Source: Telchemy, Inc

Title: Proposal for Passive QoS Monitoring Methodology for Voice over IP

Notice: The author of this document declares that ETSI **may** make the document publicly available.

Document for:

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

Contact details:

First Name, Last Name Alan Clark
e-mail: alan@telchemy.com

1. Requested actions

This document is addressed to WG 5

WG5 is requested to include this document as an annex to DTS5008 for the purpose of describing how quality of service can be monitored in a passive/ non-intrusive manner.

2. Introduction

This document describes a method of passively monitoring Voice over IP Quality of Service. The method produces an R factor using an extension to the E-Model described in ETR250 that incorporates the effects of packet loss burstiness and "recency". The implementation of this method within a Voice over IP network permits large numbers of voice calls to be monitored with minimal impact on network traffic.

IP network packet loss distribution can be modeled using a Markov process. The resulting model can be used in both analytical and numerical performance estimation and has well known and understood properties.

The channel is assumed to have high packet loss (burst) and low packet loss (gap) conditions. During the Voice over IP call the passive QoS monitor counts packet loss events and inter-loss gaps using a simple packet loss event driven algorithm. At the end of the call, or on request from a service management system, the transition probabilities of the Markov model are determined and used to compute an R factor for the call.

3. Packet Loss Model

If the number of packets received between two successive lost packets is less than a minimum value g_{min} then the sequence of the two lost packets and the intervening received packets is regarded as part of a burst. If a sequence of g_{min} or more packets are correctly received the sequence is regarded as being part of a gap.

The Markov model is defined as having the following states and associated transitions:

State 1 - gap - no loss

p_{11} - *packet received*

p_{13} - *packet loss (start of burst)*

p_{14} - isolated packet loss

State 2 - burst - no loss

p_{22} - packet received within burst

p_{23} - packet lost within burst

State 3 - burst - packet loss

p_{31} - packet received (end of burst)

p_{32} - packet received within burst

p_{33} - packet lost

State 4 - gap - packet loss

p_{41} - packet received

This model can be constructed either by accumulating packet loss information during fixed sampling intervals or at packet loss events. The latter approach is described below, and has the advantage of a low computational load for typical packet loss rates.

Assume a counter *pkt* tracks the number of received packets and that an event can be generated if a packet loss is detected:

```
Packet loss event->
    if pkt >= g_min then
        if er = 1 then
            c14 = c14 + 1
        else
            c13 = c13 + 1
        er = 1
        c11 = c11 + pkt
    else
        er = er + 1
        if pkt = 0 then
            c33 = c33 + 1
        else
            c23 = c23 + 1
            c22 = c22 + pkt
    pkt = 0
```

The key metrics needed for determining application performance are:-

$c_{31} = c_{13}$

$c_{32} = c_{23}$

$c_{11} = c_{11} + c_{14}$ (for simplicity - combine states 4 and 1)

$p_{11} = c_{11} / (c_{11} + c_{13})$

$p_{13} = 1 - p_{11}$

$p_{31} = c_{31} / (c_{31} + c_{32} + c_{33})$

$p_{32} = c_{32} / (c_{31} + c_{32} + c_{33})$

$p_{33} = 1 - p_{31} - p_{32}$

$p_{22} = c_{22} / (c_{22} + c_{23})$

$p_{23} = 1 - p_{22}$

$d = (p_{23}\ \ p_{31} + p_{13}\ \ p_{32} + p_{13}\ \ p_{23})$

$p_1 = p_{31}\ \ p_{23} / d$

$p_2 = p_{13}\ \ p_{32} / d$

$p_3 = p_{13}\ \ p_{23} / d$

frame size	F = frame size (in seconds)	
average packet loss rate	L = 100 p3	percent
gap length	$g = F / (1 - p_{11})$	seconds
gap loss density	$D_g = 100\ c_{14} / c_{11}$	percent
burst length	b = F (1 - p1) / (p1 p13)	seconds
burst loss density	D_b = 100 p23 / (p23+p32)	percent
delay since last burst	y = F pkt	seconds

4. Determining estimated voice quality

An estimate of the published “Provisional Planning values for the Equipment Impairment Factor” is given by the equation below:

$$I_e = a_1 + a_2 D / (b + D) + c D$$

where D is the packet loss rate expressed as a percentage, b = 9.26, c = 1.34 and a_1 and a_2 are CODEC dependant parameters (for G.723.1 6k a_1=15 and a_2=34)

This can be separately applied to the packet loss rates for the gap and burst state, giving I_g and I_b.

It is generally acccepted that perceived quality does not change abruptly but exponentially “decays” from one level to another. This is intuitively obvious, as a 100mS burst of noise would be less annoying than a 10S burst. The effects are described in ITU-T SG12 D.139 which measured instantaneous and average quality for a variety of impairment profiles.

Let I_1 be the quality level at the change from burst condition I_b to gap condition I_g and let I_2 be the quality level at the change from I_g to I_b

$$I_1 = I_b - (I_g - I_2)\, e^{-b/t1} \qquad \text{where } t_1 \text{ is typically } 5$$

$I_2 = I_g + (I_1 - I_g)\, e^{-g/t2}$ where t2 is typically 15

Combining these gives

$$I_2 = (I_g (1 - e^{-g/t2}) + I_b (1 - e^{-b/t1})\, e^{-g/t2}) / (1 - e^{-b/t1-g/t2})$$

Integrating the expressions for I_1 and I_2 to give a time average gives

$$I_{av} = (b\, I_b + g\, I_g - t_1 (I_b - I_2) (1 - e^{-b/t1}) + t_2 (I_1 - I_g) (1 - e^{-g/t2})) / (b + g)$$

5. Recency Effect

It has been noted by a number of researchers that the perceived quality of a call varies with the location of impairments. Impairments occurring late in a call have more effect than those occurring early in the call.

ANSI T1A1.7/98-031 described an experiment in which both mutes and noise bursts were introduced at the beginning, middle and end of a 60 second call. For the "high burst" result given:-

burst at start of call	MOS = 3.82
burst at middle of call	MOS = 3.28
burst at end of call	MOS = 3.18

ITU-T SG12 D.139 conducted an experiment in which a burst of high packet loss of duration 15, 30 or 60 seconds was introduced at the start, middle and end of a 180 second call and noted similar effects.

It is proposed that a simplified "adjustment" for recency be used, to minimise complexity. The delay since the last burst of packet loss is given above as the ending value of pkt. It is assumed that the value of Ie at the end of the previous burst is given by I_1 and that the adjusted average quality approaches Iav exponentially.

$$I_{av}\ (\text{adjusted}) = I_{av} + (I_1 - I_{av})\, e^{-y/t3}$$

where y is the delay to the previous burst and t_3 is a time constant (assumed to be 30)

6. Determining R factor

It is assumed that the delay and jitter level are constant for the duration of a call. The R factor for the call can therefore be determined as:

$R = R_o - (I_{av(adjusted)} - I_J - I_D)$ where R_o is typically 94

7. Conclusion

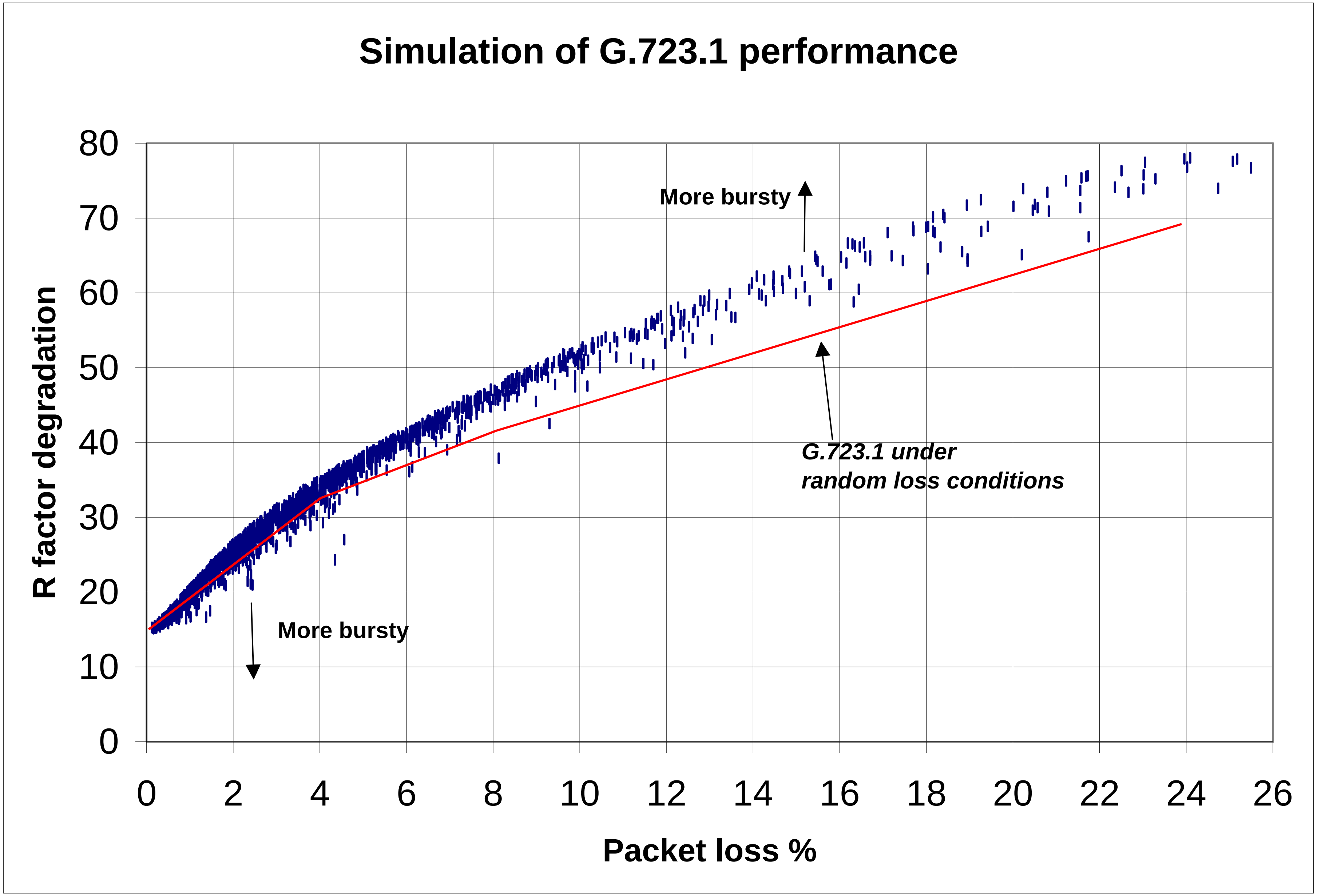
A method has been presented for passively monitoring QoS of Voice over IP connections. The method takes very little processing resource and considers effects such as packet loss burstiness and recency.

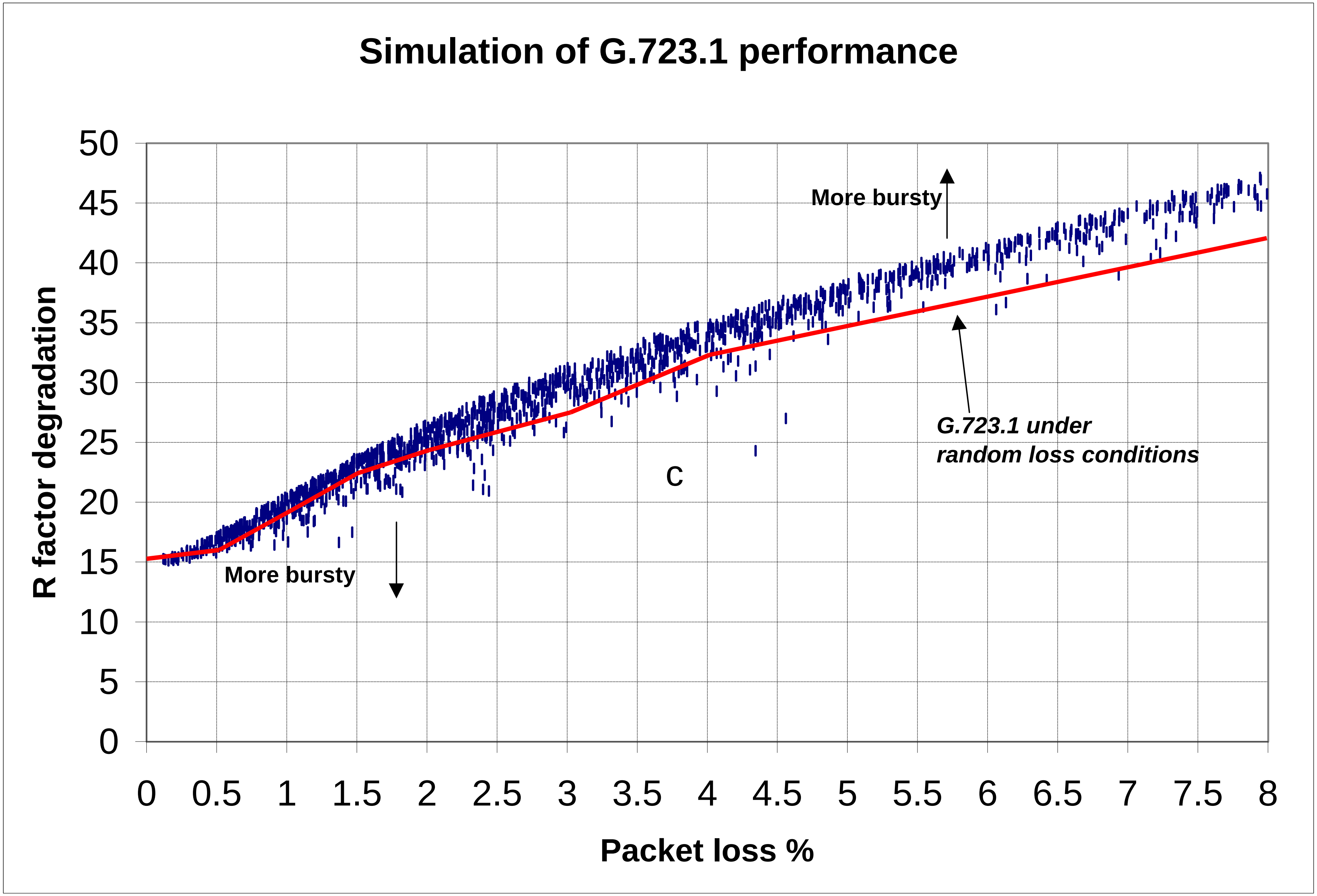
The model can be further "calibrated" as it measures effects such as consecutive packet loss which have a known impact on the operation of voice coding algorithms and which are not taken into account above.

8. References

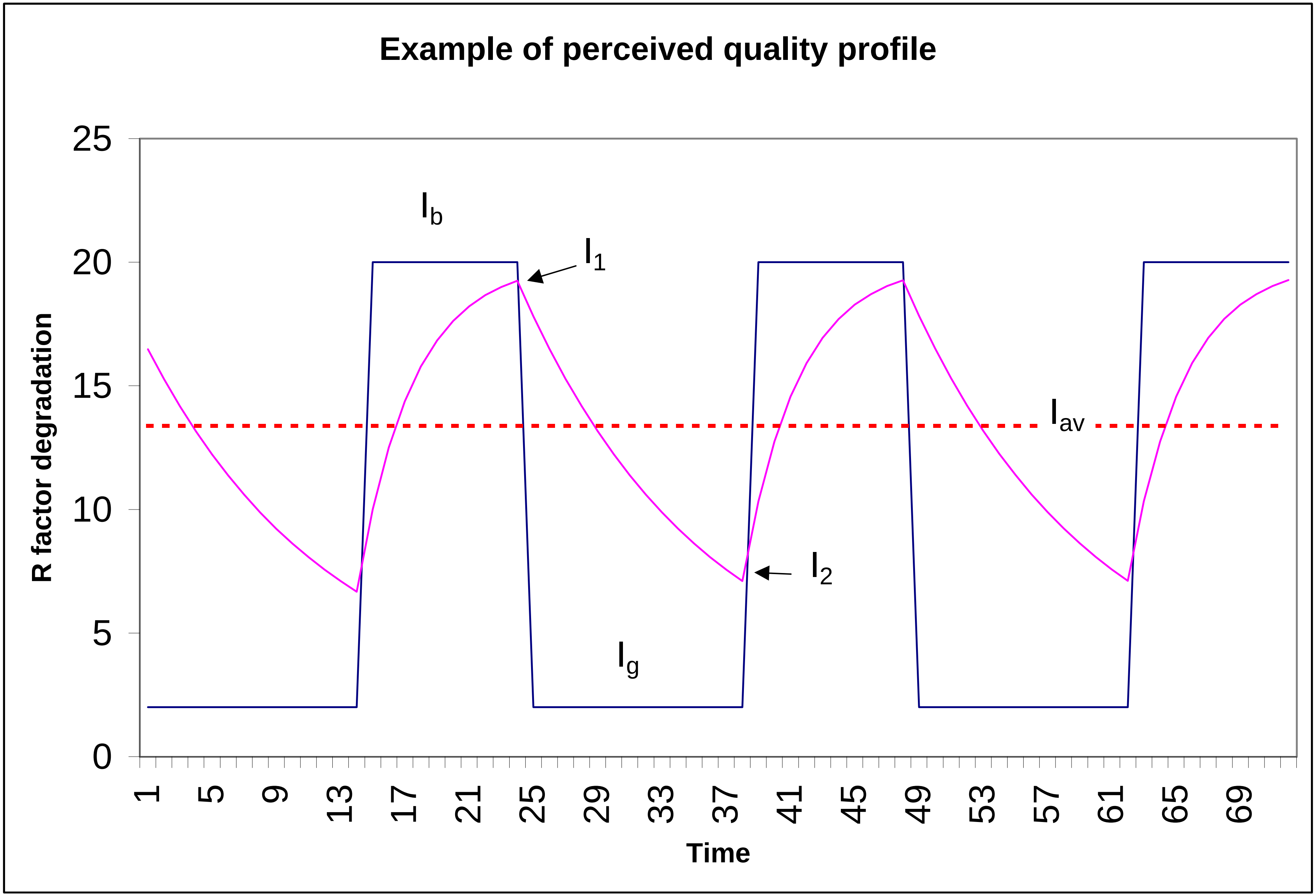
ANSI T1A1.7/98-031 Testing the quality of connections having time varying impairments, AT&T, October 1998

ITU-T SG12 D.139 Study of the relationship between instantaneous and overall subjective speech quality for time varying quality speech sequences, France Telecom, May 2000

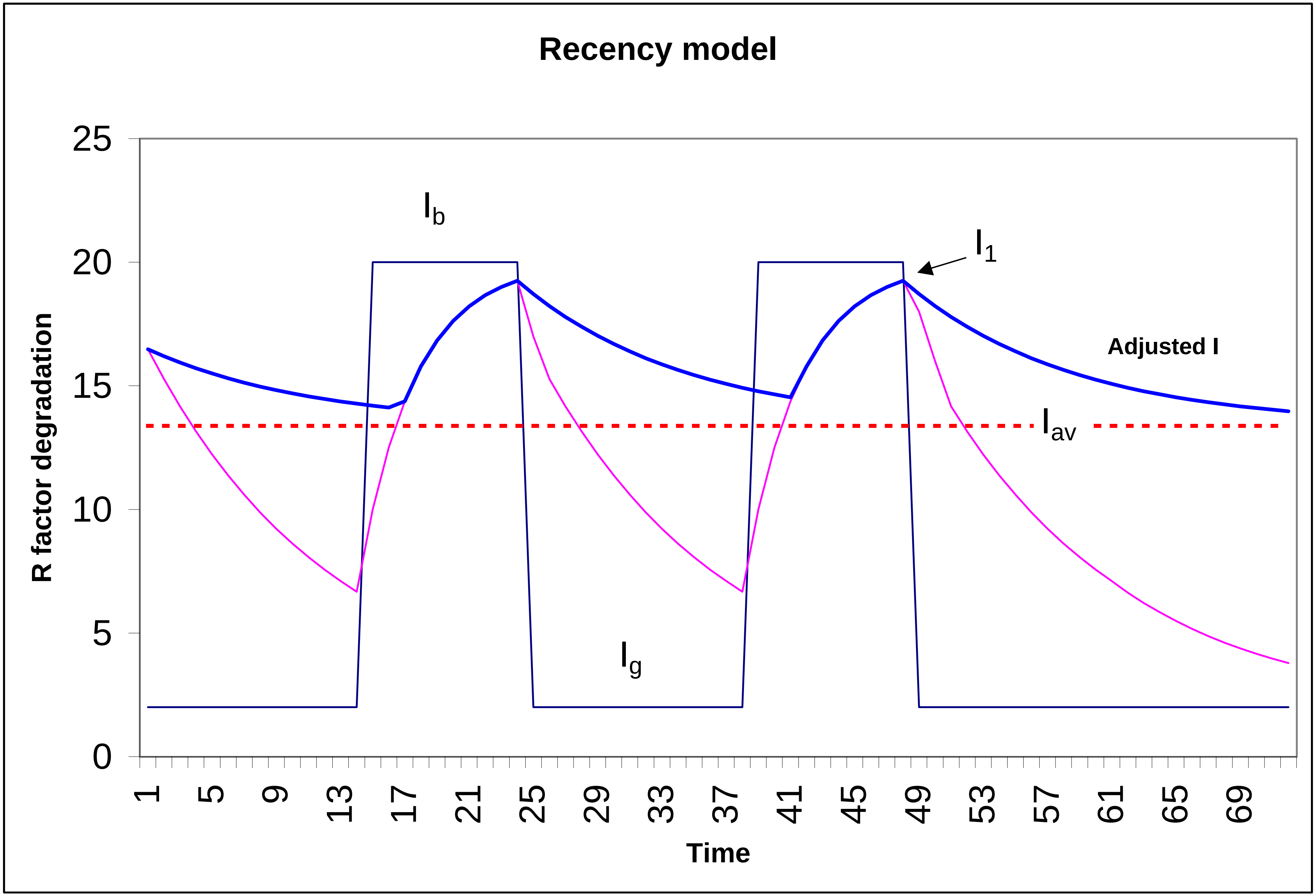

Simulation of G.723.1 performance
R factor degradation
0
10
20
30
40
50
60
70
80
0
2
4
6
8
10
12
14
16
18
20
22
24
26
Packet loss %
More bursty
More bursty
G.723.1 under
random loss conditions



Simulation of G.723.1 performance
R factor degradation
50
45
40
35
30
25
20
15
10
5
0
0
0.5
1
1.5
2
2.5
3
3.5
4
4.5
5
5.5
6
6.5
7
7.5
8
Packet loss %
More bursty
More bursty
G.723.1 under
random loss conditions
C



Example of perceived quality profile
25
20
15
10
5
0
R factor degradation
I_b
I_1
I_{av}
I_2
I_g
1
5
9
13
17
21
25
29
33
37
41
45
49
53
57
61
65
69
Time



Recency model
R factor degradation
25
20
15
10
5
0
I_b
I_1
Adjusted I
I_{av}
I_g
1
5
9
13
17
21
25
29
33
37
41
45
49
53
57
61
65
69
Time