TSG Services and System Aspects **SP-050159**

Meeting #27, 14 - 17 March 2005,Tokyo, Japan **Agenda 7.4**

3GPP TSG RAN Meeting #27 RP-050153
Tokyo, Japan, 9 - 11 March 2005

Title: **Joint meeting RAN4-SA4 on MBMS (Sophia Antipolis, 4-6 April 2005) : guidance for simulations and error patterns for the audio codec characterisation tests**

Source: TSG RAN

To: TSG SA, TSG RAN WG4, TSG SA WG4, TSG GERAN

Attached: S4-040803

Contact Person:
Name: Paolo Usai
Tel. Number: + 33 4 92 94 42 36
E-mail Address: Paolo.Usai@etsi.org

1. Introduction

At TSG#26 meeting in December 2004, SA4 asked TSG RAN and TSG GERAN guidance for simulations and the provision of adequate error patterns to complete the MBMS audio codec performance characterisation phase 2 of (subjective) testing.

At their last meeting#23 in Tampa (24-28 January 2005), TSG GERAN replied that the provision of adequate error patterns for MBMS simulations using GPRS/EGPRS was left to the action of individual (and interested) Companies, and at last SA4#34 meeting in Lisbon (21-25 February 2005) at least one Company committed to this purpose (and simulation results are expected to be provided at next SA4#35 meeting in early May 2005).

TSG RAN#26 replied to SA4 (in Liaison Statement RP-040546=TSGS#26(04)0893=S4-050009) that

> *The MBMS performance requirements defined by TSG RAN WG4 will not be completed before September 05, and this work is linked to the request from SA4*

SA4 needs the information well before September 2005 to complete the MBMS User Services Release 6 feature, and suggested (in Liaison Statement RP-050105=S4-050214) a discussion at the RAN4 MBMS ad-hoc Meeting on April 4-6, 2005. However, on a specific request raised by the SA4 Secretary during SA4#34 to avoid the repetition of a few previous (negative) experiences (i.e. no show of delegates ...), **no SA4 Company could commit to attend the Joint meeting RAN4-SA4 on MBMS**, and **THIS IS THE MATTER of CONCERN !**

Technical aspects

SA4 believes that the FEC on application layer should be part of the overall performance studies, and ROHC should also be taken into account in case it is specified for Rel-6 (which has been confirmed at TSG RAN#27). After the selection of an FEC code, still a contentious issue between two solutions, i.e. Raptor and RS codes, SA4 will provide the information required to perform simulations on the application layer (in case no decision occurs at TSG SA#27 Plenary on FEC, both candidate codes should still be considered for the simulations).

SA4 informed TSG RAN and TSG SA (in Liaison Statement RP-050105=S4-050214 and in TSGS#27(05)0132) that SA4 has made a working assumption to use the approach described in the attached Tdoc S4-040803, which was produced in origin to study the quality of multimedia services (video, audio, etc.) when delivered over IMS. The proposed methodology includes two parts: (1) Generation and use of link level PDU loss patterns for 3GPP bearers and (2) a network simulator to introduce packet losses in RTP streams.

SA4 asked RAN to review the link level PDU error masks (in Section 2.1 and Appendix C of S4-040803) and advise SA4 on the applicability of these masks. This matter can de discussed at the Joint meeting RAN4-SA4.

SA4 asked RAN (and GERAN) to review the RTP Packet Loss Simulator (in Section 2.2 and Appendix A of S4-040803) and advise SA4 on the applicability of this approach to introduce packet losses in multimedia streams delivered over RTP. This matter can de discussed at the Joint meeting RAN4-SA4.

2. Action

Companies attending TSG SA are kindly requested to assure that RAN4 and SA4 delegates will attend the Joint meeting on MBMS on April 4-6, 2005, at Sophia Antipolis, France.

3GPP TSG-SA-WG4 #33 Meeting **Tdoc S4-040803**
November 22-26, 2004, Helsinki, Finland

Agenda Item : 6.5

Source : Qualcomm Europe S.A.R.L., Toshiba Corporation

Title : Video Network Simulator and Error Masks for 3GPP Services

Document for : Information

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1 Introduction

This contribution presents error masks generated from link level simulations at various bearer rates and BLER values. These masks are expected to be used to characterize video services in 3GPP.

This contribution also proposes a method for introducing packet losses on 3GPP networks for RTP/UDP/IP bitstreams. It is based on the VCEG network simulator [1], adopted by VCEG during the development of H.264/AVC codec. This simulator primarily addresses packet losses for video streams over RTP/UDP/IP transport. However, it can be used for other data also when delivered over RTP/UDP/IP transport. Two modifications are introduced to (i) enable use of packet error masks and (ii) enable the simulator to perform correctly when lower layers support multiple PDU sizes for a given TTI [2].

2 Network Simulation software for 3GPP video applications

2.1 Packet error masks for 3GPP SA4 simulations

ASCII packet error masks for MBMS, PSC and PSS are summarized below. The error masks are a sequence of ‘1’s and ‘0’s. ‘0’ indicates that the PDU is not lost and ‘1’ indicates that the PDU is lost. These error masks can be used to inject errors at the physical layer. For every PDU transmitted one mask is read from the error mask file. If the mask is 0, the PDU is considered to be received successfully else the PDU should be discarded (i.e., not given to the receiver/video decoder). Details of generation of the error masks are in Appendix C. List of the error mask file are in Appendix B. The files contain one mask value in ASCII per line.

Table 1 Naming convention of the error masks

Naming Convention	[MBMS|PSC]_bitrate_TTI_BLER-Value
MBMS	Masks for MBMS simulations
PSC	Masks for PSC/PSS simulations (with power control)
Bitrate	16, 32, 48, 64 and 128 kbps
TTI	80ms for MBMS and 20 ms for PSC/PSS
BLER	1.5%, 1% and 0.5%

The PDU size in octets is calculated as

PDU_size_in_octets = (bitrate / 8) * TTI

e.g. MBMS_128kbps_80ms_BLER_0_5 PDU size = (128/8) * 80 = 1280 octets

PSC__64kbps_20ms_BLER_0_5 PDU size = (64/8) * 20 = 160 octets

For PSC services, the total BLER should be twice that of PSS, to account for uplink and downlink packet losses. To simulate end-to-end video quality, one needs uncorrelated error masks for uplink and downlink. In this situation, the error masks are reversed and a "logical OR" operation is performed with the forward masks. This will result in error masks of BLER=3%, 2% and 1%.

Figures 1—6 show the distribution of burst errors presented here. It can be seen that only PSC__XXXkbps_20ms_BLER_0_5 appears a little different for different kbps This is because PSC__128kbps_20ms_BLER_0_5.txt has burst errors of lengths of 1, 2, 3, and 4 while the others have burst errors of only lengths 1 and 2.

As a first approximation, it appears reasonable to assume that burst statistics do not vary much with the bearer rate. Consequently, if the video rate changes due to rate adaptation, the same error mask is applicable to all data rates to assess the effects of packet loss .

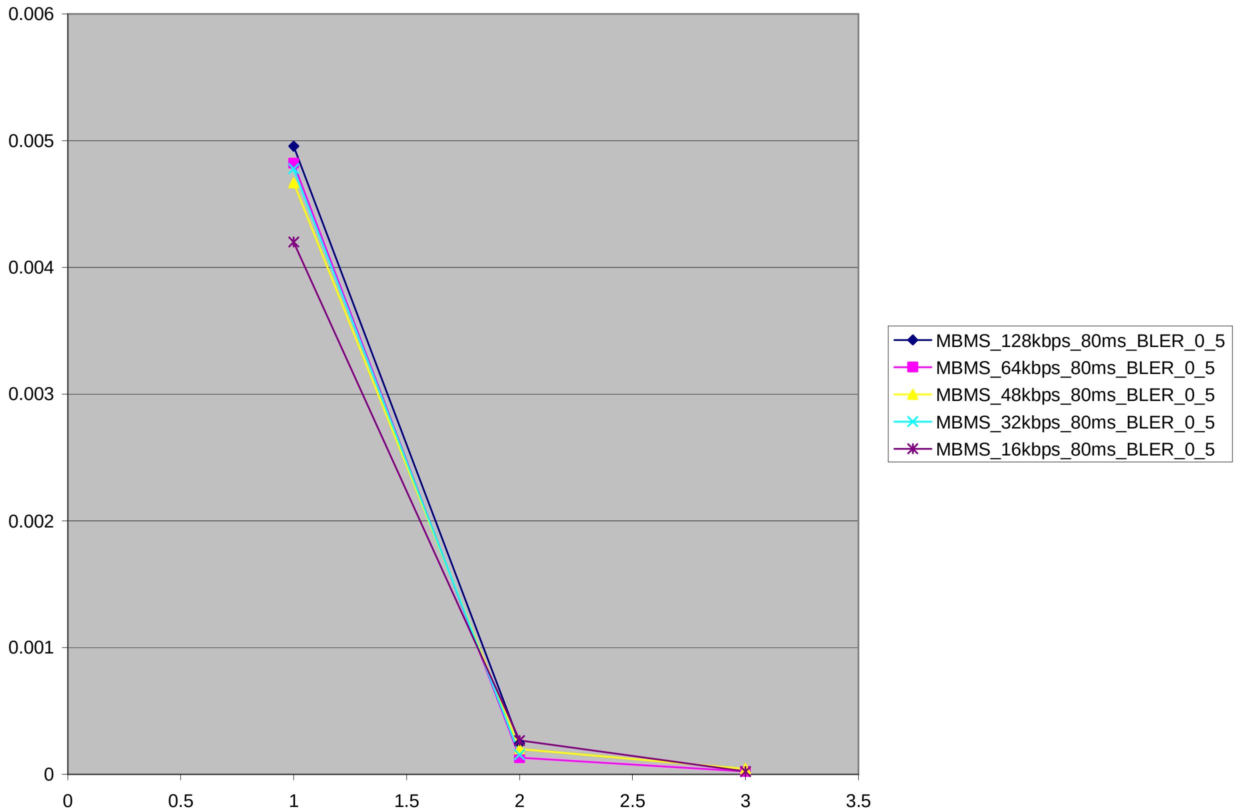

Figure 1 MBMS Burst Error Distribution at 0.5% BLER

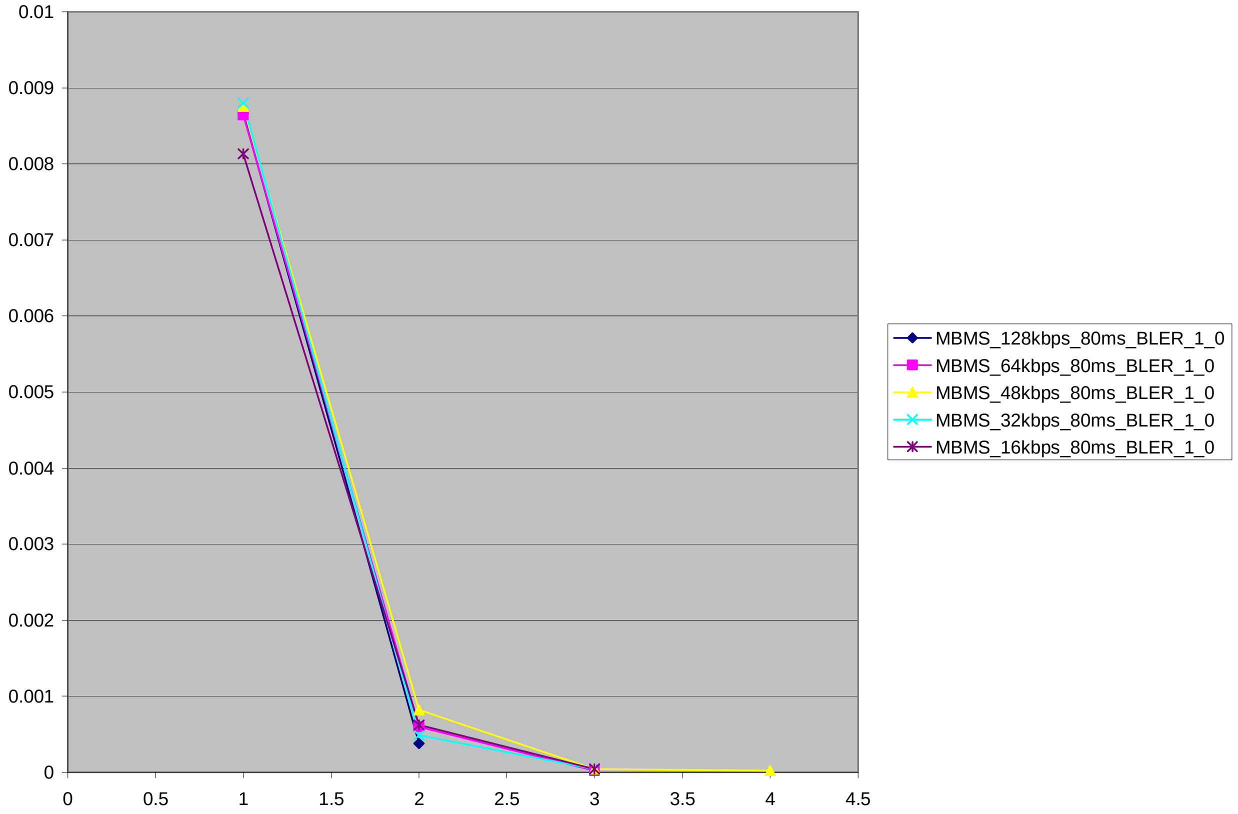

Figure 2 MBMS Burst Error Distribution at 1% BLER

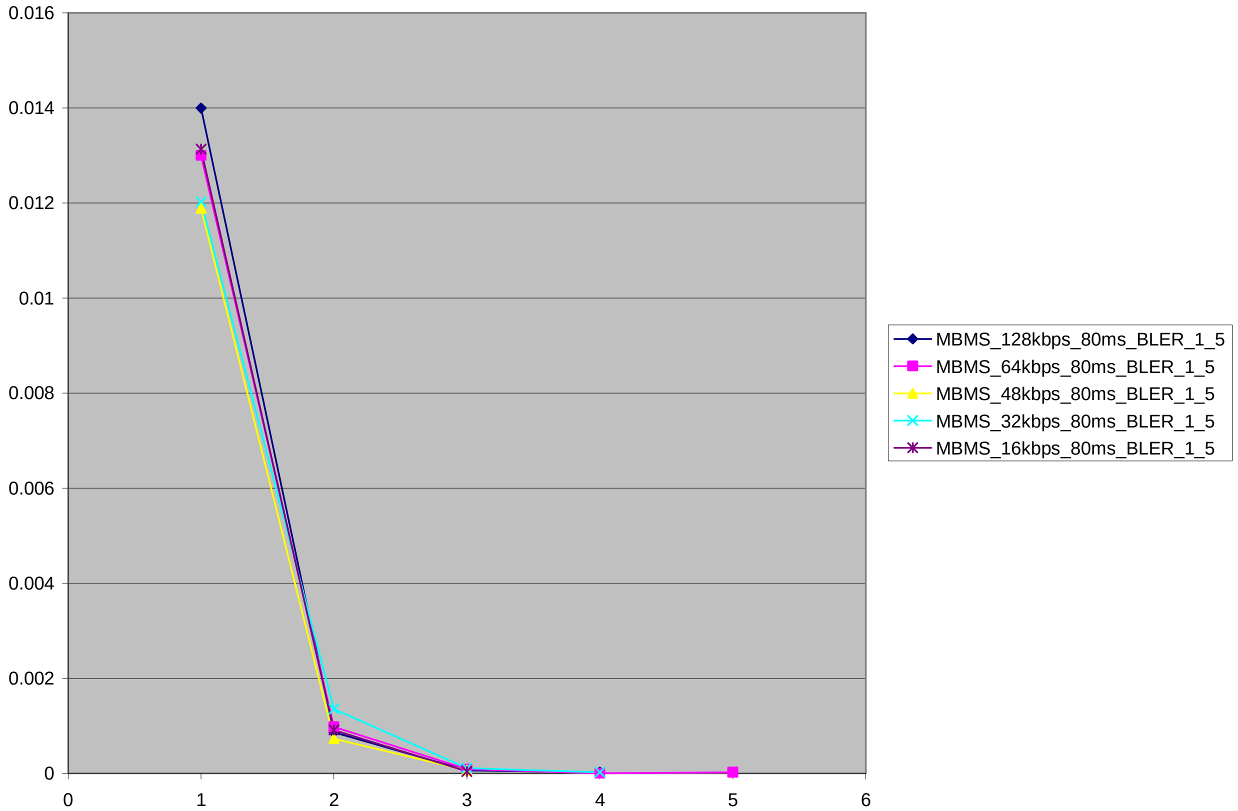

Figure 3 MBMS Burst Error Distribution at 1.5% BLER

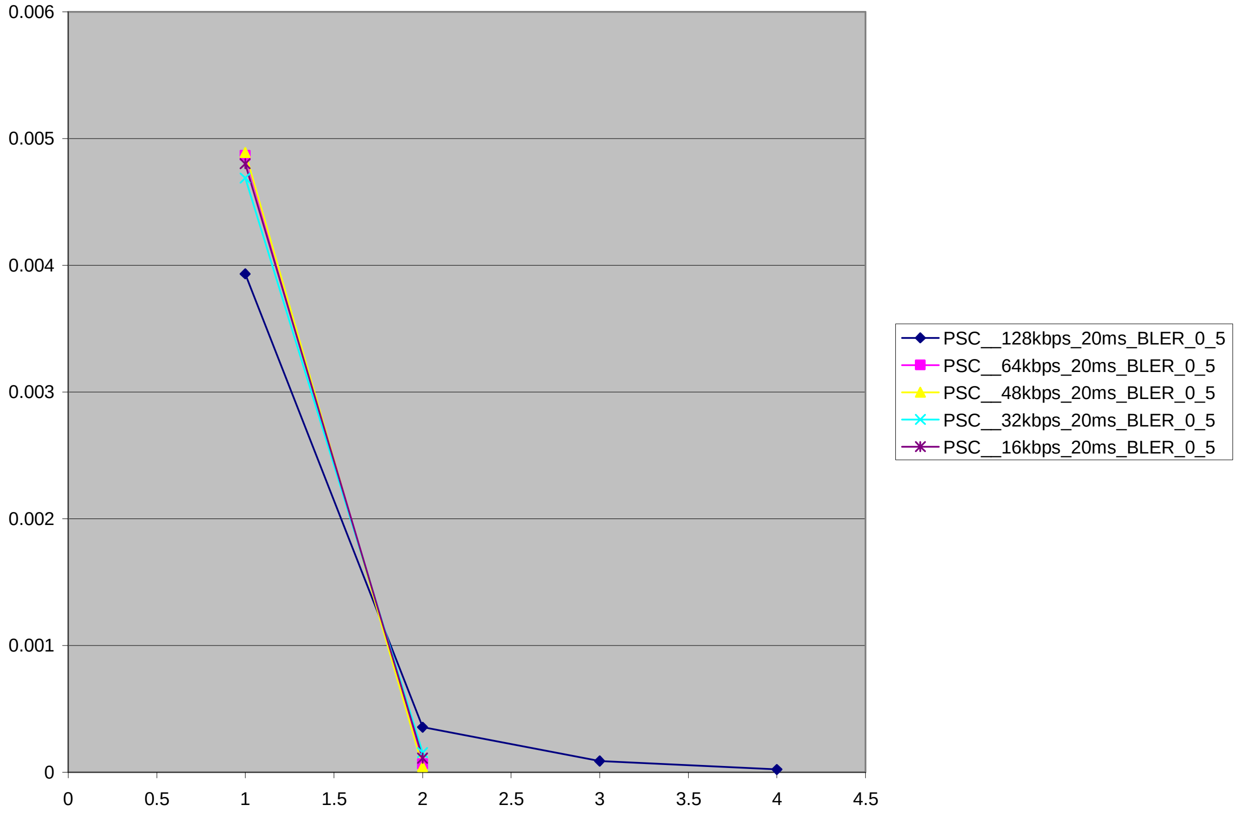

Figure 4 PSC/PSS Burst Error Distribution at 0.5% BLER

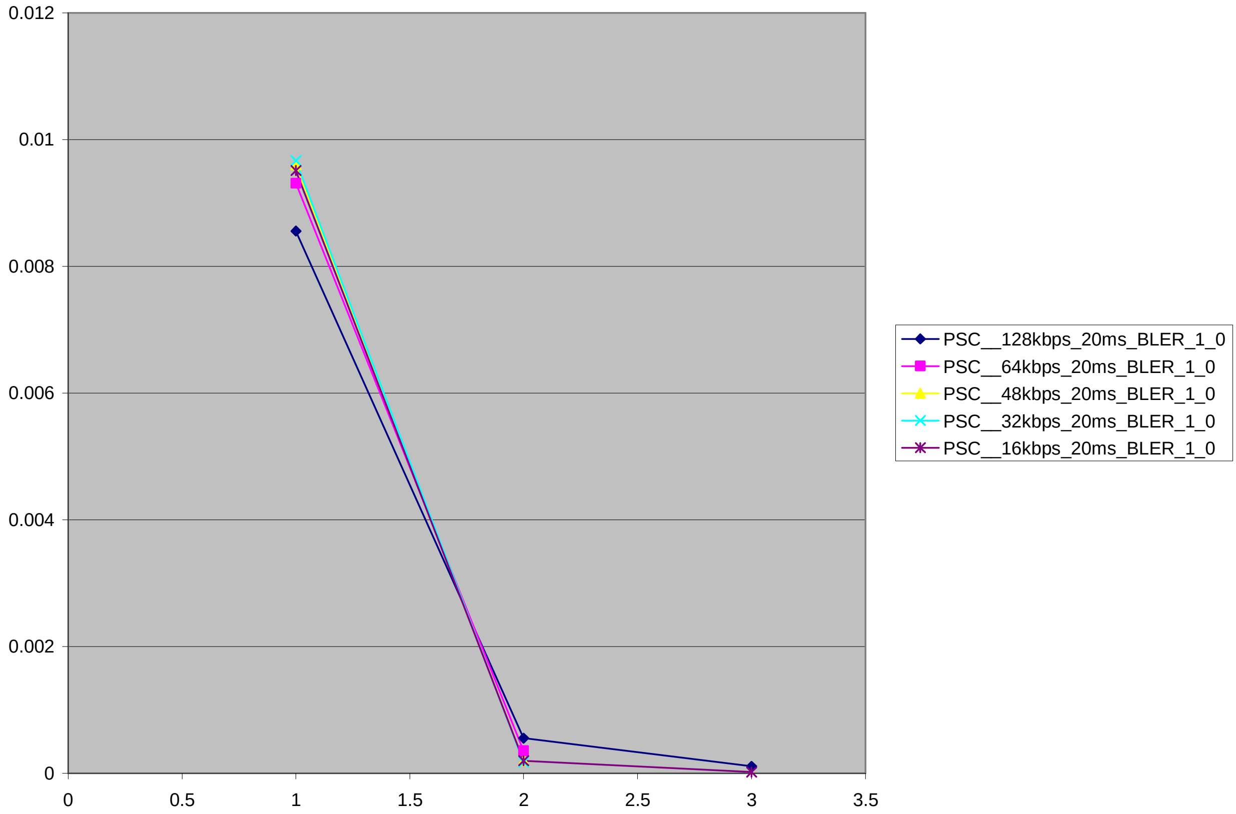

Figure 5 PSC/PSS Burst Error Distribution at 1% BLER

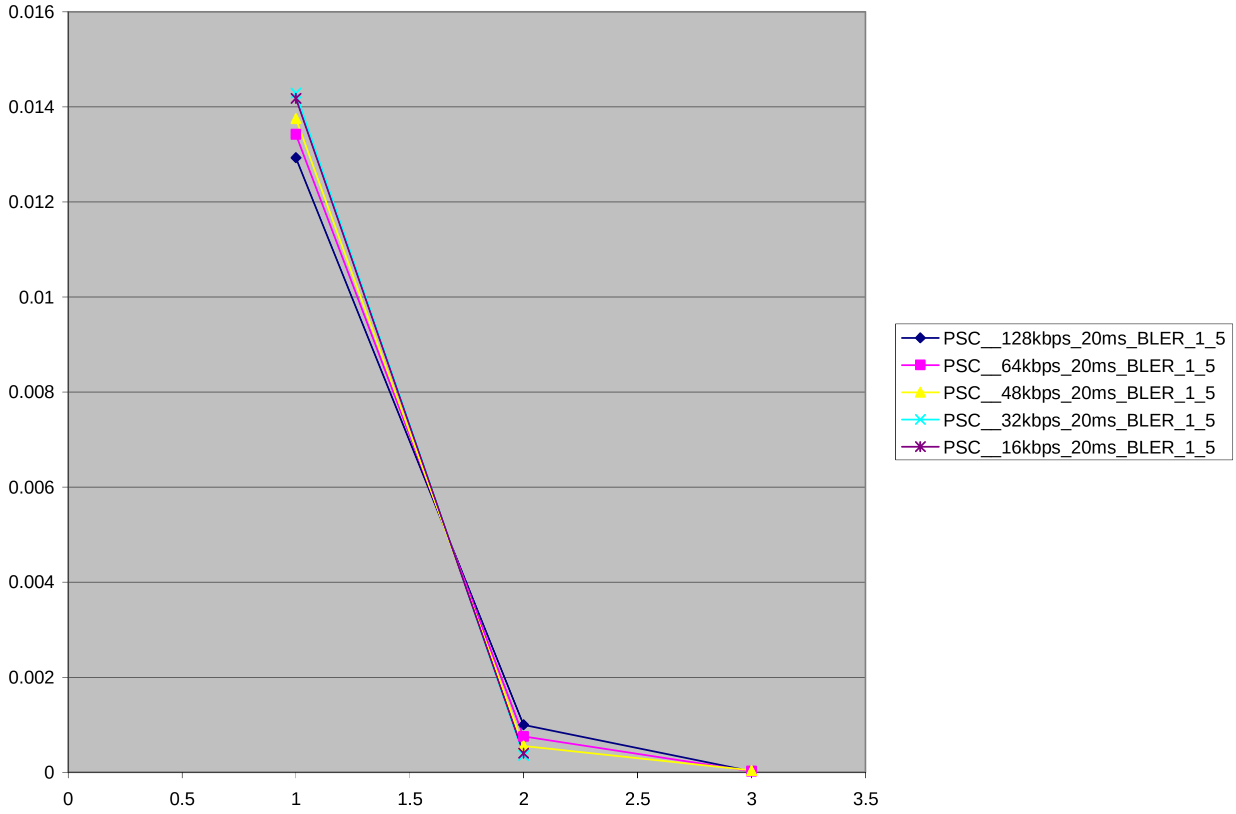

Figure 6 PSC/PSS Burst Error Distribution at 1.5% BLER

2.2 Network simulator for 3GPP packet networks

The network simulator is a modified version of the simulation software developed by VCEG [1] for video applications based on RTP/UDP/IP.

2.2.1 Frame dropping modification

The original simulator required bit error masks, i.e., the sequence of ‘0’ and ‘1’ indicated if a bit was in error or not. The error mask files were also required to be in binary format. An additional feature was added to the simulator enabling it to read ASCII error masks. Its operation was also modified so that the ASCII mask values were interpreted as corresponding to packet losses. Hence, the modified simulator only needs to read one mask for every PDU. If the mask is ‘1”, the PDU is dropped else it is not. When a PDU is in error the RTP packet(s) corresponding to it is(are) dropped (i.e., not written to output file). This operation is shown in Figure 1. As before if the end of file is reached and there are more PDUs to be transmitted, the simulator continues reading from the beginning of the error mask file.

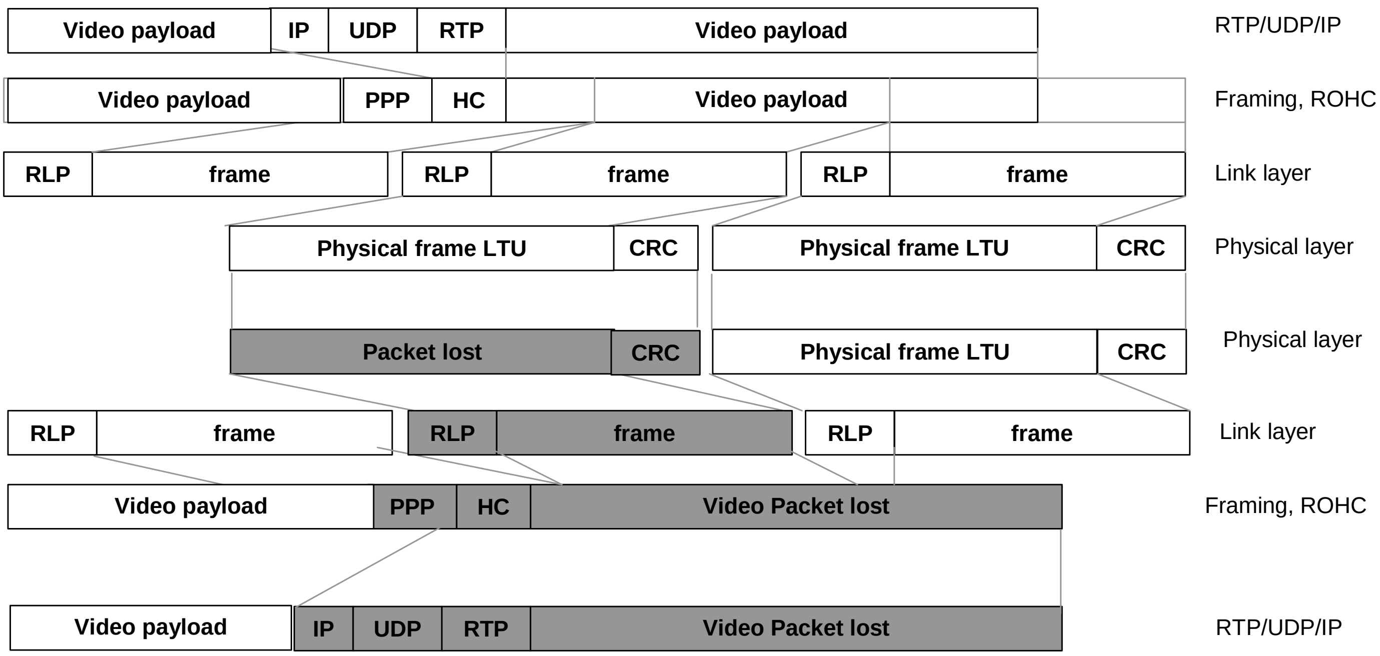

Figure 1 Network simulator operation. The dropped portions of the bitstream at different layers are shown shaded.

2.2.2 Support for Multiple PDU Sizes

For the original simulator only one PDU size needed to be specified. All RTP packets were broken up into these equal size PDUs. In EBR mode of operation a video encoder generates RTP packets corresponding to one of several fixed available PDU sizes [2]. Hence a RTP packet is completely contained in a single PDU. To enable this mode of operation all possible PDU sizes need to be specified

When multiple PDU sizes are specified, each RTP packet is expected to be transmitted in a single PDU. One mask is read for each RTP packet and the smallest PDU that can contain the entire RTP packet is considered lost over the physical layer.

In EBR mode the PDUs themselves provide framing information, hence PPP is not required for EBR. Hence the packet size expansion feature due to PPP has been turned off in the network simulator.

The modified VCEG network simulator is presented in Appendix A.

3 Conclusions

This contribution presents network simulator and packet error masks for characterizing video codecs in 3GPP services. Packet error masks for different conditions were documented. Modifications to VCEG network simulator required to use these error masks were mentioned. Also, the modifications to enable the use of simulator for channels that support multiple PDU sizes were presented.

4 References

[1] VCEG-M77.doc
[2] Tdoc S4-040477, “Video delivery on 3GPP bearers for low delay applications”

Appendix A

The VCEG software simulator along with modifications is included in the attached zip file

Appendix B: Error masks

The attached zip file contains the following error masks

Table 2 List of all error masks

MBMS_128kbps_80ms_BLER_0_5.txt
MBMS_128kbps_80ms_BLER_1_0.txt
MBMS_128kbps_80ms_BLER_1_5.txt
MBMS_16kbps_80ms_BLER_0_5.txt
MBMS_16kbps_80ms_BLER_1_0.txt
MBMS_16kbps_80ms_BLER_1_5.txt
MBMS_32kbps_80ms_BLER_0_5.txt
MBMS_32kbps_80ms_BLER_1_0.txt
MBMS_32kbps_80ms_BLER_1_5.txt
MBMS_48kbps_80ms_BLER_0_5.txt
MBMS_48kbps_80ms_BLER_1_0.txt
MBMS_48kbps_80ms_BLER_1_5.txt
MBMS_64kbps_80ms_BLER_0_5.txt
MBMS_64kbps_80ms_BLER_1_0.txt
MBMS_64kbps_80ms_BLER_1_5.txt
MBMS_64kbps_80ms_BLER_10_.txt
PSC__128kbps_20ms_BLER_0_5.txt
PSC__128kbps_20ms_BLER_1_0.txt
PSC__128kbps_20ms_BLER_1_5.txt
PSC__16kbps_20ms_BLER_0_5.txt
PSC__16kbps_20ms_BLER_1_0.txt
PSC__16kbps_20ms_BLER_1_5.txt
PSC__32kbps_20ms_BLER_0_5.txt
PSC__32kbps_20ms_BLER_1_0.txt
PSC__32kbps_20ms_BLER_1_5.txt
PSC__48kbps_20ms_BLER_0_5.txt
PSC__48kbps_20ms_BLER_1_0.txt
PSC__48kbps_20ms_BLER_1_5.txt
PSC__64kbps_20ms_BLER_0_5.txt
PSC__64kbps_20ms_BLER_1_0.txt
PSC__64kbps_20ms_BLER_1_5.txt

Appendix C: Generation of PDU loss Error Masks

This section presents the methodology used to generate PDU loss error masks.

Modelling Methodology

Frame decoding error events are generated in a link-level simulation. A link-level simulation is run and the decoding successes of each TTI block are recorded in the form of ‘0’ and ‘1’ for each TTI, thereby producing an “error mask”. The error mask is then fed into the video simulation to model air interface errors. In those simulations all the bits carried in an application layer packet containing the PDU are discarded when the error mask indicated that the block is in error. This is typically one RTP/UDP/IP packet containing the block that was in error.

The error masks are generated with 16kbps, 32kbps, 48kbps, 64kbps and 128kbps. The MBMS services can be transmitted on physical channel S-CCPCH or DPDCH. The difference between S-CCPCH and DPDCH is given as follows:

- S-CCPCH: No power control is assumed and the Node-B is sending with constant power. Single transport channel mapping is assumed, i.e., all bits on S-CCPCH are used for MBMS. Spreading factor and number of symbols per slot are given in Table 1.

Table 1 S-CCPCH channel parameters

Rate	Spreading Factor	Slot Format	S-CCPCH Bits/80ms TTI
16kbps	128	6	4560
32kbps	64	8	8640
48kbps	32	10	18240
64kbps	32	10	18240
128kbps	16	12	37440

- DPDCH: Both inner loop and outer loop power control are enabled. MBMS services are mapped to DTCH, while signalling messages are sent on DCCH. DTCH and DCCH are physically transmitted on DPDCH. In this simulation it is assumed that DCCH is always present and DCCH rate is 3.4kbps as given in 34.108[1]. Rate matching attributes for DTCH and DCCH are assumed to be the same, in other words, the code rate of DTCH and DCCH are the same.

Table 1 DPDCH channel parameters

Rate	Spreading Factor	DTCH Bits/20ms TTI
16kbps	64	766
32kbps	64	1592
48kbps	32	3861

64kbps	32	3939
128kbps	16	8361

The channel model is case 2 channel from 25.101[2]. The channel profile is given in Table 3.

Table 3 Propagation Channel Models

Case 2, speed 3 km/h	
Relative delay [ns]	Relative mean power [dB]
0	0
976	0
20000	0

Geometry: -3 dB.

- o The geometry is the ratio of the average total received power from the cells in the active set to the average of all other received power. The geometry is therefore some measure of the location of the user, in term of C/I.
- o -3dB geometry corresponds to greater than 90% cell coverage.

Active set size: 1

- In the case if the user does see more than 1 cell in the active set, selection combining or soft combining can be used to achieve better performance.
- S-CCPCH: the operating Tx power yields 0.5%, 1%, 1.5% and 10% BLER
- DPDCH: Outer-loop target BLER: 0.5%, 1% and 1.5%