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Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 4; End-to-end Quality of Service in TIPHON Systems; Part 1: General aspects of Quality of Service (QoS)

Reference

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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 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";

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";

Part 8: Void;

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

Part 10: To be published as part of release 5 (see note) "QoS Requirements for TIPHON Terminals";

Part 11: To be published as part of release 5 (see note) "Domain by domain performance planning guidelines for end-to-end QoS objectives associated with TIPHON speech QoS classes";

TS 102 024-12: "IP Telephony Service Availability".

NOTE: Quality of Service aspects of TIPHON Release 5 systems will be covered in TS 102 025 (see bibliography), and more comprehensive versions of the Release 4 documents listed above will be published as part of Release 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.

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 101 314: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 4; Abstract Architecture and Reference Points Definition; Network Architecture and Reference Points".

[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] ETSI TS 102 024-5: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 4; End-to-end Quality of Service in TIPHON Systems; Part 5: Quality of Service (QoS) measurement methodologies".

[5] ETSI TR 102 024-7: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 4; 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 point of view".

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

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

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

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

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

[11] 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:

best effort QoS: situation in which the end to end performance of the media stream is not defined by a service level specification

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.

grade of service: Number of traffic engineering variables used to provide a measure of adequacy of a group of resources under specified conditions. These grade of service variables may be probability of loss, dial tone delay, etc.

NOTE 1: The parameter values assigned as objectives for grade of service variables are called grade of service standards.

NOTE 2: The values of grade of service parameters achieved under actual conditions are called grade of service results (see ITU-T Recommendation E.600 [6]).

guaranteed QoS: situation where the end-to-end performance of the media stream, for the duration of a session, is designed to meet the requirements laid down in a service level specification

NOTE: The service level specification may include details of the percentage of the time when quality may fall below specified levels as well as the distribution and duration of the intervals when this occurs.

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 [6] 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 [8]). For calculation purposes the use of traditional telephone handsets (see ITU-T Recommendation P.310 [11]) 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 102 024-5 [4], while the requirements for these parameters with respect to the various TIPHON QoS classes are defined in TS 102 024-2 [3].

TR 102 024-7 [5] 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 [10].

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 102 024-7 [5] provides guidance on these factors with respect to the practical design phase of equipment and networks.

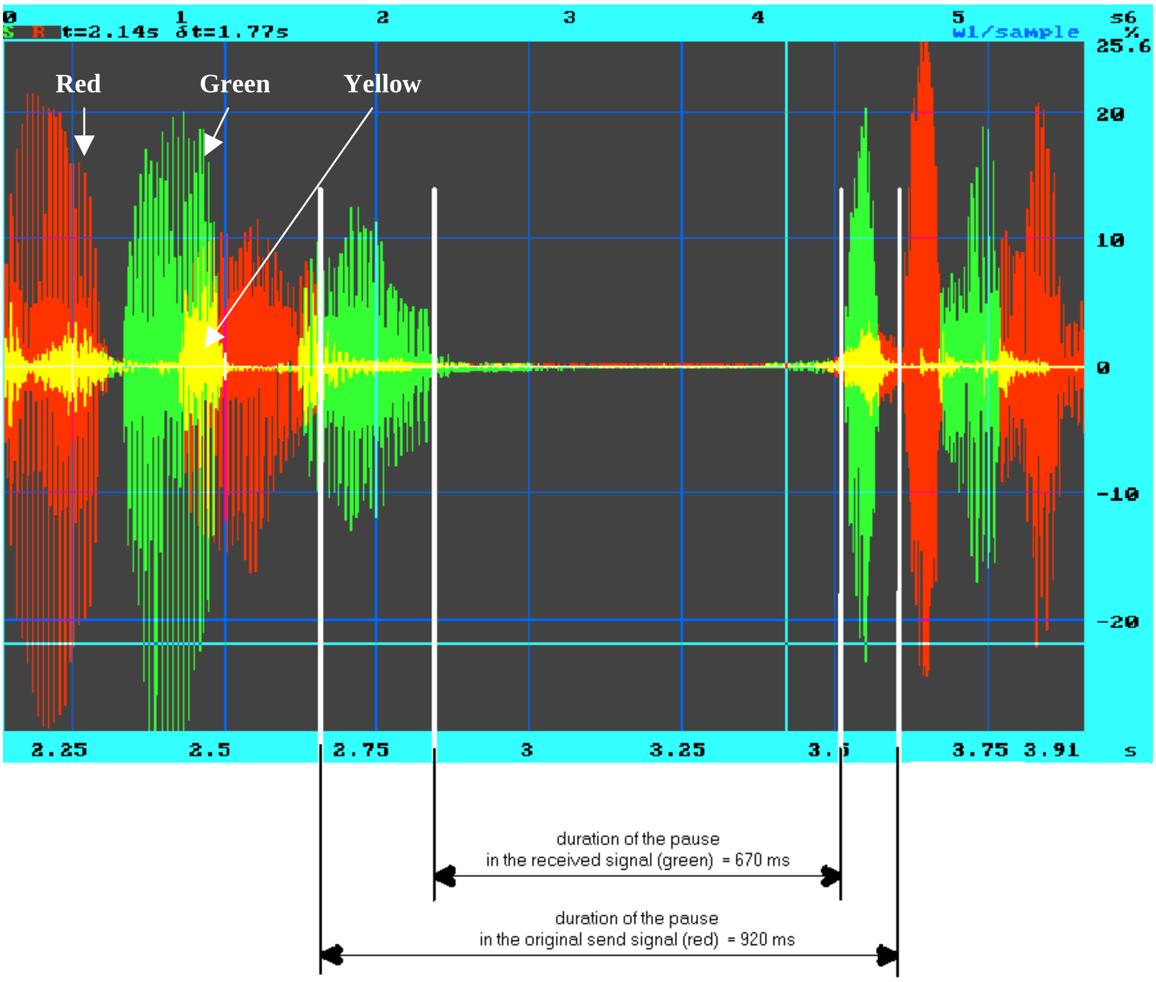
4.3.1 Delay Jitter

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.

Figure 1: Void

Figure 2 shows some measurement result regarding delay jitter:



NOTE 1: Red = Original send signal; Green = receive signal (transmitted via the test set-up).

NOTE 2: Due to the selected measurement mode the original send signal (red) is displayed 500 ms later and reduced by 6 dB in level than in reality. The resulting mean one-way delay lies between 450 ms and 700 ms and is strongly variable.

Figure 2: Delay jitter observation

Figure 2 shows the observed delay jitter. In the original speech sample (red signal) the pause sequence between the two words has approx. 920 ms duration; after having sent this speech sample over the test set-up the received speech (green signal) has a pause sequence of only 670 ms duration; i.e. the jitter amplitude is 250 ms.

4.3.2 Jitter buffers

Jitter buffers are "devices" implemented in terminal equipment or software at the receiving end of a voice connection on a packet switched network. Their function is to ensure that voice packets are delivered at regular intervals to the voice decoder, even if packets experience variation in the time they are underway in the network. In the context of IP networks, this variation of time, or rather variation of delay, is known as "jitter".

For static dejittering mechanisms dejittering delay [s], is the time between the arrival instant of the first packet and the time instant the play out (of voice/video/etc.) is started.

Jitter buffer size [byte], is the physical size of the dejittering buffer at the receiver.

4.3.3 Jitter Buffer Implementations

A jitter buffer is used to compensate for delay variation ("jitter") in a stream of incoming packets, in our case voice packets, at the receiving end of a "connection", see figures 3 and 4.

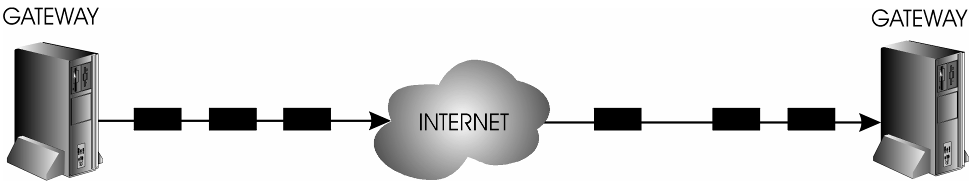


Figure 3: Jitter

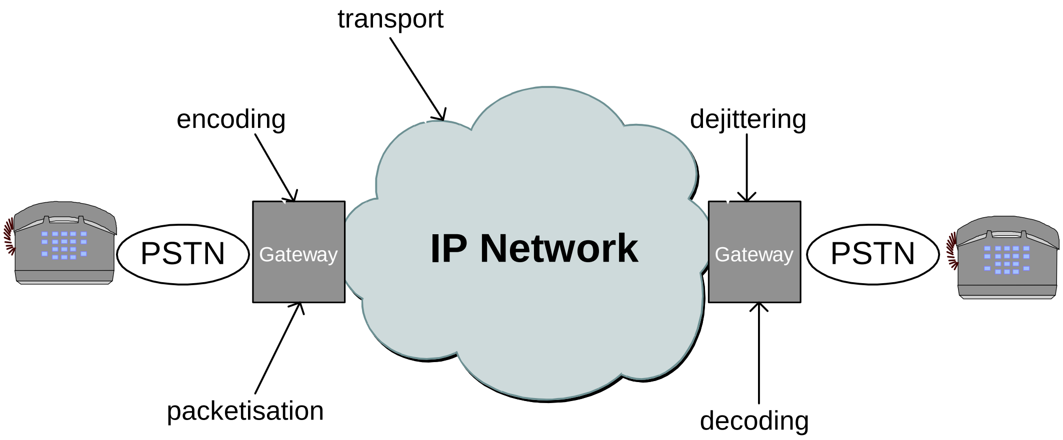


Figure 4: Jitter Buffering (dejittering)

Jitter is almost inevitable in a network with multiple data streams, even if these send packets at a regular rate. Without a jitter buffer at the receiving end, jitter would lead to starvation, i.e. no packets to process, together with loss due to too many packets arriving later on and sometimes packets arriving too late to be useful. The result on a speech connection will be a lower quality.

This text describes how jitter buffers work based on information found on the Internet and in literature. The goal of the description is to be useful for modelling a jitter buffer in the determination of its contribution to end-to-end speech quality in relation to its size.

The size of a jitter buffer (measured in time, i.e. milliseconds) is an important factor in its success or failure. If the jitter buffer is too large, unnecessary delay is introduced. If it is too small, extra loss is introduced. In both cases, voice quality may be reduced.

Two types of jitter buffers exist:

1) Static. The size of a static jitter buffer is configured once, or fixed in hardware by the manufacturer.

2) Dynamic. The size of a dynamic jitter buffer is adapted constantly by software in the receiving end to accommodate for changes in the network delay.

Some software implementations can be configured to be either static or dynamic.

4.3.4 Jitter buffer monitoring capabilities

Quantifying jitter can be done in two ways: The statistical variance of interarrival time, or the jitter amplitude. The variance is more meaningful than the amplitude, because the amplitude is only determined by the two extremes.

A packet switched network introduces not only loss and delay, but also delay variation, also called jitter. This can be very disturbing on the speech quality of a voice connection. To compensate, a receiving terminal (see note) often uses a Jitter Buffer. This can be either a static or a dynamic Jitter Buffer, with a variety of algorithms.

NOTE: A terminal can be a gateway, hardware phone, Softphone, etc.

4.4 Physical components based on which a TIPHON service may be provided

The following components may be present in a TIPHON system and may each contribute to the overall end-to-end QoS performance of the system:

- IP terminal;
- IP access network;
- IP backbone;
- IWF (gateway/gatekeeper);
- SCN;
- voice terminal(s) connected to the SCN(s).

4.4.1 IP terminal

For the purposes of the TIPHON QoS documentation only those IP terminals will be considered to which the following description of a TIPHON terminal applies.

TIPHON terminal:

A terminal that is either dedicated (e.g. a telephone set) or general purpose (e.g. a computer running an application that performs the terminal function) and that:

- is intended for connection to an IP-network;
- provides the functionality defined in TS 101 314 [2]; and
- is capable of supporting at least one of the TIPHON speech quality of service classes defined in TS 102 024-2 [3].

Examples of different terminal types which may serve as IP terminals are:

- 4-wire telephones;
- cordless telephones;
- a PC with a headset and appropriate VoIP software.

Examples of operational modes for these terminal types are:

- with traditional handset;
- with so-called modern handsets;
- with loudspeaking function;
- totally hands-free.

The way in which each of these techniques is implemented has implications for end-to-end Quality of Service.

4.4.2 IP access network

A variety of access network transport media may be used to interconnect TIPHON IP terminals with IP backbone networks. EG 202 306 [1] provides guidelines.

Examples of methods that can be used for IP access layer transport are:

- LAN Access;
- PSTN Access;
- xDSL Access;
- Cable Modem Access;
- BRAN Access;
- DECT Access;
- UMTS Access;
- ISDN Access;
- GSM Access.

The way in which each of these techniques is implemented has implications for end-to-end Quality of Service.

4.4.3 IP backbone

A variety of equipment may be used to provide IP backbone networks for TIPHON services.

Examples of equipment that can be used for IP backbones are:

- Routers;
- High-speed facilities.

The way in which each of this equipment is implemented has implications for end-to-end Quality of Service.

4.4.4 IWF (gateway/gatekeeper)

Factors affecting QoS in the Gateway mirror those in the IP terminal.

Interworking functions (IWFs) may be realized in various ways:

- local;
- distributed.

The way in which each of these techniques is implemented has implications for end-to-end Quality of Service.

4.4.5 SCN

A variety of various branches of the SCN follow a multitude of different national or international regulations or standards.

Examples of different types of branches of the SCN are:

- analogue networks;
- digital networks;
- cordless networks;
- mobile networks.

Examples of the different technologies on which those networks may be based are:

- analogue lines;
- FDM systems;
- digital lines;
- TDM systems;
- optical Fiber Systems;
- wireless Systems.

Examples of the different status which those networks may have are:

- international sections between national networks;
- public national networks, accessible for the general public;
- private networks (e.g. Corporate Networks), accessible for closed user groups only.

The way in which each of these techniques is implemented has implications for end-to-end Quality of Service (QoS).

4.4.6 Voice terminal connected to the SCN

A variety of voice telephony terminals interconnected to various branches of the SCN and follow a multitude of different national or international regulations or standards.

Examples of different terminal types interconnected to SCN are:

- 2-wire analogue telephones;
- 4-wire analogue telephones;
- digital telephones;
- cordless telephones;
- mobile telephones.

Examples of operational modes for these terminal types are:

- with traditional handset;
- with so-called modern handsets;
- with loudspeaking function;
- totally hands-free.

The way in which each of these techniques is implemented has implications for end-to-end Quality of Service.

5 Reference connections

For general transmission planning purposes ITU-T Recommendation G.103 [7] provides guidance and definitions with regard to reference connections.

For the purposes of TIPHON however, the recommendations provided by G.103 [7] are not sufficient or may not be applicable. Therefore, the present document gives additional information on the reference connection for all TIPHON scenarios.

The terms of reference of the TIPHON project set out five scenarios for interoperability between IP telephony systems and Switched Communication Networks (SCN).

The maximum geographical length of reference connections for each of the TIPHON scenarios is not defined.

Table 1 relates the reference connections for the five TIPHON scenarios to the connection arrangements defined in ITU-T Recommendation G.177 [9].

Table 1: Comparison of TIPHON Scenarios with G.177 [9]

TIPHON Scenario #	G.177 [9] connection arrangement
0	(see note)
1	(1)
2	(2)
3	(3)
4	(4)
NOTE: TIPHON scenario #0 is not covered by ITU-T Recommendation G.177 [9]; however work in this area is underway in ITU-T Study Group 12, Question, 23/12 during the Study Period 1997 to 2000.	

Figure 10 illustrates a reference model for a single transport domain.

Figure 11 depicts a generic refence connection which can be applied to all TIPHON scenarios.

5.1 Reference connection for TIPHON scenario #0

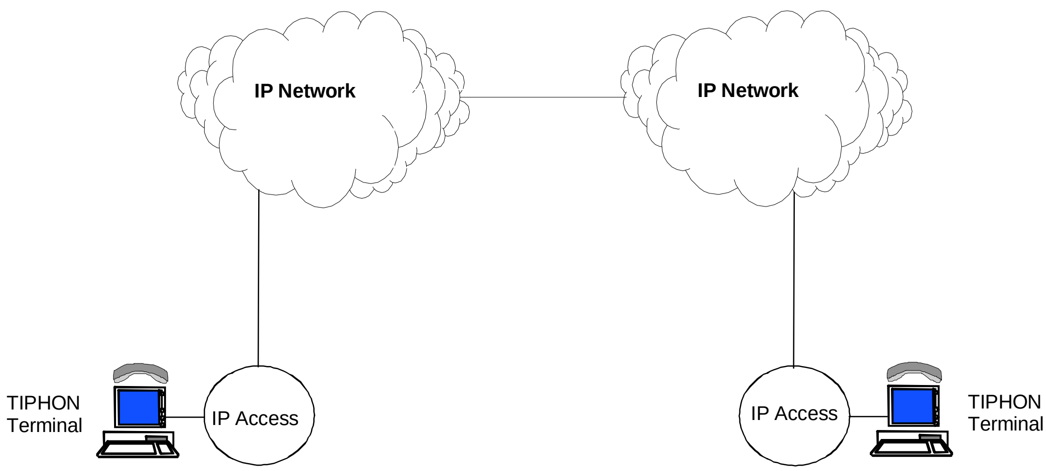


Figure 5: Scenario 0 - IP network to IP network

5.2 Reference connection for TIPHON scenario #1

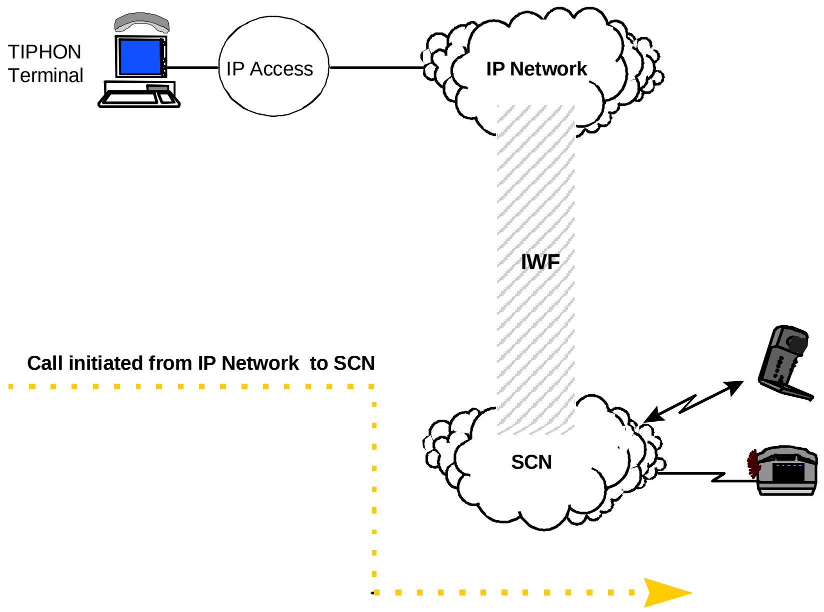


Figure 6: Scenario 1 - Call from IP Network to SCN

5.3 Reference connection for TIPHON scenario #2

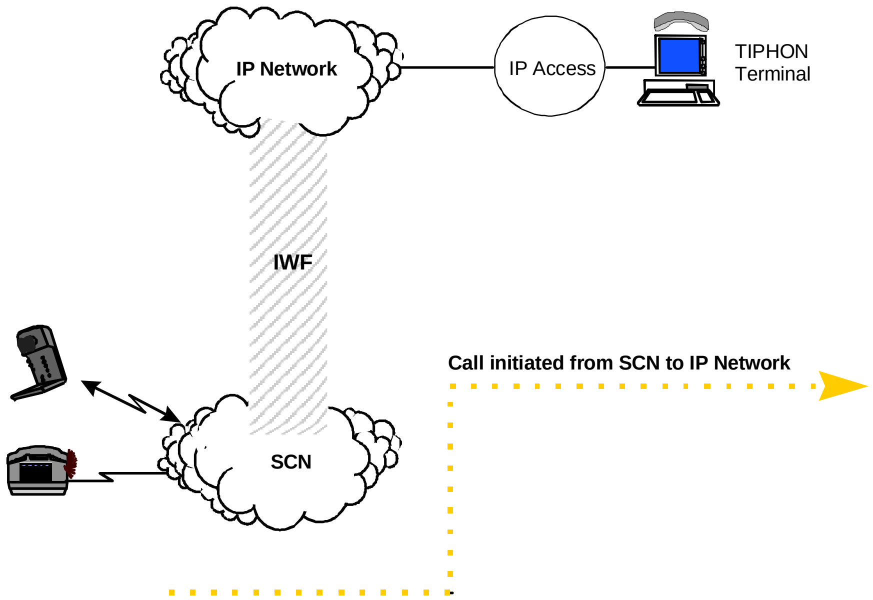


Figure 7: Scenario 2 - Call from SCN to IP Network

5.4 Reference connection for TIPHON scenario #3

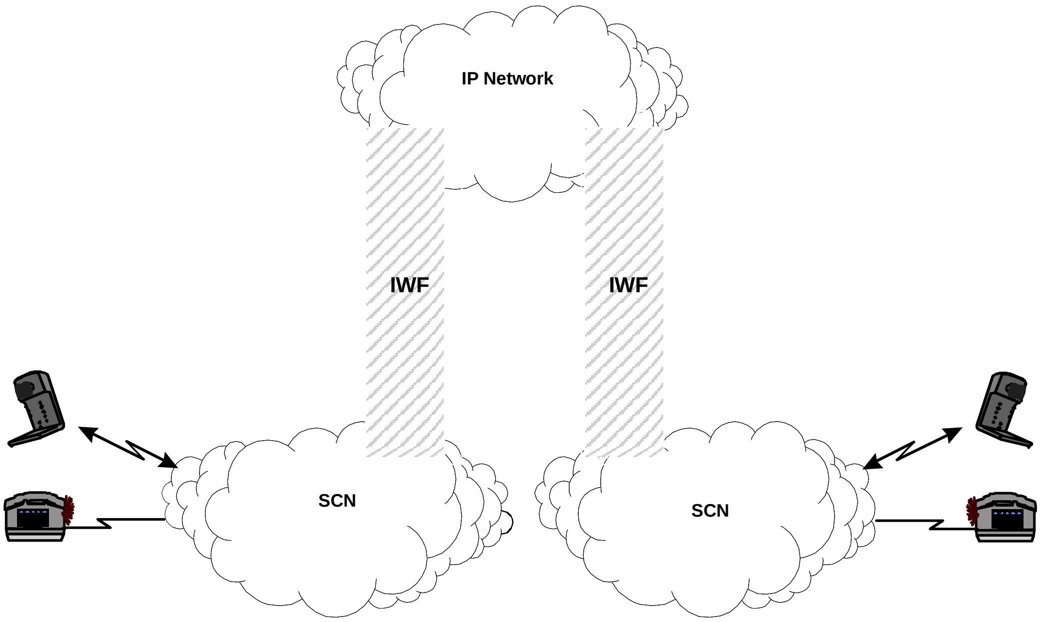


Figure 8: Scenario 3 - SCN to SCN over IP network

5.5 Reference connection for TIPHON scenario #4

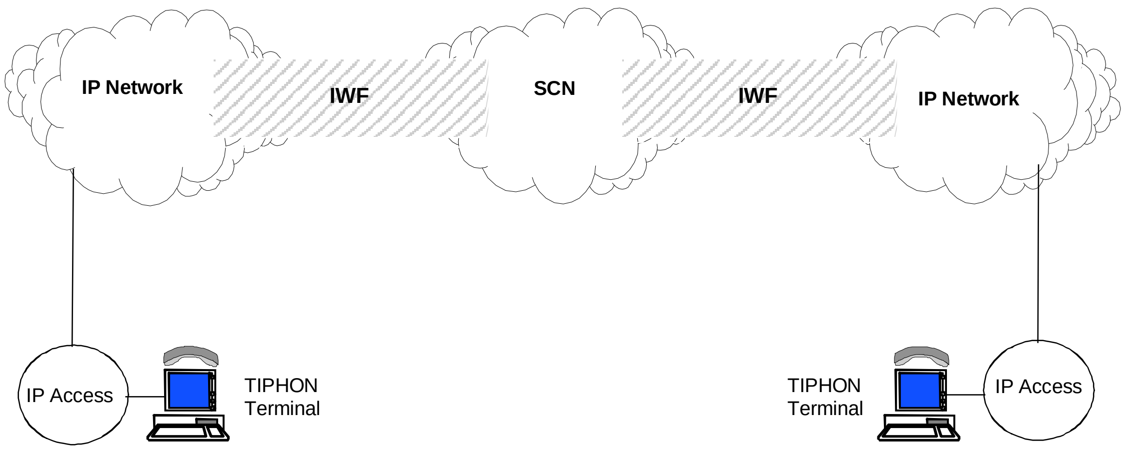


Figure 9: Scenario 4 - IP network to IP network over SCN

5.6 Reference model for a single transport domain

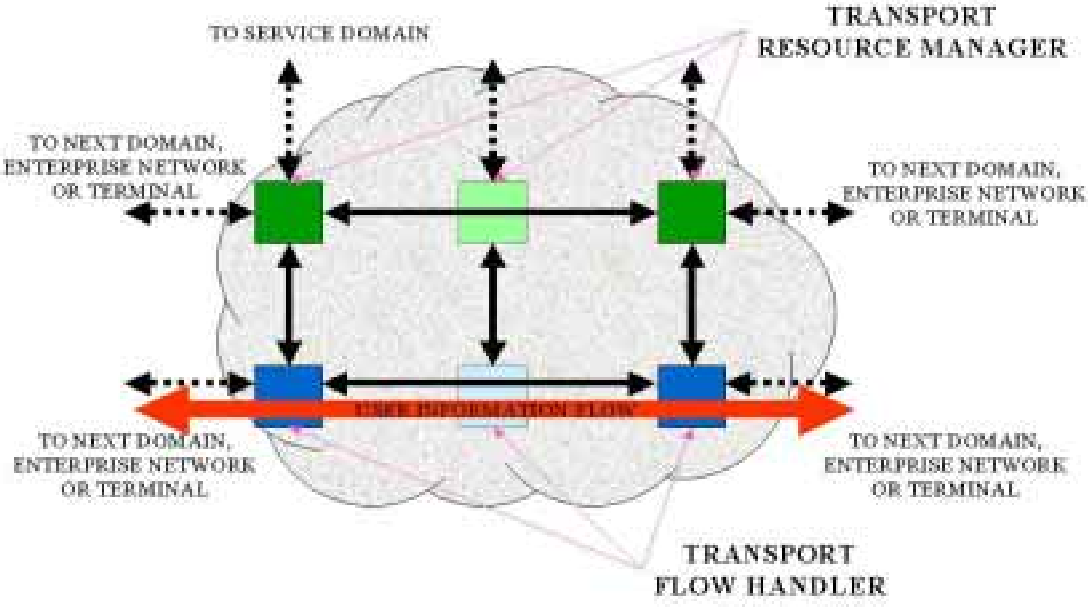


Figure 10: Reference model for a single transport domain

5.7 Generic reference connection for TIPHON scenarios

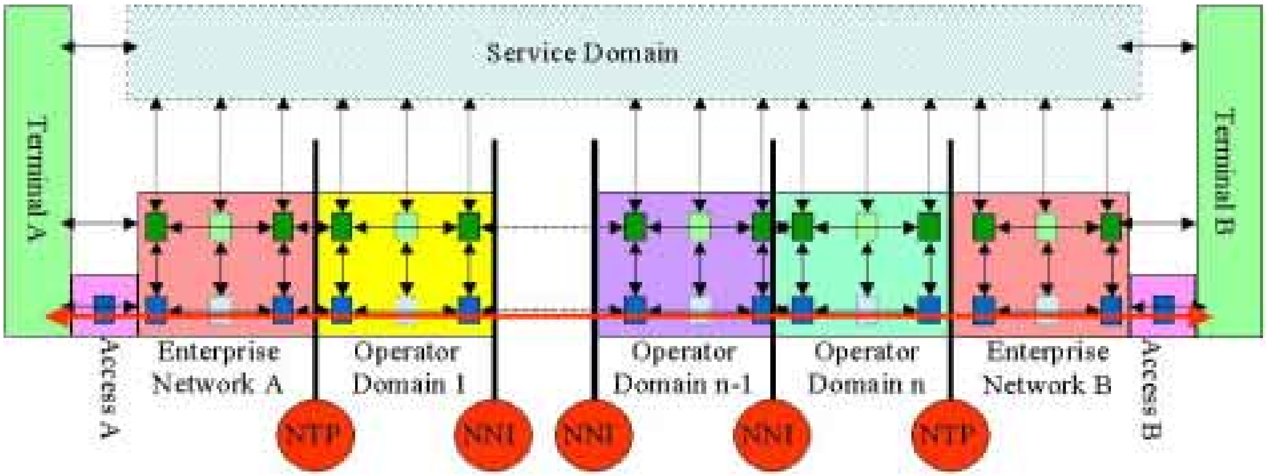


Figure 11: Generic reference connection for TIPHON scenarios

Annex A (informative): Bibliography

ETSI TS 102 025 series: TIPHON Release 5 QoS documents.

History

Document history		
V1.2.5	October 1998	Publication as TR 101 329 (Historical)
V2.1.1	June 1999	Publication as TR 101 329 (Historical)
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V3.1.2	January 2002	Publication as TR 101 329-1
V4.1.1	September 2003	Publication