International Telecommunication Union

QoS activities in ITU-T Recent Progress

Jean-Yves Monfort

ITU-T SG 12 Chairman

France Telecom R&D

Workshop on Satellites in IP and Multimedia
Geneva, 9-11 December 2002

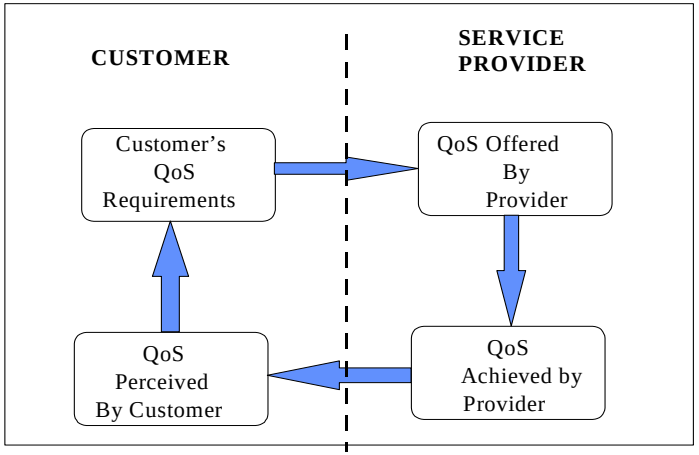
QoS and NP activities at ITU-T

- SG 12 : End to end quality, as perceived by the users. It is fully addressed to Quality, and WP3/12 is dedicated to QoS for IP.
- SG 2 : Mainly on operational aspects of QoS and SLA. New QoS handbook and activities on the impacts of routing on QoS.
  - QSDG : 1 forum meeting each year and QSDG Magazine.(Chaired by L. Cardoso)
- SG 13 : WP4/13 is dedicated to Network Performance
- SG 4 : Management of QoS and SLA.
- SG 9 : QoS for cable networks and video assessment.
- SG 11 : QoS signalling.
- SG 15 : System-specific requirements for network and transport equipment.
- SG 16 : QoS Mecanisms for H.323-based multimedia systems. Quality of speech and video coders.
- SG 17 : Frame Relay QoS.

QoS and NP. Definitions

- E.800 definitions:
  - QoS : "The collective effect of service performance which determines the degree of satisfaction of a user of a service"
  - NP :"The ability of a network or network portion to provide the functions related to *communications* between *users*."

4 viewpoints on QoS Rec. G.1000 (Communications Quality of Service : a framework and definitions)



A selection of recently approved Recommendations

- G.1000 «Communications Quality of Service : A framework and definitions »
- G.1010 «End-User multimedia QoS categories »
- G.107 « Revised « The E-Model, a computationnal model for use in transmission planning »
- P.561 Revised « In-service, Non-intrusive Measurement device – Voice services measurements »
- E.860 “Framework for service level agreement”
- Y. 1541 “IP Performance objective and allocations”

Recommendations under AAP

- Revised Y.1540 « IP Packet transfer and availability performance parameters”

A selection of works in progress in the ITU-T SGs

- On terminal equipments : P.VOIP (IP Terminals), P.GTW(IP Gateways), P. 380 (Headsets),...
- On transmission Planning and transmission parameters : G.VoIP-Islands, Revised G.114 ...
- On Quality parameters, modelling and classes : G.IPP (Transmission performance pameters for IP ...), P.562 revised, G.MMPerf (Multimedia performance requirements), H.mmclass, several Rec. for perceptual Video Quality.
- Handbooks : QoS, Subjective Testing Procedure, Telephonometry
- On Performance: Future revision -already began- of Y.1541 (IP Performance objectives and allocations), I.350 (General aspects of QoS and NP), Draft Rec. TCP Performance on IP-Based networks, Y.1530, ...
- On Management : M. QoS, H.QoS.m
- On QoS Signalling Draft TRQ "Signalling Requirements for IP-QoS", E-QSC (Signalling of proposed QoS services classes...)
- On QoS Architecture : H.QoSarch, Y.qosar

E.860 "Framework for Service Level Agreement"

- In this Recommendation, **SLA** is defined as a **tool to formalise the inter-relationships between entities** (an entity being a generic unit involved in using/delivering a service)

- E.860
  - recalls QoS terms and definitions
  - define the **one stop responsibility concept**
  - describes a generic structure of SLA
  - illustrates **how to apply in a multi-provider environment**

QoS Classes and Categories

- SG12 considers that the concept of QoS classes should be preferred to individual parameters, in general and for signalling end-to-end QoS.
- It is important that the parameters that make up a given QoS class are independent of one another.
- Progress in QoS classes signalling is in progress in the relevant SG (mainly SG 11)
- In the following slides, a special focus will be done on Rec. G.1010 and Y.1541.

Recommendation G.1010

End-User **Multimedia QoS Requirements**

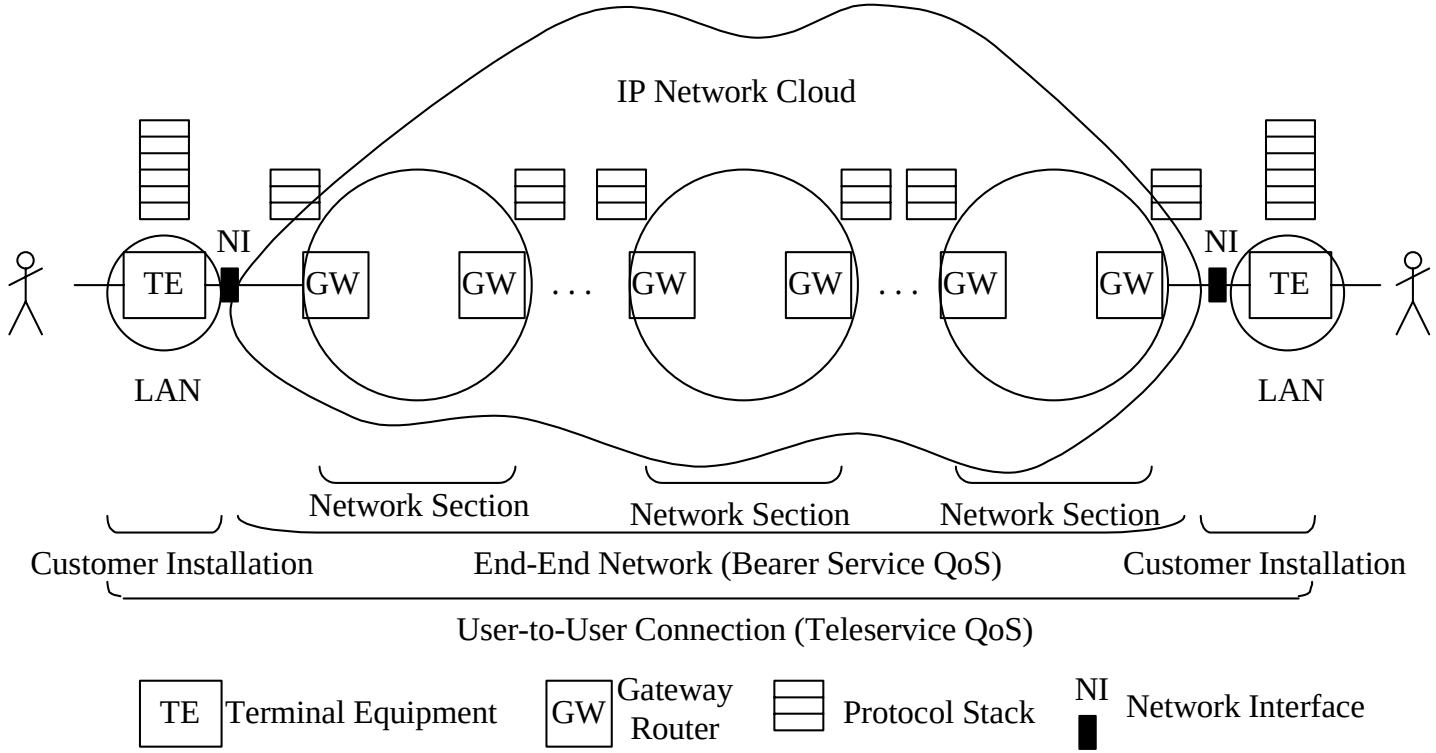
- Performance expressed by parameters
  - Focused on **user perceivable effects**
  - **Independent of the networks** internal design
- Parameters
  - Delay; Delay variation; Packet loss Ratio
- **Model for user-centric performance requirements** Mapping can be formalised into model for QoS categories

	Interactive (delay <<1 sec)	**Responsive** (delay ~2 sec)	**Timely** (delay ~10 sec)	**Non-critical** (delay >>10 sec)
Error tolerant	Conversational voice and video	Voice/video messaging	Streaming audio and video	Fax
Error intolerant	Command/control (eg Telnet, interactive games)	Transactions (eg E-commerce, WWW browsing, Email access)	Messaging, Downloads (eg FTP, still image)	Background (eg Usenet)

Benefit of end-user QoS category model

- Model is based on end to end user perception of impairments, therefore not dependent on any specific technology for its validity
- Provides basis for realistic network QoS classes (eg ITU-T Rec. Y.1541)
- Rec. G.1010 provides an indication of the upper and lower boundaries for applications to be perceived as essentially acceptable to the user (A new Recommendation G.Mmperf is under developpment)
- Shows how the underlying impairments of information loss and delay can be grouped appropriately, without implying that one class is “better” than another

Recommendation Y.1541 QoS Classes: A Basis for IP Network QoS Control



NI-to-NI Reference Path for network QoS Objectives

NOTE : Customer installation equipment is shown for illustrative purposes, only

Recommendation Y.1541 QoS Classes: A Basis for IP Network QoS Control

Network Performance Parameter	Nature of Network Performance Objective	Class 0	Class 1	Class 2	Class 3	Class 4	Class 5 (Un-specified)
IPTD	Upper bound on the mean IPTD	100 ms	400 ms	100 ms	400 ms	1s	U
IPDV	Upper bound on the 1-10^{-3} quantile of IPTD minus the minimum IPTD	50 ms	50 ms	U	U	U	U
IPLR	Upper bound on the packet loss probability	$1*10^{-3}$	$1*10^{-3}$	$1*10^{-3}$	$1*10^{-3}$	$1*10^{-3}$	U
IPER	Upper bound	$1*10^{-4}$					U

U means « unspecified » or « unbounded »

Note for IPVD : The value of 50ms is dependent on the capacity of inter-networks links. Smaller variations are possible where all the capacities are higher than primary rate, or when competing packet information fields are smaller than 1500 bytes

A new parameter as been recently included in Y.1540 (the limits are still under study:

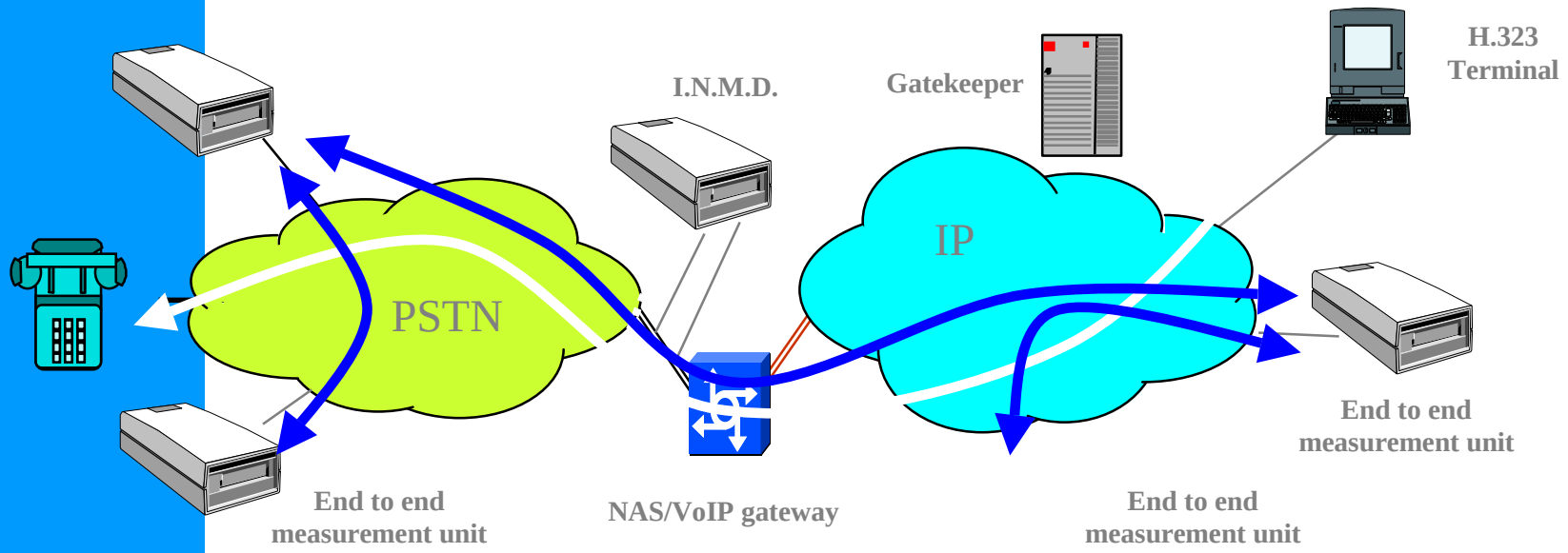
IPSLBR : IP Paket Severely Loss Block Ratio

Two types of tools to measure voice quality

:

Non-intrusive (I.N.M.D.) *: on real communication without reference signal, better for the supervision of network quality of service,*

End-to-end intrusive *: on test calls with reference signal, better fitted for the measurement of quality as perceived by end users.*



Non-intrusive techniques

- The non-intrusive measurement techniques
  - can be implemented as part of network equipments (switches) or in stand-alone devices
  - do not require tests signal since they are performed on real communications.
- The tools used to perform those measurement are generally called I.NM.D.s (for In service Non-intrusive Measurement Devices).
- Four classes depending on the type of networks and Round trip delay (Classe D is for Packet-switched networks and a maximum value of 1000 ms for Round Trip Delay)
- The parameters that one can measure with non-intrusive systems are of two kinds :
  - related to protocol or signalling
    - calling and called number, call duration, packet loss, one point and two point jitter, RTP delay, etc.
  - derived from the analysis of the voice signal
    - echo, noise and speech levels, etc…
    - IP packet loss ratio, etc…for Class D
    - clipping, voice quality, etc.

Links with perceived quality (P.562) and workplan

- The analysis of measurement results can be done for individual parameters or by combining several parameters : use of a customer opinion models :
  - E model (G.107)
  - CCI (Call Carity Index)
- A competition has started:
  - Call for model submission for the assessment of voice transmission quality from protocol analysis information in IP networks- In progress

E-Model (G.107)

- applicable to network planning of traditional, narrow-band and handset terminated networks
- estimates voice transmission quality mouth-to-ear as perceived at receive side
- renders a transmission factor *R*

The Rating factor ***R*** *is composed of*

$$R = R_0 - Is - Id - Ie,eff + A$$

- R_0 represents in principle the basic signal-to-noise ratio, including noise sources such as circuit noise and room noise.
- *Is* is a combination of all impairments which occur more or less simultaneously with the voice signal.
- *Id* represents the impairments caused by delay and the equipment impairment factor
- *Ie,eff* represents impairments caused by low bit rate codecs; it also include impairments due to packet loss of random distribution;
- *A*, the advantage factor, allows for compensation of impairment factors when there are other advantages of access to the user.

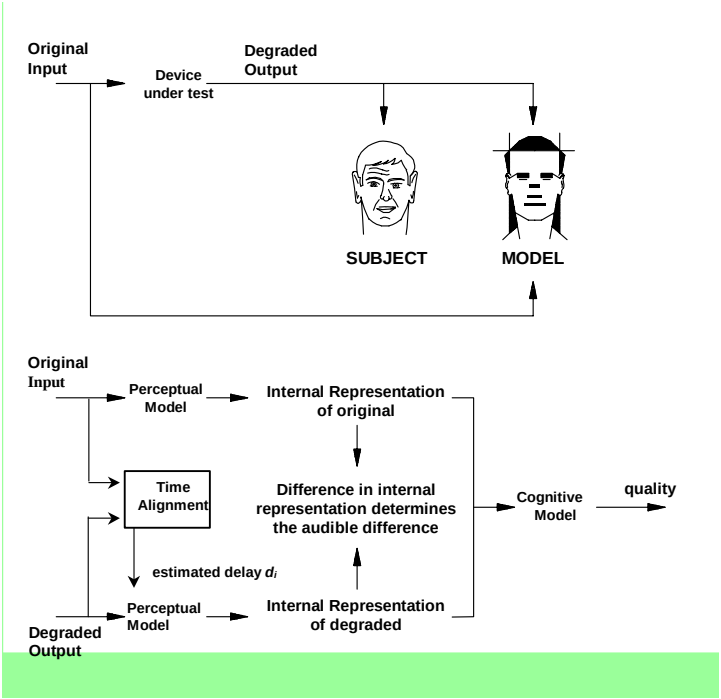
Provisional guide for the relation between R-value and user satisfaction (Annex B of G.107)

R-value (lower limit)	MOS (lower limit)	GoB (%) (lower limit)	PoW (%) (upper limit)	User satisfaction
90	4.34	97	~0	Very satisfied
80	4.03	89	~0	Satisfied
70	3.60	73	6	Some users dissatisfied
60	3.10	50	17	Many users dissatisfied
50	2.58	27	38	Nearly all users dissatisfied

A way to mesure End-to-endspeech quality

P.862 : Perceptual evaluation of speech quality (PESQ), an objective method for end-to-end speech quality assessment of narrow-band telephone networks and speech codecs

- **Overview of the basic philosophy used in PESQ : A computer model of the subject, consisting of a perceptual and a cognitive model, is used to compare the output of the device under test with the input, using alignment information as derived from the time signals in the time alignment module**



Market needs, limitations and Progress

Market needs

- Instrumental methods that cover end-to-end speech quality including the terminal
- Single-ended speech quality monitoring tools for packet-oriented networks
- Robust speech quality measures for speech quality enhancement effects in the network
- Instrumental models that consider conversational speech quality aspects
- Scalable measurement solutions (for e.g. different classes / profiles of application, price, accuracy, …)
- Speech quality assessment using knowledge of network parameters and of the psycho-acoustics
- Speech quality analysis systems

Coverage of models

… (cf. ITU-T Rec. P.862, Table 2 and 3)

PESQ has demonstarted acceptable accuracy for packet loss and packet loss concealment with CELP Codecs, but nor validated for PCM type codecs

PESQ has demonstarted acceptable accuracy for waveform codecs (G.711, G.726,…), CELP and hybrid codecs $\geq$ 4 kbit/s (G.729, G.723.1,..GSM codecs, TETRA), but has not been currently validated for CELP and hybrid Codecs < 4kbit/ or MPEG 4 HVXC..

Work Plan to expand the coverage of the model :

- New draft Recommendation P.SEAM (single ended assessment models)
- New draft Recommendation P.AAM (acoustic assessment models)

Future of QoS/NP in ITU-T

- *We have the basis for QoS Classes. We have to complete and to converge on complete figures (for the different services and layers)*
- *What have to define a set of network protocols capable of establishing IP network QoS classes. In the future, other mechanisms and protocols may enable dynamic QoS over multiple networks.*
- *We have to continue the cooperation with other standardization bodies to communicate our progress and to improve the compatibilities of the standards*
- *We have to consolidate our level of quality and of production of new recommendations for the benefit of the Industry, the Telco and Service providers and the regulatory bodies*

Thanks

- *To the organizators, speakers and participants*
- *To the Specialists whose works are summarized in this Presentation, and especially*

  M. Yam, M. Seitz, M. Kim, M. Dvorak, M. Coverdale, M. Barriac, M. Klaus, M. Möller and Mrs Jekosch
- *Thanks to all the contributors*
- *For complementary informations, don't hesitate to contact me :*
  - *Jeanyves.monfort@francetelecom.com*