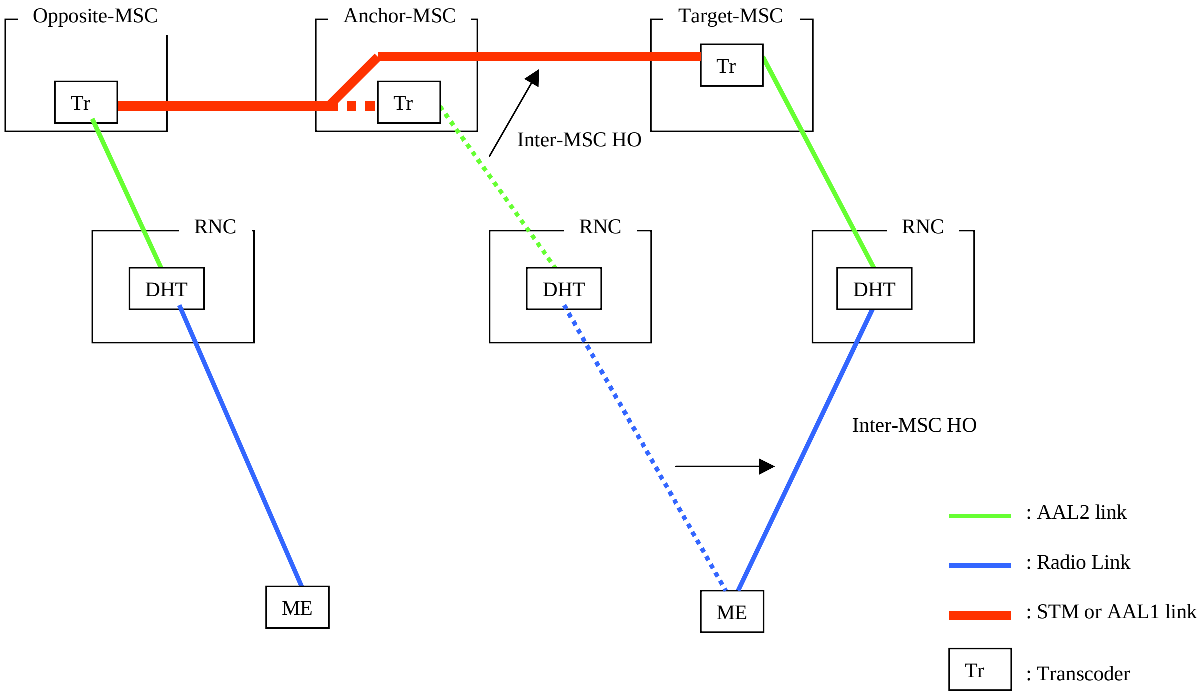

Figure: transcoder setting

3GPP TSG-CN-WG1, Meeting #10
11 - 14.January. 2000
Abiko, Japan

Tdoc N1-000201

To: TSG-S1

cc: TSG-S2, TSG-T2, TSG-T1

Source: TSG-N1

Title: Reply to LS on Requirements for Network Selection

Contact: Roland Gruber, Siemens AG
E-mail: roland.gruber@mch.siemens.de
phone: +49 89 722 46392

N1 thanks S1 for their LS on Requirements for Network Selection (S1-991056). N1 has reviewed the CR to TS 22.011 on "Network Selection" (S1-991057, attached to this LS) and like to raise the following concern:

S1 are introducing four new concepts, taking access technology, prioritisation of voice service , operator controlled PLMN selector list and Home Environment Specific Network Selection Procedure into account for PLMN selection. This is clearly against the rules which were agreed in TSG #6 when R99 was functionally frozen. These new features would risk the R99 schedule.

N1 has got only one more meeting to finalise the R99 specifications. As the requirements are still very much open, there is a risk that the requested features can not be completed for R99.

Apart from this N1 would like to raise the following concerns/questions concerning the CR to TS 22.011 on "Network Selection":

(1) As the new PLMN selection also applies to a R99 MS roaming in a R98 GSM network, should the MS assume as default that a network not indicating the new system info on speech support is an old network supporting speech?

(2) Due to the split of 23.022 into 23.122 (CN1 specific topics) and 03.22 (SMG2 specific topics) 23.022 is cancelled and 23.122 comes instead. Because of this the references to 23.022 needs to be replaced by references to 23.122. please note that 23.122 also has a title different to 23.022, when updating section 1.1 References in TS 22.011.

(3) From a technical viewpoint it is more difficult to handle two PLMN Selectors lists than one, because the storage requirements on the SIM and the mobile may be increased and it needs to be defined how to handle duplicate entries. Also it is likely that whichever list has lower priority in practice will not be used very often.

(4) In our opinion the new introduced "Operator Controlled PLMN Selector" field should not take precedence over the old now so-called "User Controlled PLMN Selector" field, because otherwise the MS behaviour concerning the Automatic PLMN selection is not any longer understandable for the user, as the users choice is probably overridden. For example it would not any longer be sure that a PLMN preferred by the user due to its cheaper price conditions is selected is available, but the more expensive one that is listed in the "Operator Controlled PLMN Selector" takes precedence. Furthermore the aim of the " Operator Controlled PLMN Selector " given in the LS from S1 : *"This will cater for those cases where the Home Environment operates more than one access technology and may use more than one network identity code."* is already fulfilled with the old PLMN selection rules, as two PLMNs with different Mobile Network Codes are treated as two independent PLMNs

also if they are both owned by the same operator. This is because the HPLMN could be extracted from the IMSI and the HPLMN has the highest priority.

(5) The new PLMN selection mode "C) Home Environment Specific Network Selection Procedure" is inserted in the middle of the old section "B) Manual network selection mode ", with the result that the last four paragraphs of the section "B) Manual network selection mode " are now not any longer applicable to B) but to the new section C). Was this the intention of S1?

(6) What was the reason that a network supporting speech service could be prioritised, but not vice versa a network supporting packet service? As there is the possibility to implement a PS only capable MS, this seems to be as useful as the prioritisation of the speech support.

(7) The technical requirements of the new introduced PLMN selection mode "C) Home Environment Specific Network Selection Procedure" are not very clear to N1. Is the intention of this feature to introduce a kind of downloading a customised PLMN selection procedure? And in order to be able to define such a procedure we like to ask for the *"set of requirements, indicated by certain parameters"*, as the kind and number of parameters needed to be implemented are not specified. As the PLMN selection procedure is tested by official type approval test case N1 would like to stress its concerns, that a MS with such a updated PLMN selection probably loose its type approval.
As a conclusion of these comments N1 does not see the feature feasible for R99.

(8) Currently the MSs are not allowed to distinguish between GSM 900 and GSM 1800 bands when performing PLMN selection. Changing this requirement would have significant impact on multiband operation feature.

(9) Based on the assumption, that cells operating in the GSM 900 band are offering the same services to the MS/user then these operating in GSM 1800, N1 does not see any use to distinguish between different GSM bands, but only between GSM and UMTS cells, as UMTS offers a enhanced service. S1 is kindly ask whether there is really a necessity to distinguish between different radio bands within GSM.

(10) As the new introduced PLMN selection rules are also applicable for R99 GSM only MS's, a default radio access technology should be defined for backwards compatibility to older SIM cards which do not specify any radio access technology.

3GPP/SMG Meeting #10 *Document* ***N1-000151***

Abiko, Japan, 11 - 14 January.2000

SMG 1 Meeting
San Diego, California, 29 Nov – 3 Dec 1999

Tdoc 1-99-1056

e.g. for 3GPP use the format TP-99xxx
or for SMG, use the format P-99-xxx

CHANGE REQUEST

Please see embedded help file at the bottom of this page for instructions on how to fill in this form correctly.

22.011 CR XXXXX Current Version: 3.1.0

GSM (AA.BB) or 3G (AA.BBB) specification number ↑ *↑ CR number as allocated by MCC support team*

For submission to:	SA#5	for approval	X	strategic	(for SMG use only)
list expected approval meeting # here ↑		for information		non-strategic	

Form: CR cover sheet, version 2 for 3GPP and SMG *The latest version of this form is available from: ftp://ftp.3gpp.org/Information/CR-Form-v2.doc*

Proposed change affects: (U)SIM [X] ME [X] UTRAN / Radio [X] Core Network [ ]
(at least one should be marked with an X)

Source: SA1 **Date:** 3/12/99

Subject: Network Selection

Work item:

Category:	F	Correction		**Release:**	Phase 2	
	A	Corresponds to a correction in an earlier release			Release 96	
(only one category	B	Addition of feature	X		Release 97	
shall be marked	C	Functional modification of feature			Release 98	
with an X)	D	Editorial modification			Release 99	X
					Release 00	

Reason for change: To clarify the requirements for network selection, including access technology type.

Clauses affected:

Other specs affected:	Other 3G core specifications		→ List of CRs:
	Other GSM core specifications	X	→ List of CRs:
	MS test specifications		→ List of CRs:
	BSS test specifications		→ List of CRs:
	O&M specifications		→ List of CRs:

Other comments:

help.doc

<--------- double-click here for help and instructions on how to create a CR.

3G TS 22.011 V3.0.1 (1999-10)

Technical Specification

3rd Generation Partnership Project;
Technical Specification Group Services and System Aspects;
Service accessibility
(3G TS 22.011 version 3.0.1)

Reference

DTS/TSGSA-0122011U

Keywords

3GPP, SA

3GPP

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Contents

Foreword

This Technical Specification has been produced by the 3GPP.

The contents of the present document are subject to continuing work within the TSG and may change following formal TSG approval. Should the TSG modify the contents of this TS, it will be re-released by the TSG with an identifying change of release date and an increase in version number as follows:

Version 3.y.z

where:

x the first digit:

1 presented to TSG for information;

2 presented to TSG for approval;

3 Indicates TSG approved document under change control.

y the second digit is incremented for all changes of substance, i.e. technical enhancements, corrections, updates, etc.

z the third digit is incremented when editorial only changes have been incorporated in the specification;

1 Scope

The purpose of this TS is to describe the service access procedures as presented to the user.

Definitions and procedures are provided in this TS for international roaming, national roaming and regionally provided service. These are mandatory in relation to the technical realization of the Mobile Station (~~MS~~UE).

~~Note: The present document covers description for GSM only. The document needs to be updated to make it applicable to 3GPP.~~

1.1 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies.
- For this Release 1999 document, references to GSM documents are for Release 1999 versions (version 8.x.y).

[1] GSM 01.04: "Digital cellular telecommunications system (Phase 2+); Abbreviations and acronyms".

[2] TR 21.905: " Vocabulary for 3GPP Specifications". features".

[3] TS 23.022: "Functions related to Mobile Station (~~MS~~UE) in idle mode and group receive mode".

[4] ITU-T Recommendation Q.1001: "General aspects of Public Land Mobile Networks".

[5] TS 22.043: " Support of Localised Service Area (SoLSA). Stage 1 ".

[6] TR 21.905: "Vocabulary for 3GPP Specifications"

1.2 Definitions and abbreviations

In addition to those below, abbreviations used in this TS are listed in GSM 01.04 [1] and TR 21.905 [6].

PLMN

A Public Land Mobile Network (PLMN) is a network established and operated by an Administration or RPOA for the specific purpose of providing land mobile communication services to the public. It provides communication possibilities for mobile users. For communications between mobile and fixed users, interworking with a fixed network is necessary.

A PLMN may provide service in one, or a combination, of frequency bands.

As a rule, a PLMN is limited by the borders of a country. Depending on national regulations there may be more than one PLMN per country.

A relationship exists between each subscriber and his home PLMN (HPLMN). If communications are handled over another PLMN, this PLMN is referred to as the visited PLMN (VPLMN).

PLMN Area

The PLMN area is the geographical area in which a PLMN provides communication services according to the specifications to mobile users. In the PLMN area, the mobile user can set up calls to a user of a terminating network. The terminating network may be a fixed network, the same PLMN, another PLMN or other types of PLMN.

Terminating network users can also set up calls to the PLMN.

The PLMN area is allocated to a PLMN. It is determined by the service and network provider in accordance with any provisions laid down under national law. In general the PLMN area is restricted to one country. It can also be determined differently, depending on the different telecommunication services, or type of ~~MS~~UE.

If there are several PLMNs in one country, their PLMN areas may overlap. In border areas, the PLMN areas of different countries may overlap. Administrations will have to take precautions to ensure that cross border coverage is minimized in adjacent countries unless otherwise agreed.

NOTE 1: ITU-T Recommendation Q.1001 [4] does not contain a definition of the PLMN area.

System Area

The System Area is defined as the group of PLMN areas accessible by ~~MS~~UEs.

Interworking of several PLMNs and interworking between PLMNs and fixed network(s) permit public land mobile communication services at international level.

NOTE 2: The System Area according to [4] Recommendation Q.1001 corresponds to the System Area.

Service Area

The Service Area is defined in the same way as the Service Area according to ITU-T Recommendation Q.1001 [4]. In contrast to the PLMN area it is not based on the coverage of a PLMN. Instead it is based on the area in which a fixed network user can call a mobile user without knowing his location. The Service Area can therefore change when the signalling system is being extended, for example.

Regionally Provided Service

Regionally Provided Service is defined as a service entitlement to only certain geographical part(s) of a PLMN, as controlled by the network operator.

Localised Service Area (LSA)

The localised service area concept shall give the operator a basis to offer subscribers different services (e.g. tariffs or access rights) depending on the location of the subscriber. A LSA consists of a cell or a number of cells within a PLMN. (TS 22.043 [5]).

2 Roaming

2.1 General requirements

A ~~MS~~UE with a valid IMEI may roam and access service in the area authorized by the entitlement of the subscription.

If a communication has been established, the ~~MS~~UE will in principle not suffer an interruption within the PLMN area (provided the entitlement of the subscription allows it). Exceptions are possible if no network resources or radio coverage are available locally.

However, if the ~~MS~~UE leaves the PLMN area, an established communication may terminate. If the user then wants to continue, another network providing service has to be selected and a new communication has to be established (see clause 3).

2.2 International roaming

International roaming is a service whereby an ~~MS~~UE of a given PLMN is able to obtain service from a PLMN of another country.

The availability of International Roaming is subject to inter-PLMN agreements.

2.3 National roaming

National Roaming is a service whereby an ~~MS~~UE of a given PLMN is able to obtain service from another PLMN of the same country, anywhere, or on a regional basis.

The availability of National Roaming depends on the home PLMN of the requesting ~~MS~~UE and the visited PLMN; it does not depend on subscription arrangements.

3 Provisions for providing continuity of service

3.1 Location registration

PLMNs shall provide a location registration function with the main purpose of providing continuity of service to ~~MS~~UEs over the whole system area. The location registration function shall be such as to allow:

- Fixed subscribers to call a ~~MS~~UE by only using the directory number of the ~~MS~~UE irrespective of where the ~~MS~~UE is located in the system area at the time of the call.
- ~~MS~~UEs to access the system irrespective of the location of the ~~MS~~UE.
- ~~MS~~UEs to identify when a change in location area has taken place in order to initiate automatic location updating procedures.

3.2 Network selection

3.2.1 General

The ~~MS~~UE shall support both manual and automatic network selection mechanisms (modes). The ~~MS~~UE shall select the last mode used, as the default mode, at every switch-on.

NOTE: By defaulting to the last mode used, e.g. manual network selection, the undesired automatic selection of an adjacent PLMN instead of the desired HPLMN in border areas, can be avoided at switch-on.

The user shall be given the opportunity to change mode at any time.

Except as defined below, the MMI shall be at the discretion of the ~~MS~~UE manufacturer.

The ~~MS~~UE shall contain display functions by which Available PLMNs and the Selected PLMN can be indicated.

3.2.2 Procedures

3.2.2.1 General

In the following procedures the ~~MS~~UE selects and attempts registration on PLMNs.

In this ETS, the term "PLMN Selection" defines a~~n~~ ~~MS~~UE based procedure, whereby candidate PLMNs are chosen, one at a time, for attempted registration.

If registration on a PLMN is successful, the ~~MS~~UE shall indicate this PLMN (the "registered PLMN") and be capable of making and receiving calls on it. The identity of the registered PLMN shall be stored on the ~~SIM~~SIM/USIM. However, if registration is unsuccessful, the ~~MS~~UE shall ensure that there is no registered PLMN stored in the ~~SIM~~SIM/USIM.

If a registration is unsuccessful because the I~~MS~~UEI is unknown in the home network, or the ~~MS~~UE is illegal, then the ~~MS~~UE shall not allow any further registration attempts on any network, until the ~~MS~~UE is next powered-up or a ~~SIM~~SIM/USIM is inserted.

If the registration is unsuccessful due to the lack to service entitlement, specific behaviour by the ~~MS~~UE may be required, see subclause 3.2.2.4.

To avoid unnecessary registration attempts, lists of forbidden PLMNs and LAs are maintained in the ~~MS~~UE, see subclause 3.2.2.4 and TS 23.022 [3].

Registration attempts shall not be made by ~~MS~~UEs without a ~~SIM~~SIM/USIM inserted.

A~~n~~ ~~MS~~UE/ME which has not successfully registered shall nevertheless be able to make emergency call attempts on an available PLMN (which supports the emergency call teleservice), without the need for the user to select a PLMN. An available PLMN is determined by radio characteristics (TS 23.022 [3]).

It shall be possible to have an Operator Controlled PLMN Selector list and a User Controlled PLMN Selector list stored on the SIM/USIM card. Both PLMN Selector lists may contain a list of preferred PLMNs in priority order. It shall be possible to have an associated Access Technology identifier e.g., UTRA, GSM900 or GSM1800 associated with each entry in the PLMN Selector lists.

NOTE 1: A PLMN in a Selector list, including HPLMN, may have multiple occurrences, with different access technology identifiers.

3.2.2.2 At switch-on or recovery from lack of coverage

If the ~~MS~~UE is within coverage (at switch-on) or returns to coverage of the PLMN on which it is already registered (as indicated by the registered PLMN stored in the SIM/USIM), the ~~MS~~UE shall perform a location update to a new location area if necessary.

If there is no registered PLMN stored in the SIM/USIM, or if this PLMN is unavailable, or the attempted registration fails, the ~~MS~~UE shall follow one of the following~~ two~~ procedures ~~depending on its~~for network selection ~~mode, automatic or manual~~:

A) Automatic network selection mode

The ~~MS~~UE shall select and attempt registration on other PLMNs, if available and allowable and the location area is not in the list of "forbidden LSs for roaming" (see TS 23.022 [3]), in the following order:

i) HPLMN for preferred access technology. It shall be possible to configure a voice capable UE so that it shall not attempt registration on a PLMN if all cells identified as belonging to the PLMN do not support the corresponding voice service;

ii) each PLMN in the "Operator Controlled PLMN Selector" data field in the SIM/USIM (in priority order). It shall be possible to configure a voice capable UE so that it shall not attempt registration on a PLMN if all cells identified as belonging to the PLMN do not support the corresponding voice service

iii) each PLMN in the "User Controlled PLMN Selector" data field in the ~~SIM~~SIM/USIM (in priority order). It shall be possible to configure a voice capable UE so that it shall not attempt registration on a PLMN if all cells identified as belonging to the PLMN do not support the corresponding voice service;

~~iii)~~iv) other PLMNs with sufficient received signal quality~~level~~ (see TS 23.022 [3]) in random order. It shall be possible to configure a voice capable UE so that it shall not attempt registration on a PLMN if all cells identified as belonging to the PLMN do not support the corresponding voice service;

~~i~~v) all other PLMNs in order of decreasing signal quality~~strength~~ . It shall be possible to configure a voice capable UE so that it shall not attempt registration on a PLMN if all cells identified as belonging to the PLMN do not support the corresponding voice service.

An allowable PLMN is one which is not in the "Forbidden PLMN" data field in the ~~SIM~~SIM/USIM . This data field may be extended in the ME memory.(see subclause 3.2.2.4).

If successful registration is achieved, the ~~MS~~UE shall indicate the selected PLMN.

If registration cannot be achieved on any PLMN, the ~~MS~~UE shall indicate "no service" to the user, wait until a new PLMN is detected, or new location areas of an allowed PLMN are found which are not in the forbidden LA list(s), and then repeat the procedure. When registration cannot be achieved, different (discontinuous) PLMN search schemes may be used in order to minimize the access time while maintaining battery life, e.g. by prioritizing the search in favour of BCCH carriers which have a high probability of belonging to an available and allowable PLMN.

B) Manual network selection mode

The ~~MS~~UE shall indicate~~ whether there are any~~ PLMNs, including "Forbidden PLMNs", which are available. If there are none, this shall also be indicated.

Any available PLMN's shall be presented in the following order:

i) ~~i)~~HPLMN;

ii) PLMNs contained in the "Operator Controlled PLMN Selector" data field in the SIM/USIM (in priority order);

ii) PLMNs contained in the "User Controlled PLMN Selector" data field in the ~~SIM~~SIM/USIM (in priority order);

iii) other PLMNs with sufficient received signal level (see TS 23.022 [3]) in random order;

iv) all other PLMNs in order of decreasing signal strength.

If a PLMN does not support voice services then this shall be indicated to the user.

The user may select his desired PLMN and the ~~MS~~UE shall attempt registration on this PLMN. (This may take place at any time during the presentation of PLMNs.)

C) Home Environment Specific Network Selection Procedure

Optionally, if provisioned by the UE and selected by the user, the home environment can add the ability to define the behaviour when selecting the required network from those available. A standardised framework for over-the-air transfer of behaviour definition is required. If enabled by the user, it shall be possible for the home environment procedure to instruct the mobile station to search for a network which meets a given set of requirements, indicated by certain parameters or to compile a list of all available networks.

A USIM/SIM shall be registered on one and only one serving network at any given time (there may be exceptions such as when preparing for an inter-network handover). Changing the serving network between two calls requires USIM/SIM de-registration from the current serving network and USIM/SIM registration on the newly selected serving network.

If the registration cannot be achieved on the selected PLMN, the ~~MS~~UE shall indicate "No Service". The user may then select and attempt to register on another or the same PLMN following the above procedure. The ~~MS~~UE shall not attempt to register on a PLMN which has not been selected by the user.

If a PLMN is selected but the ~~MS~~UE cannot register on it because registration is rejected with the cause "PLMN not allowed", the ~~MS~~UE shall not re-attempt to register on that network unless the same PLMN is selected again by the user.

If a PLMN is selected but the ~~MS~~UE cannot register on it for other reasons, the ~~MS~~UE shall, upon detection of a new LA (not in a forbidden LA list) of the selected PLMN, attempt to register on the PLMN.

If the ~~MS~~UE is registered on a PLMN but loses coverage, different (discontinuous) carrier search schemes may be used to minimize the time to find a new valid BCCH carrier and maintain battery life, e.g. by prioritizing the search in favour of BCCH carriers of the registered PLMN.

3.2.2.3 User reselection

At any time, the user may request the ~~MS~~UE to initiate reselection and registration onto an alternative available PLMN, according to the following procedures, dependent upon the operating mode.

A) Automatic Network Selection Mode

The ~~MS~~UE shall select the HPLMN. If the HPLMN is not available, the ~~MS~~UE shall follow the procedure defined in clause 3.2.2.2 A) above~~select the PLMNs in the "PLMN Selector" list in order of priority and, if necessary, other available and allowable PLMNs according to the procedure defined in TS 23.022 [3].~~

B) Manual Network Selection Mode

The procedure of 3.2.2.2 B) is followed.

C) Home Environment Specific Network Selection Procedure

The HE specific procedure as described in 3.2.2.2 C) above is followed if available.

3.2.2.4 Mobile Station reactions to indications of service restriction from the network

Different types of ~~MS~~UE behaviour is required to support, for example, national roaming, regionally provided service and temporary international roaming restrictions. The behaviour to be followed by the ~~MS~~UE is indicated by the network.

3.2.2.4.1 "Permanent" PLMN restriction

When a registration attempt by the ~~MS~~UE is rejected by a network with an indication of "permanent" PLMN restriction, the PLMN identity shall be written to a list of "Forbidden PLMNs" stored in a data field in the ~~SIM~~SIM/USIM.

If a successful registration (whilst in manual mode) is achieved on a PLMN in the "Forbidden PLMN" list, the PLMN shall be deleted from the list.

When in automatic mode, the ~~MS~~UE may indicate any PLMNs which will not be selected due to their presence in the "Forbidden PLMN" list.

3.2.2.4.2 "Partial" and "temporary" PLMN restrictions

When a registration attempt by the ~~MS~~UE is rejected by a network due to a "partial" or a "temporary" PLMN restriction, the ~~MS~~UE shall perform one of the following procedures determined by the indication in the location update reject cause sent by the network (see TS 23.022 [3]):

i) The ~~MS~~UE shall store the location area identity in the list of "forbidden LAs for regional provision of service" and shall enter the limited service state and remain in that state until it moves to a cell in a location area where service is allowed.

ii) The ~~MS~~UE shall store the location area identity in the list of "forbidden LAs for roaming" and shall use one of the following procedures according to the PLMN selection Mode:

A) Automatic network selection mode:

The procedure of 3.2.2.2. A).

B) Manual network selection mode:

The procedure of 3.2.2.2.B).

3.2.2.5 Timer for return to HPLMN

If the ~~MS~~UE in Automatic Mode has selected and registered on a VPLMN of its home country, it shall make periodic attempts to return to its HPLMN.

The interval between attempts shall be stored in the ~~SIM~~SIM/USIM. Only the service provider shall be able to set the timer value. The timer shall have a value between 6 minutes and 8 hours, with a step size of 6 minutes. One value shall be designated to indicate that no periodic attempts shall be made.

In the absence of a permitted value in the ~~SIM~~SIM/USIM, or the ~~SIM~~SIM/USIM is phase 1 and therefore does not contain the datafield, then a default value of 30 minutes, shall be used by the ~~MS~~UE.

NOTE: Use of values less than 30 minutes may result in excessive ME battery drain.

4 Access control

4.1 Purpose

Under certain circumstances, it will be desirable to prevent ~~MS~~UE users from making access attempts (including emergency call attempts) or responding to pages in specified areas of a PLMN. Such situations may arise during states of emergency, or where 1 of 2 or more co-located PLMNs has failed.

Broadcast messages should be available on a cell by cell basis indicating the class(es) of subscribers barred from network access.

The use of this facility allows the network operator to prevent overload of the access channel under critical conditions.

It is not intended that access control be used under normal operating conditions.

4.2 Allocation

All ~~MS~~UEs are members of one out of ten randomly allocated mobile populations, defined as Access Classes 0 to 9. The population number is stored in the ~~SIM~~SIM/USIM. In addition, mobiles may be members of one or more out of 5 special categories (Access Classes 11 to 15), also held in the ~~SIM~~SIM/USIM. These are allocated to specific high priority users as follows. (The enumeration is not meant as a priority sequence):

Class 15 - PLMN Staff;

-"- 14 - Emergency Services;

-"- 13 - Public Utilities (e.g. water/gas suppliers);

-"- 12 - Security Services;

-"- 11 - For PLMN Use.

4.3 Operation

If the ~~MS~~UE is a member of at least one Access Class which corresponds to the permitted classes as signalled over the air interface, and the Access Class is applicable in the serving network, access attempts are allowed. Otherwise access attempts are not allowed.

Access Classes are applicable as follows:

Classes 0 - 9 - Home and Visited PLMNs;

Classes 11 and 15 - Home PLMN only;

Classes 12, 13, 14 - Home PLMN and visited PLMNs of home country only.

Any number of these classes may be barred at any one time.

4.4 Emergency Calls

An additional control bit known as "Access Class 10" is also signalled over the air interface to the ~~MS~~UE. This indicates whether or not network access for Emergency Calls is allowed for ~~MS~~UEs with access classes 0 to 9 or without an IMEI. For ~~MS~~UEs with access classes 11 to 15, Emergency Calls are not allowed if both "Access class 10" and the relevant Access Class (11 to 15) are barred. Otherwise, Emergency Calls are allowed.

5 Support of Localised Service Area (SoLSA)

SoLSA consists of a set of service features that give the operator a basis to offer subscribers different services (e.g. tariffs or access rights) depending on the location of the subscriber. (TS22.043 [5]). The following section is only applicable to the support of SoLSA functionality.

5.1 Network selection

The standard automatic and manual network selection procedures will be used.

Manual network selection may be required when the PLMN providing the users SoLSA service is not the one on which the user is currently registered.

At manual network selection the ~~MS~~UE shall provide the means to present the subscribers LSA(s) for each PLMN presented.

5.2 The Idle-mode operation

The ~~MS~~UE shall always select a valid LSA with the highest priority.

5.2.1 Subscriber moving from a normal environment to his localised service area.

The ~~MS~~UE shall have the ability to prioritise allowed LSA cells in reselection, making it possible to camp on a LSA cell earlier (the function shall be network controlled).

5.2.2 Subscriber moving away from his localised service area to a normal environment.

The ~~MS~~UE shall have the ability to prioritise allowed LSA cells in reselection, making it possible to camp on a LSA cell longer (the function shall be network controlled).

5.2.3 Subscriber staying in his localised service area

The ~~MS~~UE shall have the ability to prioritise allowed LSA cells in reselection by being more persistent (the function shall be network controlled).

NOTE: Typically in indoor environments there are occasional reflections and "disturbances" due to macro cells, e.g. near the windows. In such a case LSA cells should be favoured even though there is higher field strength available from the outdoor cells.

5.3 LSA only access

It shall be possible to allow LSA user to access PLMN only within his LSAs. A LSA user is not allowed to receive and/or originate a call outside the allowed LSA area.

When ~~MS~~UE is out of the allowed LSA area it shall be registered in PLMN but indicate subscriber/service specific "out of LSA area" notification. It shall be a network controlled function to prevent terminated or/and originated calls. Emergency calls are however always allowed.

5.4 Exclusive access

Access to exclusive access cells is restricted to defined LSA subscribers.

Non-LSA subscriber shall consider exclusive access cells as not suitable, only allowing to camp for emergency calls (limited service state TS 23.022 [3]).

5.5 Preferential access

As a network controlled function it shall be possible in LSA to allocate resources at call setup and during the active mode to LSA users compared to non-LSA users.

Annex A: Change history

Change history						
TSG SA#	**Spec**	**Version**	**CR**	**<Phase>**	**New Version**	**Subject/Comment**
Jun 1999	GSM 02.11	7.0.0				Transferred to 3GPP SA1
SA#04	22.011				3.0.0	
SA#05	22.011	3.0.0	001	R99	3.0.1	Editorial update of references for GSM/3GPP use

History

Document history		
V3.0.0	July 1999	Transferred to TSG SA at ETSI SMG#29. Under TSG TSG SA Change Control.
V3.0.1	October 1999	CR approved at SA#05

3GPP TSG-CN-WG1, Meeting #10
11 - 14.January. 2000
Abiko, Japan

Tdoc N1-000202

From: **TSG – CN1**

To: **TSG S3 and TSG N2**

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LS on Enhanced User Identity Confidentiality – open questions

TSG-CN1 received the LS from TSG-S3 and discussed the proposed solution. To finalise the equivalent CR for 3G TS 24.008 N1 identified the following questions and restrictions.
TSG-CN1 is kindly asking for answers and guidelines:

Questions:

Which entity on the subscriber side does perform the encryption process? Is it a SIM or Terminal functionality?

Potential problems:

GSM R99 has a restriction in length (max. 20 octets, no segmentation) on lower layers for transmission of the CM service Request message, therefore introduction of the concept is possible for UTRAN only.

TSG N1 needs a finalised advise about content and coding of the requested information element.

Paging procedures can use IMSI without encryption to search for a mobile, this case is not covered by the current scenarios. This was seen as potential hole in the designed security mechanism.

If the intention is to store the HLR address on SIM / USIM card this makes moving subscribers from one HLR to another more difficult.

MSB / LSB of bit streams should be defined unless this is already specified somewhere or it is otherwise absolutely clear.

The criteria for the ME to use XEMSI or IMSI should be defined, in the current proposal this is not clear.

New concept is being introduced by the proposal and this is against the principle that was agreed in TSGN #6 that we should focus on the completion of the current open issues and not invent new ones.

TSG N1 started to draft the necessary CR for TS24.008 but put it on hold up to clarification of the given questions and problems.

3GPP TSG CN WG1 Meeting
Abiko, Japan
11-14 January 2000

N1-000205

From:	**TSG CN WG1**
To:	**TSG RAN WG3, RAN WG2**
Subject:	**Response Liaison Statement on Partial SRNS Relocation**
Contact Person:	Rouzbeh Farhoumand
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TSG CN WG1#10 reviewed the recieved Liaison Statements on Partial SRNS relocation from RAN WG2, and RAN WG3.

CN WG1 would like to inform RAN3 and RAN2 about the current Working Assumption in CN1 related to release of calls during partial SRNS relocation.

The Working Assumption in CN WG1 is that 3G_MSC keeps all the CC instances during the handover and only after the completion of handover of the selected call(s), it starts disconnecting the surplus CC instances in a controlled manner. The disconnection may occur prior to sending the RELEASE COMMAND to Source RNC by 3G_MSC. (Please see Tdoc N1-99E22)

3GPP TSG-CN-WG1, Meeting #9
30 Oct. - 3 Dec. 1999
Bad Aibling, Germany

Tdoc N1-99E22

Agenda Item: 6.1

WI / Topic: Multicall

Source: Ericsson

Title: Handover and priority settings of Multicall

Effected Specifications / Releases: -

Document for: Discussion

Date: 1999-11-29

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1. Introduction

All the working assumptions for Multicall are collected in tdoc N1-99731. Chapter 5 in that document describes the UMTS to GSM handover and the priorities for selection of the preferred call.

2. Discussion

This contribution identifies some of the problems in execution of multicall handover. It brings up some possible approaches as how to tear down the resources in the UE associated to the RABs that have not been relocated. It also discusses the priority handling of calls in case of handover to both UMTS and GSM.

2.1. Call Clearing

Following are two different approaches for discussion:

1. As it is outlined in the current working assumptions, in both UMTS to GSM, and UMTS to UMTS handover scenarios, the clearing of the corresponding calls to those RABs that target RNC/BSC is not able to support are initiated prior to sending the RELOCATION COMMAND message to the Source RNC.

The problem however is that disconnecting the surplus CC instances (DISCONNECT, RELEASE/ RELEASE COMPLETE for each call leg) may cause some delay during the handover.

After the release, even though the surplus CC instances in the UE have lost their traffical resources, it does not mean that these instances automatically will be removed.

Next, the anchor 3G_MSC will send RELOCATION COMMAND to the Source RNC with indication of the non-supported RABs to be released, see figure 1. Source RNC shall further indicate the release of the corresponding radio bearers to the UE. This in turn triggers the RRC RADIO BEARER RELEASE (one message for all RBs to be released). At reception of RADIO BEARER RELEASE COMPLETE the SRNC sends RRC HO COMMAND to UE.

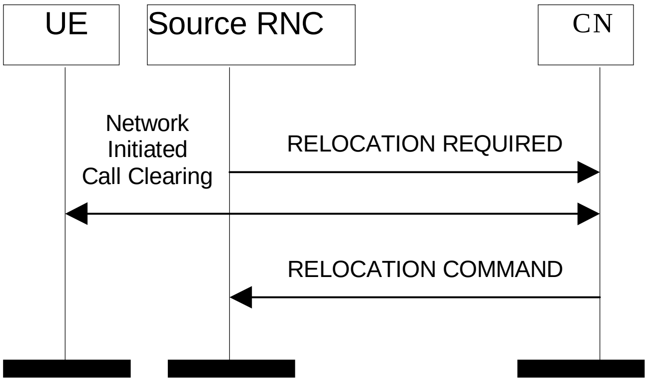

Figure 1

For UMTS to UMTS handover, Iur is used to pass the signaling messages (Direct Transfer). In this case, as the time is too short to execute handover, the signaling should be avoided prior to sending the RELOCATION COMMAND.
When Iur is not used (e.g. inter system handover) there is no satisfactory solution, i.e. the signaling is trashed which means that RELEASE/RELEASE COMPLETE is likely to never reach the UE.

2. 3G_MSC keeps all the CC instances during the handover and only after the completion of handover of the selected call(s), it starts disconnecting the surplus CC instances in a controlled manner (the user plane is gone, but the DTAP signalling is still alive). The disconnection may occur prior to sending the RELEASE COMMAND to Source RNC by 3G_MSC.

2.2. Priority Handling

Selection of those calls that can be relocated/handed over to the target RNC/BSC is specified in TS 22.135 stage 1 for Multicall.

For intra-3G_MSC UMTS to GSM handover (the 3G_MSC supports the A interface as well), the anchor 3G_MSC determines which call can be handed over.

For inter 3G_MSC handover from UMTS to GSM, the anchor 3G_MSC determines which call can be handed over and indicate that to the terminating MSC in the MAP PREPARE (SUB) HANDOVER REQUEST.

For intra-3G_MSC GSM to UMTS handover/relocation, and inter 3G_MSC relocation (3G_MSC-A to 3G_MSC-B), the anchor 3G_MSC shall map the priorities in RELOCATION_REQUEST. The target RNC shall take the priorities into consideration during the partial relocation decision.

3. Proposal

Considering the disadvantages of the first approach, the second approach is recommended to be agreed as the working assumption, and to be included in chapter 5 of N1-99731.

It is proposed also to agree and include chapter 2.2 of this paper in chapter 5 of N1-99731.

3GPP TSG-CN-WG1, Meeting #10
11 - 14.January. 2000
Abiko, Japan

Tdoc N1-000209

Source: **TSGN1**
To: **SMG2**
Cc:
Title: **Liaison statement on CR to 23.122**
Contact Person: Hannu Hietalahti
E-mail: Hannu.Hietalahti@nokia.com
Tel: +358-40-502 1724

N1 thanks SMG2 for their liaison in tdoc SMG2 500/00 / N1-000208. The LS was discussed in TSGN1 #10. Unfortunately we only received the incoming LS on Friday afternoon and thus there was not sufficient time to treat this urgent issue. As the meeting run out of time and the chairman was tasked to writing this liaison statement.

In the brief study of the matter the following observations were were made:

1. No serious objections were raised against the principle but N1 needs more time to study it and will come back to the issue in TSGN1 #11 28.2.-3.3.2000.
2. There is an open issue regarding the PLMN selector lists and N1 has sent a question on this to S1 in tdoc N1-000201 and is expecting an answer which may impact the CR.
3. There is an error in 4.4.3.2 where User controlled PLMN selector list is defined twice.
4. The originator is invited to present the CR in the next TSGN1 meeting #11 28.2.-3.3.2000 as the change impacts specification under N1 responsibility.

3GPP TSG-CN-WG1, Meeting #10
11 - 14.January. 2000
Abiko, Japan

Tdoc N1-000212

Source: **TSGN1**
To: **S1**
Cc:
Title: **Liaison statement on Emergency calls**
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Tel: +358-40-502 1724

TSGN1 has discussed the issue of emergency calls to multiple emergency service numbers.

The discussion did not lead into decision as no agreement could be reached between two proposals that were provided. The principle of each one of these is outlined in attached tdoc N1-000038 and N1-000115.

The meeting run out of time and the chairman was tasked to writing this liaison to ask on the service requirements. Both of the proposed mechanisms have their pros and cons so TSGN1 sees it S1 task to define whether service for roaming subscribers or flexibility to define operator specific emergency numbers should take higher priority.

TSGN1 has studied the stage 3 issues related with TSGN1 specifications and is waiting for S1 guidance to proceed.

The following issues were covered during the discussion:

- An alternative proposal to the one suggested by S1 has been presented in N1 but also this one could not be agreed.
- Both concepts will be provided for S1 to decide which one is more important in this tradeoff, support of roaming or flexibility to define operator specific emergency numbers.
- S1 have made stage 1 requirement to encode the Called Party BCD number in Emergency Setup.
- This requires that in roaming cases the visited MSC needs to know the emergency service numbers of the subscribers HPLMN or otherwise the emergency call must be routed to default emergency service in roaming case.
- Emergency calls may be routed incorrectly if the serving visited MSC recognises the requested emergency number but has mapped that to a different emergency service than the subscribers HPLMN.
- Do the service requirements cover also roaming case or should emergency calls in VPLMN be routed to default emergency center?
- The alternative proposal would standardise the emergency services and thus enable global support of the specified emergency services, if available, also in roaming cases.
- Only the specified emergency categories can be used by the operators in this competing proposal.

TSG-Core Network WG 1 meeting #10 **_Tdoc N1-000038_**
Abiko,Japan 11th – 14th January 2000
Agenda Item: CC related
Source: NEC
Title: Addition of called party BCD number in Emergency SETUP message
Document for: Discussion and Approval

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1. Introduction

TSG-SA#5 approved S1 requirements concerning Emergency Calls.[TSGS#5(99)435] S1 required that it should be possible for the serving network to obtain the number, which was used to initiate the emergency call. Emergency Number would be stored in the SIM/USIM and the ME would read this and use any entry of these digits to set up an emergency call. This will allow the network to include the option to route the call toward different emergency call centers if appropriate.

□The following is mentioned in TS22.101 Ver3.7.0
It shall be possible for the serving network to obtain the number, which was used to initiate the emergency call. This will allow the network the option to route the call to different emergency call centres if appropriate. If the dialled digits are not recognized as an emergency service by the serving network, the call shall be routed to the default emergency service.

2. Discussion

2.1 Mechanism

The method discussed so far transfers the dialled number from the MS to the MSC in S1 meeting. Moreover an alternative mechanism discussed so far is to transfer the actual meaning of the emergency call instead of the dialled number to the MSC in S1#6meeting. However, this requirement was rejected.[S1-991022] Due to all this reason, We proposes to declare the dialled number as an optional IE within EMERGENCY SETUP message.

2.2□Current description on EMERGENCY SETUP message

Called party BCD Number is not defined in current Emergency setup message as shown in Table 9.62. This will not allow the routing of the emergency call toward appropriate emergency center. Therefore, it should be added to Emergency setup message.

Table 9.62/TS 24.008: EMERGENCY SETUP message content

IEI	Information element	Type / Reference	Presence	Format	Length
	Call control Protocol discriminator	Protocol discriminator 10.2	M	V	1/2
	Transaction identifier	Transaction identifier 10.3.2	M	V	1/2
	Emergency setup	Message type	M	V	1

	Message type	10.4			
04	Bearer capability	Bearer capability 10.5.4.5	O	TLV	3-9

2.2 Consideration of Backward compatibility

This contribution proposal is not considered that protocol is dramatically changed from 2nd generation system. Called party BCD number should be defined by optional element. Therefore, it has been considered Emergency setup without called number and Emergency setup with called number. This will be handled by operator. If the emergency number is not recognized by the serving network , the call shall be routed to the default emergency service. S1 requirement will allow the network the option to route the call to different emergency call centers. Therefore, the only change on the protocol is the addition of the Called party BCD number.

3. Proposal

This contribution proposal is addition of Called party BCD Number in subsection 9.3.8 Emergency setup message.

3GPP TSG-CN-WG1, Meeting #10
11 - 14. Jan. 2000
Abiko, Japan

Tdoc N1-001115
(N1-99F12)

Agenda Item: 6.8 CC Related Items

WI / Topic: Emergency Call Handling

Source: Ericsson L.M.

Title: Improved Emergency Call Handling due to the Introduction of Emergency Call Categories

Effected Specifications / Releases: TS 24.008

Document for: Discussion and Approval

Date: 1999-11-22

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1 Background

Emergency calls are handled as a separate teleservice in GSM. The mobile phone maps the emergency numbers keyed in by the subscriber into teleservice 'Emergency Call'. No number is transferred from the Mobile Station (MS) to the MSC. This has the advantage that the subscriber can always dial the numbers known from his home location, but the MSC routes always to the nearest emergency centre.

2 Introduction

The major drawback of this solution is that it cannot be distinguished between different emergency authorities, for example police, ambulance, or fire brigade. Therefore S1 requires in 22.101 that emergency calls shall be routed to different emergency centres. The method discussed so far transfers the dialled number from the MS to the MSC.

However, this method does only work, if the MSC knows the meaning of the transferred emergency number.

An alternative - not discussed so far - is to transfer the actual meaning of the emergency call instead of the dialled number to the MSC. This enables unambiguous routeing to the right emergency centre.

Dialled number method

The different additional emergency numbers are stored on the SIM. If the subscriber initiates an emergency call and the number matches with one of the additional numbers on the SIM, an emergency call set-up is initiated and the dialled number is transferred to the MSC. The MSC has to know the meaning of that specific number in order to route the call to the appropriate emergency centre. Because of the fact that UMTS shall be a worldwide system, but the MSC can only know certain number sets, it is not guaranteed that the service works in every case.

Emergency category method

The different additional emergency numbers are stored at the SIM together with a meaning of the particular number, for instance police, ambulance, fire brigade, or coast guard. If the subscriber initiates an emergency call and the dialled number matches with one of the additional numbers on the SIM, an emergency call set-up is initiated and only the emergency category is transferred to the MSC. The MSC routes the call to the correct emergency centre based on the received category.

3 Discussion

Interpretation and complexity

Dialled Number Method

The subscriber has to interpret the number into a specific emergency category. This leads to the following scenario:

1 Due to international roaming, either the subscriber has to dial the emergency number valid for this PLMN which is normally not known to a roaming subscriber, or

2 the MSC/VLR has to interpret the dialled emergency number and has to map it into the appropriate address valid for this PLMN. That would mean that the MSC would have to know all emergency-numbers valid all over the world or has to map unknown emergency numbers into a default category.

Even when mapping unknown emergency numbers into a default emergency call category, one major bug will remain: If the emergency number for police is '110' in country XXX, but '110' means coast guard in country YYY, then it will be misinterpreted in the MSC when roaming.

For example a subscriber from XXX is roaming to YYY and dials ‘110’ for police, then he will be routed to the coast guard and the whole information transfer from coast guard to police will take too much precious time. There is no way for the MSC to identify the meaning of the dialled number. Note: the meaning is different from one geographical region to another.

It makes it even impossible for an operator running networks everywhere in the world to offer each subscriber the same set of services and the same level of security within his networks all over the world.

Emergency category method

The subscriber dials the emergency number he knows and which is applicable for the situation. The MSC receives only the emergency category and interprets the category - the call is routed to appropriate emergency centre. If a category is not applicable or not set up in the MSC, the MSC routes the call to a default emergency centre.

The MSC has to know only the different categories. It does not have to know the different numbers used in the world. The complexity in the MSC is reduced to a minimum.

The same set of services and security is guaranteed for all UMTS networks, even when roaming.

Signalling Load

Dialled Number method

The current proposals define up-to 19 octets for the length of the information element transferring the dialled number. This means the emergency call set-up message exceeds 20 octets, which leads to increased signalling load. This is a problem, because an emergency call set-up has to be done fast and secure.

Emergency category method

To transfer the emergency category, only one octet must be transferred. This enables up-to 255 different categories and the emergency call set-up message length remains below 20 octets. The signalling load does not increase.

Man-Machine Interface

Dialled number method

The subscriber dials the number for an emergency call set-up. Since the UE does not know about the meaning of the number, the UE cannot support the subscriber to select in an easy way the appropriate emergency number.

Emergency number method

The UE knows the meaning of the numbers and can support the user by a menu, symbols, voice dialling, etc. This can even be offered, before the user enters the PIN, and even without SIM. The level of security is even more increased.

4 Conclusion

With the emergency category method all drawbacks of the dialled number method can be avoided. As in GSM, the emergency call is set-up fast and with the same level of quality. However, the emergency call is improved by specifying different emergency call cases. The mapping to default categories must only be done for emergency cases not defined - for example dialling the coast guard number in Switzerland. Furthermore, the mechanism can be introduced in a backward compatible manner and works with and without SIM-Card.

Due to all this reasons, Ericsson proposes to declare the emergency call category as an optional IE within the EMERGENCY SETUP message as working assumption and to liase this to S1 and T2 accordingly.

3GPP TSG-CN-WG1, Meeting #11
28 Feb – 3 Mar, 2000
Umeå, Sweden

Tdoc N1-000403

Source: TSGN1

To: 3GPP TSG-RAN WG2, 3GPP TSG-RAN WG3

Cc:

Contact Person: Jaakko Rajaniemi
E-mail: jaakko.rajaniemi@nokia.com
Tel: +358 50 3391387

Subject: LS on MS initiated signaling connection release

In some abnormal cases (see N1-000401) in the mobility management layer it is a needed that the MS is able to abort the signaling connection with the CN domain with which the abnormal case occurred.

TSG-CN WG1 like to ask from RAN WG2 and RAN WG3 whether the MS initiated signaling connection release is possible according to their specifications. If not, then TSG-CN WG1 kindly asks the possibility to include this into RAN WG2 and RAN WG3 specifications for Release 99.

Agenda Item:	GSM/UMTS interworking
Title:	MS behavior on T3210 and T3330 expiry
Date	28-February-2000
Source	Nokia
Related To	Mobility Management

1. INTRODUCTION

The purpose of this contribution is to illustrate the abnormal case which may happen when the MS is executing the location or the routing area updating procedure, and for some reason the lower layers are unable to deliver the MM or GMM messages. In this case the current version of the 24.008 [1] does not specify the details of the MS behavior in UMTS, e.g. does the MS release the PS signaling connection at some point.

Additionally, it is shown in this paper that the MS should be able to request the release of the RR connection or the PS signaling connection in some cases. However, it is not clear whether this is possible by the UTRAN protocols. Therefore, it is propose that a LS is sent to RAN groups asking whether these mechanisms are available in the current RAN specifications.

2. MM CONSIDERATIONS

When the mobile station initiates the location updating procedure by sending a LOCATION UPDATING REQUEST message to the network, it starts the timer T3210. It may occur that the network does not respond to the MS and the timer T3210 expires. In this case it is specified in 24.008 that the MS aborts the location updating procedure and it aborts also the RR connection[1].

As the same behavior is defined for the MS performing location updating in UMTS then there should be similar means in the lower layer to abort the RR connection. Therefore, it is proposed that LS is sent to RAN groups asking if they have considered this in their specifications.

3. GMM CONSIDERATIONS

When the MS initiates the normal routing area updating procedure, the MS sends the ROUTING AREA UPDATE REQUEST message to the network and starts timer T3330. As described in the MM case, it may occur that the network does not respond to the MS and the timer T3330 expires. In this case it is specified in 24.008 that the MS aborts only the routing area updating procedure.

The updating of the location in the GMM is different compared to MM because the MS may have PDP contexts active and radio access bearers (RABs) setup when the MS executes the routing area updating and in MM, the location updating procedure is started only if there is no ongoing connections. The release of the PS signaling means that all RABs are also released. Therefore, it may be argued that in this case the MS should release the PS signaling connection only if the routing area updating attempt counter is greater than or equal to 5 in order to save in the signaling load. However, considering the low frequency of this abnormal case and possible fatal error in the signaling connection it is proposed that the MS releases the PS signaling connection every time the timer T3330 expires in the routing area updating procedure.

At the moment for the GPRS attach procedure it is defined that only if the GPRS attach attempt counter is greater than or equal to 5 when the timer T3310 expires then MS shall release the PS signaling connection. If the above principle for the routing area updating is accepted then also the GPRS attach should be updated according to that principle.

Moreover, it may be noted that there should be similar means as in the MM side for the MS to request the release of the PS signaling connection. This also needs to be asked from the RAN groups in the LS proposed above.

[1] In GSM, the mobile station aborts the RR connection by initiating a normal release of the main signaling link by sending L2 DISCONNECT message to the BSS.

4. CONCLUSION

It is proposed that LS to the RAN WG2 and WG3 is sent in which it is indicated that in the abnormal cases described above the MS needs to have a mechanism to request a release of the lower layer connection, i.e. the RR connection or the PS signaling connection.

Also it is proposed that the CR [2] defining the PS signaling connection release is accepted.

5. REFERENCES

[1] TS 24.008 v. 3.2.1

[2] Tdoc N1-000402, " MS behavior on T3210 and T3330 expiry", Source: Nokia

3GPP TSG-CN-WG1, Meeting #11
28 February - 03.March. 2000
Umea, Sweden

Tdoc N1-000443

From: 3GPP N1

To: GSM Association – IREG GPRSWP V. Chair Scott Bailey.
Cc: 3GPP S2, SMG2 WPA

Contact: Mark Fenton, Ericsson
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Phone: +44-1256-864376

Title: Response to LS on GPRS Terminal Support of Network Operation Modes I and II

3GPP N1 thanks the GSM Association for their LS (tdoc N1-000261) on GPRS Terminal Support of Network Operation Modes I and II.

In the incoming liaison, the question asked was "Will all Class C GPRS terminals support both Network Operation Modes II and III or will terminal manufacturers make devices which support only one of the modes?"

The N1 opinion was that it is a mandatory requirement for a GPRS mobile operating in GPRS MS operation mode C to support Network Modes of Operation I,II and III. A number of mobile manufacturers were present at the meeting, including Ericsson, Motorola, Nokia and Siemens. At least in the protocol entities for which N1 is responsible (MM, GMM) for GPRS MS operation mode C there are no differences in the behavior specified between Network Operation Modes I, II and III.

It should be noted that the term "class C" terminal has been superceded by "MS operation mode C". It may be possible for the MS operation mode to change; either as result of configuration by the user, or as a result of change of network mode of operation, or possibly for other reasons not defined.

However, once an MS is configured to operate GPRS MS operation mode C, this should not change as a result of change in network mode of operation.

3GPP TSG-CN-WG1, Meeting #11
28 February - 03.March. 2000
Umea, Sweden

Tdoc N1-000445

Title: LS on reply to Liaison Statement on access signalling and mobile station behaviour for Multicall

Source: **TSG CN WG1**
TO: **TSG CN WG2**

Cc: **TSG CN SS ad hoc, TSGN**

WI: **Multicall**

Contact Person:
Name: Kazuo Mitamura
E-mail Address: mitamura.kazuo@promote.nttcom.co.jp
Tel. Number: +81 43 211 2708

Date: **29/02/2000**

TSG CN WG1 thanks TSG CN WG2 for their LS on access signalling and mobile station behaviour for Multicall.

N2 raised three concerns in N2-000016. N2 concerns and N1 answers are below:

1. We understand that an MS which can support multicall will send a setup message including a stream identifier greater than 1 if the user requests a new call when there is at least one existing active call, but the MS can support another parallel call. If the MSC/VLR is Release 99 but it does not support multicall, it will reject the setup message because it includes a stream identifier greater than 1. This would mean that an emergency call would be rejected by the network. Have N1 considered the interaction between Multicall and handling of emergency calls?

Answer: N1 agreed that the network supporting Multicall shall inform its capability to the MS at the first call. That means the MS can recognize whether the visiting network supports Multicall or not , and ensure the emergency call establishment even if the network does not support Multicall, and there is an ongoing call.

2. We understand that N1 decided that the MS will not use the classmark to indicate to the network its capability for the number of parallel bearers which it can support. Hence, if a mobile terminated call arrives in the VMSC the MSC/VLR has to rely on the subscription information and the generic capabilities of the MSC/VLR to decide whether the new call should be offered as a parallel call or a waiting call. It is therefore necessary to define the error handling for the case where the network offers the incoming call as a new parallel call but the MS cannot accept the incoming call as a new parallel call. The service requirement appears to indicate that the network should offer the incoming call as a waiting call if it cannot be offered as a new parallel call and call waiting is active and operative (this is straightforward if **the network** has determined that the incoming call cannot be offered as a new parallel call).

Answer: N1 reconsidered, and agreed that the MS supporting Multicall shall inform its capability related to Multicall using CC capability IE in call control protocol. That means the network can know the number of parallel bearers which the MS can support, and handle the incoming calls appropriately.

3. We understand that the behaviour of the MS in the call case described in point 2 is to return a Call Confirmed message indicating UDUB. So far as we understand the behaviour of the MS, this

means that the user will be alerted for the incoming call. The behaviour of the network **could** be defined so that if the MS indicates that it cannot accept the incoming call the network will check whether the incoming call can be offered as a waiting call. This would add substantially to the complexity of the call handling in the network and the signalling procedures between the network and the MS. However it is not **certain** that the incoming call will be offered as a waiting call, so there is a possibility that the user will be alerted for the incoming call but the call will not be offered. It appears to N2 that to allow the MS to reject the offer of the incoming call as a parallel call without alerting the user would need a substantial change to the behaviour of the MS. N2 ask N1 to review their decision not to use the MS classmark to indicate to the network its capability for the number of parallel bearers which it can support. This would simplify the call handling in the network to decide whether an incoming call should be offered as a waiting call or a new parallel call, and would ensure that the possibility of undesirable service behaviour (alerting the user for a call which the network decides afterwards not to offer) can be avoided without the need for a major revision to the call handling behaviour of the MS.

Answer: As described above, N1 agreed that the MS supporting Multicall shall inform the number of parallel bearers which it can support. So, N1 think we don't have to take account of the situation that the network does not know the MS capability. In case that the MS can not accept the incoming call offered within the MS capability, it shall initiate call clearing procedure which is specified in TS 24.008. (This means N1 withdraws the approved CR (N1-99C86) related to the MS behaviour in the case that it can not accept the additional incoming call.)

3GPP TSG-CN-WG1, Meeting #11
28 February - 03.March. 2000
Umea, Sweden

Tdoc N1-000446

Source: TSG-CN WG1

To: TSG-RAN WG2

Cc: TSG-RAN WG3

Title: **Question about Idle-mode DRX control**

Contact: Fumihiko YOKOTA, Fujitsu Limited
+81 44 754 4196, yokota@ss.ts.fujitsu.co.jp

N1 received liaison statement from R2 about DRX parameter [R2-00276]. N1 would like to ask R2 for the clarification about how “k”, which is used to calculate the “DRX cycle length” is provided from the CN to the RAN. N1 would like to know if the value of “k” is only derived from broadcasting information as described in liaison, or other mechanism is also expected. In GSM, DRX related information is indicated from SGSN to BSC in paging and other messages of BSSGP. If R2 expects to receive “k” from CN node for each MS as in GSM GPRS, N1 needs to update “DRX parameter” IE in GMM message to adopt to UTRAN.

To complete the work in R99, N1 needs to receive the answer in N1#11 (from 28 Feb to 3 March). N1 expects very quick answer to this question.

[quotation from R2-00276 begin]
For the parameter “k” three different cases apply:

- for NAS paging “k” is CN specific and provided by the CN (“CN domain specific DRX cycle length coefficient” in System information,
- for AS paging “k” is either Cell specific and provided by the DRNC,
- or “k” is individually assigned to a UE and known by the SRNC.

[quotation end]

3GPP TSG-CN-WG1, Meeting #11
28 February – 3 March 2000
Umeå, Sweden

Tdoc N1-000447

Source: **CN1**

Contact Person: Robert Zaus
E-mail: robert.zaus@icn.siemens.de
Tel.: +49 170 3315485

To: **RAN2, RAN3, CN2B**

Title: **2nd LS on the Transport of Codec Information during the Codec Negotiation between MS and MSC**

Since the CN1#10 meeting in January, CN1 continued the work on the downlink transport of codec information during codec negotiation between MS and MSC.

CN1 would like to inform RAN2, RAN3 and CN2B about the progress which has been achieved (see attached Tdoc N1-000517), and kindly asks these groups to take the necessary actions to implement the procedures agreed by CN1 in the respective specifications under their responsibility.

During the further refinement of the working assumptions which were communicated to RAN2, RAN3 and CN2 (Tdoc N1-000164) in January, it was necessary to modify the concept. Our main guideline for this was to achieve a better separation between CC and RR related signalling, and to reduce the amount of codec information which has to be transported by RANAP and RRC messages other than Direct Transfer messages as far as possible. We assume that this is also in the interest of RAN2 and RAN3.

CN1 kindly asks RAN2 and RAN3 for a rapid response to be able to complete the work on codec negotiation during this week.

3GPP TSG-N WG1 **Tdoc N1-000517**
Umeå, Sweden
28 February – 3 March 2000

Source: **Ericsson, NTTDoCoMo, Siemens AG**

Agenda Item: **Out-of-Band Transcoder Control**
Title: **Concept proposal for Transport of Codec Information**

Purpose: **For Decision**

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1. Introduction

At the last CN1 meeting #10 in Abiko, CN1 agreed on a set of working assumptions for the transport of codec information during the codec negotiation between MS and MSC. As was indicated already at that meeting, the working assumptions were not complete, as they did not cover the case when RANAP instead of BSSMAP was used at the MAP E-interface for the purpose of inter 3G MSC SRNS relocation.

Meanwhile SA2 decided to select RANAP as the only protocol to be used for this purpose, and new discussions between Ericsson, NTT DoCoMo and Siemens took place which resulted in the following modified concept. The main principles followed during the development of the codec negotiation procedure were
- to achieve a clearer separation between CC and RR related signalling by reducing the amount of codec information which has to be transported by RANAP and RRC messages other than Direct Transfer messages to a minimum,
- to reduce the interruption of the speech signal during a handover or bearer re-configuration involving a codec change, as far as possible,
- to maintain the codec negotiation functionality also in the case when TDM links are used between anchor and target MSC,
- and to avoid additional MAP dialogue steps during the time critical MSC-MSC handover when TDM links are used between anchor and target MSC.

2. Concept for the Transport of Codec Information

The concept is characterized by the following features:

i) a separate CC downlink message Selected Codec which is always sent by the anchor MSC via a RANAP Direct Transfer message

ii) an information element "Codec Activation Indicator" (CAI) which is added to the message Selected Codec,

iii) an information element "NAS Synchronization Indicator" (NSI) which is added to the RANAP messages RAB Assignment Request, Relocation Request, and to the related RRC messages Radio Bearer Setup, Radio Bearer Reconfiguration and Physical Channel Reconfiguration, These information elements are used for the purpose to inform the MS about the timing of the codec activation or codec change, and to allow the MS to link the Selected Codec message to a specific RRC message.

iv) In case of MSC-MSC handover with TDM between the anchor and the target MSC:
for the handover itself: codec selection is embedded in the MAP Prepare Handover operation (to avoid an additional MAP dialogue which might jeopardize the handover)

v) for bearer reconfiguration after the handover: a separate MAP operation for codec selection is used (one MAP dialogue with request and response; the actual selection is done by the target MSC)

The details of the concept will be described in the following sections.

2.1 Linkage between the indication of the selected codec and the radio bearer change

All parties involved in the discussion have agreed that at least in case of a handover involving a codec change, the MS should apply a new selected codec only after the related bearer change has taken place.

This can be achieved by the following procedure (fig. 1):

1) An information element 'Codec activation indicator' (CAI) is included in the Selected Codec message. It is defined as a mandatory IE of type 1 with a length of 1 octet and the following two codepoints,:

   0 = immediate activation,
   1 = delayed activation (MS activates the codec after receipt of an RRC message containing a NAS Synchronization Indicator)

2) A new parameter 'NAS Synchronization Indicator' (NSI) is added to the RANAP message Relocation Command and to the related RRC messages, e.g. Physical Channel Reconfiguration.
   It is defined as 1 bit with the following codepoints:
   0 = NSI not present,
   1 = NSI present

   The MSC shall set the NSI in the RANAP message Relocation Command, if it has commanded the MS to assign a codec or to change the codec during the handover by sending a CC message Selected Codec to the MS containing a CAI indicating 'delayed activation'.

   If the information element NSI is set in the RANAP message, it has to be included by the RNC in the corresponding RRC message, e.g. Physical Channel Reconfiguration.

   If the MS receives a Selected Codec message containing a CAI indicating 'delayed activation', it shall activate the new codec only after receipt of the next RRC message containing the NSI.

   The procedure in the MS is as follows: If the RRC layer in the MS receives such an RRC message containing a NSI, this is indicated to the CC layer. Then the CC layer has to activate the codecs for all Selected Codec messages which were received with a CAI indicating ‘delayed activation’.

3) If an SRNS relocation is cancelled after the Selected Codec message has been sent, the MSC has to send a second Selected Codec message to the MS to cancel the effect of the first one. The second Selected Codec message has to be sent with a CAI indicating ‘immediate activation’. (This has to be done before any new Relocation or RAB Assignment procedure may be initiated by the MSC.)

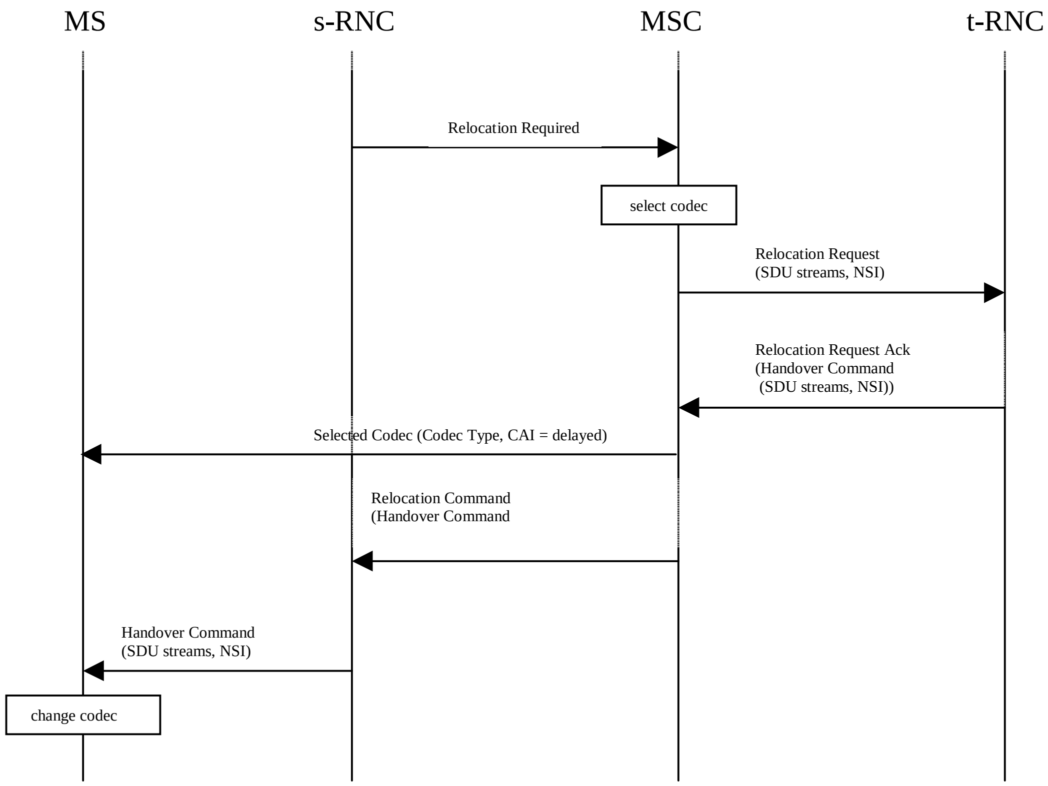

Figure 1: RNC relocation with codec change

Note: in these message flows the names of RRC message like Handover Command should be considered as ‘functional’ names; they need not be identical with what has been specified in TS 25.331.

For the first bearer assignment of a new call both an 'immediate activation' and a 'delayed activation' are possible. However, for subsequent (re-)assignments involving a codec change, a delayed activation may have advantages (in terms of loss of less speech frames).

(Note: The 'immediate activation' may be applicable in future also in case of a codec change without bearer change.)

Therefore, we propose the following procedure (fig. 2 and 3).

4) The NAS Synchronization Indicator is added also to the RANAP message RAB Assignment Request and to the RRC messages Radio Bearer Setup, Radio Bearer Reconfiguration.

   The MSC shall set the NSI in the RANAP message RAB Assignment Request, if it has commanded the MS to assign a codec or to change the codec during the radio bearer setup or reconfiguration by sending a CC message Selected Codec to the MS containing a CAI indicating 'delayed activation'.

   If the NSI is set in the RANAP message, it has to be included by the RNC in the corresponding RRC message Radio Bearer Setup or Radio Bearer Reconfiguration.

   If the MS receives a Selected Codec message containing a CAI indicating 'delayed activation', it shall activate the new codec only after receipt of the next RRC message containing the NSI and affecting the corresponding bearer.

   The procedure in the MS is as follows: If the RRC layer in the MS receives such an RRC message containing a NSI, this is indicated to the CC layer together with the Stream Identifier(s) of the bearer(s) affected by the RRC procedure. Then the CC layer has to activate the codecs for all affected bearers for which Selected Codec messages were received with a CAI indicating 'delayed activation'.

5) If a radio bearer reconfiguration fails in such a way that the RNS and MS revert to the old bearer configuration and the call can be continued afterwards, the MSC has to send a second Selected Codec message to the MS to cancel the effect of the first one. The second Selected Codec message has to be sent with a CAI indicating 'immediate activation'.

   Question to RAN3: Is such a scenario, which in GSM is indicated to the MSC by an GSM 08.08-Cause "Radio interface failure, reversion to old channel", also possible in UMTS?

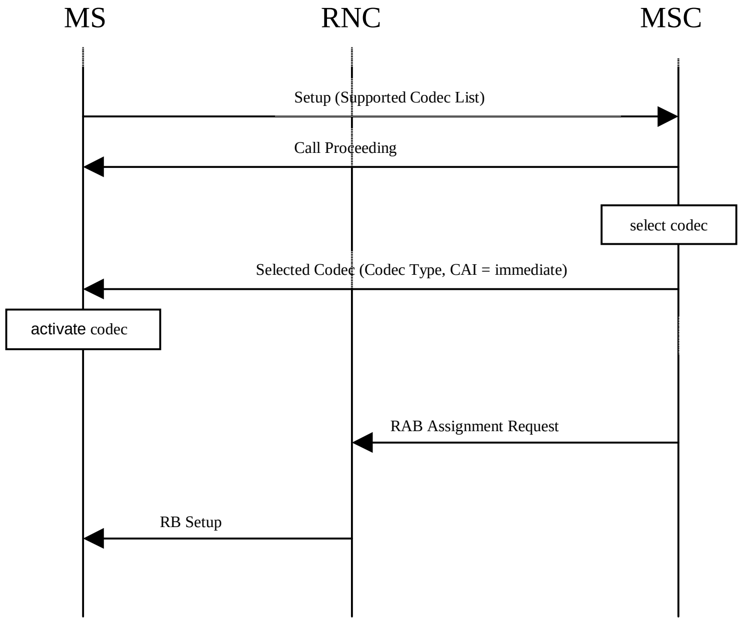

Figure 2: Radio bearer setup with immediate codec activation

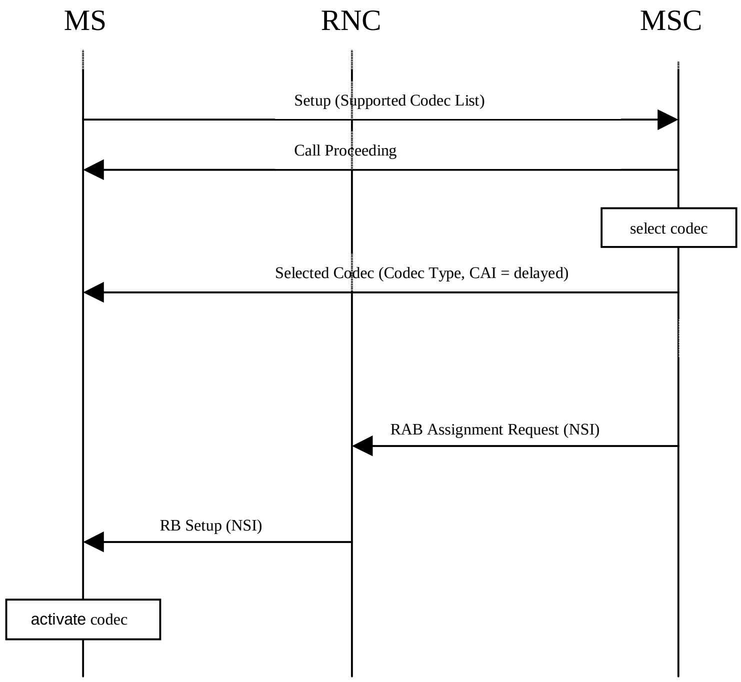

Figure 3: Radio bearer setup with delayed codec activation

2.2 Control of the transcoder after MSC-MSC Handover

The second major issue under discussion was the control of the transcoder after an MSC-MSC handover, if TDM is used on the trunks between the anchor MSC and the target MSC. In such a case, the transcoder will be located in the target MSC. Call control, however, is still located in the anchor MSC.

We think that also for TDM trunks a procedure to control the transcoder via the E interface should be provided, because even if there is only one UMTS speech codec standardized in R99, this may change quite soon. E.g. with the introduction of tandem free operation (TFO) in R00, it may become interesting to be able to support also codecs like GSM full rate or GSM enhanced full rate for 2G-3G mobile-to-mobile calls.

Secondly, the final selection of the codec type has to be performed by the target MSC which is provided by the anchor MSC with a list of the supported codecs. The availability of a certain transcoder type in the target MSC is a dynamical function of time, as the target MSC has to co-ordinate concurrent requests for codecs originating not only from one, but from several anchor MSCs and from the target MSC itself. Under these conditions, a simple and reliable procedure with no more than two MAP dialogue steps (request and response) is only possible, if the target MSC is given the freedom to select a codec from the list of supported codecs.

(Note: In most cases it should be possible to avoid a codec change during handover by adopting the following procedure: during the MAP procedure, the target MSC will be provided with the Selected Codec (Serving), i.e. the codec currently in use, and a RANAP message formatted accordingly. Only if the target MSC cannot support the Selected Codec (Serving), but can support another codec from the Supported Codec List, it selects a new one. It then opens RANAP and re-configures SDU formats accordingly. Otherwise the Selected Codec (serving) is used and RANAP is not changed.)

Thirdly, we have tried to avoid to introduce another MAP dialogue in the time-critical MSC-MSC handover procedure. As a consequence of this, if the codec type has to be changed during the handover, some of the parameters of the RANAP message Relocation Command (QoS, SDU format) will have to be provided by the target MSC (see fig. 4). We think that this is an acceptable price, because the same functionality, to derive the RANAP parameters from the selected codec type, will be needed anyway for the cases of 2G MSC ->3G MSC handover (fig. 5) and radio bearer reconfiguration after such a handover (fig. 7). (In these cases BSSMAP will be used via the E interface, and the Channel Type will provide only the information that this is a speech call.)

For the radio bearer reconfiguration after MSC-MSC we propose a different solution, involving a separate MAP operation for codec selection. With this solution we will need 4 MAP dialogue steps, but otherwise the anchor MSC will not able to inform the MS about the new selected codec in time, before the radio bearer is changed (fig. 6). Besides, the reconfiguration procedure is less time-critical than an MSC-MSC handover (in contrast to handover there is no immediate danger of loss of the radio connection), and with the proposed enhancement it will take about the same time as such a handover.

3. Proposal for decision

We suggest that N1 adopts the procedures outlined above for the finalization of the work on codec negotiation between the MS and the MSC.

Furthermore, N1 should send a liaison statement to RAN2 and RAN3, who are meeting in parallel to N1, and to N2B, to inform them about the new situation and to enable them to implement the necessary changes to their specifications.

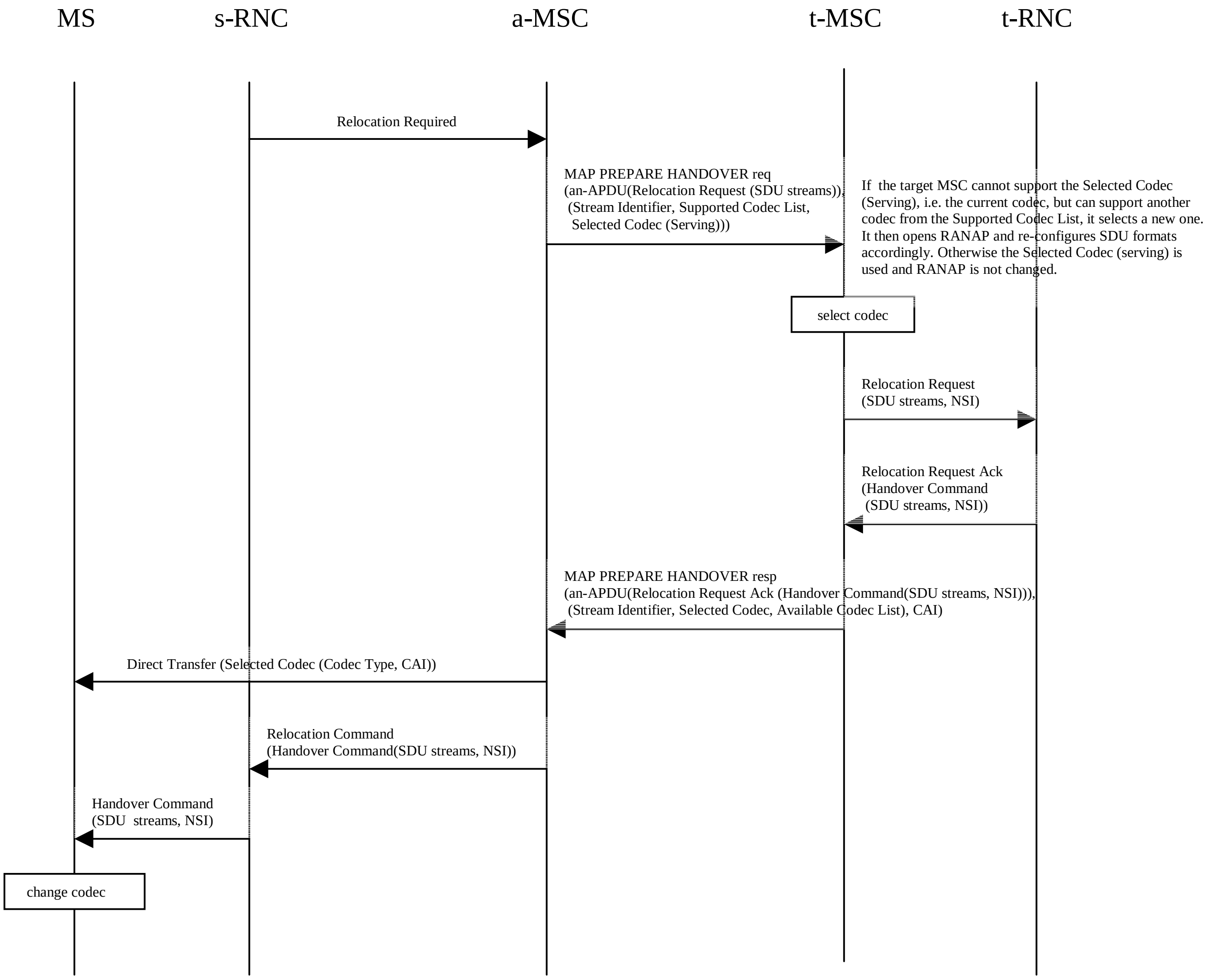

Figure 4: inter 3G MSC handover / SRNC relocation with codec change
(with TDM link between anchor and target MSC)

Note: in case of a multicall handover, the set of parameters (Stream Identifier, Supported Codec List, Selected Codec (Serving)) may have to be included more than once in the MAP Prepare Handover request. In this case each set of parameters can be assigned to a certain radio bearer by means of the Stream Identifier. The CAI and NSI are included only once per message.

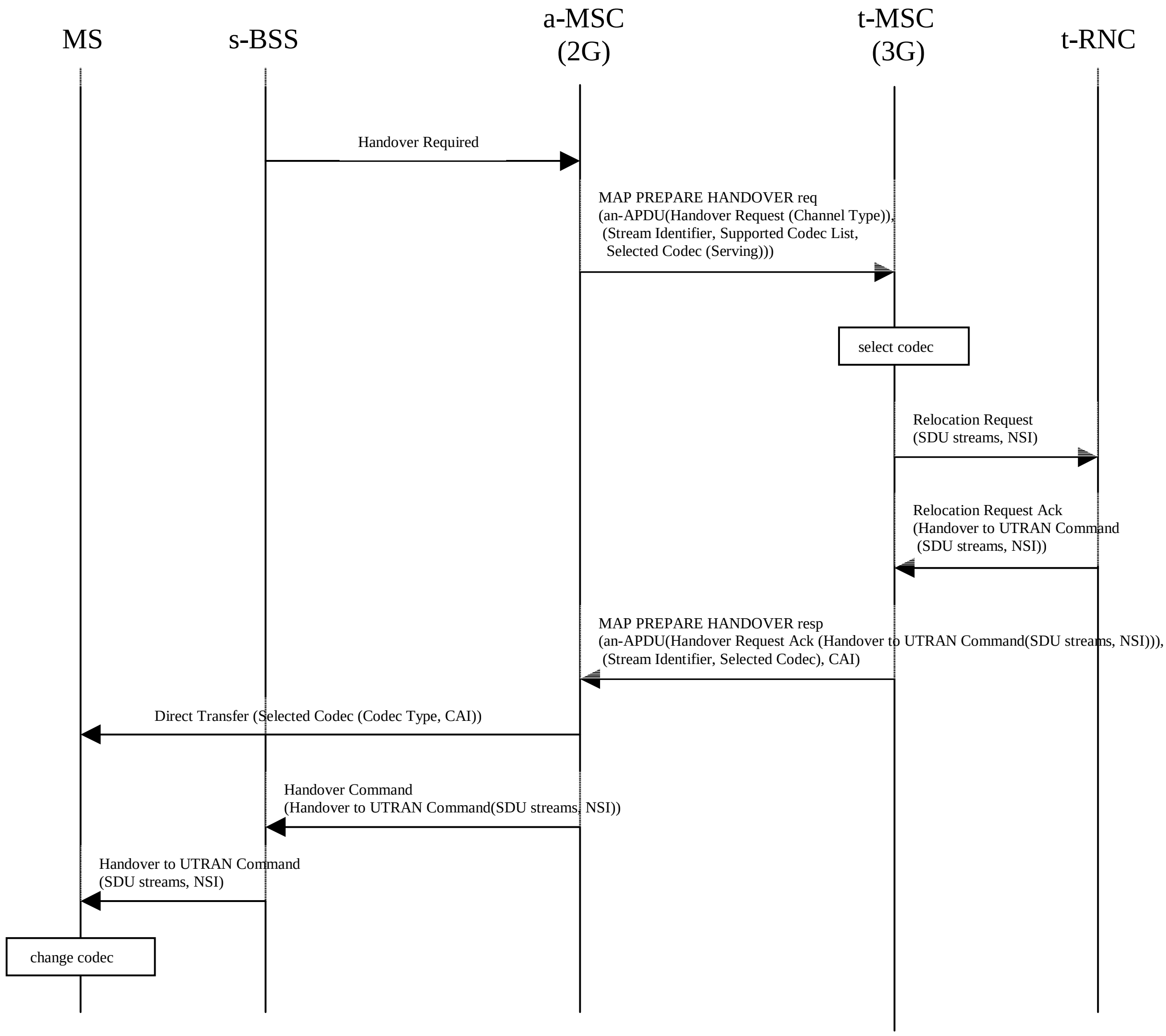

Figure 5: 2G -> 3G MSC handover with codec change
(with TDM link between anchor and target MSC; usage of BSSMAP via the E interface)

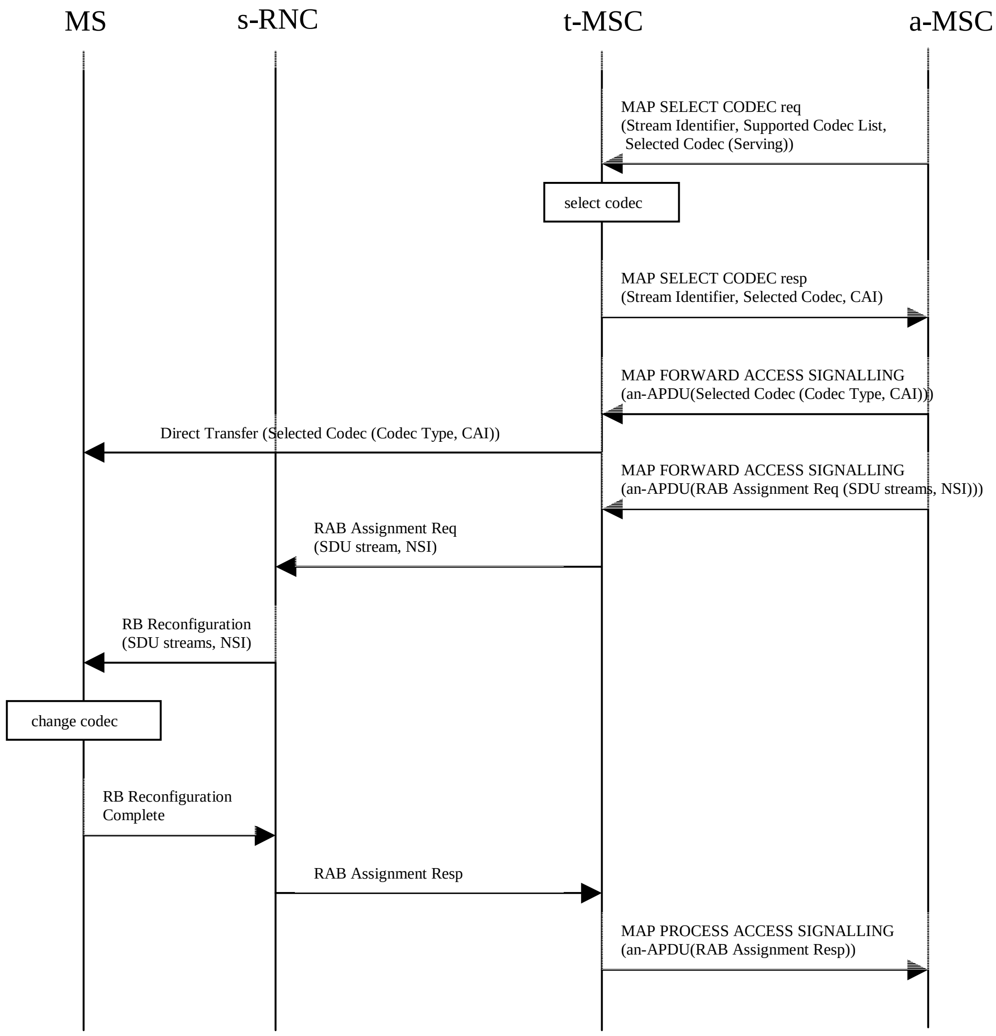

Figure 6: Radio bearer reconfiguration with codec change after 3G MSC handover / SRNC relocation (with TDM link between anchor and target MSC)

Note: in case of reconfiguration of a multicall, the set of parameters (Stream Identifier, Supported Codec List, Selected Codec (Serving)) may have to be included more than once in the MAP Select Codec request. In this case each set of parameters can be assigned to a certain radio bearer by means of the Stream Identifier. The CAI and NSI are included only once per message.

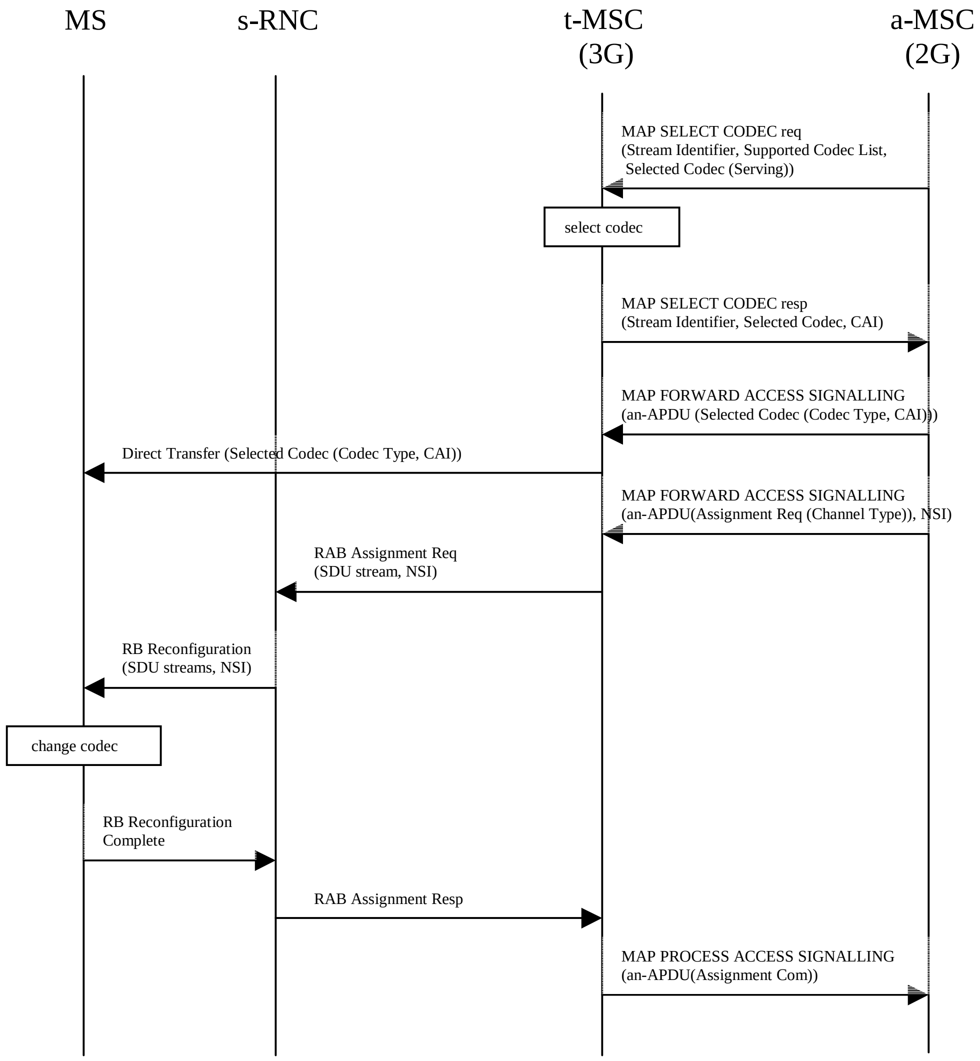

Figure 7: Radio bearer reconfiguration with codec change after 2G->3G MSC handover (with TDM link between anchor and target MSC; usage of BSSMAP via the E interface)

3GPP TSG-CN-WG1, Meeting #11
28 February - 03.March. 2000
Umea, Sweden

Tdoc N1-000449

Title: **USIM triggered authentication and key setting during PS connections**

Source: **3GPP TSG CN WG1**

TO: **3GPP TSG SA WG3**

Cc: **3GPP TSG RAN WG2, 3GPP TSG T WG3**

WI: **Security**

Contact Person:

Name: **Duncan MILLS**
E-mail Address: **duncan.mills@vf.vodafone.co.uk**
Tel. Number: **+44 1635 676074**

Date: **02-03-2000**

CN1 thanks SA3 for their Liaison Statement on the above subject. The group discussed this subject, and concluded the following:

- In CN1's opinion, the need for this feature is not great, especially in comparison with other pressing security matters.
- The work required to support such a feature needs careful consideration and has a relatively large impact on the specifications under the group's control.
- If SA3 remain of the opinion that this feature should be implemented, then CN1 would be happy to consider it as a work item for Release 2000.

In summary, CN1 predict a considerable amount of work to implement the feature and do not feel they can do this work for Release 99. Doing so may have negative affects on more important issues.

3GPP TSG-CN-WG1, Meeting #11
28 February - 03.March. 2000
Umea, Sweden

Tdoc N1-000450

From: N1

To: SMG9, T3, T2 SWG1 (MExE)
Cc:

Contact: Mark Fenton, Ericsson
Email: mailto:mark.fenton@eml.ericsson.se
Phone: +44-1256-864376

Title: Response to LS on 5 or 6 digits IMSI HPLMN

3GPP N1 thanks the T2 SWG1 (MExE) for their LS (tdoc N1-000289) on 5 or 6 digits IMSI HPLMN.

N1 would like to support T2 SWG1 (MExE) view that maintaining a list of MCCs in the MS along with the number of digits for that MCC is not a very satisfactory solution.

Adding a new field on the USIM to indicate how many digits of the IMSI must be used to extract the PLMN seems like a good solution from a MExE perspective. This solution could also be useful to MS manufacturers as it may be used also for SIM Lock purposes. However N1 would like to add the following comments.

(1) To be useful, this new field must be added to R99 GSM SIMs as well as USIMs. GSM-UMTS interworking is a fundamental requirement for R99.
(2) The ME behavior should ideally be defined for the case where this new field is **not** present on the SIM/USIM, ie. when a pre-R99 SIM is inserted. However it should be noted that at least up to R98 depending on the PLMN SIMs will contain either 5 digits or 6 digits for the IMSI HPLMN coding without any information .
(3) It would have been really useful for N1 if this field had been present on all SIMs prior to the R98 work item PCS1900 harmonisation. However as this was not the case an alternative solution had to be found to solve the signaling protocol specific problems.

In summary N1 would like to support the proposal as it is likely that it may become useful in the future when the majority of SIMs/USIMs have the new field. However until this becomes true it is unlikely that N1 will be able to use the new field for N1 protocols.

3GPP TSG-N WG1 **N1-000451**
Umea, Sweden
28 Feb – 03 Mar 2000

Source:	**N1**
To:	**N2B**
CC:	**RAN3, S4**
Agenda Item:	**LS out (OoBTC)**

Title: LS on TrFO Break procedure (N1-000264 & N1-000367)

Contact Person: Phil Hodges (phil.hodges@eed.ericsson.se)

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1. Introduction

This LS addresses LS's from N2B (N2-000012 & N2B-000387) regarding TrFO Break procedure. It is believed that the proposed solution has impacts on N1 specifications that should be considered.

The architecture presented shows the Iu interface being extended from RNC to RNC and not simply from RNC to MSC. This has impacts to the call control procedures performed by the MSC, especially with respect to the through connection of the bearer as described in N2B-000387. This would result in some impacts in TS 24.008.

The proposed solution introduces new control procedures from the MSC to RNC at certain service invocations. Although these procedures are proposed as RANAP protocol impacts, the call control entity will have impacts on how and when these procedures are executed and their interaction with CC protocol.

The proposed TrFO break procedure could result in loss of speech frames as bearers may need to be reconfigured & a new initialisation procedure is required. Extensive discussions have been performed within N1 to develop call control procedures for TrFO operation to minimise breaks in the speech connection due to handover and codec change. It is desired that our solutions should not be compromised by proposals made in N2B.

2. Conclusion

N1 believes these impacts have not been considered or coordinated with work ongoing in N1. Further it is believed that other solutions exist or are being developed (ITU framing protocols) to provide a secure TrFO that will support the call control situations where problems arise with the proposed use of Iu UP protocol within the network.

N1 is kindly asking N2B to give more considerations to the proposed architecture in N2-000012 in cooperation with the other affected working groups.

3GPP TSG-CN-WG1, Meeting #11
28 February - 03.March. 2000
Umea, Sweden

Tdoc N1-000452

Title: **Response on Liaison statement concerning the change of title of 23.060**

Source: **3GPP TSG CN WG1**

TO: **3GPP TSG SA WG2**

Cc: **3GPP TSG SA WG1**

Contact Person:

Name: **Lars Ekeroth, Ericsson Mobile Data Design AB**
E-mail: **lars.ekeroth@erv.ericsson.se**
Tel.: **+46 31 703 6566**

Date: **2000-03-02**

N1 wants to thank S2 for the Liaison statement concerning the change of title of 23.060 (Tdoc S2-000314).

N1 has in earlier meetings discussed and agreed on the terminology to be used within N1. The attached CR on TS 24.008 (N1-99D81) shows the terminology agreed. Changing the title of TS 23.060 is not in line with this.

N1 propose S2 not to change the title of TS 23.060 and instead consider adapting TS 23.060 to use the same terminology as agreed by N1.

3GPP/SMG Meeting TSG-CN WG1
Bad Aibling, Germany, 30.11.-3.12.1999

Document ***N1-99D81***
e.g. for 3GPP use the format TP-99xxx
or for SMG, use the format P-99-xxx

CHANGE REQUEST

Please see embedded help file at the bottom of this page for instructions on how to fill in this form correctly.

24.008	**CR**	**069**	Current Version:	3.1.0

GSM (AA.BB) or 3G (AA.BBB) specification number ↑ *↑ CR number as allocated by MCC support team*

For submission to:	CN#9	For approval	X	strategic		(for SMG use only)
list expected approval meeting # here ↑		for information		non-strategic		

Form: CR cover sheet, version 2 for 3GPP and SMG The latest version of this form is available from: ftp://ftp.3gpp.org/Information/CR-Form-v2.doc

Proposed change affects: *(at least one should be marked with an X)*	(U)SIM		ME	X	UTRAN / Radio		Core Network	X

Source:	Ericsson, Siemens, Nokia	**Date:**	99-11-25
Subject:	24.008 Vocabulary		
Work item:	GSM-UMTS Interworking		

Category:	F	Correction	X	**Release:**	Phase 2	
	A	Corresponds to a correction in an earlier release			Release 96	
(only one category	B	Addition of feature			Release 97	
shall be marked	C	Functional modification of feature			Release 98	
with an X)	D	Editorial modification			Release 99	X
					Release 00	

Reason for change:

TS 24.008 is based on GSM TS 04.08 and consequently the terminology has been defined for GSM and GPRS environment.

For UMTS the common Core Network specification will need to cover the requirements of GSM, UMTS and GSM-UMTS multimode.

If agreed the originators of this CR are volunteering to provide the CRs that are necessary to take these definitions to 24.008 CC, MM, SM and GMM sections.

Clauses affected: Vocabulary

Other specs affected:	Other 3G core specifications		→ List of CRs:	
	Other GSM core specifications		→ List of CRs:	
	MS test specifications		→ List of CRs:	
	BSS test specifications		→ List of CRs:	
	O&M specifications		→ List of CRs:	

Other comments:

help.doc

<--------- double-click here for help and instructions on how to create a CR.

2.2.2 Vocabulary

The following terms are used in this Technical Specification:

- **idle mode:** In this mode, the mobile station is not allocated any dedicated channel; it listens to the CCCH and the BCCH;

- **group receive mode:** (only applicable for mobile stations supporting VGCS listening or VBS listening) In this mode, the mobile station is not allocated a dedicated channel with the network; it listens to the downlink of a voice broadcast channel or voice group call channel allocated to the cell. Occasionally, the mobile station has to listen to the BCCH of the serving cell as defined in TS 23.022 and 05.08;

- **dedicated mode:** In this mode, the mobile station is allocated at least two dedicated channels, only one of them being a SACCH;

- **group transmit mode:** (only applicable for mobile stations supporting VGCS talking) In this mode, one mobile station of a voice group call is allocated two dedicated channels, one of them being a SACCH. These channels can be allocated to one mobile station at a time but to different mobile stations during the voice group call;

- **packet idle mode**: (only applicable for mobile stations supporting GPRS) In this mode, mobile station is not allocated any radio resource on a packet data physical channel; it listens to the PBCCH and PCCCH or, if those are not provided by the network, to the BCCH and the CCCH, see GSM 04.60.

- **packet transfer mode**: (only applicable for mobile stations supporting GPRS) In this mode, the mobile station is allocated radio resource on one or more packet data physical channels for the transfer of LLC PDUs.

- **main DCCH:** In Dedicated mode and group transmit mode, only two channels are used as DCCH, one being a SACCH, the other being a SDCCH or a FACCH; the SDCCH or FACCH is called here "the main DCCH";

- A channel is **activated** if it can be used for transmission, in particular for signalling, at least with UI frames. On the SACCH, whenever activated, it must be ensured that a contiguous stream of layer 2 frames is sent;

- A TCH is **connected** if circuit mode user data can be transferred. A TCH cannot be connected if it is not activated. A TCH which is activated but not connected is used only for signalling, i.e. as a DCCH;

- The data link of SAPI 0 on the main DCCH is called the **main signalling link**. Any message specified to be sent on the main signalling link is sent in acknowledged mode except when otherwise specified;

- The term **"to establish"** a link is a short form for **"to establish the multiframe mode"** on that data link. It is possible to send UI frames on a data link even if it is not established as soon as the corresponding channel is activated. Except when otherwise indicated, a data link layer establishment is done without an information field.

- **“channel set”** is used to identify TCHs that carry related user information flows, e.g., in a multislot configuration used to support circuit switched connection(s), which therefore need to

- be handled together.

- A **temporary block flow** (TBF) is a physical connection used by the two RR peer entities to support the uni-directional transfer of LLC PDUs on packet data physical channels, see GSM 04.60.

- **RLC/MAC block:** A RLC/MAC block is the protocol data unit exchanged between RLC/MAC entities, see GSM 04.60.

- A **GMM context** is established when a GPRS attach procedure is successfully completed.

-- **Network operation mode**

The three different network operation modes I, II, and III are defined in TS 23.060 [74].

The network operation mode shall be indicated as system information. For proper operation, the network operation mode should be the same in each cell of one routing area.

-- **GPRS MS operation mode**

The three different GPRS MS operation modes A, B, and C are defined in TS 23.060 [74].

- **Anonymous access** refers to limited service provisioning to an MS whose identity is unknown in the network.

- **RR connection:** A RR connection is a dedicated physical circuit switched domain connection used by the two RR or RRC peer entities to support the upper layers' exchange of information flows.

- **PS signalling connection** is a peer to peer UMTS connection between MS and CN packet domain node.

- **Inter-System change** is a change of radio access between GSM and UMTS.

- **GPRS:** Packet Services for GSM and UMTS system.

- The label **(GSM only)** indicates this section or paragraph applies only to GSM system. For multi system case this is determined by the current serving radio access network.

- The label **(UMTS only)** indicates this section or paragraph applies only to UMTS system. For multi system case this is determined by the current serving radio access network.

- **In GSM,...** Indicates this paragraph applies only to GSM System. For multi system case this is determined by the current serving radio access network.

- **In UMTS,...** Indicates this paragraph applies only to UMTS System. For multi system case this is determined by the current serving radio access network.

- **SIM,** Subscriber Identity Module (see TS GSM 02.17). This specification makes no distinction between SIM and USIM.

- **MS,** Mobile Station. This specification makes no distinction between MS and UE.

3GPP TSG-CN-WG1, Meeting #11
28 February – 3 March 2000
Umeå, Sweden

N1-000453

Source: **CN1**

To: **TSG SA2**

Cc: **TSG - SA4, CN2, RAN2, RAN3**

Title: **LS on N1 Working Status of the working plan on OoBTC in R99**

Contact Person: **Takashi KOSHIMIZU**
E-mail: koshimi@nw.yrp.nttdocomo.co.jp

CN1 has received LS from S2 titled "Liaison statement on the working plan to complete OoBTC in R99 " (S2000285 = N1000414). S2 has responsibility to coordinate on this WI and asking each WG's the final working status. N1's tasks recognized in this working item is the procedure of down link codec notification. This LS reply N1' s final working status about this for R99.

In N1#11, it is agreed no codec negotiation mechanism is necessary in R99, because UMTS-AMR is the only applicable codec type in R99-UMTS system after intensive discussion and study.

N1 would like to reply to SA2 that we have completed requested task and this is N1's final working status on OBTC in R99. Additionally, N1 recognizes the study for comprehensive codec negotiation procedure when another codec types are introduced in later release.

3GPP TSG-CN-WG1, Meeting #11
28 February - 03.March. 2000
Umea, Sweden

Tdoc N1-000460

Source: TSG-CN WG1

To: TSG-RAN WG2

Cc: TSG-RAN WG3

Title: Question about Idle-mode DRX control

Contact: Fumihiko YOKOTA, Fujitsu Limited
+81 44 754 4196, yokota@ss.ts.fujitsu.co.jp

N1 received liaison statement from R2 about DRX parameter [R2-00276]. N1 would like to ask R2 for the clarification about how “k”, which is used to calculate the “DRX cycle length” is provided from the CN to the RAN. N1 would like to know if the value of “k” is only derived from broadcasting information as described in liaison, or other mechanism is also expected. In GSM, DRX related information is indicated from SGSN to BSC in paging and other messages of BSSGP. If R2 expects to receive “k” from CN node for each MS as in GSM GPRS, N1 needs to update “DRX parameter” IE in GMM message to adapt to UTRAN.

To complete the work in R99, N1 is working on a CR in this area. In order to proceed with this question N1 needs to receive the answer in N1#11 (from 28 Feb to 3 March).

[quotation from R2-00276 begin]
For the parameter “k” three different cases apply:

- for NAS paging “k” is CN specific and provided by the CN (“CN domain specific DRX cycle length coefficient” in System information,
- for AS paging “k” is either Cell specific and provided by the DRNC,
- or “k” is individually assigned to a UE and known by the SRNC.

[quotation end]

3GPP TSG CN WG1 #11 **Tdoc N1-000481**
Umea, Sweden, 28th Feb. – 3rd Mar. 2000

Source: **TSG CN WG1**

Title: **Liaison statement on the status of Multicall**

To: **TSG CN, TSG CN WG2, TSG CN SSAdoc**

Cc: **TSG SA WG1**

Date: **3rd March, 2000**

Contact: Naoki Tani (NTT DoCoMo, tani@nw.yrp.nttdocomo.co.jp)

CN WG1 discussed the remaining open issues identified by the Multicall Adhoc (17-18, Feb. 2000) at its #11 meeting (28th Feb. – 3rd Mar. 2000). Finally CN WG1 concluded that all remaining open issues related to CN WG1 were solved. The attachment indicates the brief description of the agreed documents.

3GPP TSG-CN-WG1, Meeting #11
28 February – 03 March 2000
Umea, Sweden

Tdoc N1-000481 attachment

This table below was made based on NS-000039, which identified the open issues as an output of Multicall adhoc held in 17-18 Feb. 2000. Also, the result of N1 #11 meeting was included.

No	Issue title	Detail description	Solution	Remaining Issue	Involved WG (Mtg schedule)	TS impacted	Result of N1 #11
1	Emergency Calls	Revise to guarantee set-up by releasing resources if needed	Release a bearer or all bearers (network option) for the emergency call if no bearers are available	To be included into 24.008	N1 (28-02-2000) Multi Call Adhoc (18-02-2000)	22.135 (done) 23.135 (done) 24.008	There is no impact on CC protocol (24.008).

No	Issue title	Detail description	Solution	Remaining Issue	Involved WG (Mtg schedule)	TS impacted	Result of N1 #11
2			MS must ensure the acceptance for the emergency calls by the serving network (MS has to use stream identifier 1)	24.008?	N1 (28-02-2000) Multi Call Adhoc (18-02-2000)	22.135 (done) 23.135 (done) 24.008	Since MS shall be aware of NW capabilities, NW indicates its Multicall capability to the MS by using CC message. CR to 24.008 (N1-000560) The following description shall be added to 24.008: When MS is located in the NW not supporting Multical, it shall not request multiple bearers. CR 24.008 (N1-000489)

No	Issue title	Detail description	Solution	Remaining Issue	Involved WG (Mtg schedule)	TS impacted	Result of N1 #11
3	Transfer of MS Capability -> Network	Working assumption: in CC capability message	To be included into CC capability message. Number of bearers and number of speech bearers to be included	To be included into 24.008	N1 (28-02-2000) Multi Call Adhoc (18-02-2000)	23.135 (done) 24.008 24.135 (done)	MS capability (number of bearers and number of speech bearers) is included in *Call Control* IE. CR to 24.008 (N1-000490)
4	Simultaneous Call Set-Up	Working Assumption: Allow it in the network, error handling in MS to be defined	Network will offer the call to the MS. The MS must be able to reject that call.	To be included into 24.008 and 24.135	N1 (28-02-2000) Multi Call Adhoc (18-02-2000)	22.135 (done) 23.135 (done) 24.008	The call clearing procedures are already included in 24.008. No N1 action is needed.
5	Call Wait	Handling to be reviewed with reference to simultaneous call set-up	Network can offer the call as a waiting call. The MS must be able to reject that call.	To be included into 24.008 and 24.135	N1 (28-02-2000) Multi Call Adhoc (18-02-2000)	22.135 (done) 23.135 (done) 24.008	The call clearing procedures are already included in 24.008. No N1 action is needed.

No	Issue title	Detail description	Solution	Remaining Issue	Involved WG (Mtg schedule)	TS impacted	Result of N1 #11
6	Handover	Inter MSC Handover/multicall scenarios to be resolved	Possibility A1 has been expressed (NS-000033) Possibility B4 – Broadcast Channel should include the R'99 Multi Call scenario.	To be included into 23.135, 29.002, 24.008?	N1 (28-02-2000) N2 (02-03-2000) Multi Call Adhoc (18-02-2000)	23.135 24.008 29.002	N1's working assumption is based on possibility B2. CR on HO scenario 23.009 based on possibility B2 (N1-000491) was agreed.
7	Maximum number of calls		Number of calls is been kept in the stage 1 document. If N1 can not cope this requirement then it will be kept in release 99 (limit of call is currently set to 7 – 14)	To be added 24.007 and 24.008, 23.135, 24.135	N1 (28-02-2000) N2 (02-03-2000) SS-Adhoc (25-02-2000)	24.007 24.008 23.135 24.135	N1 discussed the requirements and agreed that TI isn't expanded in R99. Therefore no CRs to 24.007 & 24.008 are necessary.
8	Relation of stream identifiers with TI (maximum bearers versus calls)		Stream Identifier allocation should be referenced from 24.008	To be included in 24.135	SS-Adhoc (25-02-2000)	24.135 24.008	N1 discussed the requirements and agreed that TI isn't expanded in R99. Therefore no CRs to 24.007 & 24.008 are necessary.

No	Issue title	Detail description	Solution	Remaining Issue	Involved WG (Mtg schedule)	TS impacted	Result of N1 #11
9	Improvements to TS 24.135		Error Causes needs to be described (check stage 2)	To be included in 24.135 Alignment check is required to 23.135		24.135 23.135	No action is needed in N1
10	Handling of the Stream identifier by pre-release 99 MSCs and non supporting Release 99 MSCs needs to be specified		Broadcast solution raised in the meeting. No agreements yet.	To be included into 22.135, 23.135	S1 (25-02-2000) N2B (2-3-2000)	22.135 23.135	Since MS shall be aware of NW capabilities, NW indicates its Multicall capability to the MS by using CC message. CR to 24.008 (N1-000560) The following description shall be added to 24.008: When MS is located in the NW not supporting Multical, it shall not request multiple bearers. CR 24.008 (N1-000489)

No	Issue title	Detail description	Solution	Remaining Issue	Involved WG (Mtg schedule)	TS impacted	Result of N1 #11
11	Including proper definition for NDUB	Proper definition was requested.		To be included into 22.135, 23.135	S1 (25-02-2000) N2B (2-3-2000)	22.135 23.135	Since TI is not expanded in R99, no action is needed in N1.
12	Timing of TCH release		NW behaviour shall be clarified.	To be included into 23.108	N1	23.108	CR to 23.108 (N1-000492) was agreed.

3GPP TSG-CN-WG1, Meeting #11
28 February - 03.March. 2000 Umea, Sweden

N1-000487

Source: **N1**
To: **RAN2, RAN3**

Agenda Item: **LS out (OBTC)**

Title: LS on RANAP Transaction Sequence

Contact Person: Phil Hodges (phil.hodges@eed.ericsson.se)

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1. Introduction

Since the submission of LS N1-000164 to RAN2 & RAN3 regarding the support of a NAS-PDU message to carry selected codec information, further discussions have been made in N1 to minimise the impacts to RANAP. We made the assumption that the CC message Selected Codec should be sent as a Direct Transfer message. As a result of these discussions a problem with current working assumptions in the RAN was discovered.

It is understood by N1 that there is a working assumption in RAN WG2 & RAN WG3 that Direct Transfer Messages can be handled by different Radio Link Control entities and with unequal priority. Further it is understood by N1 that if a Relocation Command is received, the RNC may suspend transmission of Direct Transfer messages and initiate the change of Radio Bearer. This would lead to a change in the order of the RANAP protocol messages sent from the CN and in intersystem handover or inter MSC handover to a loss of Direct Transfer messages.

Some specific cases where this behaviour causes problems are:

i) at Network initiated clearing where the CC protocol assumes that the Release Complete message sent in Direct Transfer is received prior to any Radio Bearer change (or release);

ii) at Codec change where the Selected Codec message is sent to the UE in a Direct Transfer message which is to be activated on receipt of the associated Radio Bearer message. If the Radio Bearer message arrived before the Direct Transfer message then no codec change would be performed.

This working assumption is in conflict with the protocol principles which applied in GSM, and on which also the call control protocol for UMTS is based.

2. Conclusion

N1 requests that it is stated in the RANAP protocol that RANAP transactions are completed by the RNC in the sequence that they were received. SCCP is used on Iu to guarantee the protocol sequence integrity from the CN's call control. This should be maintained by the RNC to UE also.

This clarification would mean that prior to performing an RRC release or handover, all Direct Transfer messages received should have been successfully transmitted to the UE.

It should be noted that the CN will suspend Direct Transfer messages during Radio Bearer procedures, so it is not required that the RNC buffer separately further DTAP messages while ensuring those prior to the Radio Bearer procedure are executed.

N1 requests for a rapid response from RAN2 and 3 to be able to complete the work on codec negotiation during this week.

3GPP TSG-CN-WG1, Meeting #11
28 Feb – 3 Mar, 2000
Umeå, Sweden

Tdoc N1-000493

Source: TSG N1
To: 3GPP TSG-RAN WG2, 3GPP TSG-RAN WG3
Cc:
Contact Person: Janne Muhonen
E-mail: janne.m.muhonen@nokia.com
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E-mail: jaakko.rajaniemi@nokia.com
Tel: +358 50 3391387

Subject: LS on MS initiated signaling connection release

In some abnormal cases in the mobility management layer there is a need that the MS is able to abort the signaling connection with the CN domain with which the abnormal case occurred. These abnormal cases may happen when the MS is executing the location or the routing area updating procedure, and for some reason the lower layers are unable to deliver the MM or GMM messages.

Additionally, the MS should be able to request the release of the RR connection[1] or the PS signalling connection in some cases. However, it is not clear whether this is possible by the UTRAN protocols.

In CS domain, when the mobile station initiates the location updating procedure by sending a LOCATION UPDATING REQUEST message to the network, it starts a timer. It may occur that the network does not respond to the MS and the timer expires. In this case it is specified in 24.008 that the MS aborts the location updating procedure and it aborts also the RR connection.

As the same behavior is defined for the MS performing location updating in UMTS then there should be similar means in the lower layer to abort the RR connection.

In PS domain, when the MS initiates the normal routing area updating procedure, the MS sends the ROUTING AREA UPDATE REQUEST message to the network and starts a timer. As described in the CS domain case, it may occur that the network does not respond to the MS and the timer expires. In this case it is specified in 24.008 that the MS aborts only the routing area updating procedure. Additionally, the MS should also release the PS signalling connection, but the exact mechanism hasn't been yet decided by TSG-CN WG1.

Anyway, there should be similar means as in CS domain for the MS to request the release of the PS signaling connection from the lower layer.

---

[1] In 24.008, following terms have been defined:

- **RR connection:** A RR connection is a dedicated physical circuit switched domain connection used by the two RR or RRC peer entities to support the upper layers' exchange of information flows.

- **PS signalling connection** is a peer to peer UMTS connection between MS and CN packet domain node.

3GPP TSG-CN-WG1, Meeting #11
28 Feb – 3 Mar, 2000
Umeå, Sweden

Tdoc N1-000493

TSG-CN WG1 like to ask from RAN WG2 and RAN WG3 whether the MS initiated RR connection and PS signaling connection release is possible according to their specifications. If not, then TSG-CN WG1 kindly asks the possibility to include this into RAN WG2 and RAN WG3 specifications for Release 99.

TSG-CN Working Group 1 meeting #11 **_N1-000511_**
Umea, Sweden, 28 February – 3 March 2000

Source: **TSGN1**

Title: LS to CN WG2B proposing a new Specification “Application Part (RANAP) on the E-interface”; 29.108

Date: **2000-02-29**

Source: **CN WG1**

To: **CN WG2B**

CC: **RAN WG3**

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TSG CN WG1 would like to inform TSG CN WG2B of the need for a new specification to describe RANAP for the E-interface used at intra UMTS inter-MSC (3G-3G) Relocation. The first draft version of this new TS is attached (N1-000425) and TSG CN WG2B is hereby invited to take on the responsibility of this new TS.

The proposed new TS is based on the GSM 09.08 V.7.2.0. TS 23.009 under the responsibility of TSG CN WG1 should refer to both GSM 09.08 (BSSMAP on E interface), as well as to the new proposed TS (RANAP on E interface).

TSG-CN Working Group 1 meeting #11 *N1-00043*
Umea, Sweden, 28 Feb – 2 Mar, 2000

Agenda Item:
Source: Ericsson
Title: New TS 29.108, Application Part (RANAP) on the E-interface
Document for: Discussion

Introduction

This new TS is based on the GSM 09.08 V.7.2.0. This TS describes RANAP for the E-interface used at intra UMTS inter-MSC (3G-3G) Relocation.
This document is presented for information to N1. It is also related to the input LS Liaison Statement on "the use of RANAP for intra-UMTS inter-MSC Handover/Relocation" (tdoc N1-000266). TS 23.009 under the responsibility of N1 should refer to both GSM 09.08 (BSSMAP on E-if) and to this new proposed TS (RANAP on E-if). This new TS is recommended to be under the responsibility of TSG CN working group 2B.

3G TS 29.108 V0.0.0 (2000-01)

Technical Specification

3rd Generation Partnership Project;
Technical Specification Group Core Network;
Application Part (RANAP) on the E-interface
(3G TS 29.108 version 0.0.0 Release 99)

Reference

DTS/TSGN-0229002U

Keywords

3GPP, CN

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Contents

Foreword

This Technical Specification has been produced by the 3GPP.

This TS describes the subset of Radio Access Network Application Part (RANAP) messages and procedures within the 3rd generation mobile telecommunications system.

The contents of the present document are subject to continuing work within the TSG and may change following formal TSG approval. Should the TSG modify the contents of this TS, it will be re-released by the TSG with an identifying change of release date and an increase in version number as follows:

Version 3.y.z

where:

- x the first digit:
  - 1 presented to TSG for information;
  - 2 presented to TSG for approval;
  - 3 indicates TSG approved document under change control.
- y the second digit is incremented for all changes of substance, i.e. technical enhancements, corrections, updates, etc.
- z the third digit is incremented when editorial only changes have been incorporated in the specification.

1 Scope

The present document describes the subset of Radio Access Network Application Part (RANAP) messages and procedures defined in 3G TS 25.413 [4]. A general description can be found in 3G TS 23.002 [7] and 3G TS 23.009 [2].

For the initiation and execution of relocation between MSCs a subset of RANAP procedures are used. For the subsequent control of resources allocated to the Mobile Station (MS) RANAP procedures are used. NAS signalling messages defined in 3G TS 24.008 [8] are used for the transfer of connection management and mobility management messages between the MS and the controlling MSC.

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies.

[1] 3G TS 21.905: "3G Vocabulary".

[2] 3G TS 23.009: "Handover procedures"

[3] 3G TS 25.412: "UTRAN Iu Interface Signalling Transport".

[4] 3G TS 25.413: " UTRAN Iu Interface RANAP Signalling"

[5] 3G TS 29.002: " Mobile Application Part (MAP) specification"

[6] 3G TS 29.010: "Signalling procedures and the Mobile Application Part (MAP)"

[7] 3G TS 23.002: "Network architecture"

[8] 3G TS 24.008: "Mobile radio interface layer 3 specification, Core Network Protocols - Stage 3"

3 Abbreviations

For the purpose of this specification, the following definitions and abbreviations apply:

MS	Mobile Station
MSC	A third generation Mobile services Switching Centre that supports the Iu interface (and possibly also the A-interface)
MSC-A	The controlling MSC on which the call was originally established
MSC-B	The MSC to which the MS is handed over in a Basic Handover
MSC-B'	The MSC to which the MS is handed over in a Subsequent Handover
MSC-I	Interworking MSC
MSC-T	Target MSC
RNC	Radio Network Controller

Other abbreviations used in this specification are listed in 3G TS 21.905 [1].

4 Principles for the use of RANAP on the E-interface

4.1 General

The mechanisms for the transfer of the RANAP messages on the E-interface is defined in 3G TS 29.002 [5]. The operation of the relocation procedures between MSCs and the use of the RANAP messages for those procedures is described in 3G TS 23.009 [2] and 3G TS 29.010 [6].

In the same way as a SCCP signalling connection is used for the messages relating to one MS on the MSC-RNC interface a TCAP dialogue is used on the E-interface for messages relating to one MS. As no correspondence to the connectionless service on the MSC-RNC interface is used on the E-interface none of the connectionless procedures defined in 3G TS 25.413 [4] are applicable.

The management of the terrestrial circuits between the MSCs is outside the scope of the E-interface (see 3G TS 23.009 [2]), therefore all procedures, messages and information elements relating to terrestrial circuits are also excluded from the RANAP procedures and messages used on the E-interface.

4.2 Transfer of NAS and RANAP layer 3 messages on the E-interface

The RANAP data which on the MSC-RNC interface is contained in the user data field of the exchanged SCCP frames (see 3G TS 25.412 [3]) is on the E-interface transferred as the contents of the signalling info in a `AN-APDU` parameter as described in 3G TS 29.002 [5].

The RANAP data consists of a RANAP layer 3 message which may include NAS signalling message.

4.3 Roles of MSC-A, MSC-I and MSC-T

For the description in this 3G TS, the MSC's functionality related to the relocation between MSCs has been split into three logical parts, MSC-A, MSC-T and MSC-I. The different roles need not necessarily be performed by different MSCs.

MSC-A is the call/connection controlling part of the MSC where the call/connection was originally established and the switching point for relocation between MSCs. (This corresponds to MSC-A as defined in 3G TS 23.009 [2] and 3G TS 29.002 [5]). The MSC that is the MSC-A will not be changed during the duration of a call/connection.

MSC-T is the part relating to the transitory state during the relocation for the MSC to which the MS is handed over when Basic relocation or Subsequent relocation (see 3G TS 23.009 [2]) take place. (This corresponds, depending on the type of relocation to MSC-A, MSC-B or MSC-B' in 3G TS 23.009 [2] and 3G TS 29.002 [5]).

MSC-I is the part of an MSC through which the MSC-A, via an E-interface (or an internal interface) is in contact with the MS. (This corresponds, depending on the type of relocation to MSC-A, MSC-B or MSC-B' in 3G TS 23.009 [2] and 3G TS 29.002 [5]).

The MSC that is the MSC-A can also have the role of either the MSC-I or the MSC-T during a period of the call/connection.

The following is applicable for the involved MSCs concerning the exchange of RANAP data on an E-interface before and after a successful inter MSC Relocation:

1) At Basic relocation, two MSCs are involved, one MSC being MSC-A and one being MSC-T. When this relocation has been performed, the two MSCs interworking on the E-interface have the roles of MSC-A and MSC-I respectively, i.e. the MSC that is the MSC-T during the relocation is now the MSC-I.

2) At Subsequent relocation back to MSC-A, two MSCs are involved. The MSC having the role of MSC-A has also the role of MSC-T. The other MSC involved has the role of MSC-I. When this relocation has been completed, there is no exchange of RANAP data on the E-interface, i.e. the MSC being the MSC-I before and during the relocation is now no longer taking part.

3) At subsequent relocation to an MSC not being MSC-A, three MSCs are involved. The roles of these MSCs are MSC-A, MSC-I, and MSC-T respectively. When this relocation has been performed, the two MSCs interworking on an E-interface have the roles of MSC-A and MSC-I respectively, i.e. the MSC that is the MSC-T during the relocation is now the MSC-I and the MSC being MSC-I during the relocation is now no longer taking part.

5 Use of the RANAP on the E-interface

NAS signalling is used on the E-interface for the transfer of messages between the MSC-A and the MS.

The connection oriented RANAP procedures (3G TS 25.413 [4] chapter 6) used on the E-interface to some extent are:

- RAB Assignment;
- RAB Release Request;
- Iu Release Request;
- Iu Release;
- Relocation Preparation;
- Relocation Resource Allocation;
- Relocation Detect;
- Relocation Complete;
- Relocation Cancel;
- Common ID;
- CN Invoke Trace;
- Security Mode Control;
- Location Reporting Control;
- Location Report;
- Initial UE Message;
- Direct Transfer;
- Error Indication.

5.1 NAS Signalling

For the exchange of the NAS signalling messages (3G TS 25.413 [4] chapters 8.22 and 8.23), the involved MSCs shall act according to the following:

- the MSC-A acts as the MSC;
- the MSC-I acts as the RNC.

5.2 RAB Assignment

The RAB Assignment procedure (3G TS 25.413 [4] chapter 8.2) is applied on the E-interface with following conditions:

- the MSC-A acts as the MSC;
- the MSC-I acts as the RNC.

The handling of terrestrial resources is not applicable.

5.3 Relocation Resource Allocation

At Basic Inter-MSC Relocation (3G TS 23.009 [2]) the Relocation Resource Allocation procedure (3G TS 25.413 [4] chapter 8.7) is applied on the E-interface with the following conditions:

- the MSC-A acts as the MSC;
- the MSC-T acts as the target RNC.

At Subsequent Inter-MSC Relocation (3G TS 23.009 [2]) the Relocation Resource Allocation procedure (3G TS 25.413 [4] chapter 8.7) is applied on the E-interface with the following conditions:

- the MSC-I acts as the MSC;
- the MSC-T acts as the RNC;
- if the MSC that is the MSC-A is not also the MSC-T, then this MSC shall act as the target RNC towards the MSC-I and as the MSC towards the MSC-T.

The handling of terrestrial resources is not applicable.

5.4 Relocation execution

For the Relocation execution procedures (3G TS 25.413 [4] chapters 8.7, 8.8 and 8.9) the applicable parts on the E-interface are the transfer of RELOCATION DETECT, RELOCATION COMPLETE and RELOCATION FAILURE messages at inter MSC Relocation. For those parts, the involved MSCs shall act according to the following:

- the MSC that is the MSC-A, acts as the MSC;
- the MSC that is the MSC-I, if it is not also the MSC-A, acts as the serving RNC;
- the MSC that is the MSC-T, if it is not also the MSC-A, acts as the target RNC.

5.5 Release due to RNC generated reasons

For the Release due to RNC generated reasons (3G TS 25.413 [4] chapters 8.3 and 8.4) the involved MSCs shall act according to the following:

- the MSC-I acts as the RNC;
- further actions are taken by the MSC-A.

5.6 Release from CN

For the Release from CN (3G TS 25.413 [4] chapters 8.3 for RAB Release and 8.5 for Iu Release) the involved MSCs shall act according to the following:

- the MSC-A acts as the MSC;
- the MSC-I acts as the RNC.

The handling of terrestrial resources is not applicable.

5.7 Security Mode Control

For the Security Mode Control procedure (3G TS 25.413 [4] chapter 8.18), the involved MSCs shall act according to the following:

- the MSC-A acts as the MSC;
- the MSC-I acts as the RNC.

5.8 CN Invoke Trace

For the CN Invoke Trace procedure (3G TS 25.413 [4] chapter 8.17), the involved MSCs shall act according to the following:

- the MSC-A acts as the MSC;
- the MSC-I acts as the RNC.

5.9 Error Indication

For the Error indication (3G TS 25.413 [4] chapter 8.27), the involved MSCs shall act according to the following:

- the MSC-A acts as the MSC;
- the MSC-I acts as the RNC.

5.10 Common Id

For the Common Id procedure (3G TS 25.413 [4] chapter 8.16), the involved MSCs shall act according to the following:

- the MSC-A acts as the MSC;
- the MSC-I acts as the RNC.

5.11 Location Reporting Control

For the Location Reporting Control procedure (3G TS 25.413 [4] chapter 8.19), the involved MSCs shall act according to the following:

- the MSC-A acts as the MSC;
- the MSC-I acts as the RNC.

5.12 Location Report

For the Location Report procedure (3G TS 25.413 [4] chapter 8.20), the involved MSCs shall act according to the following:

- the MSC-A acts as the MSC;
- the MSC-I acts as the RNC.

6 RANAP messages transferred on the E-interface

The following RANAP messages, defined in 3G TS 25.413 [4] chapter 9.1, are transferred on the E-interface:

	RAB ASSIGNMENT REQUEST	(MSC-A -> MSC-I)
	RAB ASSIGNMENT RESPONSE	(MSC-I -> MSC-A)
	RAB RELEAE REQUEST	(MSC-I -> MSC-A)
*	RELOCATION REQUEST	(MSC-A -> MSC-T and MSC-I -> MSC-A)

*	RELOCATION REQUEST ACKNOWLEDGE	(MSC-T -> MSC-A and MSC-A -> MSC-I)
*	RELOCATION COMPLETE	(MSC-T -> MSC-A)
*	RELOCATION FAILURE	(MSC-T -> MSC-A and MSC-I -> MSC-A)
*	RELOCATION DETECT	(MSC-T -> MSC-A)
	IU RELEASE	(MSC-A -> MSC-I)
	IU RELEASE REQUEST	(MSC-I -> MSC-A)
	ERROR INDICATION	(MSC-A -> MSC-I and MSC-I -> MSC-A)
	COMMON ID	(MSC-A -> MSC-I)
	DIRECT TRANSFER	(MSC-A -> MSC-I)
	CN INVOKE TRACE	(MSC-A -> MSC-I)
	SECURITY MODE COMMAND	(MSC-A -> MSC-I)
	SECURITY MODE COMPLETE	(MSC-I -> MSC-A)
	SECURITY MODE REJECT	(MSC-I -> MSC-A)
	LOCATION REPORTING CONTROL	(MSC-I->MSC-A, MSC-A -> MSC-I)
	LOCATION RESPORT	(MSC-I -> MSC-A, MSC-A->MSC-I)

All other RANAP messages shall be considered as non-existent on the E-interface.

Some of the messages above are qualified by *. This signifies whether the message, when sent on the E-interface, is considered as:

- Relocation related message (*).

7 Exceptions for RANAP message contents and information element coding when transferred on the E-interface

This section is FFS.

8 RANAP message error handling when transferred on the E-interface

The RANAP error handling (3G TS 25.413 [4] chapter 10) is applicable. The handling of faults concerning the use of SCCP is not applicable.

The RANAP error messages sent on the E-interface shall only be sent as response to RANAP messages received on the same interface.

Annex A: Change history

Change history						
TSG CN#	**Spec**	**Version**	**CR**	**Phase**	**New Version**	**Subject/Comment**
Feb 2000	GSM 09.08	7.2.0		R98		Transferred to 3GPP CN
CN2B#04	29.108			R99	0.0.0	

History

Document history		
V0.0.0	February 2000	Draft of new TS

3GPP TSG-CN-WG1, Meeting #11
28 February - 03.March. 2000
Umea, Sweden

Tdoc N1-000515

From: 3GPP N1

To: 3GPP N2
Cc: 3GPP S2

Contact: Mark Fenton, Ericsson
Email: mailto:mark.fenton@eml.ericsson.se
Phone: +44-1256-864376

Title: LS on Length of QoS IE

3GPP N1 has agreed the attached CR regarding the QoS IE. This done in order to optimize the bits used on the radio interface.

N1 is of the understanding that this may impact the GTP and MAP protocols as the QoS IE carried in these protocols may need updating in order to match the 24.008 IE. In particular N1 believes the QoS length will need updating in GTP and MAP.

N1 asks N2 to make the required changes to GTP and MAP in time for R99.

3GPP/SMG Meeting TSG-CN-WG1
Umea, Sweden, 28 Feb – 03 Mar 2000

Document N1-000559
revision of N1-000514

CHANGE REQUEST

Please see embedded help file at the bottom of this page for instructions on how to fill in this form correctly.

24.008	**CR**	**114 r3**	Current Version: 3.2.1

GSM (AA.BB) or 3G (AA.BBB) specification number ↑ *↑ CR number as allocated by MCC support team*

For submission to:	CN#7	for approval	X	strategic	*(for SMG use only)*
list expected approval meeting # here ↑		For information		non-strategic X	

Proposed change affects: (U)SIM [ ] ME [X] UTRAN / Radio [ ] Core Network [X]
(at least one should be marked with an X)

Source: Ericsson **Date:** 2000-03-03

Subject: Compact coding of QoS IE

Work item: QoS

Category: *(only one category shall be marked with an X)*

			Release:	
F	Correction		Phase 2	
A	Corresponds to a correction in an earlier release		Release 96	
B	Addition of feature		Release 97	
C	Functional modification of feature	X	Release 98	
D	Editorial modification		Release 99	X
			Release 00	

Reason for change:

With CR 086r1 the new Quality of Service (QoS) information element (IE) for Release 99 was introduced in TS 24.008.
One disadvantage with the proposed coding is that the size of the QoS IE has increased considerably, from 4 bytes in R'97 to 19 in R'99 (LV coding).
This IE is used in several L3 protocol messages. Some of which were quite large already in R'97, thereby increasing the likelihood of an overflow in the messages.
A closer look shows that the IE coding has been made more spacious when necessary.
Considering that the QoS IE is also a major part of e.g. the MODIFY PDP CONTEXT REQUEST message and there are reasons to believe that QoS renegotiation will contribute significantly to the SM signalling in a GPRS network, it seems justified that a more compact coding should be strived for.
With this CR a compact coding for the QoS IE is introduced that will bring down the IE length to 13 octets (12 octets where the IE is mandatory).
QoS IE Length has also been added, were missing, in the tables in section 9.5.

Clauses affected:

9.5 GPRS Session Management Messages
10.5.6.5 Quality of Service.

Other specs affected:

Other 3G core specifications		→ List of CRs:	
Other GSM core specifications		→ List of CRs:	
MS test specifications		→ List of CRs:	
BSS test specifications		→ List of CRs:	
O&M specifications		→ List of CRs:	

Other comments: Rev 0 of this CR was based on version 3.2.0 of 24.008. This revision is based on 3.2.1

help.doc

<--------- double-click here for help and instructions on how to create a CR.

2 Normative references

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies.
- A non-specific reference to an ETS shall also be taken to refer to later versions published as an EN with the same number.

[1] GSM 01.02: "Digital cellular telecommunications system (Phase 2+); General description of a GSM Public Land Mobile Network (PLMN)".

[2] GSM 01.04: "Digital cellular telecommunications system (Phase 2+); Abbreviations and acronyms".

[2a] 3G Vocabulary

[3] TS 22.002: "Digital cellular telecommunications system (Phase 2+); Bearer Services (BS) supported by a GSM Public Land Mobile Network (PLMN)".

[4] TS 22.003: "Teleservices supported by a GSM Public Land Mobile Network (PLMN)".

[5] GSM 02.09: "Digital cellular telecommunications system (Phase 2+); Security aspects".

[6] TS 22.011: "Digital cellular telecommunications system (Phase 2+); Service accessibility".

[7] GSM 02.17: "Digital cellular telecommunications system (Phase 2+); Subscriber identity modules Functional characteristics".

[8] GSM 02.40: "Digital cellular telecommunications system (Phase 2+); Procedures for call progress indications".

[9] GSM 03.01: "Digital cellular telecommunications system (Phase 2+); Network functions".

[10] TS 23.003: "Digital cellular telecommunications system (Phase 2+); Numbering, addressing and identification".

[11] GSM 03.13: "Digital cellular telecommunications system (Phase 2+); Discontinuous Reception (DRX) in the GSM system".

[12] TS 23.014: "Digital cellular telecommunications system (Phase 2+); Support of Dual Tone Multi-Frequency signalling (DTMF) via the GSM system".

[12a] TS 23.071: "Digital cellular telecommunications system (Phase 2+); Location Services; Functional description – Stage 2".

[13] GSM 03.20: "Digital cellular telecommunications system (Phase 2+); Security related network functions".

[14] TS 23.122: "NAS Functions related to Mobile Station (MS) in idle mode".

[15] GSM 04.02: "Digital cellular telecommunications system (Phase 2+); GSM Public Land Mobile Network (PLMN) access reference configuration".

[16] GSM 04.03: "Digital cellular telecommunications system (Phase 2+); Mobile Station - Base Station System (MS - BSS) interface Channel structures and access capabilities".

[17] GSM 04.04: "Digital cellular telecommunications system (Phase 2+); layer 1 General requirements".

[18] GSM 04.05: "Digital cellular telecommunications system (Phase 2+); Data Link (DL) layer General aspects".

[19] GSM 04.06: "Digital cellular telecommunications system (Phase 2+); Mobile Station - Base Station System (MS - BSS) interface Data Link (DL) layer specification".

[20] TS 24.007: "Digital cellular telecommunications system (Phase 2+); Mobile radio interface signalling layer 3 General aspects".

[21] TS 24.010: "Digital cellular telecommunications system ; Mobile radio interface layer 3 Supplementary services specification General aspects".

[22] TS 24.011: "Point-to-Point (PP) Short Message Service (SMS) support on mobile radio interface".

[23] TS 24.012: "Short Message Service Cell Broadcast (SMSCB) support on the mobile radio interface".

[23a] TS 24.071: "Digital cellular telecommunications system (Phase 2+); Mobile radio interface layer 3 location services specification.

[23b] GSM 04.31 "Digital cellular telecommunication system (Phse 2+);Location Services;Mobile Station (MS) – Serving Mobile Location Centre (SMLC); Radio Resource LCS Protocol (RRLP)".

[24] TS 24.080: "Digital cellular telecommunications system (Phase 2+); Mobile radio interface layer 3 supplementary services specification Formats and coding".

[25] TS 24.081: "Digital cellular telecommunications system (Phase 2+); Line identification supplementary services - Stage 3".

[26] TS 24.082: "Digital cellular telecommunications system (Phase 2+); Call Forwarding (CF) supplementary services - Stage 3".

[27] TS 24.083: "Digital cellular telecommunications system (Phase 2+); Call Waiting (CW) and Call Hold (HOLD) supplementary services - Stage 3".

[28] TS 24.084: "Digital cellular telecommunications system (Phase 2+); MultiParty (MPTY) supplementary services - Stage 3".

[29] TS 24.085: "Digital cellular telecommunications system (Phase 2+); Closed User Group (CUG) supplementary services - Stage 3".

[30] TS 24.086: "Digital cellular telecommunications system (Phase 2+); Advice of Charge (AoC) supplementary services - Stage 3".

[31] TS 24.088: "Call Barring (CB) supplementary services - Stage 3".

[32] GSM 05.02: "Digital cellular telecommunications system (Phase 2+); Multiplexing and multiple access on the radio path".

[33] GSM 05.05: "Digital cellular telecommunications system (Phase 2+); Radio transmission and reception".

[34] GSM 05.08: "Digital cellular telecommunications system (Phase 2+); Radio subsystem link control".

[35] GSM 05.10: "Digital cellular telecommunications system (Phase 2+); Radio subsystem synchronization".

[36] TS 27.001: "General on Terminal Adaptation Functions (TAF) for Mobile Stations (MS)".

[37] TS 29.002: "Digital cellular telecommunications system (Phase 2+); Mobile Application Part (MAP) specification".

[38] TS 29.007: "Digital cellular telecommunications system (Phase 2+); General requirements on interworking between the Public Land Mobile Network (PLMN) and the Integrated Services Digital Network (ISDN) or Public Switched Telephone Network (PSTN)".

[39] GSM 11.10: "Digital cellular telecommunications system (Phase 2+); Mobile Station (MS) conformity specification".

[40] GSM 11.21: "Digital cellular telecommunications system (Phase 2); The GSM Base Station System (BSS) equipment specification".

[41] ISO/IEC 646 (1991): "Information technology - ISO 7-bit coded character set for information interchange".

[42] ISO/IEC 6429: "Information technology - Control functions for coded character sets".

[43] ISO 8348 (1987): "Information processing systems - Data communications - Network service definition".

[44] CCITT Recommendation E.163: "Numbering plan for the international telephone service".

[45] CCITT Recommendation E.164: "Numbering plan for the ISDN era".

[46] CCITT Recommendation E.212: "Identification plan for land mobile stations".

[47] ITU-T Recommendation F.69 (1993): "Plan for telex destination codes".

[48] CCITT Recommendation I.330: "ISDN numbering and addressing principles".

[49] CCITT Recommendation I.440 (1989): "ISDN user-network interface data link layer - General aspects".

[50] CCITT Recommendation I.450 (1989): "ISDN user-network interface layer 3 General aspects".

[51] ITU-T Recommendation I.500 (1993): "General structure of the ISDN interworking recommendations".

[52] CCITT Recommendation T.50: "International Alphabet No. 5".

[53] ITU Recommendation Q.931: ISDN user-network interface layer 3 specification for basic control".

[54] CCITT Recommendation V.21: "300 bits per second duplex modem standardized for use in the general switched telephone network".

[55] CCITT Recommendation V.22: "1200 bits per second duplex modem standardized for use in the general switched telephone network and on point-to-point 2-wire leased telephone-type circuits".

[56] CCITT Recommendation V.22bis: "2400 bits per second duplex modem using the frequency division technique standardized for use on the general switched telephone network and on point-to-point 2-wire leased telephone-type circuits".

[57] CCITT Recommendation V.23: "600/1200-baud modem standardized for use in the general switched telephone network".

[58] CCITT Recommendation V.26ter: "2400 bits per second duplex modem using the echo cancellation technique standardized for use on the general switched telephone network and on point-to-point 2-wire leased telephone-type circuits".

[59] CCITT Recommendation V.32: "A family of 2-wire, duplex modems operating at data signalling rates of up to 9600 bit/s for use on the general switched telephone network and on leased telephone-type circuits".

[60] CCITT Recommendation V.110: "Support of data terminal equipments (DTEs) with V-Series interfaces by an integrated services digital network".

[61] CCITT Recommendation V.120: "Support by an ISDN of data terminal equipment with V-Series type interfaces with provision for statistical multiplexing".

[62] CCITT Recommendation X.21: "Interface between data terminal equipment (DTE) and data circuit-terminating equipment (DCE) for synchronous operation on public data networks".

[63] CCITT Recommendation X.25: "Interface between data terminal equipment (DTE) and data circuit-terminating equipment (DCE) for terminals operating in the packet mode and connected to public data networks by dedicated circuit".

[64] CCITT Recommendation X.28: "DTE/DCE interface for a start-stop mode data terminal equipment accessing the packet assembly/disassembly facility (PAD) in a public data network situated in the same country".

[65] CCITT Recommendation X.30: "Support of X.21, X.21 bis and X.20 bis based data terminal equipments (DTEs) by an integrated services digital network (ISDN)".

[66] CCITT Recommendation X.31: "Support of packet mode terminal equipment by an ISDN".

[67] CCITT Recommendation X.32: "Interface between data terminal equipment (DTE) and data circuit-terminating equipment (DCE) for terminals operating in the packet mode and accessing a packet switched public data network through a public switched telephone network or an integrated services digital network or a circuit switched public data network".

[68] CCITT Recommendation X.75 (1988): "Packet-switched signalling system between public networks providing data transmission services".

[69] CCITT Recommendation X.121: "International numbering plan for public data networks".

[70] ETS 300 102-1: "Integrated Services Digital Network (ISDN); User-network interface layer 3 Specifications for basic call control".

[71] ETS 300 102-2: "Integrated Services Digital Network (ISDN); User-network interface layer 3 Specifications for basic call control".

[72] ISO/IEC10646: “Universal Multiple-Octet Coded Character Set (UCS)”; UCS2, 16 bit coding.

[73] TS 22.060: "General Packet Radio Service (GPRS); Service Description; Stage 1".

[74] TS 23.060: "General Packet Radio Service (GPRS); Service Description; Stage 2".

[75] GSM 03.64: "Digital cellular telecommunications system (Phase 2+); General Packet Radio Service (GPRS); Overall description of the GPRS radio interface; Stage 2".

[76] GSM 04.60: "Digital cellular telecommunications system (Phase 2+); General Packet Radio Service (GPRS); Mobile Station - Base Station System (MS-BSS) interface; Radio Link Control and Medium Access Control (RLC/MAC) layer specification".

[77] IETF RFC 1034: "Domain names - Concepts and Facilities " (STD 7).

[78] GSM 04.65: "Digital cellular telecommunications system (Phase 2+); General Packet Radio Service (GPRS); Subnetwork Dependent Convergence Protocol (SNDCP)".

[79] ITU Recommendation I.460: “Multiplexing, rate adaption and support of existing services”.

[80] TS 23.107: “3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; QoS Concept and Architecture”

9.5 GPRS Session Management Messages

9.5.1 Activate PDP context request

This message is sent by the MS to the network to request activation of a PDP context.
See table 9.5.1/TS 24.008.

Message type: ACTIVATE PDP CONTEXT REQUEST

Significance: global

Direction: MS to network

Table 9.5.1/TS 24.008: ACTIVATE PDP CONTEXT REQUEST message content

IEI	Information Element	Type/Reference	Presence	Format	Length
	Protocol discriminator	Protocol discriminator 10.2	M	V	1/2
	Transaction identifier	Transaction identifier 10.3.2	M	V	1/2
	Activate PDP context request message identity	Message type 10.4	M	V	1
	Requested NSAPI	Network service access point identifier 10.5.6.2	M	V	1
	Requested LLC SAPI	LLC service access point identifier 10.5.6.9	M	V	1
	Requested QoS	Quality of service 10.5.6.5	M	LV	~~19~~12
	Requested PDP address	Packet data protocol address 10.5.6.4	M	LV	3 - 19
28	Access point name	Access point name 10.5.6.1	O	TLV	3 - 102
27	Protocol configuration options	Protocol configuration options 10.5.6.3	O	TLV	3 - 253

9.5.1.1 Access point name

This IE is included in the message when the MS selects a specific external network to be connected to.

9.5.1.2 Protocol configuration options

This IE is included in the message when the MS provides protocol configuration options for the external PDN.

9.5.2 Activate PDP context accept

This message is sent by the network to the MS to acknowledge activation of a PDP context.
See table 9.5.2/TS 24.008.

Message type: ACTIVATE PDP CONTEXT ACCEPT

Significance: global

Direction: network to MS

Table 9.5.2/TS 24.008: ACTIVATE PDP CONTEXT ACCEPT message content

IEI	Information Element	Type/Reference	Presence	Format	Length
	Protocol discriminator	Protocol discriminator 10.2	M	V	1/2
	Transaction identifier	Transaction identifier 10.3.2	M	V	1/2
	Activate PDP context accept message identity	Message type 10.4	M	V	1
	Negotiated LLC SAPI	LLC service access point identifier 10.5.6.9	M	V	1
	Negotiated QoS	Quality of service 10.5.6.5	M	LV	~~19~~12
	Radio priority	Radio priority 10.5.7.2	M	V	1/2
	Spare half octet	Spare half octet 10.5.1.8	M	V	1/2
2B	PDP address	Packet data protocol address 10.5.6.4	O	TLV	4 - 20
27	Protocol configuration options	Protocol configuration options 10.5.6.3	O	TLV	3 - 253
34	Packet Flow Identifier	Packet Flow Identifier 10.5.6.11	O	TLV	3

9.5.2.1 PDP address

If the MS did not request a static address in the corresponding ACTIVATE PDP CONTEXT REQUEST message, the network shall include the PDP address IE in this ACTIVATE PDP CONTEXT ACCEPT message.

If the MS requested a static address in the corresponding ACTIVATE PDP CONTEXT REQUEST message, the network shall not include the PDP address IE in this ACTIVATE PDP CONTEXT ACCEPT message.

9.5.2.2 Protocol configuration options

This IE is included in the message when the network wishes to transmit protocol configuration options for the external PDN.

9.5.2.3 Packet Flow Identifier

This IE may be included if the network wants to indicate the Packet Flow Identifier associated to the PDP context.

Next Modified Section

9.5.4 Activate Secondary PDP Context Request

This message is sent by the MS to the network to request activation of a secondary PDP context. See Table 9.5.4/TS 24.008.

Message type: ACTIVATE SECONDARY PDP CONTEXT REQUEST

Significance: global

Direction: MS to network

Table 9.5.4/TS 24.008: Activate SECONDARY PDP context request message content

IEI	Information Element	Type/Reference	Presence	Format	Length
	Protocol discriminator	Protocol discriminator 10.2	M	V	½
	Transaction identifier	Transaction identifier 10.3.2	M	V	½
	Activate secondary PDP context request message identity	Message type 10.4	M	V	1
	Requested NSAPI	Network service access point identifier 10.5.6.2	M	V	1
	Requested LLC SAPI	LLC service access point identifier 10.5.6.9	M	V	1
	Requested QoS	Quality of service 10.5.6.5	M	LV	~~FFS~~ 12
	TFT	Traffic Flow Template	M	LV	FFS
	Linked TI	Linked TI 10.5.6.7	M	LV	2-3

9.5.5 Activate Secondary PDP Context Accept

This message is sent by the network to the MS to acknowledge activation of a secondary PDP context. See Table 9.5.5/TS 24.008.

Message type: ACTIVATE SECONDARY PDP CONTEXT ACCEPT

Significance: global

Direction: network to MS

Table 9.5.5/TS 24.008: ACTIVATE SECONDARY PDP CONTEXT ACCEPT message content

IEI	Information Element	Type/Reference	Presence	Format	Length
	Protocol discriminator	Protocol discriminator 10.2	M	V	½
	Transaction identifier	Transaction identifier 10.3.2	M	V	½
	Activate secondary PDP context accept message identity	Message type 10.4	M	V	1
	Negotiated LLC SAPI	LLC service access point identifier 10.5.6.9	M	V	1
	Negotiated QoS	Quality of service 10.5.6.5	M	LV	~~FFS~~ 12
	Radio priority	Radio priority	M	V	½
	Spare half octet	Spare half octet 10.5.1.8	M	V	½
34	Packet Flow Identifier	Packet Flow Identifier 10.5.6.11	O	TLV	3

9.5.5.1 Packet Flow Identifier

This IE may be included if the network wants to indicate the Packet Flow Identifier associated to the PDP context.

Next Modified Section

9.5.9 Modify PDP context request (Network to MS direction)

This message is sent by the network to the MS to request modification of an active PDP context. See table 9.5.9/TS 24.008.

Message type: MODIFY PDP CONTEXT REQUEST (NETWORK TO MS DIRECTION)

Significance: global

Direction: network to MS

Table 9.5.9/TS 24.008: MODIFY PDP CONTEXT REQUEST (NETWORK TO MS DIRECTION) message content

IEI	Information Element	Type/Reference	Presence	Format	Length
	Protocol discriminator	Protocol discriminator 10.2	M	V	1/2
	Transaction identifier	Transaction identifier 10.3.2	M	V	1/2
	Modify PDP context request message identity	Message type 10.4	M	V	1
	Radio priority	Radio priority 10.5.7.2	M	V	1/2
	Spare half octet	Spare half octet 10.5.1.8	M	V	1/2
	Requested LLC SAPI	LLC service access point identifier 10.5.6.9	M	V	1
	New QoS	Quality of service 10.5.6.5	M	LV	~~19~~12
2B	PDP address	Packet data protocol address 10.5.6.4	O	TLV	4-20
34	Packet Flow Identifier	Packet Flow Identifier 10.5.6.11	O	TLV	3

9.5.9.1 PDP address

If the MS requested external PDN address allocation at PDP context activation via an APN and this was confirmed by the network in the ACTIVATE PDP CONTEXT ACCEPT message, then the network shall include the PDP address IE in the MODIFY PDP CONTEXT REQUEST message once the address has been actually allocated, in order to update the PDP context in the MS.

9.5.9.2 Packet Flow Identifier

This IE may be included if the network wants to indicate the Packet Flow Identifier associated to the PDP context.

9.5.10 Modify PDP context request (MS to network direction)

This message is sent by the MS to the network to request modification of an active PDP context. See table 9.5.10/TS 24.008.

Message type: MODIFY PDP CONTEXT REQUEST (MS TO NETWORK DIRECTION)

Significance: global

Direction: MS to network

Table 9.5.10/TS 24.008: modify PDP context request (MS to network direction) message content

IEI	Information Element	Type/Reference	Presence	Format	Length
	Protocol discriminator	Protocol discriminator 10.2	M	V	1/2
	Transaction identifier	Transaction identifier 10.3.2	M	V	1/2
	Modify PDP context request message identity	Message type 10.4	M	V	1
32	Requested LLC SAPI	LLC service access point identifier 10.5.6.9	O	TV	2
30	Requested new QoS	Quality of service 10.5.6.5	O	TLV	~~FFS~~ 13
31	New TFT	Traffic Flow Template	O	TLV	FFS

9.5.10.1 Requested LLC SAPI

This IE may be included in the message to request a new LLC SAPI if a new QoS is requested.

9.5.10.2 Requested new QoS

This IE may be included in the message to request a modification of the QoS.

9.5.10.3 New TFT

This IE is included in the message only when the modification applies to a secondary PDP context (FFS), to request modification of the TFT.

Next Modified Section

9.5.12 Modify PDP context accept (Network to MS direction)

This message is sent by the network to the MS to acknowledge the modification of an active PDP context. See table 9.5.12/TS 24.008.

Message type: MODIFY PDP CONTEXT ACCEPT (NETWORK TO MS DIRECTION)

Significance: global

Direction: Network to MS

Table 9.5.12/TS 24.008: modify PDP context accept (NETWORK to ms direction) message content

IEI	Information Element	Type/Reference	Presence	Format	Length
	Protocol discriminator	Protocol discriminator 10.2	M	V	~~1/2~~½
	Transaction identifier	Transaction identifier 10.3.2	M	V	~~1/2~~½
	Modify PDP context accept message identity	Message type 10.4	M	V	1
30	Negotiated QoS	Quality of service 10.5.6.5	O	TLV	~~FFS~~ 13
32	Negotiated LLC SAPI	LLC service access point identifier 10.5.6.9	O	TV	2
33	New radio priority	Radio priority 10.5.7.2	O	TV	1
34	Packet Flow Identifier	Packet Flow Identifier 10.5.6.11	O	TLV	3

9.5.12.1 Negotiated QoS

This IE is included in the message if the network assigns a new QoS.

9.5.12.2 Negotiated LLC SAPI

This IE is included in the message if the network assigns a new LLC SAPI.

9.5.12.3 New radio priority

This IE is included in the message only if the network modifies the radio priority.

9.5.12.4 Packet Flow Identifier

This IE may be included if the network wants to indicate the Packet Flow Identifier associated to the PDP context.

Next Modified Section

10.5.6.5 Quality of service

The purpose of the *quality of service* information element is to specify the QoS parameters for a PDP context.

The QoS IE is defined to allow backward compatibility to earlier version of Session Management Protocol.

The *quality of service* is a type 4 information element with a length of ~~20~~13 octets.

The *quality of service* information element is coded as shown in figure 10.5.138/TS 24.008 and table 10.5.156/TS 24.008.

<table>
<tr><td>8</td><td>7</td><td>6 ~~6~~</td><td>5</td><td>4</td><td>3</td><td>2</td><td>1</td><td></td></tr>
<tr><td colspan="8">Quality of service IEI</td><td>octet 1</td></tr>
<tr><td colspan="8">Length of quality of service IE</td><td>Octet 2</td></tr>
<tr><td colspan="2">0 0
spare</td><td colspan="3">Delay
class</td><td colspan="3">Reliability
class</td><td>octet 3</td></tr>
<tr><td colspan="4">Peak
throughput</td><td>0
spare</td><td colspan="3">Precedence
class</td><td>octet 4</td></tr>
<tr><td colspan="3">0 0 0
spare</td><td colspan="5">Mean
throughput</td><td>octet 5</td></tr>
<tr><td colspan="3">Traffic Class</td><td colspan="2">~~0 spare~~
Delivery
order</td><td colspan="3">Delivery of
erroneous SDU</td><td>Octet 6</td></tr>
<tr><td colspan="8">Maximum SDU size</td><td>Octet 7
~~Octet 8~~</td></tr>
<tr><td colspan="8">Maximum bit rate for uplink</td><td>Octet 8~~9~~
~~Octet 10~~</td></tr>
<tr><td colspan="8">Maximum bit rate for downlink</td><td>Octet
9~~11~~
~~Octet 12~~</td></tr>
<tr><td colspan="4">Residual BER</td><td colspan="4">SDU error ratio</td><td>Octet
10~~3~~</td></tr>
<tr><td colspan="8">~~SDU error ratio~~</td><td>~~Octet 14~~</td></tr>
<tr><td colspan="6">Transfer delay</td><td colspan="2">Traffic
Handling
priority</td><td>Octet
11~~5~~</td></tr>
<tr><td colspan="8">Guaranteed bit rate for uplink</td><td>Octet
12~~6~~
~~Octet 17~~</td></tr>
<tr><td colspan="8">Guaranteed bit rate for downlink</td><td>Octet
13~~8~~
~~Octet 19~~</td></tr>
<tr><td colspan="5">~~0 0 0 0 0 0~~
~~spare~~</td><td colspan="3">~~Traffic handling priority~~</td><td>Octet ~~20~~</td></tr>
</table>

Figure 10.5.138/TS 24.008: *Quality of service* information element

Table 10.5.156/TS 24.008: *Quality of service* information element

Reliability class, octet 3 (see TS 23.~~060~~107) Bits 3 2 1 In MS to network direction: 0 0 0 Subscribed reliability class In network to MS direction: 0 0 0 Reserved In MS to network direction and in network to MS direction : 0 0 1 Acknowledged GTP, LLC, and RLC; Protected data 0 1 0 Unacknowledged GTP; Acknowledged LLC and RLC, Protected data 0 1 1 Unacknowledged GTP and LLC; Acknowledged RLC, Protected data 1 0 0 Unacknowledged GTP, LLC, and RLC, Protected data 1 0 1 Unacknowledged GTP, LLC, and RLC, Unprotected data 1 1 1 Reserved All other values are interpreted as *Unacknowledged GTP and LLC; Acknowledged RLC, Protected data* in this version of the protocol. Delay class, octet 3 (see TS 22.060 and TS 23.~~060~~107) Bits 6 5 4 In MS to network direction: 0 0 0 Subscribed delay class In network to MS direction: 0 0 0 Reserved In MS to network direction and in network to MS direction : 0 0 1 Delay class 1 0 1 0 Delay class 2 0 1 1 Delay class 3 1 0 0 Delay class 4 (best effort) 1 1 1 Reserved All other values are interpreted as *Delay class 4 (best effort)* in this version of the protocol. Bit 7 and 8 of octet 3 are spare and shall be coded all 0. Precedence class, octet 4 (see TS 23.~~060~~107) Bits 3 2 1 In MS to network direction: 0 0 0 Subscribed precedence In network to MS direction: 0 0 0 Reserved In MS to network direction and in network to MS direction : 0 0 1 High priority 0 1 0 Normal priority 0 1 1 Low priority 1 1 1 Reserved

All other values are interpreted as *Normal priority* in this version of the protocol.

Bit 4 of octet 4 is spare and shall be coded as 0.

Peak throughput, octet 4 (see TS 23.107)
Bits
8 7 6 5
In MS to network direction:
0 0 0 0 Subscribed peak throughput
In network to MS direction:
0 0 0 0 Reserved
In MS to network direction and in network to MS direction :

0 0 0 1	Up to 1 000 octet/s
0 0 1 0	Up to 2 000 octet/s
0 0 1 1	Up to 4 000 octet/s
0 1 0 0	Up to 8 000 octet/s
0 1 0 1	Up to 16 000 octet/s
0 1 1 0	Up to 32 000 octet/s
0 1 1 1	Up to 64 000 octet/s
1 0 0 0	Up to 128 000 octet/s
1 0 0 1	Up to 256 000 octet/s
1 1 1 1	Reserved

All other values are interpreted as *Up to 1 000 octet/s* in this version of the protocol.
Mean throughput, octet 5 (see TS 23.107)
Bits
5 4 3 2 1
In MS to network direction:
0 0 0 0 0 Subscribed mean throughput
In network to MS direction:
0 0 0 0 0 Reserved
In MS to network direction and in network to MS direction :

0 0 0 0 1	100 octet/h
0 0 0 1 0	200 octet/h
0 0 0 1 1	500 octet/h
0 0 1 0 0	1 000 octet/h
0 0 1 0 1	2 000 octet/h
0 0 1 1 0	5 000 octet/h
0 0 1 1 1	10 000 octet/h
0 1 0 0 0	20 000 octet/h
0 1 0 0 1	50 000 octet/h
0 1 0 1 0	100 000 octet/h
0 1 0 1 1	200 000 octet/h
0 1 1 0 0	500 000 octet/h
0 1 1 0 1	1 000 000 octet/h
0 1 1 1 0	2 000 000 octet/h
0 1 1 1 1	5 000 000 octet/h
1 0 0 0 0	10 000 000 octet/h
1 0 0 0 1	20 000 000 octet/h
1 0 0 1 0	50 000 000 octet/h
1 1 1 1 0	Reserved
1 1 1 1 1	Best effort

The value Best effort indicates that throughput shall be made available to the MS on a per need and availability basis.
All other values are interpreted as *Best effort* in this version of the protocol.

Bits 8 to 6 of octet 5 are spare and shall be coded all 0.

Delivery of erroneous SDUs, octet 6 (see TS 23.107)
Bits
3 2 1
In MS to network direction:
0 0 0 Subscribed delivery of erroneous SDUs
In network to MS direction:
0 0 0 Reserved
In MS to network direction and in network to MS direction :
0 0 1 No detect ('-')
0 1 0 Erroneous SDUs are delivered ('yes')
0 1 1 Erroneous SDUs are not delivered ('no')
1 1 1 Reserved

~~All other values are reserved.~~
The network shall map all other values not explicitly defined onto one of the values defined in this version of the protocol. The network shall return a negotiated value which is explicitly defined in this version of this protocol.

The MS shall consider all other values as reserved.

Delivery order, octet 6 (see TS 23.107)
Bits
5 4 ~~3~~
In MS to network direction:
0 0 Subscribed delivery order
In network to MS direction:
0 0 Reserved
In MS to network direction and in network to MS direction :
0 1 With delivery order ('yes')
1 0 Without delivery order ('no')
1 1 Reserved

~~All other values are reserved.~~
~~Bit 5 of octet 6 is spare and shall be coded all 0.~~

Traffic class, octet 6 (see TS 23.107)
Bits
8 7 6
In MS to network direction:
0 0 0 Subscribed traffic class
In network to MS direction:
0 0 0 Reserved
In MS to network direction and in network to MS direction :
0 0 1 Conversational class
0 1 0 Streaming class
0 1 1 Interactive class
1 0 0 Background class
1 1 1 Reserved

The network shall map all other values not explicitly defined onto one of the values defined in this version of the protocol. The network shall return a negotiated value which is explicitly defined in this version of this protocol.

The MS shall consider all other values as reserved.

~~All other values are reserved.~~

Maximum SDU size, octet 7 (see TS 23.107)
The Maximum SDU size value is binary coded in 8 bits, using a granularity of 10 octets.
In MS to network direction:
00000000 Subscribed maximum SDU size
11111111 Reserved
In network to MS direction:
00000000 Reserved
11111111 Reserved
In MS to network direction and in network to MS direction :

For values in the range 00000001 to 10010110 the Maximum SDU size value is binary coded in 8 bits, using a granularity of 10 octets, giving a range of values from 10 octets to 1500 octets.
Values above 10010110 are as below:
10010111 1502 octets
10011000 1510 octets

10011001 1520 octets

The network shall map all other values not explicitly defined onto one of the values defined in this version of the protocol. The network shall return a negotiated value which is explicitly defined in this version of this protocol.

The MS shall consider all other values as reserved.

~~Maximum SDU size, octet 7 and 8~~
~~In MS to network direction:~~
~~All bits 1 Subscribed maximum SDU size~~
~~In network to MS direction:~~
~~All bits 1 Reserved~~
~~In MS to network direction and in network to MS direction :~~~~interpreted as 1502~~
~~The Maximum SDU size value consists of 16 bits. Refer to TS 23.107 for the maximum value. The granularity is 1 octet.~~

Maximum bit rate for uplink, octet ~~9~~8~~ and 10~~
Bits
8 7 6 5 4 3 2 1
In MS to network direction:
0 0 0 0 0 0 0 0~~All bits 1~~ Subscribed maximum bit rate for uplink
In network to MS direction:
0 0 0 0 0 0 0 0~~All bits 1~~ Reserved
In MS to network direction and in network to MS direction :
0 0 0 0 0 0 0 1 – The maximum bit rate is binary coded in 8 bits, using a granularity of 1 kbps
0 0 1 1 1 1 1 1 giving a range of values from 1 kbps to 63 kbps in 1 kbps increments.

0 1 0 0 0 0 0 0 – The maximum bit rate is 64 kbps + ((the binary coded value in 8 bits –01000000) * 8 kbps)
0 1 1 1 1 1 1 1 giving a range of values from 64 kbps to 564 kbps in 8 kbps increments.

1 0 0 0 0 0 0 0 - The maximum bit rate is 576 kbps + ((the binary coded value in 8 bits –10000000) * 64 kbps)
1 1 1 1 1 1 1 0 giving a range of values from 576 kbps to 8640 kbps in 64 kbps increments.

1 1 1 1 1 1 1 1 Reserved
~~The Maximum bit rate for uplink value consists of 16 bits. Maximum value is 2000 kbps. The granularity is 4 kbps.~~

Maximum bit rate for downlink, octet ~~11~~9~~ and 12~~ (see TS 23.107)
~~In MS to network direction:~~
~~All bits 1 Subscribed maximum bit rate for downlink~~
~~In network to MS direction:~~
~~All bits 1 ReservedIn MS to network direction and in network to MS direction :~~
~~The Maximum bit rate for downlink value consists of 16 bits. Maximum value is 2000 kbps. The granularity is 4 kbps.~~
Coding is identical to that of Maximum bit rate for uplink.

Residual Bit Error Rate (BER), octet 1~~3~~0 (see TS 23.107)
Bits
8 7 6 5 ~~4 3 2 1~~
In MS to network direction:
0 0 0 0 ~~0 0 0 0~~ Subscribed residual BER
In network to MS direction:
0 0 0 0 ~~0 0 0 0~~ Reserved
In MS to network direction and in network to MS direction :
The Residual BER value consists of ~~8~~ 4 bits. The ranges from $5*10^{-2}$ to $6*10^{-8}$. ~~4 bits is assigned to multiplicand and exponent, respectively.~~
~~0 1 0 1 0 0 1 0~~ 0 0 0 1 $5*10^{-2}$
~~0 0 0 1 0 0 1 0~~ 0 0 1 0 $1*10^{-2}$
0 0 1 1 $5*10^{-3}$
~~0 1 0 0 0 0 1 1~~ 0 1 0 0 $4*10^{-3}$
~~0 0 0 1 0 0 1 1~~ 0 1 0 1 $1*10^{-3}$
~~0 0 0 1 0 1 0 0~~ 0 1 1 0 $1*10^{-4}$
~~0 0 0 1 0 1 0 1~~ 0 1 1 1 $1*10^{-5}$
~~0 0 0 1 0 1 1 0~~ 1 0 0 0 $1*10^{-6}$
~~0 1 1 0 1 0 0 0~~ 1 0 0 1 $6*10^{-8}$
1111 Reserved

The network shall map all other values not explicitly defined onto one of the values defined in this version of the protocol. The network shall return a negotiated value which is explicitly defined in this version of the protocol.

The MS shall consider all other values as reserved.
~~All other values are reserved.~~

SDU error ratio, octet 10~~4~~ (see TS 23.107)
Bits
~~8 7 6 5~~ 4 3 2 1
In MS to network direction:
~~0 0 0 0~~ 0 0 0 0 Subscribed SDU error ratio
In network to MS direction:
~~0 0 0 0~~ 0 0 0 0 Reserved
In MS to network direction and in network to MS direction :
The SDU error ratio value consists of ~~8~~ 4 bits. The ranges from $1*10^{-2}$ to $1*10^{-6}$. ~~4 bits is assigned to multiplicand and exponent, respectively.~~
~~0 0 0 1 0 0 1 0~~ 0 0 0 1 $1*10^{-2}$
0 0 1 0 $7*10^{-3}$
~~0 0 0 1~~ 0 0 1 1 $1*10^{-3}$
~~0 0 0 1~~ 0 1 0 0 $1*10^{-4}$
~~0 0 0 1~~ 0 1 0 1 $1*10^{-5}$
~~0 0 0 1~~ 0 1 1 0 $1*10^{-6}$
1 1 1 1 Reserved

~~All other values are reserved.~~ The network shall map all other values not explicitly defined onto one of the values defined in this version of the protocol. The network shall return a negotiated value which is explicitly defined in this version of the protocol.

The MS shall consider all other values as reserved.

Traffic handling priority, octet 11 (see TS 23.107)
Bits
2 1
In MS to network direction:
0 0 Subscribed traffic handling priority
In network to MS direction:
0 0 Reserved
In MS to network direction and in network to MS direction :
0 1 Priority level 1
1 0 Priority level 2
1 1 Priority level 3

The Traffic handling priority value is ignored if the Traffic Class is Conversation class, Streaming class or Background class.

Transfer delay, octet 11~~5~~ (See TS 23.107)
Bits
8 7 6 5 4 3
In MS to network direction:
0 0 0 0 0 0~~All bits 1~~ Subscribed transfer delay
In network to MS direction:
0 0 0 0 0 0~~All bits 1~~ Reserved
In MS to network direction and in network to MS direction :
0 0 0 0 0 1 – The Transfer delay is binary coded in 6 bits, using a granularity of 10 ms
0 0 1 1 1 1 giving a range of values from 10 ms to 150 ms in 10 ms increments

0 1 0 0 0 0 – The transfer delay is 200 ms + ((the binary coded value in 6 bits – 010000) * 50 ms)
0 1 1 1 1 1 giving a range of values from 200 ms to 950 ms in 50ms increments

1 0 0 0 0 0 – The transfer delay is 1000 ms + ((the binary coded value in 6 bits – 100000) * 100 ms)
1 1 1 1 1 0 giving a range of values from 1000 ms to 4100 ms in 100ms increments

1 1 1 1 1 1 Reserved

~~The Transfer delay value consists of 8 bits. Maximum value is 2560ms. The granularity is 10 ms.~~

The Transfer delay value is ignored if the Traffic Class is Interactive class or Background class.

Guaranteed bit rate for uplink, octet 16 and 17 (See TS 23.107)
~~In MS to network direction:~~
~~All bits 1 Subscribed guaranteed bit rate for uplink~~
~~In network to MS direction:~~
~~All bits 1 Reserved~~
~~In MS to network direction and in network to MS direction :~~

~~The Guaranteed bit rate for uplink value consists of 16 bits. Maximum value is 2000 kbps. The granularity is 4 kbps.~~
Coding is identical to that of Maximum bit rate for uplink.
The Guaranteed bit rate for uplink value is ignored if the Traffic Class is Interactive class or Background class.

Guaranteed bit rate for downlink, octet 13~~8 and 19~~ (See TS 23.107)
~~In MS to network direction:~~
~~All bits 1 Subscribed guaranteed bit rate for downlink~~
~~In network to MS direction:~~
~~All bits 1 Reserved~~
~~In MS to network direction and in network to MS direction :~~
~~The Guaranteed bit rate for downlink value consists of 16 bits. Maximum value is 2000 kbps. The granularity is 4 kbps.~~
Coding is identical to that of Maximum bit rate for uplink.
The Guaranteed bit rate for downlink value is ignored if the Traffic Class is Interactive class or Background class.

~~Traffic handling priority, octet 20 (see TS 23.107)~~
~~Bits~~
~~2 1~~
~~In MS to network direction:~~
~~0 0 Subscribed traffic handling priority~~
~~In network to MS direction:~~
~~0 0 Reserved~~
~~In MS to network direction and in network to MS direction :~~
~~0 1 Priority level 1~~
~~1 0 Priority level 2~~
~~1 1 Priority level 3~~

~~All other values are reserved.~~

~~The Traffic handling priority value is ignored if the Trafic Class is Conversation class, Streaming class or Background class.~~

3GPP TSG-N WG1 N1-000518
Umea, Sweden
28 Feb – 03 Mar 2000

Source: N1

To: RAN3, S4

CC: N2B, RAN2

Agenda Item: LS out (OoBTC)

Title: LS on AMR modes & Supported Subflow Combinations

Contact Person: Phil Hodges (phil.hodges@eed.ericsson.se)

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1. Introduction

N1 is currently undergoing the standardisation of call control procedures for support of the work item for Out Of Band Transcoder Control. These procedures assume the goal is to establish Transcoder Free connections for MS to MS calls. Multirate codecs (such as UMTS AMR) could be negotiated between end points and it is assumed by N1 that the result could in some cases be a subset of the modes applicable to a particular codec type -ACS.

2. Allocation Of RFCIs

If the endpoints of a Transcoder Free negotiated connection are connected via UTRANs then each MSC must perform a RAB assignment requesting a set of SDU formats which correspond to the negotiated modes of the selected codec type – the ACS.

It is assumed by N1 that if the RNC accepts the RAB Assignment Request then it can support all of the SDU formats requested by the MSC. If this is not the case then the TrFO connection could result in a through connected call with no compatible Sub-flow combinations because there is no possibility for the OoBTC to subsequently insert a transcoder – it doesn't know there is a problem.

N1 is concerned that this assumption has not been supported by RAN WGs due to the inclusion of point 2 in Annex A of TS 25.415:

" Allocation of RFCIs: the RNC dynamically allocates an identification (RFCI) to each permitted/possible combinations that it can offer……."

Although this statement is part of an Informative annex it suggests that it is possible for an RNC to only initialise a subset of the requested SDU formats from the RAB assignment.

3. Active Codec Set to the MS

The AMR codec types for UMTS & GSM include the defined parameter ACS – Active Codec Set. This is defined as the common total set of modes for a given connection between two AMR codecs. It is understood by N1 that rate control techniques perfomed inband may reduce this set to an Allowed Set, or to a single mode – Exact rate control. However it must always be within the agreed ACS.

It is assumed by N1 that no downlink call control message (Direct Transfer) is required because this ACS will be conveyed to the mobile station via the Transport Configuration sets sent by RNC to MS at RB set-up. These will correspond to the same modes requested by the MSC in the RAB assignment request.

4. Initial Codec Mode

It is not clear to N1 if the ICM is really needed in the OoBTC parameters. It is stated in the TS 26.103 that the ICM is optional and if not included by the originating side then the terminating side may select freely. It is assumed that this indicates the mode that one side wishes to receive as a downlink frame. Thus for a given type if ICM is important (i.e. the received frame from the other end) then it would be included in both directions (the returned, Selected Codec with ICM set would be what the terminating end requests that the originating end should send). If a codec type did not have any restriction on which modes from the active set that it could receive from the other end then it would not include the ICM parameter.

If the ICM should be conveyed to the mobile station it could be performed by downlink call control message (Direct Transfer). An alternative to this would be that the RNC informs the UE at RB setup. This would require that the MSC informs the RNC of the ICM at RAB assignment.

It is presumed by N1 that the Direct Transfer proposal is preferred by RAN groups if the MS must be informed at all. This however can only be achieved successfully if the RNC initialises all subflow combinations from the requested SDU's in the RAB assignment.

If ICM is not required then the MS is free to select a start mode from the transport configuration sets allocated at RB setup.

5. Conclusion

N1 requests that it is stated in the RAN technical specifications that the RNC shall initialise all subflow combinations requested by the MSC. In conjunction it should be stated that the MSC shall not request any SDU formats that the serving RNC cannot support.

N1 requests clarification from S4 on the relevance of the ICM in a downlink call control message to the MS. Due to time constraints for N1 to complete this WI by the end of this week, if no decision can be reached in S4, N1 will include the ICM in Selected Codec message sent in Direct Transfer to the MS. However this will be optional as described in TS 26.103.

3GPP TSG-CN-WG1, Meeting #11
28. February - 03.March. 2000
Umea, Sweden

Tdoc N1-000539

To: TSG-S3/SMG10

cc: TSG-T2, SMG9

Source: TSG-N1/SMG3

Title: Reply to LS on "Introduction of rejection of non ciphered calls for GPRS"

Contact: Roland Gruber, Siemens AG
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phone: +49 89 722 46392

N1 thanks S3 for their LS on "Introduction of rejection of non ciphered calls for GPRS" (S3-00 0206). N1 has discussed the topic and came to the conclusion, that S3 is asking for the introduction of a complex new feature that requires work to be done by several TSG working groups, which should be covered by a separate new work item. N1 assumes that S3 would be the best group to initiate and control the work item.

As the R99 and all older releases are already functionally frozen N1 do not see a possibility that, at least for the needed changes to the specifications under its responsibility, caused by this requirement can be completed for R97, 98 or 99.

N1 see that changes to the specifications under its responsibility will be needed if a new R00 work item is approved.

N1 has also discussed the attached CR (S3-000058/ N1-000287) for 04.08/ R98/ GPRS with the result that this CR is rejected by N1.

3GPP TSG-CN-WG1, Meeting #11
28 February - 03.March. 2000
Umea, Sweden

Tdoc N1-000542

Source: TSG-CN WG1

To: TSG-RAN WG2, TSG-RAN WG3

Cc:

Title: Support of Idle-mode DRX control in GMM

Contact: Fumihiko YOKOTA, Fujitsu Limited
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TSG-CN WG1 thanks for the liaison from TSG-RAN WG2 (R2-000576) and TSG-RAN WG3 (R3-000812) on Idle mode DRX control. The information given to N1 was sufficient to proceed work in N1.

The agreed CR to 24.008 for the support of configurable DRX cycle length for each MS is attached. TSG-CN WG1 kindly asks TSG-RAN WG2 and TSG-RAN WG3 for review.

"N1-000551 (DR
parameter for

3GPP TSG-CN-WG1, Meeting #11
28 February – 3 March 2000
Umeå, Sweden

Tdoc N1-000553

Source: **CN1**

Contact Person: Robert Zaus
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To: **RAN2, RAN3, CN2B**

Title: **3rd LS on the Transport of Codec Information during the Codec Negotiation between MS and MSC**

CN1 has received the answer from RAN2 (R2-000545) to CN1's liaison statement on RANAP Transaction Sequence (N1-000487) and would like to thank RAN2 for their rapid response.

CN1 understands that with the services provided by a UTRAN in release 99, in-sequence delivery can only be guaranteed for Direct Transfer messages using the same SAPI, but not between Direct Transfer and other RRC signalling messages.

As the second type of in-sequence delivery was an essential prerequisite for the modified concept for codec negotiation described in Tdoc N1-000517, this concept cannot be implemented, and CN1 asks RAN2, RAN3 and CN2B to consider the LS sent in Tdoc N1-000447 and the related concept description N1-000517 as never sent.

As in this situation one delegation in CN1 asked for more time to study a complete solution of the codec negotiation, CN1 decided not to specify any explicit signalling for the support of UMTS speech codecs in R99. This decision was made possible by the fact that in R99 there will be only one speech codec defined for UMTS, and that support of this codec type is mandatory for all mobile stations supporting UMTS speech services.

Therefore, CN1 asks RAN2 and RAN3 not to implement the working assumptions (N1-000163) which were sent earlier in January together with the liaison statement N1-000164, but to wait for more detailed requirements in R00.