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650 Route des Lucioles
F-06921 Sophia Antipolis Cedex - FRANCE

Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

Siret N° 348 623 562 00017 - NAF 742 C
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1 Scope

The present document reports the study on video telephony robustness improvements extensions in Multimedia Telephony Service for IMS (MTSI) and provides recommendation on their applicability for MTSI video telephony applications.

2 References

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

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
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- [1] 3GPP TR 21.905: "Vocabulary for 3GPP Specifications".
- [2] 3GPP TS 22.105: "Services and service capabilities".
- [3] 3GPP TS 26.114: "IP Multimedia Subsystem (IMS); Multimedia telephony; Media handling and interaction".
- [4] IETF RFC 4588: "RTP Retransmission Payload Format", July 2006.
- [5] IETF RFC 6865: "Simple Reed-Solomon Forward Error Correction (FEC) Scheme for FECFRAME", February 2013.
- [6] IETF RFC 5109: "RTP Payload Format for Generic Forward Error Correction", December 2007.
- [7] IETF RFC 4585: "Extended RTP Profile for Real-time Transport Control Protocol (RTCP)-Based Feedback (RTP/AVPF)", July 2006.
- [8] K. Yamagishi, T. Hayashi, "Parametric Packet-Layer Model for Monitoring Video Quality of IPTV Services", IEEE ICC 2008, pp. 110-114, May 2008.
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- [10] C. Wang, X. Jiang, Y. Wang, "Video Quality Assessment Models for IPTV Services", JDCTA, April 2013.
- [11] Pierre Ferre, Dimitris Agrafiotis, Tuan Kiang Chiew, Angela Doufexi, Andrew Nix, David Bull, "Packet Loss Modelling for H.264 Video Transmission over IEEE 802.11g Wireless LANs", IEEE WIAMIS 2005.
- [12] S. Holmer, M. Shemer, M. Paniconi, "Handling Packet Loss in WebRTC", pp. 1860-1864, ICIP, 2013.

3 Definitions and abbreviations

3.1 Definitions

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

3.2 Abbreviations

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

AV	Audio Video
AVC	Advanced Video Coding
AVPF	Audio-Video Profile with Feedback
ER	Error Resiliency
FPS	Frames Per Second
HEVC	High Efficiency Video Coding
IMS-VT	IP Multimedia Subsystem Video Telephony
KB	Kilo Byte
MTSI	Multimedia Telephony Service for IMS
OTT	Over The Top
PLI	Picture Loss Indication
PLR	Packet Loss Rate
QVGA	Quarter Video Graphics Array
RPS	Reference Picture Selection
RPSI	Reference Picture Selection Indication
RTT	Round Trip Time
VGA	Video Graphics Array
VT	Video Telephony
VTRI_EXT	Video Robustness Improvements Extensions
Wifi	Wireless Fidelity
Note:	Wifi is synonymous with Wi-Fi as defined by the Wi-Fi Alliance

4 Background

The present document reports the study on video telephony robustness improvements extensions in Multimedia Telephony Service for IMS and provides recommendation on their applicability for MTSI video telephony applications. These extensions target error robustness for higher bitrate MTSI video telephony as well as inter-working with WLAN use cases where error resiliency is more important. In order to be technically competitive, e.g. to some proprietary systems, MTSI should have the capability to employ mechanisms that can offer different trade-offs between rendering delay, video rendering jitter (smoothness) and video quality that can adapt to varying channel conditions for better user experience. Retransmission, Forward Error Correction (FEC), and complementary reference picture selection indication (RPSI) AVPF feedback mechanisms offer these trade-offs. The present document first provides an overview of the additional error resiliency (ER) tools that could improve the performance of the Multimedia Telephony Service for IMS (TS 26.114 [3]). Then test conditions representative of error conditions experienced in IMS Video Telephony are presented. Following the description of the test conditions, evaluation criteria for determining the benefits of proposed tools and mechanisms is presented. Performance of the proposed ER tools is evaluated under the defined testing conditions that take into account packet loss rate/pattern, end to end delay, bitrate overhead and video smoothness (dropped frames, rendering jitter). Based on the performance results, conclusions are made in terms of recommendations for support of proposed ER tools and mechanisms for Multimedia Telephony Service for IMS.

5 Overview of video robustness improvements extensions (VTRI_EXT) tools

5.1 Introduction

Multimedia Telephony Service for IMS (MTSI 3GPP TS 26.114 [3]) defines MTSI clients' sender and receiver behaviour utilizing IETF RFC 4585 [7] AVPF Generic NACK and Picture Loss Indication (PLI) feedback messages for ER. Current error correction scheme provides basic error correction through codec level error resiliency (ER) mechanisms. Transport and application level error resiliency schemes such as Retransmission (NACK), Forward Error Correction (FEC) along with advanced codec level ER schemes such as Reference Picture Selection (RPS) provide alternative error correction mechanisms that offer different performance trade-offs. The performance of error correction schemes varies with end-to-end delay, channel bandwidth and packet loss rate.

5.2 Retransmission

Retransmission (NACK) scheme [4] provides efficient error correction in terms of bandwidth under short round-trip-time (RTT) cases with low packet loss rates. The efficiency of retransmission scheme becomes more pronounced at higher bitrates since selective retransmission of lost packets instead of entire pictures are needed. Under low RTT scenarios it can provide low video rendering jitter dependent on the de-jittering mechanism at the cost of additional delay. If additional delay cannot be accommodated, then retransmission can still provide recovery from error with video freezes during recovery similar to the existing error resiliency scheme in TS 26.114.

5.3 Forward error correction

Forward Error Correction (FEC) schemes [5] and [6] provide a mechanism that balances video quality and end-to-end delay. FEC schemes can adapt to varying channel error conditions. FEC is suitable for high RTT channels with high packet loss rates where retransmission leads to high video rendering delay and codec based recovery mechanisms like RPSI, PLI lead to frequent video freezes and/or corruptions. FEC schemes are complemented by retransmission (NACK) or RPSI, PLI feedback mechanisms to address FEC failure cases.

5.4 Reference picture selection

Reference picture selection indication (RPSI) feedback message in AVPF [7] that is currently not supported in TS 26.114 offers establishment of common reference point for recovery between the sender and the receiver. In essence it provides codec level ER mechanism similar to the transport layer ER mechanism supported by the generic NACK message in TS 26.114.

6 Test cases and conditions

6.1 QoS requirements for conversational video services

Specification TS 22.105 [2] defines the range of QoS requirements and end user QoS requirements for conversational video services. According to TS 22.105, the following requirements should be supported.

Table 6.1-1: Range of QoS requirements copied from TS 22.105 (clause 5.4)

	Real Time (Constant Delay)	Non Real Time (Variable Delay)
Operating environment	BER/Max Transfer Delay	BER/Max Transfer Delay
Satellite (Terminal relative speed to ground up to 1000 km/h for plane)	Max Transfer Delay less than 400 ms BER 10-3 - 10-7 (NOTE 1)	Max Transfer Delay 1200 ms or more (NOTE 2) BER = 10-5 to 10-8
Rural outdoor (Terminal relative speed to ground up to 500 km/h) (NOTE 3)	Max Transfer Delay 20 - 300 ms BER 10-3 - 10-7 (NOTE 1)	Max Transfer Delay 150 ms or more (NOTE 2) BER = 10-5 to 10-8
Urban/ Suburban outdoor (Terminal relative speed to ground up to 120 km/h)	Max Transfer Delay 20 - 300 ms BER 10-3 - 10-7 (NOTE 1)	Max Transfer Delay 150 ms or more (NOTE 2) BER = 10-5 to 10-8
Indoor/ Low range outdoor (Terminal relative speed to ground up to 10 km/h)	Max Transfer Delay 20 - 300 ms BER 10-3 - 10-7 (NOTE 1)	Max Transfer Delay 150 ms or more (NOTE 2) BER = 10-5 to 10-8
NOTE 1: There is likely to be a compromise between BER and delay. NOTE 2: The Max Transfer Delay should be here regarded as the target value for 95% of the data. NOTE 3: The value of 500 km/h as the maximum speed to be supported in the rural outdoor environment was selected in order to provide service on high speed vehicles (e.g. trains). This is not meant to be the typical value for this environment (250 km/h is more typical).		

And the requirements for end user QoS as performance expectations for conversational/real-time services is shown in table 6.1-2.

Table 6.1-2: End-user performance expectations (copied from TS 22.105 clause 5.5)

Medium	Application	Degree of symmetry	Data rate	Key performance parameters and target values		
				End-to-end One-way Delay	Delay Variation within a call	Information loss
Audio	Conversational voice	Two-way	4-25 kb/s	<150 msec preferred <400 msec limit NOTE 1	< 1 msec	< 3% FER
Video	Videophone	Two-way	32-384 kb/s	< 150 msec preferred <400 msec limit Lip-synch: < 100 msec		< 1% FER
Data	Telemetry - two-way control	Two-way	<28.8 kb/s	< 250 msec	N.A	Zero
Data	realtime games	Two-way	< 60 kb/s NOTE 2	< 75 msec preferred	N.A	< 3% FER preferred, < 5% FER limit NOTE 2
Data	Telnet	Two-way (asymmetric)	< 1 KB	< 250 msec	N.A	Zero
NOTE 1: The overall one way delay in the mobile network (from UE to PLMN border) is approximately 100msec.						
NOTE 2: These values are considered the most demanding ones with respect to delay requirements (e.g. supporting First Person Shooter games). Other types of games may require higher or lower data rates and more or less information loss but can tolerate longer end-to-end delay						

QoS test conditions used to evaluate the proposed tools should follow the service requirements described in TS 22.105. In addition to QoS networks, test conditions addressing interworking with non-QoS networks should be considered for the following reasons:

- Interworking with non-QoS networks is a relevant deployment use case and may result in losses in the non-managed part of the delivery.
- Despite QoS, there may be circumstances for which the QoS guarantees fail and service continuity is relevant.

6.2 Channel conditions

Channels conditions from QoS LTE, best effort over the top (OTT) LTE and WiFi channels are logged from video telephony calls for video configurations defined in clause 6.4. Packet captures are conducted on video telephony (VT) calls under mobile and stationary test conditions. Sending and receiving rates, delay (RTT/2), packet loss patterns are derived from captures sending and receiving times, timestamps and sequence numbers. The sources of the packet losses are from the physical channel as well as congestion. During the channel capturing process, the operating rate of the VT calls targeted rates below the available bandwidth for avoiding congestion. It is not always possible to avoid congestion during the capturing process. Logs exhibiting frequent large variations in rate due to congestion are filtered out.

Packet losses are characterized by the burst patterns. A packet *loss-free* burst of order k_0 is observed in the loss pattern when at least k_0 consecutive packets are correctly received. A packet loss burst order k_0 starts and finishes with a missing packet ("1") and is composed of at most $k_0 - 1$ consecutive received packets [11]. In the analysis presented in the present document, $k_0 = 1$ is used for simplicity. Sequences of m (total number of logged packets) loss indicators are divided into p alternating loss-free burst (X_j) and packet loss bursts (Y_j). Average packet loss rate PLR_{avg} , average loss free duration X_{avg} and average loss duration Y_{avg} are computed as:

$$PLR_{avg} = \frac{\sum_{j=0}^{p-1} Y_j}{\sum_{j=0}^{p-1} (X_j + Y_j)}, \quad (6.2-1)$$

$$X_{avg} = \frac{1}{p} \sum_{j=0}^{p-1} X_j, \quad (6.2-2)$$

$$Y_{avg} = \frac{1}{p} \sum_{j=0}^{p-1} Y_j. \quad (6.2-3)$$

6.3 Error profiles

6.3.1 Introduction

Error profiles representing guaranteed QoS and best effort (non-QoS) cases are used for evaluation. A number of real channel capture logs from QoS and non-QoS services are provided for emulation of channel conditions and/or derivation of channel models for simulation of channel conditions. Captured channel logs are used in the simulations of channel conditions for evaluation of proposed error resiliency tools.

6.3.2 QoS LTE

IMS-VT QoS calls conducted under low speed mobile conditions covering near cell and edge cell conditions were logged for analysis. QVGA (320x240), 15 fps, 350 kbps (maximum bitrate) H.264 video is used during the IMS-VT call. 17 MO to MT and 17 MT to MO logs selected from ~100 short duration calls (less than 1 minute) are used. In Table 6.3-1, MO to MT (IMS-QoS Test1) and likewise MT to MO (IMS-QoS Test2) call statistics are consolidated into one due to short duration of the calls. Packet loss statistics are tabulated in Table 6.3-1. Clause A.1 provides packet loss patterns for the consolidated logs.

6.3.3 LTE-OTT

Video telephony calls over LTE-OTT were conducted under driving conditions. One of the UEs is positioned in a stationary office environment with good LTE signal and the other UE in a moving vehicle. VGA (640x480) 30 fps 600 kbps (VT-LTE OTT Test1 & Test2) and QVGA 15 fps 300 kbps (VT-LTE OTT Test3 & Test4) videos were used for collecting channel logs. Packet loss statistics are tabulated in Table 6.3-1. Clause A.2 provides packet loss patterns for LTE-OTT tests.

6.3.4 WiFi

Video telephony calls over WiFi are conducted in office environment. Stationary office to office call and office to walking UE calls are logged. 720p (1 280x720) 30 fps 1 000 kbps video is used for collecting channel logs. Total of 8 logs (VT-Wifi Test1-8) are collected. Packet loss statistics are tabulated in Table 6.3-1. Clause A.3 provides packet loss patterns for WiFi tests.

6.3.5 Summary

Table 6.3-1 summarizes error profiles used during the evaluation process.

Table 6.3-1: Summary of error pattern statistics

Test	Condition	Bit - rate (kbps)	Frame Rate (fps)	Resolution	Duration (sec)	No. of packets	Avg loss free duration (pkts)	Avg. loss duration (pkts)	Avg PLR (%)
IMS-QoS Test1	Low mobility	350	15	320x240	309	12032	2 007	1,5	0,07%
IMS-QoS Test2	Low mobility	350	15	320x240	309	11870	627	4,1	0,66%
VT-LTE OTT Test1	High mobility	600	30	640x480	2 291	158 699	1 521	4,6	0,30%
VT-LTE OTT Test2	High mobility	600	30	640x480	2 290	145 352	1 305	5,7	0,43%
VT-LTE OTT Test3	Walk & High mobility	300	15	320x240	982	40 305	2 672	15,1	0,56%
VT-LTE OTT Test4	Walk & High mobility	300	15	320x240	981	39 222	2 440	11,8	0,48%
VT-Wifi Test1	Stationary	1 000	30	1 280x720	766	93 771	1 801	1,9	0,10%
VT-Wifi Test2	Stationary	1 000	30	1 280x720	765	92 795	1 685	1,9	0,11%
VT-Wifi Test3	Stationary	1 000	30	1 280x720	715	53 698	292	2,7	0,92%
VT-Wifi Test4	Stationary	1 000	30	1 280x720	717	72 244	36	1,9	5,02%
VT-Wifi Test5	Stationary	1 000	30	1 280x720	620	75 946	1 724	2,2	0,13%
VT-Wifi Test6	Stationary	1 000	30	1 280x720	620	75 472	1 477	3,2	0,21%
VT-Wifi Test7	Walk	1 000	30	1 280x720	381	24 045	607	9,8	1,60%
VT-Wifi Test8	Walk	1 000	30	1 280x720	381	37 093	67	3,4	4,75%
VT-Wifi Test9	Walk	1 000	30	1 280x720	913	54 260	39	2,7	7,19%
VT-Random	Random	1 000	30	1 280x720	1 013	98 634	-	-	10,04%

6.4 Test Content

For evaluation of ER tools, the two main factors that have impact on the overall performance is the video bitrate and the frame rate. It is assumed that the video is coded in low delay configuration, i.e. IPPPPP... or IBBBB.... configuration. The video resolution, content, and codec type (AVC, HEVC) have minimal impact since as described in clause 7, the corrupted pictures will be considered as non-rendered pictures. The following video resolutions, bitrate and frame rates are used during the evaluation process.

Table 6.4-1: Test content configuration

Resolution	Bitrate (kbps)	Frame rate (fps)
320x240	300 & 350	15
640x480	600	30
1 280x720	1 000	30

7 Evaluation criteria

7.1 Testing configuration

In order to simplify the evaluation process, it will be assumed that corrupted video frames will not be rendered. When an error occurs, corrective action based on retransmission, RPSI or FEC will be taken. These proposed tools can be used alone or in combination. For example FEC and RPSI or FEC and NACK can be used in combination to complement each other (i.e. when FEC fails, NACK or RPSI can achieve recovery).

Video bitstreams are packetized into maximum packet length of 1 400 bytes. Packetization byte overhead is ignored. Packet loss patterns are applied only in one direction according to the error profiles defined in clause 6.3, i.e. feedback channel is assumed to be error free. Sender and receiver side processing (encoding/decoding + various other tasks) times are ignored. Frames are generated at uniform time interval according to the frame rate. Transmission delay of packets in each direction is equal to RTT/2. Frames are packetized and sent as soon as they are encoded (i.e. at frame timestamps) at the sender, and removed from the packet de-jitter buffer as soon as complete frame data is available.

Decoding delay ($delay_D$) is computed as the difference between the time of removal from the de-jitter buffer for decoding and the capture timestamp (RTP timestamp). End-to-end rendering delay ($delay_{e2e}$) is determined as:

$$delay_{e2e} = avg_delay_D + 3 \times std_delay_D, \quad (7.1-1)$$

where, avg_delay_D is the average decoding delay and std_delay_D is standard deviation of decoding delay. This is to accommodate variation in arrival time of frames that can be rendered due retransmission. Frames that are late by more than $delay_{e2e}$ are not rendered. A hard limit of 400 ms is also imposed according to requirements of TS 22.105. Only perfectly reconstructed frames are rendered.

Every lost packet is reported to the sender side. For RPSI based recovery, it is assumed that the recovery frame size is same as the frame size in the bitstream that occurs at the recovery point and it generates an identical picture to the picture occurring at the recovery point. This simplified assumption is necessary for simplifying the simulations. It has negligible effect on the simulation results (without this assumption, the recovery frame size will be larger than the frame size occurring at the recovery point). For NACK based recovery, missing packets are retransmitted. For FEC based recovery an adaptive perfect FEC scheme (Reed Solomon) targeting 0,95 minimum recovery probability for maximum loss rate occurring during the 10 second history window is used. There is no interleaving of packets used and FEC packets do not cross frame boundaries, i.e. FEC packets protect source data that belongs to one frame. FEC overhead rate can be adjusted according to RTT time to minimize frequency of freezes when RTT is large. In the simulation environment, this method was not used.

7.2 Performance metrics

Assuming that there will be no corrupted pictures will be rendered, then the parameters that affect the perceived video quality are:

1. Bitrate overhead
2. End-to-end rendering delay
3. Number of frames not rendered
4. Rendering smoothness measure (standard deviation of rendering time from the target rendering time), i.e.

$$std_{renderdelu} = \sqrt{\frac{1}{N-1} \sum_{n=1}^N (\Delta_n - \Delta_{avg})^2}. \quad (7.2-1)$$

Δ_n "s are the time intervals between consecutively rendered frames and Δ_{avg} is the average of Δ_n for N frames [12].

In terms of bitrate overhead, FEC and retransmission are the tools that have bitrate overhead impact, FEC being the one that may have significant overhead. Given a limited channel bandwidth, full channel utilization, bitrate overhead impacts spatial video quality. The final effect of the overhead is the reduction of effective video source rate. Although it is content dependent, generally, bitrate reductions of 15% or more are perceivable. Bitrate overhead is measured with respect to the video source rate.

End-to-end rendering delay for video, which impacts audio delay, is critical for conversational services. The upper limit for tolerable delay is 400 ms. Delays of 150 ms or below are not noticeable. During a call end-to-end delay may vary. Among the proposed tools, retransmission is the only tool that may have impact on the end-to-end delay.

Number frames not rendered convey information on the temporal video quality. A frame will not be rendered if it has a packet that is missing or it is dependent on past frames that had missing packets. It is related to ER failure rate for frames. In general the higher it is, the worse the perceived video quality is. However the distribution of non-rendered frames also impact the visual quality. In [8], [9] and [10], it is reported that the frequent short video freezes result in lower MOS scores than long infrequent video freezes. Rendering smoothness measure in combination with number of not rendered frames conveys distribution information of video freezes. These two metrics are applicable to all proposed ER tools.

During the testing process, audio-video (AV) synchronization is assumed to be preserved, i.e. long term delay in video forces audio to be delayed. End to end delay in evaluation setup remains within the bounds specified in TS 22.105.

8 Results

8.1 Test cases

The performance of proposed tools under channel conditions defined in clause 6.3 are evaluated according to metrics defined for video quality in clause 7.2. Video test content defined in clause 6.4 is generated offline using an H.264 encoder. A test setup that simulates channel conditions according to conditions defined in clause 6.3 as well as error resiliency behavior of the proposed tools is used. Evaluations of the proposed tools are conducted under different round-trip-time (RTT) conditions with the captured channel logs.

Captured channel logs are used for simulating packet losses under RTT of (100 ms, 200 ms, 300 ms, 400 ms). The proposed metrics defined in clause 7.2 are logged to characterize behavior of each tool. Each tool is tested individually and in combination with other tools. The following test cases are run:

Table 8.1-1: Test cases

Test Cases
TS 26.114 NACK or RPSI
FEC+RPSI
Retransmission (NACK)
FEC+ retransmission

8.2 Simulation (RTT= 100 ms)

Table 8.2-1 shows the evaluation results for RTT = 100 ms.

Table 8.2-1: RTT = 100 ms

Tool	Test(RTT = 100 ms)	PLR%	Bitrate Overhead %	Total Frames	e2e Delay (ms)	Std Render Delta(ms)	Rendered Frames	Rendered Frame%
TS 26.114 NACK or RPSI	IMS-QoS Test1	0,07%	0,00%	4561	50	5	4 550	99,76%
	IMS-QoS Test2	0,66%	0,00%	4497	50	24	4 453	99,02%
	VT-LTE OTT Test1	0,30%	0,00%	77 793	50	9	77 389	99,48%
	VT-LTE OTT Test2	0,43%	0,00%	71 249	50	12	70 820	99,40%
	VT-LTE OTT Test3	0,56%	0,00%	19 072	50	34	18 947	99,34%
	VT-LTE OTT Test4	0,48%	0,00%	18 560	50	24	18 458	99,45%
	VT-Wifi Test1	0,10%	0,00%	28 884	50	6	28 677	99,28%
	VT-Wifi Test2	0,11%	0,00%	28 583	50	6	28 364	99,23%
	VT-Wifi Test3	0,92%	0,00%	16 540	50	13	15 948	96,42%
	VT-Wifi Test4	5,02%	0,00%	22 253	50	62	17 584	79,02%
	VT-Wifi Test5	0,13%	0,00%	23 393	50	6	23 223	99,27%
	VT-Wifi Test6	0,21%	0,00%	23 247	50	7	23 028	99,06%
	VT-Wifi Test7	1,60%	0,00%	7 407	50	23	7 198	97,18%
	VT-Wifi Test8	4,75%	0,00%	11 425	50	62	9 797	85,75%
VT-Wifi Test9	7,19%	0,00%	16 714	50	68	13 142	78,63%	
VT-Random	10,04%	0,00%	30 381	50	79	13 791	45,39%	
FEC+RPSI	IMS-QoS Test1	0,07%	6,11%	4 300	50	5	4 290	99,77%
	IMS-QoS Test2	0,66%	7,48%	4 186	50	27	4 141	98,93%
	VT-LTE OTT Test1	0,30%	4,30%	74 587	50	7	74 269	99,57%
	VT-LTE OTT Test2	0,43%	2,83%	69 292	50	13	68 925	99,47%
	VT-LTE OTT Test3	0,56%	2,06%	18 688	50	31	18 579	99,42%
	VT-LTE OTT Test4	0,48%	2,13%	18 173	50	14	18 110	99,65%
	VT-Wifi Test1	0,10%	9,74%	26 328	50	6	26 141	99,29%
	VT-Wifi Test2	0,11%	10,84%	25 799	50	6	25 595	99,21%
	VT-Wifi Test3	0,92%	26,54%	13 097	50	12	12 746	97,32%
	VT-Wifi Test4	5,02%	43,16%	15 566	50	26	14 422	92,65%
	VT-Wifi Test5	0,13%	11,17%	21 063	50	6	20 917	99,31%
	VT-Wifi Test6	0,21%	13,04%	20 588	50	7	20 391	99,04%
	VT-Wifi Test7	1,60%	14,02%	6 502	50	22	6 331	97,37%
	VT-Wifi Test8	4,75%	32,00%	8 669	50	42	8 051	92,87%
VT-Wifi Test9	7,19%	36,46%	12 261	50	47	11 047	90,10%	
VT-Random	10,04%	62,92%	18 724	50	11	18 177	97,08%	
Retransmit (NACK)	IMS-QoS Test1	0,07%	0,08%	4 557	73	8	4 540	99,63%
	IMS-QoS Test2	0,66%	0,64%	4 468	146	16	4 440	99,37%
	VT-LTE OTT Test1	0,30%	0,31%	77 555	133	9	77 213	99,56%
	VT-LTE OTT Test2	0,43%	0,44%	70 939	131	9	70 661	99,61%
	VT-LTE OTT Test3	0,56%	0,57%	18 964	400	31	18 871	99,51%
	VT-LTE OTT Test4	0,48%	0,48%	18 470	243	19	18 414	99,70%
	VT-Wifi Test1	0,10%	0,10%	28 853	76	6	28 631	99,23%
	VT-Wifi Test2	0,11%	0,11%	28 550	76	6	28 318	99,19%
	VT-Wifi Test3	0,92%	0,92%	16 388	130	13	15 965	97,42%
	VT-Wifi Test4	5,02%	5,31%	21 134	339	18	20 804	98,44%
	VT-Wifi Test5	0,13%	0,13%	23 363	77	6	23 172	99,18%
	VT-Wifi Test6	0,21%	0,21%	23 197	91	6	23 011	99,20%
	VT-Wifi Test7	1,60%	1,54%	7 295	331	19	7 206	98,78%
	VT-Wifi Test8	4,75%	4,90%	10 892	400	44	10 620	97,50%
VT-Wifi Test9	7,19%	7,76%	15 525	400	35	14 975	96,46%	
VT-Random	10,04%	11,16%	27 330	366	7	27 029	98,90%	
FEC + NACK	IMS-QoS Test1	0,07%	5,53%	4 319	72	7	4 303	99,63%
	IMS-QoS Test2	0,66%	8,34%	4 159	151	14	4 134	99,40%
	VT-LTE OTT Test1	0,30%	4,75%	74 265	125	8	73 983	99,62%
	VT-LTE OTT Test2	0,43%	2,97%	69 196	133	9	68 948	99,64%
	VT-LTE OTT Test3	0,56%	2,13%	18 675	400	29	18 590	99,54%
	VT-LTE OTT Test4	0,48%	2,78%	18 062	185	16	18 012	99,72%
	VT-Wifi Test1	0,10%	9,85%	26 299	75	6	26 097	99,23%
	VT-Wifi Test2	0,11%	10,97%	25 769	77	6	25 552	99,16%
	VT-Wifi Test3	0,92%	27,68%	12 983	112	11	12 686	97,71%
VT-Wifi Test4	5,02%	43,03%	15 577	241	16	15 322	98,36%	

Tool	Test(RTT = 100 ms)	PLR%	Bitrate Overhead %	Total Frames	e2e Delay (ms)	Std Render Delta(ms)	Rendered Frames	Rendered Frame%
	VT-Wifi Test5	0,13%	11,32%	21 034	76	6	20 869	99,22%
	VT-Wifi Test6	0,21%	13,73%	20 468	91	7	20 304	99,20%
	VT-Wifi Test7	1,60%	17,01%	6 340	249	15	6 260	98,74%
	VT-Wifi Test8	4,75%	36,77%	8 363	400	33	8 191	97,94%
	VT-Wifi Test9	7,19%	45,28%	11 533	400	25	11 258	97,62%
	VT-Random	10,04%	63,51%	18 654	108	10	18 183	97,48%

8.3 Simulation (RTT= 200 ms)

Table 8.3-1 shows the evaluation results for RTT = 200 ms.

Table 8.3-1: RTT = 200 ms

Tool	Test(RTT = 200 ms)	PLR%	Bitrate Overhead %	Total Frames	e2e Delay (ms)	Std Render Delta(ms)	Rendered Frames	Rendered Frame%
TS 26.114 NACK or RPSI	IMS-QoS Test1	0,07%	0,00%	4 561	100	9	4 540	99,54%
	IMS-QoS Test2	0,66%	0,00%	4 497	100	25	4 448	98,91%
	VT-LTE OTT Test1	0,30%	0,00%	77 793	100	11	77 293	99,36%
	VT-LTE OTT Test2	0,43%	0,00%	71 249	100	15	70 747	99,30%
	VT-LTE OTT Test3	0,56%	0,00%	19 072	100	35	18 929	99,25%
	VT-LTE OTT Test4	0,48%	0,00%	18 560	100	25	18 448	99,40%
	VT-Wifi Test1	0,10%	0,00%	28 884	100	10	28 524	98,75%
	VT-Wifi Test2	0,11%	0,00%	28 583	100	10	28 202	98,67%
	VT-Wifi Test3	0,92%	0,00%	16 540	100	23	15 602	94,33%
	VT-Wifi Test4	5,02%	0,00%	22 253	100	76	16 164	72,64%
	VT-Wifi Test5	0,13%	0,00%	23 393	100	10	23 098	98,74%
	VT-Wifi Test6	0,21%	0,00%	23 247	100	12	22 885	98,44%
	VT-Wifi Test7	1,60%	0,00%	7 407	100	27	7 135	96,33%
	VT-Wifi Test8	4,75%	0,00%	11 425	100	81	9 220	80,70%
VT-Wifi Test9	7,19%	0,00%	16 714	100	83	12 141	72,64%	
VT-Random	10,04%	0,00%	30 381	100	137	10 052	33,09%	
FEC+RPSI	IMS-QoS Test1	0,07%	6,10%	4 301	100	8	4 285	99,63%

Tool	Test(RTT = 200 ms)	PLR%	Bitrate Overhead %	Total Frames	e2e Delay (ms)	Std Render Delta(ms)	Rendered Frames	Rendered Frame%
	IMS-QoS Test2	0,66%	7,44%	4 186	100	36	4 133	98,73%
	VT-LTE OTT Test1	0,30%	4,40%	74 515	100	10	74 083	99,42%
	VT-LTE OTT Test2	0,43%	2,41%	69 577	100	14	69 123	99,35%
	VT-LTE OTT Test3	0,56%	2,10%	18 679	100	28	18 568	99,41%
	VT-LTE OTT Test4	0,48%	2,14%	18 172	100	18	18 099	99,60%
	VT-Wifi Test1	0,10%	9,67%	26 344	100	10	26 012	98,74%
	VT-Wifi Test2	0,11%	10,88%	25 790	100	10	25 448	98,67%
	VT-Wifi Test3	0,92%	26,53%	13 097	100	18	12 570	95,98%
	VT-Wifi Test4	5,02%	40,87%	15 813	100	35	14 037	88,77%
	VT-Wifi Test5	0,13%	11,75%	20 958	100	9	20 718	98,85%
	VT-Wifi Test6	0,21%	13,03%	20 590	100	12	20 258	98,39%
	VT-Wifi Test7	1,60%	14,44%	6 480	100	23	6 262	96,64%
	VT-Wifi Test8	4,75%	32,10%	8 661	100	49	7 814	90,22%
	VT-Wifi Test9	7,19%	37,19%	12 196	100	55	10 612	87,01%
VT-Random	10,04%	62,92%	18 723	100	20	17 814	95,15%	
Retransmit (NACK)	IMS-QoS Test1	0,07%	0,08%	4 557	141	10	4 535	99,52%
	IMS-QoS Test2	0,66%	0,64%	4 468	262	20	4 434	99,24%
	VT-LTE OTT Test1	0,30%	0,31%	77 555	215	13	77 114	99,43%
	VT-LTE OTT Test2	0,43%	0,44%	70 939	211	11	70 584	99,50%
	VT-LTE OTT Test3	0,56%	0,56%	18 964	400	34	18 863	99,47%
	VT-LTE OTT Test4	0,48%	0,47%	18 470	368	22	18 404	99,64%
	VT-Wifi Test1	0,10%	0,10%	28 853	154	9	28 529	98,88%
	VT-Wifi Test2	0,11%	0,11%	28 550	156	9	28 210	98,81%
	VT-Wifi Test3	0,92%	0,93%	16 388	264	19	15 816	96,51%
	VT-Wifi Test4	5,02%	5,31%	21 134	400	34	20 189	95,53%
	VT-Wifi Test5	0,13%	0,13%	23 363	156	9	23 086	98,81%
	VT-Wifi Test6	0,21%	0,21%	23 197	171	9	22 911	98,77%
	VT-Wifi Test7	1,60%	1,50%	7 297	400	20	7 197	98,63%
	VT-Wifi Test8	4,75%	4,89%	10 892	400	53	10 411	95,58%
VT-Wifi Test9	7,19%	7,75%	15 528	400	53	14 364	92,50%	
VT-Random	10,04%	11,15%	27 331	400	44	22 540	82,47%	
FEC + NACK	IMS-QoS Test1	0,07%	6,19%	4 298	140	10	4 277	99,51%
	IMS-QoS Test2	0,66%	8,35%	4 158	286	22	4 122	99,13%
	VT-LTE OTT Test1	0,30%	4,75%	74 264	208	12	73 886	99,49%
	VT-LTE OTT Test2	0,43%	3,06%	69 134	208	11	68 798	99,51%
	VT-LTE OTT Test3	0,56%	2,13%	18 676	400	31	18 583	99,50%
	VT-LTE OTT Test4	0,48%	2,69%	18 075	278	18	18 017	99,68%
	VT-Wifi Test1	0,10%	9,86%	26 297	154	9	25 996	98,86%
	VT-Wifi Test2	0,11%	11,00%	25 763	157	9	25 438	98,74%
	VT-Wifi Test3	0,92%	27,29%	13 022	240	14	12 702	97,54%
	VT-Wifi Test4	5,02%	44,57%	15 411	400	23	15 099	97,98%
	VT-Wifi Test5	0,13%	11,81%	20 947	155	9	20 710	98,87%
	VT-Wifi Test6	0,21%	13,22%	20 558	171	9	20 305	98,77%
	VT-Wifi Test7	1,60%	15,29%	6 430	346	17	6 342	98,63%
	VT-Wifi Test8	4,75%	35,68%	8 433	400	38	8 183	97,04%
VT-Wifi Test9	7,19%	44,90%	11 552	400	35	11 047	95,63%	
VT-Random	10,04%	63,75%	18 626	215	13	18 069	97,01%	

8.4 Simulation (RTT= 300 ms)

Table 8.4-1 shows the evaluation results for RTT = 300 ms.

Table 8.4-1: RTT = 300 ms

Tool	Test(RTT = 300 ms)	PLR%	Bitrate Overhead %	Total Frames	e2e Delay (ms)	Std Render Delta(ms)	Rendered Frames	Rendered Frame%
TS 26.114 NACK or RPSI	IMS-QoS Test1	0,07%	0,00%	4 561	150	11	4 535	99,43%
	IMS-QoS Test2	0,66%	0,00%	4 497	150	34	4 442	98,78%
	VT-LTE OTT Test1	0,30%	0,00%	77 793	150	14	77 181	99,21%
	VT-LTE OTT Test2	0,43%	0,00%	71 249	150	18	70 701	99,23%
	VT-LTE OTT Test3	0,56%	0,00%	19 072	150	36	18 920	99,20%
	VT-LTE OTT Test4	0,48%	0,00%	18 560	150	26	18 442	99,36%
	VT-Wifi Test1	0,10%	0,00%	28 884	150	14	28 371	98,22%
	VT-Wifi Test2	0,11%	0,00%	28 583	150	14	28 045	98,12%
	VT-Wifi Test3	0,92%	0,00%	16 540	150	32	15 317	92,61%
	VT-Wifi Test4	5,02%	0,00%	22 253	150	94	14 940	67,14%
	VT-Wifi Test5	0,13%	0,00%	23 393	150	14	22 972	98,20%
	VT-Wifi Test6	0,21%	0,00%	23 247	150	16	22 744	97,84%
	VT-Wifi Test7	1,60%	0,00%	7 407	150	31	7 078	95,56%
	VT-Wifi Test8	4,75%	0,00%	11 425	150	95	8 786	76,90%
VT-Wifi Test9	7,19%	0,00%	16 714	150	103	11 258	67,36%	
VT-Random	10,04%	0,00%	30 381	150	191	7 890	25,97%	
FEC+RPSI	IMS-QoS Test1	0,07%	6,12%	4 300	150	11	4 275	99,42%
	IMS-QoS Test2	0,66%	7,42%	4 187	150	35	4 134	98,73%
	VT-LTE OTT Test1	0,30%	4,36%	74 547	150	12	74 056	99,34%
	VT-LTE OTT Test2	0,43%	2,80%	69 310	150	14	68 862	99,35%
	VT-LTE OTT Test3	0,56%	2,06%	18 687	150	32	18 560	99,32%
	VT-LTE OTT Test4	0,48%	2,14%	18 171	150	20	18 091	99,56%
	VT-Wifi Test1	0,10%	9,74%	26 328	150	14	25 855	98,20%
	VT-Wifi Test2	0,11%	10,84%	25 798	150	15	25 290	98,03%
	VT-Wifi Test3	0,92%	26,53%	13 097	150	26	12 360	94,37%
	VT-Wifi Test4	5,02%	41,43%	15 749	150	49	13 495	85,69%
	VT-Wifi Test5	0,13%	11,26%	21 046	150	14	20 702	98,37%
	VT-Wifi Test6	0,21%	13,08%	20 584	150	17	20 114	97,72%
	VT-Wifi Test7	1,60%	14,03%	6 501	150	28	6 227	95,79%
	VT-Wifi Test8	4,75%	32,97%	8 608	150	56	7 569	87,93%
VT-Wifi Test9	7,19%	35,88%	12 312	150	66	10 340	83,98%	
VT-Random	10,04%	62,90%	18 726	150	28	17 421	93,03%	
Retransmit	IMS-QoS Test1	0,07%	0,08%	4 557	212	12	4 530	99,41%

Tool	Test(RTT = 300 ms)	PLR%	Bitrate Overhead %	Total Frames	e2e Delay (ms)	Std Render Delta(ms)	Rendered Frames	Rendered Frame%
	IMS-QoS Test2	0,66%	0,64%	4 468	332	21	4 429	99,13%
	VT-LTE OTT Test1	0,30%	0,31%	77 555	302	14	77 032	99,33%
	VT-LTE OTT Test2	0,43%	0,44%	70 939	305	15	70 512	99,40%
	VT-LTE OTT Test3	0,56%	0,57%	18 964	400	36	18 843	99,36%
	VT-LTE OTT Test4	0,48%	0,47%	18 470	400	25	18 389	99,56%
	VT-Wifi Test1	0,10%	0,10%	28 853	239	12	28 427	98,52%
	VT-Wifi Test2	0,11%	0,11%	28 550	242	12	28 102	98,43%
	VT-Wifi Test3	0,92%	0,92%	16 388	385	18	15 819	96,53%
	VT-Wifi Test4	5,02%	5,30%	21 134	400	77	16 388	77,54%
	VT-Wifi Test5	0,13%	0,13%	23 363	242	12	23 002	98,45%
	VT-Wifi Test6	0,21%	0,21%	23 197	268	13	22 807	98,32%
	VT-Wifi Test7	1,60%	1,50%	7 297	400	25	7 102	97,33%
	VT-Wifi Test8	4,75%	4,90%	10 892	400	73	9 348	85,82%
	VT-Wifi Test9	7,18%	7,73%	15 544	400	93	12 119	77,97%
VT-Random	10,04%	11,15%	27 331	400	233	7 393	27,05%	
FEC + NACK	IMS-QoS Test1	0,07%	5,55%	4 318	211	12	4 292	99,40%
	IMS-QoS Test2	0,66%	8,28%	4 160	358	23	4 119	99,01%
	VT-LTE OTT Test1	0,30%	4,61%	74 367	293	14	73 904	99,38%
	VT-LTE OTT Test2	0,43%	3,05%	69 141	310	15	68 732	99,41%
	VT-LTE OTT Test3	0,56%	2,13%	18 675	400	33	18 563	99,40%
	VT-LTE OTT Test4	0,48%	2,56%	18 099	358	20	18 032	99,63%
	VT-Wifi Test1	0,10%	9,85%	26 299	238	11	25 920	98,56%
	VT-Wifi Test2	0,11%	10,97%	25 769	242	12	25 357	98,40%
	VT-Wifi Test3	0,92%	26,86%	13 061	346	17	12 615	96,59%
	VT-Wifi Test4	5,02%	46,46%	15 214	400	44	14 018	92,14%
	VT-Wifi Test5	0,13%	11,31%	21 036	238	12	20 731	98,55%
	VT-Wifi Test6	0,21%	13,72%	20 468	270	13	20 115	98,28%
	VT-Wifi Test7	1,60%	15,44%	6 423	400	21	6 262	97,49%
	VT-Wifi Test8	4,75%	35,77%	8 425	400	51	7 816	92,77%
VT-Wifi Test9	7,19%	44,42%	11 591	400	53	10 534	90,88%	
VT-Random	10,04%	63,95%	18 602	359	16	17 962	96,56%	

8.5 Simulation (RTT= 400 ms)

Table 8.5-1 shows the evaluation results for RTT = 400 ms.

Table 8.5-1: RTT = 400 ms

Tool	Test(RTT = 400 ms)	PLR%	Bitrate Overhead %	Total Frames	e2e Delay (ms)	Std Render Delta(ms)	Rendered Frames	Rendered Frame%
TS 26.114 NACK or RPSI	IMS-QoS Test1	0,07%	0,00%	4 561	200	16	4 525	99,21%
	IMS-QoS Test2	0,66%	0,00%	4 497	200	34	4 435	98,62%
	VT-LTE OTT Test1	0,30%	0,00%	77 793	200	15	77 097	99,11%
	VT-LTE OTT Test2	0,43%	0,00%	71 249	200	20	70 648	99,16%
	VT-LTE OTT Test3	0,56%	0,00%	19 072	200	38	18 905	99,12%
	VT-LTE OTT Test4	0,48%	0,00%	18 560	200	27	18 431	99,31%
	VT-Wifi Test1	0,10%	0,00%	28 884	200	18	28 218	97,69%
	VT-Wifi Test2	0,11%	0,00%	28 583	200	19	27 880	97,54%
	VT-Wifi Test3	0,92%	0,00%	16 540	200	39	15 035	90,90%
	VT-Wifi Test4	5,02%	0,00%	22 253	200	109	13 972	62,79%
	VT-Wifi Test5	0,13%	0,00%	23 393	200	18	22 846	97,66%
	VT-Wifi Test6	0,21%	0,00%	23 247	200	20	22 606	97,24%
	VT-Wifi Test7	1,60%	0,00%	7 407	200	36	7 012	94,67%
	VT-Wifi Test8	4,75%	0,00%	11 425	200	95	8 522	74,59%
VT-Wifi Test9	7,19%	0,00%	16 714	200	122	10 630	63,60%	
VT-Random	10,04%	0,00%	30 381	200	268	6 013	19,79%	
FEC+RPSI	IMS-QoS Test1	0,07%	6,11%	4 300	200	16	4 265	99,19%
	IMS-QoS Test2	0,66%	7,40%	4 188	200	41	4 120	98,38%
	VT-LTE OTT Test1	0,30%	4,88%	74 181	200	13	73 605	99,22%
	VT-LTE OTT Test2	0,43%	2,43%	69 566	200	17	68 993	99,18%
	VT-LTE OTT Test3	0,56%	2,11%	18 678	200	31	18 542	99,27%
	VT-LTE OTT Test4	0,48%	2,07%	18 183	200	21	18 095	99,52%
	VT-Wifi Test1	0,10%	9,70%	26 336	200	18	25 724	97,68%
	VT-Wifi Test2	0,11%	10,91%	25 782	200	19	25 133	97,48%
	VT-Wifi Test3	0,92%	26,55%	13 095	200	33	12 179	93,01%
	VT-Wifi Test4	5,02%	43,78%	15 498	200	59	12 918	83,35%
	VT-Wifi Test5	0,13%	11,75%	20 957	200	18	20 511	97,87%
	VT-Wifi Test6	0,21%	13,11%	20 575	200	21	19 987	97,14%
	VT-Wifi Test7	1,60%	14,12%	6 494	200	30	6 183	95,21%
	VT-Wifi Test8	4,75%	32,81%	8 614	200	64	7 405	85,96%
VT-Wifi Test9	7,19%	36,66%	12 248	200	75	10 004	81,68%	
VT-Random	10,04%	62,70%	18 752	200	37	17 127	91,33%	
Retransmit (NACK)	IMS-QoS Test1	0,07%	0,08%	4 557	286	14	4 525	99,30%
	IMS-QoS Test2	0,66%	0,65%	4 468	400	26	4 418	98,88%
	VT-LTE OTT Test1	0,30%	0,31%	77 555	389	16	76 953	99,22%
	VT-LTE OTT Test2	0,43%	0,44%	70 939	400	17	70 445	99,30%
	VT-LTE OTT Test3	0,56%	0,57%	18 964	400	39	18 817	99,22%
	VT-LTE OTT Test4	0,48%	0,47%	18 470	400	27	18 374	99,48%
	VT-Wifi Test1	0,10%	0,10%	28 853	330	15	28 325	98,17%
	VT-Wifi Test2	0,11%	0,11%	28 550	333	15	27 994	98,05%
	VT-Wifi Test3	0,92%	0,92%	16 388	400	36	15 175	92,60%
	VT-Wifi Test4	5,02%	5,30%	21 134	400	142	12 805	60,59%
	VT-Wifi Test5	0,13%	0,13%	23 363	333	15	22 918	98,10%
	VT-Wifi Test6	0,21%	0,21%	23 197	353	16	22 711	97,90%
	VT-Wifi Test7	1,60%	1,50%	7 297	400	33	6 983	95,70%
	VT-Wifi Test8	4,75%	4,89%	10 892	400	108	8 158	74,90%
VT-Wifi Test9	7,19%	7,53%	15 547	400	169	9 860	63,42%	
VT-Random	10,04%	11,15%	27 331	400	1756	1 478	5,41%	
FEC + NACK	IMS-QoS Test1	0,07%	6,18%	4 298	286	14	4 267	99,28%
	IMS-QoS Test2	0,66%	8,28%	4 161	400	29	4 108	98,73%
	VT-LTE OTT Test1	0,30%	4,77%	74 251	364	15	73 719	99,28%
	VT-LTE OTT Test2	0,43%	3,06%	69 138	395	16	68 668	99,32%
	VT-LTE OTT Test3	0,56%	2,13%	18 676	400	36	18 542	99,28%
	VT-LTE OTT Test4	0,48%	2,55%	18 101	400	22	18 023	99,57%
	VT-Wifi Test1	0,10%	9,86%	26 297	330	15	25 806	98,13%
	VT-Wifi Test2	0,11%	10,99%	25 765	341	14	25 301	98,20%
VT-Wifi Test3	0,92%	27,34%	13 014	400	25	12 338	94,81%	
VT-Wifi Test4	5,02%	45,79%	15 281	400	65	13 056	85,44%	

Tool	Test(RTT = 400 ms)	PLR%	Bitrate Overhead %	Total Frames	e2e Delay (ms)	Std Render Delta(ms)	Rendered Frames	Rendered Frame%
	VT-Wifi Test5	0,13%	11,81%	20 946	329	15	20 569	98,20%
	VT-Wifi Test6	0,21%	13,72%	20 469	360	16	20 043	97,92%
	VT-Wifi Test7	1,60%	16,09%	6 386	400	30	6 131	96,01%
	VT-Wifi Test8	4,75%	35,68%	8 436	400	74	7 382	87,51%
	VT-Wifi Test9	7,19%	43,94%	11 624	400	77	9 845	84,70%
	VT-Random	10,04%	64,48%	18 543	400	34	17 111	92,28%

8.6 Summary

Results in the previous clauses illustrates the behaviour of each tool under different channel conditions. As a reference the performance of RPSI tool can be taken since the behaviour of RPSI tool by itself is equivalent to the error resilience behaviour in TS 26.114 that utilizes PLI and generic NACK messages. The weakness of this tool is that for every loss point there is a freeze (not rendered frames) of at least RTT duration. As RTT increases and PLR increases, the amount of non-rendered frames increases. This can be observed in VT-Wifi Test4 and Test8 where there is around 5% packet loss. As RTT increases from 100 ms to 400 ms, the percentage of rendered frames decreases from 79 - 85% to 63 - 75%. Since no retransmission is involved in this mechanism end to end delay is preserved. The main strength of this tool is its efficient handling of large burst losses that cannot be handled efficiently with other mechanisms such as FEC and retransmission.

FEC can handle random losses and short burst losses in a way that RPSI, retransmission cannot handle by introducing bitrate overhead. This becomes more important as RTT and loss rate increases. By trading of spatial video quality to temporal smoothness (i.e. less freezes) it can provide a very robust way of handling errors. For the VT-Wifi Test4 and Test8 cases it can provide rendered frame percentage of 93% and ~84% for RTT of 100 ms and 400 ms, respectively. FEC overhead can be modulated adaptively to adapt to the channel conditions, i.e. loss rate and RTT.

Retransmission is an efficient recovery tool for low loss rates and low RTT. Under these circumstances it can provide the most efficient recovery and maintain smooth rendering without introducing high delay. This can be seen in less than 1% packet loss cases with low RTT like 100 ms. In the higher RTT cases, the end to end delay increases but can be kept under 400 ms cut off if the loss rate is low.

FEC cannot recover all error cases. It needs a backup mechanism to handle the error cases that cannot be recovered by FEC. This mechanism can be retransmission, PLI or RPSI. It can also be combined with the current generic NACK mechanism specified in TS 26.114.

9 Conclusions and recommendations

Results in clause 8 show the trade-offs of each proposed tool under various channel conditions. FEC and selective retransmission offer benefits that cannot be achieved by the existing ER tools supported in TS 26.114.

- FEC provides robustness against moderate packet loss rates at high delay scenario. FEC can especially handle random losses and short burst losses and be beneficial in environments with high packet loss rates and/or high delay (RTT). Use of FEC may however not be appropriate when packet losses are caused by insufficient throughput (over radio access or due to congestions in network) since it introduces some bit rate overhead. In order to compensate for bit rate overhead, FEC may require to be used with efficient rate adaptation mechanisms to reduce the source bit rate according to channel conditions and not increase the total RTP bitrate. FEC will be used in combination with other mechanisms to handle the error cases that cannot be recovered by FEC (like PLI or RPSI or the current generic NACK mechanism specified in TS 26.114):
- For low RTT case with relatively high packet loss, using retransmission in combination with FEC is beneficial since retransmission can efficiently handle the FEC failure case.
- For high RTT, relatively high packet loss conditions, using generic NACK based recovery in combination with FEC is beneficial since generic NACK based recovery does not introduce additional delay.

- Selective retransmission offers efficient recovery mechanism under low delay (RTT) and low failure (loss) rate conditions. Retransmission needs to ensure that retransmitted packets arrive in time to meet delay requirements of the end to end system. Higher packet loss rates may cause loss of retransmitted packets, hence leading to larger end to end delay.
- Existing generic NACK, PLI or RPSI based error correction mechanism can provide an efficient recovery for low packet loss rates with high RTT conditions. Generic NACK message can be used for indication of packets to be retransmitted as well as informing the sender of loss of particular RTP packets for sender to take necessary actions to recover from errors. These two behaviours of the system for generic NACK message should be differentiated by signalling or some other means. RPSI is a similar mechanism operating at codec level that offers, in addition, establishment of common reference point for recovery between the sender and the receiver. If retransmission based ER is being used, the support for additional RPSI or existing NACK based error correction mechanism is not essential since the failure cases for retransmission based scheme would be rare. In that case PLI message can be used to recover from errors.

FEC and retransmission provides ER mechanisms that are effective under different channel conditions that can be encountered. These tools are beneficial under non-QoS environments that are becoming more widely used with IMS-VT terminals. In order to be competitive with non-IMS based solutions, these tools should be supported. Although RPSI provides a clean mechanism to address cases where FEC or retransmission fails, the existing generic NACK based ER scheme can provide similar functionality. It is recommended that FEC and retransmission should be supported in TS 26.114. Support for these proposed tools should be negotiable during a call or at session setup.

NOTE 1: Proper implementation and usage of these different tools (e.g. trade-off between quality & delay) are still left to the MTSI client implementers taking into account the above recommendations. This has to be done according to the service requirements and expected channel conditions that may differ from the set of test cases and related error profiles defined in section 6 and used for evaluation purpose. It is recommended to update TS 26.114 to include the above text relevant for the mechanisms to recover from packet losses included in TS 26.114 to provide additional information and guidelines on usage and benefits under various channel conditions of these mechanisms.

Annex A: Error patterns

A.1 IMS-QoS

Packet loss statistics are plotted vs. packet index (X-axis).

IMS-QoS Test1 error pattern.

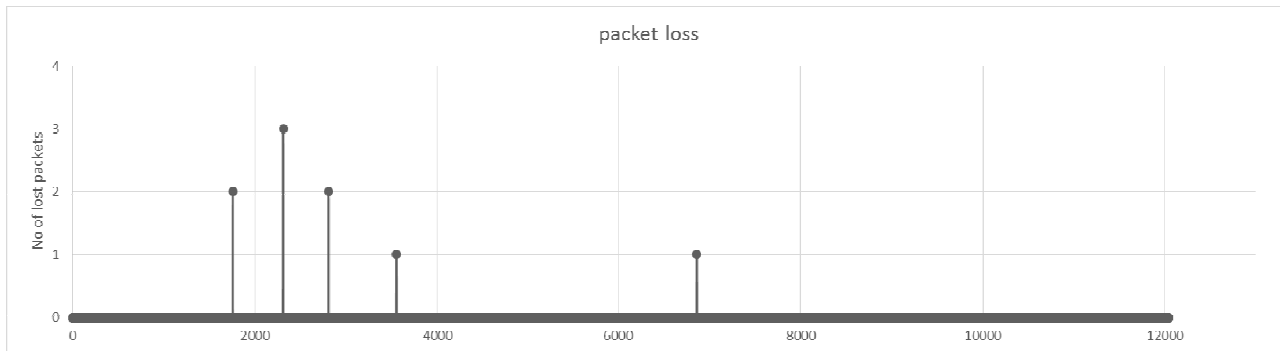


Figure A.1-1: Packet loss pattern PLR = 0,07%

IMS-QoS Test2 error pattern.

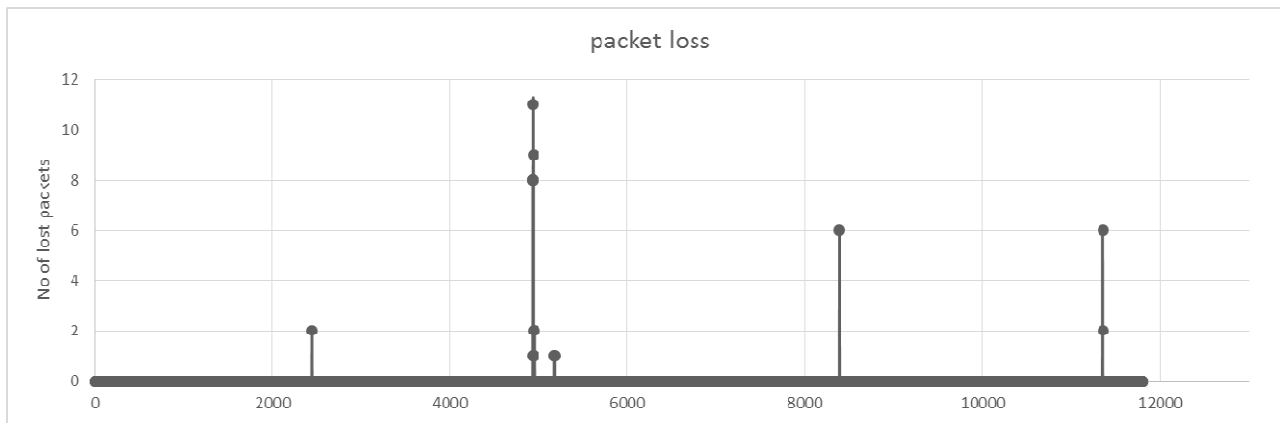


Figure A.1-2: Packet loss pattern PLR = 0,66%

A.2 VT-LTE OTT

Packet loss statistics are plotted vs. packet index (X-axis).

VT-LTE OTT Test1 error pattern.

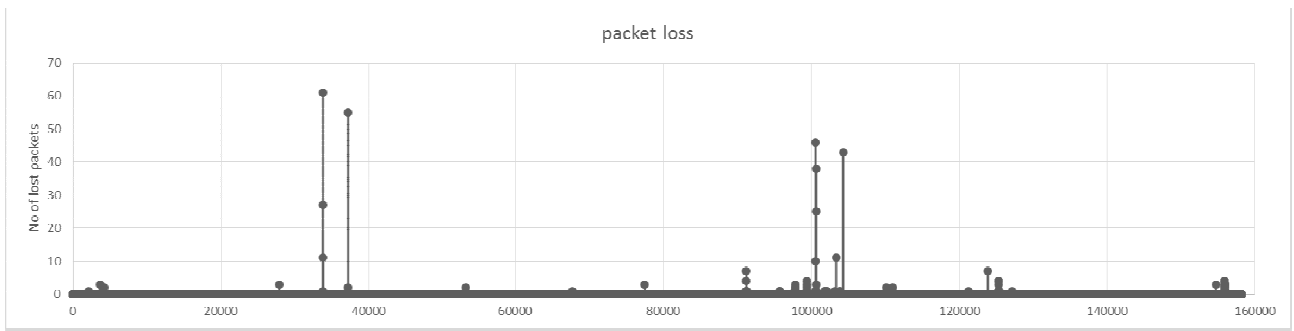


Figure A.2-1: Packet loss pattern PLR = 0,30%

VT-LTE OTT Test2 error pattern.

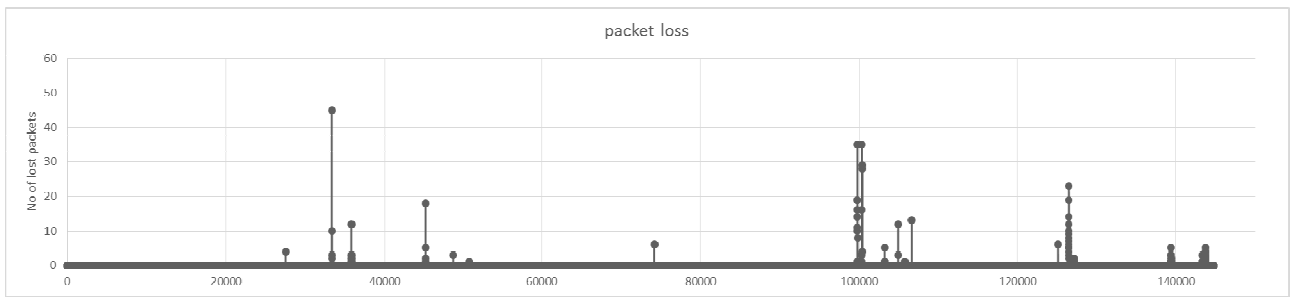


Figure A.2-2: Packet loss pattern PLR = 0,43%

VT-LTE OTT Test3 error pattern.

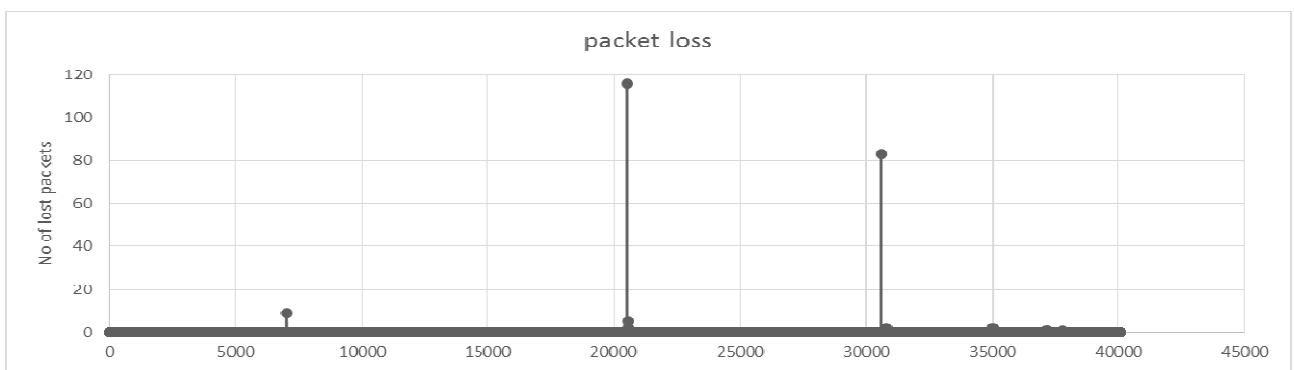


Figure A.2-3: Packet loss pattern PLR = 0,56%

VT-LTE OTT Test4 error pattern.

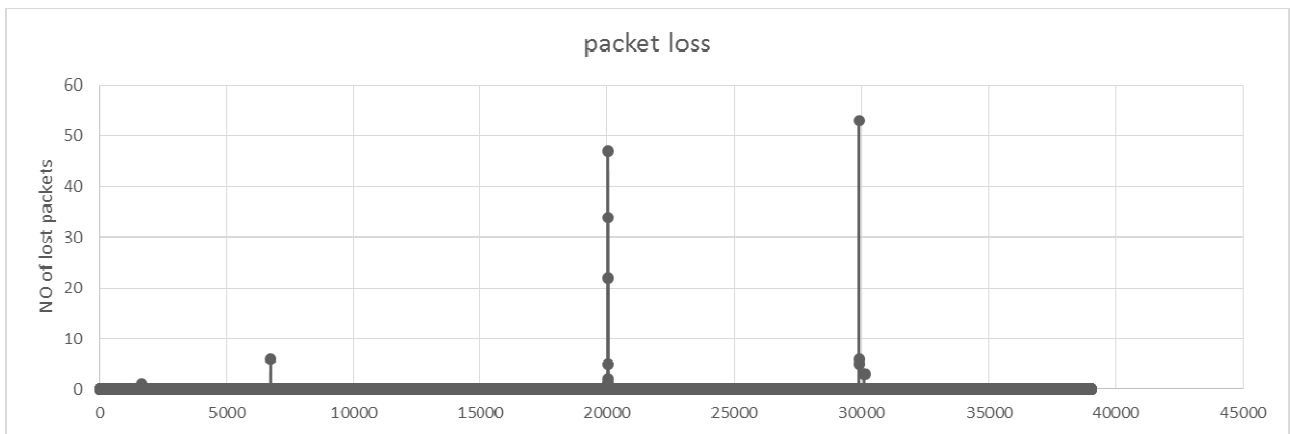


Figure A.2-4: Packet loss pattern PLR = 0,48%

A.3 VT-Wifi

Packet loss statistics are plotted vs. packet index (X-axis).

VT-Wifi Test1 error pattern.

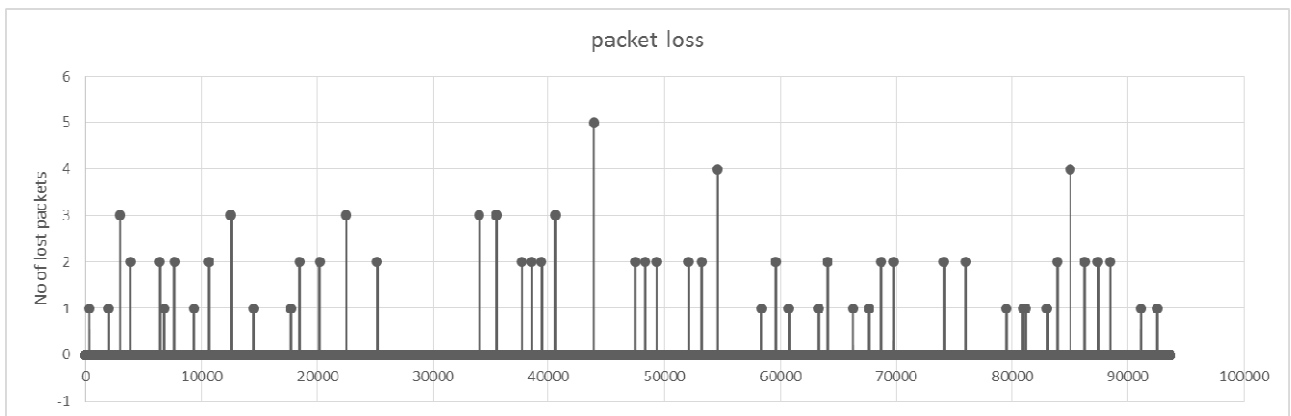


Figure A.3-1: Packet loss pattern PLR = 0,10%

VT-Wifi Test2 error pattern.

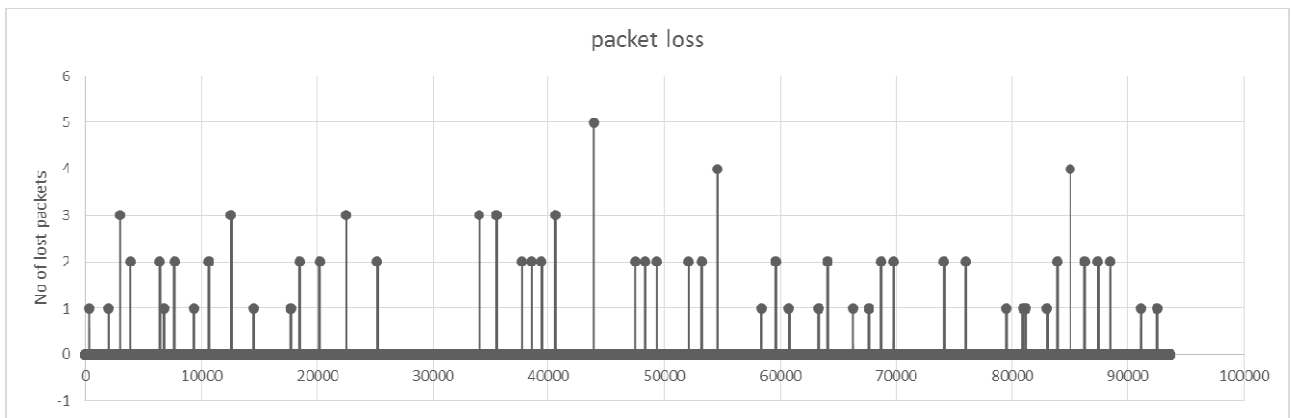


Figure A.3-2: Packet loss pattern PLR = 0,11%

VT-Wifi Test3 error pattern.

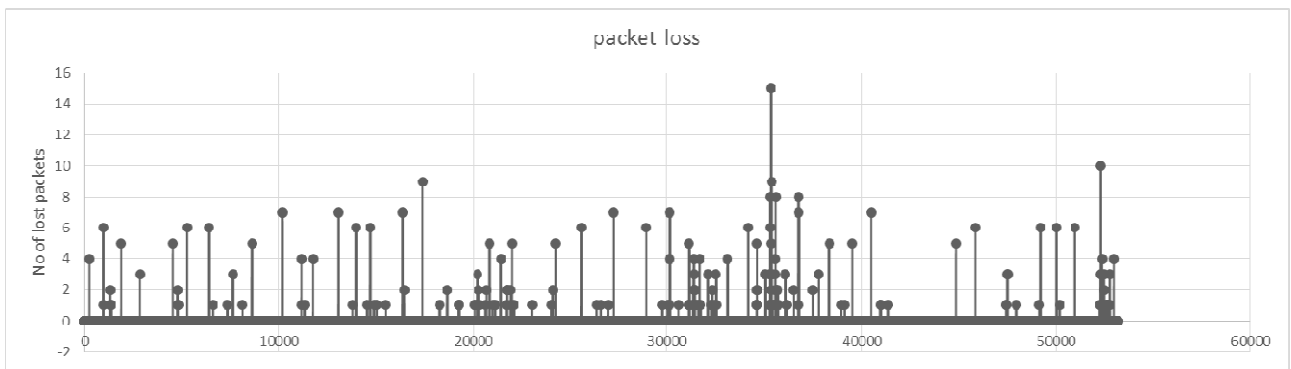


Figure A.3-3: Packet loss pattern PLR = 0,92%

VT-Wifi Test4 error pattern.

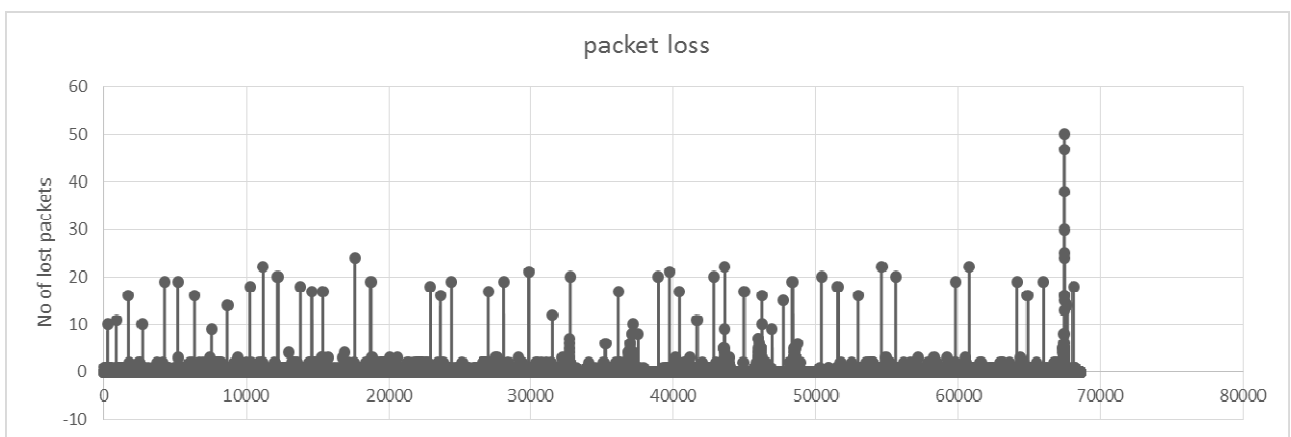


Figure A.3-4: Packet loss pattern PLR = 5,02%

VT-Wifi Test5 error pattern.

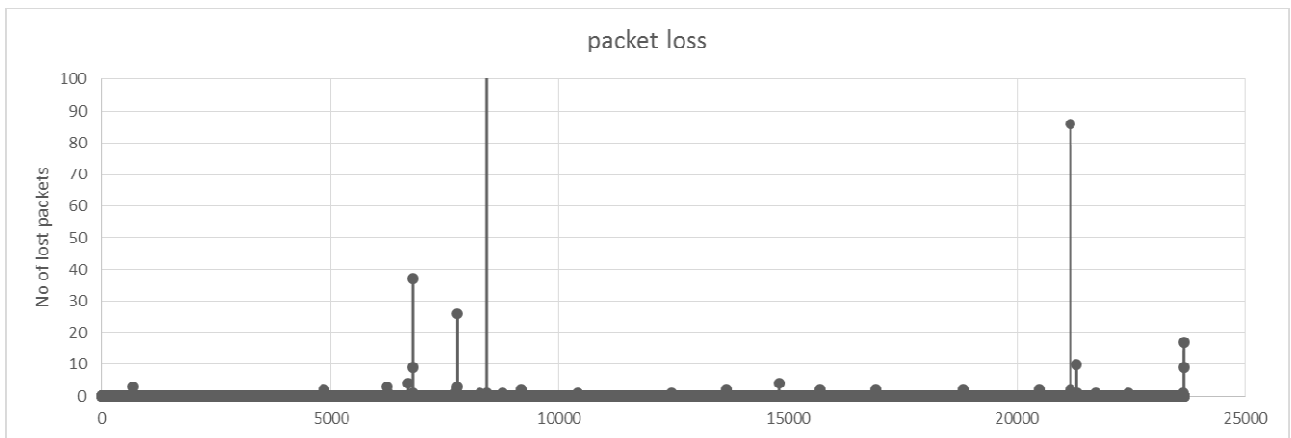


Figure A.3-5: Packet loss pattern PLR = 0,13%

VT-Wifi Test6 error pattern.

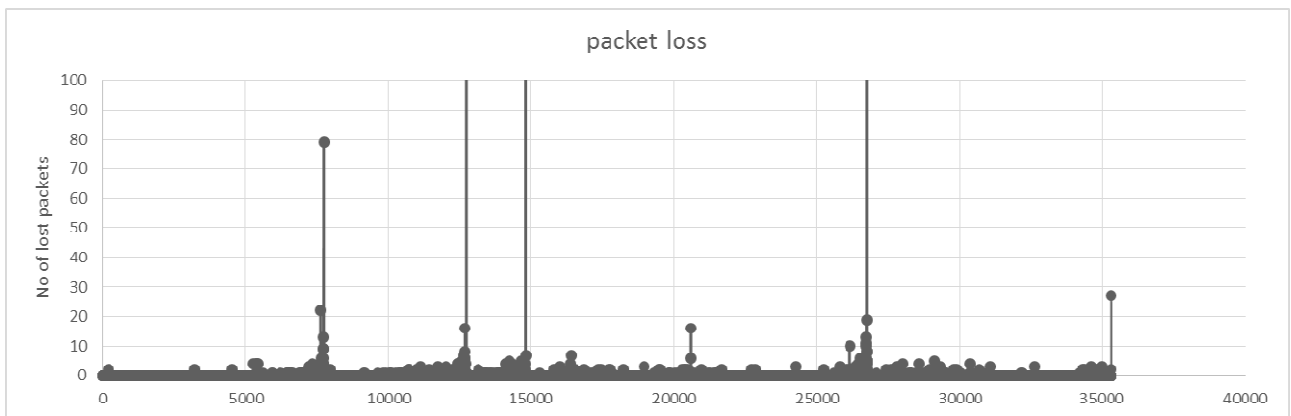


Figure A.3-6: Packet loss pattern PLR = 0,21%

VT-Wifi Test7 error pattern.

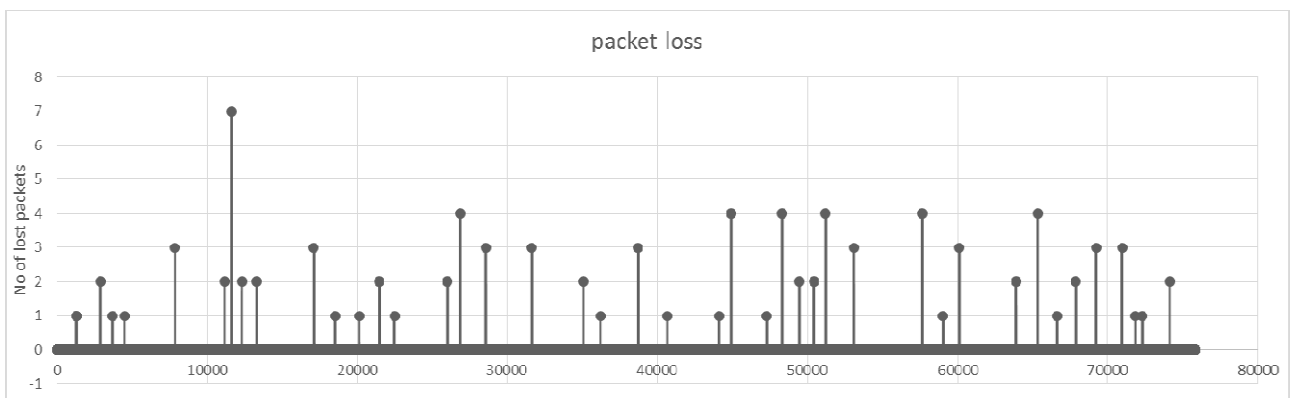


Figure A.3-7: Packet loss pattern PLR = 1,60%

VT-Wifi Test8 error pattern.

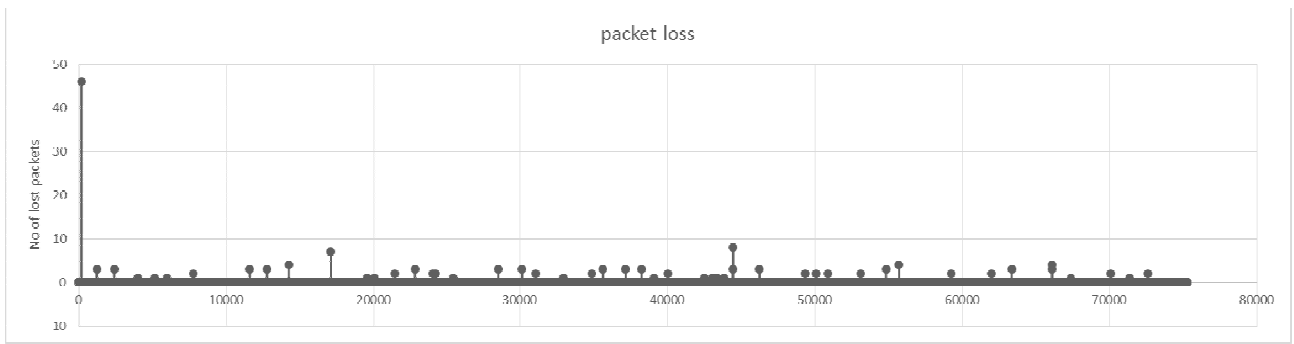


Figure A.3-8: Packet loss pattern PLR = 4,75%

VT-Wifi Test9 error pattern.

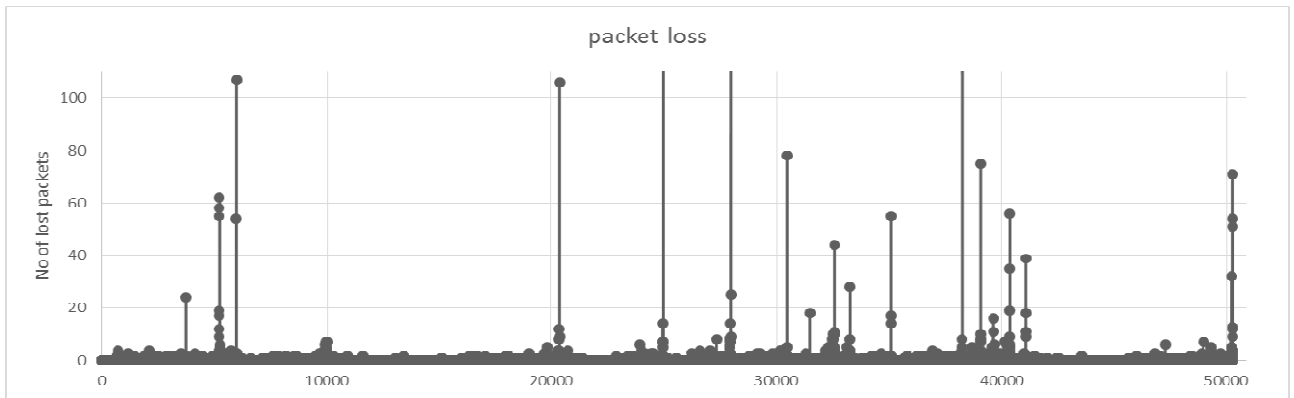


Figure A.3-9: Packet loss pattern PLR = 7,19%

Annex B: Change history

Change history							
Date	TSG SA#	TSG Doc.	CR	Rev	Subject/Comment	Old	New
2015-06	68	SP-150219			Version 1.0.0 presented at TSG SA#68 for information	n/a	1.0.0
2015-09	69	SP-150452			Version 2.0.0 presented at TSG SA#69 for approval	1.0.0	2.0.0
2015-09	69				Version 13.0.0	2.0.0	13.0.0
2016-12					LTE logo updated	13.0. 0	13.0.1

History

Document history		
V13.0.0	March 2016	Publication (withdrawn)
V13.0.1	January 2017	Publication