(S4-020227, copy TSG-RAN) LS on WCDMA reference bearers for streaming

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Title:	**LS on WCDMA reference bearers for streaming**
To:	**TSG RAN WG1, TSG RAN WG2**
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SA4 greatly acknowledges the work of RAN1 and RAN2 in specifying radio access bearers for various 3G services.

SA4 would like to inform RAN1 and RAN2 about simulation results for radio access bearer configurations, which are appropriate for Rel4/Rel5 packet-switched streaming services (PSS).

Rel4/Rel5 PSS does not contain any means for recovering lost RTP packets on the IP layer. In addition, both Rel4 and Rel5 have only limited support for content adaptation to varying link conditions. A suitable radio access bearer for Rel4/Rel5 streaming should therefore be characterized by low residual SDU losses and by fairly stable throughput behaviour in downlink direction.

At the TSG-SA4#20 meeting in Luleå simulation results for streaming over a radio access bearer using a dedicated channel and RLC running in acknowledged mode were presented. The results are included below in Annex A.

The results show that the quality-of-service provided by radio access bearers using a dedicated channel and RLC running in acknowledged mode is appropriate for PSS Rel4/Rel5 streaming services and that streaming applications can deal with the additional delay jitter caused by the RLC retransmission mechanism.

We kindly ask RAN1 and RAN2 to take those results into account when specifying suitable reference bearers for streaming.

Dates of Next SA4 Meetings:

13 – 17 May 2002	TSG-SA WG4#21	Host: France Télécom R&D, at Rennes (F)
22 – 26 July 2002	TSG-SA WG4#22	Host: Nokia (tbc)

Annex A.

Case study: Use of DCH with RLC Acknowledged Mode

For UTRAN, a Radio Bearer using a dedicated channel and RLC running in acknowledged mode could fulfil the requirements of recovering from lost RTP packets and having a fairly stable network throughput behaviour. First of all, a dedicated channel can maintain a fixed transport channel rate on the physical layer. Secondly, when used in acknowledged mode, the probability of lost IP packets is close to zero due to an efficient retransmission protocol on the RLC layer, which retransmits only the erroneous PDUs of an IP packet (note that a PDU corresponds to a small fragment of an IP packet). The increase in IP packet delay jitter caused by this RLC retransmission mechanism is acceptable for streaming services as will be shown in the following. Figure 1 shows simulations results for a UTRAN bearer in acknowledged mode with a configuration as listed in the figure. The PDU size (80 bytes) indicates the RLC block size. In addition, the figure shows only the first 15 seconds of the transmission simulation. We used the Glasgow video sequence encoded using a constant quantizer (TBC). The streaming client buffer size was set to 20000 bytes. The bitrate generated by the streaming server was limited to 58 kbps, about 10% less than the network bit rate to allow retransmission of lost RLC blocks. The maximum number of RLC retransmissions was set to be theoretically infinite (persistent retransmission). The average packet size in this example was 628 bytes (including headers).

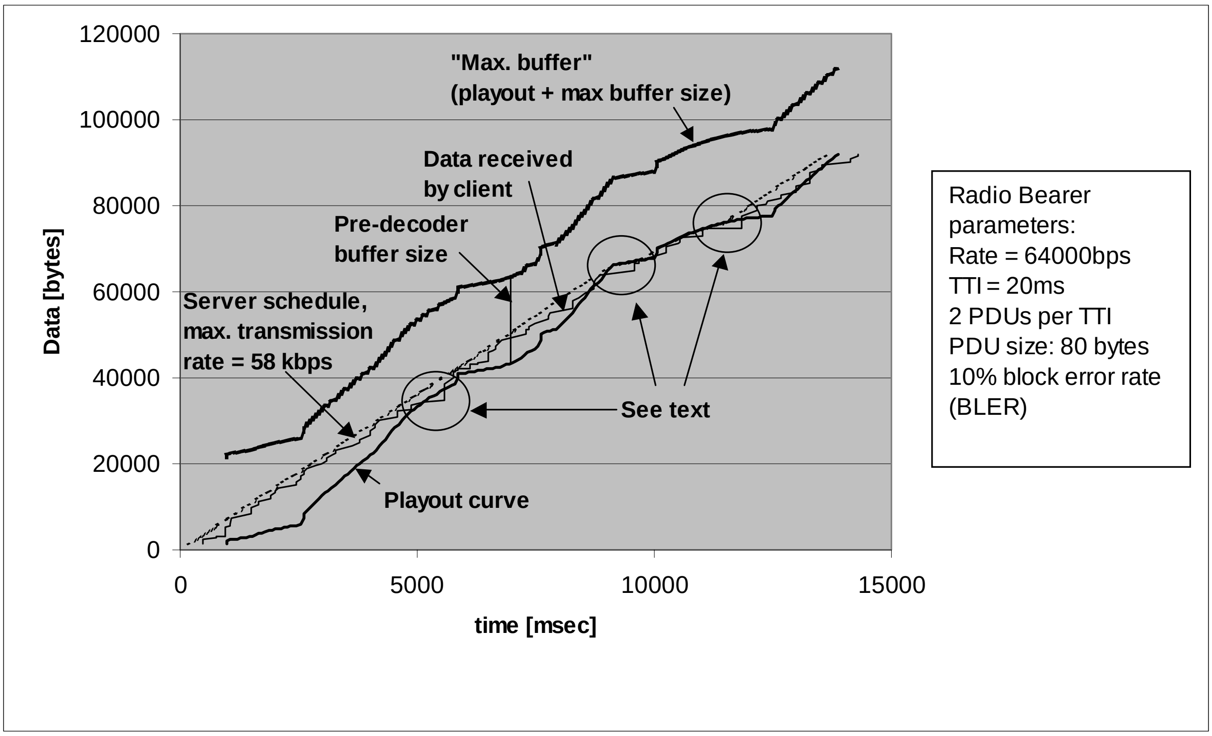

Figure 1 - Impact of the delay jitter introduced by a DCH with RLC AM on streaming playout performance.

The horizontal axis denotes time in milliseconds; the vertical axis denotes an overall amount of data in bytes. The playout curve shows the minimum amount of data that needs to be available at the decoder for smooth playout. As one can see, playout starts after an initial buffering delay of 1 second, which is needed in this example to play out the stream smoothly.
The “Max buffer” curve represents the maximum amount of bytes that can be stored at the decoder before a buffer overflow occurs. This curve is simply a vertically shifted version of the playout curve. The value by which the curve is shifted represents the client buffer size.

Between the playout and the “max buffer” curve there are two additional curves. The first one represents the amount of data as sent out by the server. The second curve represents the amount of data that is received by the client after transmission over a simulated bearer using RLC AM. Note that the curve representing the amount of data sent out by the server must not cross either the playout or the max buffer curve. Crossing the playout curve would result in a buffer underflow, which leads to a playout interruption. Crossing the “max buffer” curve would result in a buffer overflow, which leads to data losses.

The output stream of the constant quality encoder was smoothed by a traffic smoother. The traffic smoother makes sure that the maximum transmission rate of the video stream is not higher than the maximum channel capacity. Secondly it computes a schedule that minimizes the receiver buffer size by transmitting packets as late as possible (in the literature this is referred to as ‘late scheduling’ in contrast to ‘early scheduling’ where packets are sent as early as possible).

By looking at the amount of data received by the client after transmission over a simulated bearer in acknowledged mode, one can see that the delay jitter introduced by the bearer would lead to buffer underflows. In the example this happens around second 6 and 10. We want to point out that the observed maximum number of RLC retransmissions was less than or equal to 4.

To accommodate for the delay jitter, the playout curve needs to be shifted to the right (= increase in initial buffering delay) by the maximum delay introduced by the bearer. In the given example, this maximum delay was around 1 second. At the same time the buffer needs to be increased by the number of bytes that are transmitted at the maximum transmission rate during 1 second. For a 64 kbps bearer this means 8000 bytes. However, from looking at the curve, one can see that by applying a more intelligent schedule both the additional buffering time and also the additional buffer size could be further reduced. The figures presented here do not consider any further optimisations and therefore reflect a worst-case scenario.

Figure 2 shows the cumulative distribution function (C.D.F.) for the packet delays. As can be seen, in 95% of the cases the delay of a packet is less than one second.

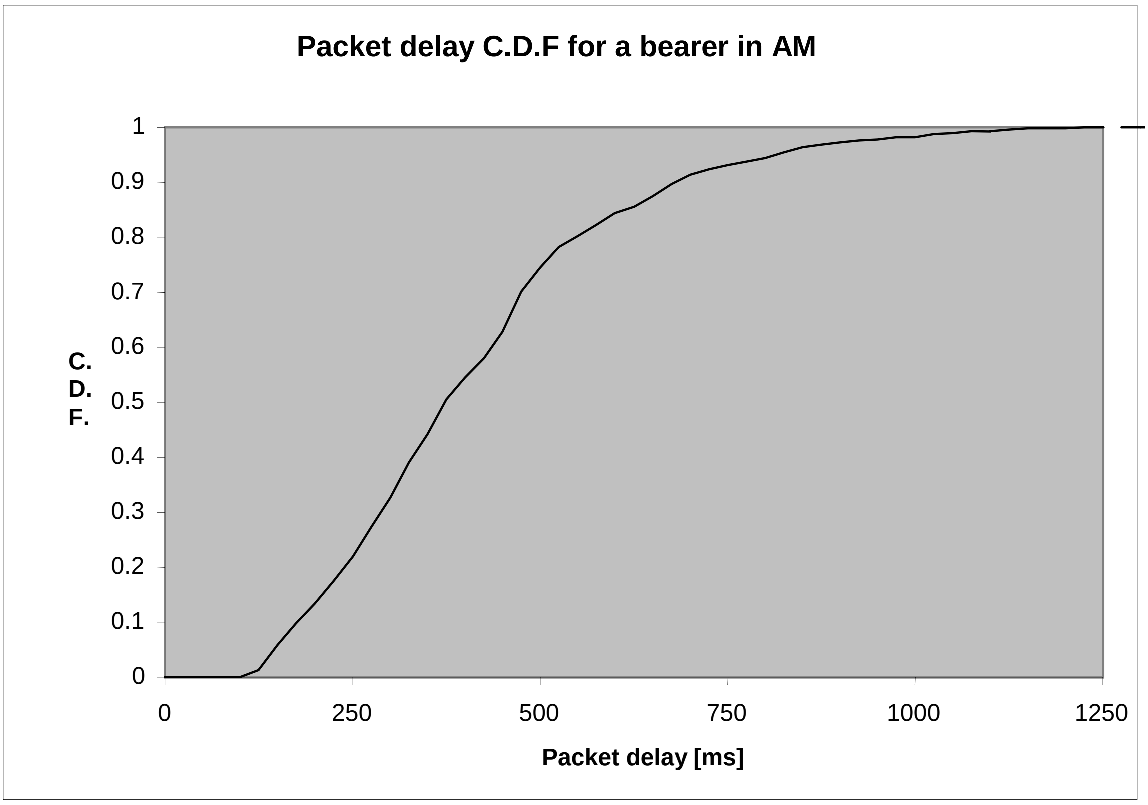

Figure 2 - Simulated packet delay C.D.F. for DCH using RLC AM