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Title: Comparison of 5008 Annex E with Average Loss Model

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Document for:

Decision:	
Discussion:	**X**
Meeting Report:	
Liaison:	
Information:	X

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1. Decision/Action Requested

Working Group 5 is asked to review this document when considering whether to make 5008 Annex E Normative.

2. Introduction

This document provides a comparison between the Average Packet Loss model implied by G.113 and the Burst Packet Loss model described in 5008 Annex E.

It is recognized that the packet loss process that occurs in an IP network is often bursty in nature, and that the loss distribution can be represented using a Markov model. When a transmission quality factor is estimated using average packet loss then inaccurate results can be obtained. For example, consider how a a packet loss rate of 2% may affect a Voice over IP system using packet loss replacement. If the probability of packet loss is independent then the packet loss replacement method can render the effects of loss almost unnoticeable to the listener. If the 2% loss occurs in bursts however then the loss rate within a burst could be quite high in which case the listener would hear distinct periods of noise or distortion.

The “ideal” method of determining the effects of packet loss would be to compute transmission quality in real time, considering all the effects described in Annex E. This represents a substantial computational load which conflicts with the increasing trend towards running many CODECs on a single high performance DSP and increasing price/cost pressure on VoIP end systems. This document provides a comparison between the approach to transmission quality estimation described in Annex E, the Average Packet Loss model and the ideal Real Time model.

3. Simulation Description

A burst loss process was simulated using a Gilbert model with state transition and packet loss probabilities selected randomly for each simulation run. This produced a wide range of packet loss distributions. For each simulation run the algorithm described in Annex E, a Real Time algorithm and an Average Packet Loss algorithm were used to produce estimated I_e factors.

The Annex E algorithm was applied with g_{min} set to 32.

The Real Time algorithm determined an Ie(ideal) factor for each 1 second time interval, applied an exponential shaping factor, as described in Annex E, and then averaged the results at the end of the call

The Average Packet Loss algorithm measured the average packet loss for the entire call and then determined an Ie(avge) factor.

In all cases a G.723.1 CODEC was assumed.

4. Simulation Results

The attached chart shows the results for 1000 successive simulation runs. Chart 1 shows the distribution of Ie computed using the Annex E algorithm. Chart 2 shows the unit error between the Annex E algorithm and Real Time algorithm, and between the Average algorithm and Real Time algorithm.

5. Discussion

The results show that under the simulated conditions the Annex E algorithm does obtain a more consistent result than the Average algorithm. It is also apparent that the Annex E algorithm sometimes over-estimates the Ie value – this is possibly due to the implicit assumption that steady state conditions exist, whereas there is in practice a ramp up period at the start of a call.

6. Conclusion

Under the simulated conditions the 5008 Annex E algorithm provides a more accurate and consistent approximation to the “ideal” value of Ie computed in real time.

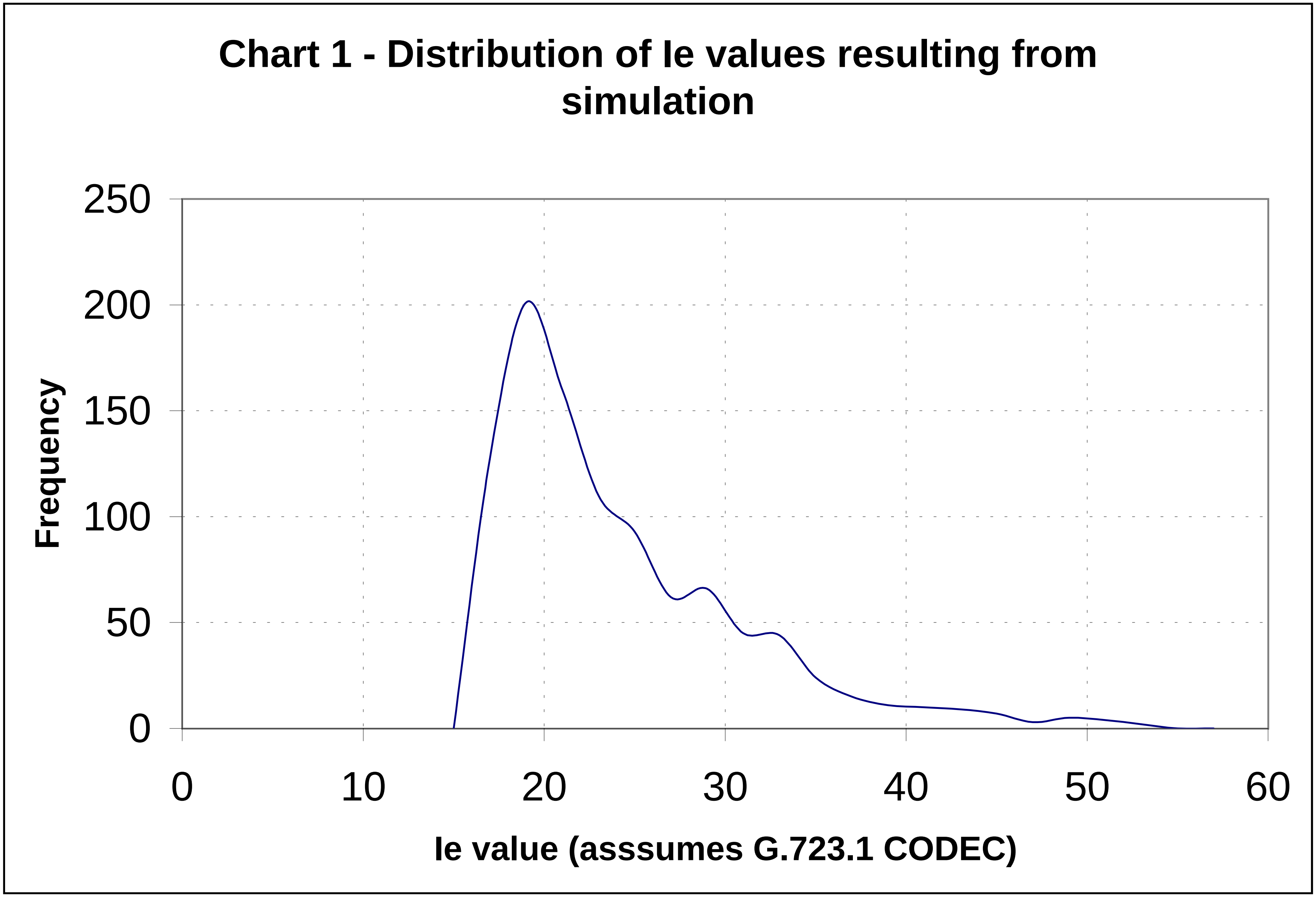
Chart 1 - Distribution of Ie values resulting from simulation
250
200
150
100
50
0
Frequency
0
10
20
30
40
50
60
Ie value (asssumes G.723.1 CODEC)

Chart 2 - Comparison of Annex E and Average Packet Loss approach to determining Ie