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Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 3; End-to-end Quality of Service in TIPHON systems; Part 1: General aspects of Quality of Service (QoS)

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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; Part 1: General aspects of Quality of Service (QoS)

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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 1 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 (see Bibliography), and more comprehensive versions of the Release 3 documents listed above will be published as part of Release 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 5addressing Quality of Service (QoS) in TIPHON Systems. The structure of this work is illustrated in figure 1.

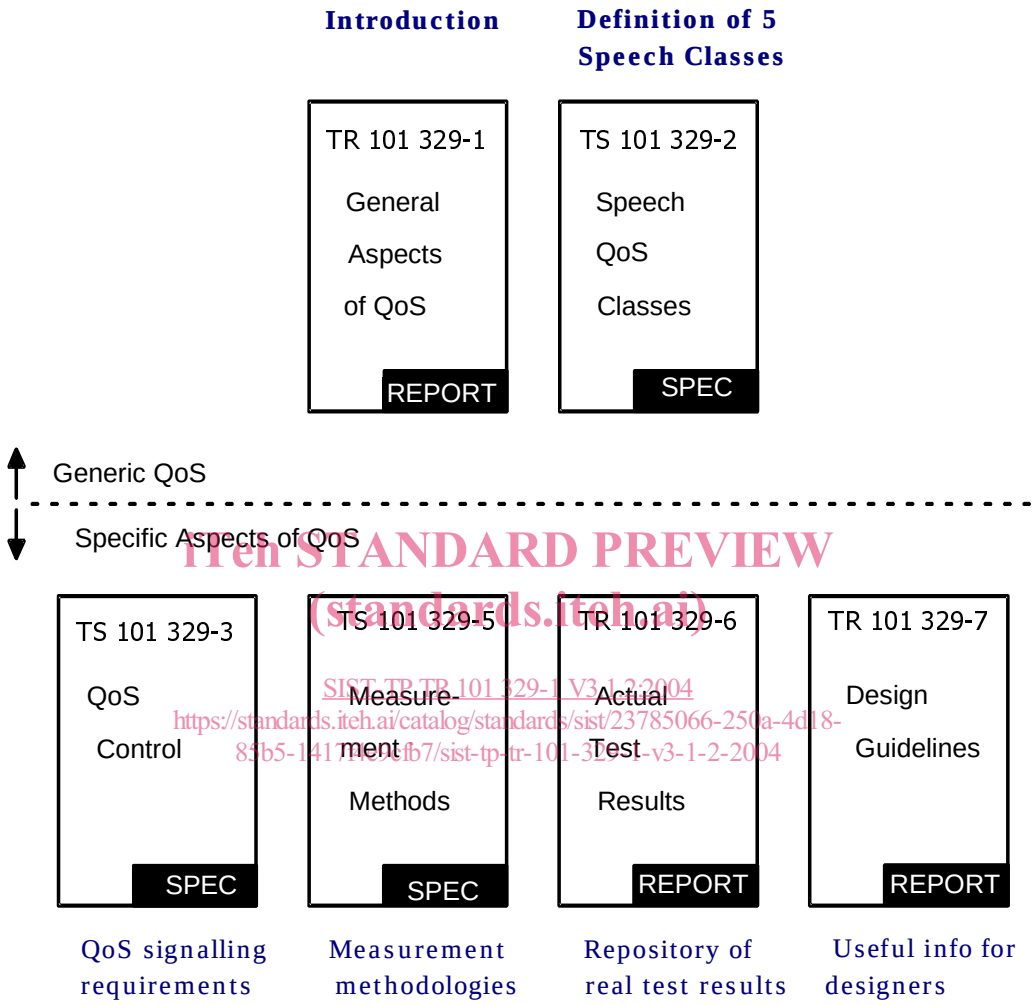

Figure 1: Structure of TIPHON QoS Documentation for Release 3

1 Scope

The present document presents QoS related background information for IP networks that provide voice telephony in accordance with all TIPHON scenarios.

It contains:

- a depiction of each TIPHON scenario by its reference connection;
- an overview of the physical components based on which a TIPHON service may be provided.

2 References

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

[1] ETSI EG 202 306: "Transmission and Multiplexing (TM); Access networks for residential customers".

[2] ETSI TS 101 312: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON); Network architecture and reference configurations; Scenario 1".

[3] ETSI TS 101 329-2: "Telecommunications and Internet Protocol Harmonization over Networks (TIPHON); End to End Quality of Service in TIPHON Systems; Part 2: Definition of Quality of Service (QoS) Classes".

[4] ETSI TS 101 329-5: "Telecommunications and Internet protocol Harmonization Over Networks (TIPHON) Release 3; Technology Compliance Specification; Part 5: Quality of Service (QoS) measurement methodologies".

[5] ITU-T Recommendation E.600: "Terms and definitions of traffic engineering".

[6] ITU-T Recommendation G.103: "Hypothetical reference connections".

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

[8] ITU-T Recommendation G.177: "Transmission planning for voiceband services over hybrid Internet/PSTN connections".

[9] ITU-T Recommendation I.350: "General aspects of quality of service and network performance in digital networks, including ISDNs".

[10] ITU-T Recommendation P.310: "Transmission characteristics for telephone-band (300-3 400 Hz) digital telephones".

3 Definitions and abbreviations

3.1 Definitions

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

end-to-end delay jitter: estimate of the statistical variance of the voice frames interarrival time measured in milliseconds and expressed as an unsigned integer

NOTE: The end-to-end delay jitter is defined to be the mean deviation (smoothed absolute value) of the difference in voice frame spacing at the receiver compared to the sender for a pair of voice frames.

interarrival jitter: estimate of the statistical variance of the RTP data packet interarrival time measured in milliseconds and expressed as an unsigned integer

NOTE: The interarrival jitter is defined to be the mean deviation (smoothed absolute value) of the difference in packet spacing at the receiver compared to the sender for a pair of packets.

jitter amplitude: absolute difference in arrival time between the fastest and the slowest data packet or voice frame

3.2 Abbreviations

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

ADSL	Asymmetric Digital Subscriber Line
BRAN	Broadband Radio Access Networks
FDM	Frequency Division Multiplex
GSM	Global System for Mobile communications
ISDN	Integrated Services Digital Network
IP	Internet Protocol
ITU	International Telecommunication Union
ITU-T	ITU Telecommunication Standardization Sector (former CCITT)
IWF	InterWorking Function
LAN	Local Area Network
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RTP	Real Time Protocol
SCN	Switched Communications Network
TDM	Time Division Multiplex
UMTS	Universal Mobile Telecommunications System
VDSL	Very High Speed Digital Subscriber Line
xDSL	ADSL, VDSL and other Digital Subscriber Line Techniques

4 Introduction to end-to-end Quality of Service (QoS)

4.1 Main QoS parameters influenced by TIPHON systems

End-to-end QoS in a TIPHON system is characterized in the TIPHON QoS documentation under two broad headings:

- call set-up quality; and
- call quality.

4.1.1 Call set-up quality

Call set-up quality is mainly characterized by the call set-up time which is perceived by the user as the responsiveness of the service. Call set-up time is the time elapsed from the end of the user interface command by the caller (keypad dialling, E-mail alias typing, etc.) to the receipt by the caller of a meaningful progress information. The present document provides the exact definition of the various call set-up times for use in TIPHON systems, whereas ITU-T Recommendation E.600 [5] provides more information on the definition of post-dialling delay in SCN systems.

4.1.2 Call quality

Call quality is characterized by the overall transmission quality rating R. Overall transmission quality rating (R) describes the full acoustic-to-acoustic (mouth to ear) quality, experienced by a user, for a typical situation using a "standard" telephony handset. The overall transmission quality rating is calculated using the E-Model (see ITU-T Recommendation G.107 [7]). For calculation purposes the use of traditional telephone handsets (see ITU-T Recommendation P.310 [10]) at both sides of the connection is assumed.

Within the overall transmission quality two major factors contribute to the overall QoS experience of the user of the TIPHON system:

- end-to-end delay: this mainly impacts the interactivity of a conversation. The measurement is done from the mouth of the speaker to the ear of the listener; and
- end-to-end speech quality: this is the one way speech quality as perceived in a non interactive situation.

The measurement methodologies for these parameters are specified in TS 101 329-5 [4], while the requirements for these parameters with respect to the various TIPHON QoS classes are defined in TS 101 329-2 [3].

TR 101 329-7 (see bibliography) provides guidance on these parameters with respect to the practical design phase of equipment and networks.

4.1.3 Conversational speech quality

The conversational quality of a telephone link is influenced by four parts:

- Listening quality, the quality of the speech received from the talker's voice at the other side, dominated by noise and speech distortion.
- Talking quality, the quality of the speech received from the talker's own voice, dominated by echo and sidetone distortion.
- Interaction quality, the quality associated with the alternation of talking and listening, dominated by end-to-end delay and noise/speech switching.
- Background noise transmission quality, the quality of the transmission of background noises received from the other side. Note that Background noise transmission quality and Listening quality are not independent: the first may or may not have been included in the latter.

4.2 Further QoS parameters

In general, Quality of Service (QoS) is determined by a multitude of further QoS parameters; guidance in this field is provided by ITU-T Recommendation I.350 [9].

For the complexity of other QoS parameters it is considered that:

- they either do not apply to TIPHON systems; or
- the TIPHON systems have similar influence on those parameters like other telephony systems.

4.3 TIPHON specific QoS relevant factors

Examples of TIPHON specific QoS relevant factors are:

- number of hops;
- possible variation of the geographical length of one connection during the talking state;
- occurrence of congestion;
- use of prioritization or bandwidth reservation schemes;
- jitter and jitter buffer behaviour (see clauses 4.3.1 through 4.3.4).

TR 101 329-7 (see bibliography) provides guidance on these factors with respect to the practical design phase of equipment and networks.

4.3.1 Delay Jitter

Packetized transmission systems exhibit variable delay in packet delivery time; this is caused by the fact that different packets carrying speech samples of the same telephone conversation may be transported via distinct routes through the network or queueing of data, voice and other voice streams on the same route: details of this effect depend strongly on the specific mechanisms for transport, queuing or prioritization, which may be implemented in such a system.

Packets which have been transported through a packet based network are collected in a buffer at the receive side. This buffer functions as the instance which re-arranges the timely order of the packets. If the delivery time of a packet exceeds the length of the receive buffer, then this packet "comes too late" with respect to the size of this buffer and will be discarded. Hence, the speech carried in this packet is lost for the decoding process. This "packet loss" impacts speech transmission quality.