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3rd Generation Partnership Project
Technical Specification Group GERAN
Support for voice optimization for the IMS in the GERAN
(Release 5)

GLOBAL SYSTEM FOR
MOBILE COMMUNICATIONS

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Contents

Foreword

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

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Version x.y.z

where:

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  - 1 presented to TSG for information;
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- y the second digit is incremented for all changes of substance, i.e. technical enhancements, corrections, updates, etc.
- z the third digit is incremented when editorial only changes have been incorporated in the document.

1 Scope

The present document provides an overview of the architecture and issues related to the provision of voice optimisation within the GERAN.

2 References

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

[1] 3G TS 26.090, "Adaptive Multi-Rate (AMR) speech transcoding".

[2] 3G TS 26.190, "AMR Wideband speech codec, Transcoding functions".

[3] IETF AVT internet-draft "RTP, A Transport Protocol for Real-Time Applications", http://www.ietf.org/internet-drafts/draft-ietf-avt-rtp-new-09.txt.

[4] IETF RFC768, "User Datagram Protocol".

[5] IETF RFC760, "Internet Protocol".

[6] IETF AVT internet-draft, "RTP payload format and file storage format for AMR and AMR-WB audio", http://www.ietf.org/internet-drafts/draft-ietf-avt-rtp-amr-10.txt.

[7] IETF RFC2543, "SIP: Session Initiation Protocol".

[8] IETF RFC2327, "SDP: Session Description Protocol".

[9] IETF AVT internet-draft "Low Delay RTCP Feedback Format", http://www.ietf.org/internet-drafts/draft-fukunaga-low-delay-rtcp-02.txt.

[10] IETF AVT internet-draft "RTCP-based Feedback: Concepts and Message Timing Rules", http://www.ietf.org/internet-drafts/draft-wenger-avt-rtcp-feedback-02.txt.

[11] 3GPP 26.101, "AMR Speech Codec Frame Structure".

[12] 3GPP 26.201, "AMR Wideband speech codec; Frame Structure".

[13] 3GPP 26.093, "AMR Speech Codec; Source Controlled Rate operation".

3 Definitions, symbols and abbreviations

3.1 Definitions

3.1.1 Terminal integrated application

In this document the term "Terminal integrated application" is used. An application is considered to be "integrated in the terminal" when the application is co-located with the PDCP and RRC protocol entities.

3.2 Symbols

Editors note: to be completed

3.3 Abbreviations

SIP Session Initiation Protocol

DTM	Dual Transfer Mode
CS	Circuit Switched
GERAN	GSM/EDGE Radio Access Network
FACCH	Fast Associated Control Channel
RTP	Real time Transport Protocol
UDP	User Datagram Protocol
IP	Internet Protocol
UL	Uplink
DL	Downlink
TS	Time Slot
CN	Core Network
SS	Subsystem

4 Overall description of voice over IP in the IMS domain when connected to GERAN

GERAN is considering the solution to provide an optimized voice bearer as well as generic bearers to support speech originating from the Iu-ps. The optimization is achieved by reusing the channel coding of CS speech channels in GSM, and by employing header removal to increase the spectrum efficiency. The consideration regarding header removal was made with the understanding that header removal is a non-transparent header adaptation scheme and that therefore optimized voice can't be used together with synchronized medias.

Optimized voice will be used in conjunction with SIP. Agreed schemes in GERAN to transport SIP are DTM (Dual transfer mode: going over to 2 half rate or full rate slots during the transmission of SIP data) or FACCH, stealing speech frames during the SIP transmission periods. Both schemes are already provided by GSM R99 or earlier.

5 Definition of optimized voice schemes

5.1 Header Removal

Transport and network level headers (e.g. RTP/UDP/IP) are completely removed. Based on information submitted at call set-up and based on information derived from lower layer (link & physical), the receiving entity can regenerate the headers. The primary application of header removal is the optimized speech bearer, and the regenerated header may not always be semantically identical to the original header.

5.2 Header Compression

Transport and network level headers (e.g. RTP/UDP/IP) are compressed in such a way that the decompressed headers are semantically identical to the original uncompressed headers. The IETF ROHC WG is responsible for standardising header compression schemes. Header compression is suited for standard Internet applications that are not designed to work only with GERAN and especially for multimedia applications therefore the scheme will be used with generic real time multimedia bearers.

6 Requirements and working assumptions for support of voice optimisation for the IMS in the GERAN

6.1 Requirements

1. It shall be possible to use a SIP based optimised voice service with a mobile terminal supporting multi slot class 1 (1 TS in DL, 1 TS in UL).
2. There shall be no performance degradation in coding and modulation compared to traditional circuit switched voice services.
3. GERAN shall not interpret SIP messages.
4. The GERAN solution shall utilise as far as possible already existing protocol means on the Iu interface for UTRAN.
   4.1 Although UTRAN has no plans to deploy header removal in Rel 5, a solution shall take into consideration UTRAN developments and UTRAN architectural principles.
5. The change between header compression and header removal shall be possible during handover.
6. Interruptions in speech due to SIP signalling, mid call, shall be kept to a minimum. SIP compression is required.
7. The GERAN solution shall be future proof and shall not exclude the support of multiple codecs.
8. Whether header regeneration is carried out in the MS shall be an implementation issue
9. It shall be possible to identify whether the terminals has requested an optimized voice bearer or a generic radio bearer for carrying voice in the call data records (CDR).

6.2 Working assumptions

1. It is unclear when/whether mid path transcoders for the IMS will be available between two SIP end users.
2. TSG GERAN is responsible to develop the header removal solution for an Optimized Voice bearer.
3. GERAN informs the MS which codecs are currently supported and the MS is in charge of identifying a single codec, which is supported by the GERAN and the other SIP endpoint (FFS). The mobile requests resources from the network.
4. The GERAN will make the final decision whether or not header removal is possible to apply, or if a generic radio bearer will have to be used.
5. It will not be possible to use header removal for bearers that are part of a multimedia session requiring synchronised media streams.
6. As RTP time stamps and sequence numbers are generated in the BSS, thus there might be an offset in the generated headers across a handover event. Positive or negative slips in sequence numbers may occur in such a situation.
7. In initial implementation it is assumed that the application that generates and receives the flow for which header removal is applied, is integrated in the terminal. Refer to 3.1.1 for the definition of an application that is integrated in the terminal.
8. Header removal cannot be used where end-to-end encryption or integrity protection is used as it does not guarantee bit-exact transfer of traffic.

7 Issues for the support of header removal within GERAN

The purpose with the following subchapters is to capture all issues related to the support of header removal within GERAN. Each subchapter is in turn divided into subchapters describing the characteristics of the problem, possible solutions and the working assumptions that have been agreed.

When a working assumption has been adopted, the solutions that have not been chosen are not removed. The reason for this approach is to avoid that discussions around matters that already have been concluded, shall pop up again at a later stage.

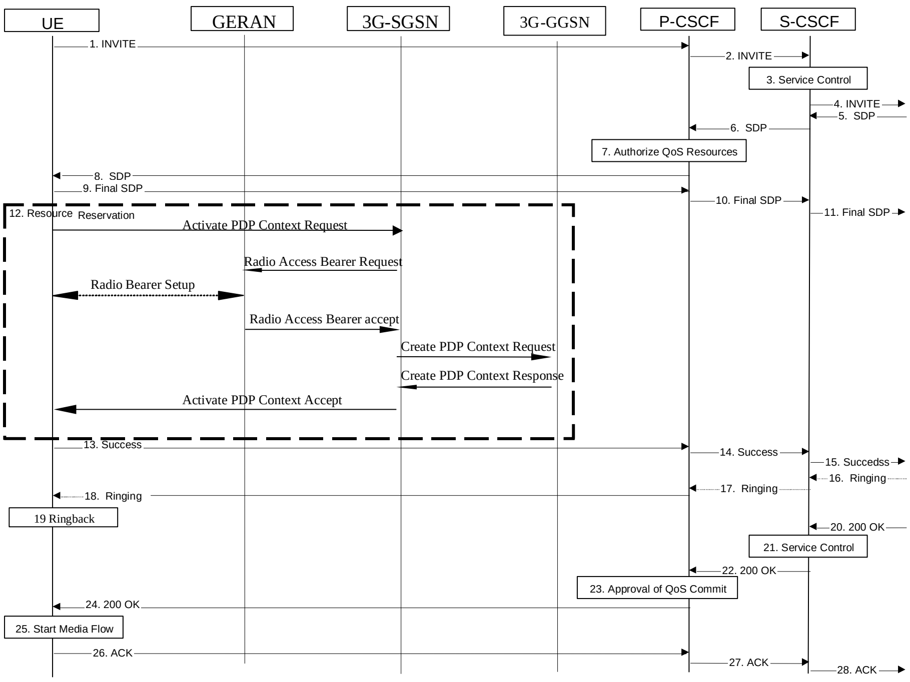


Figure 1. The figure illustrates the process when an Iu-PS voice call is set-up in GERAN. The overall principles are, where nothing else stated, basically the same in all solutions described in chapter 7.

7.0.1 Summary of GERAN/MS States for IMS Calls with Header Removal

A number of events have been identified. These are:

1. At some point, MS is made aware of current local Codec/Channel Coding Support

   Note: This information may no longer be accurate by the time that the GERAN needs to allocate a bearer, due to changes in resource availability

2. MS has PDP Context for SIP traffic

3. MS has engaged in SIP signalling; a Secondary PDP Context Activation has not yet been requested

4. MS sends the Secondary PDP Context Activation Request; SGSN sends an associated RAB Assignment Request; BSC has not selected RB and channel coding scheme yet

5. BSC selects the final RB and channel coding scheme, and initialises the PDCP entities as part of the RB (or extended RB) setup procedure. There are two variants here

5.1 Header Removal is used

Note: This event defines the "latest point" at which IP address/port and Payload Type information must be available at the BSC in order for Header Removal to be initialised, regardless of the technique used to deliver the information.

5.2 Header Compression (or no adaptation) is used.

6. RB Setup is complete, but media traffic has yet to begin

Note: It is assumed that SIP call setup is complete before media traffic transfer begins

7. Media traffic transfer is active; the access link is stable

8. Handover occurs whilst maintaining the current PDCP mode (and the same Codec/channel coding scheme). Again, there are two variants here;

8.1 Header Removal is used

8.2 Header Compression (or no adaptation) is used.

9. Handover occurs whilst maintaining Header Removal, but involving a change in Codec and channel coding scheme (including a change in ACS whilst maintaining AMR as the Codec)

10. Handover occurs, together with a PDCP mode change from Header Removal to Header Compression. There are two variants here:

10.1 PDCP Mode change from HR to HC is part of an Inter-RAT change from GERAN to UTRAN

10.2 PDCP HR to HC mode change occurs within GERAN

11. Handover occurs, together with a PDCP mode change from Header Compression to Header Removal. This has two variants:

11.1 There is also an Inter-RAT change from UTRAN to GERAN

11.2 PDCP mode change without such an Inter-RAT change"

Note: Some of these events (notably those involving complex handover cases) are not covered within the Technical Report.

7.0.2 PASNAS Information Structure

A group of data items can be used to assist the PDCP processor in selecting the appropriate scheme to be used when performing packet adaptation within the RAN. It can be viewed as a set of "suggestions" from the MS to the RAN. This information is doubly optional:

7 the MS need not send it (relying on default behaviour from the RAN),
8 the RAN need not act on it (either due to the information not being appropriate in the particular configuration, or because the requirements are not applicable to the processing it will perform).

This group of Packet Adaptation Specific Non-Access-Stratum (PASNAS) Information is intended to be valid for a given bearer and applicable over the lifetime of that bearer. Other information may also be needed to specify fully the operation of Packet Adaptation in the RAN; this need not be transferred in the same way, or even at the same time.

Data specifying the traffic type to be carried within a bearer used in a multimedia call is RAN-specific, and is needed only to improve the efficiency of RAN-based packet adaptation. It should not have an impact on the operation of the Core Network, and should not need to be modified or read by the Core Network, even if this is used to relay the information between the MS and the RAN. As a result, this information should be carried inside a Transparent Container whilst being relayed via the Core Network.

Such an (optional) PASNAS Information structure will contain a set of fields, with two initial field types being defined. If the PASNAS Information has not been provided by the MS, then the GERAN will have to assume a conservative approach to header adaptation (i.e. Header Removal will not be possible).

The possible field types that may be contained in such a structure are specified next.

7.0.2.1. Adaptation Type Requirements

It is proposed that a bit set be used to hold flags indicating the mobile application's special requirements on the adaptation scheme to be used for the associated bearer.

There are currently two situations in which a mobile application may want special treatment of packets being carried through a RAN. These are reflected in the following flags. There may be other flags added in the future, but these two cover the initial needs for GERAN and may be useful for the UTRAN case as well. It is expected that additions to these flags will be restricted, and so the Adaptation Type Requirements field can be of a fixed size (e.g. the bit set will fit into a single octet, for a total field size of two octets, including the tag).

If all these flags are set to false, the GERAN should interpret this as an explicit statement by the mobile application that it is no specific adaptation type requirements. In this case, Header Removal is allowed.

7.0.2.1.1. Synchronisation Indicator

Header Removal is not possible within a GERAN if the speech media flow is part of a multimedia application requiring synchronisation between the different media flows (see Section 7.5.1 of the TR on Optimised Voice). Thus, one reason why Header Removal might mot be allowed by the mobile application is that the associated bearer is to carry such a "synchronised stream". The RAN will be unaware of this fact, and so, if the mobile application requires special treatment for this flow, it will have to indicate this, using a "Synchronisation Indicator" flag.

More generally, it should still be possible to use Header Adaptation where a bearer is so indicated; however, the RAN should not use adaptation mechanisms that will make it difficult for such synchronisation to be maintained. The radical processing involved in Header Removal is only one such "unacceptable" technique.

7.0.2.1.2. Bit-Identical Encoding Required

There is another reason why some forms of adaptation may be unacceptable to a mobile application. It is possible to produce other "lossy compression" schemes that might be appropriate for some traffic types. For example, HTTP (web) messages use a text encoding and could be re-encoded into a canonical form with compression. The resulting message would not be bit-identical. For most purposes, this is acceptable, but there are situations in which it is not; for example, if application-level integrity protection had been applied to the HTTP message, then this would fail when checked against the message that had been re-encoded to a canonical form.

Introducing a "Bit-Identical Encoding Required" flag could allow the PDCP entities to restrict their processing to adaptation that preserved the identical bit pattern of the message. Of course, it follows that indication of such a requirement would, by definition, mean that Header Removal was not allowed as this technique does not guarantee bit-identical transfer.

7.0.2.2. Traffic Type

It is proposed that a "Traffic Type" structure be introduced. This will indicate the traffic to be used within the associated bearer, and will include a parameter set the interpretation of which is specific to the Traffic Type carried. This is the Traffic Type Parameters.

For each different Traffic Type, the parameters might have a different structure or be empty. If an implementation receives such a structure and does not recognise the Traffic Type Identity value, it can ignore the whole structure, as this implies that it does not support a specific adaptation mechanism to process this traffic.

Traffic Type Identity:

(Unknown | IP | TCP | UDP | UDP/RTP | UDP/SIP | TCP/HTTP |,…)

The interpretation would be that the associated bearer is expected to carry packets of this type. There are several values that can be considered at this point; of these, only the RTP value is required for Header Removal to function. However, the others are given as potential examples; at present, all other values should be reserved.

- “Unknown” means that the kind of data carried in this bearer is completely unknown.
- “IP” means that the bearer is known to carry IP datagrams, but these hold a mix of TCP and UDP packets.
- “TCP” means that this bearer will carry TCP packets, but the kind of application level protocols carried in the TCP packets is unknown, or is a mix of protocols.
- “UDP” means that this bearer will carry UDP packets, but the kind of application level protocols carried in the UDP packets is unknown, or is a mix of protocols.
- “UDP/RTP” means that it is known that this bearer will carry only RTP packets.
- “UDP/SIP” means that this bearer will carry only SIP messages.
- “TCP/HTTP” means that the bearer will be used to carry web requests and responses only.

Traffic Type Parameters:

– Parameters (if any) associated with this traffic type

For Header Removal, the Traffic Type Identity ‘UDP/RTP’ is required. In this case, the Traffic Type Parameters will be interpreted as carrying Codec Type information. The internal structure of this sub-field is covered next.

7.0.2.2.1. Codec Type

Where the Traffic Type Identity is ‘UDP/RTP’, the associated Traffic Type Parameters should be interpreted as a list of triple values, each consisting of the Codec Identity, ACS Modes used, and the Payload Type associated with this Codec/ACS combination. It is valid for the length of the parameter to be zero (i.e for there to be an empty list of Codecs).

Conversely, note that there might be, in the future, more than one codec used for traffic carried in a single bearer, so the parameters for this traffic type should form a list of entries. For example, data reflecting DTMF-coded signals (encoded according to RFC 2833) might be interspersed with data from speech. The situations in which such use of more than one Codec Type is valid are for further study, but using a list structure does not preclude this possibility for future systems whilst ensuring “backward compatibility”. Each list entry consists of the following tuple:

Codec Identity:

(Unknown/Unspecified | GSM-FR | GSM-EFR | GSM-HR | AMR-NB | …)

Note – other values should be reserved.

ACS Modes Used:

Bit Set, with one entry per mode, each of which is a Boolean flag indicating whether or not this mode is part of the ACS. If the associated Codec does not use Active Codec Sets, then only one mode would be expected to be set true. The default value {00000000} (i.e. no modes in this set) should be used where modes are not known or are not applicable.

Note that the mapping between particular modes and positions in the bit set is TBD.

Also note that, to ensure forward compatibility, this bit field will need to hold flags for nine modes, to allow for the future introduction of AMR-WB to the GERAN.

Payload Type:

This is a copy of the Payload Type identifier to be used in RTP packets carrying data encoded according to the associated Codec Identity. This value is an 7 bit unsigned integer.

7.0.2.3. Example PASNAS Information

Combining these two field types, the following structure might be expected for the example described above, in which a bearer was to be used exclusively to carry RTP packets with a Codec Identity of ‘GSM-FR’, a Payload Type of 96, and for which the Mobile Application decided to state explicitly that it had no special requirements on the adaptation technique applied.

‘PASNAS Info’

{

'Traffic Type'

Traffic Type Identity – 'UDP/RTP'

Traffic Type Parameters – {{'GSM-FR', {00001000}, 96}}

'Adaptation Type Requirements'

{SI= 'false', BiER= 'false'}

}

7.0.3. Summary of the issues addressed in chapter 7

- How shall the SIP negotiation between the endpoints be performed, and how to make sure that the endpoints have all necessary information in order to complete the negotiation.
- The principle of how GERAN figures out which speech codec that has been selected in order to apply the appropriate channel coding schemes.
- The principle of how to signal/negotiate a change in codec during an ongoing call.
- The principle of how to select active codec set (ACS) when AMR is used.
- How and when header regeneration shall be applied.
- The principle of how GERAN figures out whether or not header removal may be applied.
- How the IP and port numbers are communicated between the UE and the PDCP entity in the BSS.
- How GERAN-GERAN, UTRAN-GERAN, GERAN-UTRAN handovers shall be performed with regard to header removal.
- How mid call SIP communication shall be performed.
-

7.1 Optimized voice call set-up within the IMS

7.1.1 Description of

Call set-up in the CS domain is based on the following principle:

1. The terminal announces its capabilities
2. The network select speech codec to be used

Call set-up in the IMS domain is based on a fundamentally different principle:

1. The terminal endpoints negotiate speech codec (or more generally media codecs) to be used
2. The terminal request the resources required, to the network.

The IMS SIP negotiation currently does problem not take into account any access specific information concerning the codec negotiation. This is particularly the case when the access network modifies the codec packets in some way as in header removal. The BTS may lack support for some of the channel coding schemes that corresponds to the speech codecs supported by the MS.

The solutions as proposed below may be combined. For example one solution can be adopted for initial implementation and may be further improved in combination with another solution.

7.1.2 Proposed solutions

7.1.2.1 MS knowledge of GERAN channel coding capabilities at the start of or before SIP negotiation

7.1.2.1.1 Description of the solution

This solution is based on the principle of letting the peer involved in the SIP call set-up know about the capabilities of the GSM/EDGE Radio Access Network (e.g. supported channel codings in the cell). Such knowledge has to be provided prior to, or during, the SIP-based call set-up. Several solutions are possible:

- The knowledge is provided as a new Information Element appended to the RADIO BEARER SETUP message, setting up the Radio Bearer for SIP signalling.
- Other solutions are possible and may be added

This solution is also based on the principle of having a deterministic rule for the BSS to work out that the RAB being established carries SIP-signalling. Several solutions are possible:

-

- Define a new Source Descriptor choice for SIP signalling - Make an on-demand request during the SIP negotiation (FFS).

- Other solutions are possible and may be added

When the user moves to another cell after SIP negotiation has started but before it is completed, the capabilities supported by GERAN may change. Several solutions are possible to handle this:

:

- The BSS handovers the resources used for the SIP Radio Bearer and the HANDOVER COMMAND or RB RE-CONFIGURATION message, whichever is used, can include such information for the new cell (see 44.018);
- The MS re-selects the new cell and sends a CELL UPDATE to the BSS. The response from the network can include such information for the new cell (CELL UPDATE CONFIRM or RB RE-CONFIGURATION).
- Other solutions are possible and may be added

If the channel coding capabilities supported by the old cell are not the same as those supported in the new cell, this may trigger codec re-negotiation at SIP level.

The impact on SIP level codec negotiation is then the following:

- In case of Mobile Originated call the selection of QoS attributes, codec, etc for each media flow described in the SDP contained in the SIP INVITE shall then take into account not only the SIP client own capabilities but also the capabilities of the GERAN. Each media flow will be associated to a list of all the codecs that are supported by both the originating SIP client and the controlling GERAN (as far as the necessary channel codings are concerned) and which fulfil the QoS required for the media flow. The SIP negotiation then takes place according to 3GPP TS 23.228.
- In case of Mobile Terminated call, when the addressed SIP client receives the SDP contained in the SIP INVITE, it shall then take into account the codecs that it accepts itself and that are supported by its controlling GERAN (as far as the necessary channel codings are concerned) before accepting the SDP and send the reply to the originating SIP client.

Such a solution will not require any SIP level codec renegotiation in cells where the same set of channel codings is supported by all transceivers. In case transceivers of a cell do not all support the same channel codings (e.g. some support TCH/FS and TCH/AFS codings, others support only TCH/FS), it may happen that a codec is negotiated at SIP level for which there is no transceiver availability at the time the Radio Bearer is set-up (e.g. AMR NB is chosen). This would imply SIP level codec renegotiation. This solution is therefore particularly suited for network deployments where a consistent set of channel codings is supported by all transceivers of a given cell. However, this does not require all cells of the network to support the same set of channel codings. This is further described in Annex A.

This solution may, if necessary, be further improved in combination with solution 7.1.2.3.

7.1.2.1.2 Pros and Cons- The SIP radio bearer is set up when the MS makes itself available to the IP Multimedia Subsystem. However, the SIP negotiation only takes place when a call is being received or initiated by the MS. Between these two events, a substantial amount of time may expire. During this time, the set of supported codecs may change due to high network load in the current cell, or because the user is moving into a new cell. This will lead to extra signalling between the MS and network.

7.1.2.2 SDP message delayed

7.1.2.2.1 Description of the solution

In this solution the proposal as described in 7.1.2.1 is enhanced. By delaying the final SDP message sent by the calling party until the resources have been allocated within the GERAN, there is no risk that a codec is selected that requires a channel coding scheme that is not supported in the BSS.

7.1.2.2.2 Pros and cons

- This solution will not work in the case where no mid path transcoding is carried out, such as in the case of IMS MS to IMS MS call where both mobiles are accessing the network via GERAN. The reason for this is that two different GERAN entities are involved in the SIP negotiation phase, and it has to be assumed that those GERANs may come up with different codec selections.
- This proposal changes the current working model for the IMS as defined in 23.228v5.0.0. This would cause substantial changes to the currently agreed information flows and would have to be agreed both in S2 and CN1. S2 has made a clear indication (LS Tdoc S2-011577) that:
  "this solution should be removed from consideration".

7.1.2.3

non-3GPP

7.1.3 Working assumption

Solution 7.1.2.1 is the current working assumption. Several sub-alternatives exist in 7.1.2.1, Nno agreement has been reached so far on working assumption on that level,

Solution 7.1.2.2 is removed from consideration.

Solution 7.1.2.3 requires quite some changes and additions in [6]. This make this solution non feasible in short term. However this solution add a value by outlining a future proof evolution of 7.1.2.1.

7.2 Handling of ACS for AMR

In case of optimized speech with AMR codec there are additional issues that are related to managing the ACS as listed

7.2.1 Max four AMR modes can be part of an ACS I GERAN, at a time

7.2.1.1 Description of problemIn case of a session between GERAN MS and some other IP terminal, the IP terminal (somewhere in the IP cloud or in UTRAN) assumes that any of 8 modes are possible if the SIP level negotiation would result with AMR. However this is not true over the GERAN air interface as seen in R98 GSM AMR specifications. There could be maximum 4 modes

7.2.1.2 Description of solutions

7.2.1.2.1 AMR Format Parameters

This could be solved using MIME negotiation during the SIP/SDP where the ACS could be negotiated too. For example A party indicates (in SDP) ACS {12.2,7.95,7.4} and B party indicates ACS {10.2, 7.95,7.4}. So, the resulting common ACS would be {7.95,7.4}. It is clear that A party must only use modes included in the ACS that B party has indicated. Furthermore, although in general case the ACS means only the modes that a terminal is willing to receive, it seems quite clear that in GERAN case A party knows that it is only allowed to transmit modes included in its own ACS.

7.2.1.3 Working assumption

The MiME approach 7.2.1.2.1. is currently the GERAN working assumption

7.2.2 How to change the ACS at any given time

7.2.2.1 Description of problem

If we assume that only one codec and one ACS is agreed at the SIP negotiation, dynamic behavior of GERAN system (possibility to change ACS any time) would require SIP level re-negotiation of ACS. This re-negotiation is seen as incall modification of the session (SIP signaling during the speech call) and in order to transmit SIP signaling during the call, we have to use DTM like solution, so go to HR+HR constellation and this in turn requires changing ACS, since ACSs are different for FR and HR.

7.2.2.2.1.1 Description of solutions

7.2.2.2.1 use of a consistent Active Codec Set in geographic regions

In order to avoid SIP level negotiation a similar solution as described in 7.1.2.1, could be adopted. This would mean that a consistent set of ACS should be supported in the network.

7.2.2.3 working assumption

No working assumption has been reached so far

7.2.3 The encoder may have to use a more robust rate than the requested

Header removal functionality in PDCP will act as a proxy and receive AMR speech samples encapsulated in the RTP packet according to [6]. For downlink the speech samples are passed through channel encoder and the Mode Indication is set according to the information obtained from the AMR payload format for RTP. According to [6] the other end could ask using CMR (Codec Mode Request) field to receive a codec mode that would not be possible over the air interface in uplink at a certain time (or to be more precise it could be possible but the link quality could be so bad that the speech quality would be severely impacted). An example: The B party asks for 12.2, but the link conditions dictate the usage of more robust mode, for example 7.4. According to [6] GERAN PDPC header removal entity is mandated to send 12.2 in uplink, so it needs to set the Mode Command to 12.2 in the 2 AMR signalling bits. This issue is not unique and appears also in TFO cases. One simple solution would be to relax the requirement in [6].

Editors note: According to [6] this seems to already be possible. This section may be removed, or reformulated.

7.2.4 How to force a change to an AMR rate able to be carried on a HR physical channel

7.2.4.1 Description of problem

There may bea need to change from Full Rate to Half Rate channels. This may be the case in high traffic load situations. It may also be necessary if DTM is used for SIP signalling and only one TS in UL and DL can be used (also refer to chapter 7.9).

7.2.4.2 Description of solutions

,

7.2.4.2.1 Choose a HR compliant ACS

One way to avoid SIP level re-negotiation is to choose an ACS that would be compliant with half Rate channels..In case of GMSK NB AMR this would mean to restrict the highest mode in ACS to 7.95. The implications of such restrictions should be evaluated.

Editors note: An example of signalling flow for MS initiated optimized speech is provided in appendix B.

7.2.4.3 Working assumptions

No working assumptions reached so far.

7.3 Radio Bearer Identification for GERAN

7.3.1 Description of problem

When GERAN is about to apply header removal, it is necessary for GERAN to identify which codec is used, as the corresponding channel coding algorithm has to be applied. Furthermore, in the case where AMR is used, GERAN must also be informed of which *active codec set* is used. GERAN can only handle up to four rates in its active codec set.

Editors note: The relation of operation of AMR over IP and GERAN's limited active codec set needs to be clarified in cooperation with SA2.

7.3.2 Solutions

7.3.2.1 Direct communication between the UE and the BSC

7.3.2.1.1 Description of the solution

It is proposed to keep the exchange of information related to header removal completely within RRC. All required information is then transferred within extended RADIO BEARER SETUP messages as outlined in Figure 2.

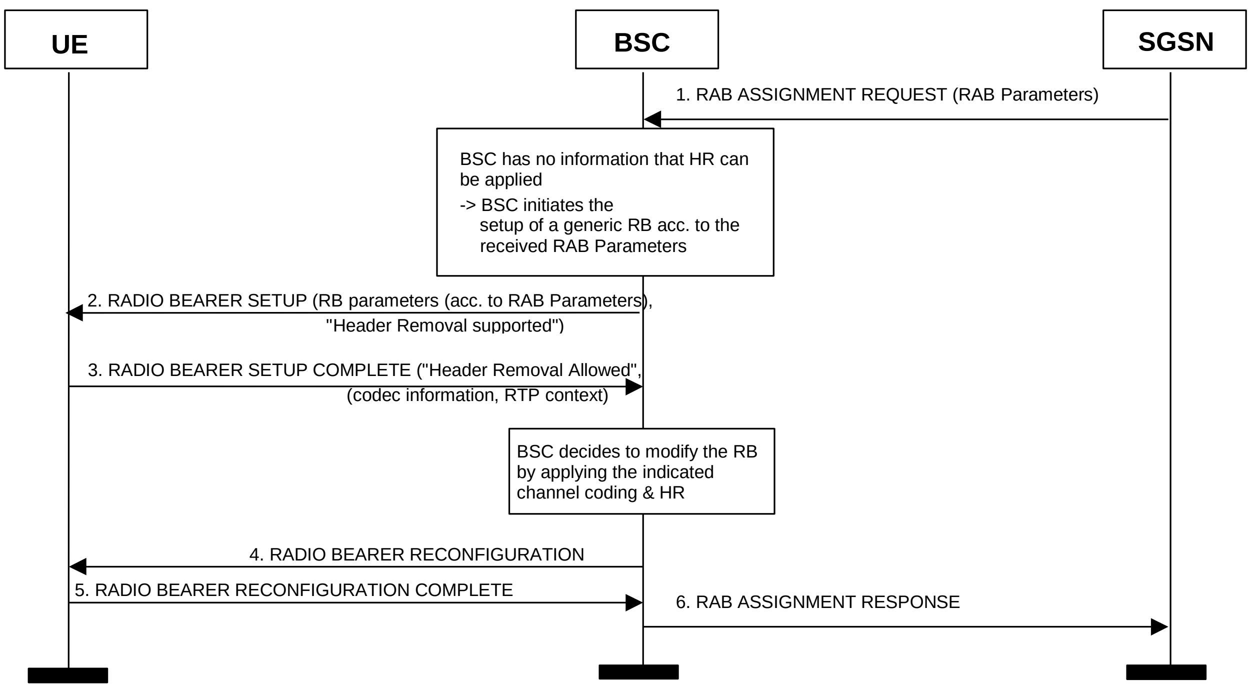


Figure 2: Extended RB set-up procedure

1. The SGSN starts the setup of the RAB with RAB ASSIGNMENT REQUEST containing a generic QoS request as received from the UE via Session Management.

2. The BSC has no knowledge so far whether header removal could be applied to this RAB. Therefore the BSC will initiate the setup of a generic radio bearer according to the received QoS received in RAB ASSIGNMENT REQUEST. Within the RADIO BEARER SETUP message the BSC may include an indication to the MS, that header removal is supported in the RAN (e.g.: by sending a flag "Header Removal Supported"). Note that it is FFS whether the BSC shall indicate the support of HR at that point in time.

3. The MS has to check whether or not header removal is possible for that media stream. If this is the case, the MS sends all information needed to be able to apply header removal within a container inside the RADIO BEARER SETUP COMPLETE message to the BSC, i.e. a flag indicating "Header Removal Allowed", negotiated codec information and the RTP context.

4. The BSC detects that header removal, i.e. optimised voice can be applied. If the BSC decides to modify the (generic, not optimised) RB according to the information received from the MS in the RB setup complete message received, it starts a RB modification procedure by sending the RADIO BEARER RECONFIGURATION message to the MS.

   If the BSC decides not to modify the RB it successfully terminates the RAB Assignment procedure instead of sending RB RECONFIGURATION to the MS.

5. MS sends back a RADIO BEARER RECONFIGURATION COMPLETE message.

6. BSC responds to SGSN with RAB ASSIGNMENT RESPONSE.

Now the establishment of the radio link is finished and a codec-specific channel coding will be applied. After this modified setup of the radio bearer, the signalling will be continued as described in Figure 2.

For MSs only supporting 1 TS in UL and 1 TS in DL the extended RB setup procedure might look different: the MS will not support the generic RB, because the amount of data for a generic RB will not fit into one timeslot. Therefore the MS has to reject the first radio bearer setup. But it could transfer all required information within the RADIO BEARER SETUP FAILURE message to the BSC and might also include the flag “Header Removal Allowed”, which indicates to the BSC that the setup of a RB with applied header removal will be successful.

7.3.2.1.2 Pros and cons

+ no impact to CN

+ if header removal is not to be applied to the media stream, the RB setup procedure remains unchanged (except for the transfer of the “Header Removal Supported” flag from the BSC to the MS and the “Header Removal Allowed” flag set to false from the MS back to the BSC).

  - Two more messages will be required to setup the RB for optimised speech. However the significance of the added delay within the whole setup procedure has to be verified.

  - It is unclear how a CDR shall be generated in order to be able to charge differently for optimized voice and voice carried over a generic radio bearer

7.3.2.2 SDU format information approach

7.3.2.2.1 Description of the solution

Detailed QoS information is provided in the ‘Activate PDP context request’ message by using the ‘SDU format information’ attribute. This information uniquely identifies the appropriate channel coding in the GERAN. However, ‘SDU format information’ would have to be introduced in R5.

For multi rate codecs such as AMR, it is important that the SDU format is provided for all rates even though only a subset has been negotiated on SIP-level, in order for GERAN to be able to identify the codec unambiguously.

7.3.2.2.2 Pros and cons

- The solution proposed does not specify how a potential future codec is uniquely identified if that codec has exactly the same bit mapping and protection for each class of bits in the payload format of an existing codec.

7.3.2.3 Activate PDP context request message approach

7.3.2.3.1 Description of the solution

Following the SIP negotiation, which needs to result in one desired codec, the UE expresses this request explicitly by stating the desired codec in the subsequent resource request to the network. A field containing the specific speech codec desired is introduced in the 'Activate PDP context request message' to the SGSN, by extending the QoS information element. More specifically, the codec information can be an extension of the 'Source Statistics Descriptor' field that will be part of the QoS IE in R5. (The R99 QoS information element included in the Activate PDP context request message is shown in section 7.5.2.).

This information is then passed to the GERAN at the 'Radio Access Bearer Request', by also extending the 'Source statistics descriptor' in the RAB QoS parameter set.

For AMR, it is assumed that the preceding SIP negotiation not only results in 'AMR', but rather AMR plus a preferred active codec set consisting of four or less rates. This active codec set information is then conveyed from the UE to GERAN. Thus, in case of AMR, the new field in the QoS information element, sent from the UE via SGSN to GERAN, comprises both AMR and the preferred active codec set.

Editors note: This section may be updated to reflect concerns expressed on service specificity. It is intended to place the codec information within a transparent container to be relayed via the SGSN.

7.3.2.3.2 Pros and cons

+ This solution is straightforward and imposes limited changes to existing standards. It is architecturally clean in that it uses existing messages for resource requests from the UE to GERAN. The codec information can potentially be used by other purposes as well, for example charging.

- Its potential drawback is that the PDP context message, which is a request for a bearer service, includes application-related information. To avoid this, one could consider the 'SDU format information' approach (section 7.3.2.2), which however introduces a bigger impact on the PDP context message size.

7.3.3 Working assumption

Currently option 7.3.2.3 seems to be the most promising solution.

7.4 Limitations due to RTP handling

Editors note: This section is to be restructured.

The Sequence Number (SEQ) and Timestamp (TS) in the RTP header determine the time instant when the contents of a packet is played out at the receiver. The SEQ is expected to increment by one at the receiver, otherwise it will be interpreted as a gap in the sequence. Also, the first TS value received is used as a reference at the receiver. This reference together with a timestamp determines the presentation time of subsequent RTP packets. During handover events (and possibly during normal operation), positive or negative slips in sequence numbers may occur. Depending on the size of the slip this may cause degradation of speech quality. A positive drift in a subsequent timestamps will cause the RTP receiver to generate a silence period. The length of this silence period will be equal to the drift in seconds. A negative drift in the timestamp will cause the RTP receiver to drop the packet, since from its perspective, the presentation time for the contents of the current packet has passed.

7.5 Identification of header removal allowed

7.5.1 Description of problem

As described in chapter 7.3, GERAN will be made aware if a supported speech codec is used, and if so, which one. However, it is also necessary for GERAN to identify whether or not it is allowed to use header removal. If the speech media flow is part of a multimedia application requiring synchronisation of the different media flows, header removal is not allowed.

7.5.2 Solutions

7.5.2.1 Activate PDP context request message approach

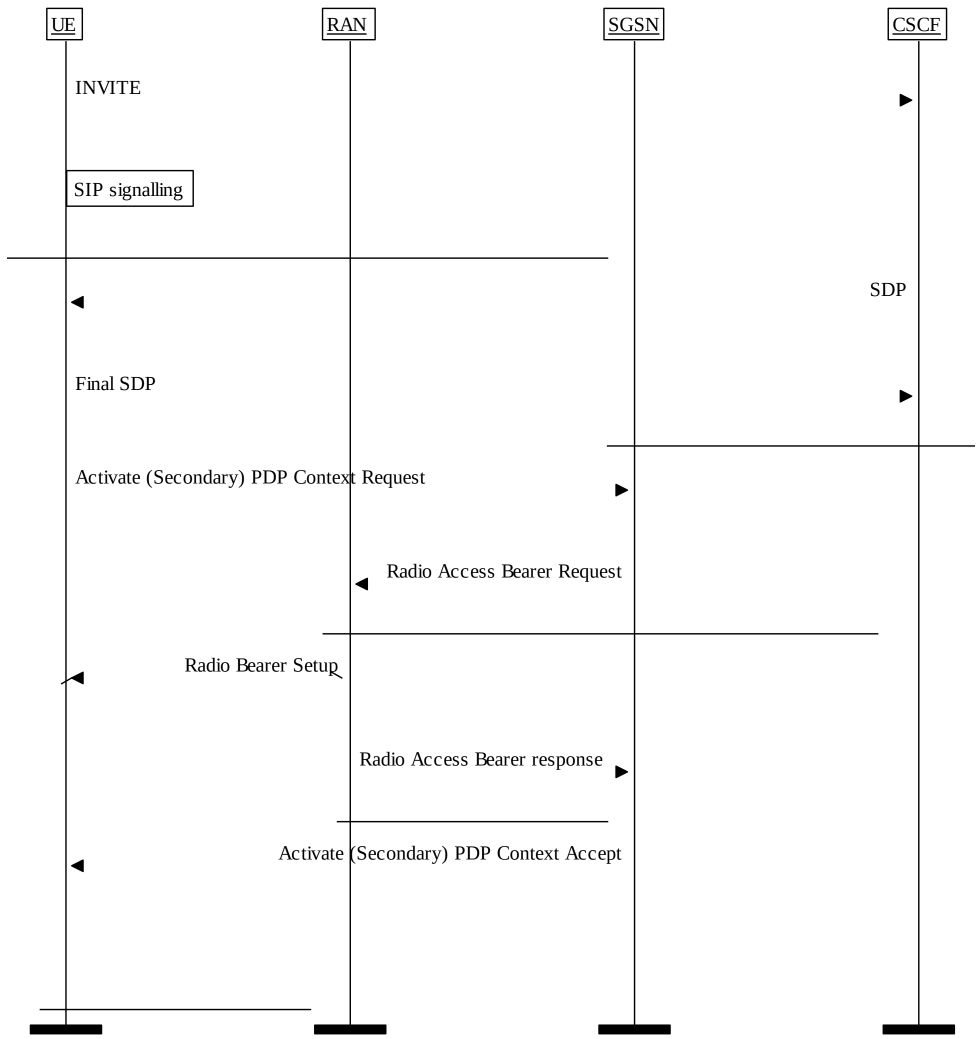
7.5.2.1.1 Description of the solution

This solution is based on the principle of providing the information whether or not header removal is allowed in the Activate PDP context request message. Several solutions have been presented how to name these bits:

1. ‘Header removal allowed’ bit.
2. ‘This flow may be synchronized with other flows’ bit. (Synchronization Indicator from PASNAS)
3. The bit may be part of a transparent container delivered to the GERAN via the SGSN.

7.5.2.1.1.1 Header removal allowed bit Since header adaptation mechanism is dependent on the application (e.g. in case of VoIP only application header removal is possible) one solution is that the MS indicates the header adaptation mechanism to be applied for a particular PDP context. The indication could be part of the Quality of Service IE, and thus the solution can be combined with the solution presented in section 7.3.2, solving also the radio bearer identification problem.

The signalling flow for the solution is given in the figure below:



The application will use the SIP signalling for setting up the session, and UE is the entity that knows the type of application used for the session.

After the initial phase of SIP signalling is completed (i.e. the session description has been agreed), the UE will activate the PDP context. Specifically in case of optimized speech (VoIP with header removal) the UE will send the Activate Secondary PDP Context Request message to the network. This message contains the Quality of Service Information Element. New field is needed in QoS IE to indicate the preference of the header adaptation mechanism for the particular PDP context. An example of the field could be as shown in the following table. Table shows the QoS IE as specified in 24.008 v4.1.1.

<table>
<tr><td>8</td><td>7</td><td>6</td><td>5</td><td>4</td><td>3</td><td>2</td><td>1</td><td></td></tr>
<tr><td colspan="8">Quality of service IEI</td><td>octet 1</td></tr>
<tr><td colspan="8">Length of quality of service IE</td><td>Octet 2</td></tr>
<tr><td colspan="2">0 0
spare</td><td colspan="3">Delay
class</td><td colspan="3">Reliability
class</td><td>octet 3</td></tr>
<tr><td colspan="4">Peak
throughput</td><td>0
spare</td><td colspan="3">Precedence
class</td><td>octet 4</td></tr>
<tr><td colspan="3">0 0 0
spare</td><td colspan="5">Mean
throughput</td><td>octet 5</td></tr>
<tr><td colspan="3">Traffic Class</td><td colspan="2">Delivery order</td><td colspan="3">Delivery of erroneous SDU</td><td>Octet 6</td></tr>
<tr><td colspan="8">Maximum SDU size</td><td>Octet 7</td></tr>
<tr><td colspan="8">Maximum bit rate for uplink</td><td>Octet 8</td></tr>
<tr><td colspan="8">Maximum bit rate for downlink</td><td>Octet 9</td></tr>
<tr><td colspan="4">Residual BER</td><td colspan="4">SDU error ratio</td><td>Octet 10</td></tr>
<tr><td colspan="6">Transfer delay</td><td colspan="2">Traffic Handling priority</td><td>Octet 11</td></tr>
<tr><td colspan="8">Guaranteed bit rate for uplink</td><td>Octet 12</td></tr>
<tr><td colspan="8">Guaranteed bit rate for downlink</td><td>Octet 13</td></tr>
<tr><td colspan="6">Spare</td><td colspan="2">Header Adaptation</td><td>Octet 14</td></tr>
</table>

Figure 10.5.138/TS 24.008: *Quality of service* information element

Table 10.5.156/TS 24.008: *Quality of service* information element

Header Adaptation (Octet 14) Bits 2 1 In MS to network direction: 0 0 No header Adaptation preferred 0 1 Header Removal preferred 1 0 Header Removal not possible 1 1 Spare

The SGSN send the RAB assignment request as specified in 25.413 and include the proposed "Header Adaptation" field in RAB Parameters IE. SGSN could as well use predefined QoS parameter combination in the RAB assignment message which would give unambiguous information to GERAN that header removal can be used.

When receiving the RAB assignment request, radio access network would choose the header adaptation mechanism according to its algorithm and inform the UE using Radio Bearer Set-up message.

The example shown above is only one possibility on how to convey the necessary information to the radio access network. If this solution is combined with the solution described in section 7.3.2.3 (dealing with the problem of radio bearer identification), there is potential room for parameter optimisation. One possible scheme is that an explicit codec indication (according to 7.3.2.3) by default implies that header removal is allowed and preferred, making a specific ‘header adaptation’ field superfluous. Such syntax details are FFS.

7.5.2.1.1.2 ‘This flow may be synchronized with other flows’ bit

Editors note: This is an alternative way of providing the necessary information (as described in chapter 7.5.1) from the terminal to the GERAN. This section is to be completed.

7.5.2.1.1.3 Information provided in a transparent container

Editors note: This is an alternative way of providing the necessary information (as described in chapter 7.5.1) from the terminal to the GERAN. This section is to be completed.

7.5.2.1.2 Pros and cons

+ This solution has the advantage that it implies very limited changes to existing specifications.

- A possible drawback is that that higher protocol messages such as the PDP context messages have to convey header adaptation information, which can be considered as being radio access related. Given the nature of optimized speech and its relation to the application setup, this drawback would seem inevitable.

7.5.2.2 Direct communication between the UE and the BSC

A different method is used to indicate that HR is possible when using the direct communications approach (see section 7.3.2.1.1).

7.5.3 Working assumption

No agreement reached so far.

7.6 IP and port number information transfer from MS to GERAN

7.6.1 Description of problem

In order to carry out header regeneration in the uplink the relevant information must be communicated with the PDCP entity in the GERAN. A number of possibilities have been identified, so far, in order to transfer IP and port numbers from the MS to PDCP in BSS.

7.6.2 Solutions

7.6.2.1 RRC signalling approach

7.6.2.1.1 Description of the solution

The information is provided by RRC signalling at RB set-up.

7.6.2.1.2 Pros and cons

Editors note: To be completed.

7.6.2.2 TFT approach

7.6.2.2.1 Description of the solution

The information is sent in a TFT from the MS to SGSN, which in turn provides the information to the BSC.

7.6.2.2.2 Pros and cons

Editors note: To be completed.

7.6.2.3 Direct communication between the UE and the BSC

A slight variation on the RRC signalling method is used when using the direct communications approach (see section 7.3.2.1.1). The data is included within the message sent by the MS during the extended RB setup procedure.

7.6.3 Working assumption

Currently solution 7.6.2.1 seems to be the most promising solution. However the expertise of TSG RAN and TSG SA is needed in order to make a decision.

7.7 Handover issues in optimized voice

7.7.1 Description of problem

When inter BSS, inter RAN or BSS-RAN handover takes place, the header generation context may have to be relocated. A mechanism for this purpose is needed. In addition, it should be clarified how slips in RTP sequence numbers and timestamps can be minimized or completely eliminated.

7.7.2 Proposed solutions

7.7.2.1 Time stamp and sequence number handling during a handover

7.7.2.1.1 Description of the solution

This solution assumes that handover is carried out as specified in 44.018 and that relocation follows the procedures that have been specified in 25.413 and 23.060. As a part of the relocation of the RNS context the location of the header removal / generation function is moved from the source BSS to the target BSS.

In the case of GERAN to GERAN handover, a way to ensure smooth continuation of the time stamp value is to utilize synchronized clocks in the network entities carrying out header removal/generation. This has been illustrated in figure 3. The MS sends voice frames 1-4 via source GERAN. Header generation function creates RTP packets and uses local clock to generate the time stamp information for each packet. After sending the relocation commit and handover command messages the "data path" is switched to go via target RAN. The clock synchronization is utilized by including the latest time stamp information and the corresponding clock time in the Relocation Commit (or Forward SRNS Context) message. When the target RAN receives the message it can, based on the local clock and the received information, deduce the right time stamp value. Some frames may be lost during the handover but that should not cause any problems as long as the time stamp value continues without disruption.

Editors's note: In here clock synchronization does not mean BTS synchronization but merely that the clocks in network entities carrying out header generation have been setup to the same time and are reasonably close to each other in rate.

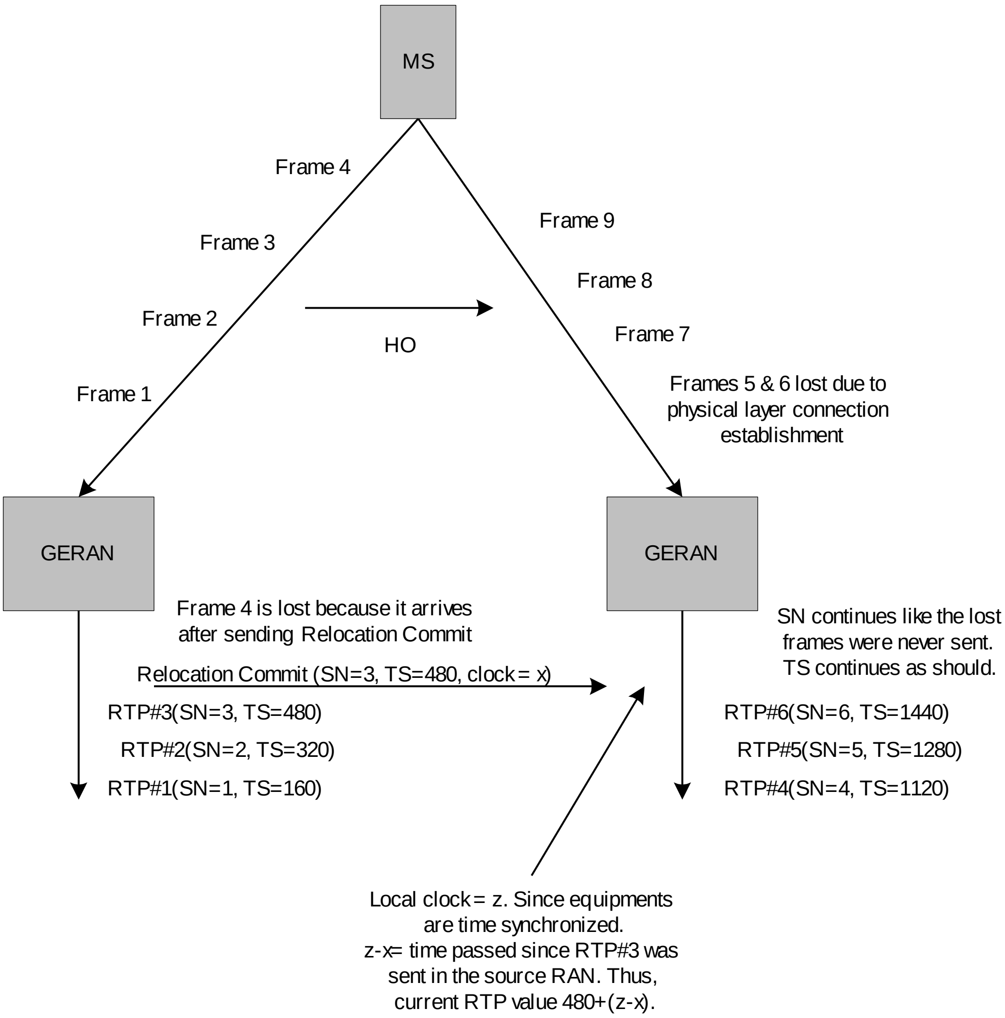


Figure 3. Time stamp synchronization in GERAN–GERAN handover (Note that in case of AMR for each 20ms frame time stamp increases by 160).

In case of GERAN to UTRAN handover the header adaptation mechanism changes from header removal to header compression and the location of the RTP end point moves from the network to the terminal. In this case large jumps in the field values are avoided by transferring the time stamp, sequence number fields and the TDMA frame number from the network to the terminal inside a container in the Handover To UTRAN Command.

When the MS receives the handover command it can deduce the correct time stamp value from the current TDMA frame number and the received information. The procedure is illustrated in figure 4. In the example the RTP packets 1-3 are sent through GERAN using header removal/generation. After sending the third RTP packet the network sends a handover command to the terminal containing the TDMA frame number when the packet 3 was sent and the corresponding time stamp and sequence number information. In the terminal the right RTP time stamp value can be deduced from the received information.

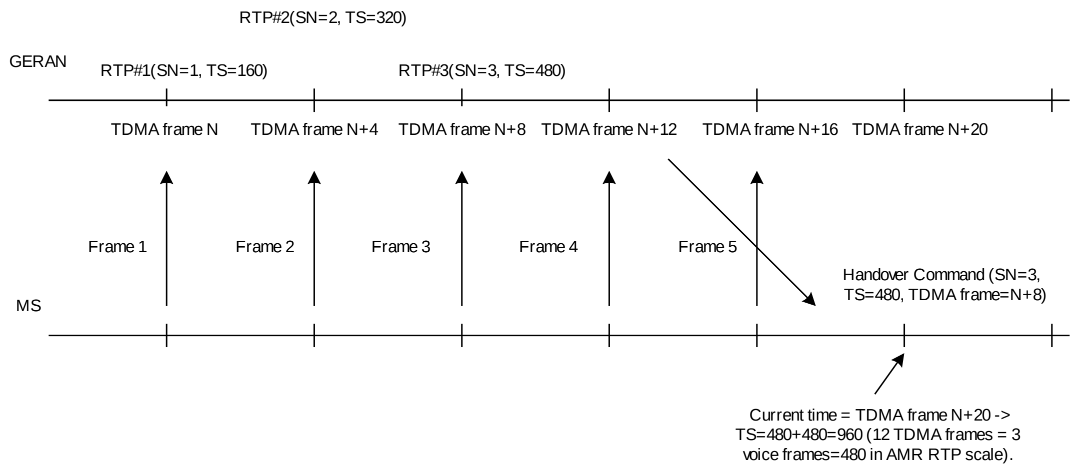


Figure 4 Time stamp synchronization in GERAN – UTRAN HO / relocation.

7.7.2.2 Pros and cons

- The proposed solution may lead to small drift in the transferred field values. It is the assumption that this does not cause large quality degradation. However, this needs to be verified from IETF AVT group. The size of the drift will be directly proportional to the number of muted or discarded frames as explained in section 7.4

7.7.3 Working assumption

No agreement reached so far.

7.8 Mid call legacy codec support

7.8.1 Description of problem

The Radio Access Network infrastructure may not support all possible channel coding schemes in all areas, and, potentially, the set of channel coding schemes supported in one area may be completely different from the set supported elsewhere. If an IMS call is active and uses Header Removal (and so relies on an unequal error protected channel coding scheme associated with the current CoDec), this can cause problems in mid-call.

7.8.2 Solutions

If, during a call, a resource that has been used is no longer available, there are two choices to resolve this problem.

Either:

- The PDCP Mode must be changed from Header Removal to Header Compression (and the radio bearer should be configured to use an equal error protected channel coding scheme), or
- The Codec used in the media stream will need to be changed to one that is associated with a supported unequal error protected channel coding scheme

7.8.2.1 PDCP Mode Change

7.8.2.1.1 Description of the solution

Editors note: To be completed

7.8.2.1.2 Pros and cons

Editors note: To be completed

7.8.2.2 Mid Call Codec Change

It is **assumed** that the call control entities must maintain a valid specification of the media transport in use.

If the codec used is to change in mid-call to one not specified in the existing session description, then the description agreed by the SIP end points at the start of the call will no longer reflect the actual media streams being exchanged. From the above assumption, this will require SIP messages to be exchanged "end to end" holding a replacement session description. This is shown in section 7.8.2.2.1 – 1.

If the codec change is to one included already in the existing session description, then alternatives not requiring SIP message exchanges may be used; these are covered in section 7.8.2.2.1-2.

Note that, if the session description includes only one codec at the end of call setup, then there is no alternative to engaging in a SIP call re-negotiation. The "non-SIP" alternatives assume that there is more than one codec included in the session description at the end of call setup.

7.8.2.2.1 Description of the solutions

1. SIP call re-negotiation

[Standard IMS procedure as will be described in TS24.228]

2. Non-SIP Codec change signalling

If a media description, *at the end of call set up phase*, includes a set of alternative CoDecs with more than one member, then a change in CoDec between these listed alternatives would not invalidate the session description agreed during call setup, and so no SIP message exchanges would be needed in this event.

It is **assumed** that listing more than one alternative within the session description does not negate the requirement that the same codec be used in both directions of a call at any one time. Although, in principle, such a session description might seem to allow different CoDecs to be used in either direction, the policy will be to only support the bi-directional case. To maintain this policy, any change to the codec used by an end point should be signalled to ensure that both end points change codec at the same time; an end point should not simply decide to swap CoDecs without agreeing this with its peer.

There are several options for signalling a codec change without the use of SIP message exchanges. These are covered next.

a. RTCP Message Exchange

This approach is based on exchanging RTCP messages between the RAN that detects a resource problem and the remote system, using the "fast feedback" scheme. It has two variants; one variant proposes to use Sender Report and Receiver Report messages to carry indications between the network-based PDCP entities of a proposed codec change. The other variant uses the "Application-specific" message type to carry the indications between the peer entities This solution proposes to use RTCP to change/re-negotiate the ACS during an RTP session. The RTP proxy or in header removal scenario the header removal/generation function would send RTCP packets containing information regarding the allowed codec modes (ACS) whenever the allowed codec modes changes. The terminal would not participate in this signalling at all because it is the GERAN who decides the ACS. The RTCP packets should not be sent over the air interface.

RTP/RTCP protocols provide two alternatives to realize this: In addition to 'regular' RTCP Sender Reports (SR) and Receiver Reports (RR), it is possible to extend the RTCP functionality with application/payload type specific feedback messages. There seems to be two mechanisms to extend RTCP to support the idea presented here:

1. Section 6.4.3 in [3] specifies a possibility to define an extension field to RTCP SR or RR.

2. Section 6.7 in [3] specifies a possibility to define an application specific RTCP packet type.

There is a work in progress in IETF AVT group on 1, see [9],[10], and it seems like a suitable mechanism to convey AMR ACS update during a session.

Higher level protocols are added on top of RTCP and RTP to allow advance indication (and negotiation) of codec/Payload Type changes. Any such scheme must provide its own reliability mechanism as RTCP and RTP are unreliable protocols.

Editors note: *An example of such higher-level protocol is outlined in G2-010020. This particular solution describes an RTP-based solution.*

Editors note: *The backup solution for the case when the scheme is not supported is an abrupt codec change, resulting in transient packet loss greater than if advance notice would have been given.*

Editors note: *A procedure for layer 3 messaging between the BSC and MS is required when a new ACS (or codec) has been agreed using RTCP or RTP signalling. This is FFS.*

"In Band" Signalling

This approach works by injecting RTP packets into the existing media stream sent towards the core network, and detecting RTP packets that have been injected by the remote peer.

[For Details, see contribution G2-010020]

Editors note: *The backup solution for the case when the scheme is not supported is an abrupt codec change, resulting in transient packet loss greater than if advance notice would have been given.*

7.8.2.2.2 Pros and Cons

- Although using SIP signalling would appear to be the simplest solution, it does have some problems. First, it requires call control signalling to be carried over the air interface. Secondly, it is not easy to see how the Terminal can be informed that it should engage in SIP message exchanges during a Handover; although the GERAN detects the resource problem, it is not a party to call control signalling and so it must have some way to instruct the Terminal to carry out these exchanges. Such an approach would require the expertise of SA WG2 and CN WG 1 groups to clarify the appropriate procedures.

- Both the non-SIP approaches have one major benefit; they do not need any extra signalling to be carried over the air interface (over and above the necessary radio bearer modification procedures that are required on any change to the bearer). Both require a specialised application protocol to be used on top of the existing RTCP or RTP transport protocols. Of the two, the RTCP-based approach would seem to require an extra PDP context to be arranged; how this is done by the BSC is unclear. In addition, this approach has raised some other concerns; it is questionable if it is wise to generate RTCP SR/RRs when the RTP protocol is terminated in the MS and RTCP is terminated in the BSS. In such an architecture, the RTCP RR will contain information about quality in the BSS, not in the MS. It is suggested that it may not be appropriate to make use of RTCP SR/RR if the termination point of the RTP protocol is not in the same node as the RTCP protocol.

- If no RTCP SR/RRs are generated (for the above mentioned reasons), then with the other variant (using "Application-specific" messages), RTCP would be used for the sole purpose of providing a possibility of informing the BSS of a change in the codec or ACS.

- Furthermore, the usage of RTCP for this task is questioned, since RTCP is not a reliable signalling protocol. There is no way of ascertaining that the ACS change has been received correctly, so that more details are required on the way in which the end points can exchange application level indications reliably.

- The RTP based approach does not have the problems of the other schemes, but (in common with the RTCP-based approach) does require that the alternatives are included in the "final" session description agreed at call setup. This solution assumes that it is allowed to negotiate multiple codecs for a SIP-session. Whether this is the case is FFS.

It cannot be ensured that the special RTCP or RTP functionality is deployed in all conceivable endpoints (also non-3GPP).

7.8.3 Working assumption

The current working assumption is based on SIP codec renegotiation 7.8.2.2.1-1. All other schemes as described in this chapter are FFS.

7.9 Bearer support for mid call SIP signalling

7.9.1 Description of problem

It is foreseen that there may be additional mid call IMS SIP communication using header removal.

7.9.2 Solutions

7.9.2.1 Solution A

7.9.2.1.1 Description of the solution

The following means can be used for SIP signalling:

3. FACCH
4. Downgrade to HR channel. This requires further analysis of:
    a. TBF allocations for signalling
    b. The codec selected at the SIP negotiation must be able to be reconfigured to support a HR channel, without SIP level renegotiation.
5. Allocation of additional timeslot

7.9.2.1.2 Pros and cons

Editors note: To be completed

7.9.3 Working assumption

Solution 7.9.2.1 has been accepted as working assumption.

8 Header compression in GERAN

Editors note: To be completed

9 Recommended work for GERAN voice optimization schemes

Editors note: To be completed.

9.1 Recommended work for particular groups

Editors note: To be completed.

10 Open issues

This section lists identified open issues related to support of voice optimisation for the IMS in the GERAN.

Ref	Description of problem	Status
1	Is it necessary to define one channel coding scheme as mandatory in the standard, required to be supported in all GERAN based IMS SIP based calls?	
2	Is there a requirement for the operators to prioritise other channel coding schemes than the default channel coding schemes to be used in the SIP negotiation?	
3	In the case of a IMS user in a communication exchange to a non SIP user where a signalling translator is needed on the control plane to translate SIP messages to the call control used by the other party. 1. What is the status regarding this work in CN3? 2. Is the signalling transition transparent to the end systems?	LS sent to TSG CN3 (TDOC: OVS-01043)
4	What information is available to the UTRAN/GERAN applicable to uniquely identify IMS signalling SIP messages to enable specific treatment by the GERAN (i.e. sending the SIP message over the FACCH). 1. What is the current status and development of the SIP signalling work? 2. Can the QoS (RAB) parameters, so far defined for RANAP be utilised to distinguish RABs for SIP messages (signalling) targeted to/from the proxy CSCF from RABs for the actual speech transport? 3. Can the QoS (RAB) parameters, so far defined for RANAP be utilised to distinguish RABs for SIP messages (signalling) not targeted to/from the P-CSCF (non IMS Signalling) from RABs for the actual speech transport? 4. Can the QoS (RAB) parameters, so far defined for RANAP be utilised to distinguish RABs for SIP messages (signalling) not targeted to/from the P-CSCF (i.e. non IMS signalling) from those targeted to/from P-CSCF? 5. How are the QoS assignments attributed to the SIP signalling in each of the cases stated above?	LS sent to TSG SA2 and TSG RAN3 (TDOC: OVS-01044))
5	In the Optimised Voice service within GERAN, only one codec (and if applicable the AMR Active Codec Set (ACS)) will be the consequence of the SIP negotiation. Is the resulting single codec decision an IMS restriction?	LS sent to TSG SA2, TSG CN1 and TSG SA4 (TDOC: OVS-01046))
6	In the Optimised Voice service using AMR, an indication of the ACS has to be made at the SIP negotiation level. In GERAN a set of four or less rates has to be selected within the AMR codec. The current solution being discussed within GERAN is that the negotiation of ACS on a SIP level is done using MIME encoding of format parameters. In order for this mechanism to work, all entities must understand the request. This raises the issue that the MIME encoding would need to be included in the standards as a mandatory requirement. Is this ability currently defined in the standards?	LS sent to TSG SA2, TSG CN1 and TSG SA4 (TDOC: OVS-01046))
7	If AMR is used; is there a mechanisms that can enforce the use of an AMR mode that can be carried on a physical HR channel (i.e. AMR 795 or lower) within the RTP for carrying Optimised Voice in GERAN?	LS sent to TSG SA2, TSG CN1 and TSG SA4 (TDOC: OVS-01046))
8	GERAN is currently looking into the analysis of the different mechanisms it can use for carrying mid call SIP messages over the GERAN.	LS sent to TSG SA2, TSG CN1 and TSG SA4

	Which are the typical mid call SIP signalling scenarios considered in CN1 call flows?	(TDOC: OVS-01046))
9	What is the current status on PDCP context transfer at handover in RAN2 and when is it planned to be completed?	

Editors note: This section has been agreed on a general level at the joint GERAN/SA2 meeting in Helsinki (1-3 of August 2001). However as it was added after the meeting, all companies have not had the opportunity to review it in detail.

Annex A: Dimensioning principles

It is assumed that legacy transceivers will not be able to support all future channel-coding schemes. The concepts as described below will allow minimizing the number of mid call codec changes. Such codec changes may involve SIP signalling, which may be very detrimental to the perceived voice quality.

A.1 The buffer zone concept

Channel coding capabilities for new codecs may be launched and introduced in limited but homogenous geographical areas. Thus when upgrading the network, all cells in a given area are updated to support the new channel coding schemes. If this is done before the operator allows the speech codec associated with this new channel coding scheme to be used for call set-up, then the number of mid call codec changes will be minimized.

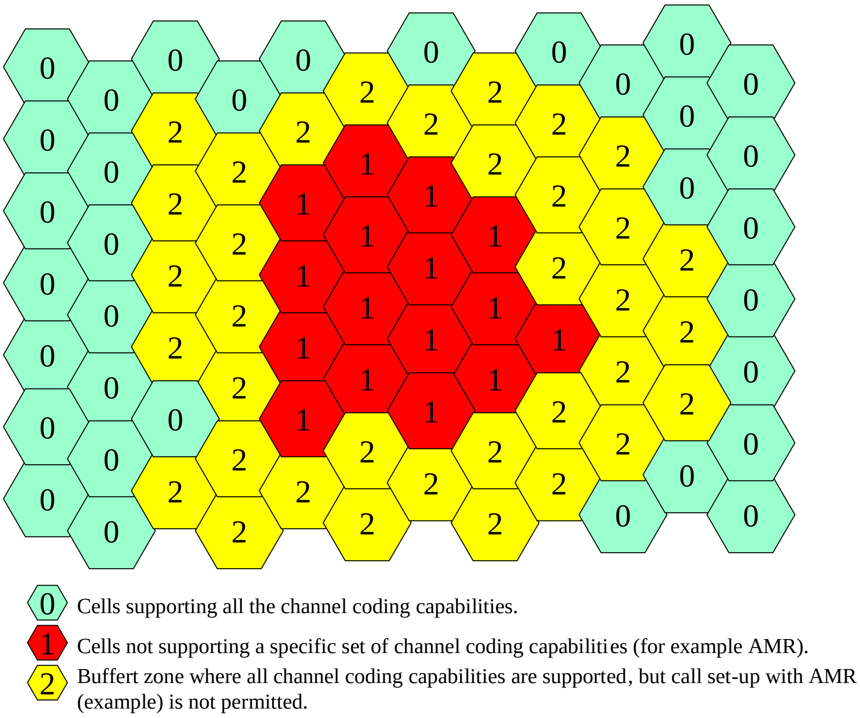
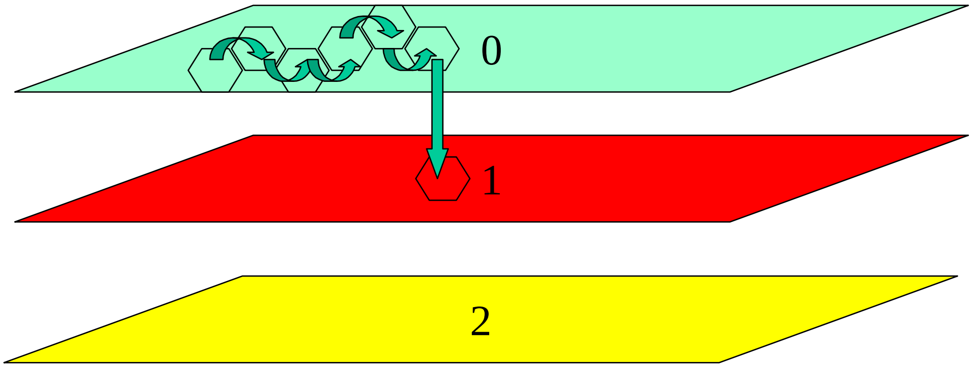


Figure X. Mid call codec renegotiation will only have to take place if a call is set up in a cell marked (0) and the customer moves into some of the cells marked (1).

A.2 The layering concept

Considering a layered cell planning, channel coding capabilities for new codecs may be launched and introduced in one or more layers of the network but not all of them simultaneously. Thus when upgrading the network, all cells in a given layer are updated to support the new channel coding schemes. At call set-up the network can direct the MS to one layer, depending on the MS capabilities. By ensuring that the MS will remain in that layer for the duration of the SIP session (e.g. forbidding handovers between different layers, or at least between layers that do not support the same set of channel coding capabilities) then the number of mid call codec changes will be minimized.



0 Layer supporting channel coding for codec 1 only. The call is set-up in that layer.

1 Layer supporting channel coding for codecs 1 and 2. The resources allocated for the call can be handed-over to that layer provided there are some resources available for the channel coding defined for codec 1.

2 Layer supporting channel coding for codec 2 only. The resources cannot be handed-over to that layer during the call.

Figure X. Mid call codec renegotiation will only have to take place if resources for a call are handed over from layer 0 to layer 2, or from layer 0 to layer 1 and there are no resources available for codec 1.

A.3 Resource dimensioning concept

Cells may be dimensioned by the operator, in such a way that a sufficient number of channel coder resources (non legacy transceivers) are available in each cell, in the areas where a certain channel coding scheme is used. This has the implication that the operator will dimension the channel coder resources in response to congestion detected in the cell.

Mid call codec change will happen in the case where no appropriate channel coding resources can be allocated. This can occur in two cases. Either as a result of resource exhaustion locally or in the remote RAN.

Annex B: An example of signalling flow for MS initiated optimized speech

The following is a simplified example of signalling for optimized speech taking into account some potential solutions described so far in the TR.

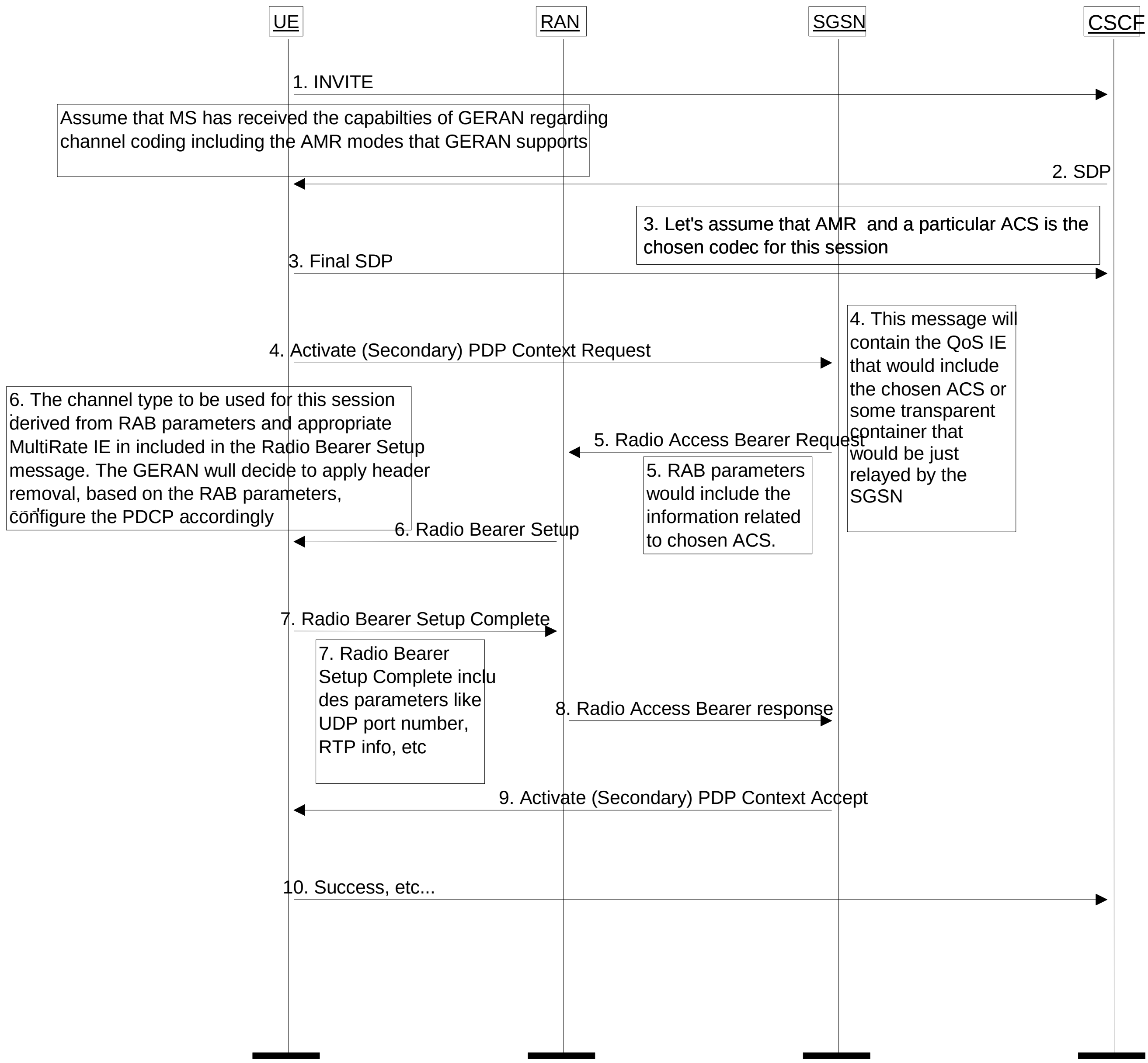


Figure X. Example of signaling flow for optimized speech

Annex C:
Change history

Change history							
Date	TSG #	TSG Doc.	CR	Rev	Subject/Comment	Old	New

Annex D: change history up to v. 1.0.0

(to be deleted when the TR reaches version 2.0.0)

Version	Date	Comment
0.0.1	June 01	Created by GERAN 2 #5bis
0.0.2	8 Aug. 01	Output from the SA2 & GERAN Joint Meeting (RAN delegates invited) on IMS issues and optimised voice services
1.0.0	August 01	Stopped at GERAN#6