3GPP TSG CN Plenary
Meeting #12, Stockholm, Sweden
13th - 15th June 2001

Tdoc NP-010279

Source: TSG CN WG 1

Title: All LSs sent from CN1 since TSG#11 meeting,- pack2

Agenda item: 6.1.1

Document for: Information

Introduction:

This document contains **10 agreed** LSs sent from **TSG CN WG1**, and are forwarded to TSG CN Plenary meeting #12 for information only.

Meeting	TDoc #	Source	Tdoc Title	Comments
CN1_17	N1-010867	Andrew	Request for information from GSM Europe on 3 digit MNC	Reply to N1-010842 To: S1, CN, SA Cc: N4
CN1_17	N1-010870	Robert	LS on Priority Selection Criteria of Calls in a Multicall	N1-010816, 817, 826, 827 and 870 are linked To: R3 Cc: S1
CN1_17	N1-010873	Xin Chen	Liaison Statement on Adding New Definitions to 21.905	To: S1, SA Cc: S2, GERAN 2
CN1_17	N1-010888	Magnus	Response to LS N1-010504 (S2-010798r2)	Reply to N1-010504 To: S2 Cc: N2, N3, N4
CN1_17	N1-010890	Sunil	Liaison Statement on the IM Call Transfer service	To: S3, S5 Cc: S2, N2, N3. N4
CN1_17	N1-010892	Milo	Liaison Statement on " Handling of ICMP messages by 3GPP SIP Implementations"	To: S3
CN1_17	N1-010902	Eiko	Response LS on Introduction of release information in the MS Radio Access Capability IE in 3GPP TS 24.008	Reply to N1-010687 To: GERAN2
CN1_17	N1-010908	Duncan Mills	Liaison Statement on "24.008 CR for Classmark Issues"	Reply to N1-010620 To: GERAN, R2
CN1_17	N1-010910	Francesco	Liaison Statement on THRESHOLD check at RRC connection establishment	To: R2, S3 Cc: T3
CN1_17	N1-010915	Robert	Liaison Statement on Indication of Intra MSC handover from 3G_MSC-B to MSC-A/3G_MSC-A	To: R3

3GPP TSG CN WG1 Meeting #17
Puerto Rico, 14th - 18th May 2001

Tdoc N1-010867

Agenda item: 9

Liaison Statement

From: **3GPP TSG CN WG1**

To: **3GPP TSG SA WG1, 3GPP TSG CN, 3GPP TSG SA**

Cc: **3GPP TSG CN WG4**

Subject: **Request for information from GSM Europe on 3 digit MNC**

Contact: **Andrew Howell, Motorola**
andrew.howell@motorola.com

3GPP TSG CN WG1 thanks 3GPP TSG SA WG1 for the LS relating to the request, from GSM Europe, for information concerning the introduction of 3 digit MNC in Europe.

3GPP TSG CN WG1 has reviewed the proposal and would like to make the following observations:

- 3 digit MNC were first defined in R98. All implementations that are based on releases older than that do not support 3 digit MNC.
- The mixture of 2 and 3 digit MNCs in the same MCC has been explicitly defined only for the MCC codes 310-316 which have been assigned for the USA.

In general the system implications of implementing the two scenarios proposed by GSM Europe are significant:

1. Mobiles implemented to specifications earlier than R98, which check the validity of any received MNC, will fail these checks if 3 digit MNCs are introduced.
2. Existing Mobile stations and SIM cards would have to be either upgraded or replaced.
3. Existing mobile implementations could potentially have problems with the displaying of 2 and 3 digit MNCs
4. While the cost of incorporating these changes in new equipment is not significant, the cost of upgrading ALL legacy equipment (which is necessary to support roaming) will be significant.
5. The introduction of 3 digit MNC will cause roaming, billing, and service outages if existing equipment (mobile and network) is not upgraded.
6. Any update to existing networks will require the change to be implemented country wide, in all PLMNs, to avoid inconsistent mobile behaviour within the networks.
7. Even though the software changes required are themselves not significant, it is possible that all currently deployed GSM/GPRS equipment will have to be upgraded. This will be a daunting task.

Attached Tdocs:

Tdoc TSG S1 (01)518 (N1-010842), incoming LS from SA1

"N1-010842_LS
IN.zip"

3GPP TSG_CN1, CN2, CN3, CN4 Joint Meeting **Tdoc N1-010888**
Rio Grande, Puerto Rico
16th May, 2001

Source: **CN1 on behalf of Joint CN meeting**

To: TSG SA2

CC: TSGs CN2, CN3, CN4

Title: **Response to LS N1-010504 (S2-010798r2)**

Agenda item: **8.1**

Document for: **LS OUT**

Contact: Magnus Olsson (Ericsson)
Magnus.Olsson@era.ericsson.se

3GPP CN1-4 thanks SA2 for their recent response to a liaison on the service provisioning interface in N1-010504 (S2-010798r2). The text of this liaison stated that a single standardized protocol be supported by the S-CSCF for service control. CN1-4 would like to propose an alternative name for this interface between the S-CSCF and the AS. Currently this interface is labelled “SIP+”. The name SIP+ is considered confusing, particularly given its previous use for the SIP-T work within the IETF. CN1-4 suggests an alternative name be accepted for this interface: IM Service Control or ISC. Note that this is not suggested as the name of the protocol to be used over this interface, merely for the interface itself. CN1-4 requests that an appropriate alternative name be adopted instead of “SIP+” for this interface within TS 23.228. ISC is one such possible alternative.

3GPP TSG CN WG1 Meeting #17 **Tdoc N1-010892**
Puerto Rico, 14th - 18th May 2001

Title: **Liaison Statement on " Handling of ICMP messages by 3GPP SIP Implementations"**

Source: **TSG CN WG1**

To: **TSG SA WG3**

cc:

Contact Person:
Name: **Milo Orsic**
E-mail Address: orsic@lucent.com
Tel. Number: +1 630 713 5161

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1. Overall Description:

At its meeting in Puerto Rico the CN WG1 discussed the handling of the ICMP messages in 3GPP SIP implementations. It was pointed out that the SIP - as defined in RFC 2543 - recommends that some ICMP messages acquire equivalent status as SIP response messages. Hence, the respective ICMP messages become an integral part of SIP.

In the IP networks, any entity is a legitimate source of ICMP messages. Hence, the ICMP system is intrinsically a non-secure mechanism. Therefore, the handling of ICMP messages - as recommended by SIP (RFC 2543) - represent a security problem for 3GPP implementations.

The 3GPP Security Requirements states that protecting the core network signalling protocols is a clear architectural requirement for 3G systems. The security association between the SIP-message sender and the receiver has to be established and all SIP traffic has to be protected by some security mechanism.

2. Action:

The CN WG1 is kindly requesting the guidance from the SA WG3 pertaining to handling of the ICMP messages in 3GPP SIP implementations. The question posed to the SA WG3 is: "Should the 3GPP SIP implementation handle the ICMP messages as recommended in RFC 2543, or should they be ignored by the 3GPP SIP implementation?"

3. Date of Next CN1 Meetings:

CN1#19	10th – 12th July 2001 Dresden, Germany.
CN1#20	27th – 31th August 2001.

Thank you very much for your kind attention to this matter. The CN WG1 is looking forward toward our future collaboration.

3GPP TSG CN WG1 Meeting #17
Puerto Rico, 14th - 18th May 2001

Tdoc N1-010~~798~~902

Title: **Response LS on Introduction of release information in the MS Radio Access Capability IE in 3GPP TS 24.008**

Reference LS **N1-010687**
(If available)

Source: **TSG CN1**

TO : **TSG GERAN WG2**

Cc:

Contact Person:

Name: Eiko Kato
E-mail Address: eiko.kato@ecs.ericsson.se
Tel. Number: +46-46-231295

Attachments:
(Please list documents numbers to be attached)

Date: **14.5.2001**

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1. Overall Description:

CN1 was asked by GERAN WG2 to endorse the introduction of a release marker in the MS Radio Access Capability IE, specified in the 3GPP TS 24.008. The LS from GERAN WG2 can be found in N1-010687.

There is a general principle agreed within CN1 that we do not want to have release indication/negotiation over the air interface. The only case when that has been done so far is to distinguish between 3G and 2G mobiles. To introduce a release marker also for other cases would be a major change and against the general principle mentioned above. Therefore CN1 regrettably cannot endorse the introduction of a release marker in the MS Radio Access Capability IE.

CN1 thanks GERAN WG2 for the proposed corrections to the identified coding errors and we will take these into consideration.

2. Actions:

3. Date of Next CN1 Meetings:

CN1_18	10th – 12th July 2001	Dresden, Germany.
CN1_19	27th – 31th August 2001	

3GPP TSG CN WG1 Meeting #17 **Tdoc N1-010910**
Puerto Rico, 14th - 18th May 2001

Title: **Liaison Statement on THRESHOLD check at RRC connection establishment**

Source: TSG_CN WG1

To: TSG_RAN WG2, TSG-SA WG3

cc: TSG-T WG3

Contact Person:
Name: **Francesco Pica**
E-mail Address: **francesco.pica@eml.ericsson.se**
Tel. Number: +447771774995

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1. Overall Description:

TSG CN WG1 thank TSG RAN WG2 for their LS *R2-010981* (N1-010766) dated 9th – 13th April 2001 on THRESHOLD check at RRC connection establishment

CN1 would like to confirm that the RAN2 assumptions in the scenario pointed out in the input LS, regarding the handling of Threshold value at RRC connection establishment, are correct from CN1 point of view.
Furthermore, CN1 do not see any need to change CN1 specifications. Instead, changes to 33.102 would be needed, so SA3 are kindly asked to present the relevant CRs to their specs.

3. Date of Next CN1 Meetings:

CN1_18	10th – 12th July 2001	Dresden, Germany.
CN1_19	27th – 31th August 2001	

3GPP TSG CN WG1 Meeting #17 **Tdoc N1-010870**
Puerto Rico, 14th - 18th May 2001

Title: **LS on Priority Selection Criteria of Calls in a Multicall**

Source: **TSG CN WG1**

To: **TSG RAN WG3**

Cc: **TSG SA WG1**

Contact Person:

Name: **Robert Zaus**
E-mail Address: **robert.zaus@icn.siemens.de**
Tel. Number: **+49 89 722 26899**

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1. Overall Description:

During their ad hoc meeting on “old stuff up to R99”, 8-9 May, and the CN1 meeting #17, 14-18 May, CN1 discussed a service requirement from SA1 concerning the selection criteria of calls in a multicall which have to be applied when it is not possible to handover all bearers belonging to a multicall.

This situation may occur in case of UMTS to GSM inter-system handover, in case of the basic inter-MSC relocation if 3G_MSC-B does not support multicall or cannot support the number of bearers requested by 3G_MSC-A, or in case of a lack of radio resources in the UMTS target cell.

According to the Multicall specification, TS 22.135, the handover requirements for multicall are specified in 3GPP TS 22.129. The current version 3.5.0 of these requirements specifies that the calls have to be selected for handover in the following order of priority:

i. The call of teleservice emergency call
ii. The call of teleservice telephony
iii. The call of any other type

According to the current version of TS 23.009, 3G_MSC-A and 3G_MSC-B have to base their decision on “the priority level as defined in RAB parameters in 25.413” (i.e. the allocation/retention priority). However, the priority field is optional in TS 25.413 and therefore may not always be available. Besides, it was claimed by one delegation that in some countries the regulator explicitly forbids the allocation of priorities to calls.

CN1 discussed two different proposals to align TS 23.009 with the requirements in TS 22.129, but could not reach an agreement. Since CN1 thinks that RAN3 is also affected by the service requirements from SA1 and by the two proposed solutions, CN1 would like to ask RAN3 for a decision which of the alternatives should be selected.

To speed up the process, CN1 conditionally agreed two alternative sets of CRs for R99 and Rel-4. CN1 kindly asks RAN3 to make a decision between these two proposals during next week’s RAN3 meeting and inform CN1 immediately about the outcome so that CN1 can forward one of the sets of CRs for approval to the CN plenary #12.

The annex of this liaison statement tries to summarise the discussion in CN1 and is intended to aid RAN3 in finding their decision. CN1 kindly asks RAN3 to answer also the questions included in the annex.

2. Actions:

To RAN WG3.

ACTION: CN1 asks RAN3 to decide which of the two solutions in the attached CRs should be chosen, and to inform CN1 immediately about their decision.

3. Date of Next CN Meetings:

CN_12 plenary 13th – 15th June 2001 Stockholm, Sweden

4. Attachments:

N1-010816 [CR 28r1 on TS 23.009].

N1-010826 [CR 38 on TS 23.009].

Annex:

1. Alternative Solutions:

Two proposals were discussed in CN1.

Proposal A (N1-010816)**:** The selection criteria in TS 23.009 shall be based on the criteria specified by SA1 in TS 22.129, which should be modified to take into account the priority levels for non-speech calls when applicable. This implies that the anchor and target 3G_MSC must be aware if the call is a speech call, which is necessary to be able to apply the criteria from TS 22.129.

Proposal B (N1-010826)**:** The allocation/retention priority shall be used to implement the requirement from SA1 in the stage 2 specification, TS 23.009. This implies that a 3G_MSC-A supporting multicall shall allocate priority levels for all bearers, and that it shall do so in such a way that the criteria from TS 22.129 are met. (In the current version of TS 25.413 the priority is an optional parameter.)

2. Discussion of the two proposals:

2.1 Proposal A

With regard to proposal A the discussion concentrated on the issues whether all the necessary information is available at 3G_MSC-B, and how it can be ensured that the requirements from TS 22.129 are fulfilled also during MSC-internal relocation.

1) In section 4.4.1 of TS 23.009, the following has been specified for 3G_MSC-B:

If 3G_MSC-B supports the optional supplementary service Multicall (See TS 23.135) and UE is engaged with multiple bearers the following description applies;

- In the basic relocation case, the 3G_MSC-B shall be able to allocate an Handover Number for each bearer. The 3G-MSC-B shall also be able to select some bearers so that the number of bearers will fulfill the maximum number of bearers supported by the 3G_MSC-B.

Note that for this selection 3G_MSC-B shall apply the selection criteria as specified in TS 22.129, although this is not mentioned explicitly.

Since the priority in the RANAP message Relocation Request is optional, and the MAP parameter Radio Resource Information (=BSSMAP Channel Type) may be missing from the MAP_Prepare_Handover request, it may be necessary for 3G_MSC-B to base its decision only on the RAB parameters. It was pointed out that the parameter Source Statistics Descriptor could be used, since it indicates whether the call is a speech call or not (note that the RNC can also use this information if needed).

According to TS 25.413 the Source Statistics Descriptor is a conditional parameter included in the RAB parameters and is specified as follows:

>Source Statistics Descriptor	C-iftrafficConv-Stream		ENUMERATED (speech, unknown, …)	**Desc.:** This IE specifies characteristics of the source of submitted SDUs **Usage:** -

IftrafficConv-Stream	This IE is only present when traffic class indicates "Conversational" or "Streaming"

There were different opinions within CN1 whether the Source Statistics Descriptor can be used to unequivocally identify whether a call is a speech call. CN1 would like to ask RAN3 for guidance:

Q1: Can we base the decision whether a call is a speech call on the Source Statistics Descriptor?

2) In section 4.3.1 of TS 23.009, the following has been specified for 3G_MSC-A:

If 3G_MSC-A supports the optional supplementary service Multicall (See TS 23.135) and UE is engaged with multiple bearers the following description applies;

- In the Intra-3G_MSC relocation case, the 3G-MSC-A tries to relocate all bearers to a new RNS.

A similar description applies to 3G_MSC-B for the case of subsequent Intra-3G_MSC-B relocation.

Q2: Is it possible in these situations that the target RNC will establish only some of the bearers requested by the MSC, e.g. for reasons of lack of resources?

If yes, how can the requirements from TS 22.129 be fulfilled, since it specifies requirements to the network, not just to the MSC? As the Source Statistics Descriptor is available also to the target RNC, the RNC could

1. follow the criteria specified in TS 22.129 for the selection of bearers in a Multicall;
2. or it could still use the priority field if available. (In this case, however, there is a risk of dropping an emergency call, if for example the priority field is not included for the emergency call but for a parallel data call and if there is congestion in RNC).

With regard to alternative 1: note that although in release 99 and release 4 there can be only one speech call in a multicall, TS 22.129 already specifies requirements for the “Support of Multicall with Simultaneous Voice Calls”. If one day it were possible to have more than one speech call in a multicall, 3G_MSC-B and the RNC would not only need to discriminate between speech calls and data calls, but also between speech calls and emergency (speech) calls.

With regard to alternative 2: if the RNC is not able to establish all the bearers requested by the MSC and the choice of bearers made by the RNC does not comply with the requirements from TS 22.129, the MSC could abort the first resource allocation and send a second Relocation Request for a subset of bearers. (I.e. in this case the actual selection of the calls would be made by the MSC). If such a procedure is not supported by RANAP (and the RNC does not support alternative 1), it cannot be guaranteed that the requirements from TS 22.129 are fulfilled if the RNC makes the selection.

Q3: Is such a procedure (repetition of the resource allocation as outlined above) supported by the current version of TS 25.413?

2.2 Proposal B

Concerning proposal B the discussion concentrated on the issue that with this proposal the allocation/ retention priority would become a conditional instead of an optional information element in RANAP.

It was noted that it may be sufficient to make the priority conditional in the procedural description, but not in the encoding of the message.

During the discussion one delegation commented: " Even if the specifications would be changed so that the priority is mandatory for multicall in TS 25.413, it would not guarantee that an emergency call can always be handed over, since one RNC is handling a big amount of calls for several subscribers and not all will use multicall (and mandatory priority). Therefore the RNC must be able to handle channel allocation for the bearers that have priority and bearers that do not have priority. It does not really matter if a bearer is for multicall or not. They are just bearers. The same principle should be used in case of multicall if not all bearers have priority.”

The answer to this comment was that TS 22.129 only specifies requirements for priorities between the different calls belonging to the same multicall. Priorities between calls belonging to different subscribers, which might e.g. trigger pre-emption, are a different issue.

3GPP TSG-CN WG1 Meeting #17
Puerto Rico, 14th - 18th May 2001

Tdoc N1-010816

R-Form-v3

CHANGE REQUEST

⌘ **23.009 CR**	028	⌘ rev	**1**	⌘ Current version:	3.6.0	⌘

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

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

Title: ⌘	Priority selection criteria of calls in a multicall		
Source: ⌘	Nokia, Alcatel		
Work item code: ⌘	Multicall	***Date:*** ⌘	08.05.01
Category: ⌘	**F**	***Release:*** ⌘	R99
	Use one of the following categories: ***F*** *(essential correction)* ***A*** *(corresponds to a correction in an earlier release)* ***B*** *(Addition of feature),* ***C*** *(Functional modification of feature)* ***D*** *(Editorial modification)* Detailed explanations of the above categories can be found in 3GPP TR 21.900.		*Use one of the following releases:* *2 (GSM Phase 2)* *R96 (Release 1996)* *R97 (Release 1997)* *R98 (Release 1998)* *R99 (Release 1999)* *REL-4 (Release 4)* *REL-5 (Release 5)*

Reason for change: ⌘ When a handover of a multicall takes place in some situations only one bearer can be handed over, for example in InterSystem handover to GSM or in basic relocation when the target MSC does not support Multicall service.
In these cases the 3G-MSC shall select one bearer among the various bearers of the multicall.

In 23.009 the following criteria is used for the selection:
the priority level as defined in RAB parameters in 25.413.

This is a problem since the priority field is an optional field in 25.413.

However in 22.129 the following is stated:

5.4 Handover of a Multicall

The handover event can trigger changes to individual calls in any multicall scenario.

It shall be possible to handover all the calls in a multicall configuration.. If the target cell is not able to accommodate all the calls in a multicall configuration, then the calls that are handed over shall be selected in following order:

i. The call of teleservice emergency call
ii. The call of teleservice telephony
iii. The call of any other type

Calls that cannot be handed over will be released.
If no single call can be selected according to the above criteria, handover shall be rejected.
A change in the availability of suitable radio resources may also occur for other reasons in addition to handover.

	Therefore both criterias should be mapped in a way that only consistent priority order can be applied for the selection of calls in the handover of a Multicall.
Summary of change: ⌘	This CR proposes to coordinate both specifications, by refering in the text in 23.009 in chapters and 4.3.1 and 4.4.1 to the selection criteria defined in TS 22.129.
Consequences if not approved: ⌘	If not approved this may lead to a situation when different selection criteria are applied in different MSCs (for instance in 3G-MSC A and 3G-MSC B) when selecting one bearer to be handed over in a Multicall situation.

Clauses affected: ⌘	4.3.1 and 4.4.1		
Other specs Affected: ⌘	X	Other core specifications	⌘ 22.129
		Test specifications	
		O&M Specifications	
Other comments: ⌘			

********************** First Modified Sections ****************************

4.3 3G_MSC-A

For roles and functional composition of the 3G_MSC-A working as pure GSM MSC, please see previous clause ("MSC-A").

4.3.1 Role of 3G_MSC-A

In the Intra-3G_MSC handover/relocation case, the 3G_MSC-A (simply termed 3G_MSC) controls the call, the mobility management and the radio resources before, during and after an Intra-3G_MSC handover/relocation. When RANAP or BSSMAP procedures have to be performed, they are initiated and driven by 3G_MSC-A.
In the case of an inter-system, intra-MSC handover of a speech call, 3G_MSC-A controls the transcoder in the core network. The 3G_MSC-A determines if a transcoder is required to be inserted or released in the CN.
In case of ATM network between 3G_MSC-A and 3G_MSC-B, 3G_MSC-A retains control of transcoder. In the case of TDM between 3G_MSC-A and 3G_MSC-B, 3G_MSC-A assumes G.711 [16] coding on the TDM E-interface. In case of UMTS to GSM handover, 3G_MSC-A assumes G.711 [16] coding on the ATM E-interface.
In the Inter-3G_MSC relocation case, 3G_MSC-A is the 3G_MSC that controls the call and the mobility management of the UE during the call, before, during and after a basic or subsequent relocation. When RANAP procedures related to dedicated resources have to be performed towards the UE, they are initiated and driven by 3G_MSC-A. The 3G_MSC-A - 3G_MSC-B interface works as a 3G_MSC - RNS interface for the RANAP procedures. The Direct Transfer signalling is relayed transparently by 3G_MSC-B between 3G_MSC-A and the UE.
During a basic relocation, 3G_MSC-A initiates and controls all the relocation procedure, from its initiation (reception of Relocation Required from RNS-A on Iu-interface) until its completion (reception of Relocation Complete from 3G_MSC-B on E-interface).
During a subsequent relocation back to 3G_MSC-A, 3G_MSC-A acts as an RNS towards 3G_MSC-B, which controls the relocation procedure until the termination in 3G_MSC-A of the handover radio resources allocation (sending of the Relocation Request Acknowledge to 3G_MSC-B from 3G_MSC-A). Then all relocation related messages shall terminate at 3G_MSC-A (e.g. Relocation Detect/Complete from RNS-B, Relocation Cancel from RNS-A).
During a subsequent relocation to a third 3G_MSC, 3G_MSC-A works towards 3G_MSC-B' as described above in the basic relocation paragraph and towards 3G_MSC-B as described above in subsequent relocation paragraph.
In the Inter-System, inter-3G_MSC handover case, 3G_MSC-A is the 3G_MSC which controls the call and the mobility management of the UE/MS during the call, before, during and after a basic or subsequent inter-system handover. When BSSAP procedures related to dedicated resources have to be performed towards the UE/MS, they are initiated and driven by 3G_MSC-A. The 3G_MSC-A – MSC-B interface works as a 3G_MSC – BSS interface for a subset of BSSMAP procedures. These BSSMAP procedures described in GSM 09 08 are those related to dedicated resources. The DTAP signalling is relayed transparently by MSC-B between 3G_MSC-A and the UE/MS.
During a basic inter-system UMTS to GSM handover, 3G_MSC-A initiates and controls all the handover procedure, from its initiation (reception of Relocation Required from RNS-A on Iu-interface) until its completion (reception of Handover Complete from MSC-B on E-interface).
During a subsequent inter-system UMTS to GSM handover back to 3G_MSC-A, 3G_MSC-A acts as a BSS towards 3G_MSC-B, which controls the handover procedure until the termination in 3G_MSC-A of the handover radio resources allocation (sending of the Handover Request Acknowledge to 3G_MSC-B from 3G_MSC-A). Then all handover related messages shall terminate at 3G_MSC-A (e.g. Handover Detect/Complete from BSS-B, Relocation Cancel from RNS-A).
During a subsequent inter-system UMTS to GSM handover to a third 3G_MSC, 3G_MSC-A works towards MSC-B' as described above in the basic inter-system handover paragraph and towards 3G_MSC-B as described above in subsequent inter-system handover paragraph.
During a basic inter-system GSM to UMTS handover, 3G_MSC-A initiates and controls all the handover procedure, from its initiation (reception of Handover Required from BSS-A on A-interface) until its completion (reception of Handover Complete from 3G_MSC-B on E-interface).
During a subsequent inter-system GSM to UMTS handover back to 3G_MSC-A, 3G_MSC-A acts as an RNS towards MSC-B, which controls the handover procedure until the termination in 3G_MSC-A of the handover radio resources allocation (sending of the Handover Request Acknowledge to MSC-B from 3G_MSC-A). Then all handover related messages shall terminate at 3G_MSC-A (e.g. Relocation Detect/Complete from RNS-B, Handover Failure from BSS-A).
During a subsequent inter-system GSM to UMTS handover to a third 3G_MSC, 3G_MSC-A works towards 3G_MSC-B' as described above in the basic inter-system handover paragraph and towards MSC-B as described above in subsequent inter-system handover paragraph.

If 3G_MSC-A supports the optional supplementary service Multicall (See TS 23.135) and UE is engaged with multiple bearers the following description applies;

- In the Intra-3G_MSC relocation case, the 3G-MSC-A tries to relocate all bearers to a new RNS.
- In the basic relocation case, the 3G-MSC-A tries to relocate all bearers to 3G_MSC-B. If 3G_MSC-A receives an indication that the 3G_MSC-B does not support multiple bearers, then 3G_MSC-A shall be able to select one bearer to be

handed over according to the ~~priority level defined as RAB parameters in 3GPP TS 25.413 [11]~~ selection criteria defined in 3GPP TS 22.129 [9]and tries again to relocate the selected bearer.

- In the subsequent relocation to a third 3G_MSC-B' case, the 3G-MSC-A tries to relocate all bearers to 3G_MSC-B'. If 3G_MSC-A receives an indication that the 3G_MSC-B' does not support multiple bearers, then 3G_MSC-A shall be able to select one bearer to be handed over according to the ~~priority level defined as RAB parameters in 3GPP TS 25.413 [11]~~ selection criteria defined in 3GPP TS 22.129 [9]and tries again to relocate the selected bearer.
- In the Intra-3G_MSC inter-system UMTS to GSM handover case and the basic inter-system UMTS to GSM handover case, the 3G_MSC-A shall be able to select one bearer to be handed over according to the ~~priority level defined as RAB parameters in 3GPP TS 25.413 [11]~~ selection criteria defined in 3GPP TS 22.129 [9]and tries to handover the selected bearer.
- In all cases described above, 3G_MSC-A shall release some calls which has been carried by the bearers failed to set up in new RNS or the bearers not to be handed over.

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

4.4 3G_MSC-B

For roles and functional composition of the 3G_MSC-B working as pure GSM MSC, please see previous clause ("MSC-B").

4.4.1 Role of 3G_MSC-B

In the Intra-3G_MSC handover/relocation case, the 3G_MSC-B keeps the control of the whole Intra-3G_MSC handover/relocation procedure.

In case of TDM networks, the role of 3G_MSC-B is also to provide transcoderresources. In the case of ATM, 3G_MSC-B has no transcoder handling.

In the Inter-3G_MSC relocation case, the role of 3G_MSC-B (3G_MSC-B') is only to provide radio resources control within its area. This means that 3G_MSC-B keeps control of the radio resources connection and release towards RNS-B. 3G_MSC-B will do some processing on the RANAP information received on the E-interface or the RANAP information received on the Iu-interface whereas it will relay the Direct Transfer information transparently between Iu-interface and E-interface. 3G_MSC-A initiates and drives RANAP procedures towards 3G_MSC-B, while 3G_MSC-B controls them towards its RNSs to the extent that 3G_MSC-B is responsible for the connections of its RNSs. The release of the dedicated resources between 3G_MSC-B and RNS-B is under the responsibility of 3G_MSC-B and RNS-B, and is not directly controlled by 3G_MSC-A. When clearing is to be performed due to information received from RNS-B, 3G_MSC-B shall transfer this clearing indication to 3G_MSC-A, to clear its connection with RNS-B, to terminate the dialogue with 3G_MSC-A through the E-interface, and to release its circuit connection with 3G_MSC-A, if any. In the same way, the release of the connection to its RNS-B, is initiated by 3G_MSC-B, when the dialogue with 3G_MSC-A ends normally and a release is received from the circuit connection with 3G_MSC-A, if any, or when the dialogue with the 3G_MSC-A ends abnormally.

When a release is received by 3G_MSC-B for the circuit connection with 3G_MSC-A then 3G_MSC-B shall release the circuit connection.

In the Inter-system UMTS to GSM Inter-3G_MSC handover case, the role of 3G_MSC-B (3G_MSC-B') is only to provide radio resources control within its area. This means that 3G_MSC-B keeps control of the radio resources connection and release towards BSS-B. 3G_MSC-B will do some processing on the BSSMAP information received on the E-interface or the BSSMAP information received on the A-interface whereas it will relay the DTAP information transparently between A-interface and E-interface. 3G_MSC-A initiates and drives a subset of BSSMAP procedures towards 3G_MSC-B, while 3G_MSC-B controls them towards its BSSs to the extent that 3G_MSC-B is responsible for the connections of its BSSs. The release of the dedicated resources between 3G_MSC-B and BSS-B is under the responsibility of 3G_MSC-B and BSS-B, and is not directly controlled by 3G_MSC-A. When clearing is to be performed due to information received from BSS-B, 3G_MSC-B shall transfer this clearing indication to 3G_MSC-A, to clear its connection with BSS-B, to terminate the dialogue with 3G_MSC-A through the E-interface, and to release its circuit connection with MSC-A, if any. In the same way, the release of the connection to its BSS-B, is initiated by 3G_MSC-B, when the dialogue with 3G_MSC-A ends normally and a release is received from the circuit connection with 3G_MSC-A, if any, or when the dialogue with the MSC-A ends abnormally.

When a release is received by 3G_MSC-B for the circuit connection with 3G_MSC-A then 3G_MSC-B shall release the circuit connection.

In the Inter-system GSM to UMTS Inter-3G_MSC handover case, the role of 3G_MSC-B (3G_MSC-B') is only to provide radio resources control within its area. This means that 3G_MSC-B keeps control of the radio resources connection and release towards RNS-B. 3G_MSC-B will do some processing on the BSSMAP information received on the E-interface or the RANAP information received on the Iu-interface whereas it will relay the Direct Transfer information transparently between Iu-interface and E-interface. MSC-A initiates and drives a subset of BSSMAP procedures towards 3G_MSC-B, while 3G_MSC-B controls them towards its RNSs to the extent that 3G_MSC-B is responsible for the connections of its RNSs. The release of the dedicated resources between 3G_MSC-B and RNS-B is under the responsibility of 3G_MSC-B and RNS-B, and

is not directly controlled by MSC-A. When clearing is to be performed due to information received from RNS-B, 3G_MSC-B shall transfer this clearing indication to MSC-A, to clear its connection with RNS-B, to terminate the dialogue with MSC-A through the E-interface, and to release its circuit connection with MSC-A, if any. In the same way, the release of the connection to its RNS-B, is initiated by 3G_MSC-B, when the dialogue with MSC-A ends normally and a release is received from the circuit connection with MSC-A, if any, or when the dialogue with the MSC-A ends abnormally.
When a release is received by 3G_MSC-B for the circuit connection with MSC-A then 3G_MSC-B shall release the circuit connection.
If 3G_MSC-B does not support the optional supplementary service Mutlicall (See TS 23.135) and 3G_MSC-A requests to relocate multiple bearers, 3G_MSC-B shall indicate that it does not support multiple bearers to 3G_MSC-A.

If 3G_MSC-B supports the optional supplementary service Multicall (See TS 23.135) and UE is engaged with multiple bearers the following description applies;

- In the basic relocation case, the 3G_MSC-B shall be able to allocate an Handover Number for each bearer. The 3G-MSC-B shall also be able to select some bearers so that the number of bearers will fulfill the maximum number of bearers supported by the 3G_MSC-B.
- In the Intra-3G_MSC relocation case, the 3G-MSC-B tries to relocate all bearers to a new RNS.
- In the subsequent relocation back to the 3G_MSC-A or to a third 3G_MSC-B' case, the 3G-MSC-B tries to request to the 3G_MSC-A to relocate all bearers to the 3G_MSC-A or to the 3G_MSC-B'.
- In the Intra-3G_MSC inter-system UMTS to GSM handover case and the subsequent inter-system UMTS to GSM handover back to the 3G_MSC-A or to a third MSC-B' case, the 3G_MSC-B shall be able to select one bearer to be handed over according to the ~~priority level defined as RAB parameters in 3GPP TS 25.413 [11]~~ selection criteria defined in 3GPP TS 22.129 [9] and tries to handover the selected bearer.

3GPP TSG-CN WG1 Meeting #17 **_Tdoc N1-010826_**
Puerto Rico, 14th - 18th May 2001

CR-Form-v3

CHANGE REQUEST

⌘	**23.009**	**CR**	**38**	⌘	rev	**-**	⌘	Current version:	**3.6.0**	⌘

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

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

Title:	⌘	Priority selection criteria of calls in a multicall		
Source:	⌘	Siemens AG		
Work item code:	⌘	Multicall	***Date:*** ⌘	14.05.01
Category:	⌘	**F**	***Release:*** ⌘	R99

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

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

Reason for change: ⌘ In case of handover of a multicall it may not be possible to handover all bearers belonging to the multicall, e.g. in case of UMTS to GSM inter-system handover, in case of inter-MSC relocation if 3G_MSC-B does not support multicall or cannot support the number of bearers requested by 3G_MSC-A, or in case of a lack of radio resources in the UMTS target cell.

In these cases the 3G_MSC-A or 3G_MSC-B shall select one or several bearers to be handed over according to the selection criteria specified in TS 22.129:

5.4 Handover of a Multicall

The handover event can trigger changes to individual calls in any multicall scenario.

It shall be possible to handover all the calls in a multicall configuration.. If the target cell is not able to accommodate all the calls in a multicall configuration, then the calls that are handed over shall be selected in following order:

i. The call of teleservice emergency call

ii. The call of teleservice telephony

iii. The call of any other type

Calls that cannot be handed over will be released.

If no single call can be selected according to the above criteria, handover shall be rejected.

A change in the availability of suitable radio resources may also occur for other reasons in addition to handover.

However, in TS 23.009 the following criterion is used for the selection:

	the priority level as defined in RAB parameters in 25.413. and currently the priority is an optional field in 25.413. In order to align both specifications, it needs to be specified that during RAB assignment and relocation request a 3G_MSC-A supporting multicall shall always assign priorities, and that the priorities shall be assigned in such a way that the requirements from TS 22.129 are fulfilled automatically.
Summary of change: ⌘	A 3G_MSC-A supporting multicall shall always assign priorities during RAB assignment and relocation request. The priorities shall be assigned in such a way that the that the requirements from TS 22.129 are fulfilled.
Consequences if not approved: ⌘	If not approved this may lead to a situation when different selection criteria are applied in different MSCs (for instance in 3G-MSC A and 3G-MSC B), and also different criteria are applied in the MSC and the RNC when selecting which bearers of a multicall will be handed over.

Clauses affected: ⌘	4.3.1		
Other specs affected: ⌘	☐	Other core specifications ⌘	
	☐	Test specifications	
	☐	O&M Specifications	
Other comments: ⌘			

How to create CRs using this form:

Comprehensive information and tips about how to create CRs can be found at: http://www.3gpp.org/3G_Specs/CRs.htm. Below is a brief summary:

1) Fill out the above form. The symbols above marked ⌘ contain pop-up help information about the field that they are closest to.

2) Obtain the latest version for the release of the specification to which the change is proposed. Use the MS Word "revision marks" feature (also known as "track changes") when making the changes. All 3GPP specifications can be downloaded from the 3GPP server under ftp://www.3gpp.org/specs/ For the latest version, look for the directory name with the latest date e.g. 2000-09 contains the specifications resulting from the September 2000 TSG meetings.

3) With "track changes" disabled, paste the entire CR form (use CTRL-A to select it) into the specification just in front of the clause containing the first piece of changed text. Delete those parts of the specification which are not relevant to the change request.

4.3 3G_MSC-A

For roles and functional composition of the 3G_MSC-A working as pure GSM MSC, please see previous clause ("MSC-A").

4.3.1 Role of 3G_MSC-A

In the Intra-3G_MSC handover/relocation case, the 3G_MSC-A (simply termed 3G_MSC) controls the call, the mobility management and the radio resources before, during and after an Intra-3G_MSC handover/relocation. When RANAP or BSSMAP procedures have to be performed, they are initiated and driven by 3G_MSC-A.

In the case of intra-MSC handover of a speech call, 3G_MSC-A controls the transcoder in the core network. The 3G_MSC-A determines if a transcoder is required to be inserted or released in the CN.

In the case of Inter-3G_MSC relocation, 3G_MSC-A links out the transcoder.

In the Inter-3G_MSC relocation case, 3G_MSC-A is the 3G_MSC that controls the call and the mobility management of the UE during the call, before, during and after a basic or subsequent relocation. When RANAP procedures related to dedicated resources have to be performed towards the UE, they are initiated and driven by 3G_MSC-A. The 3G_MSC-A - 3G_MSC-B interface works as a 3G_MSC - RNS interface for the RANAP procedures. The Direct Transfer signalling is relayed transparently by 3G_MSC-B between 3G_MSC-A and the UE.

During a basic relocation, 3G_MSC-A initiates and controls all the relocation procedure, from its initiation (reception of Relocation Required from RNS-A on Iu-interface) until its completion (reception of Relocation Complete from 3G_MSC-B on E-interface).

During a subsequent relocation back to 3G_MSC-A, 3G_MSC-A acts as an RNS towards 3G_MSC-B, which controls the relocation procedure until the termination in 3G_MSC-A of the handover radio resources allocation (sending of the Relocation Request Acknowledge to 3G_MSC-B from 3G_MSC-A). Then all relocation related messages shall terminate at 3G_MSC-A (e.g. Relocation Detect/Complete from RNS-B, Relocation Cancel from RNS-A).

During a subsequent relocation to a third 3G_MSC, 3G_MSC-A works towards 3G_MSC-B' as described above in the basic relocation paragraph and towards 3G_MSC-B as described above in subsequent relocation paragraph.

In the Inter-System, inter-3G_MSC handover case, 3G_MSC-A is the 3G_MSC which controls the call and the mobility management of the UE/MS during the call, before, during and after a basic or subsequent inter-system handover. When BSSAP procedures related to dedicated resources have to be performed towards the UE/MS, they are initiated and driven by 3G_MSC-A. The 3G_MSC-A – MSC-B interface works as a 3G_MSC – BSS interface for a subset of BSSMAP procedures. These BSSMAP procedures described in 3GPP TS 09 08 [7] are those related to dedicated resources. The DTAP signalling is relayed transparently by MSC-B between 3G_MSC-A and the UE/MS.

During a basic inter-system UMTS to GSM handover, 3G_MSC-A initiates and controls all the handover procedure, from its initiation (reception of Relocation Required from RNS-A on Iu-interface) until its completion (reception of Handover Complete from MSC-B on E-interface).

During a subsequent inter-system UMTS to GSM handover back to 3G_MSC-A, 3G_MSC-A acts as a BSS towards 3G_MSC-B, which controls the handover procedure until the termination in 3G_MSC-A of the handover radio resources allocation (sending of the Handover Request Acknowledge to 3G_MSC-B from 3G_MSC-A). Then all handover related messages shall terminate at 3G_MSC-A (e.g. Handover Detect/Complete from BSS-B, Relocation Cancel from RNS-A).

During a subsequent inter-system UMTS to GSM handover to a third 3G_MSC, 3G_MSC-A works towards MSC-B' as described above in the basic inter-system handover paragraph and towards 3G_MSC-B as described above in subsequent inter-system handover paragraph.

During a basic inter-system GSM to UMTS handover, 3G_MSC-A initiates and controls all the handover procedure, from its initiation (reception of Handover Required from BSS-A on A-interface) until its completion (reception of Handover Complete from 3G_MSC-B on E-interface).

During a subsequent inter-system GSM to UMTS handover back to 3G_MSC-A, 3G_MSC-A acts as an RNS towards MSC-B, which controls the handover procedure until the termination in 3G_MSC-A of the handover radio resources allocation (sending of the Handover Request Acknowledge to MSC-B from 3G_MSC-A). Then all handover related messages shall terminate at 3G_MSC-A (e.g. Relocation Detect/Complete from RNS-B, Handover Failure from BSS-A).

During a subsequent inter-system GSM to UMTS handover to a third 3G_MSC, 3G_MSC-A works towards 3G_MSC-B' as described above in the basic inter-system handover paragraph and towards MSC-B as described above in subsequent inter-system handover paragraph.

If 3G_MSC-A supports the optional supplementary service Multicall (See 3GPP TS 23.135) and UE is engaged with multiple bearers the following description applies;

- During RAB assignment and relocation 3G-MSC-A assigns a priority level defined as RAB parameter in 3GPP TS 25.413 [11] for each bearer. The rules for the assignment of priority levels are implementation dependent. However, the priority levels shall be assigned in such a way that the requirements from 3GPP TS 22.129 [9], subclause "Handover of a Multicall", are fulfilled if 3G_MSC-A selects the bearers to be handed over according to the priority level.

- In the Intra-3G_MSC relocation case, the 3G-MSC-A tries to relocate all bearers to a new RNS.

- In the basic relocation case, the 3G-MSC-A tries to relocate all bearers to 3G_MSC-B. If 3G_MSC-A receives an indication that the 3G_MSC-B does not support multiple bearers, then 3G_MSC-A shall be able to select one bearer to be handed over according to the priority level defined as RAB parameters in 3GPP TS 25.413 and tries again to relocate the selected bearer.

- In the subsequent relocation to a third 3G_MSC-B' case, the 3G-MSC-A tries to relocate all bearers to 3G_MSC-B'. If 3G_MSC-A receives an indication that the 3G_MSC-B' does not support multiple bearers, then 3G_MSC-A shall be able to select one bearer to be handed over according to the priority level defined as RAB parameters in 3GPP TS 25.413 [11] and tries again to relocate the selected bearer.

- In the Intra-3G_MSC inter-system UMTS to GSM handover case and the basic inter-system UMTS to GSM handover case, the 3G_MSC-A shall be able to select one bearer to be handed over according to the priority level defined as RAB parameters in 3GPP TS 25.413 [11] and tries to handover the selected bearer.

- In all cases described above, 3G_MSC-A shall release some calls which has been carried by the bearers failed to set up in new RNS or the bearers not to be handed over.

************** PROVIDED FOR INFORMATION ONLY ****************

4.4 3G_MSC-B

For roles and functional composition of the 3G_MSC-B working as pure GSM MSC, please see previous clause ("MSC-B").

4.4.1 Role of 3G_MSC-B

In the Intra-3G_MSC handover/relocation case, the 3G_MSC-B keeps the control of the whole Intra-3G_MSC handover/relocation procedure. 3G_MSC-B notifies MSC-A or 3G_MSC-A of intra-3G_MSC-B InterSystem handover by using the A_HANDOVER_PERFORMED procedure.

The role of 3G_MSC-B is also to provide transcoder resources.

In the Inter-3G_MSC relocation case, the role of 3G_MSC-B (3G_MSC-B') is only to provide radio resources control within its area. This means that 3G_MSC-B keeps control of the radio resources connection and release towards RNS-B. 3G_MSC-B will do some processing on the RANAP information received on the E-interface or the RANAP information received on the Iu-interface whereas it will relay the Direct Transfer information transparently between Iu-interface and E-interface. 3G_MSC-A initiates and drives RANAP procedures towards 3G_MSC-B, while 3G_MSC-B controls them towards its RNSs to the extent that 3G_MSC-B is responsible for the connections of its RNSs. The release of the dedicated resources between 3G_MSC-B and RNS-B is under the responsibility of

3G_MSC-B and RNS-B, and is not directly controlled by 3G_MSC-A. When clearing is to be performed due to information received from RNS-B, 3G_MSC-B shall transfer this clearing indication to 3G_MSC-A, to clear its connection with RNS-B, to terminate the dialogue with 3G_MSC-A through the E-interface, and to release its circuit connection with 3G_MSC-A, if any. In the same way, the release of the connection to its RNS-B, is initiated by 3G_MSC-B, when the dialogue with 3G_MSC-A ends normally and a release is received from the circuit connection with 3G_MSC-A, if any, or when the dialogue with the 3G_MSC-A ends abnormally.

When a release is received by 3G_MSC-B for the circuit connection with 3G_MSC-A then 3G_MSC-B shall release the circuit connection.

In the Inter-system UMTS to GSM Inter-3G_MSC handover case, the role of 3G_MSC-B (3G_MSC-B') is only to provide radio resources control within its area. This means that 3G_MSC-B keeps control of the radio resources connection and release towards BSS-B. 3G_MSC-B will do some processing on the BSSMAP information received on the E-interface or the BSSMAP information received on the A-interface whereas it will relay the DTAP information transparently between A-interface and E-interface. 3G_MSC-A initiates and drives a subset of BSSMAP procedures towards 3G_MSC-B, while 3G_MSC-B controls them towards its BSSs to the extent that 3G_MSC-B is responsible for the connections of its BSSs. The release of the dedicated resources between 3G_MSC-B and BSS-B is under the responsibility of 3G_MSC-B and BSS-B, and is not directly controlled by 3G_MSC-A. When clearing is to be performed due to information received from BSS-B, 3G_MSC-B shall transfer this clearing indication to 3G_MSC-A, to clear its connection with BSS-B, to terminate the dialogue with 3G_MSC-A through the E-interface, and to release its circuit connection with MSC-A, if any. In the same way, the release of the connection to its BSS-B, is initiated by 3G_MSC-B, when the dialogue with 3G_MSC-A ends normally and a release is received from the circuit connection with 3G_MSC-A, if any, or when the dialogue with the MSC-A ends abnormally.

When a release is received by 3G_MSC-B for the circuit connection with 3G_MSC-A then 3G_MSC-B shall release the circuit connection.

In the Inter-system GSM to UMTS Inter-3G_MSC handover case, the role of 3G_MSC-B (3G_MSC-B') is only to provide radio resources control within its area. This means that 3G_MSC-B keeps control of the radio resources connection and release towards RNS-B. 3G_MSC-B will do some processing on the BSSMAP information received on the E-interface or the RANAP information received on the Iu-interface whereas it will relay the Direct Transfer information transparently between Iu-interface and E-interface. MSC-A initiates and drives a subset of BSSMAP procedures towards 3G_MSC-B, while 3G_MSC-B controls them towards its RNSs to the extent that 3G_MSC-B is responsible for the connections of its RNSs. The release of the dedicated resources between 3G_MSC-B and RNS-B is under the responsibility of 3G_MSC-B and RNS-B, and is not directly controlled by MSC-A. When clearing is to be performed due to information received from RNS-B, 3G_MSC-B shall transfer this clearing indication to MSC-A, to clear its connection with RNS-B, to terminate the dialogue with MSC-A through the E-interface, and to release its circuit connection with MSC-A, if any. In the same way, the release of the connection to its RNS-B, is initiated by 3G_MSC-B, when the dialogue with MSC-A ends normally and a release is received from the circuit connection with MSC-A, if any, or when the dialogue with the MSC-A ends abnormally.

When a release is received by 3G_MSC-B for the circuit connection with MSC-A then 3G_MSC-B shall release the circuit connection.

If 3G_MSC-B does not support the optional supplementary service Mutlicall (See 3GPP TS 23.135) and 3G_MSC-A requests to relocate multiple bearers, 3G_MSC-B shall indicate that it does not support multiple bearers to 3G_MSC-A.

If 3G_MSC-B supports the optional supplementary service Multicall (See 3GPP TS 23.135) and UE is engaged with multiple bearers the following description applies;

- In the basic relocation case, the 3G_MSC-B shall be able to allocate an Handover Number for each bearer. The 3G-MSC-B shall also be able to select some bearers so that the number of bearers will fulfill the maximum number of bearers supported by the 3G_MSC-B.

- In the Intra-3G_MSC relocation case, the 3G-MSC-B tries to relocate all bearers to a new RNS.

- In the subsequent relocation back to the 3G_MSC-A or to a third 3G_MSC-B' case, the 3G-MSC-B tries to request to the 3G_MSC-A to relocate all bearers to the 3G_MSC-A or to the 3G_MSC-B'.

- In the Intra-3G_MSC inter-system UMTS to GSM handover case and the subsequent inter-system UMTS to GSM handover back to the 3G_MSC-A or to a third MSC-B' case, the 3G_MSC-B shall be able to select one bearer to be handed over according to the priority level defined as RAB parameters in 3GPP TS 25.413 [11] and tries to handover the selected bearer.

3GPP TSG CN WG1 Meeting #17 **Tdoc N1-010873**
Puerto Rico, 14th - 18th May 2001

Title: **Liaison Statement on Adding New Definitions to 21.905**

Source: **TSG_CN WG1**

To: **TSG_SA WG1, TSG_SA**

cc: **TSG_SA WG2, GERAN WG2**

Contact Person:
Name: Daniel,Elizabeth Mary
E-mail Address: **lizdaniel@lucent.com**
Tel. Number: +44 179388 3412

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1. Overall Description:

In 23.221 and 24.008, two terms *In Iu mode* and *In A/Gb mode* have been used. The definitions for these two terms has been defined by 23.122 clause 1.2.

The definitions are:

In A/Gb mode: Indicates this paragraph applies only to GSM System. For multi system case this is determined by the current serving radio access network.

In Iu mode: Indicates this paragraph applies only to UMTS System. For multi system case this is determined by the current serving radio access network.

Since both SA2 and CN1 use these two terms CN1 asks SA1 (alternatively to SA directly) to put these definitions to 21.905 for R99 and Rel-4.

It can be foreseen that the definitions will need to change for Rel-5 but the requirements are still open and the discussion is ongoing between GERAN 2 and CN1.

2. Attachments:

N1-010xxx_S2-010xxx_newdefinitionR99.doc

N1-010xxx_S2-010xxx_newdefinitionR4.doc

3GPP TSG-SA2 Meeting #17
14-19 May 2001, Puerto Rico, US

Tdoc S2-010xxx

CR-Form-v3

CHANGE REQUEST

⌘	**21.905**	**CR**		⌘ rev	**00**	⌘	Current version:	**4.2.0** ⌘

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

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

Title:	⌘	Adding new definitions to 21.905 for In Iu mode and In A/Gb mode		
Source:	⌘	Lucent Technologies		
Work item code:	⌘	TEI	***Date:*** ⌘	2001-05-08
Category:	⌘	**A**	***Release:*** ⌘	REL-4
		Use one of the following categories: ***F*** *(essential correction)* ***A*** *(corresponds to a correction in an earlier release)* ***B*** *(Addition of feature),* ***C*** *(Functional modification of feature)* ***D*** *(Editorial modification)* Detailed explanations of the above categories can be found in 3GPP TR 21.900.		*Use one of the following releases:* *2 (GSM Phase 2)* *R96 (Release 1996)* *R97 (Release 1997)* *R98 (Release 1998)* *R99 (Release 1999)* *REL-4 (Release 4)* *REL-5 (Release 5)*

Reason for change:	⌘	Two terms needs to be defined in 23.221and the terms are already used in other specifications
Summary of change:	⌘	Add two definitions In Iu mode and In A/Gb mode to 21.905
Consequences if not approved:	⌘	No definitions for two terms in 21.905

Clauses affected:	⌘	3		
Other specs affected:	⌘	[ ] Other core specifications [ ] Test specifications [ ] O&M Specifications	⌘	
Other comments:	⌘			

I

IC Card: A card holding an Integrated Circuit containing subscriber, end user, authentication and/or application data for one or more applications.

IC card SIM: Obsolete term for ID-1 SIM.

ID-1 SIM: The SIM having the format of an ID-1 card (see ISO 7816-1 [24]).

Idle mode: The state of UE switched on but which does not have any established RRC connection.

Implementation capability: A capability that relates to a particular technical domain. Examples: a spreading factor of 128 (in the domain of the physical layer); the A5 algorithm; a 64 bit key length (in the domain of security); a power output of 21 dBm (in the domain of transmitter performance); support of AMR Codec (in the domain of the Codec); support of CHV1 (in the domain of the USIM).

In A/Gb mode: Indicates this paragraph applies only to GSM System. For multi system case this is determined by the current serving radio access network.

Information Data Rate: Rate of the user information, which must be transmitted over the Air Interface. For example, output rate of the voice codec.

Initial paging information: This information indicates if the UE needs to continue to read more paging information and eventually receive a page message.

Initial paging occasion: The paging occasion the UE uses as starting point for its paging DRX cycle.

In Iu mode: Indicates this paragraph applies only to UMTS System. For multi system case this is determined by the current serving radio access network.

Integrity: (in the context of security) The avoidance of unauthorised modification of information.

Inter-cell handover: A handover between different cells. An inter-cell handover requires network connections to be altered.

Inter PLMN handover: Handover between different PLMNs, ie having different MCC-MNC.

Inter system handover: Handover between networks using different radiosystems , e.g. UMTS – GSM.

Interactive service: A service which provides the means for bi-directional exchange of information between users. Interactive services are divided into three classes of services: conversational services, messaging services and retrieval services (source: ITU-T I.113).

Interface: The common boundary between two associated systems (source: GSM 01.04, ITU-T I.112).

International Mobile Station Equipment Identity (IMEI): An "International Mobile Station Equipment Identity" is a unique number which shall be allocated to each individual mobile station equipment in the PLMN and shall be unconditionally implemented by the MS manufacturer.

International mobile user number (IMUN): The International Mobile User Number is a diallable number allocated to a UMTS user.

Interference Signal Code Power (ISCP): Given only interference power is received, the average power of the received signal after despreading and combining.

Intra-cell handover: A handover within one sector or between different sectors of the same cell. An intra-cell handover does not require network connections to be altered.

Intra PLMN handover: Handover within the same network, ie having the same MCC-MNC regardless of radio access system. Note: this includes the case of UMTS <>GSM handover where MCC-MNC are the same in both cases.

IRP Information Model: An IRP Information Model consists of an IRP Information Service and a Network Resource Model (see below for definitions of IRP Information Service and Network Resource Model).

IRP Information Service: An IRP Information Service describes the information flow and support objects for a certain functional area, e.g. the alarm information service in the fault management area. As an example of support objects, for the Alarm IRP there is the alarm record and alarm list.

IRP Solution Set: An IRP Solution Set is a mapping of the IRP Information Service to one of several technologies (CORBA/IDL, SNMP/SMI, CMIP/GDMO, etc.). An IRP Information Service can be mapped to several different IRP Solution Sets. Different technology selections may be done for different IRPs.

Inter System Change: a change of radio access between different radio access technologies such as GSM and UMTS.

Iu: Interconnection point between an RNC and a Core Network. It is also considered as a reference point.

Iub: Interface between an RNC and a Node B.

Iur: A logical interface between two RNC. Whilst logically representing a point to point link between RNC, the physical realisation may not be a point to point link.

3GPP TSG-SA2 Meeting #17
14-19 May 2001, Puerto Rico, US

Tdoc S2-010xxx

CR-Form-v3

CHANGE REQUEST

⌘ **21.905**	**CR**		⌘ rev	**00**	⌘ Current version:	**3.2.0**	⌘

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

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

Title: ⌘	Adding new definitions for 21.905 for Iu mode and A/Gb mode		
Source: ⌘	Lucent Technologies		
Work item code: ⌘	TEI	***Date:*** ⌘	2001-05-09
Category: ⌘	**F**	***Release:*** ⌘	R99
	Use one of the following categories: ***F*** *(essential correction)* ***A*** *(corresponds to a correction in an earlier release)* ***B*** *(Addition of feature),* ***C*** *(Functional modification of feature)* ***D*** *(Editorial modification)* Detailed explanations of the above categories can be found in 3GPP TR 21.900.		*Use one of the following releases:* *2 (GSM Phase 2)* *R96 (Release 1996)* *R97 (Release 1997)* *R98 (Release 1998)* *R99 (Release 1999)* *REL-4 (Release 4)* *REL-5 (Release 5)*

Reason for change: ⌘	Two terms needs to be defined in 23.221and the terms are already used in other specifications
Summary of change: ⌘	Add two definitions In Iu mode and In A/Gb mode to 21.905
Consequences if not approved: ⌘	No definitions for two terms in 21.905

Clauses affected: ⌘	3		
Other specs affected: ⌘	[ ] Other core specifications [ ] Test specifications [ ] O&M Specifications	⌘	
Other comments: ⌘			

I

IC Card: A card holding an Integrated Circuit containing subscriber, end user, authentication and/or application data for one or more applications.

IC card SIM: Obsolete term for ID-1 SIM.

ID-1 SIM: The SIM having the format of an ID-1 card (see ISO 7816-1 [24]).

Idle mode: The state of UE switched on but which does not have any established RRC connection.

Implementation capability: A capability that relates to a particular technical domain. Examples: a spreading factor of 128 (in the domain of the physical layer); the A5 algorithm; a 64 bit key length (in the domain of security); a power output of 21 dBm (in the domain of transmitter performance); support of AMR Codec (in the domain of the Codec); support of CHV1 (in the domain of the USIM).

In A/Gb mode,...: Indicates this paragraph applies only to GSM System. For multi system case this is determined by the current serving radio access network.

Information Data Rate: Rate of the user information, which must be transmitted over the Air Interface. For example, output rate of the voice codec.

Initial paging information: This information indicates if the UE needs to continue to read more paging information and eventually receive a page message.

Initial paging occasion: The paging occasion the UE uses as starting point for its paging DRX cycle.

In Iu mode,...: Indicates this paragraph applies only to UMTS System. For multi system case this is determined by the current serving radio access network.

Integrity: (in the context of security) The avoidance of unauthorised modification of information.

Inter-cell handover: A handover between different cells. An inter-cell handover requires network connections to be altered.

Inter PLMN handover: Handover between different PLMNs, ie having different MCC-MNC.

Inter system handover: Handover between networks using different radiosystems , e.g. UMTS – GSM.

Interactive service: A service which provides the means for bi-directional exchange of information between users. Interactive services are divided into three classes of services: conversational services, messaging services and retrieval services (source: ITU-T I.113).

Interface: The common boundary between two associated systems (source: GSM 01.04, ITU-T I.112).

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3GPP TSG CN WG1 Meeting #17 **Tdoc N1-010706**
Puerto Rico, 14th - 18th May 2001

Source: AT&T

Title: CR for 24.228, Call Transfer Procedures

Agenda item: 8.4

Document for: APPROVAL

It is proposed that the following text be added to an informative Annex (TS 24.228 Annex B) as preliminary material for section 10.5.

In addition, the following IETF Internet-Drafts should be added to the Work Item Description as IETF dependencies:

- Draft-ietf-sip-cc-transfer-04
- Draft-roach-sip-subscribe-notify-03

10.5 Session Transfer Procedures

This section gives information flows for the procedures for performing session transfers. Section 10.5.1 gives the procedures for a transfer that initiates a new session (i.e. to a new destination not previously involved in the session). Section 10.5.2 gives the procedures for a transfer that replaces an existing session (i.e. to a destination that was previously involved in the session).

10.5.1 Session Transfer initiating a new session

An IP multi-media session already exists between UE#1 and UE#2. UE#2 desires UE#1 to initiate a new session to a new destination, UE#F, and terminate the existing session. The procedures for this transfer are shown in Figure 10.5.1-1.

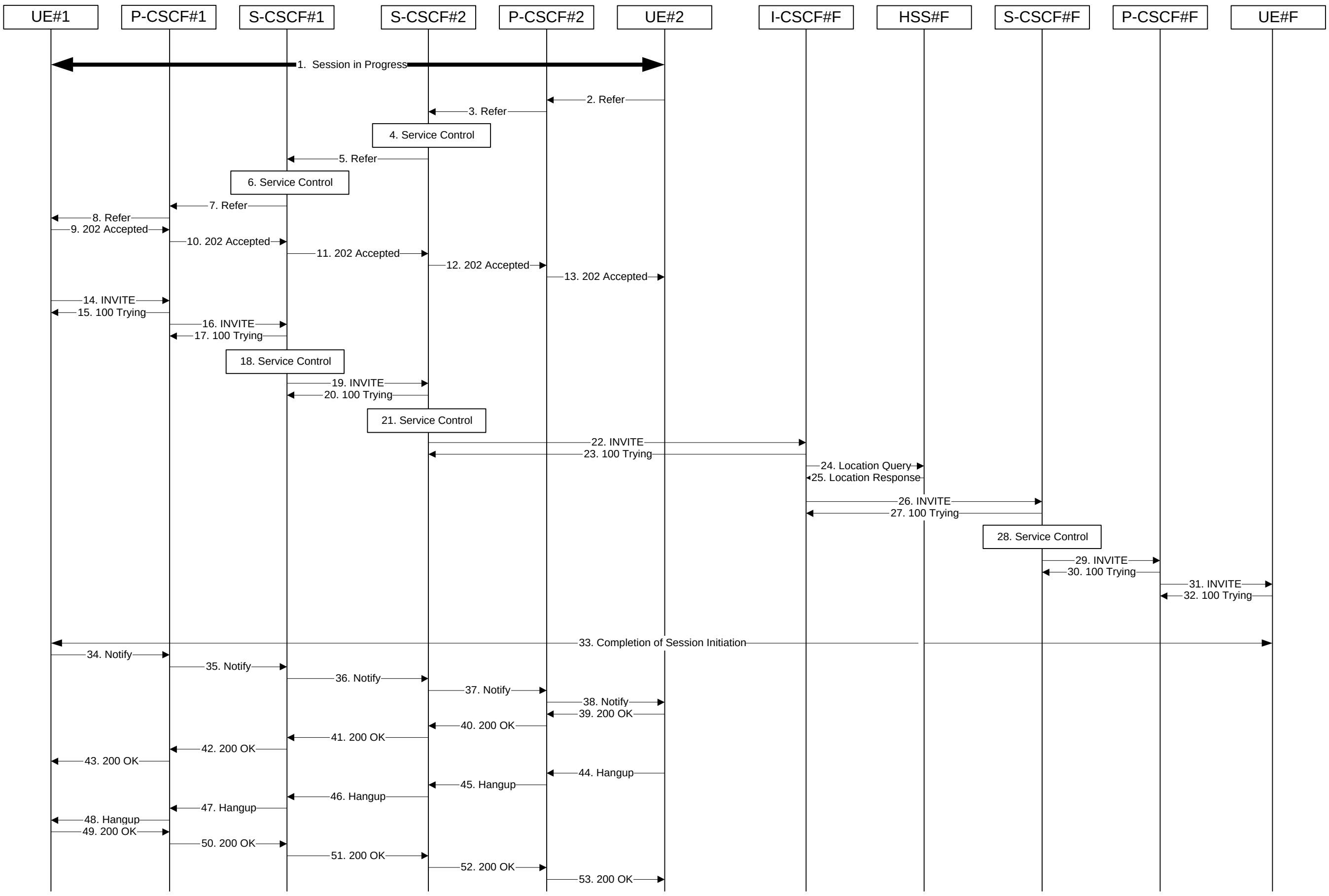

Figure 10.5.1-1 – Session Transfer initiating a new session

1. Session in Progress

UE#1 initiates a multi-media session with UE#2. As a result, the state information stored at P-CSCF#2 is shown in Table 10.5.1-1

Table 10.5.1-1: State Information

```
Request-URI: sip:token6@pcscf2.home.net

From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;
    tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost

Route: sip:scscf2.home.net, sip:scscf1.home.net,
    sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
```

2. **REFER (UE to P-CSCF) – see example in Table 10.5.1-2**

UE#2 sends a Refer request to its proxy, P-CSCF#2.

Table 10.5.1-2: REFER (UE to P-CSCF)

```
REFER sip:token6@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
From: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost;tag=314159
To: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;
    tag=171828
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 130 REFER
Contact: sip:[5555::eee:fff:aaa:bbb]
Refer-To: tel:+1-212-555-3333
Refer-By: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off
Content-length: 0
```

Request-URI: contains the value of the Contact header from the 200-OK response to the initial INVITE.

Via: contains the IP address or FQDN of the originating UE.

From:, To:, Call-ID: contain the values previously used to establish the session, including the tag value from the response.

Cseq: next higher sequential value.

Contact: the IP address or FQDN of the originating UE.

Editor's Note: Use of Remote-Party-ID in REFER is FFS.

Editor's Note: The proper value for the Refer-By header is FFS.

3. **REFER (P-CSCF to S-CSCF) – see example in Table 10.5.1-3**
P-CSCF adds a Route header, with the saved value from the previous 200-OK response. P-CSCF identifies the proper saved value by the Request-URI.
P-CSCF#2 forwards the Refer request to S-CSCF#2.

Table 10.5.1-3: REFER (P-CSCF to S-CSCF)

```
REFER sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Route: sip:scscf1.home.net, sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Refer-To:
Refer-By:
Remote-Party-ID:
Content-length:
```

Request-URI: the first component of the saved Route header.

Route: saved from the 200-OK response to the initial INVITE (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

4. Service Control

5. **REFER (S-CSCF to S-CSCF) – see example in Table 10.5.1-5**
In order to maintain the expectation of privacy of the identity of the new destination, S-CSCF#2 converts the "Refer-To" header into a private URL. S-CSCF#2 forwards the Refer request to S-CSCF#1.

Table 10.5.1-5: REFER (S-CSCF to S-CSCF)

```
REFER sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Route: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Record-Route: sip:scscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Refer-To: sip:token(tel:+1-212-555-3333)@scscf2.home.net;private
Refer-By:
Remote-Party-ID: “John Smith” <tel:+1-212-555-2222>;privacy=off;screen=yes
Content-length:
```

6. Service Control

7. **REFER (S-CSCF to P-CSCF) – see example in Table 10.5.1-7**

S-CSCF#1 forwards the Refer request to P-CSCF#1.

Table 10.5.1-7: REFER (S-CSCF to P-CSCF)

```
INVITE sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UEP scscf1.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net,
    SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route: sip:scscf1.home.net, sip:scscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Refer-To:
Refer-By:
Remote-Party-ID:
Content-length:
```

8. **REFER (P-CSCF to UE) – see example in Table 10.5.1-8**

P-CSCF#1 forwards the Refer request to UE#1.

Table 10.5.1-8: REFER (P-CSCF to UE)

```
REFER sip:[5555::aaa:bbb:ccc:ddd] SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net;branch=token3
From:
To:
Call-ID:
Cseq:
Contact: token3@pcscf2.home.net
Refer-To:
Refer-By:
Remote-Party-ID:
Content-length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

9. **202-Accepted (UE to P-CSCF) – see example in Table 10.5.1-9**

UE#2 acknowledges receipt of the Refer request (8) with a 202-Accepted final response, sent to P-CSCF#1.

Table 10.5.1-8: 202 Accepted (UE to P-CSCF)

```
SIP/2.0 202 Accepted
Via: SIP/2.0/UDP pcscf2.home.net;branch=token3
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

10. **202-Accepted (P-CSCF to S-CSCF) – see example in Table 10.5.1-10**

P-CSCF#1 forwards the 202 Accepted final response to S-CSCF#1.

Table 10.5.1-10: 202 Accepted (P-CSCF to S-CSCF)

```
SIP/2.0 202 Accepted
Via: SIP/2.0/UEP scscf1.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net,
    SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route: sip:scscf1.home.net, sip:scscf2.home.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

P-CSCF restores the Via headers and Record-Route headers from the branch value in its Via.

11. **202-Accepted (S-CSCF to S-CSCF) – see example in Table 10.5.1-11**

S-CSCF#1 forwards the 202 Accepted final response to S-CSCF#2.

Table 10.5.1-11: 202 Accepted (S-CSCF to S-CSCF)

```
SIP/2.0 202 Accepted
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-length:
```

12. **202-Accepted (S-CSCF to P-CSCF) – see example in Table 10.5.1-12**

S-CSCF#2 forwards the 202 Accepted final response to P-CSCF#2.

Table 10.5.1-12: 202 Accepted (S-CSCF to P-CSCF)

```
SIP/2.0 202 Accepted
Via: SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-length:
```

13. **202-Accepted (P-CSCF to UE) – see example in Table 10.5.1-13**

P-CSCF#2 forwards the 202 Accepted final response to UE#2.

Table 10.5.1-13: 202 Accepted (P-CSCF to UE)

```
SIP/2.0 202 Accepted
Via: SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
From:
To:
Call-ID:
CSeq:
Content-length:
```

P-CSCF removes the Record-Route header

.

14. **INVITE (UE to P-CSCF) – see example in Table 10.5.1-14**

UE#1 initiates an INVITE request based on the Refer-To header URL in the REFER request. The INVITE is sent from the UE to P-CSCF#1.

Table 10.5.1-14: INVITE (UE to P-CSCF)

```
INVITE sip:token(tel:+1-212-555-3333)@scscf2.home.net;private SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off
Proxy-Require: privacy
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=74))@localhost>;
    tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=75))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Refer-By: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=audio 3456 RTP/AVP 97 3 96
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

15. **100 Trying (P-CSCF to UE) – see example in Table 10.5.1-15**

P-CSCF#1 responds to the INVITE request (14) with a 100 Trying provisional response.

Table 10.5.1-15: 100 Trying (P-CSCF to UE)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

16. **INVITE (P-CSCF to S-CSCF) – see example in Table 10.5.1-16**

P-CSCF#1 remembers (from the registration procedure) the request routing for this UE. This becomes a Route header in the request. The next hop is the S-CSCF serving this UE. P-CSCF rewrites the Contact header with a locally defined value that identifies the UE. P-CSCF adds itself to the Via header.

Table 10.5.1-16: INVITE (P-CSCF to S-CSCF)

```
INVITE sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:token(tel:+1-212-555-3333)@scscf2.home.net;private
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Refer-By:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

17. **100 Trying (S-CSCF to P-CSCF) – see example in Table 10.5.1-17**

S-CSCF#1 responds to the INVITE request (16) with a 100 Trying provisional response.

Table 10.5.1-17: 100 Trying (S-CSCF to P-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

18. **Service Control**

S-CSCF#1 performs whatever service control logic is appropriate for this call attempt.

19. **INVITE (S-CSCF to S-CSCF) – see example in Table 10.5.1-19**

S-CSCF#1 performs an analysis of the destination address, which is a private URL generated by S-CSCF#2. Since it is a destination within the same operator’s network, S-CSCF#1 forwards the INVITE request directly to S-CSCF#2.

Table 10.5.1-19: INVITE (S-CSCF to S-CSCF)

```
INVITE sip:token(tel:+1-212-555-3333)@scscf2.home.net;private SIP/2.0
Via: SIP/2.0/UDP sip:scscf1.home.net SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf1.home.net
Supported:
Remote-Party-ID: “John Doe” <tel:+1-212-555-1111>;privacy=off;screen=yes
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Refer-By:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

20. **100 Trying (S-CSCF to S-CSCF) – see example in Table 10.5.1-20**

S-CSCF#2 responds to the INVITE request (19) by sending a 100 Trying provisional response to S-CSCF#1.

Table 10.5.1-20: 100 Trying (S-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP  [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

21. **Service Control**

S-CSCF#2 performs whatever service control logic is appropriate for this call transfer attempt.

22. **INVITE (S-CSCF to I-CSCF) – see example in Table 10.5.1-22**

S-CSCF#2 determines the destination address from the private URL contained in the INVITE request. Based on information in that URL, and information saved from step #4 above (implementation decision), S-CSCF#2 verifies the validity of the transfer request, and that it is within a short time delay from the REFER request.

S-CSCF#2 performs an analysis of the destination address, and determines the network operator to whom the destination subscriber belongs. Since (for this example) the forwarding network operator does not desire to keep their internal configuration hidden, S-CSCF#2 forwards the INVITE request directly to I-CSCF#F.

Table 10.5.1-22: INVITE (S-CSCF to I-CSCF)

```
INVITE sip:+1-212-555-3333@home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
Supported:
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off;screen=yes;party=transferor
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Refer-By:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

Editor's Note: Use of "party=transferor" in a separate Remote-Party-ID header is FFS.

23. **100 Trying (I-CSCF to S-CSCF) – see example in Table 10.5.1-23**

I-CSCF#F responds to the INVITE request (22) by sending a 100 Trying provisional response to S-CSCF#2.

Table 10.5.1-23: 100 Trying (I-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

24. **Location Query**

I-CSCF (at the border of the terminating subscriber's network) queries the HSS for current location information. It will send "Cx-location-query" to the HSS to obtain the location information for the destination.

25. **Location Response**

HSS responds with the address of the current Serving-CSCF for the terminating subscriber.

26. **INVITE (I-CSCF to S-CSCF) – see example in Table 10.5.1-26**

I-CSCF#F forwards the INVITE request to the S-CSCF (S-CSCF#F) that will handle the call termination.

Table 10.5.1-26: INVITE (I-CSCF to S-CSCF)

```
INVITE sip:scscff.home.net SIP/2.0
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP
    pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home.net;user=phone
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
Supported:
Remote-Party-ID:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Refer-By:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

NOTE: The I-CSCF does not add itself to the Record-Route header, as it has no need to remain in the signalling path once the session is established.

27. **100 Trying (S-CSCF to I-CSCF) – see example in Table 10.5.1-27**

S-CSCF#F responds to the INVITE request (26) with a 100 Trying provisional response.

Table 10.5.1-27: 100 Trying (S-CSCF to I-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP
    pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

28. **Service Control**

S-CSCF#F performs whatever service control logic is appropriate for this call attempt.

29. **INVITE (S-CSCF to P-CSCF) – see example in Table 10.5.1-29**

S-CSCF#F remembers (from the registration procedure) the next hop CSCF for this UE. It forwards the INVITE request to P-CSCF#F.

Table 10.5.1-29: INVITE (S-CSCF to P-CSCF)

```
INVITE sip:+1-212-555-3333@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscff.home.net, SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP
    scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscff.home.net, sip:scscf2.home.net, sip:scscf1.home.net
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Refer-By:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

30. **100 Trying (P-CSCF to S-CSCF) – see example in Table 10.5.1-30**

P-CSCF#F responds to the INVITE request (29) by sending a 100 Trying provisional response to S-CSCF#F.

Table 10.5.1-12: 100 Trying (P-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscff.home.net, SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP
    scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

31. **INVITE (P-CSCF to UE) – see example in Table 10.5.1-31**

P-CSCF forwards the INVITE request to the UE.

Table 10.5.1-31: INVITE (P-CSCF to UE)

```
INVITE sip:+1-212-555-3333@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscff.home.net;branch=token6
Supported:
Remote-Party-ID:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: token6@pcscff.home.net
Refer-By:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

32. **100 Trying (UE to P-CSCF) – see example in Table 10.5.1-32**

UE#F may optionally send a 100 Trying provisional response to P-CSCF.

Table 10.5.1-32: 100 Trying (UE to P-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pcscf1.home.net;branch=token6
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

33. **Completion of Session Initiation**

UE#1 and UE#F complete the session initiation, as shown in the MO, S-S, and MT procedures.

34. **NOTIFY (UE to P-CSCF) – see example in Table 10.5.1-34**

When the session with UE#F has been successfully established, UE#1 sends a Notify request to its proxy, P-CSCF#1.

Table 10.5.1-34: Notify (UE to P-CSCF)

```
NOTIFY sip:token7@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: “Alien Blaster” <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;
    tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 130 NOTIFY
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Event: refer
Content-Type: application/sip
Content-length: (…)

200 OK
```

Request-URI: contains the value of the Contact header from the 200-OK response to the initial INVITE.

Via: contains the IP address or FQDN of the originating UE.

From:, To:, Call-ID: contain the values previously used to establish the session, including the tag value from the response.

Cseq: next higher sequential value.

Contact: the IP address or FQDN of the originating UE.

35. **Notify (P-CSCF to S-CSCF) – see example in Table 10.5.1-35**

P-CSCF adds a Route header, with the saved value from the previous 200-OK response. P-CSCF identifies the proper saved value by the Request-URI.

P-CSCF#1 forwards the Notify request to S-CSCF#1.

Table 10.5.1-35: Notify (P-CSCF to S-CSCF)

```
NOTIFY sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Event:
Content-Type:
Content-length:

200 OK
```

Request-URI: the first component of the saved Route header.

Route: saved from the 200-OK response to the initial INVITE (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

36. **Notify (S-CSCF to S-CSCF) – see example in Table 10.5.1-36**

S-CSCF#1 forwards the Notify request to S-CSCF#2.

Table 10.5.1-36: Notify (S-CSCF to S-CSCF)

```
NOTIFY sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Record-Route: sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Event:
Content-Type:
Content-length:

200 OK
```

37. **Notify (S-CSCF to P-CSCF) – see example in Table 10.5.1-37**

S-CSCF#2 forwards the Notify request to P-CSCF#2.

Table 10.5.1-37: Notify (S-CSCF to P-CSCF)

```
NOTIFY sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Event:
Content-Type:
Content-length:

200 OK
```

38. **Notify (P-CSCF to UE) – see example in Table 10.5.1-38**

P-CSCF#2 forwards the Notify request to UE#2.

Table 10.5.1-38: Notify (P-CSCF to UE)

```
NOTIFY sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net;branch=token3
From:
To:
Call-ID:
Cseq:
Contact: token3@pcscf2.home.net
Event:
Content-Type:
Content-length:

200 OK
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

39. **200-OK (UE to P-CSCF) – see example in Table 10.5.1-39**

UE#2 acknowledges receipt of the Notify request (38) with a 200-OK final response, sent to P-CSCF#2.

Table 10.5.1-39: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.home.net;branch=token3
From:
To:
Call-ID:
CSeq:
Contact: sip:[5555::eee:fff:aaa:bbb]
Content-length: 0
```

40. **200-OK (P-CSCF to S-CSCF) – see example in Table 10.5.1-40**

P-CSCF#2 forwards the 200 OK final response to S-CSCF#2.

Table 10.5.1-40: 200 OK (P-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Content-length:
```

P-CSCF restores the Via headers and Record-Route headers from the branch value in its Via.

Contact: a locally defined value that identifies the UE.

41. **200-OK (S-CSCF to S-CSCF) – see example in Table 10.5.1-41**

S-CSCF#2 forwards the 200 OK final response to S-CSCF#1.

Table 10.5.1-41: 200 OK (S-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

42. **200-OK (S-CSCF to P-CSCF) – see example in Table 10.5.1-42**

S-CSCF#1 forwards the 200 OK final response to P-CSCF#1.

Table 10.5.1-42: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

43. **200-OK (P-CSCF to UE) – see example in Table 10.5.1-43**

P-CSCF#1 forwards the 200 OK final response to UE#1.

Table 10.5.1-43: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token2@pcscf1.home.net
Content-length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information

44. **BYE (UE to P-CSCF) – see example in Table 10.5.1-44**

Upon receiving the notification of successful refer operation (38), UE#2 terminates the session with UE#1..

Table 10.5.1-34: Bye (UE to P-CSCF)

```
BYE sip:token6@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
From: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost;tag=314159
To: “Alien Blaster” <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;
    tag=171828
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 131 BYE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-length: 0
```

Request-URI: contains the value of the Contact header from the initial INVITE.

Via: contains the IP address or FQDN of the originating UE.

From:, To:, Call-ID: contain the values previously used to establish the session, including the tag value from the response. Since this request is being initiated by the destination, the From and To are reversed.

Cseq: next higher sequential value.

Contact: the IP address or FQDN of the originating UE.

45. **Bye (P-CSCF to S-CSCF) – see example in Table 10.5.1-45**

P-CSCF adds a Route header, with the saved value from the previous 200-OK response. P-CSCF identifies the proper saved value by the Request-URI.

P-CSCF#2 forwards the Notify request to S-CSCF#2.

Table 10.5.1-45: Bye (P-CSCF to S-CSCF)

```
BYE sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Route: sip:scscf1.home.net, sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf1.home.net
Content-length:
```

Request-URI: the first component of the saved Route header.

Route: saved from the 200-OK response to the initial INVITE (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

46. **Bye (S-CSCF to S-CSCF) – see example in Table 10.5.1-46**

S-CSCF#2 forwards the Bye request to S-CSCF#1.

Table 10.5.1-46: Bye (S-CSCF to S-CSCF)

```
BYE sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Route: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Record-Route: sip:scscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

47. **Bye (S-CSCF to P-CSCF) – see example in Table 10.5.1-47**

S-CSCF#1 forwards the Bye request to P-CSCF#1.

Table 10.5.1-47: Bye (S-CSCF to P-CSCF)

```
BYE sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net,
    SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route: sip:scscf1.home.net, sip:scscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

48. **Bye (P-CSCF to UE) – see example in Table 10.5.1-48**

P-CSCF#2 forwards the Bye request to UE#2.

Table 10.5.1-48: Bye (P-CSCF to UE)

```
BYE sip:[5555::aaa:bbb:ccc:ddd] SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net;branch=token9
From:
To:
Call-ID:
Cseq:
Contact: token9@pcscf1.home.net
Content-length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

49. 200-OK (UE to P-CSCF) – see example in Table 10.5.1-49

UE#2 acknowledges receipt of the Bye request (48) with a 200-OK final response, sent to P-CSCF#1.

Table 10.5.1-49: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net;branch=token9
From:
To:
Call-ID:
CSeq:
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-length: 0
```

50. 200-OK (P-CSCF to S-CSCF) – see example in Table 10.5.1-50

P-CSCF#1 forwards the 200 OK final response to S-CSCF#1.

Table 10.5.1-50: 200 OK (P-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net,
    SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route: sip:scscf1.home.net, sip:scscf2.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf2.home.net
Content-length:
```

P-CSCF restores the Via headers and Record-Route headers from the branch value in its Via.

Contact: a locally defined value that identifies the UE.

51. 200-OK (S-CSCF to S-CSCF) – see example in Table 10.5.1-51

S-CSCF#1 forwards the 200 OK final response to S-CSCF#2.

Table 10.5.1-51: 200 OK (S-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

52. 200-OK (S-CSCF to P-CSCF) – see example in Table 10.5.1-52

S-CSCF#2 forwards the 200 OK final response to P-CSCF#2.

Table 10.5.1-52: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

53. **200-OK (P-CSCF to UE) – see example in Table 10.5.1-53**

P-CSCF#2 forwards the 200 OK final response to UE#2.

Table 10.5.1-53: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
From:
To:
Call-ID:
CSeq:
Contact: sip:token2@pcscf2.home.net
Content-length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information

10.5.2 Session Transfer replacing an existing session

An IP multi-media session already exists between UE#1 and UE#2, and an IP multi-media session already exists between UE#2 and UE#F. UE#2 desires UE#1 to initiate a new session to destination UE#F, and terminate the existing sessions. The procedures for this transfer are shown in Figure 10.5.2-1.

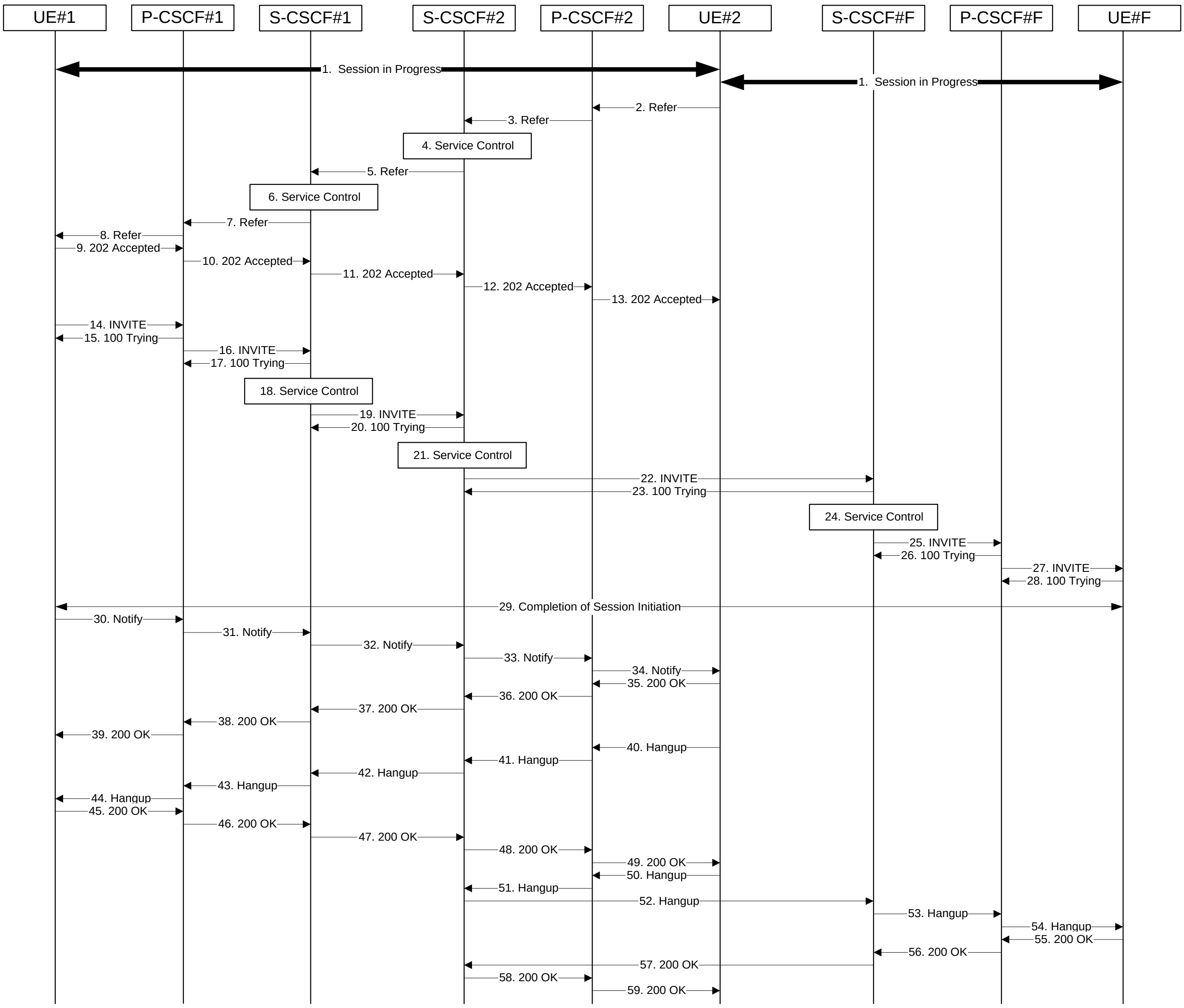

Figure 10.5.2-1 – Session Transfer replacing an existing session

1. Sessions in Progress

UE#1 initiates a multi-media session with UE#2. As a result, the state information stored at P-CSCF#2 is shown in Table 10.5.2-1a

Table 10.5.2-1a: State Information

```
Request-URI: sip:token6@pcscf2.home.net

From: “Alien Blaster” <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;
    tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost

Route: sip:scscf2.home.net, sip:scscf1.home.net,
    sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
```

UE#2 initiates a multi-media session with UE#F. As a result, the state information stored at P-CSCF#2 is shown in Table 10.5.2-1b

Table 10.5.2-1b: State Information

```
Request-URI: sip:token3@pcscf2.home.net

From: sip:B36(SHA-1(+1-212-555-2222; time=36123F05; seq=31))@localhost;tag=171828
To: sip:B36(SHA-1(+1-212-555-3333; time=36123F05; seq=32))@localhost;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=31))@localhost

Route: sip:scscf2.home.net, sip:scscff.home.net,
    sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscff.home.net
```

UE#2 has placed both of these sessions on hold.

2. **REFER (UE to P-CSCF) – see example in Table 10.5.2-2**

UE#2 sends a Refer request to its proxy, P-CSCF#2.

Table 10.5.2-2: REFER (UE to P-CSCF)

```
REFER sip:token6@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
From: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost;tag=314159
To: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;
    tag=171828
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 130 REFER
Contact: sip:[5555::eee:fff:aaa:bbb]
Refer-To: tel:+1-212-555-3333 ? Call-ID=B36(SHA-1(555-1111;time=36123E5B;seq=31))@localhost
Refer-By: sip:B36(SHA-1(+1-212-555-2222; time=36123F05; seq=31))@localhost;tag=171828
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off
Content-length: 0
```

Request-URI: contains the value of the Contact header from the 200-OK response to the initial INVITE.

Via: contains the IP address or FQDN of the originating UE.

From:, To:, Call-ID: contain the values previously used to establish the session, including the tag value from the response.

Cseq: next higher sequential value.

Contact: the IP address or FQDN of the originating UE.

Editor's Note: Use of Remote-Party-ID in REFER is FFS.

Editor's Note: The proper value for the Refer-By header is FFS. The value of the From header of the session to be replaced seems most appropriate.

3. **REFER (P-CSCF to S-CSCF) – see example in Table 10.5.2-3**

P-CSCF adds a Route header, with the saved value from the previous 200-OK response. P-CSCF identifies the proper saved value by the Request-URI.

P-CSCF#2 forwards the Refer request to S-CSCF#2.

Table 10.5.2-3: REFER (P-CSCF to S-CSCF)

```
REFER sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Route: sip:scscf1.home.net, sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Refer-To:
Refer-By:
Remote-Party-ID:
Content-length:
```

Request-URI: the first component of the saved Route header.

Route: saved from the 200-OK response to the initial INVITE (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

4. Service Control

5. **REFER (S-CSCF to S-CSCF) – see example in Table 10.5.2-5**

In order to maintain the expectation of privacy of the identity of the new destination, S-CSCF#2 converts the "Refer-To" header into a private URL. S-CSCF#2 forwards the Refer request to S-CSCF#1.

NOTE: If the network operator desired configuration independence, the REFER would be routed through an I-CSCF before leaving the operator's network. For example, see configuration S-S#1b. That I-CSCF would convert the private URL into one that specified the I-CSCF as the hostname.

Table 10.5.2-5: REFER (S-CSCF to S-CSCF)

```
REFER sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Route: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Record-Route: sip:scscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Refer-To: sip:token(tel:+1-212-555-3333)@scscf2.home.net;private ?
    Call-ID=B36(SHA-1(555-1111;time=36123E5B;seq=31))@localhost
Refer-By:
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off;screen=yes
Content-length:
```

6. Service Control

7. **REFER (S-CSCF to P-CSCF) – see example in Table 10.5.2-7**

S-CSCF#1 forwards the Refer request to P-CSCF#1.

Table 10.5.2-7: REFER (S-CSCF to P-CSCF)

```
INVITE sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UEP scscf1.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net,
    SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route: sip:scscf1.home.net, sip:scscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Refer-To:
Refer-By:
Remote-Party-ID:
Content-length:
```

8. **REFER (P-CSCF to UE) – see example in Table 10.5.2-8**

P-CSCF#1 forwards the Refer request to UE#1.

Table 10.5.2-8: REFER (P-CSCF to UE)

```
REFER sip:[5555::aaa:bbb:ccc:ddd] SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net;branch=token3
From:
To:
Call-ID:
Cseq:
Contact: token3@pcscf2.home.net
Refer-To:
Refer-By:
Remote-Party-ID:
Content-length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

9. **202-Accepted (UE to P-CSCF) – see example in Table 10.5.2-9**

UE#2 acknowledges receipt of the Refer request (8) with a 202-Accepted final response, sent to P-CSCF#1.

Table 10.5.2-8: 202 Accepted (UE to P-CSCF)

```
SIP/2.0 202 Accepted
Via: SIP/2.0/UDP pcscf2.home.net;branch=token3
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

10. **202-Accepted (P-CSCF to S-CSCF) – see example in Table 10.5.2-10**

P-CSCF#1 forwards the 202 Accepted final response to S-CSCF#1.

Table 10.5.2-10: 202 Accepted (P-CSCF to S-CSCF)

```
SIP/2.0 202 Accepted
Via: SIP/2.0/UEP scscf1.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net,
    SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route: sip:scscf1.home.net, sip:scscf2.home.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

P-CSCF restores the Via headers and Record-Route headers from the branch value in its Via.

11. **202-Accepted (S-CSCF to S-CSCF) – see example in Table 10.5.2-11**

S-CSCF#1 forwards the 202 Accepted final response to S-CSCF#2.

Table 10.5.2-11: 202 Accepted (S-CSCF to S-CSCF)

```
SIP/2.0 202 Accepted
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-length:
```

12. **202-Accepted (S-CSCF to P-CSCF) – see example in Table 10.5.2-12**

S-CSCF#2 forwards the 202 Accepted final response to P-CSCF#2.

Table 10.5.2-12: 202 Accepted (S-CSCF to P-CSCF)

```
SIP/2.0 202 Accepted
Via: SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-length:
```

13. **202-Accepted (P-CSCF to UE) – see example in Table 10.5.2-13**

P-CSCF#2 forwards the 202 Accepted final response to UE#2.

Table 10.5.2-13: 202 Accepted (P-CSCF to UE)

```
SIP/2.0 202 Accepted
Via: SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
From:
To:
Call-ID:
CSeq:
Content-length:
```

P-CSCF removes the Record-Route header

.

14. **INVITE (UE to P-CSCF) – see example in Table 10.5.2-14**

UE#1 initiates an INVITE request based on the Refer-To header URL in the REFER request. The INVITE is sent from the UE to P-CSCF#1.

Table 10.5.2-14: INVITE (UE to P-CSCF)

```
INVITE sip:token(tel:+1-212-555-3333)@scscf2.home.net;private SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Supported: 100rel
Remote-Party-ID: “John Doe” <tel:+1-212-555-1111>;privacy=off
Proxy-Require: privacy
Anonymity: Off
From: “Alien Blaster” <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=74))@localhost>;
    tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=75))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=31))@localhost
Cseq: 127 INVITE
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Refer-By: sip:B36(SHA-1(+1-212-555-2222; time=36123F05; seq=31))@localhost;tag=171828
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=audio 3456 RTP/AVP 97 3 96
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

Call-ID: value taken from the URL parameter attached to the Refer-To header

15. **100 Trying (P-CSCF to UE) – see example in Table 10.5.2-15**

P-CSCF#1 responds to the INVITE request (14) with a 100 Trying provisional response.

Table 10.5.2-15: 100 Trying (P-CSCF to UE)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

16. **INVITE (P-CSCF to S-CSCF) – see example in Table 10.5.2-16**

P-CSCF#1 remembers (from the registration procedure) the request routing for this UE. This becomes a Route header in the request. The next hop is the S-CSCF serving this UE. P-CSCF rewrites the Contact header with a locally defined value that identifies the UE. P-CSCF adds itself to the Via header.

Table 10.5.2-16: INVITE (P-CSCF to S-CSCF)

```
INVITE sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:token(tel:+1-212-555-3333)@scscf2.home.net;private
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Refer-By:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

17. **100 Trying (S-CSCF to P-CSCF) – see example in Table 10.5.2-17**

S-CSCF#1 responds to the INVITE request (16) with a 100 Trying provisional response.

Table 10.5.2-17: 100 Trying (S-CSCF to P-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

18. **Service Control**

S-CSCF#1 performs whatever service control logic is appropriate for this call attempt.

19. **INVITE (S-CSCF to S-CSCF) – see example in Table 10.5.2-19**

S-CSCF#1 performs an analysis of the destination address, which is a private URL generated by S-CSCF#2. Since it is a destination within the same operator's network, S-CSCF#1 forwards the INVITE request directly to S-CSCF#2.

Table 10.5.2-19: INVITE (S-CSCF to S-CSCF)

```
INVITE sip:token(tel:+1-212-555-3333)@scscf2.home.net;private SIP/2.0
Via: SIP/2.0/UDP sip:scscf1.home.net SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf1.home.net
Supported:
Remote-Party-ID: “John Doe” <tel:+1-212-555-1111>;privacy=off;screen=yes
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Refer-By:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

20. **100 Trying (S-CSCF to S-CSCF) – see example in Table 10.5.2-20**

S-CSCF#2 responds to the INVITE request (19) by sending a 100 Trying provisional response to S-CSCF#1.

Table 10.5.2-20: 100 Trying (S-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP  [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

21. **Service Control**

S-CSCF#2 performs whatever service control logic is appropriate for this call transfer attempt.

22. **INVITE (S-CSCF to S-CSCF) – see example in Table 10.5.2-22**

S-CSCF#2 determines the destination address from the private URL contained in the INVITE request. Based on information in that URL, and information saved from step #4 above (implementation decision), S-CSCF#2 verifies the validity of the transfer request, and that it is within a short time delay from the REFER request.

S-CSCF#2 builds a Route header based on stored state information for the Refer’d session (as determined by the Call-ID and Refer-By values in step #4 above).

S-CSCF#2 forwards the INVITE request to S-CSCF#F.

Table 10.5.2-22: INVITE (S-CSCF to S-CSCF)

```
INVITE sip:scscff.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
Route: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscff.home.net, sip:+1-212-555-3333@home.net
Supported:
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off;screen=yes;party=transferor
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Refer-By:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

Editor's Note: Use of "party=transferor" in a separate Remote-Party-ID header is FFS.

23. **100 Trying (S-CSCF to S-CSCF) – see example in Table 10.5.2-23**
S-CSCF#F responds to the INVITE request (22) by sending a 100 Trying provisional response to S-CSCF#2.

Table 10.5.2-23: 100 Trying (S-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

24. **Service Control**
S-CSCF#F performs whatever service control logic is appropriate for this call attempt.

25. **INVITE (S-CSCF to P-CSCF) – see example in Table 10.5.2-25**
S-CSCF#F uses the Route header value to determine the next hop CSCF for this UE. It forwards the INVITE request to P-CSCF#F.

Table 10.5.2-25: INVITE (S-CSCF to P-CSCF)

```
INVITE sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscff.home.net SIP/2.0
Via: SIP/2.0/UDP scscff.home.net, SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP
    scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscff.home.net, sip:scscf2.home.net, sip:scscf1.home.net
Route: sip:+1-212-555-3333@home.net
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Refer-By:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

26. **100 Trying (P-CSCF to S-CSCF) – see example in Table 10.5.2-26**
    P-CSCF#F responds to the INVITE request (25) by sending a 100 Trying provisional response to S-CSCF#F.

Table 10.5.2-26: 100 Trying (P-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscff.home.net, SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP
    scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

27. **INVITE (P-CSCF to UE) – see example in Table 10.5.2-27**
    P-CSCF forwards the INVITE request to the UE.

Table 10.5.2-27: INVITE (P-CSCF to UE)

```
INVITE sip:+1-212-555-3333@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscff.home.net;branch=token6
Supported:
Remote-Party-ID:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: token6@pcscff.home.net
Refer-By:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

Editor's Note: The mechanism by which UE#F knows it is to replace an existing session with this new session is FFS. A match of the Call-ID to an existing session, and the Refer-By to either From or To of the same session should be sufficient.

28. **100 Trying (UE to P-CSCF) – see example in Table 10.5.2-28**

UE#F may optionally send a 100 Trying provisional response to P-CSCF.

Table 10.5.2-28: 100 Trying (UE to P-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pcscf1.home.net;branch=token6
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

29. **Completion of Session Initiation**

UE#1 and UE#F complete the session initiation, as shown in the MO, S-S, and MT procedures.

30. **NOTIFY (UE to P-CSCF) – see example in Table 10.5.2-30**

When the session with UE#F has been successfully established, UE#1 sends a Notify request to its proxy, P-CSCF#1. The call leg identification for this Notify is taken from that used in the Refer, earlier.

Table 10.5.2-30: Notify (UE to P-CSCF)

```
NOTIFY sip:token7@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;
    tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 130 NOTIFY
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Event: refer
Content-Type: application/sip
Content-length: (…)

200 OK
```

Request-URI: contains the value of the Contact header from the 200-OK response to the initial INVITE.

Via: contains the IP address or FQDN of the originating UE.

From:, To:, Call-ID: contain the values previously used to establish the session, including the tag value from the response.

Cseq: next higher sequential value.

Contact: the IP address or FQDN of the originating UE.

31. **Notify (P-CSCF to S-CSCF) – see example in Table 10.5.2-31**

P-CSCF adds a Route header, with the saved value from the previous 200-OK response. P-CSCF identifies the proper saved value by the Request-URI.

P-CSCF#1 forwards the Notify request to S-CSCF#1.

Table 10.5.2-31: Notify (P-CSCF to S-CSCF)

```
NOTIFY sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Event:
Content-Type:
Content-length:

200 OK
```

Request-URI: the first component of the saved Route header.

Route: saved from the 200-OK response to the initial INVITE (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

32. **Notify (S-CSCF to S-CSCF) – see example in Table 10.5.2-32**

S-CSCF#1 forwards the Notify request to S-CSCF#2.

Table 10.5.2-32: Notify (S-CSCF to S-CSCF)

```
NOTIFY sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Record-Route: sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Event:
Content-Type:
Content-length:

200 OK
```

33. **Notify (S-CSCF to P-CSCF) – see example in Table 10.5.2-33**

S-CSCF#2 forwards the Notify request to P-CSCF#2.

Table 10.5.2-33: Notify (S-CSCF to P-CSCF)

```
NOTIFY sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Event:
Content-Type:
Content-length:

200 OK
```

34. Notify (P-CSCF to UE) – see example in Table 10.5.2-34

P-CSCF#2 forwards the Notify request to UE#2.

Table 10.5.2-34: Notify (P-CSCF to UE)

```
NOTIFY sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net;branch=token3
From:
To:
Call-ID:
Cseq:
Contact: token3@pcscf2.home.net
Event:
Content-Type:
Content-length:

200 OK
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

35. 200-OK (UE to P-CSCF) – see example in Table 10.5.2-35

UE#2 acknowledges receipt of the Notify request (34) with a 200-OK final response, sent to P-CSCF#2.

Table 10.5.2-35: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.home.net;branch=token3
From:
To:
Call-ID:
CSeq:
Contact: sip:[5555::eee:fff:aaa:bbb]
Content-length: 0
```

36. 200-OK (P-CSCF to S-CSCF) – see example in Table 10.5.2-36

P-CSCF#2 forwards the 200 OK final response to S-CSCF#2.

Table 10.5.2-36: 200 OK (P-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Content-length:
```

P-CSCF restores the Via headers and Record-Route headers from the branch value in its Via.

Contact: a locally defined value that identifies the UE.

37. **200-OK (S-CSCF to S-CSCF) – see example in Table 10.5.2-37**

S-CSCF#2 forwards the 200 OK final response to S-CSCF#1.

Table 10.5.2-37: 200 OK (S-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

38. **200-OK (S-CSCF to P-CSCF) – see example in Table 10.5.2-38**

S-CSCF#1 forwards the 200 OK final response to P-CSCF#1.

Table 10.5.2-38: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

39. **200-OK (P-CSCF to UE) – see example in Table 10.5.2-39**

P-CSCF#1 forwards the 200 OK final response to UE#1.

Table 10.5.2-39: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token2@pcscf1.home.net
Content-length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information

40. **BYE (UE to P-CSCF) – see example in Table 10.5.2-40**

Upon receiving the notification of successful refer operation (34), UE#2 terminates the session with UE#1.

Table 10.5.2-40: Bye (UE to P-CSCF)

```
BYE sip:token6@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
From: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost;tag=314159
To: “Alien Blaster” <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;
    tag=171828
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 131 BYE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-length: 0
```

Request-URI: contains the value of the Contact header from the initial INVITE.

Via: contains the IP address or FQDN of the originating UE.

From:, To:, Call-ID: contain the values previously used to establish the session, including the tag value from the response. Since this request is being initiated by the destination, the From and To are reversed.

Cseq: next higher sequential value.

Contact: the IP address or FQDN of the originating UE.

41. **Bye (P-CSCF to S-CSCF) – see example in Table 10.5.2-41**

P-CSCF adds a Route header, with the saved value from the previous 200-OK response. P-CSCF identifies the proper saved value by the Request-URI.

P-CSCF#2 forwards the Notify request to S-CSCF#2.

Table 10.5.2-41: Bye (P-CSCF to S-CSCF)

```
BYE sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Route: sip:scscf1.home.net, sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf1.home.net
Content-length:
```

Request-URI: the first component of the saved Route header.

Route: saved from the 200-OK response to the initial INVITE (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

42. **Bye (S-CSCF to S-CSCF) – see example in Table 10.5.2-42**

S-CSCF#2 forwards the Bye request to S-CSCF#1.

Table 10.5.2-42: Bye (S-CSCF to S-CSCF)

```
BYE sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Route: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Record-Route: sip:scscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

43. **Bye (S-CSCF to P-CSCF) – see example in Table 10.5.2-43**

S-CSCF#1 forwards the Bye request to P-CSCF#1.

Table 10.5.2-43: Bye (S-CSCF to P-CSCF)

```
BYE sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net,
    SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route: sip:scscf1.home.net, sip:scscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

44. **Bye (P-CSCF to UE) – see example in Table 10.5.2-44**

P-CSCF#2 forwards the Bye request to UE#2.

Table 10.5.2-44: Bye (P-CSCF to UE)

```
BYE sip:[5555::aaa:bbb:ccc:ddd] SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net;branch=token9
From:
To:
Call-ID:
Cseq:
Contact: token9@pcscf1.home.net
Content-length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

45. **200-OK (UE to P-CSCF) – see example in Table 10.5.2-45**

UE#2 acknowledges receipt of the Bye request (44) with a 200-OK final response, sent to P-CSCF#1.

Table 10.5.2-45: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net;branch=token9
From:
To:
Call-ID:
CSeq:
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-length: 0
```

46. **200-OK (P-CSCF to S-CSCF) – see example in Table 10.5.2-46**

P-CSCF#1 forwards the 200 OK final response to S-CSCF#1.

Table 10.5.2-46: 200 OK (P-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net,
    SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route: sip:scscf1.home.net, sip:scscf2.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf2.home.net
Content-length:
```

P-CSCF restores the Via headers and Record-Route headers from the branch value in its Via.

Contact: a locally defined value that identifies the UE.

47. **200-OK (S-CSCF to S-CSCF) – see example in Table 10.5.2-47**

S-CSCF#1 forwards the 200 OK final response to S-CSCF#2.

Table 10.5.2-547: 200 OK (S-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

48. **200-OK (S-CSCF to P-CSCF) – see example in Table 10.5.2-48**

S-CSCF#2 forwards the 200 OK final response to P-CSCF#2.

Table 10.5.2-48: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

49. **200-OK (P-CSCF to UE) – see example in Table 10.5.2-49**

P-CSCF#2 forwards the 200 OK final response to UE#2.

Table 10.5.2-49: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
From:
To:
Call-ID:
CSeq:
Contact: sip:token2@pcscf2.home.net
Content-length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information

50. **BYE (UE to P-CSCF) – see example in Table 10.5.2-50**

Upon receiving the notification of successful refer operation (38), UE#2 terminates the session with UE#F.

Table 10.5.2-50: Bye (UE to P-CSCF)

```
BYE sip:token4@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
From: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost;tag=314159
To: “Alien Blaster” <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;
    tag=171828
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 131 BYE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-length: 0
```

Request-URI: contains the value of the Contact header from the initial INVITE.

Via: contains the IP address or FQDN of the originating UE.

From:, To:, Call-ID: contain the values previously used to establish the session, including the tag value from the response. Since this request is being initiated by the destination, the From and To are reversed.

Cseq: next higher sequential value.

Contact: the IP address or FQDN of the originating UE.

51. **Bye (P-CSCF to S-CSCF) – see example in Table 10.5.2-51**

P-CSCF adds a Route header, with the saved value from the previous 200-OK response. P-CSCF identifies the proper saved value by the Request-URI.

P-CSCF#2 forwards the Notify request to S-CSCF#2.

Table 10.5.2-51: Bye (P-CSCF to S-CSCF)

```
BYE sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Route: sip:scscff.home.net, sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscff.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Content-length:
```

Request-URI: the first component of the saved Route header.

Route: saved from the 200-OK response to the initial INVITE (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

52. Bye (S-CSCF to S-CSCF) – see example in Table 10.5.2-52

S-CSCF#2 forwards the Bye request to S-CSCF#F.

Table 10.5.2-52: Bye (S-CSCF to S-CSCF)

```
BYE sip:scscff.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Route: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscff.home.net
Record-Route: sip:scscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

53. Bye (S-CSCF to P-CSCF) – see example in Table 10.5.2-53

S-CSCF#F forwards the Bye request to P-CSCF#F.

Table 10.5.2-53: Bye (S-CSCF to P-CSCF)

```
BYE sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscff.home.net SIP/2.0
Via: SIP/2.0/UDP scscff.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net,
    SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route: sip:scscff.home.net, sip:scscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

54. Bye (P-CSCF to UE) – see example in Table 10.5.2-54

P-CSCF#F forwards the Bye request to UE#F.

Table 10.5.2-54: Bye (P-CSCF to UE)

```
BYE sip:[5555::aaa:bbb:ccc:ddd] SIP/2.0
Via: SIP/2.0/UDP pcscff.home.net;branch=token9
From:
To:
Call-ID:
Cseq:
Contact: token9@pcscff.home.net
Content-length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

55. **200-OK (UE to P-CSCF) – see example in Table 10.5.2-55**

UE#F acknowledges receipt of the Bye request (58) with a 200-OK final response, sent to P-CSCF#F.

Table 10.5.2-55: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscff.home.net;branch=token9
From:
To:
Call-ID:
CSeq:
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-length: 0
```

56. **200-OK (P-CSCF to S-CSCF) – see example in Table 10.5.2-56**

P-CSCF#F forwards the 200 OK final response to S-CSCF#F.

Table 10.5.2-56: 200 OK (P-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscff.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net,
    SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route: sip:scscff.home.net, sip:scscf2.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscff.home.net
Content-length:
```

P-CSCF restores the Via headers and Record-Route headers from the branch value in its Via.

Contact: a locally defined value that identifies the UE.

57. **200-OK (S-CSCF to S-CSCF) – see example in Table 10.5.2-57**

S-CSCF#F forwards the 200 OK final response to S-CSCF#2.

Table 10.5.2-57: 200 OK (S-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

58. **200-OK (S-CSCF to P-CSCF) – see example in Table 10.5.2-58**

S-CSCF#2 forwards the 200 OK final response to P-CSCF#2.

Table 10.5.2-58: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

59. **200-OK (P-CSCF to UE) – see example in Table 10.5.2-59**

P-CSCF#2 forwards the 200 OK final response to UE#2.

Table 10.5.2-59: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
From:
To:
Call-ID:
CSeq:
Contact: sip:token2@pcscf2.home.net
Content-length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information

3GPP TSG CN WG1 Meeting #17 **Tdoc N1-010890**
Puerto Rico, 14th - 18th May 2001

Title:	**Liaison Statement on the IM Call Transfer service**
Source:	Joint CN WG1,2, 3, 4 groups
To:	TSG SA3, SA5
cc:	TSG SA2, TSG CN2, TSG CN3, TSG CN4

Contact Person:
Name: Sunil Chotai
E-mail Address: sunil.chotai@bt.com
Tel. Number: +44 1473 605603

---

1. Introduction

The joint CN1-4 groups reviewed proposed message flows for an IM Call Transfer service to be included in TS 24.228. It was noted that this service is similar to the Explicit Call Transfer service in GSM (TS23.091). Concerns were raised that there could be significant fraud problems with this service. Concerns were also raised that the charging principles for this service are not known.

The proposed detailed message flows are shown in contribution N1-010706. These build upon high level Call Transfer message flows described in TS 23.228. The message flows assume that the initial Mobile (UE) to Mobile (UE) call is established between mobiles that are served by a 3GPP IM network. However, it should be noted that it is possible for a Mobile to call a terminal which is not served by a 3GPP network.

CN1 is keen to progress the work on the IM Call Transfer service but needs guidance from security experts on how best to ensure that operators do not experience significant fraud problems with this service. CN1 also needs guidance from charging experts on the charging principles for this service. It seems possible that the security and charging aspects may be somehow related.

2. Actions:

To SA3.

SA3 security experts are requested to review the proposed IM Call Transfer service message flows (described in TS 23.228 and N1-010706) from a fraud potential viewpoint, and to provide guidance on any essential modifications that are necessary to ensure potential fraud problems are minimised.

To SA5.

SA5 charging experts are requested to review the proposed IM Call Transfer service message flows (described in TS 23.228 and N1-010706) from a charging viewpoint, and to provide guidance on charging principles.

3. Date of Next CN1 Meetings:

CN1_18	10th – 12th July 2001	Dresden, Germany.
CN1_19	27th – 31th August 2001	

4. Attachments:

N1-010706

3GPP TSG_CN WG1 Meeting #17 **Tdoc N1-010837**
Puerto Rico, USA **(rev of N1-010810 rev of N1-010621)**
14 - 18 May, 2001

CR-Form-v3

CHANGE REQUEST

⌘	**24.008 CR**	**402**	⌘ rev	**2**	⌘ Current version:	**3.7.0**	⌘

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

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

Title:	⌘	CLASSMARK1, 2 and 3 corrections.
Source:	⌘	VODAFONE
Work item code:	⌘	TEI and GSM-UMTS interworking. ***Date:*** ⌘ 2nd May 2001.
Category:	⌘	F ***Release:*** ⌘ R99
		Use one of the following categories: ***F*** *(essential correction)* ***A*** *(corresponds to a correction in an earlier release)* ***B*** *(Addition of feature),* ***C*** *(Functional modification of feature)* ***D*** *(Editorial modification)* Detailed explanations of the above categories can be found in 3GPP TR 21.900. *Use one of the following releases:* *2 (GSM Phase 2)* *R96 (Release 1996)* *R97 (Release 1997)* *R98 (Release 1998)* *R99 (Release 1999)* *REL-4 (Release 4)* *REL-5 (Release 5)*

Reason for change:	⌘	When access is performed via UMTS channels, information on GSM MS capability is *not* irrelevant as this is used by the RNC for decision on **handover from UMTS to GSM**. The required specification of the coding of CLASSMARK doesn't allow the network to distinguish unambiguously single banded GSM mobiles. This may lead to dropped calls if the network can't understand the precise implementation in the mobile. Also it is impossible for a UMTS mobile not supporting GSM to signal its capability without misleadingly indicate that it supports GSM as well.
Summary of change:	⌘	It is clarified how a single band GSM mobile and a UMTS only mobile need to code the classmarks 1, 2 and 3.
Consequences if not approved:	⌘	There will be a misunderstanding between network and mobile as to what coding schemes and handovers are supported. This can lead to the network attempting to hand the mobile some where or send information which the mobile is unable to decode.

Clauses affected:	⌘	10.5.1.5, 10.5.1.6 & 10.5.1.7
Other specs affected:	⌘	[ ] Other core specifications ⌘ [ ] Test specifications [ ] O&M Specifications
Other comments:	⌘	This CR is a copy of that found within Tdoc N1-010620 (GP-010847). It has been created using the most recent reference version of 24.008, but the changes are the same as those identified in N1-010620 (GP-010847). A CR proposing almost

	identical changes for Release 4 appears in CR403.

10.5.1.5 Mobile Station Classmark 1

The purpose of the *Mobile Station Classmark 1* information element is to provide the network with information concerning aspects of high priority of the mobile station equipment. This affects the manner in which the network handles the operation of the mobile station. The Mobile Station Classmark information indicates general mobile station characteristics and it shall therefore, except for fields explicitly indicated, be independent of the frequency band of the channel it is sent on.

The *Mobile Station Classmark 1* information element is coded as shown in figure 10.5.5/3GPP TS 24.008 and table 10.5.5/3GPP TS 24.008.

The *Mobile Station Classmark 1* is a type 3 information element with 2 octets length.

8	7	6	5	4	3	2	1	
	Mobile Station Classmark 1 IEI							octet 1
0 spare	Revision level		ES IND	A5/1	RF power capability			octet 2

Figure 10.5.5/3GPP TS 24.008 *Mobile Station Classmark 1* information element

~~A MS supporting GSM shall always encode all fields relevant for GSM radio access technology, even when accessing UMTS radio access technology. A UMTS MS which does not support GSM shall encode fields relevant only for GSM radio access tecnology using any value which has been defined for this version of the protocol and is not reserved.~~

Table 10.5.5/3GPP TS 24.008: *Mobile Station Classmark 1* information element

Revision level (octet 2)
~~Required for MS supporting GSM and UMTS.~~
Bits
7 6
0 0 Reserved for GSM phase 1
0 1 Used by GSM phase 2 mobile stations
1 0 Used by mobile stations supporting R99 or later versions of the protocol
1 1 Reserved for future use

ES IND (octet 2, bit 5) "Controlled Early Classmark Sending" option implementation
~~Required for MS supporting GSM.~~
An MS not supporting GSM shall set this bit to '0'.
An MS supporting GSM shall indicate the associated GSM capability (see table):

0 "Controlled Early Classmark Sending" option is not implemented in the MS
1 "Controlled Early Classmark Sending" option is implemented in the MS

NOTE: The value of the ES IND gives the implementation in the MS. It's value is **not** dependent on the broadcast SI 3 Rest Octet <Early Classmark Sending Control> value.

A5/1 algorithm supported (octet 2, bit4)
~~Required for mobile station supporting GSM.~~
An MS not supporting GSM shall set this bit to '1'.
An MS supporting GSM shall indicate the associated GSM capability (see table):

0 encryption algorithm A5/1 available
1 encryption algorithm A5/1 not available

RF power capability (octet 2)
~~Required for mobile stations supporting GSM.~~
When GSM 450, GSM 480, GSM 850, GSM 900 P, E [or R] band is used (for exceptions see GSM 04.18), the MS shall indicate the RF power capability of the band used (see table);
When UMTS is used, a single band GSM 450, GSM 480, GSM 850, GSM 900 P, E [or R] MS shall indicate the RF power capability corresponding to the (GSM) band it supports (see table); in this case information on which single band is supported is found in classmark 3.
Bits
3 2 1
0 0 0 class 1
0 0 1 class 2
0 1 0 class 3
0 1 1 class 4
1 0 0 class 5

All other values are reserved.

When the DCS 1800 or PCS 1900 band is used (for exceptions see 3GPP TS 04.18, sub-clause 3.4.18), the MS shall indicate the RF power capability of the band used (see table) ;~~:~~
When UMTS is used, a single band DCS1800 or PCS 1900 MS shall indicate the RF power capability corresponding to the (GSM) band it supports (see table); in this case information on which single band is supported is found in classmark 3.
Bits
3 2 1
0 0 0 class 1
0 0 1 class 2
0 1 0 class 3
All other values are reserved.
~~All other values are reserved.~~
When UMTS is used, an MS not supporting any GSM band or a multiband GSM MS shall code this field as follows (see table):
Bits
3 2 1
1 1 1 ~~Shall be sent;~~ RF Power capability is irrelevant in this information element
All other values are reserved.

10.5.1.6 Mobile Station Classmark 2

The purpose of the *Mobile Station Classmark 2* information element is to provide the network with information concerning aspects of both high and low priority of the mobile station equipment. This affects the manner in which the network handles the operation of the mobile station. The Mobile Station Classmark information indicates general mobile station characteristics and it shall therefore, except for fields explicitly indicated, be independent of the frequency band of the channel it is sent on.

The *Mobile Station Classmark 2* information element is coded as shown in figure 10.5.6/3GPP TS 24.008, table 10.5.6a/3GPP TS 24.008 and table 10.5.6b/3GPP TS 24.008.
The *Mobile Station Classmark 2* is a type 4 information element with 5 octets length.

8	7	6	5	4	3	2	1	
	Mobile station classmark 2 IEI							octet 1
Length of mobile station classmark 2 contents								octet 2
0 spare	Revision Level		ES IND	A5/1	RF power capability			octet 3
0 spare	PS capa.	SS Screen. Indicator		SM ca Pabi.	VBS	VGCS	FC	octet 4
CM3	0 spare	LCSVA CAP	UCS2	SoLSA	CMSP	A5/3	A5/2	octet 5

NOTE: Owing to backward compatibility problems, bit 8 of octet 4 should not be used unless it is also checked that the bits 8, 7 and 6 of octet 3 are not "0 0 0".

Figure 10.5.6/3GPP TS 24.008 *Mobile Station Classmark 2* information element

Table 10.5.6a/3GPP TS 24.008: *Mobile Station Classmark 2* information element

Revision level (octet 3)
~~Required for MS supporting GSM and UMTS.~~
Bits
7 6
0 0 Reserved for GSM phase 1
0 1 Used by GSM phase 2 mobile stations
1 0 Used by mobile stations supporting R99 or later versions of the protocol
1 1 Reserved for future use

ES IND (octet 3, bit 5) "Controlled Early Classmark Sending" option implementation
~~Required for MS supporting GSM.~~
An MS not supporting GSM shall set this bit to '0'.
An MS supporting GSM shall indicate the associated GSM capability (see table):

0 "Controlled Early Classmark Sending" option is not implemented in the MS
1 "Controlled Early Classmark Sending" option is implemented in the MS

NOTE: The value of the ES IND gives the implementation in the MS. It's value is **not** dependent on the broadcast SI 3 Rest Octet <Early Classmark Sending Control> value

Table 10.5.6a/3GPP TS 24.008: *Mobile Station Classmark 2* information element

A5/1 algorithm supported (octet 3, bit 4)
~~Required for MS supporting GSM.~~
An MS not supporting GSM shall set this bit to '1'.
An MS supporting GSM shall indicate the associated GSM capability (see table):

0 encryption algorithm A5/1 available
1 encryption algorithm A5/1 not available

RF Power Capability (Octet 3)
~~Required for MS supporting GSM.~~
When GSM 450, GSM 480, GSM 850, GSM 900 P, E [or R] band is used (for exceptions see GSM 04.18)~~:~~, the MS shall indicate the RF power capability of the band used (see table);
When UMTS is used, a single band GSM 450, GSM 480, GSM 850, GSM 900 P, E [or R] MS shall indicate the RF power capability corresponding to the (GSM) band it supports (see table); in this case information on which single band is supported is found in classmark 3.

Bits
3 2 1
0 0 0 Class 1
0 0 1 Class 2
0 1 0 Class 3
0 1 1 Class 4
1 0 0 Class 5
All other values are reserved.
~~All other values are reserved.~~

When the DCS 1800 or PCS 1900 band is used (for exceptions see GSM 04.18~~3~~)~~:~~, the MS shall indicate the RF power capability of the band used (see table) ;
When UMTS is used, a single band DCS1800 or PCS 1900 MS shall indicate the RF power capability corresponding to the (GSM) band it supports (see table); in this case information on which single band is supported is found in classmark 3.
Bits
3 2 1
0 0 0 Class 1
0 0 1 Class 2
0 1 0 Class 3
All other values are reserved.
~~All other values are reserved.~~
When UMTS is used, an MS not supporting any GSM ban d or a multiband GSM MS shall code this field as follows (see table):
Bits
3 2 1
1 1 1 ~~Shall be sent;~~ RF Power capability is irrelevant in this information element
All other values are reserved.
~~All other values are reserved.~~

PS capability (pseudo-synchronization capability) (octet 4)
~~Required for MS supporting GSM~~
An MS not supporting GSM shall set this bit to '0'.
An MS supporting GSM shall indicate the associated GSM capability (see table):
Bit **7**
0 PS capability not present
1 PS capability present

SS Screening Indicator (octet 4)
~~Required for MS supporting GSM and UMTS~~
Bits
6 5
0 0 defined in 3GPP TS 24.080
0 1 defined in 3GPP TS 24.080
1 0 defined in 3GPP TS 24.080
1 1 defined in 3GPP TS 24.080

Table 10.5.6a/3GPP TS 24.008: *Mobile Station Classmark 2* information element

SM capability (MT SMS pt to pt capability) (octet 4)
~~Required for MS supporting GSM.~~
Bit **4**
0 Mobile station does not support mobile terminated point to point SMS
1 Mobile station supports mobile terminated point to point SMS

~~Table 10.5.6a/3GPP TS 24.008: *Mobile Station Classmark 2* information element~~

VBS notification reception (octet 4)
~~Required for MS supporting GSM.~~
An MS not supporting GSM shall set this bit to ‘0’.
An MS supporting GSM shall indicate the associated GSM capability (see table):
Bit **3**
0 no VBS capability or no notifications wanted
1 VBS capability and notifications wanted

VGCS notification reception (octet 4)
~~Required for MS supporting GSM.~~
An MS not supporting GSM shall set this bit to ‘0’.
An MS supporting GSM shall indicate the associated GSM capability (see table):
Bit **2**
0 no VGCS capability or no notifications wanted
1 VGCS capability and notifications wanted

FC Frequency Capability (octet 4)
~~Required for MS supporting GSM.~~
When the GSM 400 or GSM 850 or DCS 1800 or PCS 1900 band or UMTS is used (for exceptions see GSM 04.18, for definitions of frequency band see GSM 05.05), this bit shall be sent with the value ‘0’.~~:~~
~~Bit **1**~~
~~0 Reserved for future use (for definition of frequency bands see GSM 05.05)~~
Note: This bit conveys no information about support or non support of the E-GSM or R-GSM bands when ~~transmitted on a~~ GSM 400, GSM 850, DCS1800, PCS1900 band or UMTS is used~~channel~~.

~~When GSM 850 band is used (for exceptions see GSM 04.18:~~
~~Bit **1**~~
~~0 Reserved for future use (for definition of frequency bands see GSM 05.05)~~

~~Note: This bit conveys no information about support or non support of the E-GSM or R-GSM band when transmitted on a GSM 850 channel.~~

When a GSM 900 band is used (for exceptions see GSM 04.18):
Bit **1**
0 The MS does not support the E-GSM or R-GSM band (For definition of frequency bands see GSM 05.05)
1 The MS does support the E-GSM or R-GSM (For definition of frequency bands see GSM 05.05)
Note: For mobile station supporting the R-GSM band further information can be found in MS Classmark 3.

~~When the DCS 1800 band is used (for exceptions see GSM 04.18:~~
~~Bit 1~~
~~0 Reserved for future use (for definition of frequency bands see GSM 05.05)~~

~~Note: This bit conveys no information about support or non support of the E-GSM or R-GSM band when transmitted on a DCS 1800 channel.~~

~~When the PCS 1900 band is used (for exceptions see GSM 04.18:~~
~~Bit **1**~~
~~0 Reserved for future use (for definition of frequency bands see GSM 05.05)~~

~~Note: This bit conveys no information about support or non support of the E-GSM or R-GSM band when transmitted on a PCS 1900 channel.~~

Table 10.5.6a/3GPP TS 24.008: *Mobile Station Classmark 2* information element

CM3 (octet 5, bit 8) ~~Required for MS supporting GSM.~~ 0 The MS does not support any options that are indicated in CM3 1 The MS supports options that are indicated in classmark 3 IE
LCS VA capability (LCS value added location request notification capability) (octet 5,bit 6) ~~Required for MS supporting GSM.~~~~and UMTS~~ 0 LCS value added location request notification capability not supported 1 LCS value added location request notification capability supported
UCS2 treatment (octet 5, bit 5) ~~Required for MS suppporting UMTS.~~ This information field indicates the likely treatment by the mobile station of UCS2 encoded character strings. For backward compatibility reasons, ~~I~~if this field is not included, the value 0 shall be assumed by the receiver. 0 The ME has a preference for the default alphabet (defined in GSM 03.38) over UCS2. 1 The ME has no preference between the use of the default alphabet and the use of UCS2.
SoLSA (octet 5, bit 4) ~~Required for MS supporting GSM.~~ An MS not supporting GSM shall set this bit to '0'. An MS supporting GSM shall indicate the associated GSM capability (see table): 0 The ME does not support SoLSA. 1 The ME supports SoLSA.
CMSP: CM Service Prompt (octet 5, bit 3) $(CCBS)$ ~~Required for MS supporting GSM and UMTS.~~ 0 "Network initiated MO CM connection request" not supported. 1 "Network initiated MO CM connection request" supported for at least one CM protocol.
A5/3 algorithm supported (octet 5, bit 2) ~~Required for MS supporting GSM.~~ An MS not supporting GSM shall set this bit to '0'. An MS supporting GSM shall indicate the associated GSM capability (see table): 0 encryption algorithm A5/3 not available 1 encryption algorithm A5/3 available
A5/2 algorithm supported (octet 5, bit 1) ~~Required for MS supporting GSM.~~ An MS not supporting GSM shall set this bit to '0'. An MS supporting GSM shall indicate the associated GSM capability (see table): 0 encryption algorithm A5/2 not available 1 encryption algorithm A5/2 available

~~A MS supporting GSM shall always encode all fields relevant for GSM radio access technology, even when accessing UMTS radio access technology. A UMTS MS which does not support GSM shall encode fields relevant only for GSM radio access tecnology using any value which has been defined for this version of the protocol and is not reserved.~~

NOTE: Additional mobile station capability information might be obtained by invoking the classmark interrogation procedure when GSM is used~~the mobile station is accessing the GSM radio access technology~~.

10.5.1.7 Mobile Station Classmark 3

The purpose of the *Mobile Station Classmark 3* information element is to provide the network with information concerning aspects of the mobile station. The contents might affect the manner in which the network handles the operation of the mobile station. The Mobile Station Classmark information indicates general mobile station characteristics and it shall therefore, except for fields explicitly indicated, be independent of the frequency band of the channel it is sent on.
The *MS Classmark 3* is a type 4 information element with a maximum of 14 octets length.

The value part of a *MS Classmark 3* information element is coded as shown in figure 10.5.7/3GPP TS 24.008 and table 10.5.7/3GPP TS 24.008.

NOTE: The 14 octet limit is so that the CLASSMARK CHANGE message will fit in one layer 2 frame.

SEMANTIC RULE : a multiband mobile station shall provide information about all frequency bands it can support. A single band mobile station shall not indicate the band it supports in the *Multiband Supported, GSM 400 Bands Supported, GSM 850 Associated Radio Capability* or PCS *1900 Associated Radio Capability* fields in the MS Classmark 3. Due to shared radio frequency channel numbers between DCS 1800 and PCS 1900, the mobile should indicate support for either DCS 1800 band OR PCS 1900 band.

SEMANTIC RULE : a mobile station shall include the MS Measurement Capability field if the *Multi Slot Class* field contains a value of 19 or greater (see GSM 05.02).

Typically, the number of spare bits at the end is the minimum to reach an octet boundary. The receiver may add any number of bits set to "0" at the end of the received string if needed for correct decoding.

```
<Classmark 3 Value part> ::=
        < spare bit >                 [ These 2 lines added are to be removed at the incorporation of CR ]
        {       < Multiband supported : { 000 } >   bitmap indicating DCS 1800; EGSM/RGSM; PGSM.
                        < A5 bits >
        |       < Multiband supported : { 101 | 110 } >
                        < A5 bits >
                        < Associated Radio Capability 2 : bit(4) >
                        < Associated Radio Capability 1 : bit(4) >
        |       < Multiband supported : { 001 | 010 | 100 } >
                        < A5 bits >
                        < spare bit >(4)
                        < Associated Radio Capability 1 : bit(4) > }
        { 0 | 1 < R Support > }
        { 0 | 1 < Multi Slot Capability > }
        < UCS2 treatment: bit >
        < Extended Measurement Capability : bit >
        { 0 | 1 < MS measurement capability > }
        { 0 | 1 < MS Positioning Method Capability > }
        { 0 | 1 < EDGE Multi Slot Capability > }
        { 0 | 1 < EDGE Struct > }
        { 0 | 1 < GSM 400 Bands Supported : { 01 | 10 | 11 } >
                        < GSM 400 Associated Radio Capability: bit(4) > }

        { 0 | 1 <GSM 850 Associated Radio Capability : bit(4) > }
        { 0 | 1 <PCS 1900 Associated Radio Capability : bit(4) > }
        < UMTS FDD Radio Access Technology Capability : bit >
        < UMTS TDD Radio Access Technology Capability : bit >
        < CDMA 2000 Radio Access Technology Capability : bit >

        { 0 | 1     < DTM GPRS Multi Slot Sub-Class : bit(2) >
                        < MAC Mode Support : bit >
                        { 0 | 1< ~~EGPRS Support : bit~~ DTM EGPRS Multi Slot Sub-Class : bit(2) > } }
        { 0 | 1 < Single Band Support > }
        < spare bit >** ;

< A5 bits > ::=
        < A5/7 : bit > < A5/6 : bit > < A5/5 : bit > < A5/4 : bit >  ;

<R Support>::=
        < R-GSM band Associated Radio Capability : bit(3) > ;

< Multi Slot Capability > ::=
        < Multi Slot Class : bit(5) >  ;

< MS Measurement capability > ::=
        < SMS_VALUE : bit (4) >
        < SM_VALUE : bit (4) > ;

< MS Positioning Method Capability > ::=
        < MS Positioning Method : bit(5) > ;

< EDGE Multi Slot Capability > ::=
        < EDGE Multi Slot Class : bit(5) > ;

<EDGE Struct> ::=
        < Modulation Capability : bit >
        { 0 | 1 < EDGE RF Power Capability 1: bit(2) > }
        { 0 | 1 < EDGE RF Power Capability 2: bit(2) > } ;

< Single Band Support > ::=
        < GSMBand : bit(4) > ;
```

Figure 10.5.7/3GPP TS 24.008 *Mobile Station Classmark 3* information element

Table 10.5.7/3GPP TS 24.008: *Mobile Station Classmark 3* information element

Multiband Supported (3 bit field)

Band 1 supported (third bit of the field)

Bit	3	
	0	P-GSM not supported
	1	P-GSM supported

Band 2 supported (second bit of the field)

BIT	2	
	0	E-GSM or R-GSM not supported
	1	E-GSM or R-GSM supported

Band 3 supported (first bit of the field)

Bit	1	
	0	DCS 1800 not supported
	1	DCS 1800 supported

The indication of support of P-GSM band or E-GSM or R-GSM band is mutually exclusive.

When the 'Band 2 supported' bit indicates support of E-GSM or R-GSM, the presence of the <R Support> field, see below, indicates if the E-GSM or R-GSM band is supported.

In this version of the protocol, the sender indicates in this field either none, one or two of these 3 bands supported. ~~If only one band is indicated, the receiver shall ignore the Associated Radio Capability 2.~~

For single band mobile station or a mobile station supporting none of the GSM 900 bands(P-GSM, E-GSM and R-GSM)~~P-GSM (and hence none of E-GSM and R-GSM)~~ and DCS1800 bands, all bits are set to 0.

A5/4

Bit	1	
	0	Encryption algorithm A5/4 not available
	1	Encryption algorithm A5/4 available

A5/5

Bit	1	
	0	Encryption algorithm A5/5 not available
	1	Encryption algorithm A5/5 available

A5/6

Bit	1	
	0	Encryption algorithm A5/6 not available
	1	Encryption algorithm A5/6 available

A5/7

	0	Encryption algorithm A5/7 not available
	1	Encryption algorithm A5/7 available

Associated Radio capability 1 and 2 (4 bit fields)

If either of P-GSM or E-GSM or R-GSM is supported, the radio capability 1 field indicates the radio capability for P-GSM, E-GSM or R-GSM, and the radio capability 2 field indicates the radio capability for DCS1800 if supported, and is spare otherwise.

If none of P-GSM or E-GSM or R-GSM are supported, the radio capability 1 field indicates the radio capability for DCS1800, and the radio capability 2 field is spare.

The radio capability contains the binary coding of the power class associated with the band indicated in multiband support bits (see GSM 05.05).

(continued...)

Table 10.5.1.7/3GPP TS 24.008 (continued): *MS Classmark 3* information element

R Support In case where the R-GSM band is supported the R-GSM band associated radio capability field contains the binary coding of the power class associated (see GSM 05.05) (regardless of the number of GSM bands supported). A mobile station supporting the R-GSM band shall also when appropriate, (see 10.5.1.6) indicate its support in the 'FC' bit in the Mobile Station Classmark 2 information element. Note: the coding of the power class for P-GSM, E-GSM, R-GSM and DCS 1800 in radio capability 1 and/or 2 is different to that used in the Mobile Station Classmark 1 and Mobile Station Classmark 2 information elements.   Multi Slot Class (5 bit field) In case the MS supports the use of multiple timeslots then the Multi Slot Class field is coded as the binary representation of the multislot class defined in 3GPP TS GSM 05.02. **UCS2 treatment** (1 bit field) This information field indicates the likely treatment by the mobile station of UCS2 encoded character strings. If not included, the value 0 shall be assumed by the receiver. Bit 1 0 the ME has a preference for the default alphabet (defined in GSM 03.38) over UCS2. 1 the ME has no preference between the use of the default alphabet and the use of UCS2. **Extended Measurement Capability (1 bit field)** This bit indicates whether the mobile station supports 'Extended Measurements' or not Bit 1 0 the MS does not support Extended Measurements 1 the MS supports Extended Measurements **SMS_VALUE (Switch-Measure-Switch)** (4 bit field) The SMS field indicates the time needed for the mobile station to switch from one radio channel to another, perform a neighbour cell power measurement, and the switch from that radio channel to another radio channel. Bits 4 3 2 1 0 0 0 0 1/4 timeslot (~144 microseconds) 0 0 0 1 2/4 timeslot (~288 microseconds) 0 0 1 0 3/4 timeslot (~433 microseconds) . . . 1 1 1 1 16/4 timeslot (~2307 microseconds) **SM_VALUE (Switch-Measure)** (4 bit field) The SM field indicates the time needed for the mobile station to switch from one radio channel to another and perform a neighbour cell power measurement. Bits 4 3 2 1 0 0 0 0 1/4 timeslot (~144 microseconds) 0 0 0 1 2/4 timeslot (~288 microseconds) 0 0 1 0 3/4 timeslot (~433 microseconds) . . . 1 1 1 1 16/4 timeslot (~2307 microseconds) **MS Positioning Method Capability (1 bit field)** This bit indicates whether the MS supports Positioning Method or not for the provision of Location Services. **MS Positioning Method** (5 bit field) This field indicates the Positioning Method(s) supported by the mobile station. MS assisted E-OTD Bit 5 0 MS assisted E-OTD not supported 1 MS assisted E-OTD supported

Table 10.5.1.7/3GPP TS 24.008 (continued): *MS Classmark 3* information element

<u>MS based E-OTD</u>
Bit 4
0 MS based E-OTD not supported
1 MS based E-OTD supported

<u>MS assisted GPS</u>
Bit 3
0 MS assisted GPS not supported
1 MS assisted GPS supported

<u>MS based GPS</u>
Bit 2
0 MS based GPS not supported
1 MS based GPS supported

<u>MS conventional GPS</u>
Bit 1
0 conventional GPS not supported
1 conventional GPS supported

EDGE Multi Slot class (5 bit field)

In case the EDGE MS supports the use of multiple timeslots and the number of supported time slots is different from number of time slots supported for GMSK then the EDGE Multi Slot class field is included and is coded as the binary representation of the multislot class defined in 3GPP TS GSM 05.02.

Modulation Capability

Modulation Capability field indicates the supported modulation scheme by MS in addition to GMSK
Bit 1
0 8-PSK supported for downlink reception only
1 8-PSK supported for uplink transmission and downlink reception

EDGE RF Power Capability 1 (2 bit field)

If 8-PSK is supported for both uplink and downlink, the **EDGE RF Power Capability 1** field indicates the radio capability for GSM900.

The radio capability contains the binary coding of the EDGE power class(see GSMß05.05).

EDGE RF Power Capability 2 (2 bit field)
If 8-PSK is supported for both uplink and downlink, the **EDGE RF Power Capability 2** field indicates the radio capability for DCS1800 or PCS1900 if supported, and is not included otherwise.
The radio capability contains the binary coding of the EDGE power class (see GSM 05.05).

Table 10.5.1.7/3GPP TS 24.008 (continued): *MS Classmark 3* information element

GSM 400 Bands Supported (2 bit field)
See the semantic rule for the sending of this field.
Bits
2 1

0 1	GSM 480 supported, GSM 450 not supported
1 0	GSM 450 supported, GSM 480 not supported
1 1	GSM 450 supported, GSM 480 supported

GSM 400 Associated Radio Capability (4 bit field)
If either GSM 450 or GSM 480 or both is supported, the GSM 400 Associated Radio Capability field indicates the radio capability for GSM 450 and/or GSM 480.

The radio capability contains the binary coding of the power class associated with the band indicated in GSM 400 Bands Supported bits (see GSM 05.05).

Note: the coding of the power class for GSM 450 and GSM 480 in GSM 400 Associated Radio Capability is different to that used in the Mobile Station Classmark 1 and Mobile Station Classmark 2 information elements.

GSM 850 Associated Radio Capability (4 bit field)

See the semantic rule for the sending of this field. This field indicates whether GSM 850 band is supported and its associated radio capability.

The radio capability contains the binary coding of the power class associated with the GSM 850 band (see GSM 05.05).

Note: the coding of the power class for GSM 850 in GSM 850 Associated Radio Capability is different to that used in the Mobile Station Classmark 1 and Mobile Station Classmark 2 information elements.

PCS 1900 Associated Radio Capability (4 bit field)

See the semantic rule for the sending of this field. This field indicates whether PCS 1900 band is supported and its associated radio capability.

The radio capability contains the binary coding of the power class associated with the PCS 1900 band (see GSM 05.05).

Note: the coding of the power class for PCS 1900 in PCS 1900 Associated Radio Capability is different to that used in the Mobile Station Classmark 1 and Mobile Station Classmark 2 information elements.

Table 10.5.1.7/3GPP TS 24.008 (continued): *MS Classmark 3* information element

UMTS FDD Radio Access Technology Capability (1 bit field)
Bit 1
0 UMTS FDD not supported
1 UMTS FDD supported

UMTS TDD Radio Access Technology Capability (1 bit field)
Bit 1
0 UMTS TDD not supported
1 UMTS TDD supported

CDMA 2000 Radio Access Technology Capability (1 bit field)
Bit 1
0 CDMA2000 not supported
1 CDMA2000 supported

DTM GPRS Multi Slot Sub-Class (2 bit field)
This field indicates the GPRS DTM capabilities of the MS. The DTM GPRS Multi Slot Sub-Class is independent from the Multi Slot Capabilities field. It is coded as follows:
Bit 2 1
0 0 Sub-Class 1 supported
0 1 Sub-Class 5 supported
1 0 Sub-Class 9 supported
1 1 Reserved for future extension. If received, the network shall interpret this as '00'

DTM EGPRS Multi Slot Sub-Class (2 bit field)
This field indicates the EGPRS DTM capabilities of the MS. The DTM EGPRS Multi Slot Sub-Class is independent from the Multi Slot Capabilities field. This field shall be included only if the mobile station supports EGPRS DTM. This field is coded as the DTM GPRS Multi Slot Sub-Class field.

MAC Mode Support (1 bit field)
This field indicates whether the MS supports Dynamic and Fixed Allocation or only supports Exclusive Allocation. It is coded as follows:
Bit 1
0 Dynamic and Fixed Allocation not supported
1 Dynamic and Fixed allocation supported

Single Band Support

This field shall be sent if the mobile station supports one and only one GSM band with the exception of R-GSM; this field shall not be sent otherwise.

GSMBand (4 bit field)
Bits
4 3 2 1
0 0 0 0 E-GSM is supported
0 0 0 1 P-GSM is supported
0 0 1 0 DCS 1800 is supported
0 0 1 1 GSM 450 is supported
0 1 0 0 GSM 480 is supported
0 1 0 1 GSM 850 is supported
0 1 1 0 PCS 1900 is supported
All other values are reserved for future use.

NOTE: When this field is received, the associated RF Power capability is found in Classmark1 or 2.

***** Extract of TS 04.18 Version 8.8.0 for information *****

9.1.11 Classmark change

This message is sent on the main DCCH by the mobile station to the network to indicate a classmark change or as a response to a classmark enquiry. See Table 9.1.11.1/3GPP TS 04.18.

Message type: CLASSMARK CHANGE

Significance: dual

Direction: mobile station to network

Table 9.1.11.1/3GPP TS 04.18: CLASSMARK CHANGE message content

IEI	Information element	Type / Reference	Presence	Format	length
	RR management Protocol Discriminator	Protocol Discriminator 10.2	M	V	1/2
	Skip Indicator	Skip Indicator 10.3.1	M	V	1/2
	Classmark Change Message Type	Message Type 10.4	M	V	1
	Mobile Station Classmark	Mobile Station Classmark 2 10.5.1.6	M	LV	4
20	Additional Mobile Station Classmark Information	Mobile Station Classmark 3 10.5.1.7	C	TLV	3-14

9.1.11.1 Additional Mobile Station Classmark Information

This IE shall be included if and only if the CM3 bit in the *Mobile Station Classmark* IE is set to 1.

9.1.11.2 Mobile Station Classmark

This IE shall include for multiband MS the Classmark 2 corresponding to the frequency band in use.

3GPP TSG_CN WG1 Meeting #17
Puerto Rico, USA
14 - 18 May, 2001

Tdoc N1-010838
(Rev of N1-010811 rev of N1-010622)

CR-Form-v3

CHANGE REQUEST

⌘	**24.008**	**CR**	**403**	⌘	rev	**2**	⌘	Current version:	**4.2.0**	⌘

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

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

Title:	⌘	CLASSMARK1, 2 and 3 corrections.		
Source:	⌘	Vodafone		
Work item code:	⌘	TEI	***Date:*** ⌘	2nd May 2001
Category:	⌘	**A**	***Release:*** ⌘	REL-4
		Use one of the following categories: ***F*** *(essential correction)* ***A*** *(corresponds to a correction in an earlier release)* ***B*** *(Addition of feature),* ***C*** *(Functional modification of feature)* ***D*** *(Editorial modification)* Detailed explanations of the above categories can be found in 3GPP TR 21.900.		*Use one of the following releases:* *2 (GSM Phase 2)* *R96 (Release 1996)* *R97 (Release 1997)* *R98 (Release 1998)* *R99 (Release 1999)* *REL-4 (Release 4)* *REL-5 (Release 5)*

Reason for change:	⌘	When access is performed via UMTS channels, information on GSM MS capability is *not* irrelevant as this is used by the RNC for decision on **handover from UMTS to GSM**. The required specification of the coding of CLASSMARK doesn't allow the network to distinguish unambiguously single banded GSM mobiles. This may lead to dropped calls if the network can't understand the precise implementation in the mobile. Also it is impossible for a UMTS mobile not supporting GSM to signal its capability without misleadingly indicate that it supports GSM as well.
Summary of change:	⌘	It is clarified how a single band GSM mobile and a UMTS only mobile need to code the classmarks 1, 2 and 3.
Consequences if not approved:	⌘	There will be a misunderstanding between network and mobile as to what coding schemes and handovers are supported. This can lead to the network attempting to hand the mobile some where or send information which the mobile is unable to decode.

Clauses affected:	⌘	10.5.1.5, 10.5.1.6 & 10.5.1.7	
Other specs affected:	⌘	[ ] Other core specifications [ ] Test specifications [ ] O&M Specifications	⌘
Other comments:	⌘	An almost identical CR against Release 99 appears in CR402	

How to create CRs using this form:

Comprehensive information and tips about how to create CRs can be found at: http://www.3gpp.org/3G_Specs/CRs.htm. Below is a brief summary:

1) Fill out the above form. The symbols above marked ⌘ contain pop-up help information about the field that they are closest to.

2) Obtain the latest version for the release of the specification to which the change is proposed. Use the MS Word "revision marks" feature (also known as "track changes") when making the changes. All 3GPP specifications can be downloaded from the 3GPP server under ftp://www.3gpp.org/specs/ For the latest version, look for the directory name with the latest date e.g. 2000-09 contains the specifications resulting from the September 2000 TSG meetings.

3) With "track changes" disabled, paste the entire CR form (use CTRL-A to select it) into the specification just in front of the clause containing the first piece of changed text. Delete those parts of the specification which are not relevant to the change request.

10.5.1.5 Mobile Station Classmark 1

The purpose of the *Mobile Station Classmark 1* information element is to provide the network with information concerning aspects of high priority of the mobile station equipment. This affects the manner in which the network handles the operation of the mobile station. The Mobile Station Classmark information indicates general mobile station characteristics and it shall therefore, except for fields explicitly indicated, be independent of the frequency band of the channel it is sent on.

The *Mobile Station Classmark 1* information element is coded as shown in figure 10.5.5/3GPP TS 24.008 and table 10.5.5/3GPP TS 24.008.

The *Mobile Station Classmark 1* is a type 3 information element with 2 octets length.

<table>
<tr><td>8</td><td>7</td><td>6</td><td>5</td><td>4</td><td>3</td><td>2</td><td>1</td><td></td></tr>
<tr><td></td><td colspan="7">Mobile Station Classmark 1 IEI</td><td>octet 1</td></tr>
<tr><td>0
spare</td><td colspan="2">Revision
level</td><td>ES
IND</td><td>A5/1</td><td colspan="3">RF power
capability</td><td>octet 2</td></tr>
</table>

Figure 10.5.5/3GPP TS 24.008 *Mobile Station Classmark 1* information element

~~A MS supporting GSM shall always encode all fields relevant for GSM radio access technology, even when accessing UMTS radio access technology. A UMTS MS which does not support GSM shall encode fields relevant only for GSM radio access tecnology using any value which has been defined for this version of the protocol and is not reserved.~~

Table 10.5.5/3GPP TS 24.008: *Mobile Station Classmark 1* information element

Revision level (octet 2)
~~Required for MS supporting GSM and UMTS.~~
Bits
7 6
0 0 Reserved for GSM phase 1
0 1 Used by GSM phase 2 mobile stations
1 0 Used by mobile stations supporting R99 or later versions of the protocol
1 1 Reserved for future use

ES IND (octet 2, bit 5) "Controlled Early Classmark Sending" option implementation
~~Required for MS supporting GSM.~~
An MS not supporting GSM shall set this bit to '0'.
An MS supporting GSM shall indicate the associated GSM capability (see table):

0 "Controlled Early Classmark Sending" option is not implemented in the MS
1 "Controlled Early Classmark Sending" option is implemented in the MS

NOTE: The value of the ES IND gives the implementation in the MS. It's value is **not** dependent on the broadcast SI 3 Rest Octet <Early Classmark Sending Control> value.

A5/1 algorithm supported (octet 2, bit4)
~~Required for mobile station supporting GSM.~~
An MS not supporting GSM shall set this bit to '1'.
An MS supporting GSM shall indicate the associated GSM capability (see table):

0 Encryption algorithm A5/1 available
1 Encryption algorithm A5/1 not available

RF power capability (octet 2)
~~Required for mobile stations supporting GSM.~~
When GSM 450, GSM 480, GSM 700, GSM 850, GSM 900 P, E [or R] band is used (for exceptions see 04.18), the MS shall indicate the RF power capability of the band used (see table):
When UMTS is used, a single band GSM 450, GSM 480, GSM 700, GSM 850, GSM 900 P, E [or R] MS shall indicate the RF power capability corresponding to the (GSM) band it supports (see table). In this case information on which single band is supported is found in classmark 3.

Bits
3 2 1
0 0 0 class 1
0 0 1 class 2
0 1 0 class 3
0 1 1 class 4
1 0 0 class 5

All other values are reserved.

When the DCS 1800 or PCS 1900 band is used (for exceptions see 3GPP TS 44.018, sub-clause 3.4.18), the MS shall indicate the RF power capability of the band used (see table):
When UMTS is used, a single band DCS 1800 or PCS 1900 MS shall indicate the RF power capability corresponding to the (GSM) band it supports (see table). In this case, information on which single band is supported is found in classmark 3.

Bits
3 2 1
0 0 0 class 1
0 0 1 class 2
0 1 0 class 3

All other values are reserved.

When UMTS is used, an MS not supporting any GSM band or a multiband GSM MS shall code this field as follows (see table):

Bits
3 2 1
1 1 1 RF power capability is irrelevant in this information element.
All other values are reserved.

10.5.1.6 Mobile Station Classmark 2

The purpose of the *Mobile Station Classmark 2* information element is to provide the network with information concerning aspects of both high and low priority of the mobile station equipment. This affects the manner in which the network handles the operation of the mobile station. The Mobile Station Classmark information indicates general mobile station characteristics and it shall therefore, except for fields explicitly indicated, be independent of the frequency band of the channel it is sent on.

The *Mobile Station Classmark 2* information element is coded as shown in figure 10.5.6/3GPP TS 24.008, table 10.5.6a/3GPP TS 24.008 and table 10.5.6b/3GPP TS 24.008.

The *Mobile Station Classmark 2* is a type 4 information element with 5 octets length.

8	7	6	5	4	3	2	1	
	Mobile station classmark 2 IEI							octet 1
Length of mobile station classmark 2 contents								octet 2
0 spare	Revision level		ES IND	A5/1	RF power Capability			octet 3
0 spare	PS capa.	SS Screen. Indicator		SM ca pabi.	VBS	VGCS	FC	octet 4
CM3	0 spare	LCSVA CAP	UCS2	SoLSA	CMSP	A5/3	A5/2	octet 5

NOTE: Owing to backward compatibility problems, bit 8 of octet 4 should not be used unless it is also checked that the bits 8, 7 and 6 of octet 3 are not "0 0 0".

Figure 10.5.6/3GPP TS 24.008 *Mobile Station Classmark 2* information element

Table 10.5.6a/3GPP TS 24.008: *Mobile Station Classmark 2* information element

Revision level (octet 3)
~~Required for MS supporting GSM and UMTS.~~
Bits
7 6
0 0 Reserved for GSM phase 1
0 1 Used by GSM phase 2 mobile stations
1 0 Used by mobile stations supporting R99 or later versions of the protocol
1 1 Reserved for future use

ES IND (octet 3, bit 5) "Controlled Early Classmark Sending" option implementation
~~Required for MS supporting GSM.~~
AN MS not supporting GSM shall set this bit to ‘0’.
An MS supporting GSM shall indicate the associated GSM capability (see table):

0 "Controlled Early Classmark Sending" option is not implemented in the MS
1 "Controlled Early Classmark Sending" option is implemented in the MS

NOTE: The value of the ES IND gives the implementation in the MS. It's value is **not** dependent on the broadcast SI 3 Rest Octet <Early Classmark Sending Control> value

Table 10.5.6a/3GPP TS 24.008: *Mobile Station Classmark 2* information element

A5/1 algorithm supported (octet 3, bit 4)
~~Required for MS supporting GSM.~~
An MS not supporting GSM shall set this bit to '1'.
An MS supporting GSM shall indicate the associated GSM capability (see table)

0	encryption algorithm A5/1 available
1	encryption algorithm A5/1 not available

RF Power Capability (Octet 3)
~~Required for MS supporting GSM.~~
When GSM 450, GSM 480, GSM 700, GSM 850, GSM 900 P, E [or R] band is used (for exceptions see 3GPP TS 44.018), the MS shall indicate the RF power capability of the band used (see table).
When UMTS is used, a single band GSM 450, GSM 480, GSM 700, GSM 850, GSM 900 P, E [or R] MS shall indicate the RF power capability corresponding to the (GSM) band it supports (see table). In this case, information on which single band is supported is found in classmark 3.~~:~~

Bits

3	**2**	**1**	
0	0	0	class 1
0	0	1	class 2
0	1	0	class 3
0	1	1	class 4
1	0	0	class 5

All other values are reserved.

When the DCS 1800 or PCS 1900 band is used (for exceptions see ~~3~~44.018)~~:~~. The MS shall indicate the RF power capability of the band used (see table).
When UMTS is used, a single band DCS 1800 or PCS 1900 MS shall indicate the RF power capability corresponding to the (GSM) band it supports (see table). In this case, information on which single band is supported is found in classmark 3.

Bits

3	**2**	**1**	
0	0	0	class 1
0	0	1	class 2
0	1	0	class 3

All other values are reserved.
When UMTS is used, an MS not supporting any GSM band or a multiband GSM MS shall code this field as follows (see table):

Bits

3	**2**	**1**	
1	1	1	RF Power capability is irrelevant in this information element

All other values are reserved.

PS capability (pseudo-synchronization capability) (octet 4)
~~Required for MS supporting GSM~~
An MS not supporting GSM shall set this bit to '0'.
An MS supporting GSM shall indicate the associated GSM capability (see table):

Bit **7**	
0	PS capability not present
1	PS capability present

SS Screening Indicator (octet 4)
~~Required for MS supporting GSM and UMTS~~
Bits

6	**5**	
0	0	defined in 3GPP TS 24.080
0	1	defined in 3GPP TS 24.080
1	0	defined in 3GPP TS 24.080
1	1	defined in 3GPP TS 24.080

Table 10.5.6a/3GPP TS 24.008: *Mobile Station Classmark 2* information element

SM capability (MT SMS pt to pt capability) (octet 4)
~~Required for MS supporting GSM.~~
Bit **4**

0	Mobile station does not support mobile terminated point to point SMS
1	Mobile station supports mobile terminated point to point SMS

~~Table 10.5.6a/3GPP TS 24.008: *Mobile Station Classmark 2* information element~~

VBS notification reception (octet 4)
~~Required for MS supporting GSM.~~
An MS not supporting GSM shall set this bit to '0'.
An MS supporting GSM shall indicate the associated GSM capability (see table):

Bit **3**

0	no VBS capability or no notifications wanted
1	VBS capability and notifications wanted

VGCS notification reception (octet 4)
~~Required for MS supporting GSM.~~
An MS not supporting GSM shall set this bit to '0'.
An MS supporting GSM shall indicate the associated GSM capability (see table):

Bit **2**

0	no VGCS capability or no notifications wanted
1	VGCS capability and notifications wanted

FC Frequency Capability (octet 4)
~~Required for MS supporting GSM.~~
When the GSM 400, or GSM 700, or GSM 850, or DCS 1800, or PCS 1900 band or UMTS ~~band~~ is used (for exceptions see 3GPP TS 44.018, for definitions of frequency band see 3GPP TS 45.005), this bit shall be sent with the value '0'.~~:~~

~~Bit 1~~
~~0~~ ~~Reserved for future use (for definition of frequency bands see 3GPP TS 05.05)~~

Note: This bit conveys no information about support or non support of the E-GSM or R-GSM bands when ~~transmitted on a~~ GSM 400, GSM 700, GSM 850, DCS 1800, PCS 1900 band or UMTS is used ~~channel~~.

~~When GSM 700 band is used (for exceptions see 3GPP TS 44.018):~~
~~Bit 1~~
~~0~~ ~~Reserved for future use (for definition of frequency bands see 3GPP TS 05.05)~~

~~Note: This bit conveys no information about support or non support of the E-GSM or R-GSM band when transmitted on a GSM 700 channel.~~

~~When GSM 850 band is used (for exceptions see 3GPP TS 44.018):~~
~~Bit 1~~
~~0~~ ~~Reserved for future use (for definition of frequency bands see 3GPP TS 05.05)~~

~~Note: This bit conveys no information about support or non support of the E-GSM or R-GSM band when transmitted on a GSM 850 channel.~~

Table 10.5.6a/3GPP TS 24.008: *Mobile Station Classmark 2* information element

When a GSM 900 band is used (for exceptions see 3GPP TS 44.018):
Bit **1**

0	The MS does not support the E-GSM or R-GSM band (For definition of frequency bands see 3GPP TS 05.05)
1	The MS does support the E-GSM or R-GSM (For definition of frequency bands see 3GPP TS 05.05)

Note: For mobile station supporting the R-GSM band further information can be found in MS Classmark 3.

~~When the DCS 1800 band is used (for exceptions see 3GPP TS 44.018):~~
~~Bit **1**~~

~~0~~	~~Reserved for future use (for definition of frequency bands see 3GPP TS 05.05)~~

~~Note: This bit conveys no information about support or non support of the E-GSM or R-GSM band when transmitted on a DCS 1800 channel.~~

~~When the PCS 1900 band is used (for exceptions see 3GPP TS 44.018):~~
~~Bit **1**~~

~~0~~	~~Reserved for future use (for definition of frequency bands see 3GPP TS 05.05)~~

~~Note: This bit conveys no information about support or non support of the E-GSM or R-GSM band when transmitted on a PCS 1900 channel.~~

CM3 (octet 5, bit 8)
~~Required for MS supporting GSM.~~

0	The MS does not support any options that are indicated in CM3
1	Classmark 3 information is available

LCS VA capability (LCS value added location request notification capability) (octet 5,bit 6)
~~Required for MS supporting GSM and UMTS.~~

0	LCS value added location request notification capability not supported
1	LCS value added location request notification capability supported

UCS2 treatment (octet 5, bit 5)
~~Required for MS suppporting UMTS.~~
This information field indicates the likely treatment by the mobile station of UCS2 encoded character strings. For backward compatibility reasons, i~~I~~f this field is not included, the value 0 shall be assumed by the receiver.

0	the ME has a preference for the default alphabet (defined in 3GPP TS 03.38) over UCS2.
1	the ME has no preference between the use of the default alphabet and the use of UCS2.

Table 10.5.6a/3GPP TS 24.008: *Mobile Station Classmark 2* information element

SoLSA (octet 5, bit 4)	
~~Required for MS supporting GSM.~~	
An MS not supporting GSM shall set this bit to '0'.	
An MS supporting GSM shall indicate the associated GSM capability (see table):	
0	The ME does not support SoLSA.
1	The ME supports SoLSA.
CMSP: CM Service Prompt (octet 5, bit 3) $(CCBS)$	
~~Required for MS supporting GSM and UMTS.~~	
0	"Network initiated MO CM connection request" not supported.
1	"Network initiated MO CM connection request" supported for at least one CM protocol.
A5/3 algorithm supported (octet 5, bit 2)	
~~Required for MS supporting GSM.~~	
An MS not supporting GSM shall set this bit to '0'.	
An MS supporting GSM shall indicate the associated GSM capability (see table):	
0	encryption algorithm A5/3 not available
1	encryption algorithm A5/3 available
A5/2 algorithm supported (octet 5, bit 1)	
~~Required for MS supporting GSM.~~	
An MS not supporting GSM shall set this bit to '0'.	
An MS supporting GSM shall indicate the associated GSM capability (see table):	
0	encryption algorithm A5/2 not available
1	encryption algorithm A5/2 available

~~A MS supporting GSM shall always encode all fields relevant for GSM radio access technology, even when accessing UMTS radio access technology. A UMTS MS which does not support GSM shall encode fields relevant only for GSM radio access tecnology using any value which has been defined for this version of the protocol and is not reserved.~~

NOTE: Additional mobile station capability information might be obtained by invoking the classmark interrogation procedure when GSM is used~~the mobile station is accessing the GSM radio access technology~~.

10.5.1.7 Mobile Station Classmark 3

The purpose of the *Mobile Station Classmark 3* information element is to provide the network with information concerning aspects of the mobile station. The contents might affect the manner in which the network handles the operation of the mobile station. The Mobile Station Classmark information indicates general mobile station characteristics and it shall therefore, except for fields explicitly indicated, be independent of the frequency band of the channel it is sent on.

The *MS Classmark 3* is a type 4 information element with a maximum of 14 octets length.

The value part of a *MS Classmark 3* information element is coded as shown in figure 10.5.7/3GPP TS 24.008 and table 10.5.7/3GPP TS 24.008.

NOTE: The 14 octet limit is so that the CLASSMARK CHANGE message will fit in one layer 2 frame.

SEMANTIC RULE : a multiband mobile station shall provide information about all frequency bands it can support. A single band mobile station shall not indicate the band it supports in the *Multiband Supported, GSM 400 Bands Supported, GSM 700 Associated Radio Capability, GSM 850 Associated Radio Capability* or PCS *1900 Associated Radio Capability* fields in the MS Classmark 3. Due to shared radio frequency channel numbers between DCS 1800 and PCS 1900, the mobile should indicate support for either DCS 1800 band OR PCS 1900 band.

SEMANTIC RULE : a mobile station shall include the MS Measurement Capability field if the *Multi Slot Class* field contains a value of 19 or greater (see 3GPP TS 05.02).

Typically, the number of spare bits at the end is the minimum to reach an octet boundary. The receiver may add any number of bits set to "0" at the end of the received string if needed for correct decoding.

```
<Classmark 3 Value part> ::=
    < spare bit >
    {    < Multiband supported : { 000 } >
            < A5 bits >
    |    < Multiband supported : { 101 | 110 } >
            < A5 bits >
            < Associated Radio Capability 2 : bit(4) >
            < Associated Radio Capability 1 : bit(4) >
    |    < Multiband supported : { 001 | 010 | 100 } >
            < A5 bits >
            < spare bit >(4)
            < Associated Radio Capability 1 : bit(4) > }
    { 0 | 1 < R Support > }
    { 0 | 1 < Multi Slot Capability > }
    < UCS2 treatment: bit >
    < Extended Measurement Capability : bit >
    { 0 | 1 < MS measurement capability > }
    { 0 | 1 < MS Positioning Method Capability > }
    { 0 | 1 < EDGE Multi Slot Capability > }
    { 0 | 1 < EDGE Struct > }
    { 0 | 1 < GSM 400 Bands Supported : { 01 | 10 | 11 } >
            < GSM 400 Associated Radio Capability: bit(4) > }

    { 0 | 1 <GSM 850 Associated Radio Capability : bit(4) > }
    { 0 | 1 <PCS 1900 Associated Radio Capability : bit(4) > }
    < UMTS FDD Radio Access Technology Capability : bit >
    < UMTS 3.84 Mcps TDD Radio Access Technology Capability : bit >
    < CDMA 2000 Radio Access Technology Capability : bit >

    { 0 | 1  < DTM GPRS Multi Slot Sub-Class : bit(2) >
            < MAC Mode Support : bit >
            {0 | 1< DTM EGPRS Multi Slot Sub-Class : bit(2) > } }
    { 0 | 1 < Single Band Support > } -- Release 4 starts here:
    {0 | 1 <GSM 700 Associated Radio Capability : bit(4)>}

    < UMTS 1.28 Mcps TDD Radio Access Technology Capability : bit >
    < spare bit > ;

< A5 bits > ::=
    < A5/7 : bit > < A5/6 : bit > < A5/5 : bit > < A5/4 : bit >  ;

<R Support>::=
    < R-GSM band Associated Radio Capability : bit(3) > ;

< Multi Slot Capability > ::=
    < Multi Slot Class : bit(5) >  ;

< MS Measurement capability > ::=
    < SMS_VALUE : bit (4) >
    < SM_VALUE : bit (4) > ;

< MS Positioning Method Capability > ::=
    < MS Positioning Method : bit(5) > ;

< EDGE Multi Slot Capability > ::=
    < EDGE Multi Slot Class : bit(5) > ;

 <EDGE Struct> : :=
    < Modulation Capability : bit >
    { 0 | 1 < EDGE RF Power Capability 1: bit(2) > }
    { 0 | 1 < EDGE RF Power Capability 2: bit(2) > }

< Single Band Support > ::=
    < GSM Band : bit (4) > ;
```

Figure 10.5.7/3GPP TS 24.008 *Mobile Station Classmark 3* information element

Table 10.5.7/3GPP TS 24.008: *Mobile Station Classmark 3* information element

Multiband Supported (3 bit field)

Band 1 supported (third bit of the field)
Bit 3
0 P-GSM not supported
1 P-GSM supported

Band 2 supported (second bit of the field)
BIT 2
0 E-GSM or R-GSM not supported
1 E-GSM or R-GSM supported

Band 3 supported (first bit of the field)
Bit 1
0 DCS 1800 not supported
1 DCS 1800 supported

The indication of support of P-GSM band or E-GSM or R-GSM band is mutually exclusive.

When the 'Band 2 supported' bit indicates support of E-GSM or R-GSM, the presence of the <R Support> field, see below, indicates if the E-GSM or R-GSM band is supported.

In this version of the protocol, the sender indicates in this field either none, one or two of these 3 bands supported. ~~If only one band is indicated, the receiver shall ignore the Associated Radio Capability 2.~~

For single band mobile station or a mobile station supporting none of the GSM 900 bands(P-GSM, E-GSM and R-GSM) and DCS 1800 bands, all bits are set to 0.

A5/4
Bit 1
0 Encryption algorithm A5/4 not available
1 Encryption algorithm A5/4 available

A5/5
Bit 1
0 Encryption algorithm A5/5 not available
1 Encryption algorithm A5/5 available

A5/6
Bit 1
0 Encryption algorithm A5/6 not available
1 Encryption algorithm A5/6 available

A5/7
0 Encryption algorithm A5/7 not available
1 Encryption algorithm A5/7 available

Associated Radio capability 1 and 2 (4 bit fields)

If either of P-GSM or E-GSM or R-GSM is supported, the radio capability 1 field indicates the radio capability for P-GSM, E-GSM or R-GSM, and the radio capability 2 field indicates the radio capability for DCS1800 if supported, and is spare otherwise.

If none of P-GSM or E-GSM or R-GSM are supported, the radio capability 1 field indicates the radio capability for DCS1800, and the radio capability 2 field is spare.

The radio capability contains the binary coding of the power class associated with the band indicated in multiband support bits (see GSMß05.05).

(continued...)

Table 10.5.1.7/3GPP TS 24.008 (continued): *MS Classmark 3* information element

R Support

In case where the R-GSM band is supported the R-GSM band associated radio capability field contains the binary coding of the power class associated (see GSM ~~0~~45.005) (regardless of the number of GSM bands supported). A mobile station supporting the R-GSM band shall also when appropriate, (see 10.5.1.6) indicate its support in the 'FC' bit in the Mobile Station Classmark 2 information element.

Note: the coding of the power class for P-GSM, E-GSM, R-GSM and DCS 1800 in radio capability 1 and/or 2 is different to that used in the Mobile Station Classmark 1 and Mobile Station Classmark 2 information elements.

Multi Slot Class (5 bit field)

In case the MS supports the use of multiple timeslots then the Multi Slot Class field is coded as the binary representation of the multislot class defined in TS GSM 05.02.

UCS2 treatment (1 bit field)

This information field indicates the likely treatment by the mobile station of UCS2 encoded character strings. If not included, the value 0 shall be assumed by the receiver.
Bit 1
0 the ME has a preference for the default alphabet (defined in 3GPP TS 03.38) over UCS2.
1 the ME has no preference between the use of the default alphabet and the use of UCS2.

Extended Measurement Capability (1 bit field)

This bit indicates whether the mobile station supports 'Extended Measurements' or not
Bit 1
0 the MS does not support Extended Measurements
1 the MS supports Extended Measurements

SMS_VALUE (Switch-Measure-Switch) (4 bit field)
The SMS field indicates the time needed for the mobile station to switch from one radio channel to another, perform a neighbour cell power measurement, and the switch from that radio channel to another radio channel.
Bits
4 3 2 1
0 0 0 0 1/4 timeslot (~144 microseconds)
0 0 0 1 2/4 timeslot (~288 microseconds)
0 0 1 0 3/4 timeslot (~433 microseconds)
. . .
1 1 1 1 16/4 timeslot (~2307 microseconds)

SM_VALUE (Switch-Measure) (4 bit field)
The SM field indicates the time needed for the mobile station to switch from one radio channel to another and perform a neighbour cell power measurement.
Bits
4 3 2 1
0 0 0 0 1/4 timeslot (~144 microseconds)
0 0 0 1 2/4 timeslot (~288 microseconds)
0 0 1 0 3/4 timeslot (~433 microseconds)
. . .
1 1 1 1 16/4 timeslot (~2307 microseconds)

MS Positioning Method Capability (1 bit field)
This bit indicates whether the MS supports Positioning Method or not for the provision of Location Services.

MS Positioning Method (5 bit field)
This field indicates the Positioning Method(s) supported by the mobile station.
MS assisted E-OTD
Bit 5
0 MS assisted E-OTD not supported
1 MS assisted E-OTD supported

Table 10.5.1.7/3GPP TS 24.008 (continued): *MS Classmark 3* information element

MS based E-OTD
Bit 4
0 MS based E-OTD not supported
1 MS based E-OTD supported

MS assisted GPS
Bit 3
0 MS assisted GPS not supported
1 MS assisted GPS supported

MS based GPS
Bit 2
0 MS based GPS not supported
1 MS based GPS supported

MS conventional GPS
Bit 1
0 conventional GPS not supported
1 conventional GPS supported

EDGE Multi Slot class (5 bit field)

In case the EDGE MS supports the use of multiple timeslots and the number of supported time slots is different from number of time slots supported for GMSK then the EDGE Multi Slot class field is included and is coded as the binary representation of the multislot class defined in TS GSM 05.02.

Modulation Capability

Modulation Capability field indicates the supported modulation scheme by MS in addition to GMSK
Bit 1
0 8-PSK supported for downlink reception only
1 8-PSK supported for uplink transmission and downlink reception

EDGE RF Power Capability 1 (2 bit field)

If 8-PSK is supported for both uplink and downlink, the **EDGE RF Power Capability 1** field indicates the radio capability for GSM700, GSM850 or GSM900.

The radio capability contains the binary coding of the EDGE power class(see GSMß05.05).

EDGE RF Power Capability 2 (2 bit field)
If 8-PSK is supported for both uplink and downlink, the **EDGE RF Power Capability 2** field indicates the radio capability for DCS1800 or PCS1900 if supported, and is not included otherwise.
The radio capability contains the binary coding of the EDGE power class (see 3GPP TS 05.05).

Table 10.5.1.7/3GPP TS 24.008 (continued): *MS Classmark 3* information element

GSM 400 Bands Supported (2 bit field)
See the semantic rule for the sending of this field.

Bits
2 1
0 1 GSM 480 supported, GSM 450 not supported
1 0 GSM 450 supported, GSM 480 not supported
1 1 GSM 450 supported, GSM 480 supported

GSM 400 Associated Radio Capability (4 bit field)
If either GSM 450 or GSM 480 or both is supported, the GSM 400 Associated Radio Capability field indicates the radio capability for GSM 450 and/or GSM 480.

The radio capability contains the binary coding of the power class associated with the band indicated in GSM 400 Bands Supported bits (see 3GPP TS 05.05).

Note: the coding of the power class for GSM 450 and GSM 480 in GSM 400 Associated Radio Capability is different to that used in the Mobile Station Classmark 1 and Mobile Station Classmark 2 information elements.

GSM 850 Associated Radio Capability (4 bit field)

See the semantic rule for the sending of this field.

This field indicates whether GSM 850 band is supported and its associated radio capability.

The radio capability contains the binary coding of the power class associated with the GSM 850 band (see 3GPP TS 05.05).

Note: the coding of the power class for GSM 850 in GSM 850 Associated Radio Capability is different to that used in the Mobile Station Classmark 1 and Mobile Station Classmark 2 information elements.

PCS 1900 Associated Radio Capability (4 bit field)

See the semantic rule for the sending of this field. This field indicates whether PCS 1900 band is supported and its associated radio capability.

The radio capability contains the binary coding of the power class associated with the PCS 1900 band (see 3GPP TS 05.05).

Note: the coding of the power class for PCS 1900 in PCS 1900 Associated Radio Capability is different to that used in the Mobile Station Classmark 1 and Mobile Station Classmark 2 information elements.

Table 10.5.1.7/3GPP TS 24.008 (continued): ***MS Classmark 3*** **information element**

UMTS FDD Radio Access Technology Capability (1 bit field)
Bit 1
0 UMTS FDD not supported
1 UMTS FDD supported

UMTS 3.84 Mcps TDD Radio Access Technology Capability (1 bit field)
Bit 1
0 UMTS 3.84 Mcps TDD not supported
1 UMTS 3.84 Mcps TDD supported

CDMA 2000 Radio Access Technology Capability (1 bit field)
Bit 1
0 CDMA2000 not supported
1 CDMA2000 supported

DTM GPRS Multi Slot Sub-Class (2 bit field)
This field indicates the GPRS DTM capabilities of the MS. The GPRS DTM Multi Slot Sub-Class is independent from the Multi Slot Capabilities field. It is coded as follows:

Bit 2 1
0 0 Sub-Class 1 supported
0 1 Sub-Class 5 supported
1 0 Sub-Class 9 supported
1 1 Reserved for future extension. If received, the network shall interpret this as '00'

DTM EGPRS Multi Slot Sub-Class (2 bit field)
This field indicates the EGPRS DTM capabilities of the MS. The DTM EGPRS Multi Slot Sub-Class is independent from the Multi Slot Capabilities field. This field shall be included only if the mobile station supports EGPRS DTM. This field is coded as the DTM GPRS Multi Slot Sub-Class field.

MAC Mode Support (1 bit field)
This field indicates whether the MS supports Dynamic and Fixed Allocation or only supports Exclusive Allocation. It is coded as follows:

Bit 1
0 Dynamic and Fixed Allocation not supported
1 Dynamic and Fixed allocation supported

Single Band Support
This field shall be sent if the mobile station supports one and only one GSM band with the exception of R-GSM; this field shall not be sent otherwise.

GSM Band (4 bit field)
Bits
4 3 2 1
0 0 0 0 E-GSM is supported
0 0 0 1 P-GSM is supported
0 0 1 0 DCS 1800 is supported
0 0 1 1 GSM 450 is supported
0 1 0 0 GSM 480 is supported
0 1 0 1 GSM 850 is supported
0 1 1 0 PCS 1900 is supported
0 1 1 1 GSM 700 is supported
All other values are reserved for future use.

NOTE: When this field is received, the associated RF power capability is found in Classmark 1 or 2.

GSM 700 Associated Radio Capability (4 bit field)

See the semantic rule for the sending of this field.

This field indicates whether GSM 700 band is supported and its associated radio capability.

The radio capability contains the binary coding of the power class associated with the GSM 700 band (see 3GPP TS 05.05).

Note: the coding of the power class for GSM 700 in GSM 700 Associated Radio Capability is different to that used in the Mobile Station Classmark 1 and Mobile Station Classmark 2 information elements.

UMTS 1.28 Mcps TDD Radio Access Technology Capability (1 bit field)

Bit 1
0 UMTS 1.28 Mcps TDD not supported
1 UMTS 1.28 Mcps TDD supported

***** Extract of TS 44.018 for information *****

9.1.11 Classmark change

This message is sent on the main DCCH by the mobile station to the network to indicate a classmark change or as a response to a classmark enquiry. See Table 9.1.11.1/3GPP TS 44.018.

Message type: CLASSMARK CHANGE

Significance: dual

Direction: mobile station to network

Table 9.1.11.1/3GPP TS 04.18: CLASSMARK CHANGE message content

IEI	Information element	Type / Reference	Presence	Format	length
	RR management Protocol Discriminator	Protocol Discriminator 10.2	M	V	1/2
	Skip Indicator	Skip Indicator 10.3.1	M	V	1/2
	Classmark Change Message Type	Message Type 10.4	M	V	1
	Mobile Station Classmark	Mobile Station Classmark 2 10.5.1.6	M	LV	4
20	Additional Mobile Station Classmark Information	Mobile Station Classmark 3 10.5.1.7	C	TLV	3-14

9.1.11.1 Additional Mobile Station Classmark Information

This IE shall be included if and only if the CM3 bit in the *Mobile Station Classmark* IE is set to 1.

9.1.11.2 Mobile Station Classmark

This IE shall include for multiband MS the Classmark 2 corresponding to the frequency band in use.

3GPP TSG CN WG1 Meeting #17 **Tdoc N1-010908**
Puerto Rico, 14th - 18th May 2001 (Rev of N1-010797)

Title: Liaison Statement on "24.008 CR for Classmark Issues"

Source: TSG_CN WG1

To: TSG_GERAN, TSG_RAN WG2

cc:

Contact Person:

Name:	Duncan Mills	Hannu Hietalahti
E-mail Address:	duncan.mills@vf.vodafone.co.uk	Hannu.Hietalahti@nokia.com
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1. Overall Description:

TSG CN WG1 thanks TSG GERAN for their LS N1-010620 (GP-010847) dated 4th April 2001. CN1 would like to acknowledge the considerable work GERAN has done to create the CR to 24.008 on MS Classmark issues.

CN1 also recognises the need to ensure that all future changes to Classmark fields are liaised to GERAN for their expert review.

CN1 has discussed the CR at length at both the 'R99 and older' ad hoc meeting (8th – 9th May 2001) and at this meeting.

CN1 would like to inform GERAN that following these discussions CN1 has been able to agree a revised version of the Release 99 CR, as well as a Rel-4 version. Both of these CRs are attached, and below is a brief summary of the revisions made to the R99 CR and the differences between this and the Rel-4 CR.

CN1 proposes that these two CRs be forwarded in the normal way to the TSG CN plenary in June, for approval, and asks GERAN to review the revised R99 CR and the Rel-4 CR. Should GERAN find any errors in the CRs we ask these to be made known to the TSG CN plenary meeting. Otherwise it will be assumed that the CRs will both be approved at that plenary meeting on the 13th-15th of June.

2. Summary of CN1's changes to the GERAN's R99 CR

- Correct reference version of 24.008 was used (3.7.0)
- 'MS not supporting GSM' changed to 'MS not supporting any GSM band' in several places
- 'Multibanded' changed to 'Multiband' (in order to align with the rest of the specifications).
- RF Power capability (Classmark 1, octet 2 and Classmark 2, octet 3)- '111 shall be sent' is deleted because it is irrelevant. The codepoint must be sent in order to keep the coding syntactically correct. (The text already reads '...MS shall code the field as follows:')
- The description of the CM3 bit in MS Classmark 2 is reverted back to it's original form, as GERAN's proposal made it unclear whether the absence of the classmark 3 IE indicated that all the features are supported or none of the features are supported.

3. Summary of CN1's Rel-4 equivalent CR

- Same changes as above; and
- The addition of GSM 700 band to various sections where GSM 400, 480, 850 and 900 bands are mentioned, and to the 'Single Band Support' field.

4. Date of Next CN1 Meetings:

CN1_19	27th – 31st August
CN1_20	15th – 19th October 2001 U.K.

5. Attachments:

N1-010837 [R99 CR against 24.008 (v.3.7.0)] and N1-010838 [Rel-4 CR against 24.008 (v.4.2.0)].

CR-Form-v3

CHANGE REQUEST

⌘	**23.009 CR**	034	⌘ rev	3	⌘	Current version:	3.6.0 ⌘

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

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

Title: ⌘	Indication of Intra MSC handover from 3G_MSC-B to MSC-A/3G_MSC-A
Source: ⌘	Nokia
Work item code: ⌘	Handover
Date: ⌘	14.05.~~.~~2001
Category: ⌘	**F**
Release: ⌘	R99

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

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

Reason for change: ⌘ The GSM channel mode configuration may change during intra GSM handovers. The chosen channel needs to be known by the interworking function (IWF) located in the anchor MSC (MSC-A or 3G-MSC-A).

Also the selected encryption algorithm may change during intra GSM handovers. This information needs also to be known by the anchor MSC (MSC-A or 3G-MSC-A).

~~For 3G-MSC-B to inform MSC-A or 3G-MSC-A of chosen channel and selected encryption algorithm shall be sent to MSC-A or 3G-MSC-A in A_Handover_Performed and MAP_PROCESS_ACCESS_SIGNALLING_REQUEST By the 3G-MSC-B.~~

Summary of change: ⌘ In section 4.4.1 "Role of 3G_MSC-B" the text which is underlined below has been added:

3G_MSC-B notifies MSC-A or 3G_MSC-A of intra-3G_MSC-B InterSystem handover and intra GSM handovers, by using the A_HANDOVER_PERFORMED message.

~~If BSSMAP is used on the E-interface, 3G_MSC-B notifies MSC-A or 3G_MSC-A of intra-3G_MSC-B intra UMTS relocations (if security algorithms have been changed), by using the A_HANDOVER_PERFORMED message and indicating the selected UMTS algorithm(s) in MAP_PROCESS_ACCESS_SIGNALLING_REQUEST.~~
~~If RANAP is used on the E-interface, 3G_MSC-B notifies 3G_MSC-A of intra-3G_MSC-B intra UMTS relocations (if security algorithms have been changed), by using the LOCATION REPORT message and indicating the selected UMTS algorithm(s) in MAP_PROCESS_ACCESS_SIGNALLING_REQUEST.~~

In case of intra-3G_MSC-B ~~intra UMTS~~SRNS relocation, if security algorithms have been changed then:
a) When BSSMAP is used on the E interface, the A_HANDOVER_PERFORMED

		message shall be sent. b) When RANAP is used on the E interface, the LOCATION REPORT message shall be sent. In both cases, the selected UMTS algorithm(s) shall be indicated in the MAP_PROCESS_ACCESS_SIGNALLING_REQUEST.
Consequences if not approved:	⌘	IWF located in anchor MSC can not adapt to the changes of channel configuration potentially taking place during intra 3G-MSC intra GSM handovers. Anchor MSC is not aware of currently used encryption algorithm.

Clauses affected:	⌘	4.4.1			
Other specs affected:	⌘	X	Other core specifications	⌘	29.002, 29.010
			Test specifications		
			O&M Specifications		
Other comments:	⌘	There are related CRs to TS 29.002 and TS 29.010 which will be handled in the CN4 # 8. This change request is also related to the discussion paper presented in this meeting (TSG CN1 R99 and older Ad-hoc), N1-010597. ~~Additionally, in the situation where an inter MSC SRNC relocation has been performed and a subsequent handover to GSM is performed, the location of MS may not be known by the anchor MSC since the location reporting does not support cell based location reporting (only based on SAI, which does not exist in GSM).~~			

************************ First Modified Sections ********************************

4.4 3G_MSC-B

For roles and functional composition of the 3G_MSC-B working as pure GSM MSC, please see previous clause ("MSC-B").

4.4.1 Role of 3G_MSC-B

In the Intra-3G_MSC handover/relocation case, the 3G_MSC-B keeps the control of the whole Intra-3G_MSC handover/relocation procedure. 3G_MSC-B notifies MSC-A or 3G_MSC-A of intra-3G_MSC-B InterSystem handover and intra GSM handovers, by using the A_HANDOVER_PERFORMED message.

~~If BSSMAP is used on the E-interface, 3G_MSC-B notifies MSC-A or 3G_MSC-A of intra-3G_MSC-B intra UMTS relocations (if security algorithms have been changed), by using the A_HANDOVER_PERFORMED message and indicating the selected UMTS algorithm(s) in MAP_PROCESS_ACCESS_SIGNALLING_REQUEST.~~
~~If RANAP is used on the E-interface, 3G_MSC-B notifies 3G_MSC-A of intra-3G_MSC-B intra UMTS relocations (if security algorithms have been changed), by using the LOCATION REPORT message and indicating the selected UMTS algorithm(s) in MAP_PROCESS_ACCESS_SIGNALLING_REQUEST.~~

In case of intra-3G_MSC-B ~~intra UMTS~~SRNS relocation, if security algorithms have been changed then:
a) When BSSMAP is used on the E interface, the A_HANDOVER_PERFORMED message shall be sent.
b) When RANAP is used on the E interface, the LOCATION REPORT message shall be sent.
In both cases, ~~if security algorithms have been changed~~, the selected UMTS algorithm(s) shall be indicated in the MAP_PROCESS_ACCESS_SIGNALLING_REQUEST.

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

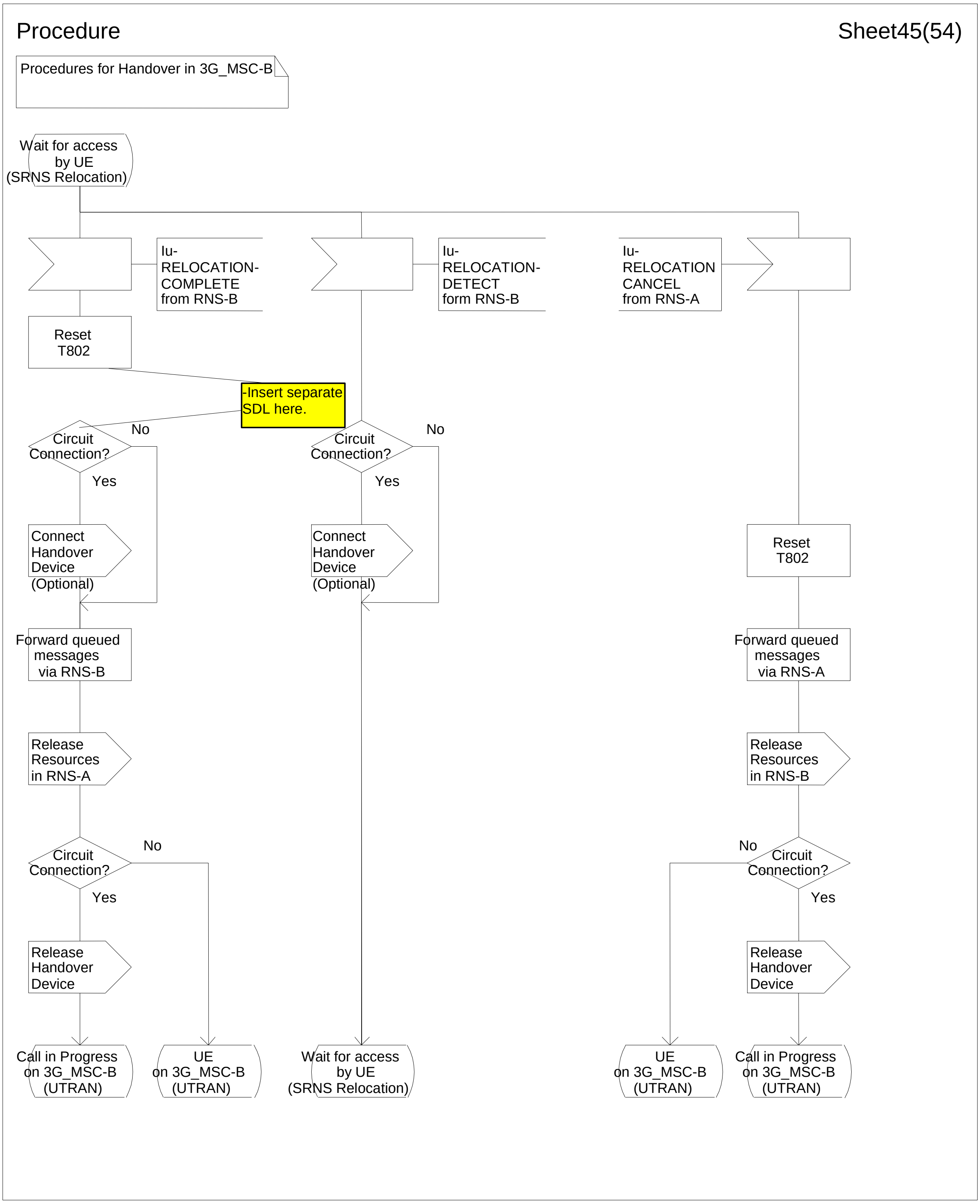
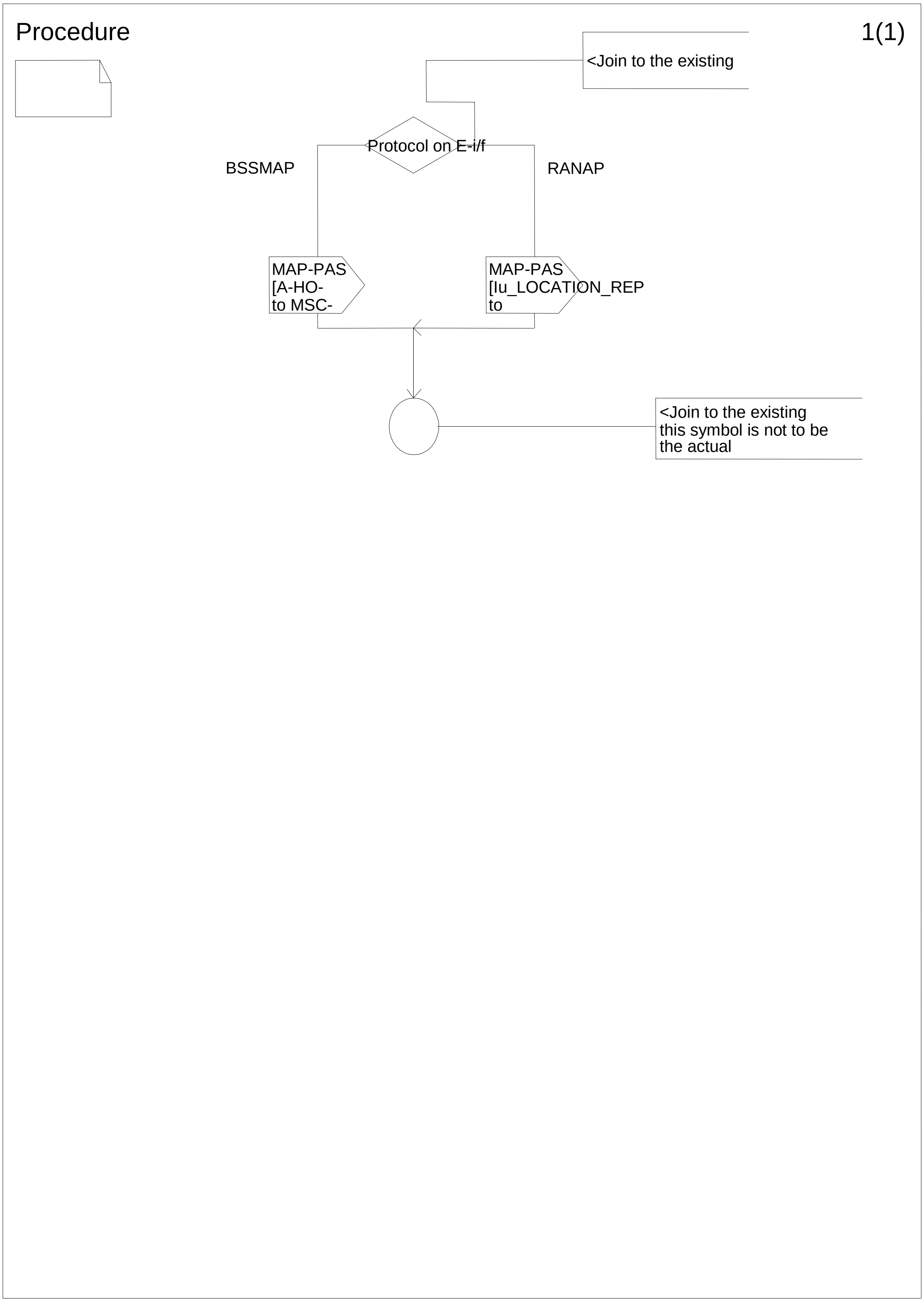


Figure 44 (sheet 45 of 54): Handover control procedure in 3G_MSC-B



<New SDL to be inserted between task box and decision diamond>

3GPP TSG CN WG1 Meeting #17 **Tdoc N1-010915**
Puerto Rico, 14th - 18th May 2001

Title:	**Liaison Statement on Indication of Intra MSC handover from 3G_MSC-B to MSC-A/3G_MSC-A**
Source:	**TSG_CN WG1**
To:	**TSG_RAN WG3**
cc:	

Contact Person:

Name:	**Robert Zaus**
E-mail Address:	**robert.zaus@icn.siemens.de**
Tel. Number:	**+49 89 722 26899**

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1. Overall Description:

CN1 would like to inform RAN3 that they have agreed the attached CR to TS 23.009, v 3.6.0, and a corresponding mirror CR to Rel-4.

According to this CR, 3G_MSC-B shall inform the anchor MSC in case of a change of UMTS security algorithms during a subsequent intra-3G_MSC-B SRNS relocation by sending a Location Report message, if RANAP is used at the E-interface. Please note that in this case it may happen that the Location Report message contains neither an *Area Identity* IE nor a *Cause* IE.

CN1 kindly asks RAN3 to note the CR and to adapt the specifications under responsibility of RAN3, if any changes are necessary.

2. Actions:

To RAN3

ACTION: CN1 asks RAN3 to adapt the specifications under their responsibility, if necessary.

3. Date of Next CNx Meetings:

CN1_18	10th – 12th July 2001	Dresden, Germany.
CN1_19	27th – 31th August 2001	

4. Attachment:

N1-010913 [CR 34r3 on TS 23.009, v3.6.0].