International Telecommunication Union

ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

G.109

Amendment 1
(01/2007)

SERIES G: TRANSMISSION SYSTEMS AND MEDIA, DIGITAL SYSTEMS AND NETWORKS

International telephone connections and circuits – General definitions

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Definition of categories of speech transmission quality

Amendment 1: New Appendix I – The E-model-based quality contours for predicting speech transmission quality and user satisfaction from time-varying transmission impairments

ITU-T Recommendation G.109 (1999) – Amendment 1

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ITU-T Recommendation G.109

Definition of categories of speech transmission quality

Amendment 1

New Appendix I – The E-model-based quality contours for predicting speech transmission quality and user satisfaction from time-varying transmission impairments

Source

Amendment 1 to ITU-T Recommendation G.109 (1999) was agreed on 25 January 2007 by ITU-T Study Group 12 (2005-2008).

FOREWORD

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New Appendix I – The E-model-based quality contours for predicting speech transmission quality and user satisfaction from time-varying transmission impairments

(This appendix does not form an integral part of this Recommendation)

I.1 Abstract

This appendix introduces contours of speech transmission quality (or contours of user satisfaction) that can be used to predict speech transmission quality from time-varying transmission impairments. Quality contours are derived from the ITU-T E-model [ITU-T G.107] upon reducing it to the transport layer only (i.e., with assumed default values characterizing perfect terminals). The shape of quality contours is determined by the Delay Impairment I_{dd} that covers loss of interactivity and the Effective Equipment Impairment $I_{e\text{-}eff}$ that covers information loss due to encoding scheme and packet loss. The proposed quality contours determine the rating factor R for all possible combinations of packet loss (assuming a given encoding scheme) and mouth-to-ear delay (assuming echo-free connections). Quality contours can be used in cross-layer optimization of various communications layers (e.g., adaptive playout scheduling at the application layer, traffic differentiation at the MAC layer) when predicting end-to-end speech transmission quality from time-varying transmission impairments.

I.2 The E-model reduced to the transport layer

The ITU-T E-model combines individual impairments (loss, delay, echo, codec type, noise, etc.) due to both the signal's properties and the network characteristics into a single, overall measure of conversational voice quality called the R factor:

$$R = R_o - I_s - I_d - I_{e-eff} + A \qquad \text{(I-1)}$$

The calculations of the rating factor R are based on a number of input parameters and include complex mathematical formulas. To assist with calculations, ITU has established a number of default parameters (Table 2 in [ITU-T G.107]). If all parameters that do not depend on the transmission over the network are set to the default values (i.e., assuming perfect terminals) or ideal ones (e.g., TELR which is set to infinity), the expression for the R factor can be reduced to the transport layer as follows:

$$R = R_o - I_{dd} - I_{e-eff} \qquad \text{(I-2)}$$

In this equation, I_{dd} is the Delay Impairment factor which captures the effect of transmission delay (mouth-to-ear) in echo-free connections and $I_{e\text{-}eff}$ is the Effective Equipment Impairment factor which captures the effect of information loss due to both encoding scheme and packet loss. In the context of VoIP transmission assessment, both impairments are relevant. Since both impairments are isolated, their contributions to the quality degradation can be studied separately.

NOTE – All figures displayed in this appendix have been calculated with the current (2005) version of the E-model. The most recent version should be used in conjunction with this appendix.

I.2.1 Delay Impairment I_{dd}

Delay Impairment I_{dd} represents impairment due to absolute transmission delay (e.g., mouth-to-ear) in echo-free connections. [ITU-T G.107] gives a fully analytical expression for calculating I_{dd}. Figure I.1 shows calculated Delay Impairment I_{dd} as a function of absolute transmission delay T_a.

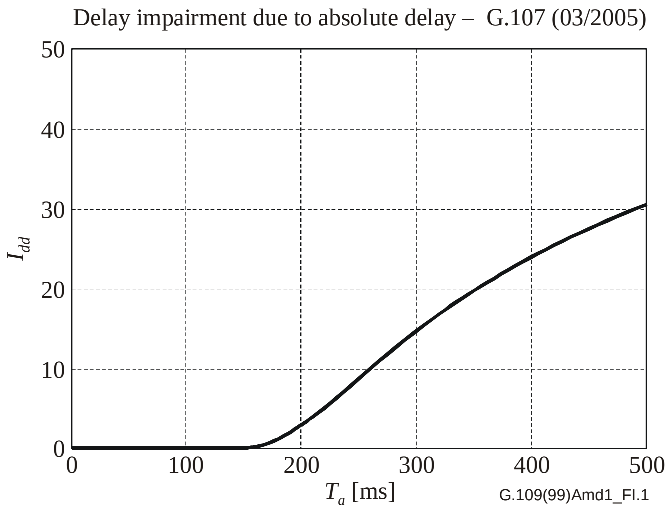

Figure I.1 – Delay impairment I_{dd} as a function of absolute transmission delay T_a

I.2.2 Effective Equipment Impairment $I_{e\text{-}eff}$

The Effective Equipment Impairment $I_{e\text{-}eff}$ captures the effect of information loss due to both encoding scheme and packet loss. The current version of the E-model is able to quantify the effect of random (and for general predictions also bursty) packet loss in narrow-band speech transmission [ITU-T G.107] and [ITU-T G.113ApI]. Figure I.2 shows for several codecs how the Effective Equipment Impairment increases as packet loss increases. This figure deals only with one specific packetization interval per codec: 10 ms for G.711 random loss, 20 ms for G.711 bursty loss, G.729A/E (both bursty and random loss), GSM-EFR, and 30 ms for G.723.1 (random loss only).

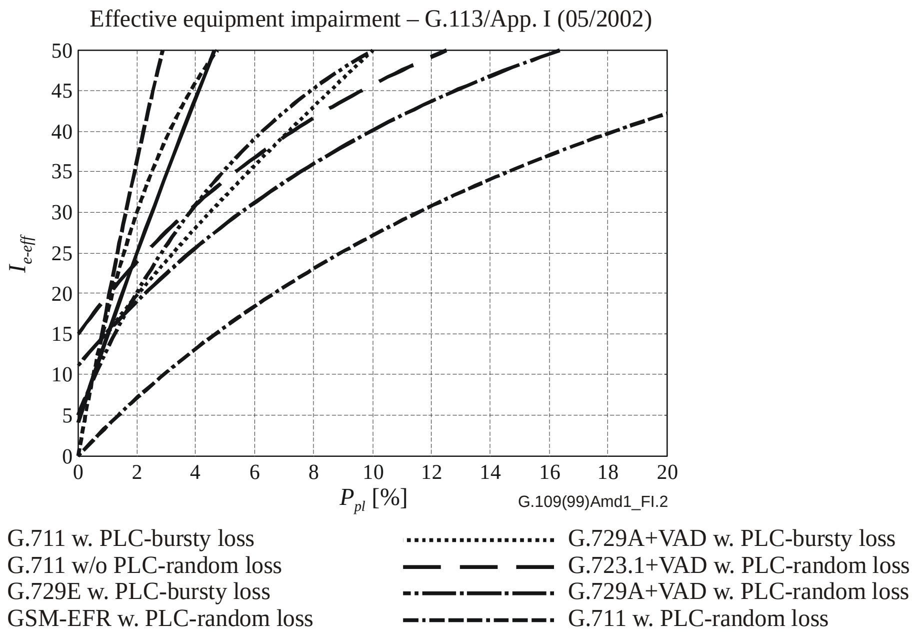

Figure I.2 – Effective Equipment Impairment $I_{e\text{-}eff}$ as a function of packet loss P_{pl}

I.3 Contours of speech transmission quality (contours of user satisfaction)

The R factor reduced to transmission layer can be calculated from transmission impairments (e.g., I_{dd} and $I_{e\text{-}eff}$) described in clause I.2 as follows:

$$R = R_0 - I_{dd}(T_a) - I_{e-eff}(P_{pl}) \tag{I-3}$$

Equation I-3 can be viewed as a function of two variables: absolute transmission delay T_a (e.g., mouth-to-ear delay) and Packet-loss Probability P_{pl} (with assumed encoding scheme). By placing T_a on the X-axis and P_{pl} on the Y-axis it is possible to draw values of R for all possible combinations of loss and delay. Figure I.3 shows the contours of Rating Factor R calculated for the G.711 encoding scheme (random packet loss) with Packet Loss Concealment (PLC) implemented.

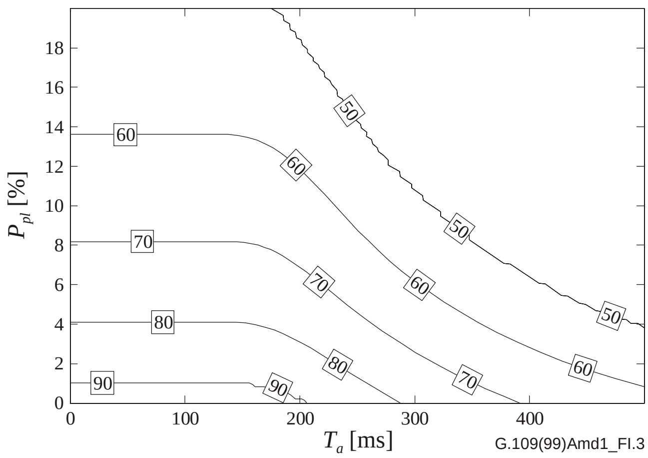

Figure I.3 – Contours of rating factor *R* calculated for G.711 encoding scheme (random packet loss, PLC)

[ITU-T G.109] defines categories of speech transmission quality and categories of user satisfaction in terms of ranges of R. Consequently, it is possible to draw contours of speech transmission quality to be expected by end user (contours of user satisfaction) as shown in Figure I.4.

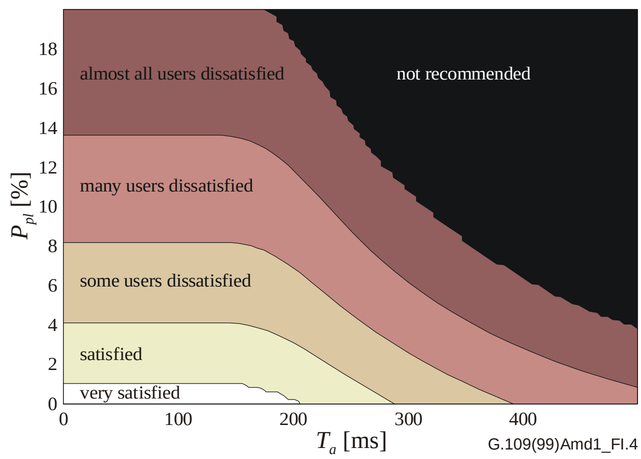

Figure I.4 – Contours of user satisfaction (contours of speech transmission quality) for G.711 encoding scheme (random packet loss, PLC)

Figures I.5 to I.12 show contours of quality calculated for a number of encoding schemes. They are based on the ITU-T E-model analytical formulas (Equations 3-27, 3-28, 3-29 in [ITU-T G.107]) and tabulated values (Tables I.3, I.4 in [ITU-T G.113ApI]) for calculating I_{dd} and $I_{e\text{-}eff}$.

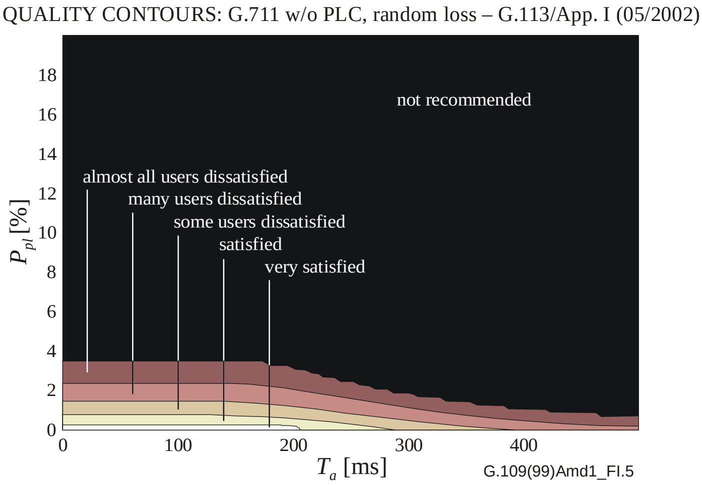

Figure I.5 – Quality contours calculated for G.711 encoding scheme (random packet loss) without PLC

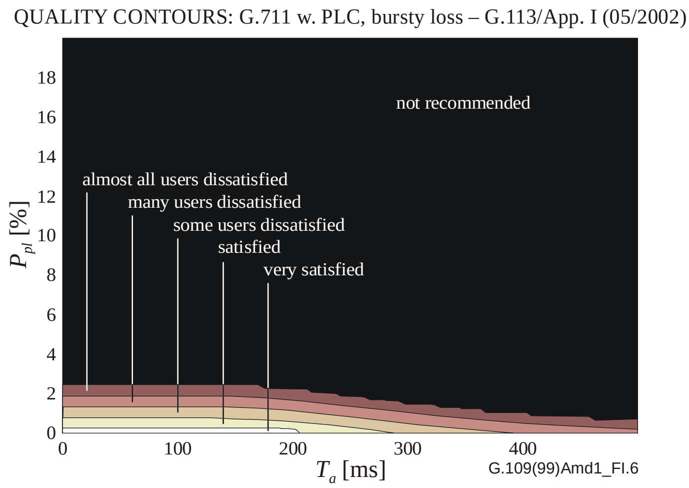

Figure I.6 – Quality contours calculated for G.711 encoding scheme (bursty packet loss) with PLC

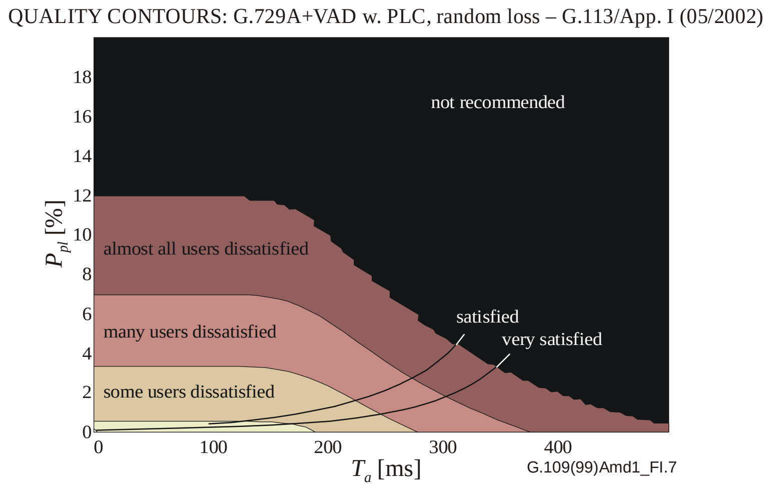

Figure I.7 – Quality contours calculated for G.729A encoding scheme (random packet loss) with PLC

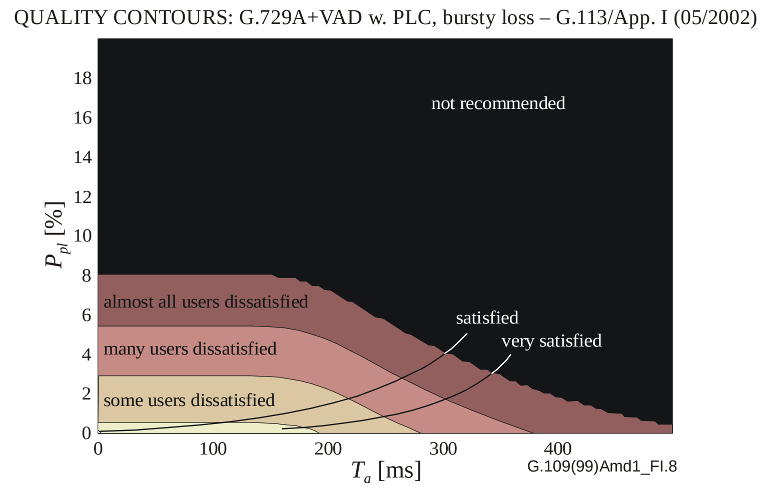

Figure I.8 – Quality contours calculated for G.729A encoding scheme (bursty packet loss) with PLC

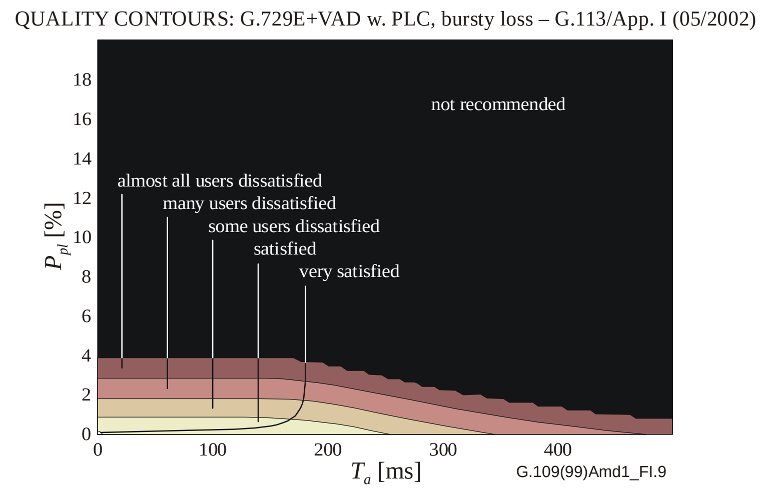

Figure I.9 – Quality contours calculated for G.729E encoding scheme (bursty packet loss) with PLC

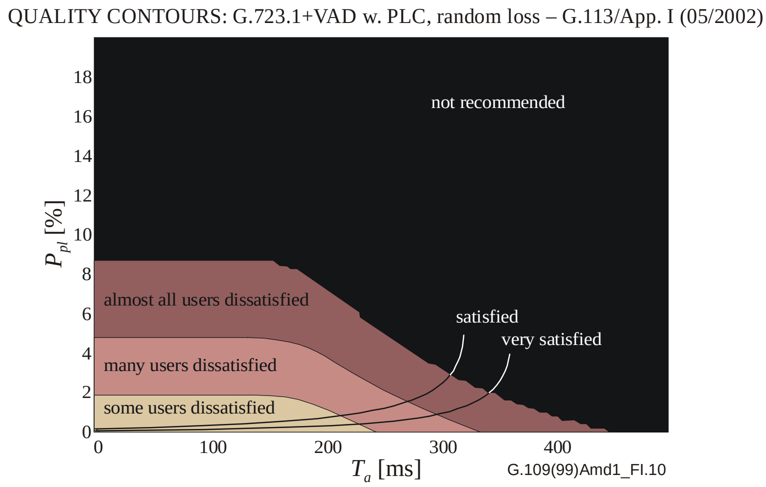

Figure I.10 – Quality contours calculated for G.723.1 encoding scheme (random packet loss) with PLC

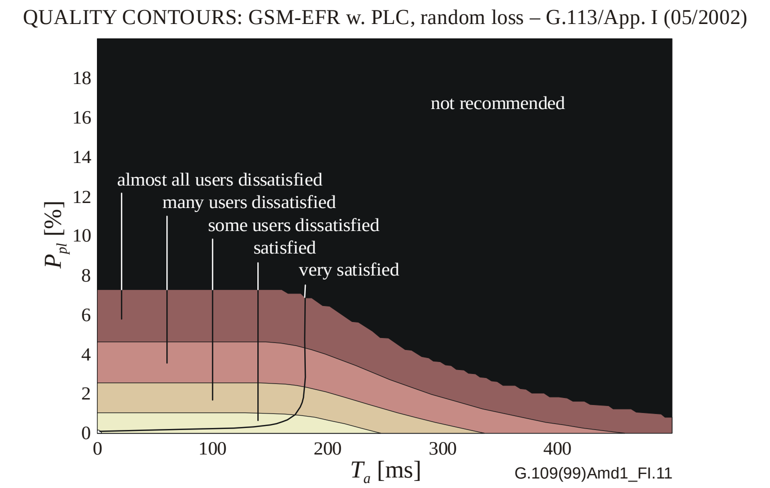

Figure I.11 – Quality contours calculated for GSM 06.60 (EFR) encoding scheme (random packet loss) with PLC

I.4 Predicting speech transmission quality from time-varying transmission impairments with the use of quality contours

Quality contours are useful in predicting speech transmission quality (and user satisfaction) from time-varying transmission impairments. The procedure of predicting speech transmission quality with the use of quality contours is described below and illustrated in Figure I.12:

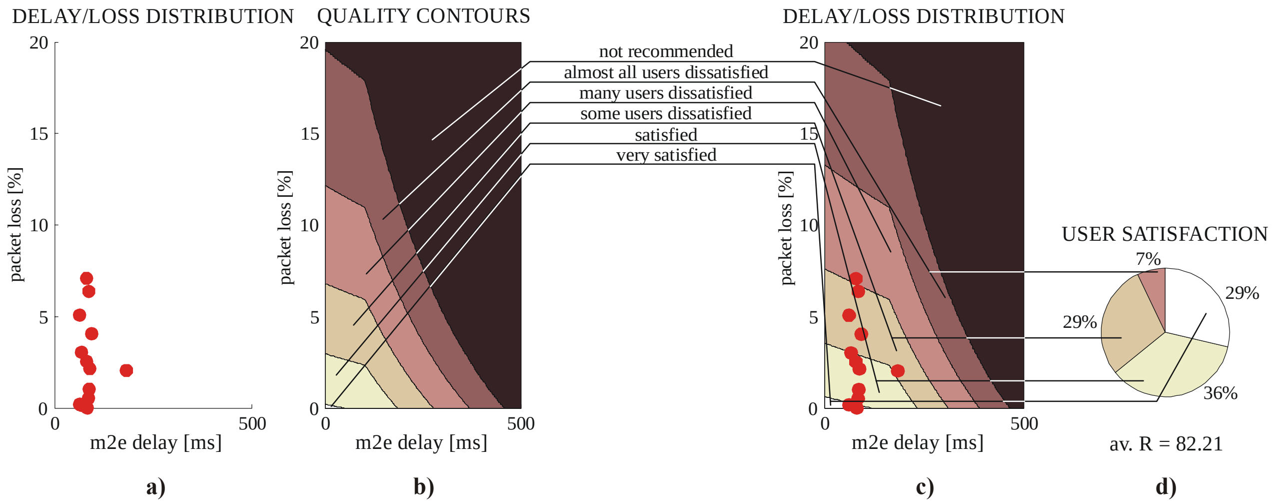

Figure I.12 – Predicting user satisfaction from time-varying transmission

a) playout delays (i.e., mouth-to-ear delays) and packet loss are calculated over non-overlapping time windows of 10 seconds (or for every talkburst) at the output of the de-jitter buffer;

b) quality contours are chosen for a specific encoding scheme;

c) playout delays and packet losses are mapped onto a chosen quality contour;

d) overall user satisfaction regarding speech transmission quality in the form of pie chart can be derived from the distribution of playout delays on quality contours.

With quality contours, the impact of delay and packet loss on conversational speech quality can be studied in two ways: either as the combined effect of loss and delay on overall quality, or as individual contributions of packet loss to speech degradation and playout delay to interactivity degradation. This is especially useful in the process of parameter tuning when a trade-off exists between packet delays and loss and efforts are focused on finding the operating point where conversational quality is maximized. An additional advantage of this method is to show percentages of user satisfaction instead of giving one quality score. The proposed method of predicting user satisfaction from time-varying transmission impairments has already shown to be particularly effective in evaluating various playout buffer algorithms [Narbutt 1], [Narbutt 2] and in assessing VoIP performance in WLAN systems [Narbutt 3], [Narbutt 4].

I.4.1 Example: Choosing adaptive de-jitter playout algorithm

Traditionally, the choice of a de-jitter buffer algorithm was based purely on the trade-off between buffering delay and the resulting late-packet loss. Given that the purpose of playout buffering is to improve conversational speech quality, a more informed choice of algorithm can be made by considering its effect on user satisfaction.

The method could be used to predict user satisfaction for a number of playout algorithms as follows: For an hour of transmission, all experimental data (packet arrival times, timestamps, sequence numbers, and marker bits) are collected at the receiver and processed later (offline) with a program that simulates various playout algorithms' behaviour (see Figure I.13).

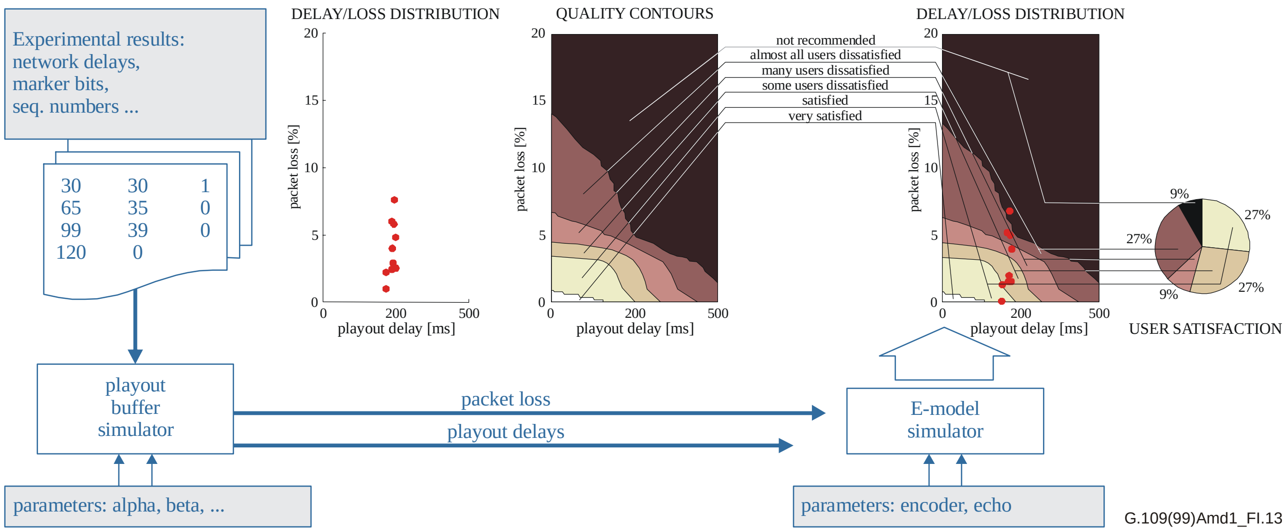

Figure I.13 – Evaluating playout buffer algorithms with the use of the proposed method

The playout buffer simulator calculates the average playout delays (i.e., mouth-to-ear) and resulting packet loss for non-overlapping time windows of 10 seconds over the transmission. These time-varying transmission impairments are the input values of E-model simulator that predicts overall user satisfaction. Figure I.14 compares the performance of six various playout buffer algorithms in terms of percentages of user satisfaction [Narbutt 1].

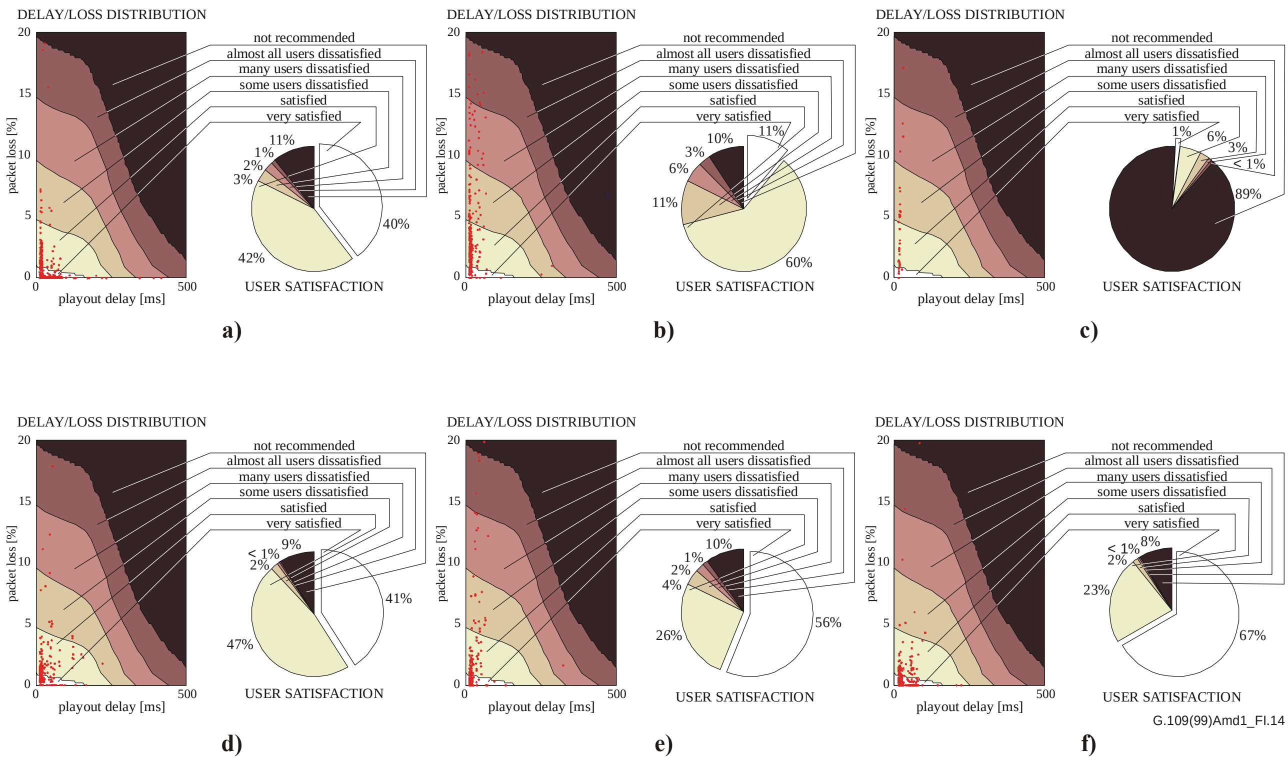

Figure I.14 – Performance of various playout algorithms in terms of user satisfaction: a) Ramjee's alg. with α = 0.99, b) Ramjee's alg. with α = 0.9, c) "Concord" alg., d) Moon's alg., e) Bolot's alg., f) the "dynamic α" alg.

I.5 Appendix I references

[ITU-T G.107] ITU-T Recommendation G.107 (2005), *The E-Model, a computational model for use in transmission planning*.

[ITU-T G.109] ITU-T Recommendation G.109 (1999), *Definition of categories of speech transmission quality*.

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[Narbutt 1] NARBUTT (M.), DAVIS (M.): Assessing the Quality of VoIP Transmission Affected by Playout Buffer Scheme, *Proc. of the ETSI/IEEE Measurement of Speech and Audio Quality in Networks Conference 2005 (MESAQIN 2005)*, Prague, June 2005.

[Narbutt 2] NARBUTT (M.), KELLY (A.), MURPHY (L.), PERRY (P.): Adaptive VoIP Playout scheduling: Assessing User Satisfaction, *IEEE Internet Computing Magazine*, vol. 09, No. 4, July/August 2005.

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[Narbutt 4] NARBUTT (M.), DAVIS (M.): The Experimental investigation on VoIP performance and the resource utilization in 802.11b WLANs, *IEEE Conference on Local Computer Networks (LCN'06)*, Tampa, November 2006.

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