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Introduction

Many 3GPP operators regard circuit switched 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.

Following a study of alternative mechanisms, this Technical Report describes the idle mode redial mechanism which 3GPP based systems can use to enhance switching between voice and video services.

1 Scope

The present document describes the idle mode redial mechanism which 3GPP based systems can use to enhance switching between circuit switched voice and video services.

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.
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- [1] 3GPP TR 21.905: "Vocabulary for 3GPP Specifications".
 - [2] 3GPP TS 44.018: "Mobile radio interface layer 3 specification, Radio Resource Control Protocol".
 - [3] 3GPP TS 48.008: "Mobile-services Switching Centre – Base Station System (MSC – BSS) interface; layer 3 specification".
 - [4] 3GPP TS 25.331: "Radio Resource Control (RRC) protocol specification".
 - [5] 3GPP TS 22.004: "General on supplementary services".
 - [6] 3GPP TS 45.008: "Radio subsystem link control".
 - [7] 3GPP TS 24.008: "Mobile radio interface Layer 3 specification; Core network protocols; Stage 3".
 - [8] 3GPP TS 23.009: "Handover procedures".
 - [9] 3GPP TS 32.205: "Telecommunication management; Charging management; Charging data description for the Circuit Switched (CS) domain".
 - [10] 3GPP TS 22.082: "Call Forwarding (CF) supplementary services - Stage 1".
 - [11] 3GPP TS 23.082: "Call Forwarding (CF) supplementary services - Stage 2".
 - [12] 3GPP TS 23.205: "Bearer-independent CS Core Network".
-

3 Definitions, symbols and abbreviations

3.1 Definitions

For the purposes of the present document, the following terms and definitions given in TR 21.905 [1] and the following apply.

A party: the calling party.

B party: the called party.

4 Description of idle mode redial switching between voice and video

4.1 General Description

This "Idle-Mode Redial" mechanism is a combination of existing Voice call and Video call standards and services. It is a "terminal-centric" solution that requires only minimal additional support from the UTRAN, GERAN and Core Network. There are 3 main components to the solution:

- a) Switching between voice and video during an established call is achieved by one UE releasing the voice (or video) call and then that UE establishing a video (or voice) call with the same destination UE.
- b) During a video call, the radio coverage at one end may degrade such that the video call cannot be maintained. In this case the video call will be released, and, the UE that initiated the video call can offer to its user an attempt to establish a voice call with the other party.
- c) The initial establishment of a video call may be unsuccessful, in which case the A party can automatically establish a voice call.

For all these 3 cases, optimisation of the Man Machine Interface in the UE is possible and desirable.

NOTE: There may be tradeoffs between 'speed of redial' and interactions with content servers that only support video calls. In particular, the use of the term "automatic" does not prevent the UE requesting confirmation from the user.

It is not the intention of this TR to specify the MMI of UEs, however, this TR does describe some MMI actions solely for the purpose of easing the description of the procedures.

4.2 User initiated switching during an established call

4.2.1 Signalling flows and procedures

The successful case of the "B party initiated switch to video" for UMTS is illustrated in Figure 1. The notation "UE 1" and "UE 2" is used (rather than A and B party) because, if the B party (of the voice call) presses the "switch to video" button, it then becomes the A party of the subsequent video call.

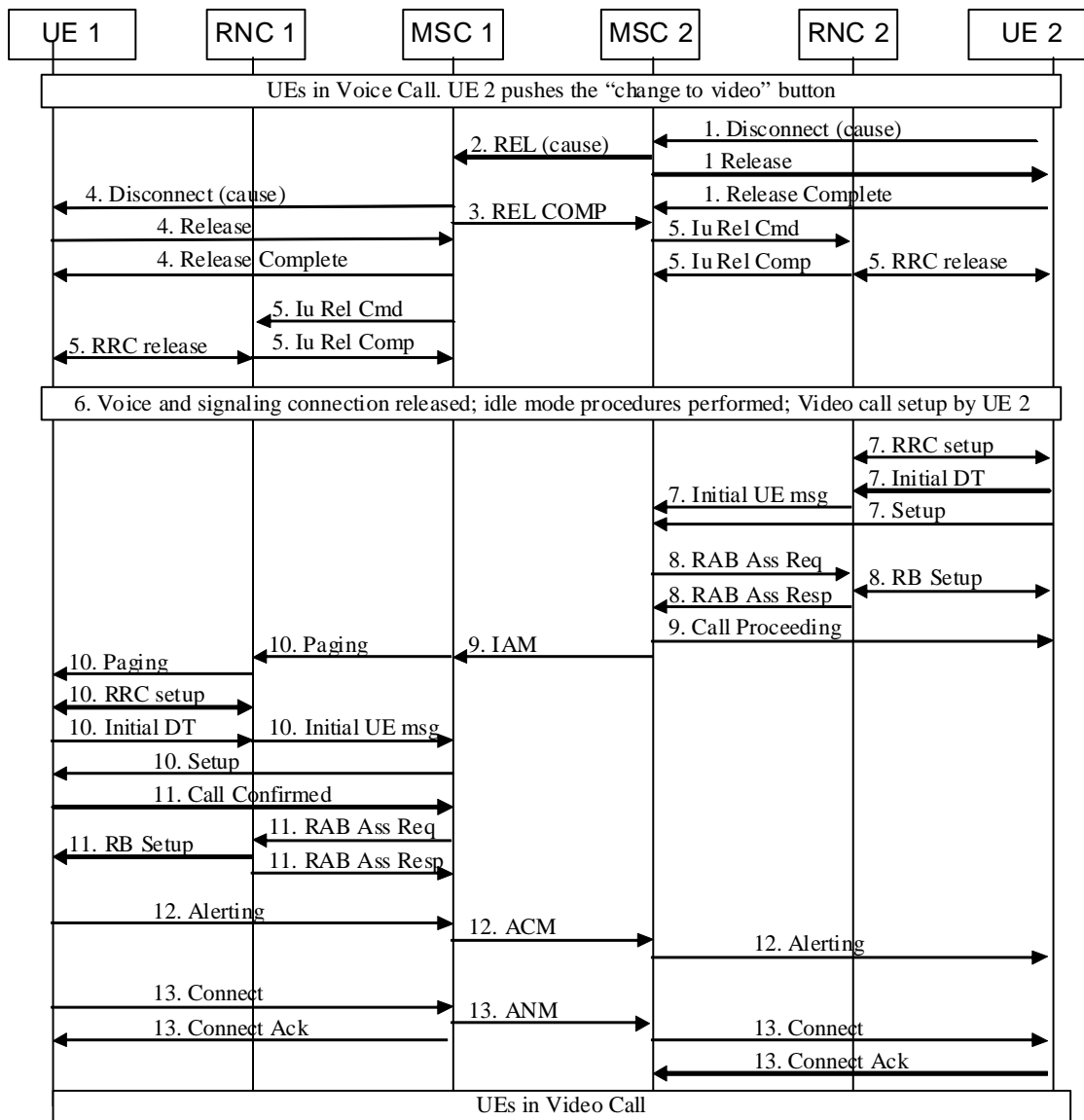


Figure 1: B party initiated switch to video

0) The UEs are in a voice call initiated by UE 1. Following verbal dialogue between User 1 and User 2, they agree that a switch to video is likely to be successful (e.g. because both User 1 and User 2 have video capable handsets, and, they both have "using 3G" indications visible on their screens). The User of UE 2 pushes the "switch to video" button (or uses other MMI that provides the same functionality) and UE 2 stores the CLI of UE 1 for use in step 7.

NOTE 1: If UE 1 (instead of UE 2) initiated the "switch to video", then UE 1 stores the previously dialed number for use in step 7.

- 1) UE 2 releases the voice call.
- 2) MSC 2 sends a RELEASE message with the release cause. The cause value in the RELEASE message might be changed by transit networks prior to arrival at MSC 1.
- 3) MSC 1 sends a RELEASE COMPLETE message to MSC 2.
- 4) MSC 1 releases the voice call with UE 1.
- 5) MSC 2 and MSC 1 locally release the Iu connections to RNC 2 and RNC 1 (assuming that there are no other CM connections active). RNC 2 and RNC 1 then release the RRC connections (assuming that their mobiles are not in PMM connected state).

In order to minimise idle tasks performed by the UE, the following should be performed by Operators which preferentially camp UEs on UMTS:

the BSC should include the "Cell selection indicator after release of all TCH and SDCCH" IE in the Channel Release message (see TS 44.018 [2]) if the release is done on a GERAN cell and the call was not started on this BSC and the local 2G and 3G cells are in different RAs.

NOTE 2: The above text does mean RA. Change of LA always involves change of RA.

- 6) UE 2 and UE 1 camp on suitable cells and perform the required idle mode tasks.
- 7) UE 2 initiates RRC connection setup and sends the Setup message for the video call to MSC 2. The Called Party number is the one stored in step 0. The Setup message should carry extra information to indicate that it is the result of a redial attempt.
- 8) MSC 2 requests the allocation of radio resources.

If the GERAN does not support CS video, then the MSC 2 shall send an A interface Assignment Request message to the BSC including the Service Handover IE set to "Handover to UTRAN or cdma2000 should be performed" and a Channel Type indicating any value acceptable to the BSC. The BSS handles the Assignment Request message as defined in TS 48.008 [3]. The interactions between MSC server and MGW can be minimised if the A interface circuit is not allocated. This can be achieved by setting the Channel Type = signalling. The interactions between the MSC server and the MGW are defined in TS 23.205 [12].

If the GERAN does support CS video then the MSC 2 should send an A interface Assignment Request message to the BSC including the relevant channel type. The BSS proceeds as defined in TS 48.008 [3].

- 9) If radio resource allocation was successful, the IAM message is sent to MSC 1 and the Call Proceeding message is sent to UE 2.
- 10) MSC 1 pages UE 1. UE 1 responds by establishing an RRC connection and sending the Paging Response message to MSC 1 within the RRC-Initial Direct Transfer and RANAP-Initial UE messages. MSC 1 then sends the Setup message for the video call to UE 1. The Setup message carries the CLI of UE 2.
- 11) UE 1 confirms its capability to handle the video call in the Call Confirmed message and MSC 1 requests the allocation of radio resources.

If UE 1 responded to paging by establishing an RR connection on a GERAN cell, then MSC 1 initiates handover to UTRAN as described in step 8.
- 12) If the radio resource allocation was successful, UE 1 sends an Alerting message to MSC 1. MSC 1 sends the ACM message to MSC 2 and MSC 2 sends the Alerting message to UE 2.
- 13) User 1 answers the video call. For privacy reasons, the "switch to video" and "switch to voice" answer functions should not be automated.
- 14) The video call is established.

With obvious small differences, some of which are described above, the above procedure is also used for switching from a video call to a voice call.

NOTE 3: When "redial-switching" from a video call to a voice call, the voice call Setup message should carry the extra information to indicate that it is a redial attempt.

4.2.2 Future enhancements

4.2.2.1 Detection of video calling capability during a voice call

As a minimum, it will be useful if mobiles can indicate whether they are using UMTS for their ongoing voice call.

However additional functionality would be useful, specifically:

- when using UTRAN, an indication of whether 64 kbit/s video can be supported in the mobile's current location;
- when using GERAN, an indication of whether UMTS coverage is available, and, ideally whether the UMTS coverage can support 64 kbit/s video.

4.3 Fallback to voice during an established video call

4.3.1 Signalling flows and procedures

4.3.1.1 Fallback from UTRAN Video to UTRAN Voice call

The successful case of the "radio degradation at UE 2 leading to fallback to voice" for UMTS is illustrated in Figure 2.

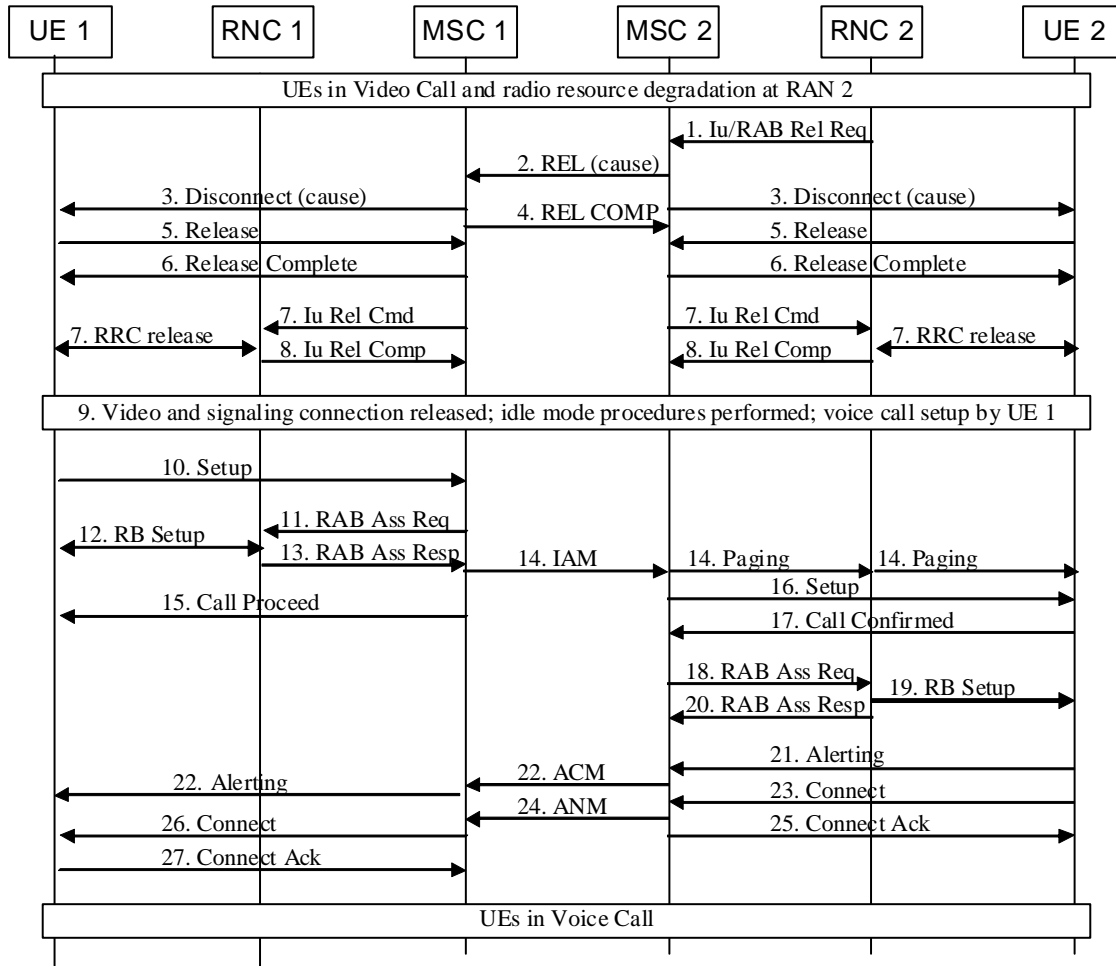


Figure 2: Radio degradation at UE 2 leading to fallback to voice

NOTE 1: Figure 2 does not show all of the RRC and Iu connection establishment messages.

1. RNC 2 is configured such that it knows that the local GERAN does not (or does) support video calls (i.e. does not (or does) support 64 kbit/s conversational QoS on the CS domain).

NOTE 2: For an interim period, an alternative solution is that MSC 2 sets the Service Handover IE to 'Handover to GSM shall not be performed' in the RAB Assignment Request message. However, this solution is not future proof if (all or part of) the network's GERAN is upgraded to support CS Video telephony.

When RNC 2 detects that the 64kbit/s bearer cannot be maintained any longer (e.g. radio link failure, TS 25.331 [4]), RNC 2 sends either an Iu RELEASE REQUEST message or a RAB Release Request message to MSC 2, indicating that the Iu connection should be released

2. MSC 2 sends a REL message with the release cause. This cause value in the REL message might be changed by transit networks prior to arrival at MSC 1.
3. The MSCs send Disconnect messages to the UEs.

In case RNC 2 sent Iu Release Request in step 1 but not a RAB Release Request, MSC 2 will not perform step 3. (i.e. not send Disconnect, which causes that also steps 5 and 6 are not performed), but MSC 2 will immediately proceed with step 7 (i.e. send Iu Release Command).

4. MSC 1 sends a Release Complete message to MSC 2.
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 and the RNCs release the RRC connections (unless their UE is in PMM connected state).
8. The RNCs confirm the release of all resources.
9. When the signalling connections with the UEs are released, UE 1 and UE 2 camp on suitable cells and perform the required idle mode tasks.
10. UE 1 prompts its user to attempt a Redial. When UE 1 gets the Redial confirmation from its user it sends a SETUP message (containing the Redial indication) to MSC 1 to setup a Voice call.
11. A RAB Assignment Request message is sent from the MSC 1 to the RNC 1, requesting the setup of a RAB for a Voice call.
12. The radio bearer is established between RNC 1 and UE 1.
13. RNC 1 responds to MSC 1 with an RAB Assignment Response message.
14. MSC 1 sends an IAM message to MSC 2 to establish a Voice Call with UE 2. MSC 2 pages RNC 2 and RNC 2 pages UE 2.
15. MSC 1 sends a Call Proceeding message to UE 1.
16. The MSC 2 sends a Setup message to UE 2 indicating the establishment of a Voice Call.
17. UE 2 sends Call Confirmed to MSC 2.
18. The RAB Assignment Request message is sent from MSC 2 to the RNC 2, requesting the establishment of a RAB for a Voice Call.
19. The radio bearer is established between the RNC 2 and UE 2.
20. RNC 2 responds to MSC 2 with a RAB Assignment Response message.
21. UE 2 sends Alerting message to MSB 2.
22. MSC 2 sends ACM message to MSC 1. MSC 1 sends Alerting to UE 1.
23. User 2 accepts the Voice Call and UE 2 sends Connect message to MSC 2.
24. MSC 2 sends ANM message to MSC 1.
25. MSC 2 sends Connect Ack message to UE 2.
26. MSC 1 sends Connect message to UE 1.
27. UE 1 acknowledges with a Connect Ack message to MSC 1 and the Voice call is established.

4.3.1.2 Fallback from UTRAN Video to GERAN Voice call

The procedure in 4.3.1.1 can also be used if UE 2 drops out of UTRAN coverage into GERAN coverage. However there are additional complications:

- a) If the UTRAN and GERAN cells are attached to different MSCs then problems are likely to occur due to the time required for the UE 2 to perform Location Area and Routeing Area updating.

In particular, if UE 2 is updating with a new MSC and/or new SGSN, UE 2 will not respond to MSC 2's paging. This will only become known to UE 1 when MSC 2's "no response to paging" timer expires. Typically, an

operator configures this timer with values in the range of 8 to 25 seconds. When this timer expires, it is quite likely that UE 2 will have "call forward on not reachable" set and so UE 1's call will be diverted to a voice mail server. To avoid these problems, it seems important that the operator configures the UTRAN to minimise the number of video calls that lead to fallback to GERAN coverage and/or to configure the network such that geographically similar UTRAN and GERAN cells are in the same Routeing Area.

- b) In order to avoid useless handover attempts, the RNCs are frequently configured so that no GERAN neighbour cells are given to the mobile during a video call. This will slow down re-selection from 3G to 2G in the case of total loss of 3G coverage. Conversely, in many other scenarios the lack of 2G neighbour cells might force UE 2 to camp on the 3G cell for long enough for it to receive the paging for the voice call.

Overall, the operator may need to adapt the neighbour cell lists provided in video calls on cell by cell basis and/or dependent upon whether or not combined 2G/3G MSCs AND combined SGSNs are in use.

4.3.2 Future enhancements

4.3.2.1 RRC behaviour following degradation of the video RAB

Additional study may be beneficial on how to 'synchronise' the release of the RRC connection in the RNC and in the mobile when the radio connection is degraded or lost. If synchronisation cannot be guaranteed then timers may be needed to delay the redial attempt by the A party.

4.3.2.2 Interaction with voice mail servers

If a redial attempt is unsuccessful, it may (or may not) be appropriate to avoid being forwarded to a voice mail server.

One potential solution to this is that the A party signals (e.g. by appending a suffix to the dialled digits) to MSC-A and MSC-A then uses the Call Diversion Not Allowed setting in the Call Diversion Treatment Indicators to inhibit call forwarding. One problem with the use of a suffix in the dialled digits is that the digit analysis in MSC-A usually only examines the leading digits. Conversely, if the UE adds a prefix to the dialled digits, then the call will not be handled properly if MSC-A does not support this functionality.

4.4 Fallback to voice during video call establishment

4.4.1 Signalling flows and procedures

The following diagram shows the various cases where video call establishment can fail or be unsuccessful. At many points, alternative signalling message flows are shown by the "a" and "b" arrows.

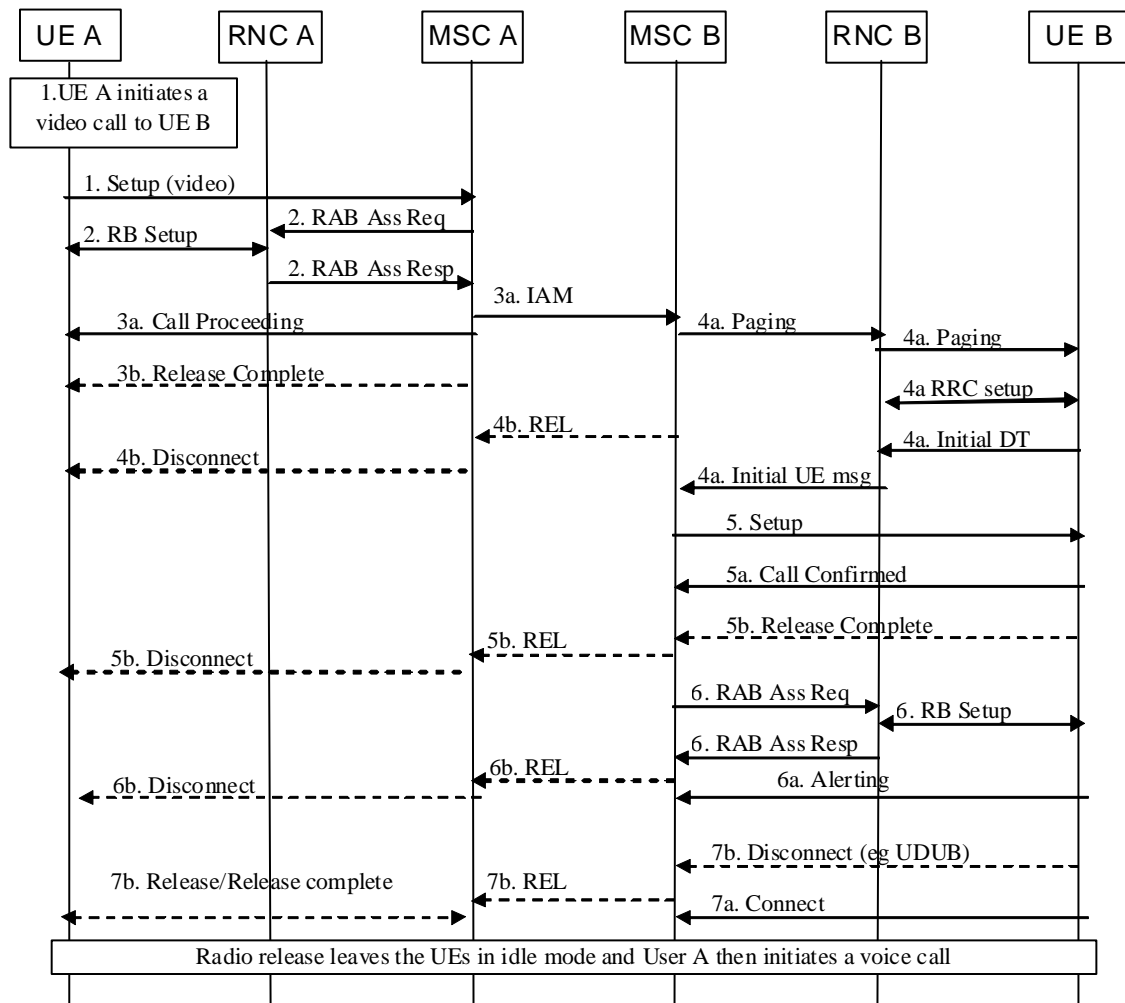


Figure 3: Video call establishment failure scenarios

1. UE A sends a SETUP message to MSC A to setup a Video call.
2. A RAB Assignment Request message is sent from the MSC A to the RNC A, requesting the setup of a RAB for a Video call. The radio bearer is established between RNC A and UE A.

If the UE is using a GERAN that does not support CS video calls, the MSC sends an Assignment Request with an encoding that prompts the BSC to initiate a handover to UTRAN.
- 3a. (Successful case) MSC A sends an IAM message to MSC B to establish a Video Call with UE B. And, MSC A sends a Call Proceeding message to UE A.
- 3b. (Failure case) The video call fails because of lack of radio resources on the A side. MSC A sends a Release Complete message (e.g. with cause #47, Resources unavailable, unspecified) to UE A. The RRC connection is released. UE A can then prompt the user to initiate a voice call to UE B.
- 4a. (Successful case) MSC B pages UE B. UE B responds to paging.
- 4b. (Failure case) The video call fails because MSC B or the transit network do not support video calls. MSC B sends a REL message to MSC A. MSC A sends a Disconnect message (e.g. with cause #79, Service or option not implemented, unspecified) to UE A. The RRC connection is released. UE A can then prompt the user to initiate a voice call to UE B.
5. MSC B sends a Setup message to UE B indicating the establishment of a Video Call.
- 5a. (Successful case) UE B sends Call Confirmed to MSC B.
- 5b. (Failure case) The video call fails because UE B does not support video calls. UE B sends a Release Complete message with cause #88 (Incompatible destination) to MSC B. MSC B sends a REL message to MSC A. MSC A

sends a Disconnect message (e.g. with cause #88, Incompatible destination) to UE A. The RRC connections are released. UE A can then prompt the user to initiate a voice call to UE B.

6a. (Successful case) The RAB Assignment Request message is sent from MSC B to the RNC B, requesting the establishment of a RAB for a Video Call. The radio bearer is established between the RNC B and UE B. RNC B responds to MSC B with a RAB Assignment Response message. Following the allocation of the radio resources, UE B sends an Alerting message to MSC B.

6b (Failure case) The video call fails because of lack of radio resources on the B side. MSC B sends a Release Complete message to UE B. MSC B sends a REL message to MSC A. MSC A sends a Disconnect message to UE A (e.g. with cause #47, Resources unavailable, unspecified). The RRC connections are released. UE A can then prompt the user to initiate a voice call to UE B.

7a (Successful case) The user of UE B accepts the call and UE B sends a Connect message to MSC B.

7b (Failure case) UE B is alerting the user of UE B. The user of UE B may choose not to accept the video call and either Reject the call or indicate User Determined User Busy. UE B then sends a Disconnect message (with cause #21, Call rejected or cause #17, User busy) to MSC B. MSC B sends a REL message to MSC A. MSC A sends a Disconnect message (with cause #21, Call rejected or cause #17, User busy) to UE A. The RRC connections are released. UE A can then prompt the user to initiate a voice call to UE B.

Many other cause values can be returned to UE A in the Disconnect/Release Complete messages. It is a matter of UE implementation as to whether or not these cause values are used to prompt the user to initiate a re-dial.

NOTE: There is no cause value that the user of UE B can use to decline the video call but indicate that they might accept a voice call.

If (following a failed video call establishment attempt) the UE prompts the user to initiate a Redial, and, the user requests the Redial attempt, then, the resulting SETUP message should carry extra information to indicate that it is a redial attempt.

4.4.2 Future enhancements

4.4.2.1 How to detect that a voice call might be successful?

There are many reasons (User Determined User Busy, Barring of all incoming calls, an existing active video call) why a video call and a re-dialled voice call might both be unsuccessful. It would be useful if the rejection for the video call carried information that helped UE A to determine whether or not a subsequent voice call might succeed.

NOTE: For UDUB, the A party will have received the Alerting message prior to the release message.

4.4.2.2 How to avoid video mail box problems?

Can users of video phones be provided with a full suite of supplementary services, e.g. can automatic forwarding to a video mail box when not in UMTS coverage be performed? Analysis of the current Call Forwarding specifications (TS 22.082 [10], TS 23.082 [11]) shows that MSC B should reject the call rather than forward it (because the UE is reachable but is not busy and has not been offered the chance to reply).

Once a video call has been stored by the mailbox, it can only be retrieved from the mailbox when the UE is in coverage that allows for a 64kbit/s bearer. As long as the UE is not in such a coverage area it cannot receive the video call from the mailbox.

One potential solution to this is that the mailbox provides a voice call comprising only the audio part of the video call. An alternative is that the user deactivates all call forwarding for video calls (c.f. CS data calls).

4.5 Automatic upgrade to video following fallback to voice

This appears difficult to achieve in an automatic manner. However, given the privacy issues associated with automatic switching from voice to video, this does not appear to be a serious deficiency.

For manual switching, good indications of UTRAN coverage, and, its ability (or not) to handle 64 kbit/s video would be useful. This is the same requirement as described in clause 4.2.2.1.

5 Interaction with Supplementary Services and other 3GPP Features

5.1 General

Within the 3GPP specifications the CS video call is regarded as a Bearer Service and not a Teleservice.

Annex A of TS 22.004 [5] defines the applicability of supplementary services to both "Telephony" (a Teleservice) and "cct data" (Bearer Services).

Most supplementary services are applicable to both the voice and video calls. For this 'redial' solution to function effectively, the user and operator needs to configure supplementary services and network features (such as ODB and Roaming Restrictions) in a sensible manner (e.g. avoid barring voice calls but permit video calls). Sensible configurations do not seem to be prohibited by the standard.

It is worth noting that some supplementary services cannot be used while in the video component of a call (e.g. Call Hold; Multiparty; Explicit Call Transfer and Calling Name Presentation).

5.2 Calling Line Identity and Single Numbering Scheme

To make this feature work smoothly, CLIs should not be restricted by the user, and, CLIs should be transferred by the networks. In addition, single numbering schemes should be used (i.e. the user's videophone and telephone numbers need to be the same).

When detecting that a redialled call is the same as the original, the terminating UE should be aware that the number format might change from, for example, national to international format. This could be because the networks route the voice and video calls differently, or, because the terminating UE was the original A party.

The impacts of CAMEL based number translation services, e.g. for office VPNs are for further study.

5.3 CAMEL based services triggered by call release and CCBS

Release of the call can trigger other services, for example, voice mail delivery attempts and/or Call Completion to a Busy Subscriber. These services should have been designed to permit customers to make 'one call followed by another', however, increased usage of closely coupled calls might mean that these other services need re-tuning.

5.4 Provision of Multi-Media "Ring Back" tones to the A party

When initiating a video call, the terminating network might provide "multi-media" content to the A party while attempting to contact the B party.

In order to achieve this, a bi-directional user plane is needed between the A party and the terminating network, i.e., the Connect message must have been sent. In this situation, the original video call establishment has been successful, and, as such, there is no need to "redial as a voice call".

How the terminating network transfers the established video call to the "real B party" is a matter for the terminating network (e.g. with regard to H.324 codec negotiation matters).

6 O+M, charging and inter-operator and roaming accounting

6.1 Call Detail Records and statistics

With this redial mechanism, the voice and video calls are totally separate. Hence each should be able to use the existing CDR and billing machinery without modification.

However, it is worth noting that the billing system will probably not be able to provide any correlation between video and voice components of the same call.

The easy ability to switch between voice and video calls may have an impact on, for example, calculations of 'mean call holding time' if this is calculated from CDRs. As a consequence there may be a need to add extra information onto the CDRs to indicate the reason for release/reason for establishment of a call.

In particular, it would be valuable to know the proportion of redial attempts that were successful.

The addition of "reason for establishment" requires extra TS 24.008 [7] signalling from the mobile. It is recommended that such changes to the TS 24.008 [7] signalling should be made in this release of the specification.

An MSC generates call statistics information from its locally available release information. Some of the abnormal releases on the B party side may be seen only as normal releases on the A party side as some problems of B party side are not signalled between MSCs. Therefore, the probability of the different B party problems has to be taken into account when the statistics for A party side is evaluated.

An MSC may collect call statistics information from video calls for:

- unsuccessful mobile originating video call setup because 64 kbit/s RAB is currently not available
- unsuccessful mobile terminating video call setup because 64 kbit/s RAB is currently not available
- unsuccessful mobile originating video call setup because of lack of video service support by the UE
- unsuccessful mobile terminating video call setup because of lack of video service support by the
- unsuccessful mobile originating video call setup because UE or network do not support video call
- disrupted video calls because of loss of the radio connection

and, if TS 24.008 [7] is upgraded, the MSC can collect statistics for:

- the number of mobile originating video call attempts indicating redial
- the number of mobile originating voice call attempts indicating redial.

Additionally, it is recommended that a "redial flag" is an optional parameter in the CDR produced for Mobile Originated calls as specified in TS 32.205 [9].

6.2 Online charging

With this redial mechanism, the voice and video calls are totally separate. Hence each should be able to use the existing pre-pay mechanisms without modification.

6.3 Tariffing

With this redial mechanism, each call will be subject to its own tariffs. If 'per second' charging is in operation this should not matter. However if "per minute" billing is used, a customer might be slightly disappointed with the charge raised following 20 seconds of voice communication and 40 seconds of video communication.

6.4 Inter-operator accounting

With this redial mechanism, the voice and video calls are totally separate. Hence there should be no impact on existing inter-operator agreements for either voice or for video calls.

6.5 Roaming accounting

With this redial mechanism, the voice and video calls are totally separate. Hence there should be no impact on existing roaming agreements for either voice or for video calls.

Annex A: List of CRs needed to other TSs

A.1 General

The functionality that is described in this TR is based on the R'99 core specifications, plus, some additional CRs already agreed in later releases, plus, some CRs that have yet to be drafted and/or agreed by TSG Plenaries.

A.2 CRs to other specifications already agreed by TSG Plenaries

These CRs are believed to include (and believed to be limited to):

- a) From release 6, CR 287r3 to TS 44.018 [2] (see GP-040518) and CR 198r4 to TS 45.008 [6] (see GP-040542). These are the CRs that permit the BSC to "release the mobile and ask the mobile to camp on 3G" (re: clause 4.2.1, step 5).
- b) From release 6, CR 133 to TS 48.008 [3] (see GP-041421). This CR relates to "Service handover for services not supported in GERAN" (re: to clause 4.2.1, step 8 and clause 4.4.1, step 2).

A.3 Future CRs that are needed and/or are desirable

The drafting of the following CRs is believed to be useful:

- a) To TS 24.008 [7], a CR for the inclusion of "redial attempt" information in the Setup messages sent by a UE (re: clause 4.2.1 step 7 and clause 6.1).
- b) To TS 23.009 [8] and TS 48.008 [3], to provide additional clarification on service based handover from GERAN (re: clause 4.2.1, step 8 and clause 4.4.1, step 2).
- c) To TS 32.205 [9] to add a "redial flag" to the MSC CDR for Mobile Originated calls (re: clause 6.1).

Annex B: Change history

Change history							
Date	TSG #	TSG Doc.	CR	Rev	Subject/Comment	Old	New
2004-12	SP-26	SP-040822			Creation of version 6.0.0	2.0.0	6.0.0
2005-03	SP-27	SP-050	001		Incorrect sequence of events in 4.3.1.1	6.0.0	6.1.0
2007-06	-	-	-	-	Update to Rel-7 version (MCC)	6.1.0	7.0.0
2008-12	SP-42	-	-	-	Update to Rel-8 version (MCC)	7.0.0	8.0.0
2009-12	SP-46	-	-	-	Update to Rel-9 version (MCC)	8.0.0	9.0.0
2011-03	SP-51	-	-	-	Update to Rel-10 version (MCC)	9.0.0	10.0.0
2012-09	-	-	-	-	Update to Rel-11 version (MCC)	10.0.0	11.0.0
2014-09	SP-65	-	-	-	Update to Rel-12 version (MCC)	11.0.0	12.0.0
2015-12	-	-	-	-	Update to Rel-13 version (MCC)	12.0.0	13.0.0
2017-03	-	-	-	-	Update to Rel-14 version (MCC)	13.0.0	14.0.0
2018-06	SP-80	-	-	-	Update to Rel-15 version (MCC)	14.0.0	15.0.0

History

Document history		
V15.0.0	June 2018	Publication