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Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 4; End-to-end Quality of Service in TIPHON Systems; Part 12: IP Telephony Service Availability

Reference

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Foreword

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

The present document is part 12 of a multi-part deliverable covering end-to-end Quality of Service in TIPHON Systems, as identified below:

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

Part 2: "Definition of Speech Quality of Service (QoS) Classes";

Part 3: "Signalling and Control of end-to-end Quality of Service";

Part 4: "Quality of Service Management";

Part 5: "Quality of Service (QoS) measurement methodologies";

Part 6: "Actual measurements of network and terminal characteristics and performance parameters in TIPHON networks and their influence on voice quality";

Part 7: "Design Guide for elements of a TIPHON connection from an end-to-end speech transmission performance point of view";

Part 8: (to be developed during TIPHON Release 5): "QoS multimedia services in TIPHON systems";

Part 9: "Call Performance Classification (Voice)";

Part 10 (to be developed during TIPHON Release 5): "Requirements for TIPHON Terminals and User Network Aspects";

Part 11: (to be developed during TIPHON Release 5): "Domain by domain performance planning guidelines for end-to-end QoS objectives associated with TIPHON speech QoS classes";

Part 12: "IP Telephony Service Availability".

Introduction

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

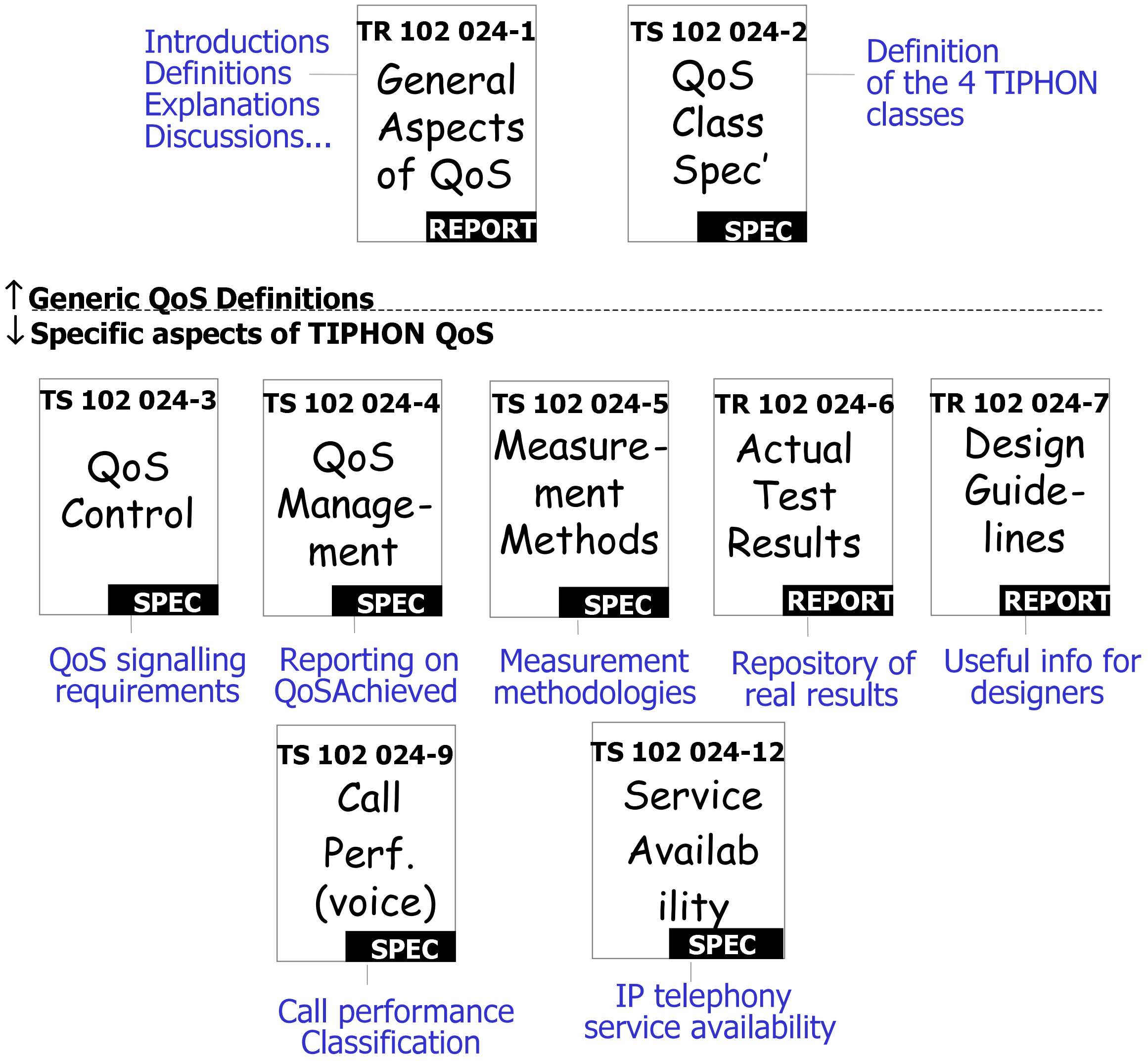

Figure 1: TIPHON WG5 QoS Documentation Structure

1 Scope

The present document defines performance parameters and related criteria in order to specify the Availability performance of the IP Telephony service.

The present document defines Service Availability, as in a multiservice IP network it is useful to define Availability from a service point of view (which can be related to a particular service) rather than from a network perspective.

The present document defines performance parameters for determining IP Telephony Service Availability for an established IP Telephony session. The performance parameters refer to IP Packet Transfer as provided by an IP data communication service.

The Application Plane issues of service availability are out of the scope of the present document. They are addressed in TS 102 024-9 [9], where control related parameters have been identified.

In the present document the methodology used to define IP Telephony Service Availability is illustrated in detail. Values of performance parameters, if not defined, are for further study.

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication and/or edition number or version number) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies.

Referenced documents which are not found to be publicly available in the expected location might be found at http://docbox.etsi.org/Reference.

[1] ITU-T Recommendation I.380 (Y.1540) "Internet protocol data communication service - IP Packet transfer and availability performance parameters".

[2] Void.

[3] ETSI TS 102 024-2: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 4; End-to-end Quality of Service in TIPHON Systems; Part 2: Definition of Speech Quality of Service (QoS) Classes".

[4] ITU-T Recommendation E.855: "Connection Integrity objective for the international Telephone Service".

[5] ETSI TR 101 300: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON); Description of Technical Issues".

[6] Void

[7] ETSI TS 102 024-6: "Telecommunication and Internet Protocol Over Networks (TIPHON); End-to-end Quality of Service in TIPHON System Part 6: Actual measurements of network and terminal characteristics and performance parameters in TIPHON networks and their influence on voice quality".

[8] ITU-T Recommendation D.22 SG13 Geneva, 20-24 November 2000: "Service Availability definition for VoIP".

[9] ETSI TS 102 024-9: "Telecommunication and Internet Protocol Over Networks (TIPHON); End-to-end Quality of Service in TIPHON System Part 9: Call performance Classification".

[10] ITU-T Recommendation E.850: "Connection Retainability Objective for the International Telephone Service".

[11] ITU-T Recommendation E.800: "Terms and definitions related to quality of service and network performance including dependability".

[12] IETF RFC 1889: "RTP: A Transport Protocol for Real-Time Applications".

3 Definitions and abbreviations

3.1 Definitions

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

IP Telephony service: telephony related service that is supported on a managed IP network

Population of interest: total set of packets sent from SRC to DST

3.2 Abbreviations

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

CoS	Class of Service
DST	DeSTination
IP	Internet Protocol
IPDV	IP Packet Delay Variation
IPLBD	IP Packet Loss Burst Duration
IPLR	IP Packet Loss Ratio
IPNLR	IP Network packet Loss Ratio
IPTD	IP Packet Transfer Delay
MPs	Measurement Points
MTBSO	Mean Time Between Service Outage
PLC	Packet Loss Concealment
PSTN	Public Switched Telephone Network
QoS	Quality of Service
SA	Service Availability
SRC	SouRCe

4 Generic IP telephony service performance model

In ITU-T Recommendation E.800 [11] the Availability performance is defined as the ability of an item to be in a state to perform a required function at a given instant of time or at any instant of time within a given time interval. The item can be any part, device, equipment of the network, that can be individually considered and perform a predefined function; it may consist of hardware, software or both: the item may be a switch, a link or a service.

The Availability of a multiservice IP network (IP-based network capable of delivering different types of service) can be usefully defined from a service point of view (can be referred to the particular service), thus providing a discussion of the concept of service outage based on a given service.

EXAMPLE: A 10-second network outage may result in a failure for IP telephone service, but go unnoticed for email.

For this reason, in IP-based telecommunications networks, we consider the **Service Availability**. For the scope of the present document the Service Availability requirements for IP Telephony service are analysed.

4.1 Service availability model

A service is considered to be available when the values of a set of selected performance parameters are deemed acceptable. The value of a specific parameter is deemed acceptable if it is greater (or lower) than of a pre-specified threshold.

IP Telephony Service Availability is impacted by the Application plane performance as well as the Transport plane performance: in order to make the telephone service available for the user the call has first to be actually set up and then (when the call has been established) the packet flow has to be transferred between the end-users. Failure of one or both of these processes heavily impacts the service availability.

For the purpose of the present document we consider the IP telephony service at the Transport Plane level, which is the user information transfer communication function as defined in ITU-T Recommendation I.380 [1]. Therefore the performance parameters and the relative thresholds, used to determine the service availability, refer to the transfer of IP packet over the IP-based network (see clause5).

The Application Plane performance parameters are considered in TS 102 024-9 [9].

The combination of these IP Packet transfer performance parameters and their thresholds is called the Availability Function. The relation between a performance parameter and the threshold(s) defines an outage criterion. Every time that an outage criterion is met there is a transition to the unavailable state. The available state is re-entered when all parameters are once again acceptably functioning.

The outage criteria are defined considering network performance that falls below TIPHON Narrowband acceptable class.

An associated two-state model provides a basis for describing IP telephone service availability (see figure 2).

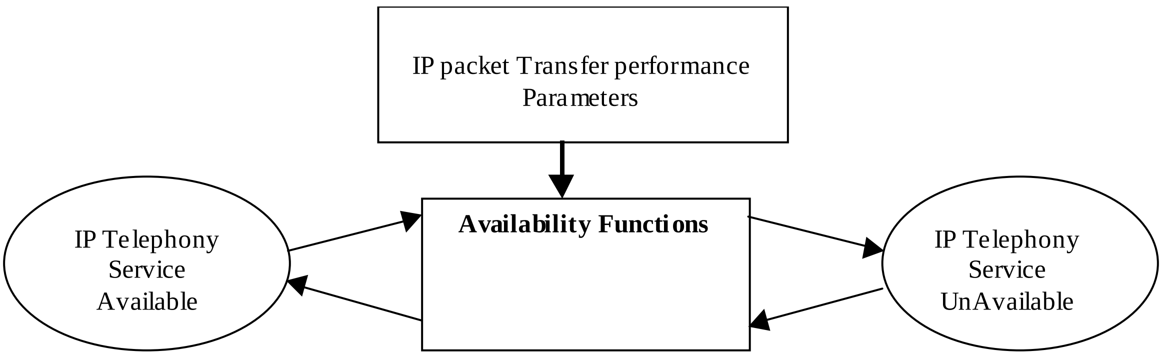

Figure 2: Two-state model for Service Availability

4.2 Reference Model

According to ITU-T Recommendation I.380 [1] a generic IP service performance model is defined in order to provide the basis for:

- identifying the blocks with which the IP Telephony service may be represented;
- identifying the measurement points to be considered.

The model and the performance parameters defined are independent from the transport technologies (es. ATM, SDH).

The Reference Model is defined in TS 102 024-7.

5 IP packet transfer performance parameters

As indicated in clause 4.1, the IP Telephony service performance parameters are defined on the basis of IP packet transfer reference events that may be observed at measurement points (MPs) associated with specified functional and jurisdictional boundaries.

This clause defines the set of IP packet transfer performance parameters that could be used to define the IP telephony service availability.

5.1 IP packet Loss Ratio (IPLR)

IP packet loss ratio is the ratio of total lost IP packet outcomes to total transmitted IP packets in a population of interest.

A lost packet outcome occurs when:

- a packet is never delivered or delivered to an unpermitted egress point (IPNLR);
- a packet is delivered to a permitted egress point out of the specified threshold value for IPTD.

IPDV is not compensated by the dejitter buffer (the voice application does not consider packet with jitter more than a fixed threshold).

5.2 IP Network packet Loss Ratio (IPNLR)

IP network packet loss ratio is the ratio of total number of never delivered packets and/or packets delivered to an unpermitted egress point to total transmitted IP packets in a population of interest.

5.3 IP packet Transfer Delay (IPTD)

The IP packet transfer delay is defined as the one-way delay between the MPs at the ingress and egress of IP network.

This delay is defined for all packet outcomes.

5.4 IP packet Delay Variation (IPDV)

Variations in IP packet transfer delay (jitter) according to RFC 1889 [12].

5.5 IP packet Loss Burst Duration (IPLBD)

IP packet Loss Burst Duration is defined as the time duration of the transmission interruption due to a continuity loss of packets occurs in a population of interest.

6 IP telephony availability

Service Availability is applied to the IP telephony service between two functional elements, identified by their IP addresses, located at the ingress point and at the egress point of the IP network.

IP telephony Availability requirements are defined to the user to network interfaces shown in the reference model. These requirements could be applied to all scenarios defined in TR 101 300 [5].

6.1 Availability criteria

This clause describes functions that define outage criteria.

6.1.1 IP Telephony service availability function

The IP Telephone Service Availability function serves to classify the total scheduled service time into available (up time) and unavailable (down time) periods.

The basis for the availability function is the definition of the threshold values for the performance parameters described above.

The availability function for IP Telephony can be defined in two ways, depending on which performance parameters the measurement systems are able to determine.

These two methods are:

- The definition of a threshold on the IPLR performance. In this case, if the outage criterion is met (see table 1), the IP telephone service is in the unavailable state. The service is in the available state if the outage criterion is not satisfied.

- The definition of thresholds on the IPNLR, IPTD and IPDV performances. In this case, if one or more of the outage criteria are satisfied (see table 2) the service is in the unavailable state. The service is available if all the outage criteria are not satisfied.

Table 1 defines the outage criterion on IPLR.

Table 1: IP telephony service availability function (case 1)

Outage criterion	Threshold
IPLR > C1	C1 = X

When IPLR > C1 the IP telephone service is unavailable. The threshold C1 is only to be used for determining when the IP network resources are temporarily incapable of supporting a useful IP packet transfer for telephony service; thus this threshold is related to the telephony service characteristics.

IPLR parameter is evaluated on the basis of:

- 1) M_{av} is the minimum number of packets that should be used in evaluating the IP telephone service availability function (the value of Mav is for further study).

- 2) T_{av} is the minimum duration of a time interval during which the IP telephone service availability function is to be evaluated (in ITU-T Recommendation I.380 [1] is defined a provisional value of 5 mins).

The value C1 takes into account the condition of random packet loss. A provisional value for C1 is 5 %. According to TS 102 024-6 [7] G.729A with PLC and framing of 20 ms at 5 % of packet loss has a MOS value lesser than the minimum value allowable for the Narrowband Low CoS.

However, the definition of the threshold value C1, for the evaluation of IP telephony service unavailability with IPLR performance parameter, is for further study.

Table 2 defines the outage criteria on IPNLR, IPTD and IPDV performance parameters.

Table 2: IP telephony service availability function (case 2)

Outage criteria	Threshold
IPNLR > C2	C2 = Y [%]
IPTD > C3	C3 = 400 ms
IPDV > C4	C4 = Z [ms]

The value C2 only takes into account the packet loss within the IP network (never delivered packets and/or packets delivered to unpermitted egress point). Its value (percentage of lost packets in the network) is for further study.

The value C3 is 400 ms because this is the maximum delay for the Narrowband CoS [3].

The value C4 depends on IP network performance and it is for further study.

6.1.2 IP Connection integrity function

An important characteristic of the Service Availability is Service integrity, that is the uninterrupted delivery of the telephone service from the end-user's perspective. Therefore the Service Availability not only requires systems that are highly available, but also carries the caveat that, in order to provide continuity of service to the end-user, the customer data and session state must be preserved during short-term fault conditions .

In actual IP networks, routes are dynamically changed due to congestion or failure of the link. During these events, the routes temporarily become unavailable, causing bursty packet loss. This instantaneous event could not be taken into account by the evaluation of IPNPL or IPLR, which are average values of packet loss. So the network could be in the available state (we are referring to the outage criterion on packet loss performance) but user experiments interruption of the IP Telephony Service.

Speech loss events occur when there is a loss of some consecutive packets between two MPs.

According to ITU-T Recommendation E.850 [10] an established call is prematurely released when:

1) a single interruption, lasting for longer than ten seconds (IPLBD > 10 s), occurs causing the connection transmission quality to be unsuitable for voice communications;

2) a succession of interruptions occurs lasting less than ten seconds and the product of the average duration of each interruption and the occurrence frequency (i.e. average number of interruptions/seconds) exceeds 0,005.

Estimation of the speech loss probability is defined as the ratio of accumulated interruptions duration to the total observation period of time (Tsl). According to ITU-T Recommendation E.855 [4] is:

$$P=\sum_{i=1..N} IPLBDi / Tsl$$

Where Tsl is the observation time and IPLBDi is the time duration of the ith transmission interruption of N transmissions measured during Tsl.

The lower value of IPLBDi to be considered for evaluating P is for further study.

A function based on these two parameters is proposed in the following table:

Table 3: IP connection integrity function

Outage criterion	Threshold
IPCLD > C_5 and/or P > C_6	C_5 = Y C_6 = X

According to ITU-T Recommendation E.855 [4] it has been evaluated that the quality perceived by the user due to the frequency of interruptions is acceptable if P is less than 0,5 %.

A call lasting 3 minutes means for example 3 interruptions that last less than 300 ms or 6 interruptions that last less than 150 ms.

The values of C_5 and C_6 are for further study.

6.2 IP Telephony service availability parameters

Both the availability functions defined in clause 6.1 can be used, depending on the measurement systems, to classify the total scheduled service time into available and unavailable periods for IP Telephone Service. On the basis of this classification we can define the percent IP network availability for the IP Telephony Service.

Another important parameter to evaluate the availability of the network for the IP Telephony Service is the Mean Time Between Service Outages, defined clause 6.2.1.

6.2.1 Service Availability (SA)

Service Availability (SA) is the percentage of scheduled IP telephony service Time (T) during which the network is available for the IP telephony service. This time is categorized as Available (Up time) using the availability criteria defined in clause 6.1.

$$SA = \frac{\text{Accumulated UpTime}}{\text{Accumulated UpTime} + \text{Accumulated DownTime}} = \frac{\text{Accumulated UpTime}}{T}$$

Consequently, the service unavailability is defined as:

SU = 100 - SA

The values of SA and T are for further study.

NOTE: The following values are given as an example of the lower and the upper boundaries for SA.

- for PSTN telephone service SA = 99.998 %; this means that if T = 1 year, the unavailability period is 10 minutes/year.
- According to ITU-T Recommendation D.22 [8], For IP best effort service SA = 99 %, this means that if T = 1 year, the unavailability period is 88 hours/year.

6.2.2 Mean Time Between Service Outage (MTBSO)

Mean time between service outage is the average duration of any continuous interval during which the network is available.

The value of MTBSO is for further study.

Annex A (informative): Bibliography

- D.106 (WP 4/13) Proposal for a new IP performance parameter: "IP Packet severely Loss Ratio (IPSLR) to Y.1540"

History

Document history		
V4.1.1	November 2003	Publication