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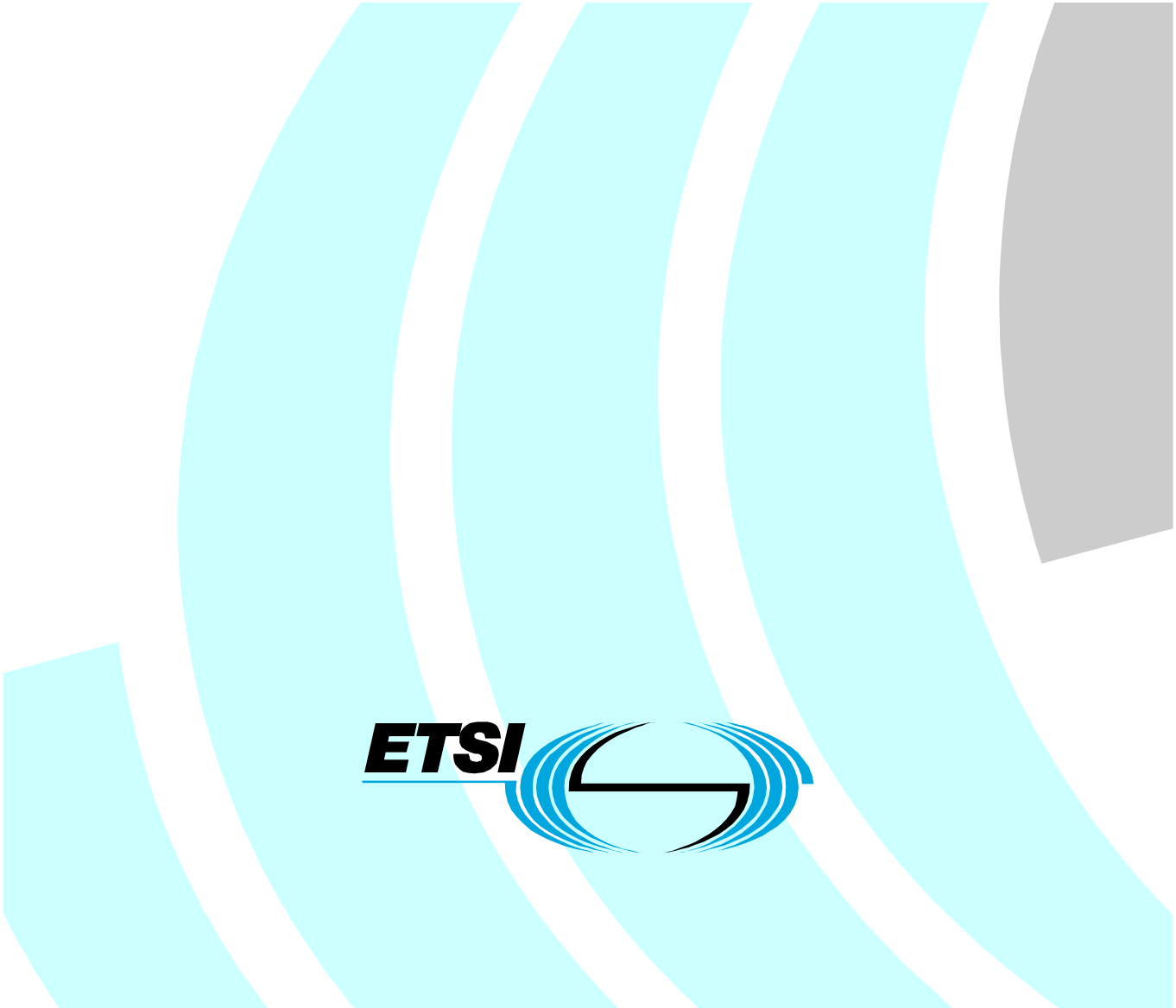
Technical Report

Speech Processing, Transmission and Quality Aspects (STQ); Application and enhancements of the E-Model (ETR 250); Overview of available documentation and ongoing work



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Contents

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Foreword

This Technical Report (TR) has been produced by ETSI Technical Committee Speech Processing, Transmission and Quality Aspects (STQ).

Introduction

In 1996 ETSI launched with an unprecedented effort a unique transmission rating model which famously combined all available knowledge in the field of transmission planning.

Under the leadership of Nils Olof Johannesson (1923-1997) a very concise document was elaborated by a special ETSI rapporteur's group and it was finally published as ETR 250 [1].

In the time thereafter this transmission rating model was promoted to many other organization in the field, including the ITU-T, where Recommendation G.175 [12] recommended for the first time in 1997, the use of the method described in ETR 250 [1] for the transmission planning of Corporate and Enterprise Networks.

After reconciliation and positive feedback from users of ITU-T Recommendation G.175 [12], the ITU-T, Study Group 12, in 1998 approved ITU-T Recommendation G.107 [3] giving the principles and the algorithm of the now so-called "E-Model" as the ITU-T's unique and global transmission planning tool.

Consequently, revisions and new recommendations with respect to the E-Model were launched and are currently being maintained.

In appreciation of the tremendous success of the "E-Model" in the ITU-T and based thereon in many other organizations, ETSI has never attempted, to parallelize this development by means of re-publishing documents from these organizations.

The present document is intended to keep track of the development of the "E-Model" outside ETSI.

1 Scope

The present document provides an overview of the application and enhancements of the E-Model taking place in various standardization bodies; in addition it points to the most recent version of the E-Model.

2 References

For the purposes of this Technical Report (TR), the following references apply:

[1] ETSI ETR 250: "Transmission and Multiplexing (TM); Speech communication quality from mouth to ear for 3,1 kHz handset telephony across networks".

[2] ETSI EG 201 050 (V1.2.2): "Speech Processing, Transmission and Quality Aspects (STQ); Overall Transmission Plan Aspects for Telephony in a Private Network".

[3] ITU-T Recommendation G.107 (2003): "The E-Model, a computational model for use in transmission planning".

[4] ITU-T Recommendation G.108 (1999): "Application of the E-model: A planning guide".

[5] ITU-T Recommendation G.108.1 (2000): "Guidance for assessing conversational speech transmission quality effects not covered by the E-model".

[6] ITU-T Recommendation G.109 (1999): "Definition of categories of speech transmission quality".

[7] ITU-T Recommendation G.113 (2001): "Transmission impairments due to speech processing".

[8] ITU-T Recommendation G.114 (2003): "One-way transmission time".

[9] ITU-T Recommendation G.131 (2003): "Talker echo and its control".

[10] TIA/EIA/TSB32-A (1998, reaffirmed 2002): "Overall Transmission Plan Aspects for Telephony in a Private Network".

[11] TIA/EIA/TSB116 (2001): "Telecommunications - IP Telephony - Voice Quality Recommendations for IP Telephony".

[12] ITU-T Recommendation G.175: "Transmission planning for private/public network interconnection of voice traffic".

[13] ITU-T Recommendation P.562: "Analysis and interpretation of INMD voice-services measurements".

[14] ITU-T Recommendation P.833: "Methodology for derivation of equipment impairment factors from subjective listening-only tests".

3 Definition

For the purposes of the present document, the following definition applies:

E-Model: model based on the equipment impairment factor method, following previous transmission rating models

NOTE 1: It was developed by an ETSI ad hoc group called "Voice Transmission Quality from Mouth to Ear".

NOTE 2: The reference connection is split into a send side and in a receive side. The model estimates the conversational quality from mouth to ear as perceived by the user at the receive side, both as listener and talker.

According to the equipment impairment factor method, the fundamental principle of the E-model is based on a concept given in the description of the OPINE model (see Supplement 3 to ITU-T Series P Recommendations (1988), Models for predicting transmission quality from objective measurements.):

- ***Psychological Factors on the psychological scale are additive.***

The result of any calculation with the E-model in a first step is a transmission rating factor R, which combines all transmission parameters relevant for the considered connection. This rating factor R is composed of:

$$R = Ro - Is - Id - Ie,eff + A \quad (1)$$

Ro represents in principle the basic signal-to-noise ratio, including noise sources such as circuit noise and room noise. The factor *Is* is a combination of all impairments which occur more or less simultaneously with the voice signal. Factor *Id* represents the impairments caused by delay and the effective equipment impairment factor *Ie,eff* represents impairments caused by low bit rate codecs. It also includes impairment due to packet-losses of random distribution. The advantage factor *A* allows for compensation of impairment factors when there are other advantages of access to the user.

4 Related documents

ETR 250: "Transmission and Multiplexing (TM); Speech communication quality from mouth to ear for 3,1 kHz handset telephony across networks" [1].

ETR 250 [1] applies to mouth to ear narrowband (nominally 3,1 kHz) telephony connections via handsets across one or more telecommunications networks of any type, public or private, fixed or mobile. ETR 250 [1] provides:

a) a compendium of established and well recognized transmission planning information on individual parameters, including limits for satisfactory performance and information on equipment with special transmission characteristics;

b) a new simple computational planning method to consider the combination effect of transmission parameters when evaluating transmission impairments for telephony services, including estimation of users' opinion using the scales percentage poor or worse, percentage good or better, percentage terminating early and mean opinion score.

The computational planning method is an adaptation of computation models published in ITU-T (former CCITT) documentation, complemented with the result from some recently presented results of subjective tests. The objective is to assist network operators in ensuring that users will be satisfied with the transmission performance whilst avoiding over-engineering of the networks. The aim is to give realistic, practical guidance rather than a scientifically exact treatment of quality factors.

It should be noted that the computational planning method, which is described in clause 9, has not yet been fully verified and may give incorrect results. Until this verification has been completed, predictions resulting from the use of this planning method should therefore be treated with caution.

EG 201 050 (V1.2.2): "Speech Processing, Transmission and Quality Aspects (STQ); Overall Transmission Plan Aspects for Telephony in a Private Network" [2].

EG 201 050 [2] represents a joint effort between TIA Subcommittee TR-41.1 (Multi-line Telecommunications Systems) and ETSI Technical Committee CN7 (Corporate Telecommunication Networks) (see note) to develop a global private network transmission planning guide. The intent of the document is to provide the planner of multi-national and transcontinental private networks with the transmission planning principles and planning examples which will enable the planner to design a loss and level plan for the network, optimized to the desired level of voice quality between any two users of the network as well as between the network and other - e.g. public networks.

NOTE: The main effort on the part of ETSI in the development of EG 201 050 [2] occurred during the existence of CN7. That working group was a successor to the transmission working group in ETSI working group BTC2. Working group CN7 has since closed down and the work now continues as part of the program of ETSI Technical Committee STQ (Speech Processing, Transmission and Quality Aspects).

The text of EG 201 050 [2] was largely generated by ETSI as an update to an earlier ETSI Technical Report, ETR 004. Concurrent with that effort, TIA TR-41.1 started a project to update the document "DPBX Loss Plan Application Guide" (TSB-32-1989). TIA TR-41.1 and ETSI CN7 agreed in 1997 that there was a need for a common global transmission planning guide for private networks; therefore TIA TR-41.1 elected to work with the ETR 004 update project to generate the present document by adding text (network characteristics, pertinent regional parameters, planning examples, and references) which would be of value to planners in both the North American and European regions. For North American users, the document is designated as the update of TSB-32-1989. Meanwhile, ETSI published the document as ETSI Guide, EG 201 050 [2].

The underlying foundation for the planning guidelines in EG 201 050 [2] is the ETSI E-Model for calculating and evaluating end-to-end connection voice quality. The E-Model is gaining universal acceptance as the tool for transmission planning. It is expected that the E-Model will undergo refinements over time; the present document may require updating accordingly.

As a component of the TSB-32-1989 update project, TIA TR-41.1 also undertook to examine a number of connection scenarios specific to North America for the illustration of loss design based on the E-Model. The rationale for this work was to provide continuity from the presentation in the initial publication of the TSB. Rather than including these scenarios in the document, it is intended to separately publish these scenarios as a companion document for demonstrating practical applications of loss design for a variety of connection scenarios.

ITU-T Recommendation G.107 (2003): "The E-Model, a computational model for use in transmission planning" [3].

ITU-T Recommendation G.107 [3] gives the algorithm for the so-called E-model as the common ITU-T Transmission Rating Model. This computational model can be useful to transmission planners, to help ensure that users will be satisfied with end-to-end transmission performance. The primary output of the model is a scalar rating of transmission quality. A major feature of this model is the use of transmission impairment factors that reflect the effects of modern signal processing devices.

In the year 2000 revision, an enhanced version of the E-model was provided, in order to better take into account the effects of room noise at the send side, and quantizing distortion. With the year 2002 revision, the impairment due to random packet-loss has been included in a parametric way for different codecs. The current version (2003) provides an enhanced modelling of the quality in case of low talker sidetone levels.

ITU-T Recommendation G.108 (1999): "Application of the E-model: A planning guide" [4].

The intent of ITU-T Recommendation G.108 [4] is to demonstrate how the E-Model (described in ITU-T Recommendation G.107 [3]) can be used in end-to-end transmission planning for a wide range of networks - local, national, multinational and transcontinental.

ITU-T Recommendation G.108.1 (2000): "Guidance for assessing conversational speech transmission quality effects not covered by the E-model" [5].

ITU-T Recommendation G.108.1 [5] provides guidance for transmission planners on how to evaluate those effects impacting end-to-end speech transmission performance which are not covered by the E-model (ITU-T Recommendation G.107 [3]) and its associated Planning Guide (ITU-T Recommendation G.108 [4]). Procedures for informal subjective and objective evaluations that can be used to complement the E-model are provided here.

ITU-T Recommendation G.109 (1999): "Definition of categories of speech transmission quality" [6].

ITU-T Recommendation G.109 [6] defines five categories of end-to-end speech transmission quality for 3,1 kHz handset telephony. These categories are defined as ranges of speech transmission quality in terms of "user satisfaction", tied to the ratings given by the transmission planning tool of ITU-T Recommendation G.107 [3] which takes into account the combined effects of various transmission impairments. The definitions provided here are independent of any specific technology that may be used in different types of network scenarios under consideration.

ITU-T Recommendation G.113 (2001): "Transmission impairments due to speech processing" [7].

ITU-T Recommendation G.113 [7] provides guidance regarding transmission impairments introduced by digital speech processing systems. The information provided is for use in conjunction with the transmission planning approach described in ITU-T Recommendations G.107 [3], G.108 [4] and G.109 [6]. The Impairment Factor method, used by the E-model of ITU-T Recommendation G.107 [3], is now recommended. The earlier method that used Quantization Distortion Units is no longer recommended. Updated Impairment Factor values for various digital processing systems are provided in Appendix I (05/2002). Appendix II contains guidance on how an Advantage Factor can be used to reflect the variation in user expectation of quality for different communications systems (e.g. mobile).

ITU-T Recommendation G.114 (2003): "One-way transmission time" [8].

ITU-T Recommendation G.114 [8] provides guidance on the effect of end-to-end one-way delay (sometimes termed latency), and an upper bound one-way network delay.

While it is recommended that a one-way delay of 400 ms should not be exceeded for general network planning, it is important to appreciate that highly interactive tasks (e.g., many voice calls, interactive data applications, video conferencing) can be affected by much lower delays.

The effects of delays below 500 ms on conversational speech are estimated using a curve derived from the E-Model (ITU-T Recommendation G.107 [3]).

This version constitutes a major revision of this Recommendation in order to align with other Recommendations of the G.100 series.

ITU-T Recommendation G.131 (2003): "Talker echo and its control" [9].

ITU-T Recommendation G.131 [9] provides guidance on the effect of talker echo, and its control. Talker echo is considered independently of all other impairments. Furthermore the conjunction of talker echo and the E-Model of ITU-T Recommendation G.107 [3] is explained as well as the reference to ITU-T Recommendation G.108.2 on transmission planning aspects of echo cancellers is provided.

Previous versions of ITU-T Recommendation G.131 [9] included a clause on stability that has been deleted because modern networks are largely all four-wire.

Earlier versions of ITU-T Recommendation G.131 [9] contained several planning rules for connections with echo control devices. As many of those rules are now obsolete, they are not reproduced here.

A new Appendix III on the combined effects of talker echo in the presence of absolute delay has been added.

TIA/EIA/TSB32-A (1998, reaffirmed 2002): "Overall Transmission Plan Aspects for Telephony in a Private Network" [10].

TIA/EIA/TSB32-A [10] represents a joint effort between TIA Subcommittee TR-41.1 (Multi-line Telecommunications Systems) and ETSI Technical Committee CN7 (Corporate Telecommunication Networks) to develop a global private network transmission planning guide. The intent of the document is to provide the planner of multi-national and transcontinental private networks with the transmission planning principles and planning examples which will enable the planner to design a loss and level plan for the network, optimized to the desired level of voice quality between any two users of the network as well as between the network and other - e.g. public networks.

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TIA/EIA/TSB116 (2001): "Telecommunications - IP Telephony - Voice Quality Recommendations for IP Telephony" [11].

The objectives of TIA/EIA/TSB116 [11] are to provide end-to-end voice quality guidelines for North American IP Telephony and to an E-Model tutorial for IP scenarios. IP Telephony introduces several impairments, some of which are familiar and some new. The E-Model (ITU-T Recommendation G.107 [3]) is a tool that can estimate the end-to-end voice quality, taking the IP Telephony parameters and impairments into account. This TSB first describes how the E-Model handles IP Telephony impairments and then it provides general design recommendations for the best possible voice quality performance irrespective of cost, available technology or customer requirements. These recommendations are illustrated with specific IP scenarios to provide an E-Model tutorial for analysing real networks.

Since initially, IP telephony is a replacement technology for the existing wireline PSTN, the focus of this document is on wireline scenarios. The impairments introduced by wireline IP packet technology can be significant. The reader should be aware that wireless and satellite technologies also introduce significant impairments and that only a few of the combined effects are illustrated here. This TSB builds on similar work done for North American PBX private networks that was published in TIA/EIA/TSB32-A [10] and the focus of this document remains on providing guidelines for engineered private networks as opposed to the Internet.

The E-Model scenarios detailed in TIA/EIA/TSB116 [11] are included as two Microsoft Excel workbooks:

- TSB116NETEM1.xls; and
- TSB116NETEM2.xls.

Each workbook includes Version 19 of the E-Model application, which the reader can use to model other scenarios.

5 Further improvement of the E-Model

Such work is currently underway in ITU-T Study Group 12:

Question 8/12 - Extension of the E-Model

- (Continuation of Question 20/12 studied in 1997-2000)

Type of question

Task-oriented leading to a revision of ITU-T Recommendation G.107 [3] and to new ITU-T Recommendations.

Background

In the last years, Study Group 12 has established a new concept of impairment factors which aim at predicting the perceptive effects of different type of degradations on overall speech communication quality, for network planning purposes. The core algorithm related to this concept is the so-called E-model, a computational model for use in transmission planning. This model has been analysed in detail, and standardized as ITU-T Recommendation G.107 [3]. It can be applied in network planning of traditional, narrow-band and handset terminated networks. Additionally, a methodology has been set up to derive impairment factors for codecs from subjective listening-only tests.

With respect to the application of the E-model to the planning of modern networks, however, there are still several open questions, which limit the model's usability. They result from the terminal and transmission characteristics of modern networks, which could not have been taken into account at the time the model was established. In order to guarantee that the achieved concept keeps track with the technological progress in both the transmission system and the terminal equipment areas, it is highly desirable to maintain and update the model. The validity range of the E-model should be extended so that it cannot only be applied to traditional networks, but e.g. to packet-based transmission, wide-band systems, or non-handset terminal equipment, too.

The extension of the E-model is also a prerequisite in order to keep other Recommendations up to date, which deal with the impairment factor concept and with categories of speech transmission quality. These Recommendations include but are not limited to:

- ITU-T Recommendations G.108 [4] and G.108.1 [5] on the application of the E-model, and on conversational aspects not covered by the current version;
- ITU-T Recommendation G.109 [6] on classes of speech transmission quality;
- ITU-T Recommendation G.113 [7] on transmission impairments;
- ITU-T Recommendation G.175 [12] on the interconnection of the PSTN with private and/or IP based networks;
- ITU-T Recommendation P.562 [13] on the analysis and interpretation of INMD voice-service measurements.

Text of the question

The main question is how the E-model can be extended in order to cover the effects of an impaired digital transmission and of terminal equipment other than handsets.

- Does practical experience with the new methodology to derive equipment impairment factors from subjective tests lead to a consistent framework of Ie values? What scaling method should be applied for this purpose?
- How can instrumental models currently developed and standardized by Q.J/12 be fruitfully used to derive equipment impairment factors for:
  - new codecs in single operation;
  - codecs in tandem operation; and
  - codecs under conditions of random bit errors, frame erasures or packet loss?
- Is it useful to derive a table of total Ie values for codec tandems, which do not satisfy the additivity property, assumed by the impairment factor principle? Can new instrumental measures be used to establish such a table?
- Can new formulae be derived for codecs under transmission error conditions, in order to keep the model simple and manageable?
- Which quality issues have to be taken into account when extending the E-model to terminal equipment other than handset telephones (e.g. HFTs, headsets)? Problems to be resolved may include:
  - the determination of new input parameters to the model, e.g. for SLR and OLR, STMR, as well as room noise (send and receive side);
  - the coverage of acoustic echo and sidetone;

  - the pickup of ambient room noise by this terminal equipment; and
  - the perceptual effects of signal-processing equipment which forms part of such terminals, as well as their coverage in instrumentally measurable input parameters to the model (clipping, sound degradation, impact on the conversation, etc.).

- How can or should quality dimensions other than 'impairment' be covered, e.g.:
  - "speech sound quality", e.g. due to terminal equipment other than handsets, due to transmission bandwidth other than the normal 3,1 kHz band, e.g. wide-band transmission, or due to frequency distortion or non-linear codecs;
  - conversational quality features (cf. draft ITU-T Recommendation G.108.01).
- What is the influence of user expectation on the overall quality, e.g. for terminal equipment other than handsets, or for computer operated VoIP services?
  - What are the psychological dimensions commonly handled by the term "expectation"?
  - How will user expectation develop with time (cf. COM 12-98, COM 12-111, study period 1997-2000)?
- Study items
  - Analysis and verification of Ie values obtained with the methodology described in draft new ITU-T Recommendation P.833 [14].
  - Definition of a new method for deriving equipment impairment factors from instrumental prediction models (e.g. P.86x).
  - Analysis of the additivity property of Ie values for codecs in tandem or codecs under transmission error conditions.
  - Establishment of new formulae describing impairments for codecs under transmission error conditions.
  - Definition of instrumentally measurable input parameter to the E-model describing non-handset terminal equipment.
  - Investigation of perceptual effects related to terminal equipment other than handsets, e.g. due to signal processing in such terminals.
  - Investigation of the so-called "speech sound quality" and its relation to impairment as predicted by the E-model.
  - Investigation of the effects of wide-band transmission on "speech sound quality" and speech communication quality.
  - Investigation of conversational quality features currently not covered by the model.
  - Analysis of dimensions related to user expectation.
- Objectives and schedule:
  - Revision of ITU-T Recommendation G.107 [3].
  - New Recommendation on the derivation of equipment impairment factors from instrumental measures.
  - Potential revision of other related ITU-T Recommendations as indicated above.
  - Potentially further new Recommendations as outcome from the study items.

History

Document history		
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