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Contents

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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.

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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.

[9] Q. Huynh-Thu, M. Ghanbari, "Impact of Jitter and Jerkiness on Perceived Video Quality", Proc. of the Second International Workshop on Video Processing and Quality Metrics for Consumer Electronics (VPQM), 2006.

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[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 Allicance

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: Thesevalues 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:

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

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

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

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.