3GPP TR 44.933 V1.3.0(2003-09)

Technical Report

3rd Generation Partnership Project;
Technical Specification Group GSM/EDGE;
Radio Access Network;
Seamless Support of Streaming Services in GERAN A/Gb Mode
(Release 6)

GLOBAL SYSTEM FOR
MOBILE COMMUNICATIONS

Keywords

Digital cellular telecommunications system,
Global System for Mobile communications,
GSM, GERAN, radio

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Contents

Foreword

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Introduction

GPRS Release 97 design focused on the support of best effort QoS, which was later defined as Background class. Since then, some improvements have been introduced into the specifications in order to enable the support of Interactive and, to a lesser extent, Streaming services (e.g. NACC, delayed TBF release, Gb flow control per PFC). However, support of the Streaming class may face large interruptions to the service at cell change.

Requirements for the support of streaming services in GERAN A/Gb mode and the performance of existing mechanisms (e.g. for cell change) need to be analysed before appropriate enhancements can be selected and introduced.

The objective of this TR is to collect the available information on these study areas in order to aid the decision on which improvements to existing standards to pursue.

1 Scope

The present document provides a study on requirements, performance and possible enhancements for seamless support of streaming services in GERAN A/Gb mode.

2 References

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

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

[1] 3GPP TR 21.905: " Vocabulary for 3GPP Specifications ".

[2] 3GPP TS 26.233: "Transparent end-to-end packet switched streaming service (PSS); General description".

[3] 3GPP TS 23.107: " QoS Concept and Architecture ".

[4] 3GPP TS 22.105: " Service aspects; Services and Service Capabilities 1".

[5] 3GPP TR 26.937: " Streaming service (PSS); RTP usage model ".

[6] 3GPP TS 22.060: "General Packet Radio Service (GPRS) stage 1".

[7] 3GPP TS 26.234: “Transparent end-to-end Packet Switched Streaming Service (PSS); Protocols and codecs“

[8] 3GPP TS 22.233: "Transparent end-to-end Packet Switched Streaming Service Stage 1"

[9] 3GPP TS 22.105: "Services and service capabilities"

[10] 3GPP TS 44.064: "Logical Link Control (LLC) layer specification"

[11] 3GPP TS 43.064: "Overall description of the GPRS radio interface; Stage 2"

[12] 3GPP TS 24.008: "Core Network Protocols – Stage 3"

3 Definitions, symbols and abbreviations

3.1 Definitions

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

Network: In the context of the RTP/UDP/IP or HTTP/TCP/IP based streaming usage models, network refers to the both GERAN and UMTS bearer service between the entry-point of the GERAN/UMTS network and the UE.

Streaming: The ability of an application to play synchronized media streams like audio and video streams in a continuous way while those streams are being transmitted to the client over a data network.

3.2 Symbols

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

<symbol> *<Explanation>*

3.3 Abbreviations

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

AM	Acknowledged Mode
CBR	Constant Bit Rate
CCN	Cell Change Notification
DCH	Dedicated Channel
DSCH	Dedicated Shared Channel
GPRS	General Packet Radio Service
GTP	GPRS Tunneling Protocol
HSDPA	High Speed Downlink Packet Access
HTTP	Hypertext Transfer Protocol
IP	Internet Protocol
LLC	Logical Link Control
MTU	Maximum Transmission Unit
PCCO	Packet Cell Change Order
PDCP	Packet Data Convergence Protocol
PDU	Protocol Data Unit
PSS	Packet-switched Streaming Service
QoS	Quality of Service
RA	Routing Area
RAN	Radio Access Network
RAU	Routing Area Update
RLC	Radio Link Control
RNC	Radio Network Controller
RTCP	RTP Control Protocol
RTP	Real-time Transport Protocol
SDU	Service Data Unit
SNDCP	Subnetwork Dependent Convergence Protocol
TCP	Transmission Control Protocol
UDP	User Datagram Protocol
UTRAN	UMTS Terrestrial RAN
VBR	Variable Bit Rate

4 Streaming Services in GERAN *A/Gb mode*

4.1 General

The streaming class is defined to accommodate applications for on demand real-time unidirectional multimedia streaming. The main QoS characteristic for real time multimedia streams is the time relation between the information entities (i.e., packets, samples), which imposes requirements on the limitation of end-to-end delay variation of the flows, in addition to the transfer delay. A streaming application at the receiving end will compensate for the delay variation induced by the network up to the limits acceptable by the end user by means of time alignment functions.

The Packet-switched Streaming Service (PSS) ([2] and [7]) provides a framework for Internet Protocol (IP) based streaming applications in 3G networks. In TSG SA WG4 there is on going work on characterizing the PSS such that the underlying network characteristics, streaming mechanism itself and its possible optimizations are taken into consideration. In [5] there are 4 major PSS use cases identified: Voice only streaming (AMR at 12.2 kbps), High-quality voice/low quality music only streaming (AMR-WB at 23.85 kbps), Music only streaming (AAC at 52 kbps) and Voice and video streaming (AMR at 7.95 kbps + video at 44 kbps).

[8] defines the PSS service requirements, among which the PSS service transport requirements as well. The most relevant ones for the seamless support of streaming service in GERAN *A/Gb mode* are listed here:

- Quality of Service (e.g. time delay) requirements shall be in accordance with requirements in [9].
- PSS should be able to work over different QoS bearers.
- PSS should provide mechanisms for streaming servers and clients to adapt to the network conditions in order to achieve significant improvement in the quality of streaming, e.g. using information on end-to-end transport quality from the network.
- PSS should provide a reliable delivery mechanism that enables the user to receive the content without any errors due to the transport mechanism.

4.2 PSS Overall Architecture and the Network Elements

The most important network entities involved in a mobile packet based streaming service is illustrated in Figure 1. A streaming service requires at least a content server and a streaming client. Additional components like portals, profile servers caching servers and proxies might be involved as well to provide additional services or to improve the overall service quality.

Portals are servers allowing convenient access to streamed media content. For instance, a portal might offer content browse and search facilities. In the simplest case, it is simply a Web/WAP-page with a list of links to streaming content servers. The content itself is usually stored on content servers, which can be located elsewhere in the network.

User and terminal profile servers are used to store user preferences and terminal capabilities. This information can be used to control the presentation of streamed media content to a mobile user.

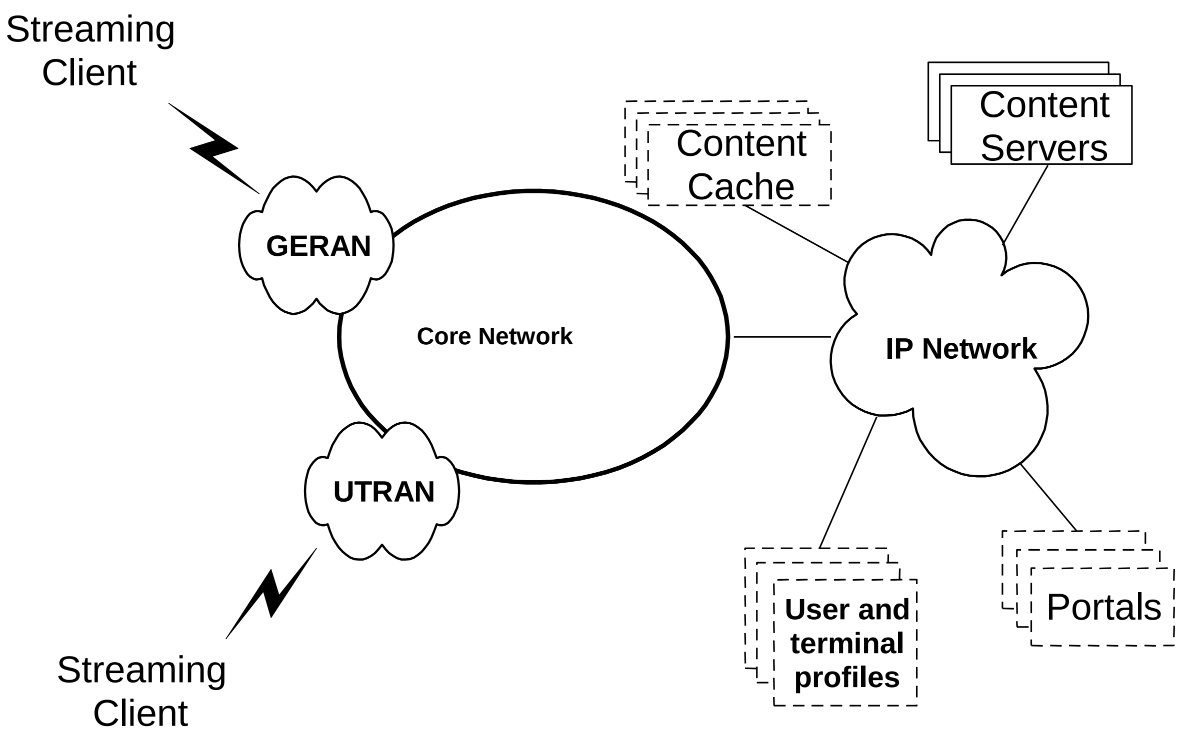

Figure 1. Network elements involved in a mobile packet switched streaming service

4.2.1 Application layer protocols

Most available streaming applications today use the Real-Time Streaming Protocol (RTSP). RTSP is used for controlling a streaming session, i.e. setting up the individual streams and controlling the streaming. This signaling takes place between the content server and the streaming client in Figure 1 (or via the content cache if one is used).

The structure of RTSP looks very much like HTTP. Many header fields and methods in RTSP are left optional to the implementers. Which of the methods and header fields that are required also depends on if the client should be able to control both playback and recording.

4.2.2 Media transport protocols

A well-established transport protocol used for streaming services that require real-time data is the Real time Transport Protocol (RTP). RTP was designed to run on top of a connection-less protocol like the User Datagram Protocol (UDP). RTP is used by many applications, e.g. many IP-telephony solutions. Currently it is the preferred way of sending audio and video data over packet based networks. Associated with RTP is the Real time Transport Control Protocol (RTCP), which provides feedback to the server, about the transmission quality.

4.3 QoS Architecture and Streaming Class Characteristics

A generic QoS architecture covering UTRAN and GERAN is defined in [3], as well as the Streaming Class and its QoS profile. The QoS profile settings for different PSS use cases are given in [5].

When establishing a UMTS bearer and the underlaying Radio Access Bearer for support of a service request, some attribute on UMTS level does typically not have the same value as corresponding attribute on Radio Access Bearer level. For the following attributes the attribute value for the UMTS bearer will normally not be the same as the corresponding attribute value for the Radio Access Bearer. The relation between the attribute values for UMTS Bearer service and Radio Access Bearer service is implementational and depends for example on network dimensioning.

According to [3], the QoS attribute value that will change are:

- Residual BER
- SDU error ratio
- Transfer delay.

4.4 Considerations on Streaming Service Support in GERAN *A/Gb mode*

The values of these QoS attributes for the Radio Access Bearer Service in GERAN *A/Gb mode* will be depending on the following characteristics:

- Residual BER

  In GERAN *A/Gb mode* the target average Residual BER should be given for both LLC acknowledged mode and LLC un-acknowledged mode, considering that LLC operates in protected mode, i.e. FCS covers header plus information field [10]. In GERAN *A/Gb mode* 6*10-8 for 24 bits CRC is the worst-case scenario. Note that LLC un-acknowledged mode can operate in unprotected mode as well, where the FCS field protects the frame header and only the first octets of the information field.

- SDU error ratio

  In GERAN *A/Gb mode* same as with the Residual BER parameter, the value of SDU error ratio is dependent on whether the LLC acknowledged mode or unacknowledged mode is used. For the LLC unacknowledged mode the SDU error ratio will be limited by the 12 bits CRC check on the RLC layer.

- Transfer delay

  In GERAN *A/Gb mode* the transfer delay indicates maximum delay for 95th percentile of the distribution of delay for all delivered LLC PDU, where delay for an LLC PDU is defined as the time from a request to transfer an LLC PDU from one NSAPI to the other NSAPI once the BSS Packet Flow Context (PFC) is set up.

The change of the cell/Routing Area in GERAN *A/Gb mode* requires activation/modification of the BSS Packet Flow Context (PFC) in the new routing path from the SGSN to the MS. Thus for the flows not belonging to the pre-defined packet flow contexts, the BSS PFC needs to be re-initiated and/or modified. During these procedures there will be no data transfer until a logical link is established between the MS and the SGSN.

The interruption of data transfer during these procedures will cause delay variation in the data transfer and possibly packet losses, which may affect the performance of the streaming service. The induced delay variation depends on the duration of data transfer interruption during Cell update and/or Routing Area update procedure.

For Streaming Service support in GERAN *A/Gb mode* it is desirable to limit the data transfer interruption in order to minimize service interruption times and packet losses.

5 Requirements for Support of Streaming Services in GERAN A/Gb mode

5.1 General

The main requirements for Streaming Service support in GERAN *A/Gb mode* are related to:

- Setting the appropriate values of SDU error ratio, Residual BER and Transfer Delay
- The duration of the data transfer interruption due to Cell Update and RAU procedures.

5.2 Streaming Class QoS Parameters

In GERAN *A/Gb mode* the concept of a dedicated packet channel (i.e. radio resources dedicated to one given packet flow only) does not exist. The GPRS capacity (i.e. number of timeslots allocated to packet based services) available is to be shared between all mobiles in the system. The radio resource is to be managed by the RLC/MAC layer. The GPRS capacity is shared by allocating PDTCH(s) on timeslots (PDCH(s)) to the different application packet flows associated with the set of active mobile stations.

The data rate that can be supported for a packet based streaming service depends on the number of time slots within a TDMA frame allocated to a given mobile station (e.g. 3 DL + 1 UL timeslot) and the Modulation and Coding Scheme (MCS) used in the timeslot [11]. The values for the data rates in GERAN *A/Gb mode* are already covered by the ranges of the Radio Access Bearer Service attributes for UTRAN in [3], where the Maximum Bitrate and Guaranteed Bitrate are ≤ 2048 kbps for all classes therefore there is no need to discuss it further.

As given in Section 4.4 the values that need to be defined for Streaming Service support in GERAN *A/Gb mode* are the Residual BER, SDU error ratio and Transfer Delay value:

- Residual BER

  The UMTS Streaming Class Residual BER is 10^{-5}. This value can be set for GERAN *A/Gb mode* as well, because 24 bit CRC check at LLC will satisfy this requirement (see also GP-030782)

- SDU error ratio

  In [5], the value for SDU Error Ratio is 10^{-4}. However, this value cannot be met in case of LLC unacknowledged mode due to 12bit CRC limitations at RLC layer. Therefore the 10^{-3} is agreed (see GP-030782, GP-030784). Note that this is also an acceptable value for the UTRAN Radio Access Bearer Service attributes for streaming class defined in [3]

- Transfer delay

  From the simulation results presented in (GP-030844) transfer delay of 2s required in the UMTS QoS Streaming Service profile and [5] is sufficiently well met by GERAN *A/Gb mode*.

5.3 Service Interruption due to Cell Update and RAU procedures and Packet Losses

The key challenge in providing multimedia-streaming services is the service outage time during cell change or RAU procedure

However defining appropriate values for service outage is not an easy and straightforward task and therefore extensive simulations are needed in order to determine the values that offer the best performance of the streaming service class.

The data transfer interruption because of misrouting due to RAU will most probably cause packet losses. The acceptable values for the packet losses will depend on the type of the streaming application, transport protocols used and in GERAN *A/Gb mode* the LLC mode used. Thus finding out the acceptable values for packet losses will also require extensive simulations, same as for service outage.

The outcome of related studies is summarized in section 6.

6 Performance Study of Streaming Services

6.1 General

This clause provides a study of the performance of the solutions already available in the existing specifications for the support of streaming services.

6.2 Simulation Assumptions

< NOTE: This clause describes simulation assumptions for the performance investigations made in the following sub-clauses. >

6.3 Achievable Service Interruption Times

As outlined in section 5.3, service interruption times caused by cell change and RAU have a major impact on streaming service support in GERAN A/Gb mode.

Service interruption time at cell change will affect the performance of any service and has therefore been studied and also been the target for improvements added to release 4 and 5. The improvements already done are mainly related to the Network Assisted Cell Change (NACC) feature where the MS can be served with required system information for the target cell already before leaving the serving cell.

The NACC support for intra BSC Cell Change was introduced in release 4 of the 3GPP specifications and then in Rel 5 the NACC for inter BSC Cell Change followed. With introduction of NACC the radio outage time is in most cases reduced to less than 1second.

TDoc G2-020778 presents estimated values for service outage in two cases:

- Case 1: Cell change performed between cells within the same BSC and Routing Area;
- Case 2: Cell change performed between cells belonging to BSCs that are controlled by different SGSNs.

The values for total user data outage time that are presented in G2-020778 for Case 1 and Case 2 are 715ms and 1120ms respectively. Similar (lower) results are presented in GP-020765. In both documents it is concluded that these values are ideal and assume no errors in the radio link, however no values have been agreed.

Tdoc GP-030056 hints at adopting some values:

- The service outage time for cell change within BSC and RA in order of 0.5-4 seconds
- The service outage time for cell change between BSC and RA in order of 1-15 seconds, with typical values of 2-3 seconds.

The effect of NMO II and NMO III on service interruption where combined RA/LA updates are not possible must also be considered. Also change of cells between GERAN and UTRAN will sustain long interrupts. These cell changes will cause major disruption of the service for the user.

6.4 Achievable Packet Loss

The reason to avoid packet loss is that most Streaming applications can accept only minimal loss of data. Protocols like RTP/UDP are not providing error detection/recovery. In this case lower layers in GPRS must guarantee nearly loss-less operation.

Loss-less operation can be provided in existing GPRS standards by LLC protocol operating in ABM (acknowledged operation) mode. The implementation of ABM mode of LLC is more complex compared to ADM (un-acknowledge) mode and it is not widely used in existing implementations. Furthermore, due to the need to re-transmit each and every un-acknowledged frame (intrinsic in the concept of “Ack mode”) LLC Ack mode is expected to show some basic difficulties in meeting tight real-time requirements.

On the contrary, LLC unacknowledged mode is generally considered as the natural solution to carry delay-sensitive streaming services, since it does not require time-consuming retransmissions. On the other hand, unack LLC cannot cope in any way with packet loss during a generic cell change.

In the following some simulation results showing performance of LLC ack and LLC unack modes are presented.

6.4.1 Simulation model

A basic streaming service will be considered during the simulations, having the following characteristics:

- 1 RTP packet generated each 133.3 ms (7.5 RTP packets/sec)
- constant RTP packet size leading to an "RTP/UDP/IP/SNDCP/LLC packet" of 500 bytes
- streaming sessions lasting 100 seconds

With the above mentioned numbers a 30 kbit/s (at LLC layer) constant bit rate streaming service can be simulated.

Multislot class 4 (i.e. 3 DL TS, 1 UL TS) mobiles are considered here. For each session, three PDCHs are reserved over the radio interface to guarantee the required bit rate (i.e. 30 kbit/s). The adopted coding is MCS2, providing a bandwidth of 11.2 kbit/s per timeslot, and therefore 33.6 kbit/s over 3 PDCHs (if BLER is zero!).

It is assumed that the "application layer" at the MS side is characterized by a de-jittering buffer. The application starts reading (i.e. extracting packets from the buffer) after a "Start-up Delay" or "Buffering Time" of a few seconds since the reception of the first RTP packet. Over the air interface, both directions (uplink and downlink) are simulated and control messages (PDAs, PUAs, etc.) are taken into account. RLC Acknowledged mode is assumed.

As regards radio channel quality, the following scenario is considered: BLER(MCS1) = 2%, BLER (MCS2) = 10% (MCS1 is used for control messages, BLER is the same for UL and DL). The (DL) bandwidth over the Um interface (nearly 3 * 11.2kbit/s * 90% = 30.24 kbit/s) should be still enough to maintain the real-time requirements (i.e. convey the 30 kbit/s packet stream without introducing delays).

No residual BER is taken into account in simulations. Considering an EGPRS RLC CRC size of 12 bits, the probability to not detect an error is in any case lower than $(1/2)^{12} = 2.4*10^{-4}$ and has been neglected during simulations (note that Residual BER increases the number of lost RTP packets in LLC UM, but also triggers more (possibly useless) retransmissions for LLC AM).

6.4.2 Simulations results with cell change

Simulations were run to analyse the impact of cell changes taking place during streaming sessions. In the following the assumption is that each RTP stream is affected by one cell change, introducing an interruption time of 1 second in the middle of the packet stream. An interruption time of 1 second is only one of the assumptions among the range of simulations performed for the cell change behaviours.

6.4.2.1 LLC unack mode

As already mentioned, at the moment LLC unack is not able to cope with a generic cell change. Therefore, in the following, it is assumed that a mechanism to re-route unsent LL-PDUs from the old cell to the new one is always available. This is currently the case for intra-RA, intra-NSE cell changes (even inter-NSE if "Inter-NSE re-routing" is supported). For other cases (e.g. inter-RA, inter-BSC) a new mechanism would be needed.

In the following, the assumption is that every LL-PDUs that could not be transmitted in the old cell is re-routed to the new one.

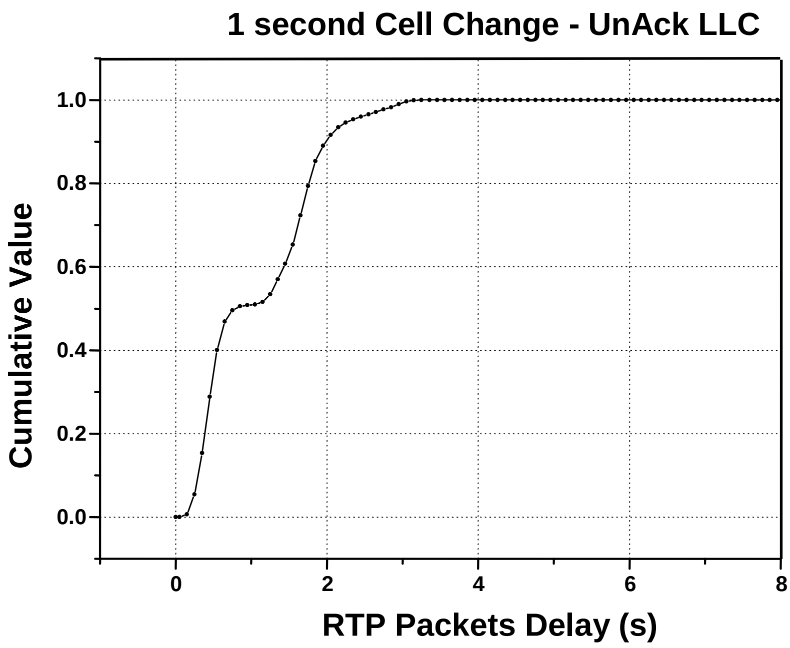

Figure 2: RTP packet delay CDF

In figure 2, the RTP packet delay CDF is shown. It can be seen that every RTP packet is received in less than 3 seconds, so that a "Buffering Time" of 3 seconds would be enough to avoid any packet loss. Again, from the figure it can be seen that the delay of nearly half of the packets is not influenced by the cell change (as expected, since the interruption has been placed in the middle of the packet stream).

6.4.2.2 LLC ack mode

The same simulations were also run using an LLC ack mode strategy. In this case a few parameters have to be properly chosen.

Several values for retransmission timers (T200 and T201) have been taken into account (from 2 seconds to 10 seconds). Using short values increases the probability that a T201-triggered retransmission is received in time at the application layer (according to the Buffering Time), but it increases the number of retransmissions as well. Higher values limit the number of retransmissions but at the same time prevent T201-triggered retransmissions from being received in time at the MS.

In all the cases the PDU lifetime is set equal to T201. Again, N200 is set to 5 in order to avoid ABM re-establishments as much as possible, an infinite buffer size (M) is assumed at LLC layer, k (i.e. the "Maximum number of outstanding I frames") is set to 64, while the A (Acknowledgement) bit is set to 1 (and T201 is started!) every 6^{th} I+S frame.

In the LLC ack mode case no re-routing mechanism is assumed (i.e. it is assumed that, during the cell update, the FLUSH-LL-ACK PDU indicates that the LLC-PDUs associated with the old BVC have been "deleted").

Figure 3 shows the RTP packet delay CDF with different values for retransmission times. From the figure it can be derived that Buffering Times higher than 6 seconds (and T201 >= 6 seconds) are needed to cope with an interruption time of 1 second (in the middle of a 100 second packet stream). Figure 4 shows the number of "lost" RTP packet, i.e. the number of packets received too late at the application buffer, for different "Buffering Time" values (T201 is set equal to 6 seconds).

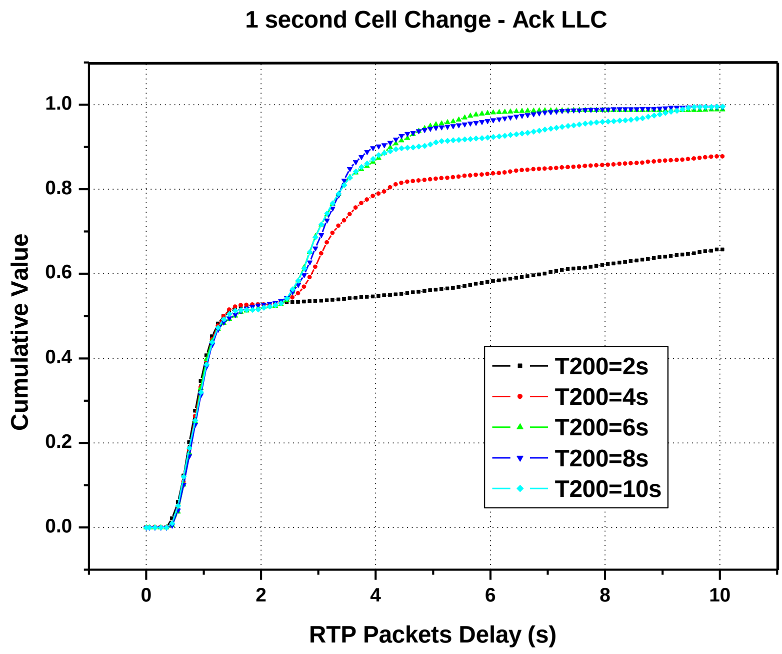

Figure 3: RTP packet delay CDF

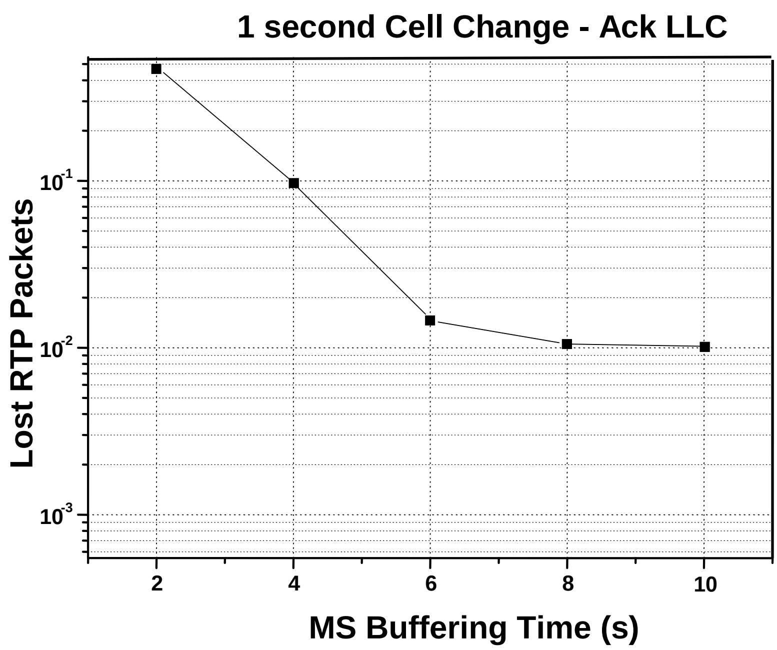

Figure 4: Number of lost RTP packets, for different Buffering Time values (T201 = 6 seconds)

The explanation for the introduced delay is linked to the fact that, after the cell change, several LLC frames can be received out-of-sequence at the MS (see G2-030091 for further information). As soon as an out-of-sequence condition is detected, the receiving LLE will immediately ask for intermediate frames to fill the gap. Nevertheless some time (seconds!) will be needed to receive all the missing frames, because all the already transmitted frames hanging in the BSS buffer have to be transmitted first. In the meanwhile, already received out-of-sequence data (which would be "on time") cannot be passed up to SNDCP->...->RTP layer (because LLC AM needs to perform re-sequencing). When packets are finally sent up to the application (i.e. when all missing LLC frames in between are received), it may be "too late". Briefly: the in-sequence delivery feature of LLC AM may delay even packets that were received in time. This implies the need for higher Application Buffering times. However it should be noted that a larger application buffer may not imply a shorter transfer delay.

6.4.3 Concluding remarks

Streaming services could be supported with LLC ADM maintaining real-time requirements, anyway some enhancements are needed, precisely:

- A new procedure to avoid packet loss during inter-RA/inter-BSC cell changes

On the other hand, some of the identified problems with LLC ABM, when supporting streaming services, are:

- Time to establish/re-establish ABM mode. This could be a serious problem if T200 (=T201) is set to an high value: if the first SABM is lost due to whatever reason, T200 has to expire before sending another one, clearly not respecting real-time requirements.
- Need to re-transmit each and every un-acknowledged frame. To maintain real-time requirements, this implies that longer buffering times must be considered.
- Extra re-transmissions after ABM re-establishments. According to [44.064]: *in the case of LLC layer and peer-initiated re-establishment, the LLE shall issue an LL-ESTABLISH-IND primitive to layer 3 and discard all outstanding LL-DATA-REQ primitives and all queued I frames.* This means that SNDCP should re-submit all SN-PDUs that did not receive any confirmation, possibly increasing the number of useless re-transmissions.
- LLC in-sequence delivery feature, that may delay packets that were received in time
- Stronger dependency (than in LLC unack mode) on UL channel quality.
- Setting of proper values for all the possible parameters (N200, T201, k, etc.). For instance k should be set at least high enough to account for the number of LL-DATA-REQs that can be received during one “LLC Round Trip Time”. This means that the optimal value for k should be linked to the RTP packet arrival rate at the SGSN (which is an unknown parameter). Again, in this case the “transfer delay” QoS parameter cannot be used to configure any parameter/timer (i.e. no action can be taken according to that).

Therefore some enhancements would be needed for LLC AM as well, in order to make it feasible for streaming service support.

As a working assumption, LLC ADM is considered as the most straightforward solution to carry streaming services. Investigations on possible enhancements should focus on this mode (this does not prevent LLC ABM to be further considered).

6.5 Streaming Service Performance Assuming Fixed MS buffer Size

6.5.1 General

In this section a slightly different approach from the above is presented, that is assuming fixed MS play-out buffer of 8 seconds (Play out is started after 8 seconds; Content rate = Server rate)and cell-reselection and RAU outage times, what is the percentage of users experiencing re-buffering during a streaming session? In this section the simulation results for the simulation parameters agreed in (G2-030263) with higher Routing Area Update probability are presented. Unlike in G2-030263, fixed outage times for Routing Area Update and Cell Change are assumed. The video traffic model (frame size) that has been used is based on measurements using H263 video coding algorithm, and the model obtained from measurement is second order autoregressive AR(2) model and is described as:

$$X_n = a_1 X_{n-1} + a_2 X_{n-2} + \varepsilon_n$$

where the terms ε_n are independent Gamma(λ,r) distributed random variables. The following parameter values are estimated from measurements: a1 = 0.664598, a2 = 0.196255, λ = 0.002655, r = 0.07527. Values for mean and variance are: E(X) = 203.7 bytes, std(X) = 187.408.

The average bit rate is 32 kbps with 27 frames per second.

The RTT value at RLC layer used in these simulations is 240ms.

NOTE: The value of RTT in these simulations require more explanation, especially in conjunction with the results presented in section 6.5.2.

For the purpose of this analysis two notions are defined:

- Proportion of users having N zero-playout periods: This is representing the number of re-bufferings that the user experienced during the session, meaning that the play-out buffer in the MS is empty and the user experiences a break in a session. After the play-out buffer is empty the rebuffering is initiated. Note that streaming clients usually have an "underflow threshold" (i.e. 1 second) so that rebuffering is supposed to happen before the buffer is completely empty.
- Distribution of zero-playout times during the session: This is the distribution of cumulative duration of zero-playout periods. Note that this includes the re-buffering times. Re-buffering does not necessarily take as long as the buffer length (in this document this means that it does not necessarily take 8 seconds to re-fill the buffer, since the scheduler typically allocates more resources for re-buffering).

Following abbreviations are used in figure titles: T_call = Session duration, T_ra = RAU outage time, v = MS speed.

6.5.1.2 3km/h, Probability for RAU of 10% and 15%

As it is intuitively expected the results for slow mobility model provide satisfactory user QoS throughout the range of outage times.

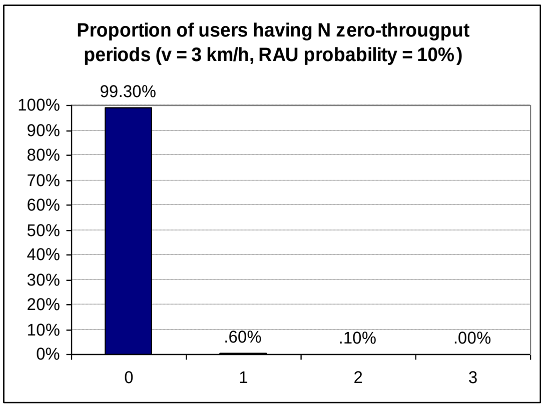

Figure 5. RAU outage 3 seconds with 10% probability, MS speed 3km/h, clip duration 3 minutes

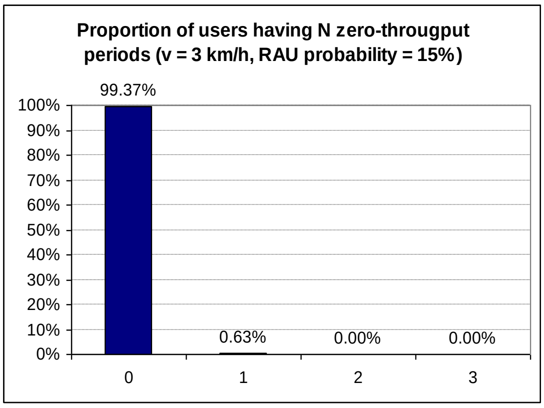

Figure 6. RAU outage 3 seconds with 15% probability, MS speed 3km/h, clip duration 3 minutes

Obviously, the RAU probability does not affect much the results for slow moving mobiles. Even with the higher RAU outage time (5 seconds) the results are very similar.

6.5.1.3 50km/h, Probability for RAU of 10% and 15%

Simulation results for fast mobility model, shows different results. It is clear that with faster users the number of cell reselections and RAU increases rapidly having an effect on the overall performance.

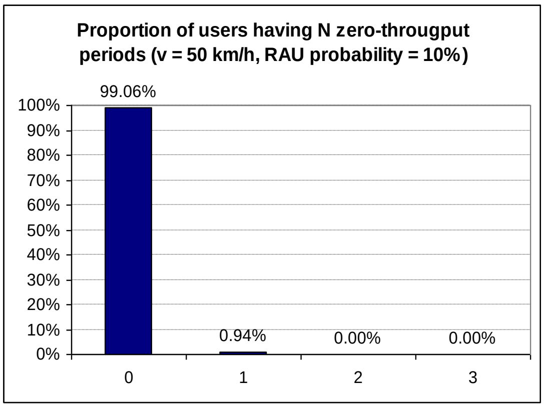

Figure 7. RAU outage 3 seconds with 10% probability, MS speed 50km/h, clip duration 3 minutes

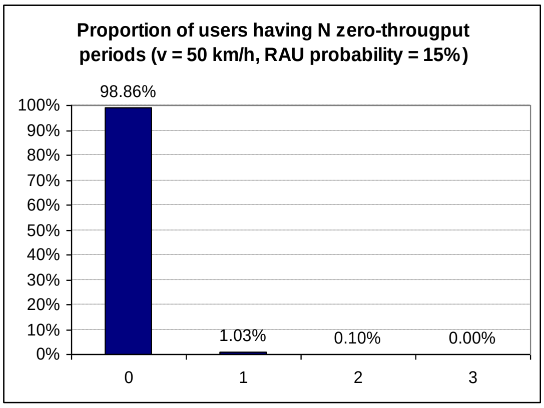

Figure 8. RAU outage 3 seconds with 15% probability, MS speed 50km/h, clip duration 3 minutes

Figures 8 and 9 show very small difference when using different values for the probability of RAU. To confirm this,

Clearly the performance of the network in the 50km/h scenario under the assumptions is acceptable. The 8 seconds buffer length seems sufficient for almost all of the users.

6.5.2 Transfer delay

Figure 9 shows the one-way transfer delay for the case of RAU outage of 3 seconds and different probabilities for RAU and very high load in the network. Very high load in these simulations is considered to be 75% TSL utilization. TSL utilization means the average number of active time slots divided by all available time slots.

The transfer delay represents LLC frame delay value in such a way that it describes the time difference between the moment when the frame has arrived in SGSN's buffer and the moment when all the RLC blocks carrying the bits of certain LLC frame have been received (not acknowledged) by the mobile.

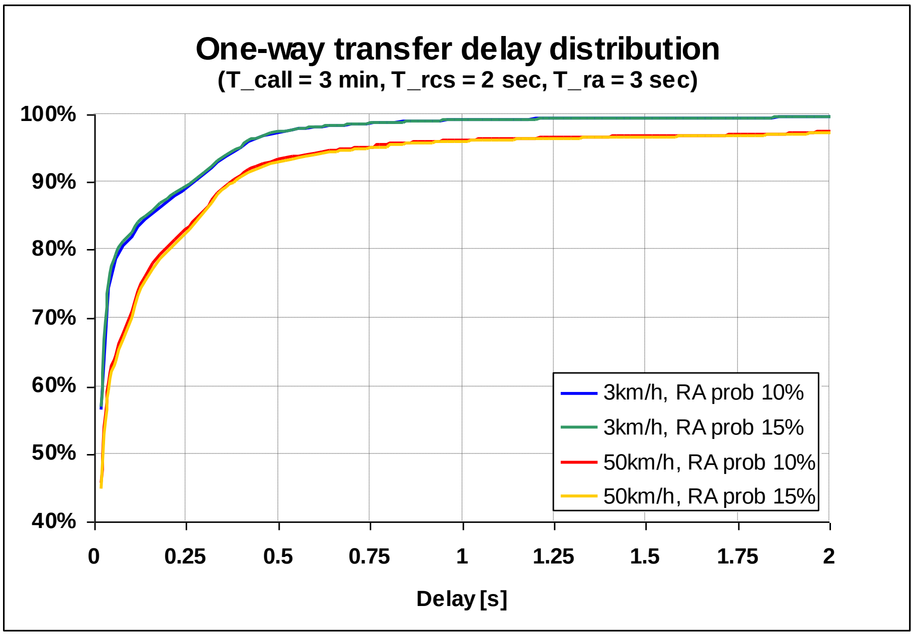

Figure 9. One Way Transfer Delay Distribution for RAU service outage of 3s

From the results we see that in 95% the transfer delay is well below 1 seconds. It is interesting to note that there is no big dependency on the RAU probability. Figure 10 presents the visualization of the transfer delay over the area.

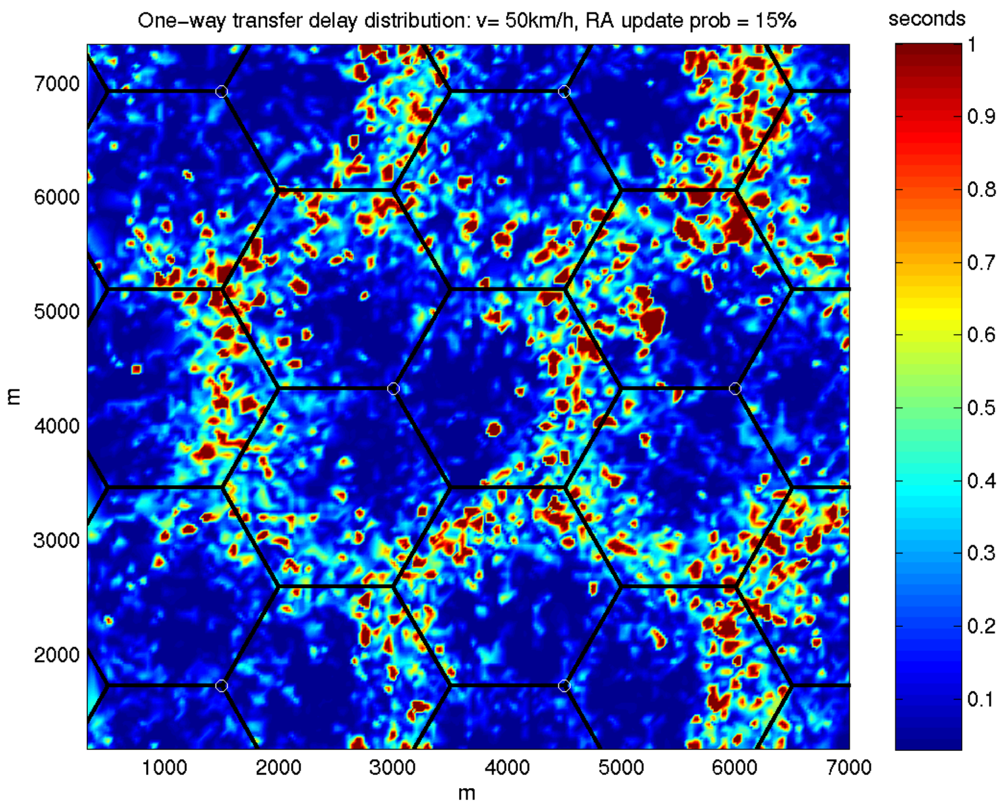

Figure 10. Visualization of One Way Transfer Delay Distribution for RAU service outage of 3s

6.5.3 Concluding Remarks and Recommendations

Results from the simulations under assumptions that were agreed in TSG GERAN WG2#13bis, Tdoc G2-030263, lead to the following conclusions (unlike in G2-030263, fixed outage times for Routing Area Update and Cell Change are assumed):

1. For 3 km/h and 50 km/h mobility model, PS streaming service over GERAN *A/Gb mode* could be successfully provided, i.e. a high number of satisfied users.

2. The RAU probability values that have been used in these simulations have no effect on the streaming performance in terms of zero-throughput periods and transfer delay.
   NOTE: The impact of RAU probability values on streaming performance in terms of packet loss during RA change is not agreed.

From these results it is recommended that:

a. It has to be ensured that Release 5 mechanisms of GERAN *A/Gb mode* indeed provide for RAU interruption of no longer than 3 to 5 seconds.

b. Packet loss during cell change (inter-BSS, when using unacked LLC) is minimized. The number of lost packets could be minimized either with proper scheduling mechanism in combination with flow control or when using enhancement presented in section 7.3 (SGSN Suspend procedure). Therefore the SGSN Suspend procedure should be specified.

c. Transfer delay of 2s is sufficiently well met by GERAN *A/Gb mode* in case of streaming service.

7 Reduction of Service Interruption Times and Packet Loss during Mobility Procedures

7.1 General

This clause provides an overview of the possible changes and additions required for the efficient support of streaming services in the GERAN specifications. Different solutions are provided. For each of them, impact on protocols and network entities should be outlined as well as the expected performance gain compared to the results in section 5.

7.2 Enhancement 1: Enhanced Flush Procedure (Packet Recovery)

This procedure is based on limiting packet loss at inter-BSS cell change for the case when unacknowledged LLC is used. The basic principle is that the BSSGP flush procedure is extended to also include an indication of the LLC PDUs, which has not yet been received by the MS, from the BSS back to the SGSN. This would make it possible for the SGSN to forward the IP packets, which the old BSS does not considered received by the MS to the new BSS (in the Inter-SGSN case this is done via normal packet forwarding procedure to the new SGSN).

Some re-transmission of packets may occur however if the procedure is working correctly no packets will be lost. (i.e. upon being flushed the old BSS may decide to return LLC PDUs to the SGSN that were actually already successfully delivered to the MS)

This solution has no impact on terminals and does not require any use of NACC, CCN and PS Handover. More detailed description has been presented in GP-023127, GP-020776 and GP-030200.

The current working assumptions is that there is no need to send the complete LLC PDUs back from the BSS to the SGSN. Instead some form of indication will be used. Either the BSS sends the UI frame number of the oldest LLC PDU that has not yet been received by the MS or the BSS just returns the first bytes of the oldest sent LLC PDU. The first solution requires the BSS to look into the LLC protocol.

The SGSN, who buffers all downlink packets for this particular flow, can then re-start the LLC transmission accordingly.

Following open issues can be subject to further investigation:

- The impact to the SGSN
- The duration that packets should be buffered in the SGSN.

7.3 Enhancement 2: Radio Status (SGSN suspend) procedure

This is an alternative solution to avoid packet loss at inter-BSS cell re-selection. It is used in combination with Cell Change Notification (used for NACC), or alternatively PS Handover.

When the BSS detects that the MS (in CCN mode) is planning to make a cell change to a cell in another BSS it will send a BSSGP Radio Status message with cause value "cell re-selection ordered" to the SGSN.

The SGSN will then temporarily stop the downlink transmission of LLC PDUs on the Gb interface. The BSS then has time to finish all downlink LLC PDUs it has in its buffer for that particular MS. At GERAN2 #12bis it was noted that this might require the BSS to order the MS into NC2 if the emptying of the buffer takes longer than one second. Possible enhancements not involving NC2 can also be considered.

When the buffer is empty the BSS can let the MS go to the target cell (either automatically or by sending a PCCO). The MS will perform a cell update when it enters the target cell. The cell update will be considered as an implicit resume by the SGSN and it then will start the downlink transmission in the target cell.

At GERAN2 #12bis the case when the PCCO fails and the MS enters the old cell was also discussed. As it is specified today the SGSN has no direct way of knowing that the PCCO failed since the MS is not required to perform a Cell update procedure or may not have any uplink data to send. This failure case can be solved if the MS is required to always perform a Cell update after receiving a PCCO. For more details see GP-030211.

In GP-030211 it is also proposed that this solution shall be introduced for cell changes towards UTRAN. This includes modifying the CCN procedures to also support CCN towards UTRAN. So far no fundamental problems have been raised with this proposal however the gains of introducing this needs to be analyzed further.

7.3.1 Investigation on the achieved gains using the SGSN Suspend

According to document GP-030781 presented in 3GPP TSG GERAN#14, performance of DL suspend solution for Streaming QoS was simulated and the summary of the results indicated loss less behavior and can be considered feasible in most cases. The document presents further studies on the solutions called BSSGP Radio Status/SGSN suspend. It studies the performance of emptying the BSS buffer further by analyzing a worst-case scenario on how much data will be in the BSS buffer prior to cell change. From the results, the BSSGP Radio Status/SGSN suspend solution can be considered feasible in most cases.

7.4 Enhancement 3: PS Handover

PS Handover is feature that lets the network support an almost seamless cell change (as seen by the application) by pre-allocating radio resources and then ordering the MS to go directly to a traffic channel in the target cell. The feature is mainly applicable to streaming and conversational services with high requirements on transfer delay. It may get initiated by the network based on RF criteria as measured by the MS, or by the network based on traffic criteria. (e.g. current traffic load per cell, interference levels, maintenance requests, etc.).

This feature will reduce the service outage time down to < 200 ms. It will be possible to steer mobiles based on load and services as well as reserve resources for mobiles in the target cell. This increases the possibilities to support guaranteed bit rate services significantly. PS handover provides benefits both for lossy and loss-less services and TCP time outs also can be kept to a minimum.

PS Handover can be used in combination with other enhancements described in this section.

7.5 Enhancement 4: MS Controlled restart

The basic principle for this solution is that the MS informs the SGSN in target cell which packets it has received correctly in the old Cell. The SGSN, who buffers all downlink packets for this particular flow, can then re-start the transmission from the first LLC PDU that did not reach the MS in the old cell.

In GP-020200 various variations of this solution was presented. Below is a summary of the preferred solution.

In this case the MS informs the SGSN about the status of LLC receiver by including the indication of the "Last Received UI Frame" when performing Cell Updates and when the MS performs Routing Area Updates it informs the SGSN about the SNDCP status by indicating the "Last Received N-PDU sequence number".

After receiving such information, the SGSN will send the FLUSH message to the old BVCI. If it will be informed (by the FLUSH ACK) that LL-PDUs were transferred (i.e BSS re-routing is possible), it will neglect the information coming from the mobile, otherwise it will start transmitting towards the new BVCI starting from "Last Received UI Frame"+1 in case of cell update and from the "Last Received N-PDU" +1 in case of RAU.

This solution moves any impact from the BSS to the MS, and could be more accurate. In this case, the MS provides a sort of acknowledgement, with the difference that it provides information only on the last received frame (and not on the frames possibly lost in between) so that no (possibly useless) retransmission is performed. In this way the packet transmission can be resumed exactly from where it was suspended in the old cell.

Inter-SGSN cell changes could also be handled by re-using procedures foreseen for LLC Ack mode. The new SGSN retrieves all the N-PDUs buffered at the old SGSN and then starts the transmission in the new cell from the "Last Received N-PDU sequence number".

Following open issues can be subject to further investigation:

- The impact to the SGSN.
- The impact to the MS

7.6 Enhancement 5: Sending of Downlink Data During RAU

As specified in [12] up to REL-5 it is not allowed that the SGSN and the MS sends data when the P-TMSI is changed during the RAU update procedure between the RAU Accept and the RAU Complete message. From REL-6 upwards the SGSN is allowed to send Downlink data during the RAU procedure between the RAU Accept and the RAU Complete message since the restriction was removed in [12]. More information is given in G2-030047, which shows that downlink data transfer is possible (see the figure below) and it is recommended that at least for streaming services to make use of that. The data transfer can be started as soon as the SGSN contexts are available and security functions have been executed.

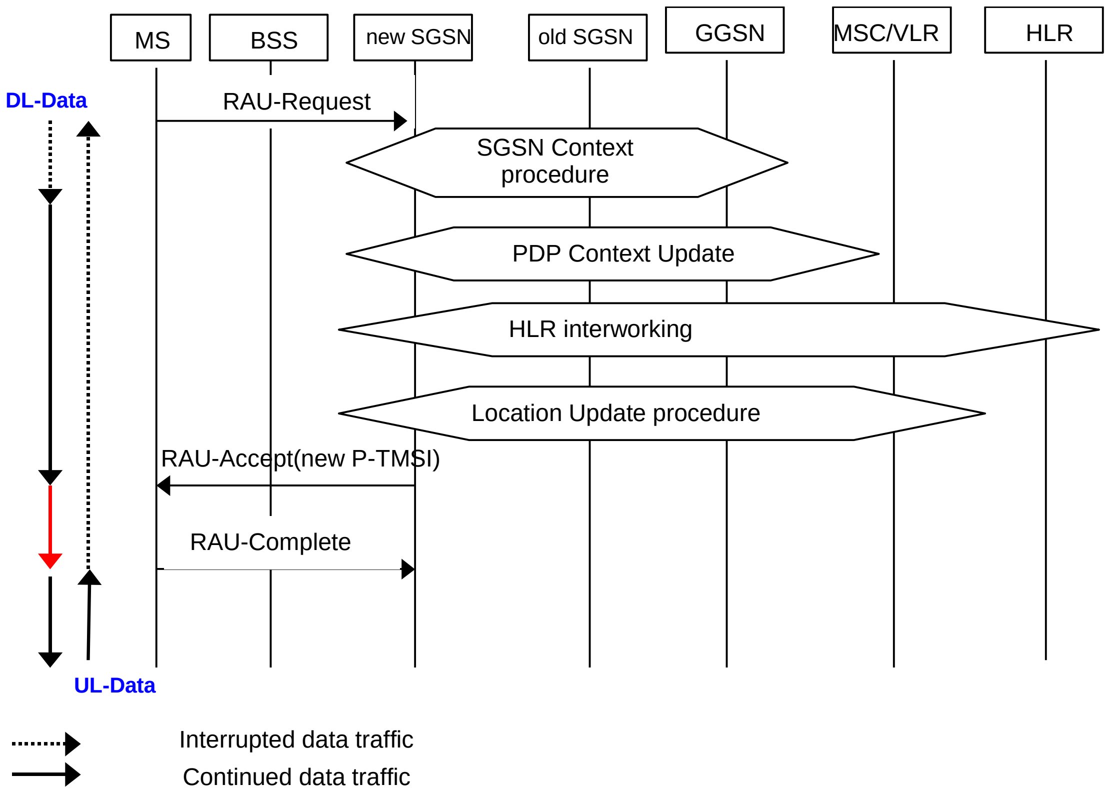

Figure 11: Inter-SGSN RAU and corresponding UL/DL-data traffic according to proposal

8 Support of header compression

IP header compression for GERAN is provided by SNDCP in the MS and the SGSN. SNDCP currently supports the following header compression schemes:

- RFC 1144
- RFC 2507
- RFC 3095

For more details on the use of header compression schemes see 3GPP TS 44.065.

Annex A (informative): Change history

Change history							
Date	**TSG #**	**TSG Doc.**	**CR**	**Rev**	**Subject/Comment**	**Old**	**New**
2002-11	G #12	GP(02)3373			Initial version provided.		0.1.0
2003-01	G2 #12bis	GP(03)0057rev2			Update with revised version of GP(03)0057after email approval (27.1.2003)	0.1.0	0.2.0
2003-02	G#13	Email			Inclusion of GP-030201 and GP-030323	0.2.0	0.3.0
2003-03	G#14	GP(03)0654			Note 4 in section 5.2 put FFS Clause 6.3 corrected	0.3.0	0.4.0
2003-05	G2#14bis	Email			Linkage between Transfer Delay and Buffer size removed	0.4.0	0.5.0
2003-05	G2#14bis	Email			Inclusion of agreed sections of G2-030309 and G2-030328 and additional clarifications according to G2#14bis meeting Editorial revisions and numbering of TR	0.5.0	0.6.0
2003-06	G#15	GP(03)1598			Inclusion of GP-031085	0.6.0	0.7.0
2003-07	G#15	Email			Inclusion of GP-031193	1.0.0	1.1.0
2003-09	G#16	Email			Inclusion of GP-031862 and GP-032034 from GERAN#16	1.1.0	1.2.0
2003-09	G#16	Email			Update of section 6.5.3	1.2.0	1.3.0