

**Source:** Telchemy incorporated  
**Title:** Practical Call Quality Metrics  
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<b>Document for:</b>	Decision:	<b>X</b>
	Discussion:	<b>X</b>
	Meeting Report:	
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	Information:	

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## 1. Decision/Action Requested

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Update Annex E to TS 101 329-5 to redefine call quality metrics.

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## 2. References

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[1] ETSI TIPHON Draft TS 101 329 – 5 Annex E

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## 3. Introduction

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As a result of feedback from service providers on the use of TS101 329-5 Annex E it appears to be useful to provide three call quality metrics in place of the one currently calculated. It is proposed that Annex E be modified to incorporate two metrics.

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## 4. Discussion

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Discussions with service providers have identified that there are several applications for real time call quality monitoring, each of which has slightly different requirements. The primary applications are:-

- (i) Billing systems require an estimate of the call quality that would be reported by the user at the end of a call, which represents what they would remember of call quality.
- (ii) Network or operations management systems require an estimate of call quality as it is directly affected by network introduced impairments. It can be disadvantageous to include some subjective effects as they can mask the impact of raw network performance.
- (iii) SLA monitoring systems need to estimate call quality at intermediate points in a Voice over IP stream and may require both end user oriented and network oriented metrics.

Based on discussions with service providers the following metrics appear to fit the requirements:

#### ***User R Factor and MOS***

The Annex E algorithm incorporates the effects of packet loss distribution, jitter, delay, CODEC type and recency. This does provide a reasonable estimate of call quality in the form of an R Factor, as it would be reported by an end user. It is proposed that this metric be termed the "*User R Factor*" to denote this. As many people are more familiar with MOS scores than R Factors it is also useful to convert the User R Factor into a MOS score, using the equation given in G.107 and ETR250.

#### ***Network R Factor***

The effects of recency, and to some extent delay, are out of the control of the network operator and can mask the effects of raw packet loss. Network managers would like to be able to see the effects of packet loss and jitter, which they can potentially affect through router configuration. The R factor calculated using Annex E before the effects of recency and delay are incorporated provides this information - it is proposed that this be called the "*Network R Factor*".

It is proposed that section E.8 of TS 101 329-5 be modified as follows:

### **E.8 Calculation of Call Quality Metrics**

The  $I_e$  values calculated above can be used as inputs to the E Model. It is recommended that two separate metrics are calculated:-

***User R Factor*** – calculated using  $I_e$ (end of call) and incorporating delay effects

This factor provides an estimate of the call quality that would be reported by a subscriber at the end of a call and would typically be used for Billing or SLA verification applications. It is also appropriate to convert this factor into an estimated MOS score, using the equation given in G.107 or ETR250.

***Network R Factor*** – calculated using  $I_e$ (av)

This factor provides an estimate of the call quality as impacted only by network impairments such as packet loss and jitter and would typically be used for network management applications. It is not recommended to convert this factor into an estimated MOS score as it does not include the effects of delay or recency which can significantly impact subjective quality.

An R Factor calculated by the E Model using only average packet loss, i.e. not applying the model described in this Annex, would primarily be used for transmission planning purposes as the distribution of packet loss would not be known in advance.