Technical Specification Group Services and System Aspects **TSGS#24(04)0336**

Meeting #24, Seoul, Korea 7-10 June 2004

Presentation of Specification to TSG SA

Presentation to:	**TSG SA Meeting #24**
Document for presentation:	**TR 23.801 Version 1.0.0**
Presented for:	**Information**

Abstract of document:

Scope of TR 23.801

This TR studies different mechanisms which might permit 3GPP based systems to rapidly enhance their customers' Circuit Switched Videotelephony experience.

Many operators regard circuit video services as a key part of UMTS. However there is a strong desire to have an effective and user friendly method of switching between voice and video services when the user desires and/or when radio conditions change and video mode is no longer available. There are several situations where swapping between video and voice calls is needed.

Within the TR, different mechanisms are described along with their characteristics.

Current conclusion of TR 23.801

None of the mechanisms fully meets both, all the service requirements, and the requirements of operators for fast deployment.

The only mechanism that offers scope for deployment of a system complying to the anticipated R'6 specification, and, aligns with existing inter-operator commercial arrangements is "re-dial with release of the radio connection".

However "re-dial with release of the radio connection" does not totally fulfil all the service requirements. Hence work on enhancing/completing SCUDIF will also continue.

Changes since last presentation to TSG SA:

This is the first presentation of TR 23.801 to TSG SA.

Outstanding Issues:

Evaluation of ISUP-based SCUDIF enhancements;

The impact on RAN of voice-video switching (eg Iu interface enhancements);

Section 4 "Architectural Requirements and Evaluation Criteria" needs to be completed;

In section 6.1, a more complete comparison of the pros and cons of the different mechanisms may be included.

A new section on issues relating to inter-operator roaming/interconnect accounting might need to be added.

Contentious Issues:

None.

3GPP TR 23.801 V1.0.0 (2004-06)

Technical Report

3rd Generation Partnership Project;
Technical Specification Group Services and System Aspects;
Potential Mechanisms for CS Domain Video and Voice Service Improvements
(Release 6)

GLOBAL SYSTEM FOR MOBILE COMMUNICATIONS

Keywords

UMTS, terminal, testing

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Contents

Foreword

This Technical Specification has been produced by the 3rd Generation Partnership Project (3GPP).

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

Version x.y.z

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x the first digit:

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z the third digit is incremented when editorial only changes have been incorporated in the document.

1 Scope

This TR studies different mechanisms which might permit 3GPP based systems to rapidly enhance their customers Circuit Switched Videotelephony experience.

Many operators regard circuit video services as a key part of UMTS. However there is a strong desire to have an effective and user friendly method of switching between voice and video services when the user desires and/or when radio conditions change and video mode is no longer available. There are several situations where swapping between video and voice calls is needed.

Within the TR, different mechanisms are described along with their characteristics. A comparison of the pros and cons of the different mechanisms is included.

2 References

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

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies. In the case of a reference to a 3GPP document (including a GSM document), a non-specific reference implicitly refers to the latest version of that document *in the same Release as the present document*.

[1] 3GPP TS 23.172 v6.0.0: “Technical Realisation of the Circuit Switched (CS) multimedia service: UDI/RDI Fallback & Service Modification; Stage 2

[2] ITU-T Q.765.5 Signalling system No. 7 – Application transport mechanism: Bearer Independent Call Control (BICC)

3 Definitions, symbols and abbreviations

3.1 Definitions

For the purposes of the present document, the following terms and definitions given in TS 21.905 and the following apply.

<keyword> <Definition>

3.2 Symbols

For the purposes of the present document, the following symbols apply:

3.3 Abbreviations

For the purposes of the present document, the following abbreviations apply:

SCUDIF	**S**ervice **C**hange and **UDI F**allback
APM	Access Transport Mechanism

APP	Access Transport Parameter
USI	User Service Information
TMR	Transmission Medium Requirement

4 Architectural Requirements and Evaluation Criteria

Editor's note: this is a kind of requirements section: eg it should indicate the need to handle loss of UMTS 64 kbit/s bearer; the need for roaming and inter-operator accounting; etc.

4.1 Background

4.2 Criteria

5 Mechanisms

5.1 Mechanism 1: enhanced SCUDIF

5.1.1 Description

SCUDIF signals at multimedia call setup information elements for speech and multimedia. The user may decide which mode (speech or multimedia) is used initially and may also modify the call later. Also the originating and terminating networks may change the service to speech when multimedia is not supported as a service or because of lack of resources. Inter MSC and therefore also inter PLMN signalling bases on the BICC codec negotiation.

SCUDIF consists of the following functional components:

- The existing fallback to speech application within SCUDIF, as defined in 23.172 from rel5 onwards.
- The existing user-initiated service change application within SCUDIF, as defined in 23.172 from rel5 onwards.
- A network-initiated service change procedure, as defined in 23.172 from rel6 onwards.

Issues	**SCUDIF Solution**		
	Fallback to speech (rel5)	**User initiated service change (rel5)**	**Network initiated service change (rel6)**
Issue 1: 3G VT user to 3G non-VT user	Yes	-	-
Issue 2: 3G VT user to user on non-VT network	Yes	-	-
Issue 3: 3G VT user invoked call change from VT to speech (and vice versa)	-	Yes	-
Issue 4: VT call failure during 3G-2G handover	-	-	Yes
Issue 5: VT degradation in 3G cell	-	Yes	Yes

Table 1: Solution to Issue mapping

The basic premise of Network Initiated service change is that under certain conditions the radio access network makes the determination that a service change to speech is required, and invokes the signalling required to change the call type from 64K UDI to speech. There is no manual interaction required on the part of the user. There are several manners in which this could occur:

- In the case of 3G cell performance degradation, potentially after unsuccessful handover/relocation trials, the radio access network could trigger the service change (assuming the user has not already done so via user initiated service change).
- In the case of a 3G-2G handover, the radio access ensures that the service change to speech occurs prior to processing the relocation required.

The Iu RANAP trigger to indicate the video call cannot be maintained even on neighbour cells needs to be clarified.

In more details, the SCUDIF feature provides the following:

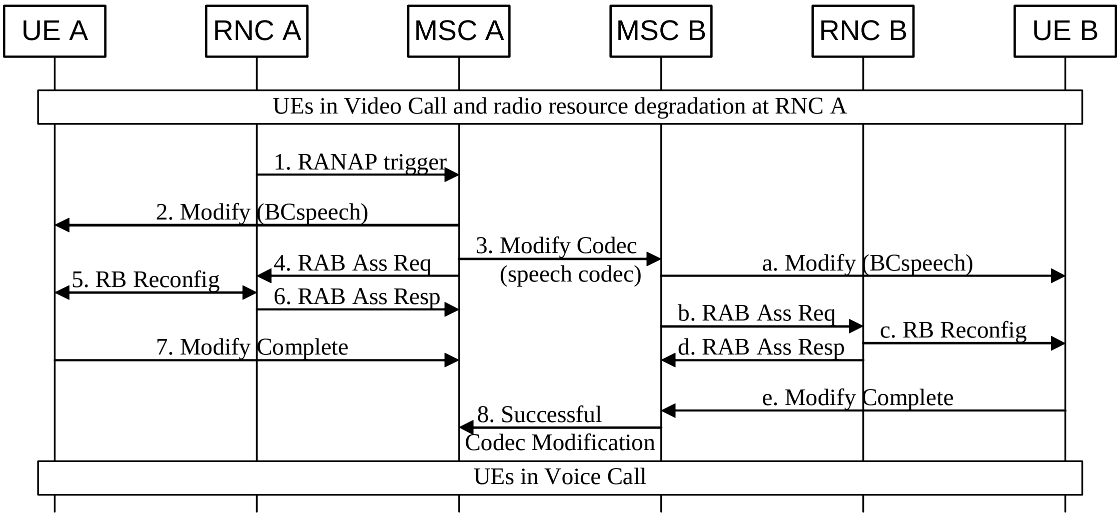
- At call set-up the user equipment identifies itself to the network as requesting a SCUDIF call, i.e. a call with the possibility to modify the call at any time between a "speech" call and a "multimedia" call.
- The originating user may request "speech" first or "multimedia" first, i.e. the user defines its preferred service.
- If any part of the network cannot support the multimedia service it falls automatically back to "speech".
- If any part of the network cannot support SCUDIF it falls back to the preferred service (i.e. could become "multimedia-only" only or "speech-only").
- Call setup is in all cases comparably quick, because the networks and terminals and users can select and decide in one single attempt.
- During a successful SCUDIF call both users can switch between "speech" and "multimedia" (via network signalling – BICC Codec Negotiation/Modification). In any case the originating user (the one who started the SCUDIF call) is charged for all services, even if the terminating user switches to "multimedia".
- Due to the already negotiated capabilities both terminals and the network know, whether or not a "multimedia" call is in principle possible.
  It is, however, not in all cases guaranteed that the "swap" from "speech" to "multimedia" will be successful, when "BICC Codec Modification" is applied. Changed radio or other network resource conditions may prohibit that the modification is successful from end-to-end.
  If the modification is successful, then it is done rather quickly and the call interruption time is short.
  If the modification is not successful, then an unfavourable speech call interruption occurs, but it is not very long. But of course the users know that they are trying to switch to video and will expect the break in speech.
- A user can deny a request for the initial set-up or can deny to "swap" to "multimedia" via MMI and force "speech" to be established/retained. This is a primary requirement for privacy reasons. The fallback to "speech" is performed, if one or both users wish to do so. It is anticipated that in most cases both users will agree beforehand whether they attempt or not to set up the multimedia connection. Unnecessary speech call interruptions due to a user not accepting the multimedia should therefore be minimal.
- All supplementary services as known for "speech" are supported, but for some services "multimedia" will then be impossible, because only a few supplementary services for "multimedia" are defined.

To deploy and run SCUDIF several prerequisites must be in place:

- Both terminals must support SCUDIF
- Both RANs must support RAB modification & UDI bearer
- The Core Network must support SCUDIF
  It is not really necessary to run a layered network architecture, but it is highly recommendable.
- In case of inter-operator calls potential Transit Networks must support SCUDIF

If one of these prerequisites is missing the call will be either in "speech-only" (most likely) or in "Multimedia-only", i.e. without the possibility to swap to "speech".

5.1.2 Fallback from video to voice



1. The RNC detects that for UE A the transmission quality has moved below the quality threshold set for the 64k VT bearer. After potential handover/relocation trials, the RNC A then sends a RANAP message to MSC A. MSC A understands the reception of the message to mean that the video call cannot be maintained and should be switched to speech.

2. The MSC A sends the MODIFY message to the UE A indicating the change to bearer capability speech.

3. The MSC A sends a BICC message MODIFY CODEC to the MSC B indicating that the bearer should be modified to support one of the previously negotiated speech codecs.

4. The RAB ASSIGNMENT REQUEST message is sent from the MSC A to the RNC A, requesting the modification of the RAB for VT to a RAB for a Voice call.

5. The radio bearer is modified between RNC A and UE A.

6. RNC A responds to MSC A with an RAB ASSIGNMENT RESPONSE message indicating that the radio bearer was modified.

7. The MODIFY COMPLETE message is sent by UE A to acknowledge the change of the bearer capability. Note: this response can happen before 4.

a. The MSC B sends the MODIFY message to UE B indicating the change to bearer capability speech.

b. The RAB ASSIGNMENT REQUEST message is sent from MSC B to the RNC B, requesting the modification of the RAB for VT to a RAB for a Voice call.

c. The radio bearer is modified between the RNC B and UE B.

d. RNC B responds to MSC B with a RAB ASSIGNMENT RESPONSE message indicating that the radio bearer was modified. Note: this response can happen before b.

e. The MODIFY COMPLETE message is sent by UE B to acknowledge the change of the bearer capability.

8. MSC B sends a BICC message SUCCESSFUL CODEC MODIFICATION to MSC A indicating that the bearer has been successfully modified.

5.1.3 Advantages of the mechanism

SCUDIF is optimised towards the videotelephony successful setup-path and for fast switching between voice and video, thereby increasing the chances of using the service.

Any time the users may change between speech and multimedia. The user at the terminating end may refuse a service change without terminating the call.

At call setup and during the call the network may fallback to speech due to lack of resources.

The general benefits of the SCUDIF feature is that it allows users to attempt to initiate combined “Speech” and “Multi-media” calls and if the network or called subscriber can not or prefers not to support the multimedia service the user still gets through-connected with “speech” quickly, thus retaining some quality of service and customer satisfaction. In addition it allows two parties conversing via normal “speech” to transmit UDI payload (video, still pictures, other data) and return to normal “speech”, thus only using and paying for the higher rate services, when needed.

Solutions for prepaid and post-paid charging are standardized. The network correlates speech and multimedia, allowing flexible charging solutions.

The users at the originating and terminating side do not experience additional delay compared to an ordinary call setup attempt. Users with SCUDIF capable terminals are therefore more likely to configure their terminals to attempt a video call with fallback by default.

At call setup, the called party is informed about the choice between speech and multimedia.

SCUDIF is one call or service of two modes, which minimises impacts by other services during mode change compared to other mechanisms.

Rules for the interacting of speech and multimedia supplementary services are standardized, which simplify the configuration. To a certain extent supplementary services like call forwarding, barring, roaming restriction have to be configured for TS11 and BS30 with the same settings. Otherwise, a call with the preferred service but no fallback or service change capability is established.

5.1.4 Issues that need to be resolved

The ability to traverse transit networks is limited by the requirement of BICC. Usage with an ISUP transit network needs to be studied.

5.2 Mechanism 2: Dual Call

5.2.1 Description

The caller establishes a speech call between both UEs first. When one of the users wants to initiate a multimedia call, that users’s UE then puts the speech call on hold and sets up a multimedia call. The receiving UE gets a call wait indication. The receiving UE derives from Caller Line Identification that the multimedia call is associated with the established voice call and asks the user whether the change to multimedia shall be accepted. When the user accepts the multimedia call, the UE puts the voice call on hold and accepts the multimedia call.

The dual call is assumed to be an extension of the UE and network software that uses only R99 functionality: TS11, BS30/MM, call hold, call wait and caller line identification. When the network resources can no longer support the video call, e.g. due to load increase in UTRAN or handover to GERAN, the multimedia call is dropped and the UEs retrieve the speech call automatically.

Also at any time during the multimedia call, either party can request to switch the multimedia call to a speech call and then the (held) voice call is retrieved automatically.

One message flow is shown below, but a more complete set of message flows is shown in Annex A.

5.2.2 Fallback from Video to Voice

Figure X below shows the case of fallback from Multimedia to voice due to degrading UTRAN coverage.

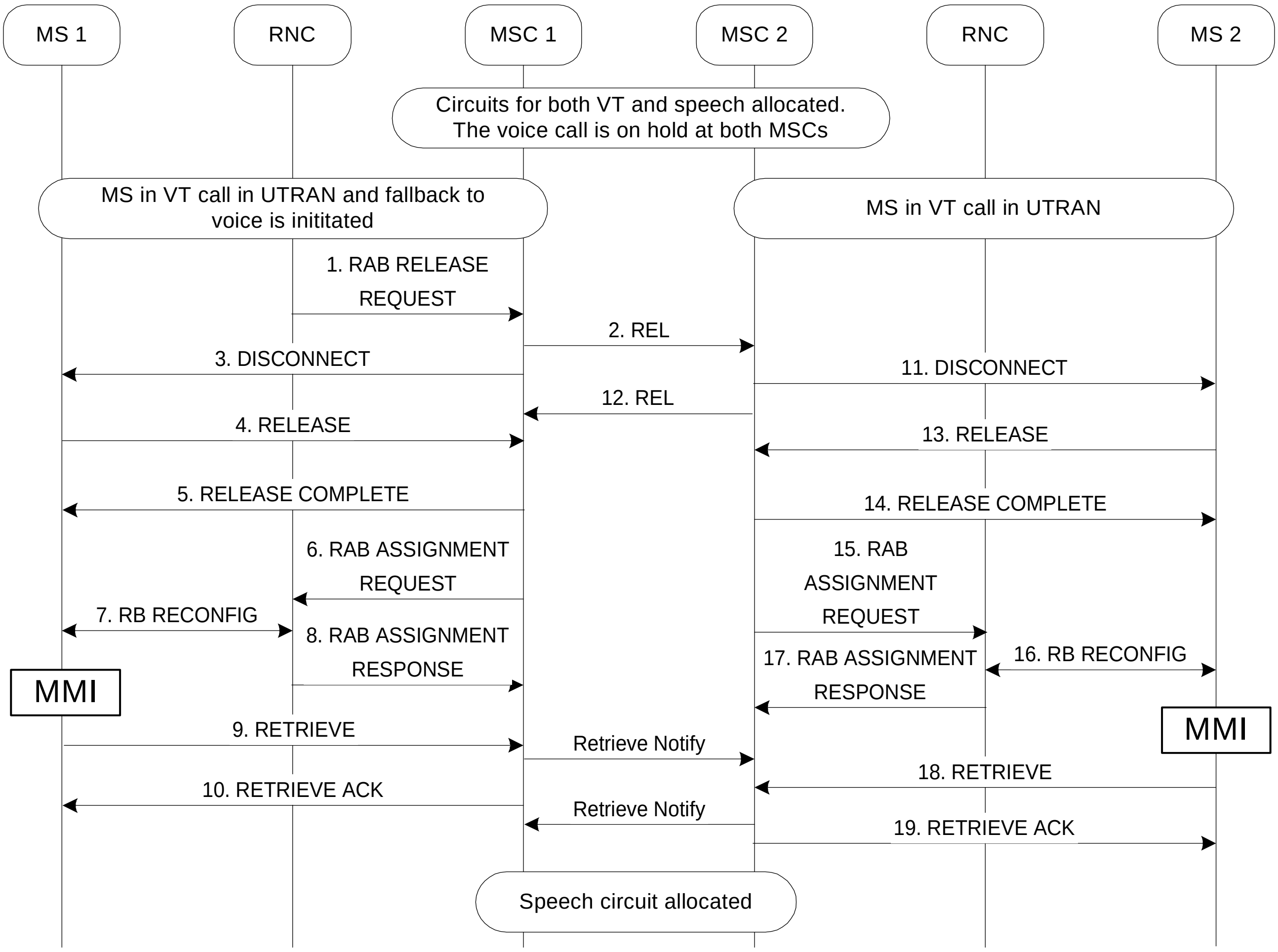


Figure X: fallback from Multimedia to voice due to degrading UTRAN coverage

1 The RNC detects that MS 1 has moved below the quality threshold set for the 64k VT bearer. After having tried unsuccessfully handovers/relocations, the RNC then sends the RAB RELEASE REQUEST message to MSC 1 indicating that the bearer should be released. MSC 1 understands the reception of the RAB RELEASE REQUEST message to mean that MS 1 should be switched to speech because e.g. it is moving out of coverage for the active service.

2 The MSC 1 sends the REL message to MSC 2 controlling the MS 2, indicating that the video call should be released.

3 A DISCONNECT message is sent from the MSC 1 to MS 1, instructing MS 1 that the video call should be released.

4 MS 1 responds with a RELEASE message for the video call.

5 MSC 1 sends the RELEASE COMPLETE message to MS 1 for the video call.

6 The RAB ASSIGNMENT REQUEST message is sent from the MSC 1 to the RNC, requesting the modification of the VT bearer to a bearer for a speech call.

7 The radio access bearer is modified between the RNC and MS.

8 The RNC responds to the MSC 1 with a RAB ASSIGNMENT RESPONSE message indicating that the radio bearer is being modified.

It is possible that the handsets could be automated to retrieve a speech call in this situation, i.e. when both calls are from the same CLI and the VT call is terminated.

9 The RETRIEVE message is sent by MS 1 to MSC 1 to reconnect the voice call to MS 1, which had been on hold at MSC 1. A Retrieve Notify message is sent from MSC 1 to MSC 2 indicating that the MS 1 has taken the voice call off hold.

10 MSC 1 responds to MS 1 with a RETRIEVE ACK message once the call has been connected.

11 A DISCONNECT message is sent from MSC 2 to the MS 2, instructing MS 2 that the video call should be released.

12 MSC 2 sends the REL message to MSC 1, acknowledging that the video call is being released.

13 MS 2 responds with a RELEASE message for the video call

14 MSC 2 sends the RELEASE COMPLETE message to MS 2 for the video call.

15 The RAB ASSIGNMENT REQUEST message is sent from MSC 2 to the RNC, requesting the modification of the VT bearer to a bearer for a speech call.

16 The radio bearer is modified between the RNC and MS 2.

17 The RNC responds to MSC 2 with a RAB ASSIGNMENT RESPONSE message indicating that the radio bearer is being modified.

It is possible that the handsets could be automated to retrieve a speech call in this situation, i.e. when both calls are from the same CLI and the VT call is terminated.

18 The RETRIEVE message is sent by MS 2 to MSC 2 to reconnect the voice call to the MS, which had been on hold at the MSC. A Retrieve Notify message is sent from MSC 2 to MSC 1 indicating that the MS 2 has taken the voice call off hold.

19 MSC 2 responds with a RETRIEVE ACK message to MS 2 once the call has been connected.

5.2.3 Advantages of the mechanism

Allows for fallback and service change without new network functionality. Fallback and service change are not delayed by cell selection or mobility management procedures as the UE does not enter idle mode.

5.2.4 Issues that need to be resolved

5.2.4.1 General

Billing issues

1- When the call corresponds to a path that crosses a fixed network (e.g. France Telecom), two 64 kbit/s circuits are established. The mobile operators will have to adjust their charging policies to give the user the intended billing.
2- Moreover, ideally, a correlation has to be performed between the speech call and the video call in the mobile network.
3- Dual Call assumes that the person who initiates the video session pays for it. If, instead, the video session has to be paid for by the person who initiated the speech call then, extra functionality is required.

Supplementary services

1- Even if it is not an important constraint, call hold (for the calling party), and call wait (for the called party) must be activated for speech and video.
2- Voice and BS30 services must be forwarded to the same entity, except for answering machines.
3- When the caller does not signal its identity, the recipient has to treat the incoming call as any other unidentified waiting data call.

General

1- Other transactions may delay or complicate dual call, e.g. an ongoing call and the voice call of the dual call have to be put on hold just that the user has a short communication before retrieving the interrupted call.

2- The RAB modifications that need to be supported together with handling of Call Hold/Wait/Retrieve have to be investigated.

3- The use of service based handover between GERAN and UTRAN may cause additional implications.

Calls to/from IP users

1- Calls to fixed network videophones or non-dual call UEs are questionable as it puts the burden on the user to emulate dual call behavior. The B-party user may be confused by hold announcements as the speech call goes on hold when the user answers.

2- To establish the video call while engaged in the speech call, when the calling party is an IP terminal, it must be able to support H.450 HOLD supplementary service; and when the called party is an IP terminal, it must be able to support the H.450 WAIT supplementary service. The gatekeepers must support both H.450 HOLD and WAIT services.

3- At fallback from video to speech for e.g. radio conditions, the message DISCONNECT [video] is sent to the IP terminal. This requires the implementation of a specific "automatic" RETRIEVE [speech call] in the IP terminal, because the user is not aware of the event at mobile side.

Calls to/from H.324 fixed users

Same kind of issues as for IP users.

5.2.4.2 Video call established automatically once the speech call has been setup

1- Call setup delay is increased by several seconds, corresponding to the HOLD procedure, followed by a modification of the RAB ordered to the UTRAN. However, this additional delay is not equivalent to a full call setup since authentication and paging procedures are not required in a Dual Call.

2- A called user with Dual Call mobile is rang when the speech call arrives. At hook-off, that speech call is put on hold for several seconds before being switched to video call. The called user hears nothing in most of the cases, and would probably hook-on before the video call is established.

3- Moreover, when the called user has a R99 mobile, it has to accept the video call manually. But this would be strange to the user to receive a speech call immediately put on hold for a while with no specific indication, and would be rung a second time for a video call he has to answer to.

4- When the called user has a non-capable video mobile, it would be put on hold for several seconds before being sent back to speech call. He would probably hook-on as well.

5- If there is a lack of video coverage at call setup, fallback to voice takes time as typically two RAB modifications are performed.

6- Automatic dual call does not work properly if CLIR is activated because correlation between the established speech call and the incoming video call is needed in the terminal, and CLIR is not service dependent. This means CLIR should be deactivated for all the services.

7- Since the speech call is established first, the speech answering machine will answer first. The caller may miss the announcements while the A party’s UE attempts to establish the video call. How does the UE react when the video call is not answered? Only user or timer controlled release of the video call and retrieve of the voice call seems possible.

8- With different answering machines for speech and video: Even if the speech answering machine is put on hold immediately, there is a problem if the video call cannot be setup or is released due to radio reasons. Indeed, the call goes back to speech answering machine, but part of announcement has been lost. The user would not understand.

5.3 Mechanism 3: Re-dial with release of the radio connection

5.3.1 Description

This simple "Idle-Mode Redial" mechanism is a combination of existing standards and services, Voice call and Video call, in a "terminal-centric" way, i.e. without any or with only minimal additional support by the RAN or CN. The MMI on the UE initiates automatically or on user request the setup of a Voice call, when the Video call fails. The MMI on the UE releases the active service and initiates the setup of the alternate service, when a user requests a service change. It is subject to the various terminal implementations how smart and efficient the service setup and the service changes appear to the user. The user's MMI may provide different terminal options and preferences, which control the automation of Redial and call acceptance in case of user-initiated service change and network-caused Video call release.

The operator may optimise the behaviour by keeping dual-mode terminals as much as possible in UTRAN.

Definition: In the following the calling user and his terminal are named "user A" and "UE A"; the called user and his terminal are named "user B" and "UE B".

Assumptions:
At call setup the MMI offers (somehow): "Setup Voice" and "Setup Video".
During a Voice call the MMI offers: "Terminate Call" and "Change to Video".
During a Video call the MMI offers: "Terminate Call" and "Change to Voice".

If a call is terminated by user B, then UE A will get the release cause "normal termination".
The same release cause will be received by UE A in most networks, when the Video call cannot be maintained any longer. Only in networks with enhanced MSC-MSC and MSC-UE signalling the cause code "service not longer available" may be received by UE A, when the network releases the Video call.

Detailed description:

UE B shall never initiate an automated Redial, when a call is released.
UE B shall perform a Redial only on explicit request by user B.

If a Voice call setup fails, then no automated Redial shall be initiated.

If a Voice call is terminated by the users or released by the network, then no automated Redial shall be initiated.

If a Video call is terminated by either user, then no Redial should initiated.

UE A may initiate an automated Redial in one of the following situations:

a) Automatic fallback to Voice when the setup of a Video call is not possible,

b) Automatic fallback to Voice when the Video call is released by the network,
and the release cause "service not longer available" is received by UE A,

c) User A initiated a change from Video to Voice,

d) User A initiated a change from Voice to Video.

UE B may initiate an automated Redial in one of the following situations:

e) User B initiated a change from Video to Voice,

f) User B initiated a change from Voice to Video.

In case a) UE A initiates automatically the setup of a Voice call with UE B, when the Video call attempt fails. The reasons may be that UE A has no Video coverage, or UE B does not want to accept the Video call, or UE B might be not capable of receiving a Video call, or UE B might be unreachable for a Video call or it might be not reachable at all. If the Redial attempt of a Voice call fails, then no further attempt is performed.

In case b), when the ongoing Video call is released with "service not longer available",
then UE A automatically initiates a Redial of a Voice call to the B-number. If this Redial attempt fails, then no further attempt is performed. If this release cause is not supported by the network or UE, then an abnormal call release by the network cannot be distinguished from an intentional call termination by User B. In that case UE A may prepare and offer to Redial a Voice call, but should not automatically initiate the Redial, unless user A confirms it.

In cases c), d), e) and f) either user A or user B may initiate the change in either direction.
On user A request ("change") UE A initiates a termination of the ongoing call and then initiates a new call of the alternate service to the same B-number. UE B performs normal call clearing and call accept procedures.
User B may only initiate the change, when the caller line identification has indicated the A-number.
On user B request ("change") UE B initiates a termination of the ongoing call and then initiates a new call to the A-number. In that case UE A receives the cause code "normal termination" and performs normal call clearing and call accept procedures.

Note: when the change is successfully initiated by user B, then this user becomes automatically the calling user for the new call. This has implications on the charging and the behavior of the UEs.

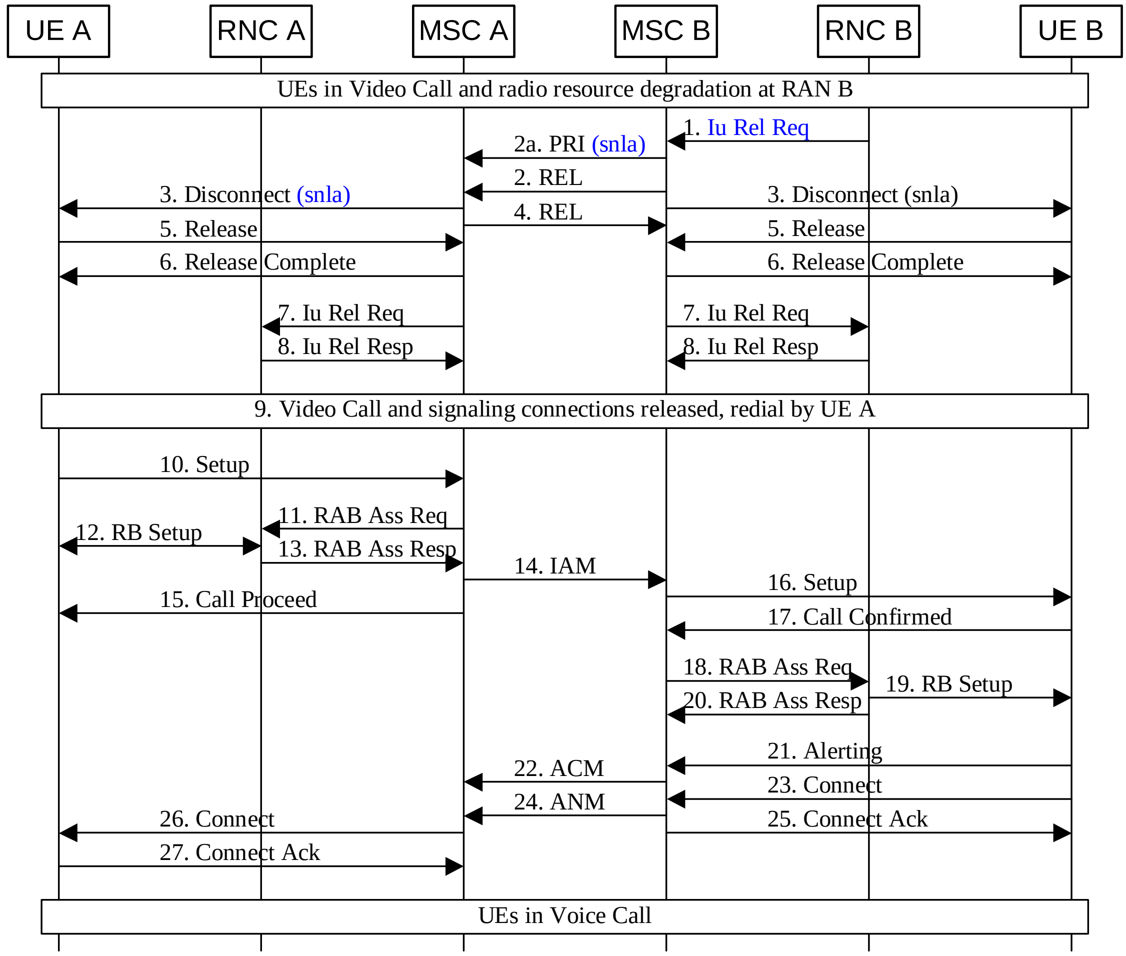
5.3.2 Automated Fallback from Video to Voice

When UE A in an ongoing Video call looses radio connection or the call is released with cause "service not longer available", then UE A may display '*Video lost, reestablishing Voice!*' for the attention of user A. Then UE A reuses the B- number, as of the previous Video call, to Redial a Voice call to the same previous UE B. If user A presses '*No*' in that phase, then the terminal remains/goes idle and no fallback to Voice will occur.

The terminating network and the terminating UE B apply normal (Video) call termination and (Voice) call establishment procedures, e.g. paging, alerting and answer. UE B may display *"Video lost, accept Voice?"*, when it receives the incoming Voice call from the previous UE A. Alternatively it may accept the incoming Voice call from UE A automatically, as long as the Redial comes within a time-out window. If user B presses '*No*' in that phase, then the terminal remains/goes idle and no fallback to Voice will occur.

When UE B looses radio contact or understands that the Video call is dropped for any other reason, then it only applies normal call clearing procedures and waits for a potential Redial attempt by UE A. For a short while UE B may display *"Video lost, waiting for Voice?"* and user B may already now decide to terminate the session.

The signaling flow below shows this automated Redial in case of radio resource degradation at RAN B. It assumes that an optional ISUP "PRI" message is sent between the MSCs with the cause code "service not longer available". Without that PRI message the REL message should indicate a proper release cause. Otherwise MSCA can most likely not distinguish between normal and abnormal call termination by the B-side. Both, the transfer of PRI messages and the modification of release causes are subject to operator policies. Their usage is therefore for further study. The signaling flow also assumes that MSC A informs UE A about normal or abnormal call termination ("service not longer available"). Without that differentiation between normal and abnormal video call termination UE A should not automatically perform a voice call redial, but wait for confirmation by user A.



1. After potential handover/relocation trials RNC B finally detects that for UE B the 64k bearer cannot be maintained any longer. RNC B then sends the Iu RELEASE REQUEST message to MSC B, indicating that the Iu should be released.
2. MSC B sends an optional PRI message to MSC A with a release cause, e.g. "service not longer available". This is followed by a REL message to the MSC B. This mandatory REL message contains a second release cause, but this might be less precise than the other one.
3. The MSCs send Disconnect messages to the UEs with an error cause indicating "service not longer available". This error cause is of no relevance for UE B, because it anyway shall not Redial. If this cause code is not send or understood by UE A, then UE A cannot decide, whether to Redial or not. In that case user A should first confirm the Redial attempt (not shown here).
4. MSC A sends a REL message to MSC B.
5. The UEs release the Video Call.
6. The MSCs confirm the release of the Video Call.
7. The MSCs request the release of all resources.
8. The RNCs confirm the release of all resources.
9. When the signalling connections with the UEs are released it may take some time before an UE can place or receive a new call because of cell selection and registration procedures.
10. UE A, which received for the Video call the release cause "insufficient radio resources", sends a SETUP message to MSC A to setup a Voice call.
11. A RAB Assignment Request message is sent from the MSC A to the RNC A, requesting the setup of a RAB for a Voice call.
12. The radio bearer is established between RNC A and UE A.
13. RNC A responds to MSC A with an RAB Assignment Response message.

14. MSC A sends an IAM message to MSC B to establish a Voice Call with UE B.
15. MSC A sends a Call Proceed message to UE A.
16. The MSC B sends a Setup message to UE B indicating the establishment of a Voice Call.
17. UE B sends Call Confirmed to MSC B.
18. The RAB Assignment Request message is sent from MSC B to the RNC B, requesting the establishment of a RAB for a Voice Call.
19. The radio bearer is established between the RNC B and UE B.
20. RNC B responds to MSC B with a RAB Assignment Response message.
21. UE B sends Alert message to MSB B.
22. MSC B sends ACM message to MSC A.
23. User or UE accepts the Voice Call and UE B sends Connect message to MSC B.
24. MSC B sends ANM message to MSC A.
25. MSC B sends Connect Ack message to UE B.
26. MSC A sends Connect message to UE A.
27. UE A acknowledges with a Connect Ack message to MSC A and the Voice call is established.

5.3.3 Advantages of the mechanism

The strength of this simple Redial behavior is that it covers all situations, where the radio connection may be lost, e.g. due to movement from good 3G coverage into ‘fringe 3G coverage’, as well as UE movement from good 3G coverage into 2G coverage.

It is up to the users whether they first set up a Voice call and change then – after verbal agreement – to Video, or whether they directly try to establish a Video call. The direct call setup time for Voice is as fast as today. The direct call setup for Video will be as fast – if successful – otherwise the Redial will setup the Voice call some seconds delayed, but without additional effort by the user. This can be judged as a major benefit for the user. Also just the fact that they get a “connection” rather than a “failure” is much more appreciated.

The Redial uses always only one bearer for one service at a time. No special care must be taken for more than one call or bearer during any kind of handover or whatsoever. This is important for operators.

Redial after loss of radio coverage for Video results in a silence period of some seconds, before the Voice call is re-established. It can be assumed that the Video call was released in the coverage area of an UTRAN cell. With quite high likelihood this same UTRAN cell will have sufficient resources to setup the Voice call, because the Video call just released substantial radio resources. It seems to be quite unlikely that the UE can move out of the UTRAN cell during the Redial attempt. Therefore the fallback from Video to Voice will in most cases succeed. This is an important benefit for the users.

Without the mentioned cause code “service not longer available” an automatic Redial by UE A after network-initiated Video call release is possible, but then also a Video call termination by user B would result in a Redial attempt by UE A – and this is not wanted.

All existing Video interworking and conferencing equipment as well as Voice supplementary services can be used, but without change between Voice and Video.

Fallback and Voice/Video service change are accomplished with no or minimum new network functionality, optionally added to Video call functionality.

The application or MMI that combines Voice and Video call on the UE is comparably simple.

Redial interoperates also with UEs that implement Voice and Video services completely separated. And it interoperates with non-Video UEs and with fixed network Video terminals.

The most important positive effect is fastest possible deployment.

5.3.4 Issues that need to be resolved

To allow for automated Redial and to support automated call answering in case of network initiated video call release the release messages need to or should indicate proper release causes on ISUP and from MSC to UE, e.g. “service not longer available”. This might require the sending of an ISUP PRI message, as causes indicated by ISUP release messages are typically modified at network borders. Otherwise, if these proper release cause codes are not available, then the network initiated release of the Video call cannot be distinguished from the normal call termination by user B.

User-initiated change between the services will, however, work well with existing messages. And, as it can be assumed that the users will anyway initiate a fallback from Video to Voice in case of radio resource degradation before the call is released by the network, it does not seem to be a big disadvantage, if these release causes are not always available. The user changes the services most likely before the network has to release the video call.

Alternatively User-to-User signaling could be used to inform the distant UE about a user-initiated call modification or termination. With that U-to-U signaling no additional ISUP messages like PRI would be needed. But also U-to-U signaling is not always available today.

The majority of the fallbacks are quite likely already at Video call setup, so that the UE performs the Redial in the cell where the Video call setup was initiated. This early fallback without cell change is likely as the majority of the UEs are GSM or other terminals that are not Video capable. Furthermore Video capable UEs are not always in UTRAN coverage for Video and also user B may refuse the Video call. Once a Video call is established the user is probably not moving too much as being engaged in a Video call makes it difficult to walk or drive a car.

It remains the change of a Voice to a Video call, which may be delayed because of cell changes during the Voice call. The delay to setup a Video call from an ongoing Voice call may be long.

The called user can only refuse the Video call and not indicate that also the Redial of a Voice call will not be accepted. Only when the Redialled Voice call is also rejected by the B-party, then no further Redial is attempted.

Other transactions may delay or prevent Redial, e.g. waiting calls. But this will anyhow be a problem that the user has to sort out

Race conditions might occur, when both users initiate a service change in parallel. The users will have to learn to agree on changes beforehand by verbal communication. This is anyhow a matter of etiquette. Probably the MMI could give some support.

The user that initiated the service change will be charged for the service that is setup by the service change.

Supplementary services like call forwarding, barring, roaming restriction should be configured for Voice and Video with the same settings, which enables both services to reach the same endpoint in case of Redial.

The use of service based handover between GERAN and UTRAN may cause additional implications. If one or both terminals are in GERAN, then first a service-based handover to UTRAN must be performed – if possible, and potentially on both radio legs.

It might be an operator’s strategy to keep dual-mode terminals preferably in UTRAN also during voice calls. If it still ended up in GERAN, then Video is just impossible! in that moment and the change will not be attempted and the Voice call is not interrupted. The user, who tried to change to Video can be informed accordingly. But typically the users should see on their displays, whether they are in “3G coverage”or not – before they try to change to Video.

Charging data are collected only for the individual services. This requires additional effort when combination to a Voice/Video service is needed.

The duration of a fully automated service change when the Video call is lost due to degrading radio conditions can be considerable in certain situations, especially when the users do not act themselves. First, it is necessary to wait for the normal radio link failure, before automatically going to idle mode. Otherwise, a handover to another 3G cell in Video would be jeopardised: the mobile and UTRAN would have tried a handover, i.e. the received power would have been below the configured measurement threshold, the mobile would have performed measurements under cells chosen by UTRAN, and UTRAN would have decided to do nothing, because the Video call could not be handed over. This is a problem for all solutions.

Further: when a GSM or UTRAN mobile releases its last connection with the MSC, it enters a state where it waits for the MSC to release the radio connection (RR connection in GSM; RRC connection in UTRAN).

Following the release of the radio connection, the mobile has to complete various tasks before it is permitted to initiate a new call. These tasks can include receiving information broadcast by the serving cell, measuring neighbour cells, performing Location Area Updates to an MSC, performing Routeing Area Updates to an SGSN; performing inter Radio

Access Technology changes, etc., and they can take a long and variable period of time. During this time UE A and/or UE B may be unable to make or receive calls. These effects can make the Redial service unattractive to the customer.

The necessity (or otherwise) to perform a location update largely depends upon the operator's strategy. One such strategy is to use the 2G capacity for Voice calls and to keep the 3G capacity for data. In this scenario, mobiles camp on 3G cells, start their Voice calls on 3G and are then handed over to 2G. If the 2G and 3G cells are on different MSCs, then they will be in different Location Areas and Routeing Areas. In this scenario Redial can systematically lead to a time consuming sequence of 2G Location and Routeing Area updating followed by 3G Location and Routeing Area updating. Another strategy could be to keep all dual-mode terminals as long as possible in UTRAN. Then everything is substantially simpler. This might be especially interesting at the beginning of UMTS services, where radio capacity is not the primary issue, until a better Voice-Video swap is available.

The largest component of delay and delay variation is likely to be caused by overlaid, not identical 2G and 3G Location Areas. One solution to this problem is contained in the recently agreed release 6 GERAN changes to 44.018 and 45.008 (see GP-040518 and GP-040542), which permit the BSC to "release the mobile and ask the mobile to camp on 3G". By using this enhancement, a mobile that started a Voice call on 3G, was inter-MSC handed over to 2G, and then released, could immediately return to 3G and avoid the need to perform Location Area and Routeing Area Updates.

Although this is a REL-6 change, there is nothing to prevent the BSC from sending this information to R99, REL-4, REL-5 mobiles (in fact, the mobile does not indicate its support/non-support of this feature to the BSC, and so upgraded BSCs *will* send this information to older mobiles). However, strictly speaking, an R99 mobile that acted upon this information would be failing to comply with the letter of R99 05.08 (but existing test specifications should not be impacted by an UE that did implement this REL-6 feature).

For other operational scenarios (e.g. camping all mobiles on 2G, and inter-MSC hand over of Video calls to 3G), the RRC signalling appears to contain signalling, which is able to push the mobile from 3G to 2G when the UTRAN channel is released (see 25.331 section 10.3.8.15, "RPLMN information").

Note that the A and B party networks may be different and neither operator should have control over the operational strategy of the other operator.

Different time delays at the separate ends of the link will make it difficult to provide high success rates for Redialled calls. It is likely that two attempts at re-establishing a Voice call may be needed to ensure high probabilities of success. The timings of these attempts need to be evaluated.

5.4 Mechanism 4: Re-dial using the same radio connection

5.4.1 Description

This mechanism is similar to Idle Mode Redial, except that the UEs do not release the RR/RRC connection in between the "release" and "setup" messages.

Release of the RR connections is inhibited by a UE indication to the MSC or by the UEs exchanging User-User signaling and then both initiating USSD transactions toward their HLRs prior to the release of the call.

In the first approach the UEs indicate in their release messages that the signaling connection should not immediately be released by indicating a flag comparable to "follow on proceed". This causes the MSCs to change the RABs to signaling only and allows for subsequent service setup.

The second approach can be summarised as following:

a) When one user initiates "re-dial", that user's mobile avoids the release of the RR/RRC radio connection by initiating a new CM level connection (eg for USSD) between the mobile and the network.

b) The USSD string causes the initiator's HLR to invoke a Mobile Terminating Supplementary Service Transaction with the terminating mobile.

c) Only once the initiating mobile has received an acknowledgement from the HLR that the HLR has contacted the terminating mobile, does the initiator release the voice (or video) call and setup the video (or voice) call.

This approach uses existing phase 2 GSM 24.008/24.010 signalling, plus some bespoke HLR/IN node functionality. Whether or not existing MSCs can handle this is an issue of MSC functionality, not standardisation.

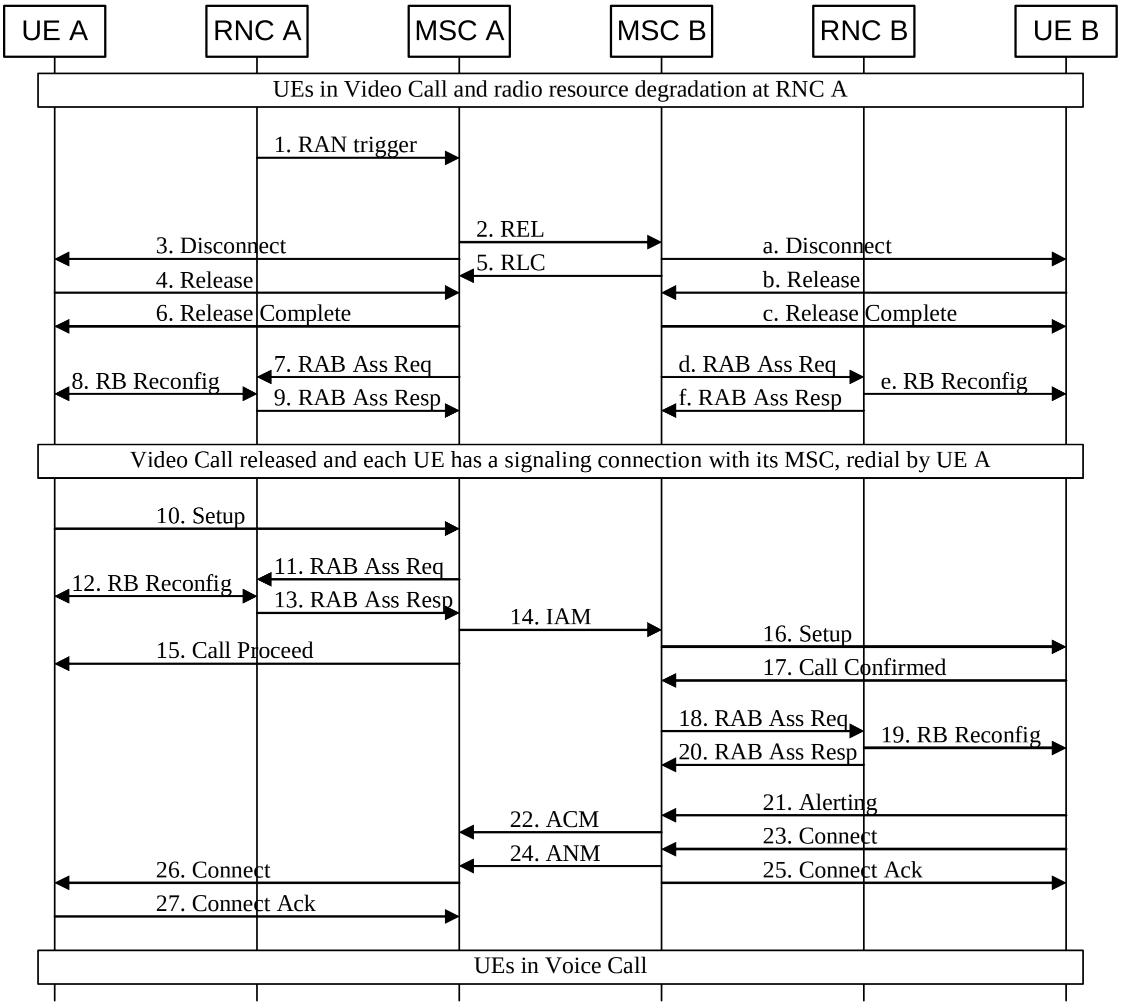
For the cases where 3G coverage degrades, some IN (CAMEL) based techniques could be imagined:

a) When the quality of the 64 kbit/s radio bearer degrades to an unacceptable level, potentially after unsuccessful handover/relocation trials, the RNC sends a trigger, e.g. a RAB Release Request, to the MSC.
b) During the call set up procedure, the MSC has had an IN detection point armed for the Disconnect procedure. When the trigger is received, the MSC then contacts an IN platform.
c) The IN platform causes MT-USSD to be initiated towards BOTH mobiles and then lets the Video
d) Disconnection take place.
e) The IN platform then uses its conferencing capabilities to initiate MT voice calls towards both mobiles, and when they answer, to join the two calls together (following a suitable announcement).

Because an IN state machine is needed in the B party's V-MSC, at least CAMEL phase 3 is needed for this feature. There is also a dependency on the RNC, in that, it needs to send a trigger, for example a "RAB release" message rather than an "Iu release" message, when the radio bearer degrades below 64 kbit/s quality.

5.4.2 Fallback from video to voice

The signaling flow below shows the approach that maintains the signaling connection on explicit UE request. The side, which experiences the radio resource degradation performs the redial. The other side has typically no knowledge about the release reasons. ISUP signaling enhancements, e.g. the use of ISUP PRI messages would be needed to transfer such information, which would allow an automatic redial.



1. After potential handover/relocation trials the RNC detects that for UE A the transmission quality has moved below the quality threshold set for the 64k VT bearer. The RNC A then sends a message to MSC A that triggers the network-initiated fallback of the video call to a voice call.
2. MSC A sends a REL message to the MSC B.
3. MSC A sends a DISCONNECT message to the UE A probably with an error cause indicating “insufficient radio resources”.
4. UE A releases the Video Call. The release message may indicate something like “follow on proceed” to delay the release of the signaling connection between UE and MSC.
5. MSC B sends a RLC message to the MSC A.
6. MSC B confirms the release of the Video Call.
7. The RAB ASSIGNMENT REQUEST message is sent from the MSC A to the RNC A, requesting the modification of the RAB for VT to a signaling RAB.
8. The radio bearer is modified between RNC A and UE A.
9. RNC A responds to MSC A with an RAB ASSIGNMENT RESPONSE message indicating that the radio bearer was modified.
10. Here is the difference between Connected mode and Idle mode redial. On the still established signaling connection UE A, which received the release cause “insufficient radio resources”, sends a SETUP message to establish a Voice Call with UE B.
11. A RAB ASSIGNMENT REQUEST message is sent from the MSC A to the RNC A, requesting the modification of the RAB for a Voice call.
12. The radio bearer is modified between RNC A and UE A.
13. RNC A responds to MSC A with an RAB ASSIGNMENT RESPONSE message indicating that the radio bearer was modified.
14. MSC A sends an IAM message to MSC B to establish a Voice Call with UE B.
15. MSC A sends a CALL PROCEED message to UE A.
16. The MSC B sends a SETUP message to UE B indicating the establishment of a Voice Call.
17. UE B sends CALL Confirmed to MSC B.
18. The RAB ASSIGNMENT REQUEST message is sent from MSC B to the RNC B, requesting the modification of the RAB for to a RAB for a Voice Call.
19. The radio bearer is modified between the RNC B and UE B.
20. RNC B responds to MSC B with a RAB ASSIGNMENT RESPONSE message indicating that the radio bearer is being modified.
21. UE B sends ALERT message to MSB B.
22. MSC B sends ACM message to MSC A.
23. User or UE accept the Voice Call and UE B sends CONNECT message to MSC B.
24. MSC B sends ANM message to MSC A.
25. MSC B sends CONNECT ACK message to UE B.
26. MSC A sends CONNECT message to UE A.
27. UE A acknowledges with a CONNECT ACK message to MSC A and the Voice call is established.

a. MSC B sends a DISCONNECT message to UE B indicating the release of the Video Call.

b. UE B sends a RELEASE message to MSC B. The UE may indicate the support of Redial to delay the release of the signaling connection between UE and MSC.

c. MSC B sends a RELEASE COMPLETE message to UE B.

d. A RAB ASSIGNMENT REQUEST message is sent from MSC B to the RNC B, requesting the modification of the RAB for to a signaling RAB.

e. The radio bearer is modified between the RNC B and UE B.

f. RNC B responds to MSC B with a RAB ASSIGNMENT RESPONSE message indicating that the radio bearer is being modified.

5.4.3 Advantages of the mechanism

The UE does not enter Idle Mode at a fallback or at a change between video and voice. A return to Idle Mode may result in delays before new services can start, as the UE may need to perform cell selection and mobility management procedures first.

Redial interoperates also with UEs that implement TS11 and BS30/MM completely separated. And it interoperates with fixed network terminals.

5.4.4 Issues that need to be resolved

To allow for automated redial and automated call answering in case of fallback the release messages needs to indicate proper release causes on ISUP and from MSC to UE, e.g. “resources not available”. This might require the sending of a ISUP PRI message as causes indicated by ISUP release messages are typically modified at network borders. Without that PRI message the REL message should indicate a proper release cause. Otherwise an A-side MSC can most likely not distinguish between normal and abnormal call termination by the B-side. Both, the transfer of PRI messages and the modification of release causes are subject to operator policies.

Also user initiated service change requires specific release causes for automated call answering otherwise it is a new call for the B-party, i.e. the user has to accept a new call.

The called user can only refuse the video call and not indicate that also the redial of a voice call will not be accepted. Shall the UE or the user initiate redial when the B-party refuses the video call?

Other transactions may delay or prevent redial, e.g. waiting calls.

Race conditions might occur when both users initiate a service change in parallel. The user that initiated the service change will be charged for the service that is setup by the service change.

Supplementary services like call forwarding, barring, roaming restriction have to be configured for TS11 and BS30 with the same settings, which enables both services to reach the same endpoint in case of redial.

The RAN needs to indicate to the MSC that the 64kbit/s RAB can no longer be maintained, which differs from current RAN behavior that maintains the RAB as long as possible and releases otherwise.

At call release the MSC and the RAN have to perform a RAB modification by release of the RAB and maintaining the signaling connection with the UE, which modifies call handling. Potentially a multimedia UE needs to request that the signaling connection is maintained or the UE starts transactions that prevent the release of the signaling connection.

The use of service based handover between GERAN and UTRAN may cause additional implications.

Charging data collected only for the individual services TS11 and BS30/MM. Requires additional effort when combination to a voice/video service is needed.

UEs need to behave user friendly also when the other side does not behave according to “redial”, e.g. when no specific release causes are signaled. In this case more user interaction seems needed.

5.5 Mechanism 5: SCUDIF with ISUP

5.5.1 Description

SCUDIF with ISUP provides the same service as the already specified SCUDIF. It signals at multimedia call setup information elements for speech and multimedia. The user may decide which mode (speech or multimedia) is used initially and may also modify the call later. Also the originating and terminating networks may change the service to speech when multimedia is not supported as a service or because of lack of resources. In difference to the already specified SCUDIF, which uses BICC codec negotiation for inter-MSC and inter-PLMN signalling, it uses ISUP for inter-MSC and inter-PLMN signalling.

SCUDIF with ISUP consists of the following functional components:

- Normal call setup.
- Fallback to speech at call setup.
- Service change in the active state.
- Network initiated fallback to speech.

SCUDIF with ISUP provides the same services feature details as described for SCUDIF. All functionality and signaling between UE and MSC is the same as for SCUDIF. Only the inter-MSC signalling and related functionality is changed to ISUP.

To deploy and run SCUDIF with ISUP several prerequisites must be in place:

- Both terminals must support SCUDIF
- Both RANs must support RAB modification and 64 kbit/s bearer
- The Core Network must support SCUDIF.

If one of these prerequisites is missing the call will be either in “voice-only” (most likely) or in “multimedia-only”, i.e. without the possibility to modify or fallback to a voice call.

The ability to traverse ISUP transit networks can be done by using the ISUP Application Transport mechanism, see ITU-T Q.765.5 Signalling system No. 7 – Application transport mechanism: Bearer Independent Call Control (BICC) [2].

If the external network does not support ISUP Access Transport Mechanism then a fallback to a single service shall be made as defined in the 3GPP TS 23.172 [1], subclause 4.3.8 “Interworking with external networks”.

5.5.2 Normal call setup

The procedures for the Access Call Control Signalling and HLR interrogation are as defined for SCUDIF but the network procedures are performed using ISUP. The circuit reserved for the SCUDIF call shall be a 64 kbit/s UDI circuit so that there is no need to modify or change the physical circuit between the MSCs.

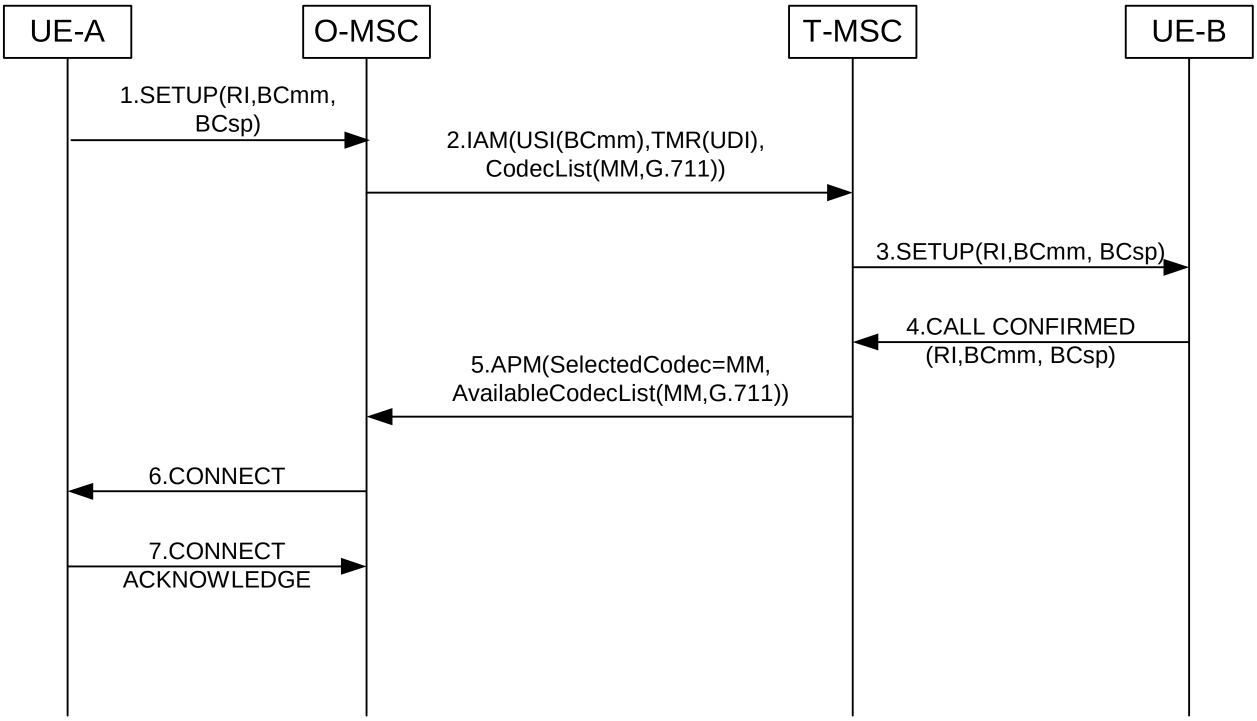


Figure 5.n: Message flow for normal call setup

1. UE-A sends a SETUP message with a Repeat Indicator set to "support of service change and fallback", a multimedia BC-IE, and a speech BC-IE to O-MSC.

2. After checking the provisioning and verifying that the SCUDIF functionality is supported, the MSC reserves a 64 kbit/s UDI circuit and sends and ISUP IAM message to T-MSC with the USI set to the preferred service ISDN BC and TMR set to value UDI. Additionally the codec list containing the multimedia codec and the G.711 codec is sent in the Application Transport (APP) parameter of the IAM message.

3. When the IAM is received by the T-MSC it shall check if the called user is provisioned for the service and sends the SETUP to the UE-B with a Repeat Indicator set to "support of service change and fallback", a multimedia BC-IE, and a speech BC-IE.

4. The UE-B returns the two BC-IEs in the same order (to indicate that it accepts the proposed settings) to the terminating MSC in the CALL CONFIRMED message.

5. The T-MSC sends an ISUP APM message to the O-MSC indicating the SelectedCodec (MM) and the AvailableCodecList (MM,G.711).

6. The O-MSC sends CONNECT to the UE-A.

7. The UE-A replies with a CONNECT ACKNOWLEDGE to the O-MSC.

5.5.3 Fallback to speech at call setup

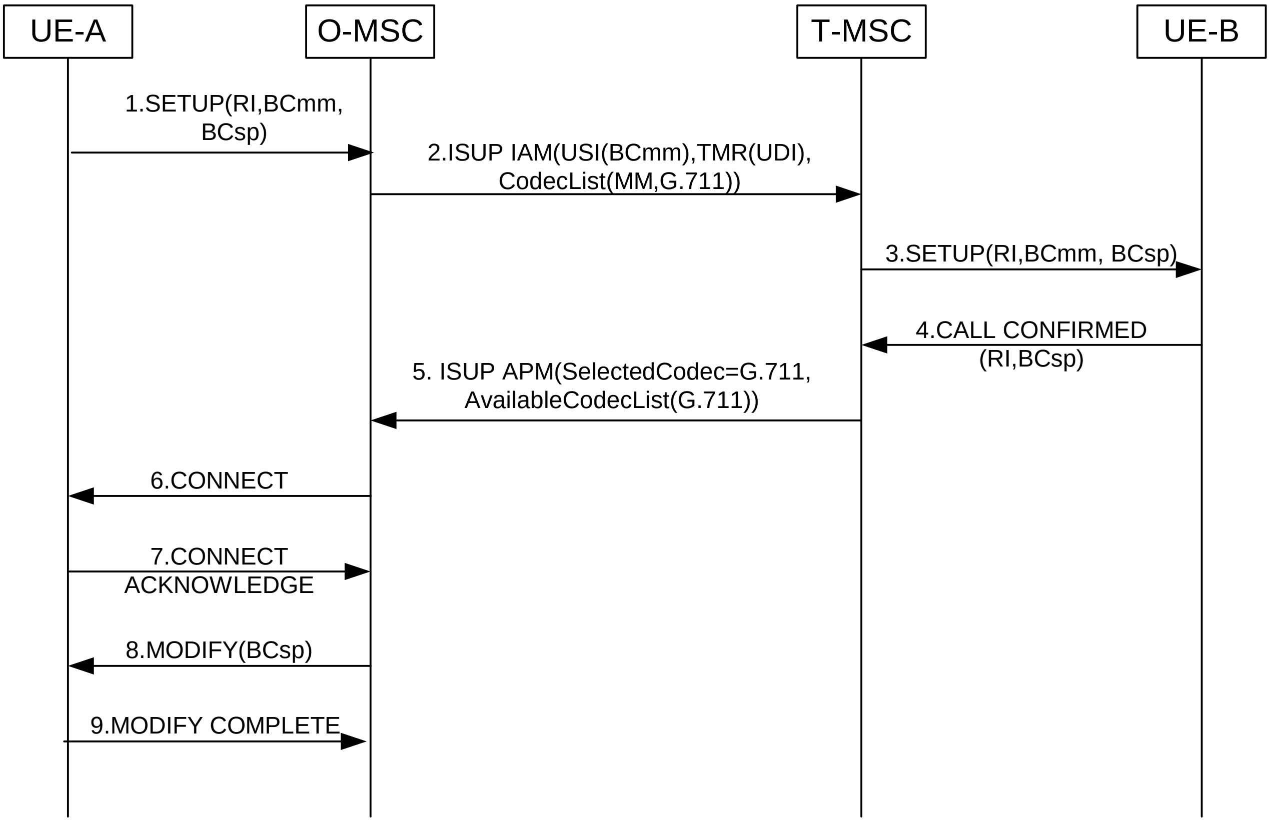


Figure 5.n+1: Message flow for fallback to speech at call setup

1. UE-A sends a SETUP message with a Repeat Indicator set to "support of service change and fallback", a multimedia BC-IE, and a speech BC-IE to O-MSC.

2. After checking the provisioning and verifying that the SCUDIF functionality is supported, the MSC reserves a 64 kbit/s UDI circuit and sends and ISUP IAM message to T-MSC with the USI set to the preferred service ISDN BC and TMR set to value UDI. Additionally the codec list containing the multimedia codec and the G.711 codec is sent in the Application Transport (APP) parameter of the IAM message.

3. When the IAM is received by the T-MSC it shall check if the called user is provisioned for the service and the T-MSC sends the SETUP to the UE-B with a Repeat Indicator set to "support of service change and fallback", a multimedia BC-IE, and a speech BC-IE.

4. The UE-B makes a fallback to speech and returns the BCsp to the T-MSC in the CALL CONFIRMED message.

5. The T-MSC sends an ISUP APM message to the O-MSC indicating the SelectedCodec (G.711) and the AvailableCodecList (G.711).

6. The O-MSC sends CONNECT to the UE-A.

7. The UE-A replies with a CONNECT ACKNOWLEDGE to the O-MSC.

8. Next the O-MSC initiates an In-Call Modification procedure towards the originating UE after the call control entity has entered the active state, i.e. the CONNECT message has been sent.

9. The UE-A sends the MODIFY COMPLETE message to acknowledge the modification.

5.5.4 Service change in the active state

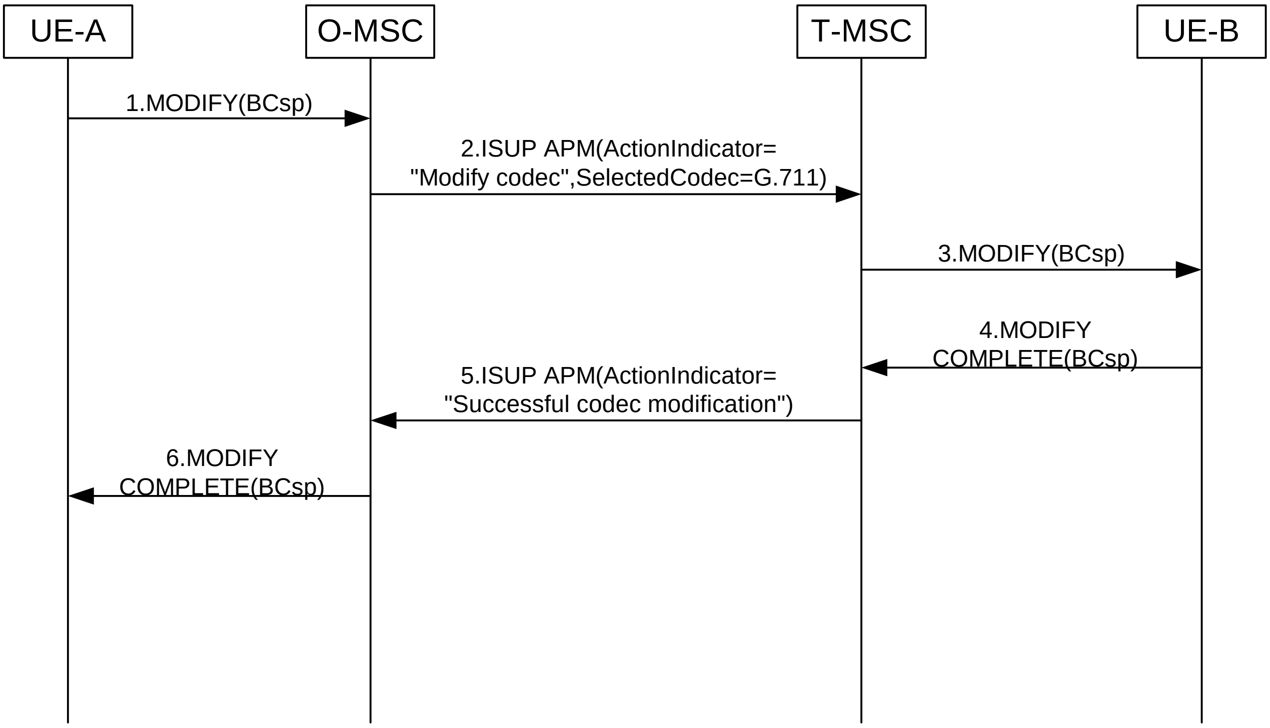


Figure 5.n+2: Message flow for service change in active state

1. UE-A activates an In-Call Modification procedure to change from multimedia to speech. The BC included in the MODIFY message shall be one of those already negotiated at call setup.

2. The O-MSC shall then invoke the service change procedure towards the UE-B by sending the ISUP APM message to the T-MSC containing a modify indication and the proposed codec.

3. The T-MSC shall initiate an In-Call Modification procedure towards the UE-B using the MODIFY message containing the BCsp.

4. If the change is accepted, the UE-B shall reply to the MSC with a MODIFY COMPLETE message containing BCsp, whereas a MODIFY REJECT message shall be sent if the change is rejected.

5. The T-MSC sends the ISUP APM message to the O-MSC indicating successful modification.

6. Next the O-MSC shall reply with a MODIFY COMPLETE message to the UE-A. to acknowledge the modification initiated by the UE-A.

5.5.5 Network initiated fallback to speech

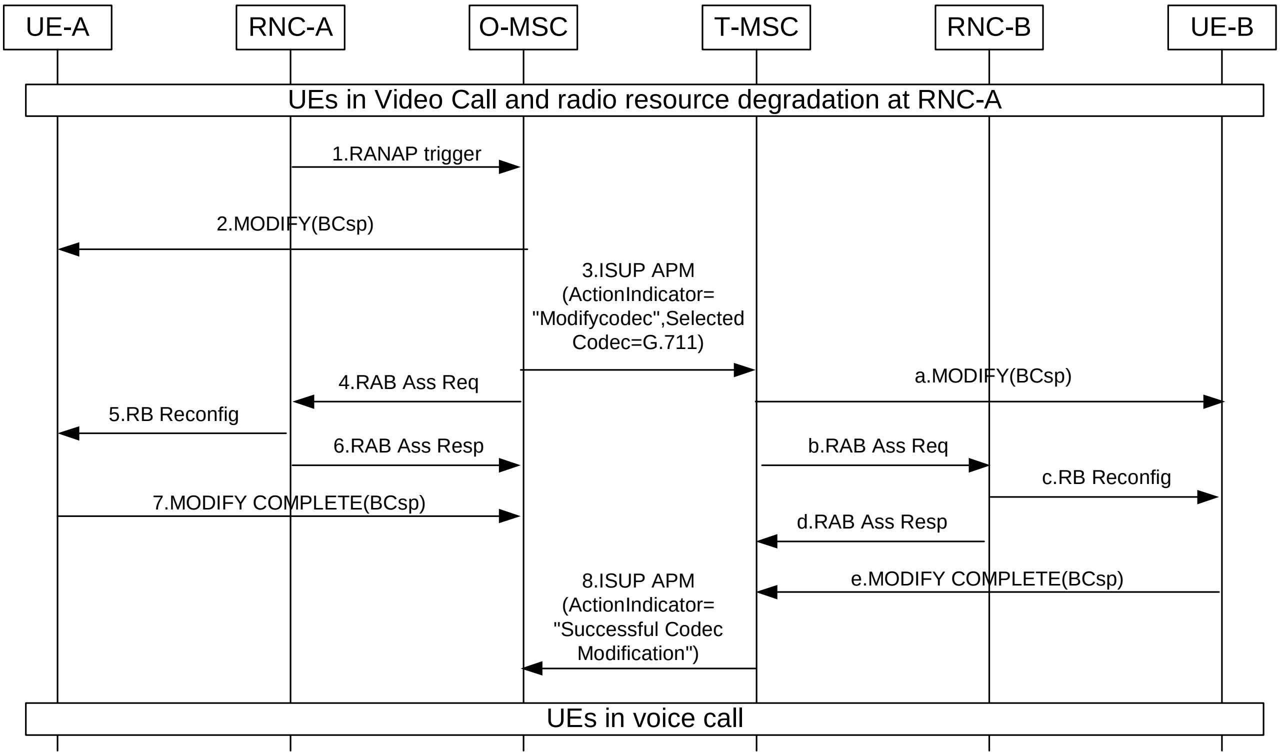


Figure 5.n+3: Message flow for network initiated fallback to speech

1. The RNC detects that for UE A the transmission quality has moved below the quality threshold set for the 64 kbit/s VT bearer. After potential handover/relocation trials, the RNC A then sends a RANAP message to MSC A. MSC A understands the reception of the message to mean that the video call cannot be maintained and should be switched to speech.

2. The MSC A sends the MODIFY message to the UE A indicating the change to bearer capability speech.

3. The MSC A sends an ISUP APM message with the parameter values: Action Indicator= “Modify codec”, Selected Codec=”value indicating one of the previously negotiated speech codecs ”.

4. The RAB ASSIGNMENT REQUEST message is sent from the MSC A to the RNC A, requesting the modification of the RAB for VT to a RAB for a Voice call.

5. The radio bearer is modified between RNC A and UE A.

6. RNC A responds to MSC A with an RAB ASSIGNMENT RESPONSE message indicating that the radio bearer was modified.

7. The MODIFY COMPLETE message is sent by UE A to acknowledge the change of the bearer capability.

NOTE: This response can happen before step 4.

a) The MSC B sends the MODIFY message to UE B indicating the change to bearer capability speech.

b) The RAB ASSIGNMENT REQUEST message is sent from MSC B to the RNC B, requesting the modification of the RAB for VT to a RAB for a Voice call.

c) The radio bearer is modified between the RNC B and UE B.

d) RNC B responds to MSC B with a RAB ASSIGNMENT RESPONSE message indicating that the radio bearer was modified.

NOTE: This response can happen before step b).

e) The MODIFY COMPLETE message is sent by UE B to acknowledge the change of the bearer capability.

8. MSC B sends an ISUP APM message with the parameter values: Action Indicator= "Successful Codec Modification" to MSC A indicating that the bearer has been successfully modified.

5.5.6 Advantages of the mechanism

The advantages are the same as for SCUDIF. Any time the users may change between speech and multimedia. The user at the terminating end may refuse a service change without terminating the call.

At call setup and during the call the network may fallback to speech due to lack of resources.

The general benefits of the SCUDIF feature is that it allows users to attempt to initiate combined "Speech" and "Multi-media" calls and if the network or called subscriber can not or prefers not to support the multimedia service the user still gets through-connected with "speech" quickly, thus retaining some quality of service and customer satisfaction. In addition it allows two parties conversing via normal "speech" to transmit UDI payload (video, still pictures, other data) and return to normal "speech", thus only using and paying for the higher rate services, when needed.

Solutions for prepaid and post-paid charging are standardized. The network correlates speech and multimedia, allowing flexible charging solutions.

The users at the originating and terminating side do not experience additional delay compared to an ordinary call setup attempt. Users with SCUDIF capable terminals are therefore more likely to configure their terminals to attempt a video call with fallback by default.

At call setup, the called party is informed about the choice between speech and multimedia.

SCUDIF is one call or service of two modes, which minimises impacts by other services during mode change compared to other mechanisms.

Rules for the interacting of speech and multimedia supplementary services are standardized, which simplify the configuration. To a certain extent supplementary services like call forwarding, barring, roaming restriction have to be configured for TS11 and BS30 with the same settings. Otherwise, a call with the preferred service but no fallback or service change capability is established.

5.5.7 Issues that need to be resolved

The new ISUP parameters require the reservation of codepoints within ITU-T SG11 to avoid a usage of the same codepoints for other purposes. The semantics could also be defined with ITU-T SG11, or within 3GPP if ITU-T SG 11 accepts that the service is mostly of interest for mobile networks (where a mid-call service change may be motivated by expensive resources or a degrading connection).

ISUP messages and parameters are subject to operator policies. The new parameter and APM message need to be signaled unmodified between PLMNs. This might require specific configuration of all functions that inspect and modify signalling. Otherwise the APM message and its parameters do not traverse PLMN borders or transit networks.

A transparent 64 kbit/s bearer is established between the MSCs. Any interworking, e.g. echo cancellation or A-law/u-law conversion, has to be performed by the MSCs or specific gateways are required that react on the ISUP APM message and parameters.

How legal intercept would work with SCUDIF needs to be studied.

If SCUDIF with BICC and ISUP is used interworking may be required.

6 Comparison of the different mechanisms

6.1 Differences between the different mechanisms

Editor's note:< text to be added >

6.2 Common Issues

6.2.1 Trigger for fallback to voice

This trigger is intended to initiate the fallback from video to voice when the quality of the 64kbit/s radio bearer degrades to an unacceptable level.

A number of the information flows in this TR assume that the RAN provides a trigger for network-initiated fallback to voice. Generally, the RAN is service agnostic and maintains the radio bearer by handover or relocation to available resources. If no better cell is available, the 64kbit/s RAB is maintained as long as possible in the current cell and finally the connection towards the UE is lost and the related RAB is released. This release is not suited to trigger a fallback as in this situation the signalling connection is lost. Moreover, the video quality was then already for some time below an enjoyable level.

A combined 2G/3G network might use the service based handover mechanism "handover to GERAN should not be performed". In this case a handover required to GERAN indicates obviously that no suitable 3G is available. And when GERAN does not support 64kbit/s the MSC may use the handover required or a handover failure as a trigger for the network initiated fallback. Problematic might be the use of the compressed mode that is necessary for monitoring of GERAN cells, which worsens the radio bearer quality in addition.

When the RAN shall trigger the fallback to voice the radio bearer quality degradation needs to be indicated in time to the MSC. In order for the RNC to be able to indicate towards the MSC about the need for the fallback to speech well in time, the RNC needs to be aware of the possibility to initiate a service change. The MSC indicates during the RAB setup towards the RAN about the possibility for the network initiated service change during the call. Based on this information the RNC can fine-tune the thresholds of RRM features for the RAB having the service change possibility. In degrading radio conditions the RNC indicates towards the MSC about the need for the fallback to speech.

It might also be difficult to define such a criterion for the RAN, as the Video service quality is not directly proportional to the radio bearer quality. It depends, for example, on the video codec and its error correction performance. In the current architecture the selected video codec and its characteristics are not known to the CN and RAN. The RAN can only implement thresholds for the radio bearer but not dependent on the specific video service. Furthermore, users have different level of acceptance of service degradation.

Another approach for a fallback trigger may be provided by the application on the UE. It may measure the video service quality close to the user perception. The application on the UE could trigger a service change more reliable. This fallback looks like a user initiated service change from the network's perspective.

Alternatively the application on the UE could propose to the user a service change, which may adapt better to different user acceptance of service degradation than a threshold implemented in the UE application.

6.2.2 Trigger for return to video

This trigger is intended to initiate the return to video once a fallback to voice happened and network resources are available for a 64kbit/s radio bearer.

Generally, the RAN is service agnostic and uses UE measurements and RAN resource control to maintain the established RAB, i.e. it aims at maintaining the parameters requested for the RAB.

When the RAN shall trigger the return to video the radio bearer availability needs to be indicated to the MSC. In order for the RNC to be able to indicate towards the MSC about a chance to return to video, the RNC needs to be aware of the possibility to initiate a service change. The MSC indicates during the RAB setup towards the RAN about the possibility for the network initiated service change during the call. Based on this information the RNC can tune its RRM features.

When radio conditions allow a return to video the RNC indicates this towards the MSC. This requires the RNC to continuously monitor the radio resources looking for the opportunity to provide a radio bearer for video.

It might be difficult to define such a criterion for the RAN, as the Video service quality is not directly proportional to the radio bearer quality. It depends, for example, on the video codec and its error correction performance. In the current architecture the selected video codec and its characteristics are not known to the CN and RAN. The RAN can only implement thresholds for the radio bearer but not dependent on the specific video service. Furthermore, users have different level of acceptance of service degradation.

Such an automatic return attempt to video should be made visible to the users. Otherwise, the users might experience a service interruption of probably a few second and don't know why. This would be the case when the RAN on one side indicates 64kbit/s are available, the connected MSC modifies the service and the RAB to video, switches successfully to a 64kbit/s bearer and sends the modify to the other MSC. The other MSC or the other RAN might reject the modification, for example because of lack of resources or the UE is on GERAN. The modification is rejected to the first MSC and this MSC modifies the bearer back to voice. At least the UE on the initiating side should inform the user and display, for example "service change attempt".

The network may need mechanisms to avoid too frequent attempts of changing back to video, which might happen when only one side has resources available for video.

Preferably the users are asked before a service change. So the UEs could inform the user when a change back to video is possible, e.g. derived from UTRAN coverage and additional radio criteria. Then the users can decide to try a change to video and would be aware that the communication is interrupted during the return to video attempt.

7 Summary and Conclusions

7.1 Summary of open issues with the different mechanisms

7.1.1 SCUDIF (either BICC based or ISUP based)

This mechanism probably requires changes to operator’s existing commercial arrangements before it can be used. Specification changes may be needed to ensure the correct interaction between mobiles and networks. Interaction with SA 1 is required to clarify the requirement on who should pay for eg a video call when the B party “pushes to video”.

7.1.2 Dual Call

This mechanism requires some changes to the MSC/RNC/BSC interactions and limits some tariff models. However existing commercial arrangements need not be changed.

7.1.3 Re-dial with release of the radio connection

This is the only mechanism that does not mandate network changes. However, at least some BSC changes are needed to avoid poor performance (eg long gaps at service change and/or high redialled call setup failure rate) caused by ‘accidental’ or ‘systematic’ handover of voice calls to 2G.

7.1.4 Re-dial using the same radio connection

This mechanism requires some changes to the MSC/RNC/BSC interactions and some HLR functionality. However, existing commercial arrangements need not be changed.

7.2 Conclusions

None of the approaches fully meets both, all the service requirements, and the requirements of operators for fast deployment.

The only mechanism that offers scope for deployment of a system complying to the anticipated R’6 specification, and, aligns with existing inter-operator commercial arrangements is “re-dial with release of the radio connection”.

However "re-dial with release of the radio connection" does not totally fulfil all the service requirements. Hence work on enhancing/completing SCUDIF will also continue.

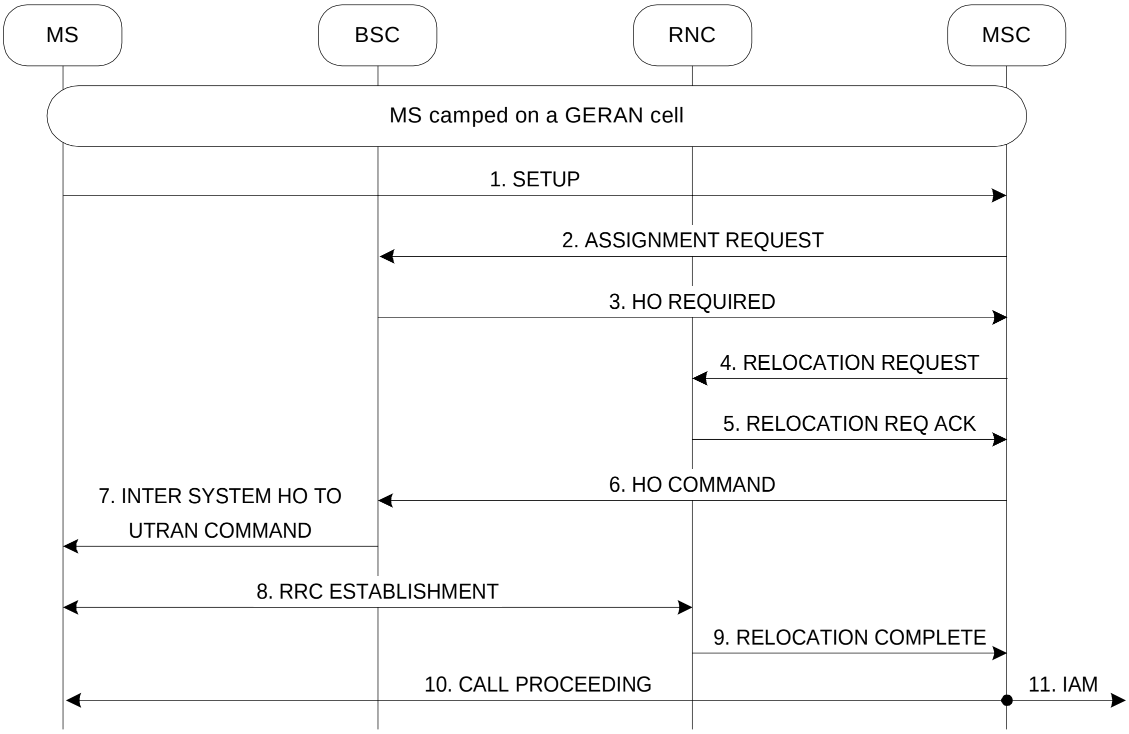
Annex A (informative): Dual Call Message Flows

A.1 Service based handover in GERAN

A.1.1 Mobile originated VT call

A.1.1.1 Idle mode

A.1.1.1.1 Single bearer capability – successful



1. The MS requests a video call by passing the SETUP message to the MSC. The SETUP message includes one BCIE with Other Rate Adaption set to "H.233 & H.245".

2. The MSC, after completing subscription verification for Bearer Service 37, indicates to the BSC in an ASSIGNMENT REQUEST message that a signalling channel is required. This message also includes the Service Handover IE, which is set to "Handover to UTRAN should be performed". The Circuit Identity Code is not included because the Channel Type is set to signalling.

3. The BSC, after gathering the correct UTRAN neighbour cell measurements from the MS, passes a HANDOVER REQUIRED message to the MSC. The Cell ID Discriminator field in the CI List IE lets the MSC know the identity of the target RNC. The Source RNC to Target RNC transparent information container identifies to the target RNC the identity of the target cell.

4. The MSC indicates, in the RAB parameters of the RELOCATION REQUEST message, to the RNC that a 64k bearer is required.

5. The target RNC sends a RELOCATION REQUEST ACK message to the MSC informing the MSC that the resources for the MS have been successfully allocated in the target cell.

6. The MSC sends the HANDOVER COMMAND message to the BSC indicating that the MS should be instructed to move to UTRAN.

7. The BSC sends the INTER SYSTEM HANDOVER TO UTRAN COMMAND message to the MS commanding the MS to move to the new cell.

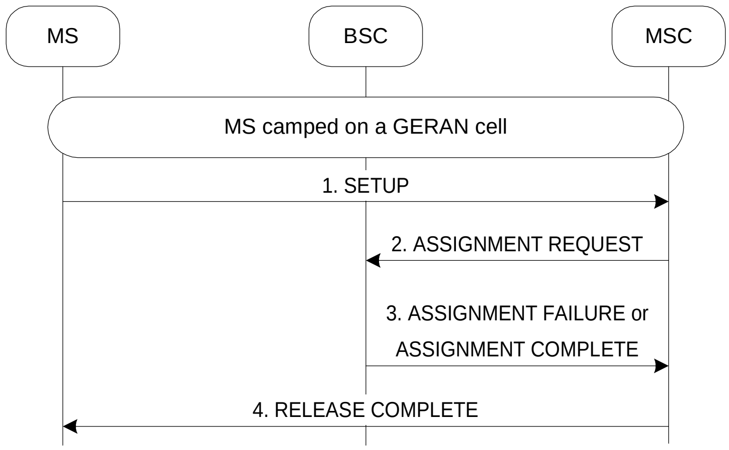
8. Once the MS arrives in UTRAN coverage, the MS synchronises with the Node B and establishes the RRC connection.

9. The target RNC informs the MSC, with the RELOCATION COMPLETE message, that the MS has been successfully completed the handover to UTRAN procedures.

10. After the MS arrives on the UTRAN cell the MSC indicates to the MS that the establishment of the call is progressing by sending the CALL PROCEEDING message.

11. The MSC sends the IAM message towards the B-party.

A.1.1.1.2 Single bearer capability – failure



1. The MS requests a video call by passing the SETUP message to the MSC. The SETUP message includes one BCIE with Other Rate Adaption set to “H.233 & H.245”.

2. The MSC looks up the Bearer Service (BS) to which the BCIE aligns. The BCIE aligns to BS37 and the MSC verifies that this MS is provisioned for BS37.

   If the MS is not provisioned the MSC skips to message 4, RELEASE COMPLETE including cause #57 "bearer capability not authorized".

   If the MS is provisioned the MSC indicates to the BSC in an ASSIGNMENT REQUEST message that a signalling channel is required. This message would also include the Service Handover IE, which is set to “Handover to UTRAN should be performed” and the Circuit Identity Code is not included when the Channel Type is set to signalling.

3. The BSC will respond to the MSC with an ASSIGNMENT COMPLETE message if the Service Handover IE is not understood by the BSC.

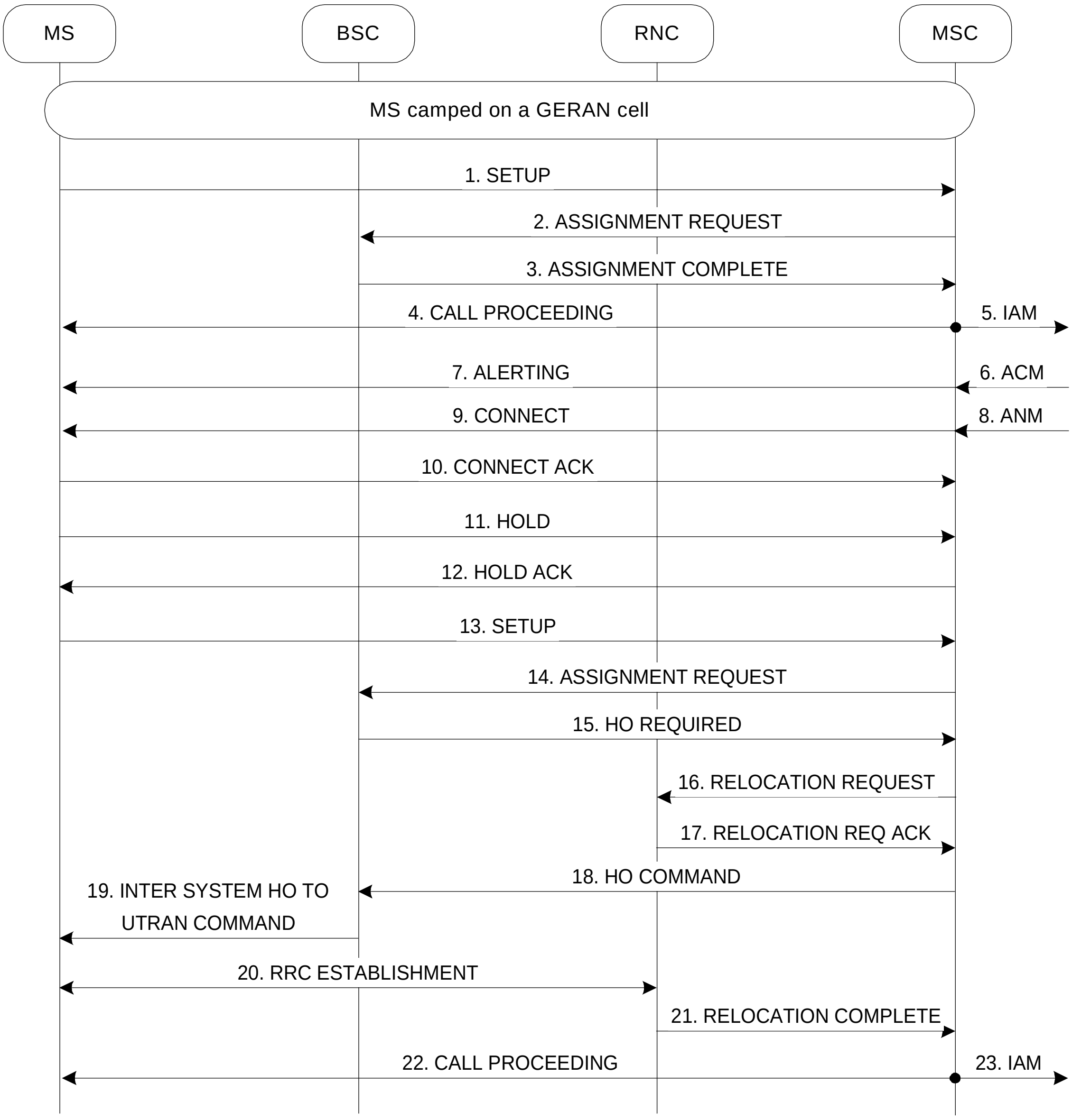
   The BSC may also respond to the MSC with an ASSIGNMENT FAILURE message if the handover to UTRAN cannot be completed due to one of the following reasons:

a) MS not it UTRAN coverage: cause sent to MSC in ASSIGNMENT FAILURE = “No radio resource available”, cause #58 returned to MS in RELEASE COMPLETE = “bearer capability not presently available“.

b) If BSC is not capable of initiating Handover to UTRAN: cause sent to MSC in ASSIGNMENT FAILURE = “BSS not equipped”, cause #65 returned to MS in RELEASE COMPLETE = "bearer service not implemented".

4. The MSC sends a RELEASE COMPLETE message to the MS.

A.1.1.1.3 Dual Call - successful voice establishment



The User decides to initiate a video call by hitting the “Video” button. The MMI in the handset is programmed to first request a speech call and then request a video call.

1. The MS sends a SETUP message to the MSC requesting a speech call.

2. The MSC sends an ASSIGNMENT REQUEST message to the BSC requesting that radio resources be allocated to the MS for a speech call.

3. The BSC responds to the MSC with an ASSIGNMENT COMPLETE message informing the MSC that the radio resources have been allocated to the MS.

4. The CALL PROCEEDING message is then transmitted to the MS from the MSC indicating that the MS has been allocated a bearer for speech.

5. The MSC sends the IAM message requesting the establishment of a speech call towards the B-party.

6. The MSC receives the ACM message from the B-party.

7. The MSC informs the MS that the B-party has alerted the user using the ALERTING message.

8. The ANM message is received by the MSC when the B-party user has answered.

9. The CONNECT message is sent to the MS by the MSC to indicate that the call has been through connected.

10. The MS acknowledges the reception of the CONNECT message with the CONNECT ACK message.

After the speech call has successfully been set up the MMI in the MS could be programmed to automatically put this speech call on hold and initiate a video call.

11. The MS sends the HOLD message to the MSC to place the current voice call on hold.

12. The MSC responds with a HOLD ACK message indicating that the voice call has been placed on hold.

13. The MS requests a video call by passing the SETUP message to the MSC. The SETUP message includes one BCIE with Other Rate Adaption set to "H.233 & H.245" and the called number is the same as for the voice call.

14. The MSC, after completing subscription verification for Bearer Service 37, sends to the BSC an ASSIGNMENT REQUEST message identical (including CIC) to the previous ASSIGNMENT REQUEST message except that it also includes the Service Handover IE set to "Handover to UTRAN should be performed".

15. The BSC, after gathering the correct UTRAN neighbour cell measurements from the MS, passes a HANDOVER REQUIRED message to the MSC. The Cell ID Discriminator field in the CI List IE lets the MSC know the identity of the target RNC. The Source RNC to Target RNC transparent information container identifies to the target RNC the identity of the target cell.

16. The MSC indicates, in the RAB parameters of the RELOCATION REQUEST message, to the RNC that a 64k bearer is required.

17. The target RNC sends a RELOCATION REQUEST ACK message to the MSC informing the MSC that the resources for the MS have been successfully allocated in the target cell.

18. The MSC sends the HANDOVER COMMAND message to the BSC indicating that the MS should be instructed to move to UTRAN.

19. The BSC sends the INTER SYSTEM HANDOVER TO UTRAN COMMAND message to the MS commanding the MS to move to the new cell.

20. Once the MS arrives in UTRAN coverage, the MS synchronises with the Node B and establishes the RRC connection.

21. The target RNC informs the MSC, with the RELOCATION COMPLETE message, that the MS has been successfully completed the handover to UTRAN procedures. Upon successful handover to UTRAN the MSC clears the resources allocated in the BSC (both radio resources and A interface).

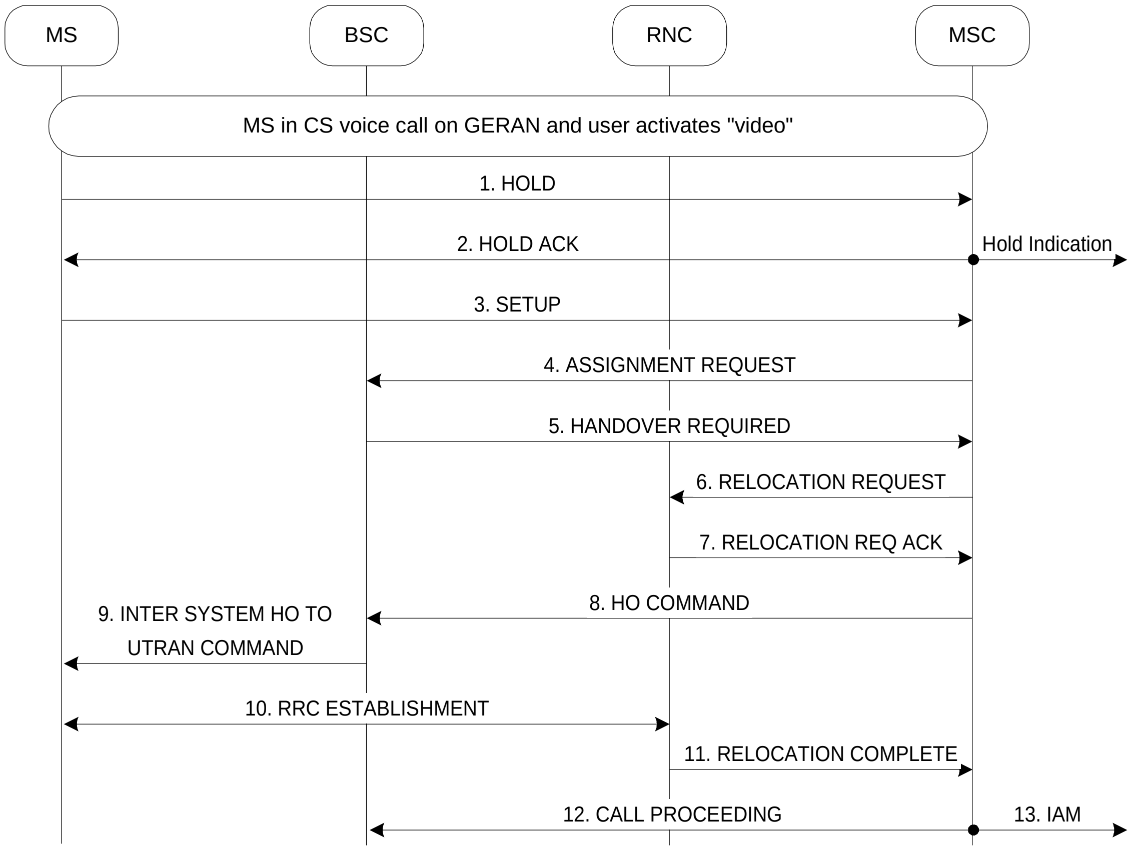
22. After the MS arrives on the UTRAN cell the MSC indicates to the MS that the establishment of the video call is progressing by sending the CALL PROCEEDING message.

23. The MSC sends the IAM message towards the B-party.

A.1.1.2 Connected mode

A.1.1.2.1 VT call originating in GERAN

A.1.1.2.1.1 VT call originating in GERAN – successful



The MS is in a voice call (either MT or MO) and the user presses the “video” button in order to have a video call with the person they are currently speaking to. The MMI in the phone is configured to then initiate a CALL HOLD and then send the SETUP message for the video call.

1. The MS sends the HOLD message to the MSC to place the current voice call on hold.

2. The MS responds with a HOLD ACK message indicating that the voice call has been placed on hold.

3. The MS requests a video call by passing the SETUP message to the MSC. The SETUP message includes one BCIE with Other Rate Adaption set to “H.233 & H.245” and the called number is the same as for the voice call.

4. The MSC, after completing subscription verification for Bearer Service 37, sends to the BSC an ASSIGNMENT REQUEST message identical (including CIC) to the previous ASSIGNMENT REQUEST message except that it also includes the Service Handover IE set to “Handover to UTRAN should be performed”.

5. The BSC, after gathering the correct UTRAN neighbour cell measurements from the MS, passes a HANDOVER REQUIRED message to the MSC. The Cell ID Discriminator field in the CI List IE lets the MSC know the identity of the target RNC. The Source RNC to Target RNC transparent information container identifies to the target RNC the identity of the target cell.

6. The MSC indicates, in the RAB parameters of the RELOCATION REQUEST message, to the RNC that a 64k bearer is required.

7. The target RNC sends a RELOCATION REQUEST ACK message to the MSC informing the MSC that the resources for the MS have been successfully allocated in the target cell.

8. The MSC sends the HANDOVER COMMAND message to the BSC indicating that the MS should be instructed to move to UTRAN.

9. The BSC sends the INTER SYSTEM HANDOVER TO UTRAN COMMAND message to the MS commanding the MS to move to the new cell.

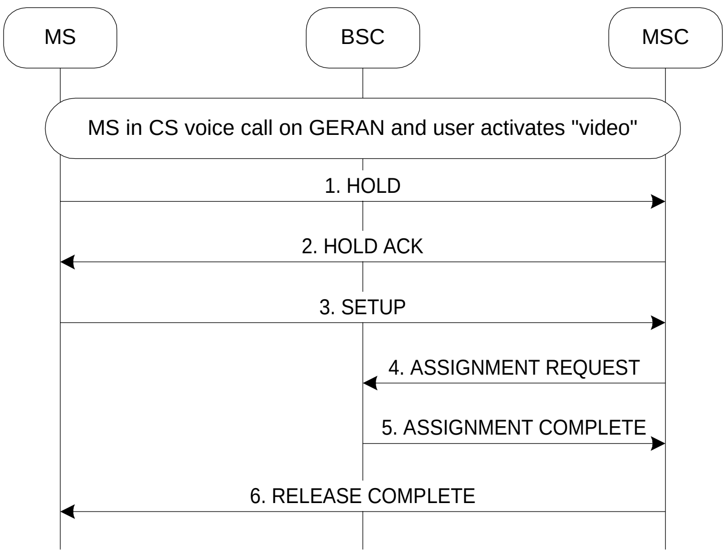
10. Once the MS arrives in UTRAN coverage, the MS synchronises with the Node B and establishes the RRC connection.

11. The target RNC informs the MSC, with the RELOCATION COMPLETE message, that the MS has been successfully completed the handover to UTRAN procedures. Upon successful handover to UTRAN the BSC clears the allocated circuit.

12. After the MS arrives on the UTRAN cell, the MSC indicates to the MS that the establishment of the video call is progressing by sending the CALL PROCEEDING message.

13. The MSC sends the IAM message towards the B-party.

A.1.1.2.1.2 VT call originating in GERAN – failure



The MS is in voice call (either MT or MO) and the user presses the "video" button in order to have a video call with the person they are currently speaking to. The MMI in the phone is configured to then initiate a CALL HOLD and then send the SETUP message for the video call.

1. The MS sends the HOLD message to the MSC to place the current voice call on hold.

2. The MS responds with a HOLD ACK message indicating that the voice call has been placed on hold.

3. The MS requests a video call by passing the SETUP message to the MSC. The SETUP message includes one BCIE with Other Rate Adaption set to "H.233 & H.245".

4. The MSC, after completing subscription verification for Bearer Service 37, sends to the BSC an ASSIGNMENT REQUEST message including no change to previous ASSIGNMENT REQUEST message except the inclusion of the Service Handover IE which is set to "Handover to UTRAN should be performed". If the MS is not provisioned the MSC skips to step 6.

4. If the MS is not in UMTS coverage, or, if the BSC does not support Service Handover, then the BSC passes an ASSIGNMENT COMPLETE message to the MSC indicating that the resource "modification" has been completed. The MSC understands the reception of this message as a failure case as the Service Handover has not been initiated.

Note: Adding the Cause IE with a new cause value to the ASSIGNMENT COMPLETE message on the A interface maybe beneficial to inform the MSC whether the MS is not in UTRAN coverage or that the functionality in the BSC is not implemented.

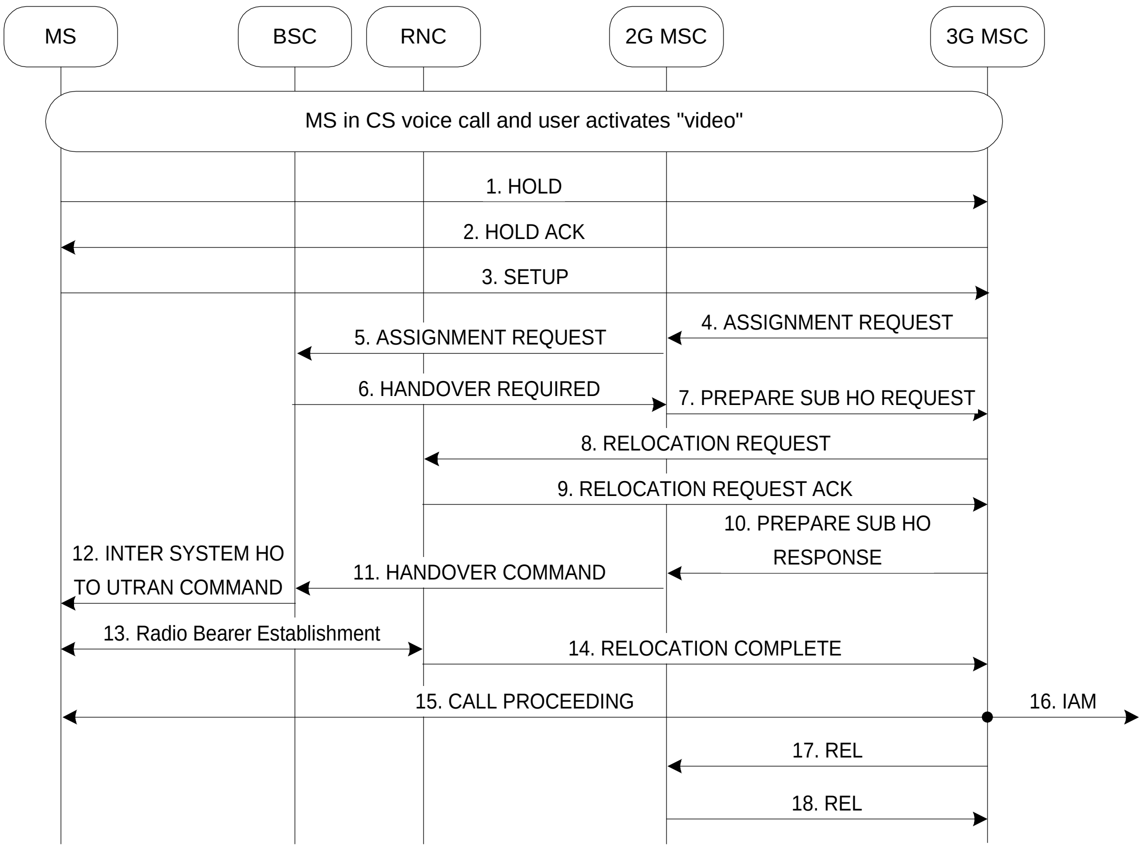
6. The RELEASE COMPLETE message for the video call is then returned to MS including, either:

   a) Cause #65, "bearer service not implemented" when the MS does not support handover; or

   b) Cause #49, "Quality of service unavailable" when the MS is not in UTRAN coverage; or

   c) Cause #57, "bearer capability not authorized".

A.1.1.2.1.3 VT call originating in GERAN after inter-MSC handover from 3G MSC – successful



The MS is in voice call (either MT or MO) that started in a UTRAN cell. The call has been handed over to GERAN coverage and then the user presses the "video" button in order to have a video call with the person they are currently speaking to. The MMI in the phone is configured to then initiate a CALL HOLD and then send the SETUP message for the video call.

1. The MS sends the HOLD message to the anchor 3G MSC to place the current voice call on hold.

2. The anchor MSC responds with a HOLD ACK message indicating that the voice call has been placed on hold.

3. The MS requests a video call by passing the SETUP message to the anchor MSC. The SETUP message includes one BCIE with Other Rate Adaption set to "H.233 & H.245" and the called number is the same as for the voice call.

4. The Anchor MSC, after completing subscription verification for Bearer Service 37, sends to the 2G MSC an ASSIGNMENT REQUEST message with the same configuration as the previous HANDOVER REQUEST message except the message includes the Service Handover IE set to "Handover to UTRAN should be

performed". Note that on the E interface between the MSCs, neither the ASSIGNMENT REQUEST nor HANDOVER REQUEST contain the Circuit Identity Code IE.

5. The 2G MSC sends an ASSIGNMENT REQUEST message to the BSC with the same configuration as message 4 except the relay MSC also includes the same Circuit Identity Code as is currently in use.

6. The BSC, after gathering the correct UTRAN neighbour cell measurements from the MS, passes a HANDOVER REQUIRED message to the relay MSC. The Cell ID Discriminator field in the CI List IE lets the relay MSC know the identity of the target RNC. The Source RNC to Target RNC transparent information container identifies to the target RNC the identity of the target cell.

7. Delete one of the options:

a) The RELOCATION REQUEST message is generated by the 2G MSC and passed to the Anchor MSC in the PREPARE SUBSEQUENT HANDOVER REQUEST message.

b) The HANDOVER REQUIRED message is passed by the 2G MSC to the Anchor MSC in the PREPARE SUBSEQUENT HANDOVER REQUEST message.

8. The anchor MSC indicates, in the RAB parameters of the RELOCATION REQUEST message, to the RNC that a 64k bearer is required.

9. The target RNC sends a RELOCATION REQUEST ACK message to the MSC informing the MSC that the resources for the MS have been successfully allocated in the target cell.

10. Delete one of the options:

a) The HANDOVER COMMAND message is generated by the 2G MSC and passed to the Anchor MSC in the PREPARE SUBSEQUENT HANDOVER RESPONSE message.

b) The RELOCATION REQUEST ACK message is passed by the 2G MSC to the Anchor MSC in the PREPARE SUBSEQUENT HANDOVER RESPONSE message.

11. The relay MSC sends the HANDOVER COMMAND message to the BSC indicating that the MS should be instructed to move to UTRAN.

12. The BSC sends the INTER SYSTEM HANDOVER TO UTRAN COMMAND message to the MS commanding the MS to move to the new cell.

13. Once the MS arrives in UTRAN coverage, the MS synchronises with the Node B and establishes the RRC connection.

14. The target RNC informs the MSC, with the RELOCATION COMPLETE message, that the MS has been successfully completed the handover to UTRAN procedures. Upon successful handover to UTRAN the anchor MSC initiates release of resources in the relay MSC and the old BSC.

15. After the MS arrives on the UTRAN cell, the 3G MSC indicates to the MS that the establishment of the call is progressing by sending the CALL PROCEEDING message.

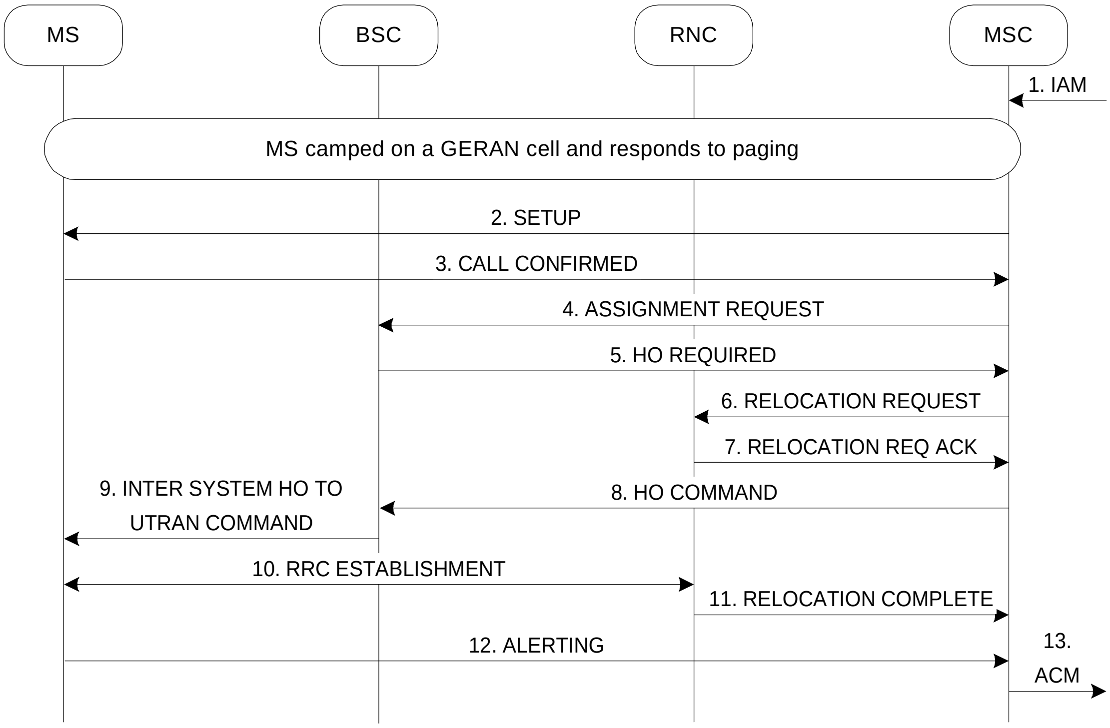
16. The MSC sends the IAM message towards the B-party.

17. The REL message is sent from the Anchor MSC to the 2G MSC instructing the MSC to release the connection.

18. The REL message is sent from the 2G MSC to the Anchor MSC acknowledging the release of the connection.

A.1.2 Mobile terminated VT call

A.1.2.1 Idle mode



1. The MSC receives an IAM message from the A-party indicating that a Video call is required. The MSC verifies the subscription of the MS to check that it is provisioned and currently allowed to receive Video calls.

2. The SETUP message shall contain all the information required by the MS to process the call. Upon receipt of a SETUP message, the MS will perform compatibility checking on the contained information and shall allow the Video call to continue even though the MS is in GERAN coverage. The SETUP message does not contain the Signal IE (and hence the MS does not Alert the user until the appropriate resources have been allocated in step 10).

3. The CALL CONFIRMED message is an acknowledgement to the SETUP message. The message can contain information about the MS capabilities.

4. The MSC indicates to the BSC in an ASSIGNMENT REQUEST message that a signalling channel is required. This message does not include a CIC because the Channel Type is set to signalling. The message includes the Service Handover IE set to “Handover to UTRAN should be performed”.

5. The BSC, after gathering the correct UTRAN neighbour cell measurements from the MS, passes a HANDOVER REQUIRED message to the MSC. The Cell ID Discriminator field in the CI List IE lets the MSC know the identity of the target RNC. The Source RNC to Target RNC transparent information container identifies to the target RNC the identity of the target cell.

6. The MSC indicates, in the RAB parameters of the RELOCATION REQUEST message, to the RNC that a 64k bearer is required.

7. The target RNC sends a RELOCATION REQUEST ACK message to the MSC informing the MSC that the resources for the MS have been successfully allocated in the target cell.

8. The MSC sends the HANDOVER COMMAND message to the BSC indicating that the MS should be instructed to move to UTRAN.

9. The BSC sends the INTER SYSTEM HANDOVER TO UTRAN COMMAND message to the MS commanding the MS to move to the new cell.

10. Once the MS arrives in UTRAN coverage, the MS synchronises with the Node B and establishes the RRC connection.

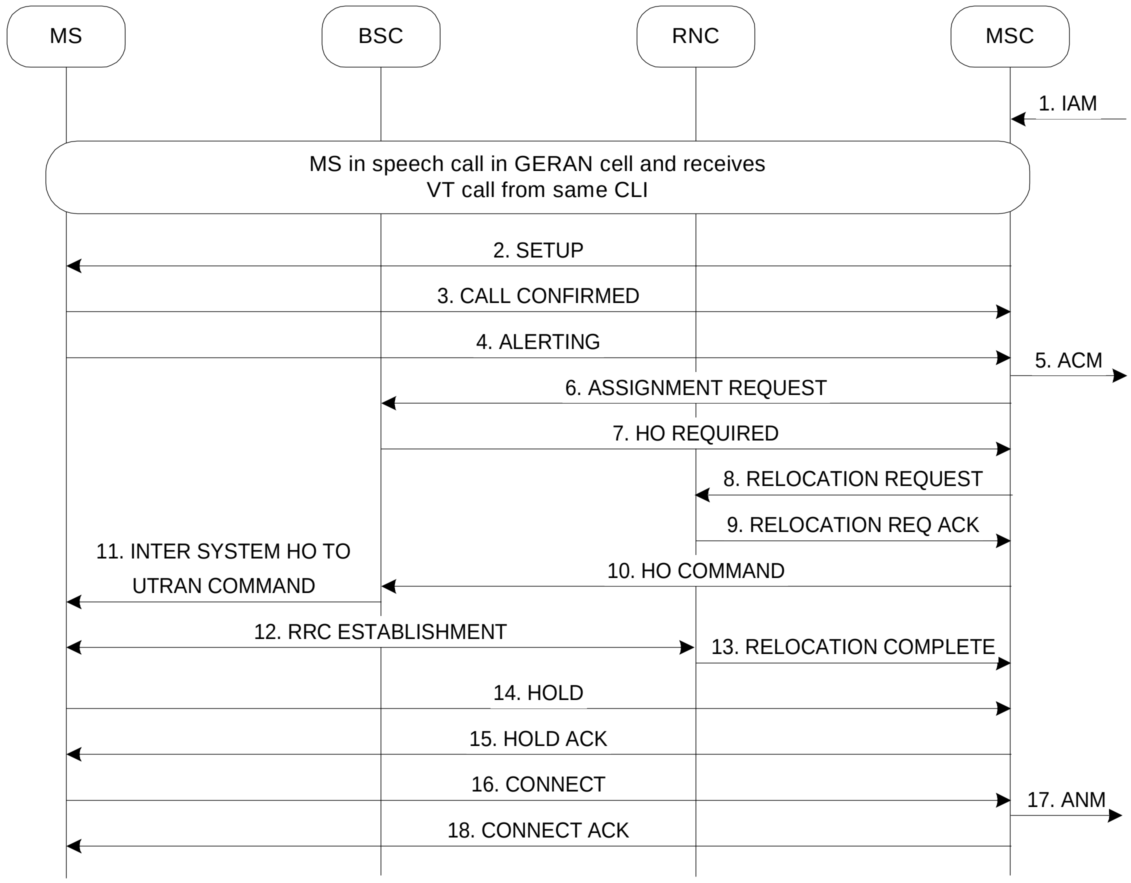
11. The target RNC informs the MSC, with the RELOCATION COMPLETE message, that the MS has successfully completed the handover to UTRAN procedures.

12. After being allocated 'appropriate' resources on the UTRAN cell, the MS indicates to the MSC that the call can progress by sending the ALERTING message.

13. The MSC sends the ACM message towards the A-party.

A.1.2.2 Connected Mode

A.1.2.2.1 Connected Mode – Voice call originating in GERAN



The MS is in voice call to user B and then the user B presses the "video" button in order to have a video call with this MS. The MSC controlling the MS has to inform the MS that there is an incoming video call and then move the MS to UTRAN coverage to allow the video call to be connected.

1. The MSC receives an IAM message from the A-party indicating that a Video call is required. The MSC verifies the subscription of the MS to check that it is provisioned and currently allowed to receive Video calls.

2. The SETUP message shall contain all the information required by the MS to process the call. Upon receipt of a SETUP message, the MS will perform compatibility checking on the contained information and shall allow the Video call to continue even though the MS is in GERAN coverage. As required by TS 48.083, the SETUP message contains the Signal IE and hence the MS immediately alerts the user.

3. The CALL CONFIRMED message is an acknowledgement to the SETUP message. The message can contain information about the MS capabilities.

4. The MS indicates to the MSC that the user is being alerted of the waiting video call by sending the ALERTING message.

5. The MSC sends the ACM message towards the A-party.

6. The MSC, after completing subscription verification for Bearer Service 37, sends an ASSIGNMENT REQUEST message with the same configuration as the previous ASSIGNMENT REQUEST message except the message includes the Service Handover IE set to “Handover to UTRAN should be performed”.

7. The BSC, after gathering the correct UTRAN neighbour cell measurements from the MS, passes a HANDOVER REQUIRED message to the MSC. The Cell ID Discriminator field in the CI List IE lets the MSC know the identity of the target RNC. The Source RNC to Target RNC transparent information container identifies to the target RNC the identity of the target cell.

8. The anchor MSC indicates, in the RAB parameters of the RELOCATION REQUEST message, to the RNC that a 64k bearer is required.

9. The target RNC sends a RELOCATION REQUEST ACK message to the MSC informing the MSC that the resources for the MS have been successfully allocated in the target cell.

10. The MSC sends the HANDOVER COMMAND message to the BSC indicating that the MS should be instructed to move to UTRAN.

11. The BSC sends the INTER SYSTEM HANDOVER TO UTRAN COMMAND message to the MS commanding the MS to move to the new cell.

12. Once the MS arrives in UTRAN coverage, the MS synchronises with the Node B and establishes the RRC connection.

13. The target RNC informs the MSC, with the RELOCATION COMPLETE message, that the MS has successfully completed the handover to UTRAN procedures. Upon successful handover to UTRAN the MSC initiates release of resources in the old BSC.

14. The MS sends the HOLD message to the anchor 3G MSC to place the current voice call on hold.

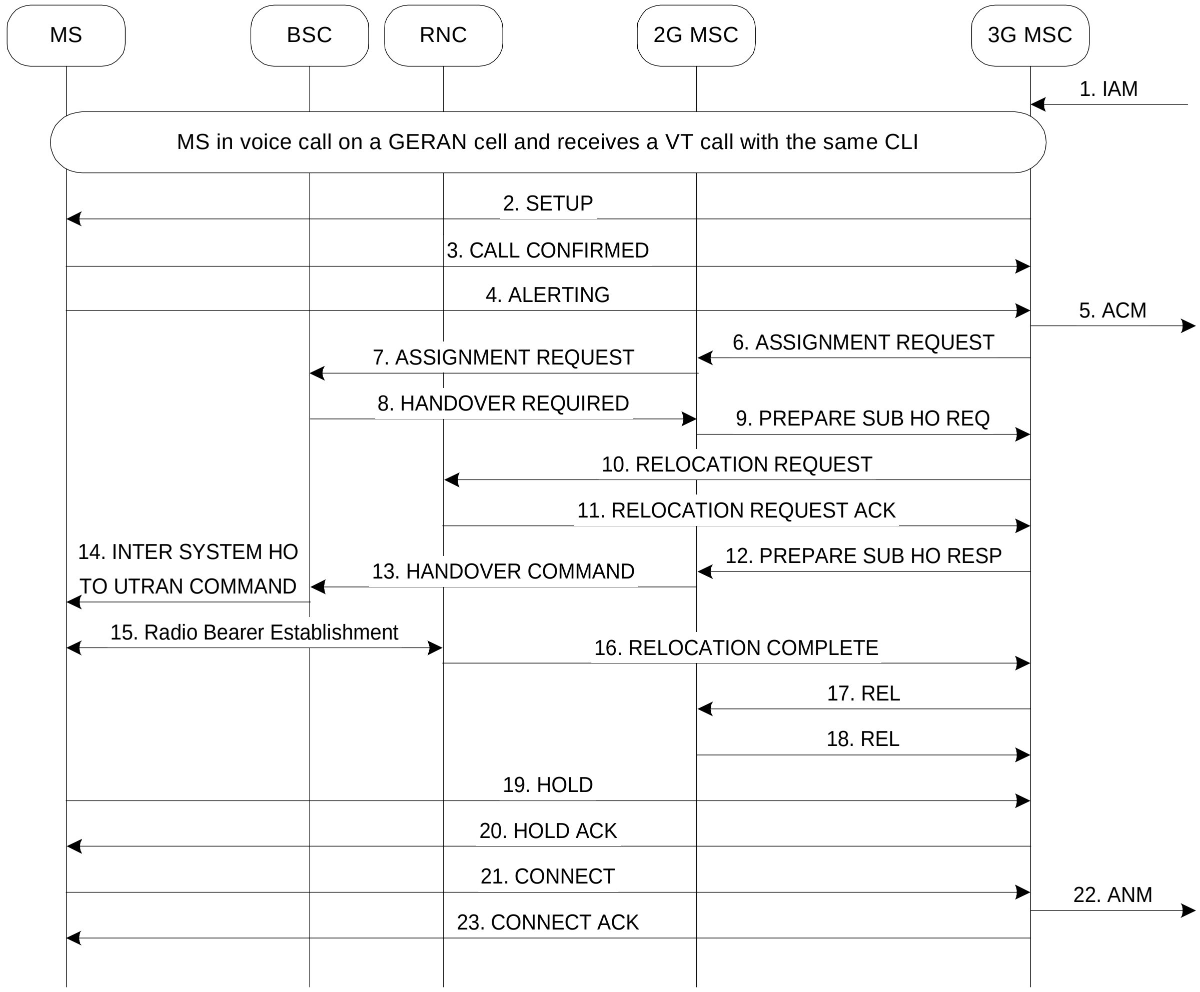
15. The anchor MSC responds with a HOLD ACK message indicating that the voice call has been placed on hold.

16. The CONNECT message is sent by the MS to indicate to the MSC that the user has accepted the video call.

17. The ANM message is sent by the MSC to indicate that the video call has been accepted.

18. The CONNECT ACK message is sent by the MSC to acknowledge to the MS the reception of the CONNECT message.

A.1.2.2.2 Connected Mode – MT video call after inter MSC handover from 3G



The MS is in voice call to user B and then the user B presses the "video" button in order to have a video call with this MS. The MSC controlling the MS has to inform the MS that there is an incoming video call and then move the MS to UTRAN coverage to allow the video call to be connected.

1. The MSC receives an IAM message from the A-party indicating that a Video call is required. The MSC verifies the subscription of the MS to check that it is provisioned and currently allowed to receive Video calls.

2. The SETUP message shall contain all the information required by the MS to process the call. Upon receipt of a SETUP message, the MS will perform compatibility checking on the contained information and shall allow the Video call to continue even though the MS is in GERAN coverage. As required by TS 48.083, the SETUP message contains the Signal IE and hence the MS immediately alerts the user.

3. The CALL CONFIRMED message is an acknowledgement to the SETUP message. The message can contain information about the MS capabilities.

4. The MS indicates to the MSC that the user is being alerted of the waiting video call by sending the ALERTING message.

5. The MSC sends the ACM message towards the A-party.

6. The MSC, after completing subscription verification for Bearer Service 37, sends to the 2G MSC an ASSIGNMENT REQUEST message with the same configuration as the previous HANDOVER REQUEST message except the message includes the Service Handover IE set to "Handover to UTRAN should be performed". Note that on the E interface between the MSCs, neither the ASSIGNMENT REQUEST nor HANDOVER REQUEST contain the Circuit Identity Code IE.

7. The 2G MSC sends an ASSIGNMENT REQUEST message to the BSC with the same configuration as message 4 except the relay MSC also includes the same Circuit Identity Code as is currently in use.

8. The BSC, after gathering the correct UTRAN neighbour cell measurements from the MS, passes a HANDOVER REQUIRED message to the relay MSC. The Cell ID Discriminator field in the CI List IE lets the relay MSC know the identity of the target RNC. The Source RNC to Target RNC transparent information container identifies to the target RNC the identity of the target cell.

9. Delete one of the options:

a) The RELOCATION REQUEST message is generated by the 2G MSC and passed to the Anchor MSC in the PREPARE SUBSEQUENT HANDOVER REQUEST message.

b) The HANDOVER REQUIRED message is passed by the 2G MSC to the Anchor MSC in the PREPARE SUBSEQUENT HANDOVER REQUEST message.

10. The anchor MSC indicates, in the RAB parameters of the RELOCATION REQUEST message, to the RNC that a 64k bearer is required.

11. The target RNC sends a RELOCATION REQUEST ACK message to the MSC informing the MSC that the resources for the MS have been successfully allocated in the target cell.

12. Delete one of the options:

a) HANDOVER COMMAND message is generated by the 2G MSC and passed to the Anchor MSC in the PREPARE SUBSEQUENT HANDOVER RESPONSE message.

b) The RELOCATION REQUEST ACK message is passed by the 2G MSC to the Anchor MSC in the PREPARE SUBSEQUENT HANDOVER RESPONSE message.

13. The relay MSC sends the HANDOVER COMMAND message to the BSC indicating that the MS should be instructed to move to UTRAN.

14. The BSC sends the INTER SYSTEM HANDOVER TO UTRAN COMMAND message to the MS commanding the MS to move to the new cell.

15. Once the MS arrives in UTRAN coverage, the MS synchronises with the Node B and establishes the RRC connection.

16. The target RNC informs the MSC, with the RELOCATION COMPLETE message, that the MS has been successfully completed the handover to UTRAN procedures. Upon successful handover to UTRAN the anchor MSC initiates release of resources in the relay MSC and the old BSC.

17. The REL message is sent from the Anchor MSC to the 2G MSC instructing the MSC to release the connection.

18. The REL message is sent from the 2G MSC to the Anchor MSC acknowledging the release of the connection.

19. The MS sends the HOLD message to the anchor 3G MSC to place the current voice call on hold.

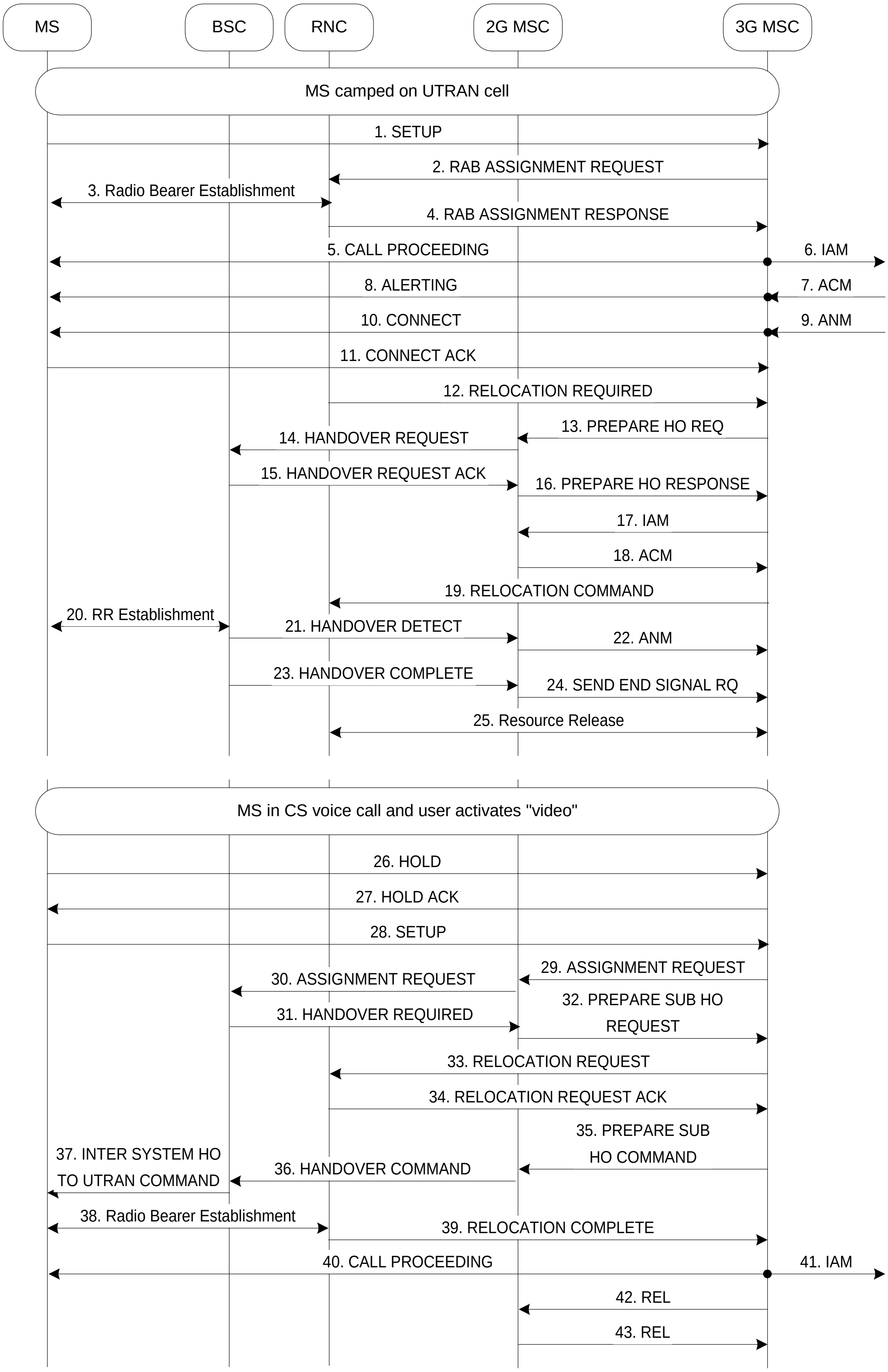
20. The anchor MSC responds with a HOLD ACK message indicating that the voice call has been placed on hold.

21. The CONNECT message is sent by the MS to indicate to the MSC that the user has accepted the video call.

22. The ANM message is sent by the MSC to indicate that the video call has been accepted.

23. The CONNECT ACK message is sent by the MSC to acknowledge to the MS the reception of the CONNECT message.

A.2 Service based handover in UTRAN



When an MS is camped on a UTRAN cell and establishes a speech call, the MS may be moved to GERAN coverage. Whilst the MS is in a voice call and the user presses the “video” button in order to have a video call with the person they are currently speaking to. The MMI in the phone is configured to then initiate a CALL HOLD and then send the SETUP message for the video call. To complete the establishment of the video call the MS will need to be moved back to UTRAN coverage.

1. The MS requests a speech call by passing the SETUP message to MSC.
2. The 3G MSC sends the RAB ASSIGNMENT REQUEST message to the RNC to request the establishment of a Radio Access Bearer between the MS and the MGW.
3. The RNC establishes the radio bearer between the RNC and the MS.
4. The RNC sends the RAB ASSIGNMENT RESPONSE message to the 3G MSC reporting the outcome of the assignment request.
5. The CALL PROCEEDING message is then transmitted to the MS from the MSC indicating that the MS has been allocated a bearer for speech.
6. The MSC sends the IAM message requesting the establishment of a speech call towards the B-party.
7. The MSC receives the ACM message from the B-party.
8. The MSC informs the MS that the B-party has alerted the user using the ALERTING message.
9. The ANM message is received by the MSC when the B-party user has answered.
10. The CONNECT message is sent to the MS by the MSC to indicate that the call has been through connected.
11. The MS acknowledges the reception of the CONNECT message with the CONNECT ACK message.
12. The RNC sends the RELOCATION REQUIRED message to the MSC initiating the handover of the MS to GERAN.
13. The HANDOVER REQUEST message is generated by the 3G MSC and passed to the 2G MSC in the PREPARE HANDOVER REQUEST message.
14. The HANDOVER REQUEST message is passed by the 2G MSC to the BSC requesting that resources are allocated for the MS.
15. The target BSC sends a HANDOVER REQUEST ACK message to the MSC informing the MSC that the resources for the MS have been successfully allocated in the target cell
16. The HANDOVER REQUEST ACK message is passed by the 2G MSC to the 3G MSC in the PREPARE HANDOVER RESPONSE message.
17. The 3G MSC sends an IAM message to the 2G MSC to request a call between the MSCs.
18. The ACM message is returned by the 2G MSC to the 3G MSC to acknowledge the IAM message.
19. The 3G MSC sends the RELOCATION COMMAND message to the RNC indicating that the handover can be completed and to provide the RNC with the new assignment information.
20. The RNC commands the MS to move to the GERAN resources.
21. Once the MS has accessed the new resources the BSC sends a HANDOVER DETECT message to the 2G MSC.
22. The 2G MSC sends the ANM message to the 3G MSC indicating the through connection of the call should go ahead.
23. The BSC sends the HANDOVER COMPLETE message to the 2G MSC to acknowledge the completion of the relocation.
24. The SEND END SIGNAL RQ message is a container that transports the HANDOVER COMPLETE message between the 2G MSC and the 3G MSC.

25. The resources allocated the MS in UTRAN can then be removed.

Whilst the MS is in a voice call and the user presses the “video” button in order to have a video call with the person they are currently speaking to. The MMI in the phone is configured to then initiate a CALL HOLD and then send the SETUP message for the video call.

26. The MS sends the HOLD message to the anchor 3G MSC to place the current voice call on hold.

27. The anchor MSC responds with a HOLD ACK message indicating that the voice call has been placed on hold.

28. The MS requests a video call by passing the SETUP message to the anchor MSC. The SETUP message includes one BCIE with Other Rate Adaption set to “H.233 & H.245” and the called number is the same as for the voice call.

29. The Anchor MSC, after completing subscription verification for Bearer Service 37, sends to the 2G MSC an ASSIGNMENT REQUEST message with the same configuration as the previous HANDOVER REQUEST message except the message includes the Service Handover IE set to “Handover to UTRAN should be performed”. Note that on the E interface between the MSCs, neither the ASSIGNMENT REQUEST nor HANDOVER REQUEST contain the Circuit Identity Code IE.

30. The 2G MSC sends an ASSIGNMENT REQUEST message to the BSC with the same configuration as message 4 except the relay MSC also includes the same Circuit Identity Code as is currently in use.

31. The BSC, after gathering the correct UTRAN neighbour cell measurements from the MS, passes a HANDOVER REQUIRED message to the relay MSC. The Cell ID Discriminator field in the CI List IE lets the relay MSC know the identity of the target RNC. The Source RNC to Target RNC transparent information container identifies to the target RNC the identity of the target cell.

32. Delete one of the options[cdp194]:

    a) The RELOCATION REQUEST message is generated by the 2G MSC and passed to the Anchor MSC in the PREPARE SUBSEQUENT HANDOVER REQUEST message.

    b) The HANDOVER REQUIRED message is passed by the 2G MSC to the Anchor MSC in the PREPARE SUBSEQUENT HANDOVER REQUEST message.

33. The anchor MSC indicates, in the RAB parameters of the RELOCATION REQUEST message, to the RNC that a 64k bearer is required.

34. The target RNC sends a RELOCATION REQUEST ACK message to the MSC informing the MSC that the resources for the MS have been successfully allocated in the target cell.

35. Delete one of the options:

    a) The HANDOVER COMMAND message is generated by the 2G MSC and passed to the Anchor MSC in the PREPARE SUBSEQUENT HANDOVER RESPONSE message.

    b) The RELOCATION REQUEST ACK message is passed by the 2G MSC to the Anchor MSC in the PREPARE SUBSEQUENT HANDOVER RESPONSE message.

36. The relay MSC sends the HANDOVER COMMAND message to the BSC indicating that the MS should be instructed to move to UTRAN.

37. The BSC sends the INTER SYSTEM HANDOVER TO UTRAN COMMAND message to the MS commanding the MS to move to the new cell.

38. Once the MS arrives in UTRAN coverage, the MS synchronises with the Node B and establishes the RRC connection.

39. The target RNC informs the MSC, with the RELOCATION COMPLETE message, that the MS has been successfully completed the handover to UTRAN procedures. Upon successful handover to UTRAN the anchor MSC initiates release of resources in the relay MSC and the old BSC.

40. After the MS arrives on the UTRAN cell, the 3G MSC indicates to the MS that the establishment of the call is progressing by sending the CALL PROCEEDING message.

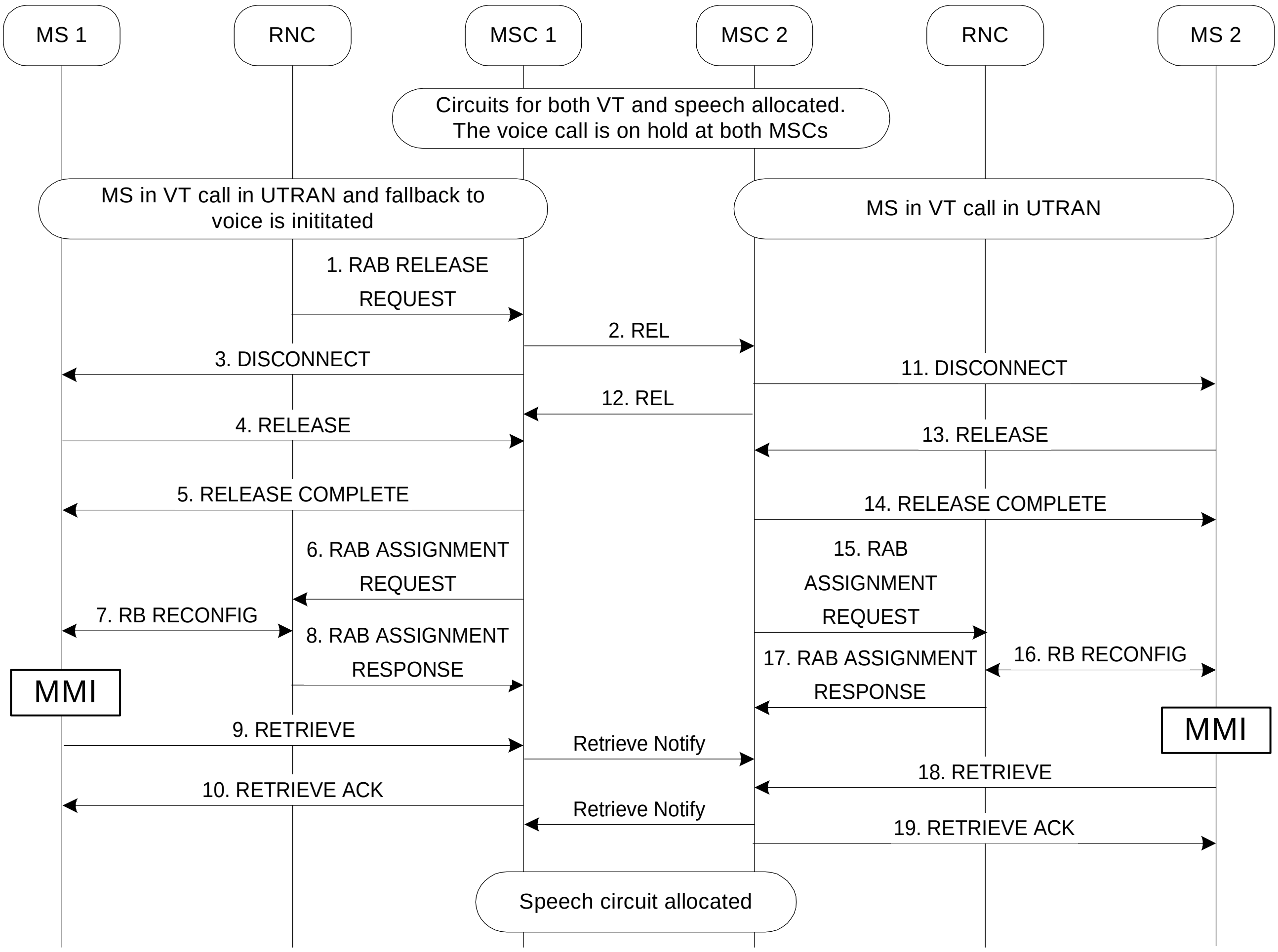
41. The MSC sends the IAM message towards the B-party.

42. The REL message is sent from the Anchor MSC to the 2G MSC instructing the MSC to release the connection.

43. The REL message is sent from the 2G MSC to the Anchor MSC acknowledging the release of the connection.

A.3 VT fallback in UTRAN

A.3.1 VT fallback to UTRAN voice: Fallback to voice during VT operation



1. The RNC detects that MS 1 has moved below the quality threshold set for the 64k VT bearer. The RNC then sends the RAB RELEASE REQUEST message to MSC 1 indicating that the bearer should be released. MSC 1 understands the reception of the RAB RELEASE REQUEST message to mean that MS 1 is moving out of coverage for the active service.

2. The MSC 1 sends the REL message to MSC 2 controlling the MS 2, indicating that the video call should be released.

3. A DISCONNECT message is sent from the MSC 1 to MS 1, instructing MS 1 that the video call should be released.

4. MS 1 responds with a RELEASE message for the video call.

5. MSC 1 sends the RELEASE COMPLETE message to MS 1 for the video call.

6. The RAB ASSIGNMENT REQUEST message is sent from the MSC 1 to the RNC, requesting the modification of the VT bearer to a bearer for a speech call.

7. The radio bearer is modified between the RNC and MS.

8. The RNC responds to the MSC 1 with a RAB ASSIGNMENT RESPONSE message indicating that the radio bearer is being modified.

It is possible that the handsets could be automated to retrieve a speech call in this situation, i.e. when both calls are from the same CLI and the VT call is terminated.

9. The RETRIEVE message is sent by MS 1 to MSC 1 to reconnect the voice call to MS 1, which had been on hold at MSC 1. A Retrieve Notify message is sent from MSC 1 to MSC 2 indicating that the MS 1 has taken the voice call off hold.

10. MSC 1 responds to MS 1 with a RETRIEVE ACK message once the call has been connected.

11. A DISCONNECT message is sent from MSC 2 to the MS 2, instructing MS 2 that the video call should be released.

12. MSC 2 sends the REL message to MSC 1, acknowledging that the video call is being released.

13. MS 2 responds with a RELEASE message for the video call

14. MSC 2 sends the RELEASE COMPLETE message to MS 2 for the video call.

15. The RAB ASSIGNMENT REQUEST message is sent from MSC 2 to the RNC, requesting the modification of the VT bearer to a bearer for a speech call.

16. The radio bearer is modified between the RNC and MS 2.

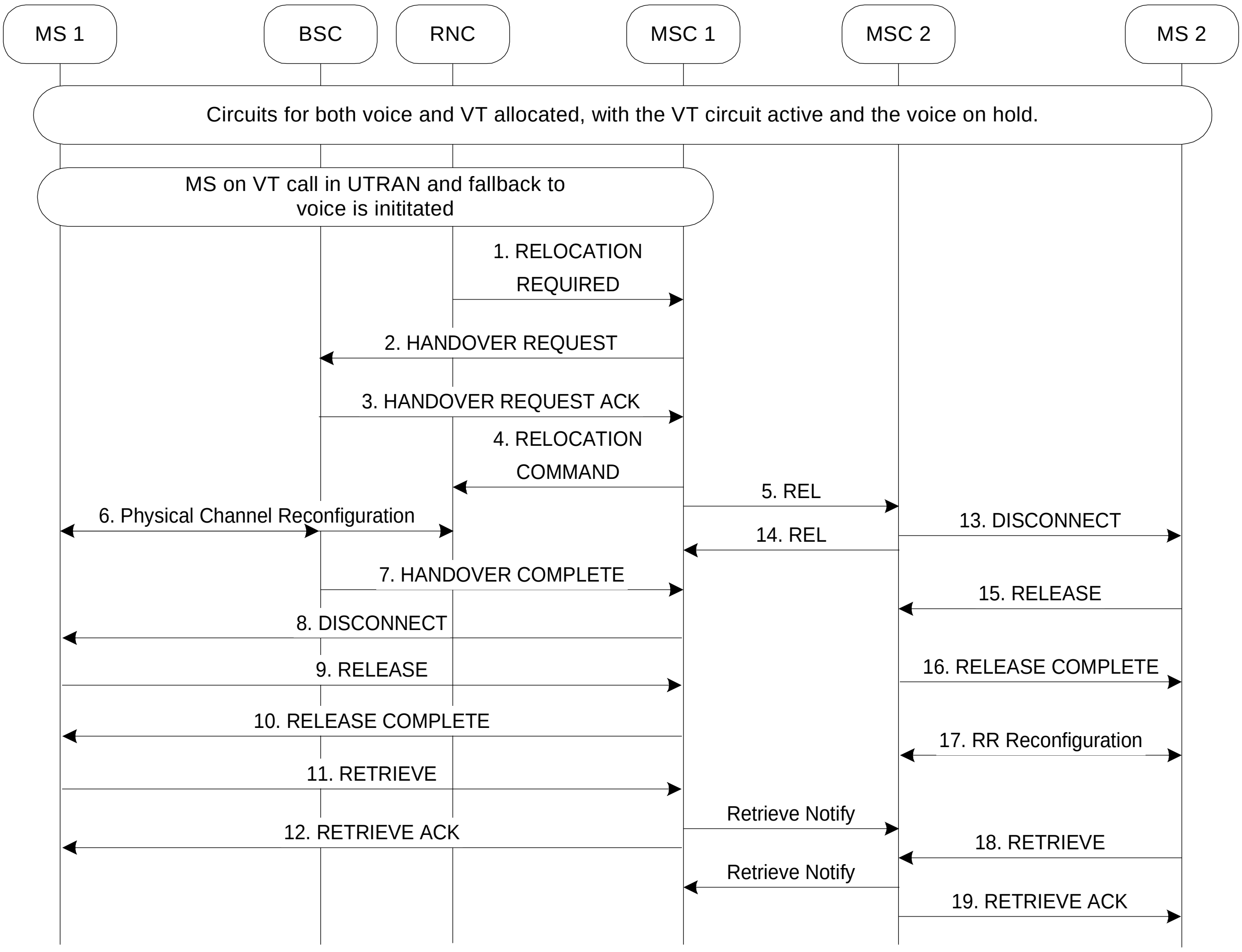
17. The RNC responds to MSC 2 with a RAB ASSIGNMENT RESPONSE message indicating that the radio bearer is being modified.

It is possible that the handsets could be automated to retrieve a speech call in this situation, i.e. when both calls are from the same CLI and the VT call is terminated.

18. The RETRIEVE message is sent by MS 2 to MSC 2 to reconnect the voice call to the MS, which had been on hold at the MSC. A Retrieve Notify message is sent from MSC 2 to MSC 1 indicating that the MS 2 has taken the voice call off hold.

19. MSC 2 responds with a RETRIEVE ACK message to MS 2 once the call has been connected.

A.3.2 VT fallback to GERAN voice: Fallback to voice during VT operation



1. When the RNC detects the quality for the 64k VT bearer has fallen below a certain threshold and the MS is in GERAN coverage, the RNC sends a RELOCATION REQUIRED message to MSC 1.

2. MSC 1 sends a HANDOVER REQUEST message to the BSC indicating that MSC 1 has a MS to be handed over to that BSC.

3. The BSC responds to MSC 1 with a HANDOVER REQUEST ACK message indicating that the request to handover an MS to this BSC can be supported by the BSC, and also to which radio channel the MS should be directed.

4. The RELOCATION COMMAND is sent by MSC 1 to the RNC to instruct the RNC to move the MS to the allocated resources on GERAN.

5. The MSC 1 sends the REL message to MSC 2 controlling the MS 2, indicating that the video call should be released

6. The MS is instructed to move to the GERAN and is assigned resources.

7. When the MS successfully arrives in GERAN the BSC informs MSC 1 that the handover has been successful by sending the HANDOVER COMPLETE message

8. A DISCONNECT message is sent from the MSC 1 to MS 1, instructing MS 1 that the video call should be released.

9. MS 1 responds with a RELEASE message for the video call.

10. MSC 1 sends the RELEASE COMPLETE message to MS 1 for the video call.

It is possible that the handsets could be automated to retrieve a speech call in this situation, i.e. when both calls are from the same CLI and the VT call is terminated.

11. The RETRIEVE message is sent by MS 1 to MSC 1 to reconnect the voice call to MS 1, which had been on hold at MSC 1. A Retrieve Notify message is sent from MSC 1 to MSC 2 indicating that the MS 1 has taken the voice call off hold.

12. MSC 1 responds to MS 1 with a RETRIEVE ACK message once the call has been connected.

13. A DISCONNECT message is sent from MSC 2 to the MS 2, instructing MS 2 that the video call should be released.

14. MSC 2 sends the REL message to MSC 1, acknowledging that the video call is being released.

15. MS 2 responds with a RELEASE message for the video call

16. MSC 2 sends the RELEASE COMPLETE message to MS 2 for the video call.

17. RR Reconfiguration

It is possible that the handsets could be automated to retrieve a speech call in this situation, i.e. when both calls are from the same CLI and the VT call is terminated.

18. The RETRIEVE message is sent by MS 2 to MSC 2 to reconnect the voice call to the MS, which had been on hold at the MSC. A Retrieve Notify message is sent from MSC 2 to MSC 1 indicating that the MS 2 has taken the voice call off hold.

19 MSC 2 responds with a RETRIEVE

Annex B: Change History

Change history							
Date	**TSG #**	**TSG Doc.**	**CR**	**Rev**	**Subject/Comment**	**Old**	**New**
16/5/04	SA2 #40				Text incorporated from S2-041326; S2-042027; S2-042022 and S2-042029.	0.0.1	0.1.0
20/5/04	SA2 #40				Text incorporated from S2-042037, S2-042038, S2-042040 and the first sentence in section 5.1.3 of S2-042031.	0.1.0	0.2.0
21/5/04	SA2 #40				Text incorporated from S2-042044.	0.2.0	0.3.0
07/06/04	SA#24	SP-040336			Presented for information	0.3.0	1.0.0