

<Ufa cb]nUWYUHY_Y_ca i b]_UWY^]b`]bhYfbYbY[Udfcfc_c`UdfY_`ca fYj]`fH-D<CBtZ' "
]nXUWU!`?U_cj cghghcf]h YcX`_cbWUXc`_cbWUj `g]ghYa] `H-D<CB`!`+"XY. `JcX]`c`nU
bU flcj UbY`Y`Ya Ybhcj `dcj YnUj `H-D<CB`n`j]X]_U`na c[`]j cgh]`dfYbcgU[cj cfUcX
_cbWUXc`_cbWU

Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON)
Release 3; End-to-end Quality of Service in TIPHON systems; Part 7: Design guide for
elements of a TIPHON connection from an end-to-end speech transmission performance
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Technical Report

**Telecommunications and Internet Protocol
Harmonization Over Networks (TIPHON) Release 3;
End-to-end Quality of Service in TIPHON systems;
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ETSI

650 Route des Lucioles
F-06921 Sophia Antipolis Cedex - FRANCE

Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

Siret N° 348 623 562 00017 - NAF 742 C
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Foreword

This Technical Report (TR) has been produced by ETSI Project Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON).

The present document is part 7 of a multi-part deliverable covering End-to-end Quality of Service in TIPHON systems, as identified below:

- TR 101 329-1: "General aspects of Quality of Service (QoS)";
- TS 101 329-2: "Definition of speech Quality of Service (QoS) classes";
- TS 101 329-3: "Signalling and control of end-to-end Quality of Service (QoS)";
- TS 101 329-5: "Quality of Service (QoS) measurement methodologies";
- TR 101 329-6: "Actual measurements of network and terminal characteristics and performance parameters in TIPHON networks and their influence on voice quality";
- TR 101 329-7: "Design guide for elements of a TIPHON connection from an end-to-end speech transmission performance point of view".**

Quality of Service aspects of TIPHON Release 4 and 5 Systems will be covered in TS 102 024 and TS 102 025 respectively, and more comprehensive versions of the Release 3 documents listed above will be published as part of Releases 4 and 5 as work progresses.

Introduction

The present document forms one of a series of technical specifications and technical reports produced by TIPHON Working Group 5 addressing Quality of Service (QoS) in TIPHON Systems. The structure of this work is illustrated in figure 1.

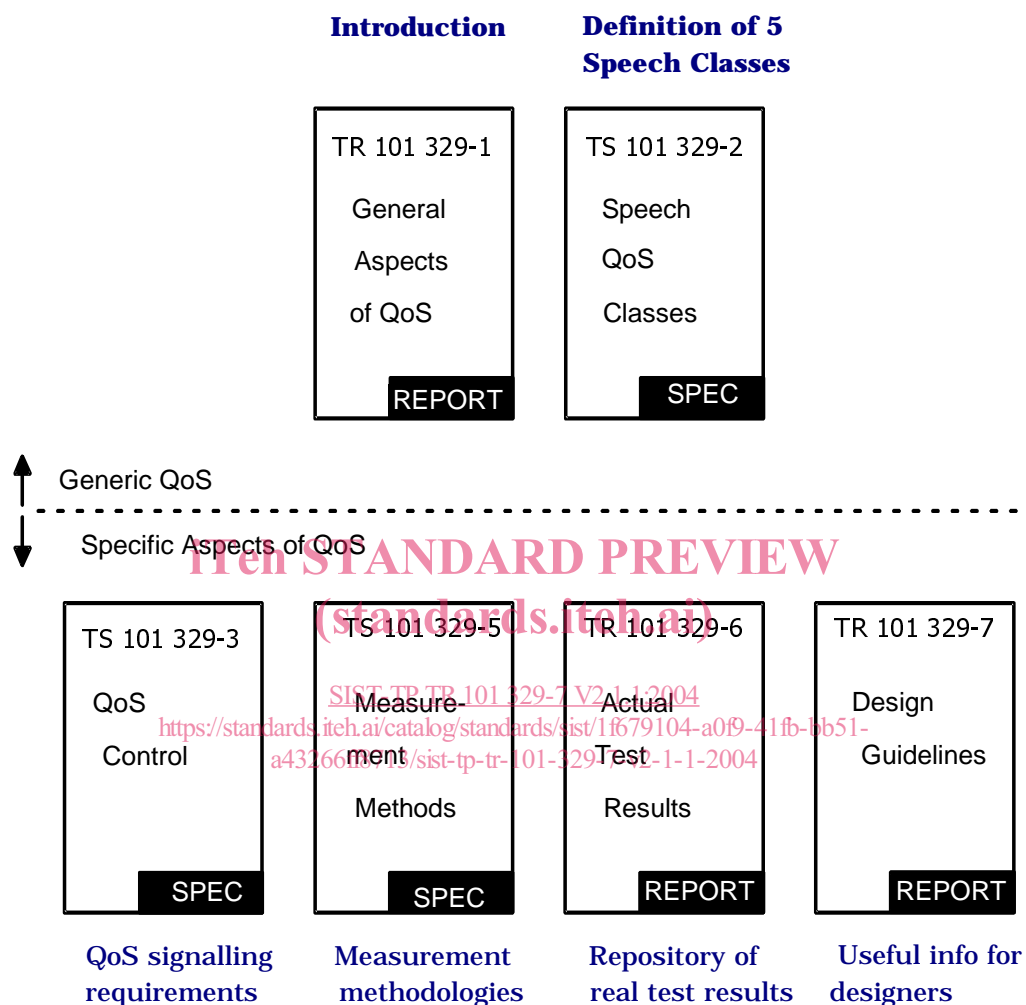


Figure 1: Structure of TIPHON QoS Documentation for Release 3

For a concise understanding of the guidance provided in the present document it is strongly recommended that the reader be aware of the content of the most recent version of TS 101 329-6 [3] which is a repository of real results.

Figure 2: Void

1 Scope

The present document provides a collection of informative background information and guidance to supplement parts 1 to part 6 of TS 101 329. The issues covered concern the practical design phases for both equipment and networks with respect to speech performance, and therefore is relevant to TIPHON equipment manufacturers, service providers and network designers.

2 References

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

- [1] ETSI TS 101 329-2: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 3; End-to-end Quality of Service in TIPHON Systems; Part 2: Definition of Speech Quality of Service (QoS) Classes".
- [2] ETSI TS 101 329-5: "Telecommunications and Internet protocol Harmonization Over Networks (TIPHON) Release 3; End-to-end Quality of Service in TIPHON Systems; Part 5: Quality of Service (QoS) measurement methodologies".
- [3] ETSI TS 101 329-6: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 3; End-to-end Quality of Service in TIPHON systems; Part 6: Actual measurements of network and terminal characteristics and performance parameters in TIPHON networks and their influence on voice quality".
- [4] ITU-T Recommendation G.100: "Definitions used in Recommendations on general characteristics of international telephone connections and circuits".
- [5] ITU-T Recommendation G.122: "Influence of national systems on stability and talker echo in international connections".
- [6] ITU-T Recommendation G.131: "Control of talker echo".
- [7] ITU-T Recommendation G.111: "Loudness ratings (LRs) in an international connection".
- [8] ITU-T Recommendations P.64: "Determination of sensitivity/frequency characteristics of local telephone systems".
- [9] ITU-T Recommendation G.109: "Definition of categories of speech transmission quality".
- [10] ITU-T Recommendation G.726: "40, 32, 24, 16 kbit/s adaptive differential pulse code modulation (ADPCM)".
- [11] ITU-T Recommendation G.114: "One-way transmission time".
- [12] ETSI I-ETS 300 245-2: "Integrated Services Digital Network (ISDN); Technical characteristics of telephony terminals; Part 2: PCM A-law handset telephony".
- [13] ITU-T Recommendation P.800: "Methods for subjective determination of transmission quality".
- [14] ITU-T Recommendation P.310: "Transmission characteristics for telephone-band (300-3400 Hz) digital telephones".
- [15] ANSI/TIA/EIA-810-A-2000: "Telecommunications-Telephone Terminal Equipment-Transmission Requirements for Narrowband".
- [16] ITU-T Recommendation G.168: "Digital network echo cancellers".
- [17] ITU-T Recommendation G.107: "The E-Model, a computational model for use in transmission planning".
- [18] ITU-T Recommendation G.113: "Transmission impairments due to speech processing".

- [19] ETSI EG 201 050: "Speech Processing, Transmission and Quality Aspects (STQ); Overall Transmission Plan Aspects for Telephony in a Private Network".
- [20] ITU-T Recommendation G.108: "Application of the E-model: A planning guide".
- [21] ETSI ETR 250: "Transmission and Multiplexing (TM); Speech communication quality from mouth to ear for 3,1 kHz handset telephony across networks".
- [22] ITU-T Recommendation P.833: "Methodology for derivation of equipment impairment factors from subjective listening?only tests".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the following terms and definitions apply:

dBm: power level with reference to 1 mW

dBm0: at the reference frequency (1 020 Hz), L dBm0 represents an absolute power level of L dBm measured at the transmission reference point (0 dB_r point), and a level of L + x dBm measured at a point having a relative level of x dB_r

NOTE: See ITU-T Recommendation G.100 [4], annex A.4.

echo: unwanted signal delayed to such a degree that it is perceived as distinct from the wanted signal

talker echo: echo produced by reflection near the listener's end of a connection, and disturbing the talker

listener echo: echo produced by double reflected signals and disturbing the listener

Loudness Rating (LR): as used in the G-Series Recommendations for planning; loudness rating is an objective measure of the loudness loss, i.e. a weighted, electro-acoustic loss between certain interfaces in the telephone network

NOTE: If the circuit between the interfaces is subdivided into sections, the sum of the individual section LRs is equal to the total LR. In loudness rating contexts, the subscribers are represented from a measuring point of view by an artificial mouth and an artificial ear respectively, both being accurately specified.

Overall Loudness Rating (OLR): loudness loss between the speaking subscriber's mouth and the listening subscriber's ear via a connection

Talker Echo Loudness Rating (TELRL): loudness loss of the speaker's voice sound reaching his ear as a delayed echo

NOTE: See ITU-T Recommendation G.122 [5], clause 4.2 and ITU-T Recommendation G.131 [6], figure I.1.

Terminal Coupling Loss weighted (TCLw): weighted coupling loss between the receiving port and the sending port of a terminal due to acoustical coupling at the user interface, electrical coupling due to crosstalk in the handset cord or within the electrical circuits, seismic coupling through the mechanical parts of the terminal

NOTE: For a digital handset it is commonly in the order of 40 db to 46 dB.

weighted Terminal Coupling Loss-single talk (TCLwst): weighted loss between Rin and Sout network interfaces when AEC is in normal operation, and when there is no signal coming from the user

weighted Terminal Coupling Loss-double talk (TCLwdt): weighted loss between Rin and Sout network interfaces when AEC is in normal operation, and when the local user and the far-end user talk simultaneously

Send Loudness Rating (SLR) (from ITU-T Recommendation G.111): loudness loss between the speaking subscriber's mouth and an electric interface in the network

NOTE: The loudness loss is defined here as the weighted (dB) average of driving sound pressure to measured voltage. The weighted mean value for ITU-T Recommendations G.111 [7] and G.121 (see Bibliography) is 7 to 15 in the short term, 7 to 9 in the long term. The rating methodology is described in ITU-T Recommendations P.64 [8], P.76 (see Bibliography) and P.79 (see Bibliography).

Receive Loudness Rating (RLR) (from ITU-T Recommendation G.111): loudness loss between an electric interface in the network and the listening subscriber's ear

NOTE: The loudness loss is here defined as the weighted (dB) average of driving e.m.f. to measured sound pressure. The weighted mean value for ITU-T Recommendations G.111 [7] and G.121 (see Bibliography) is 1 to 6 in the short term, 1 to 3 in the long term. The rating methodology is described in ITU-T Recommendations P.64 [8], P.76 (see Bibliography), P.79 (see Bibliography).

Circuit Loudness Rating (CLR): loudness loss between two electrical interfaces in a connection or circuit, each interface terminated by its nominal impedance which may be complex

toll quality: In general "toll quality" is a term which is not well defined. Currently, there are two different views:

- ITU-T Recommendation G.109 [9] provides the following guidance:

"Finally, to relate the definitions provided by this Recommendation to concepts and terminology used in the past, a comment about "toll quality" is in order. "Toll quality" has been used by many different people to mean different things, but to network planners it really meant that technology being introduced into the network was robust to the effects of transmission impairments from other sources, and could thus be used in many configurations where inter-working with other systems would be necessary. In this context, the term "toll quality" does not have any absolute relation to speech transmission quality today, because, for example, the impairments of systems such as wireless access or packet-based transport will have the same impact regardless of whether on a local or a long-distance connection. Instead, the terminology provided here is recommended (i.e. "best" for R in the range from 90 to 100, "high" in the range from 80 to 90 and "medium" in the range from 70 to 80)."

- Experts on low bit-rate coding (members of ITU-T Study Group 16 and SQEG) use the following explanation:

"In summary, we define toll quality as equivalent to wire-line telephone quality. Basically the 32 kb/s ADPCM (ITU-T Recommendation G.726 [10]) is considered to be a toll quality coder, and when some low rate coders get standardized in ITU-T, 32 kb/s ADPCM is used as reference, and if a low rate coder produce equivalent performance to the 32 kb/s ADPCM, then this is considered to be toll quality."

Consequently, at this time the term "toll quality" should be considered as an internal term of speech coder experts only which is obsolete and which should be avoided in conjunction with the TIPHON QoS documentation. TIPHON equipment manufacturers and network designers should rather use the Quality Categories defined in ITU-T Recommendation G.109 [9] or the QoS Classes specified in TIPHON (TS 101 329-2 [1]).

NOTE: Harmonization of the views regarding the term "toll quality" are envisaged to be discussed during the Study Period 2001-2004.

3.2 Abbreviations

For the purposes of the present document, the following abbreviations apply:

ACR	Absolute Category Rating
ADSL	Asymmetric Digital Subscriber Line
AEC	Acoustic Echo Control
ALC	Automatic Level Control
ALC	Automatic Level Control
ASL	Active Speech input Level
ATM	Asynchronous Transfer Mode
DCME	Digital Circuit Multiplication Equipment
DTMF	Dual Tone Multi Frequency
DTX	Discontinuous Transmission
ECD	Echo Control Devices
GSM EFR	GSM Enhanced Full Rate Speech Coder
GSM FR	GSM Full Rate Speech Coder
GSM HR	GSM Half Rate Speech Coder
GSM	Global System for Mobile communications

IP	Internet Protocol
ISDN	Integrated Services Digital Network
ISP	Internet Service Provider
IWF	Inter Working Function
LAN	Local Area Network
MOS	Mean Opinion Score
MPLS	Multi Protocol Layer Switching
NIC	Network Interface Card
NS	Noise suppressors
PPP	Point to Point Protocol
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RSVP	Resource Reservation Set-Up Protocol
RTP	Real-Time Transport Protocol
SBM	Subnet Bandwidth Manager
SCN	Switched Communications Network
TCP	Transmission Control Protocol
TRM	Transmission Rating Model
UDP	User Datagram Protocol
VAD	Voice Activity Detection
VAD	Voice Activity Detectors
VDSL	Very High Speed Digital Subscriber Line
VED	Voice Enhancement Devices
VoIP	Voice over IP
VTOA	Voice and Telephony Over ATM
WAN	Wide Area Network
xDSL	ADSL, VDSL and other Digital Subscriber Line Techniques

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4 General considerations

SIST-TP TR 101 329-7 V2.1.1:2004

The realization of end to end speech quality in a TIPHON system is determined by a combination of user equipment design, service provider equipment performance and network transmission planning. To guarantee end to end speech quality:

- user equipment must meet specified performance requirements; and
- service provider equipment must meet specified performance requirements and be correctly configured by service providers;
- underlying transport networks, involved in the call end-to-end (IP as well as SCN), must be designed to deliver specific performance criteria at all times. It is implicit that guarantees can only be achieved over managed IP networks, engineered to deliver a given level of performance, and where traffic levels are controlled.

The issues of end-to-end speech transmission quality need to be considered from various perspectives therefore:

- user equipment and service provider equipment design by manufactures;
- system configuration by service providers;
- transmission planning by network operators.

The purpose of the present document is to provide design guidelines in each of these areas.

The following steps are likely to be involved in implementing a TIPHON system:

- Planning and configuration;
- Pre-qualification;
- User interaction;

- Maintenance;
- Monitoring and Verification,

which are summarized in the following clauses.

4.1 Transmission planning

In order to deliver the intended end-to-end speech transmission quality in TIPHON systems, transmission planning should be performed during the design phase of TIPHON related equipment. It is not sufficient to design equipment or networks just along the requirement limits of the respective TIPHON class.

An advantage factor A (see ITU-T Recommendation G.107 and Appendix II to ITU-T Recommendation G.113) which is sometimes discussed for Internet-Telephony does not generally apply for business applications and consequently does also not apply for TIPHON systems. However, this is not a general planning rule, but a business and customer related decision for this single example case or for a specific service.

Any variation of transmission parameters should only be judged on the basis of E-model calculations for critical end-to-end connections. Any assumption whether or whether not a specific parameter variation will be perceived by the user should always be based on E-model calculations.

Special care should be taken with devices which dynamically vary one or more transmission parameters, e.g. Automatic Level Control (ALC) devices; experiences with such devices have shown that they have the potential to impact end-to-end speech transmission quality, severely.

4.2 User interaction

User interaction with regard to the change of certain transmission parameters may be provided by equipment which forms part of a TIPHON connection, e.g. a TIPHON terminal may include a PC client software which provides adjustment of Loudness Rating to the user (see clause 5.1.2 for further guidance).

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4.3 Maintenance

After TIPHON equipment and networks have been designed, planned and rendered operative in compliance with one the TIPHON QoS classes it might - nevertheless - occur that users complain about too low speech quality.

In such cases, it is very important to be able to carry through a diagnosis of end-to-end speech transmission performance. For that it will be needed to keep track of all parameter changes (e.g. of Send and Receive Loudness Rating) carried out either automatically or by user interaction.

This should be considered already during the design phase of TIPHON equipment and networks, e.g. by providing tools to set parameters back to default values or by providing a log file function.

4.4 Monitoring & verification

Even if a specific TIPHON system has been operated for some time at the desired level of customer satisfaction it will be required to continuously monitor and check the end-to-end speech transmission quality.

Verification will require access to the actual settings of all major transmission parameters - including those which were accessible to the user.

4.5 Interconnection of TIPHON systems with other IP networks

Implementers of TIPHON networks are advised that IP networks other than those following the TIPHON regime, eventually may employ different QoS classification schemes than the one defined in [1].

As an example, in the following the TIPHON approach is compared to the TIA approach taken in [2].

Explanation of the arrangement of the figures

The figures are drawn with LSQ as the Y-axis and delay as the X-axis as these are the main design parameters over which the user has some control.

TIPHON sets three criteria for its QoS classes as shown in the following table which is an excerpt from [1].

	2H	2M	2A
OVR	80	70	50
Delay	100 ms	150 ms	400 ms
LSQ	86	73	50

Whilst the criteria for LSQ and delay are independent of each other, the criterion on OVR depends on LSQ, delay and other parameters. Some assumptions therefore have to be made for the other parameters as follows:

- Perfect echo cancellation is assumed for all TIPHON systems therefore, TELR = 65 dB and WEPL = 110 dB (default as per G.107). Whereas the need for proper echo control is recognized in general, other IP networks may consider the quality of echo cancellation as actually achieved in their QoS classification.
- The E-model default values are assumed for all terminal related parameters.
- The Overall E-Model Rating R is calculated according to E-model, with TELR, WEPL and terminal related parameters as above (G.107 default) and the values of delay and LSQ as on the graph.

Figure 3 illustrates the TIPHON QoS classes as described in TS 101 329-2 [1].

TIPHON QoS classes are defined by a combination of all three metrics, LSQ, delay and OVR. Therefore, the colored areas represent the TIPHON QoS classes as follows:

Green Narrowband High (2H)

Yellow Narrowband Medium (2M)

Red Narrowband Acceptable (2A)

Mauve Not Recommended (1)

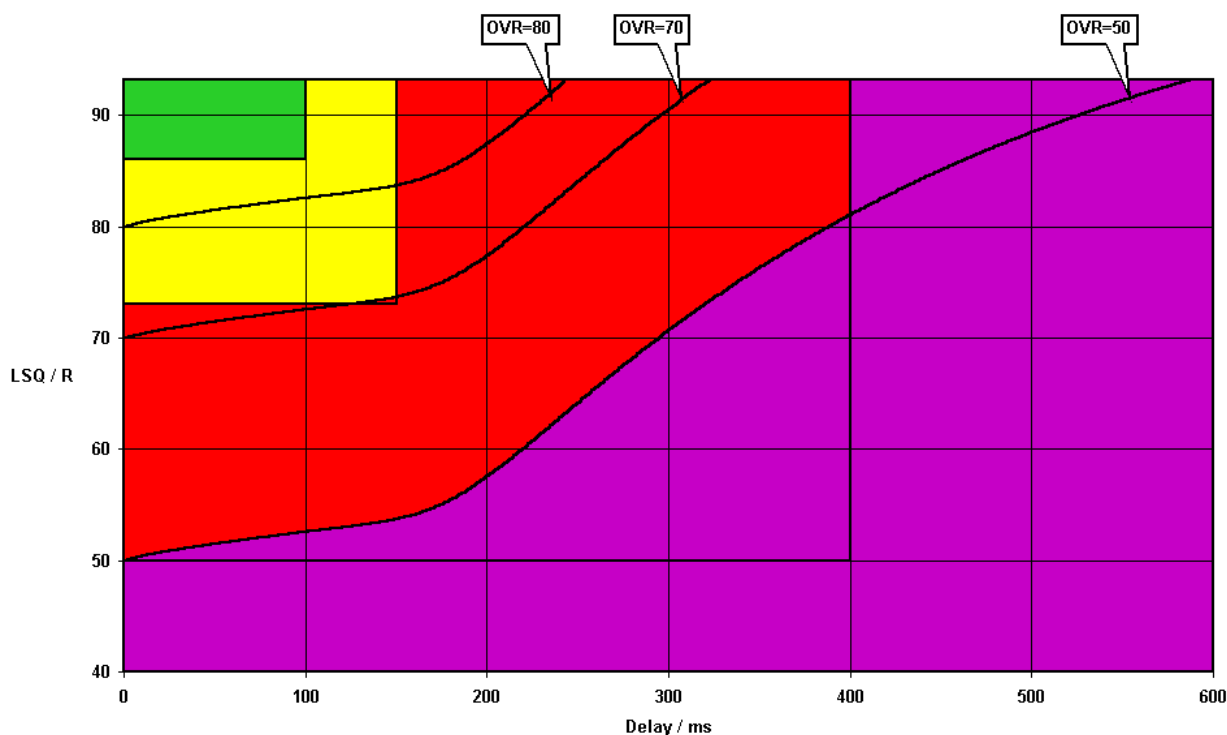


Figure 3: TIPHON QoS classes

Figure 3 shows how the criteria for LSQ and delay are more stringent for the high and medium quality classes than the criterion on OVR under the other conditions chosen. But for acceptable and not recommended quality, OVR is the most stringent criterion.

Figure 4 illustrates the TIA approach as contained in [2].

TIA QoS classes are defined by a pure E-Model approach, employing OVR, only. Therefore, the colored areas represent the TIA QoS classes as follows:

Green	High
Yellow	Medium
Red	Low
Mauve	3Not Recommended

Please, note that, the vertical and horizontal lines which indicate the TIPPHON requirements on LSQ and delay have been added to the TIA diagram in figure 4 in order to accomplish a more convenient comparability with figure 3.

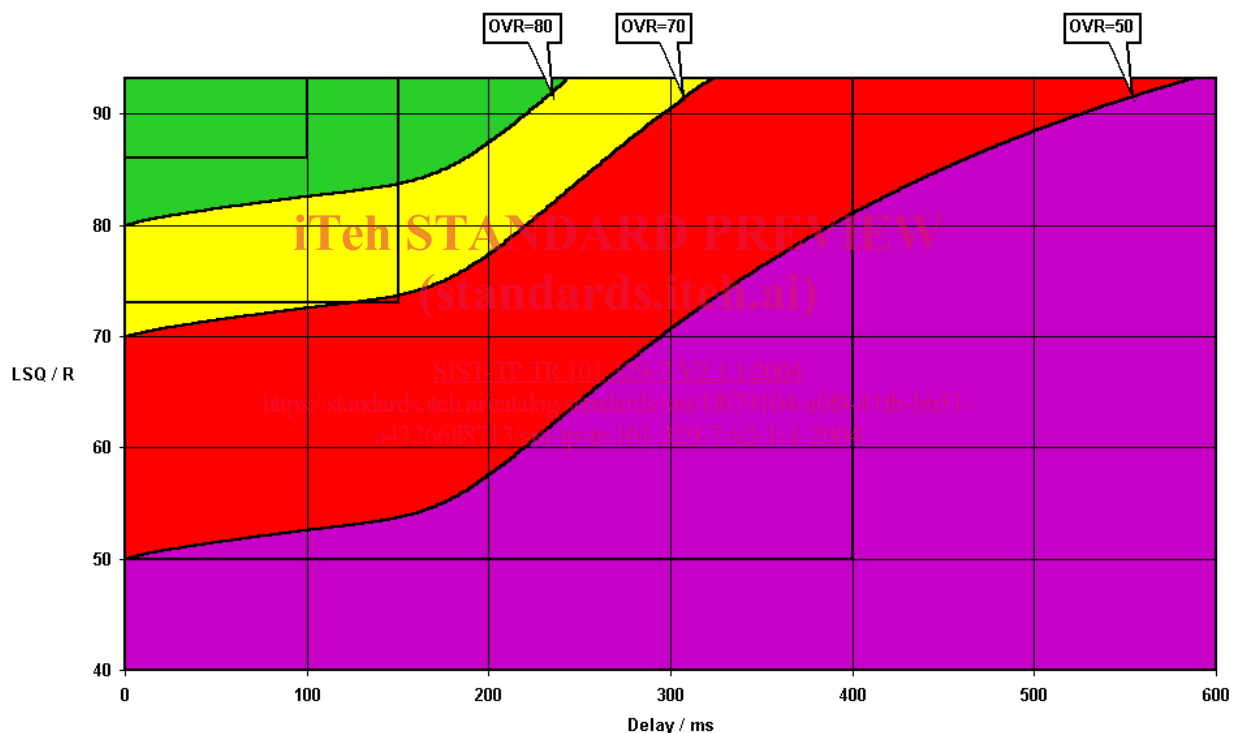


Figure 4: TIA QoS classes

Furthermore, it should be noted that in IP networks other than TIPPHON one or all of the following may differ:

- name or description of QoS classes;
- guarantees provided or given for each individual class;
- availability and description of wideband classes.

In cases where TIPPHON systems are interconnected to other IP networks, implementers of TIPPHON systems, therefore, should thoroughly inspect the basic transmission parameters of the end-to-end connection.

Any conclusion of the resulting end-to-end speech transmission performance of such an interconnection scenario should be based on end-to-end E-model calculations. A mapping or transformation of TIPPHON QoS classes with classes of other IP networks is, in general, not feasible.