SIST

SLOVENSKI STANDARD
SIST-TS TS 102 024-9 V4.1.1:2004

01-april-2004

Harmonizacija telekomunikacij in internetnega protokola prek omrežij (TIPHON), 4. izdaja - Kakovost storitve od konca do konca v sistemih TIPHON - 9. del: Klasifikacija zmogljivosti klica (govor)

Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 4; End-to-End Quality of Service in TIPHON Systems; Part 9: Call performance Classification (Voice)

Ta slovenski standard je istoveten z: TS 102 024-9 Version 4.1.1

ICS:

33.020	Telekomunikacije na splošno	Telecommunications in general

SIST-TS TS 102 024-9 V4.1.1:2004 en

ETSI TS 102 024-9 V4.1.1 (2003-01)

Technical Specification

Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 4; End-to-end Quality of Service in TIPHON Systems; Part 9: Call performance Classification (Voice)

Reference

DTS/TIPHON-05016R4

Keywords

IP, performance, QoS, telephony, voice, VoIP

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
Association à but non lucratif enregistrée à la
Sous-Préfecture de Grasse (06) N° 7803/88

Important notice

Individual copies of the present document can be downloaded from:
http://www.etsi.org

The present document may be made available in more than one electronic version or in print. In any case of existing or perceived difference in contents between such versions, the reference version is the Portable Document Format (PDF). In case of dispute, the reference shall be the printing on ETSI printers of the PDF version kept on a specific network drive within ETSI Secretariat.

Users of the present document should be aware that the document may be subject to revision or change of status. Information on the current status of this and other ETSI documents is available at
http://portal.etsi.org/tb/status/status.asp

If you find errors in the present document, send your comment to:
editor@etsi.org

Contents

Intellectual Property Rights

IPRs essential or potentially essential to the present document may have been declared to ETSI. The information pertaining to these essential IPRs, if any, is publicly available for **ETSI members and non-members**, and can be found in ETSI SR 000 314: *"Intellectual Property Rights (IPRs); Essential, or potentially Essential, IPRs notified to ETSI in respect of ETSI standards"*, which is available from the ETSI Secretariat. Latest updates are available on the ETSI Web server (http://webapp.etsi.org/IPR/home.asp).

All published ETSI deliverables shall include information which directs the reader to the above source of information.

Foreword

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

The present document is part 9 of a multi-part deliverable covering the End-to-end Quality of Service in TIPHON Systems, as identified below:

TR 102 024-1: "General aspects of Quality of Service (QoS)";

TS 102 024-2: "Definition of Speech Quality of Service (QoS) Classes";

TS 102 024-3: "Signalling and Control of End-to-end Quality of Service (QoS) in a multi-media environment";

TS 102 024-4: "Quality of Service Management";

TS 102 024-5: "Quality of Service (QoS) measurement methodologies";

TR 102 024-6: "Actual measurements of network and terminal characteristics and Performance parameters in TIPHON networks and their influence on voice quality";

TR 102 024-7: "Design Guide for elements of a TIPHON connection from an End-to-end speech transmission performance point of view";

TS 102 024-9: "Call performance Classification (Voice)";

TS 102 024-10: "QoS Requirements for TIPHON Terminals".

1 Scope

The present document, "TIPHON call performance classification", specifies the signalling aspects associated with the control of End-to-end QoS within and between TIPHON domains for Voice over IP. It defines call and media processing parameters, bounds on the values of these parameters and a possible classification system. The specified parameters apply to session and media flow set-up and close-down as well as to mid-session changes and include issues of signalling accuracy, signalling latency and signalling reliability.

NOTE: The present document only applies to guaranteed QoS classes. Hence it is not applicable to the best effort class.

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] ETSI TS 101 314: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 3; Abstract Architecture and Reference Points Definition; Network Architecture and Reference Points".

[2] ETSI TS 101 882: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 3; Protocol Framework Definition; General (meta-protocol)".

[3] ITU-T Recommendation Y.1530 (February 2002): "Call processing performance for voice service in hybrid IP networks".

[4] ITU-T Recommendation Y.1541: "Network performance objectives for IP-based services".

[5] ITU-T Recommendation Q.766: "Performance objectives in the integrated services digital network application".

[6] ITU-T Recommendation Q.706: "Message transfer part signalling performance".

[7] ITU-T Recommendation Q.709: "Hypothetical signalling reference connection".

[8] Directive 98/10/EC of the European Parliament and of the Council of 26 February 1998 on the application of open network provision (ONP) to voice telephony and on universal service for telecommunications in a competitive environment.

3 Definitions and abbreviations

3.1 Definitions

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

unsuccessful call: call attempt to a valid number, properly dialled following dial tone, where neither called party busy tone, nor ringing tone, nor answer signal, is recognized on the access line of the calling user within 30 s from the instant when the address information required for setting up a call is received by the network

unsuccessful call ratio: ratio of unsuccessful calls to the total number of call attempts in a specified time period

3.2 Abbreviations

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

CASD	Call Answer Signal Delay
CPDP	Call Premature Disconnect Probability
CPRP	Call Premature Release Probability
CRFP	Call Release Failure Probability
CRD	Call Release Delay
CSD	Call Setup Delay
CSEP	Call Setup Error Probability
CSFP	Call Setup Failure Probability
DNS	Domain Name Server
FFS	For Further Study
ISDN	Integrated Service Digital Network
QoS	Quality of Service
SCN	Switched Circuit Network
TE	Terminal Equipment

4 Parameters

In this clause a set of parameters and values is proposed for consideration for the specification of the call completion performance classification.

4.1 Signalling failure probability

4.1.1 Call Setup Failure Probability

The Call Setup Failure Probability (CSFP) is the ratio of total call setup attempts that result in call setup failure to the total call setup attempts in a population of interest.

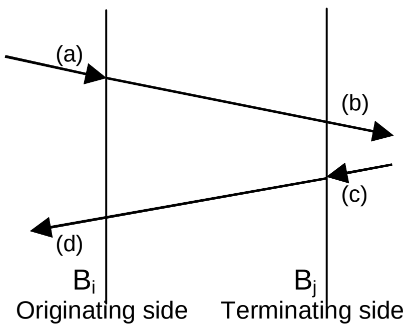

Figure 1: Reference events occurring during successful call setup

With reference to figure 1, the CSFP is defined to occur on any call setup attempt in which either one of the following outcomes is observed prior to expiration of timer T_{max}:

- both events (b) and (d) do not occur;
- events (b) and (c) occur, but event (d) does not.

Events (a) to (d) are observable events at specific reference points:

(a) illustrates setup at originating side;

(b) illustrates setup at terminating side;

(c) illustrates alerting at terminating side;

(d) illustrates alerting at originating side.

The reference points for TIPHON Release 4, corresponding to reference points B_i and B_j in figure 1 are specified in TS 101 314 [1]. The events occurring at those reference points correspond to specific TIPHON Metaprotocol messages as defined in TS 101 882 [2].

In case of TIPHON Scenario 0, reference point B_i in figure 1 relates to reference point C1 at the originating side, while B_j relates to reference point C1 at the terminating side. Reference point B_j relates to TIPHON reference point C3 in case of TIPHON scenario 1. B_i relates to TIPHON reference point C3 in case of TIPHON scenario 2.

The relationship between events (a) to (d) and the TIPHON Metaprotocol messages are as follows:

(a) corresponds to the entry of *U_CallRequest* at reference point C1 between the originating terminal device and the SpoA for call control;

(b) corresponds to the exit of *D_CallRequest* at reference point C1 between the terminating SpoA and the terminating terminal device;

(c) corresponds to the entry of U_*CallAltert* at reference point C1 between the terminating terminal device and the terminating SpoA;

(d) corresponds to the exit of *D_CallReport* at reference point C1 between the originating SpoA and the originating terminal device.

NOTE 1: The exact value of T_{max} is for further study.

NOTE 2: Call setup attempts that are cleared by the network portion as a result of incorrect performance or non-performance on the part of an entity outside the network portion are excluded.

NOTE 3: A call setup attempt can fail as a result of user blocking. Such failures are excluded from network performance measurement. Examples of user blocking include the following:

- the called user issues a message to reject the call setup attempt;
- the connect message reference event fails to occur at the originating MPT boundary due to the lack of a connect message reference event at the terminating MPT boundary;
- the called user delays excessively in generating the connect message reference event during the call period, with the result that a call is not established before the time-out;
- all channels at the called TE are in use.

Objective: The Call Setup Failure Probability as defined above for a TIPHON network shall not exceed 3×10^{-5}.

NOTE 4: This is a provisional value. The actual target value is for further study.

4.1.2 Call setup error probability

Call setup error probability is the ratio of total call setup attempts that result in call setup error to the total call setup attempts in a population of interest.

With reference to figure 1, a call setup error is defined to occur on any call setup attempt in which event (d) occurs, but event (c) does not occur at an appropriate boundary prior to expiration of timer T.

Call setup error is essentially the case of a network-caused "wrong number". It occurs when the network responds to a valid call request by erroneously establishing a call to a destination Terminal Equipment (TE) other than the one designated in the call request, and does not correct the error prior to the user information transfer. It may be caused, for example, by network operator administrative or maintenance actions.

Call setup error is distinguished from successful call setup by the fact that the intended called user is not contacted and not committed to the session during the call setup attempt.

4.1.3 Signalling route unavailability

Important remark: The signalling route unavailability parameter is only applicable to the SCN network portions of TIPHON systems. Hence it is not an End-to-end parameter.

The availability of a signalling route set is determined by the availability of the individual components of the signalling network and by the structure of the signalling network. (i.e. the arrangements for redundancy and alternative routing in the event of the unavailability of a component.)

ITU-T recommendation [6] Q.706, clause 1.1, recommends that the availability of a signalling route set should be not less than 0,99998. This corresponds to a downtime of not more than ten minutes per year.

4.1.4 Call Premature Release Probability

The Call Premature Release Probability (CPRP) for a network portion is the probability, in any given second, that a call experiences a premature disconnect generated within that portion.

A premature disconnect is defined as an occurrence of a *Disconnect request* or a *Release* message [2] exiting the network portion at its originating or terminating network boundary,

- either in the absence of a previous corresponding *Modify* or *Release* message occurring at the other boundary of the network;
- or when a premature disconnect stimulus occurs within the network boundaries and is transferred across its originating or terminating boundaries.

The network portion boundaries are the same (B_i, B_j) as shown in figure 1. They also correspond to the reference point between FE5 and FE6 for the originating side of a TIPHON network and the reference point between FE8 and FE9 for the terminating side.

Objective: The Call Premature Disconnect Probability (CPDP) as defined in this clause should not exceed 1,5 x 10^{-5}.

NOTE: This is a provisional value. The actual target value is for further study.

4.1.5 Call Release Failure Probability

The Call Release Failure Probability (CRFP) is the ratio of total call clearing failures to the total call clearing attempts in a population of interest.

A call clearing failure is defined with reference to events at the originating and terminating boundaries of a network portion (B_i and B_j respectively as indicated in figure 1).

A call clearing attempt occurs when a *U_CallClear* message [2] enters the network portion at the originating reference point B_i. A call clearing failure occurs when no corresponding *D_CallClear* event [2] occurs at the terminating reference point B_j within $\mathbf{T}_{ccf}$ seconds.

Objective: For CCFP: for further study. In clause 2.5.1.2 of [3] a value of 2 out of 10^5 is specified.

NOTE: The value of T_{ccf} is for further study.

4.1.6 Percentage of unsuccessful call attempts

The percentage of unsuccessful call attempts is the ratio of the number of unsuccessful call attempts divided by the total number of call attempts in a population of interest. It is the sum of 2 elements as defined above:

- CSFP (Call Setup Failure Probability);
- CSEP (Call Setup Error Probability).

4.2 Call signalling delays

The call setup time depends on a number of elements:

- The number of times messages must traverse the signalling path between the calling party (or its local exchange, depending on the access technology used) and the called party or the called party's local exchange.
- The propagation time within the signalling path.
- The processing delay for signalling messages within each routing point within the signalling domain.
- The delay involved in accessing database information, for instance in number portability scenarios.

NOTE: Although mandatory objectives can be set by national regulators according to the ONP Telephony Directive 98/10/EC [8], no regulations specifying call setup times have been identified.

4.2.1 Call setup delay

The Call Setup Delay (CSD) is the time between the calling terminal providing sufficient address information to set up the call, and the calling party receiving a confirmation from the called terminal that the called party is being alerted.

Objective:

Draft Recommendation Y.1530 [3] defines the following objectives for the CSD:

Table 1: Call setup delay

Statistic	Objective
Mean	7 500 ms
95 % ile	8 450 ms
NOTE: Provisional values; the actual target values are for further study.	

4.2.2 Call Answer Signal Delay

The Call Answer Signal Delay (CASD) is the time between the called terminal indicating that it is ready to initiate the call and receipt of that indication by the calling terminal.

Table 2: Call Answer Signal Delay objectives

Statistic	Objective
Mean	FFS
95 % ile	FFS
NOTE: Provisional values; the actual target values are for further study.	