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SERIES Y: GLOBAL INFORMATION INFRASTRUCTURE, INTERNET PROTOCOL ASPECTS AND NEXT GENERATION NETWORKS

Internet protocol aspects – Quality of service and network performance

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Call processing performance for voice service in hybrid IP networks

ITU-T Recommendation Y.1530

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ITU-T Recommendation Y.1530

Call processing performance for voice service in hybrid IP networks

Summary

This Recommendation defines performance parameters and objectives for point-to-point call processing in voice service provided by Hybrid IP networks. The parameter definitions are based on the principles and generic performance parameters defined in ITU-T Rec. I.350, and make use of relevant definitions from ISDN call processing performance Recommendations where appropriate.

Source

ITU-T Recommendation Y.1530 was approved on 7 May 2004 by ITU-T Study Group 13 (2001-2004) under the ITU-T Recommendation A.8 procedure.

FOREWORD

The International Telecommunication Union (ITU) is the United Nations specialized agency in the field of telecommunications. The ITU Telecommunication Standardization Sector (ITU-T) is a permanent organ of ITU. ITU-T is responsible for studying technical, operating and tariff questions and issuing Recommendations on them with a view to standardizing telecommunications on a worldwide basis.

The World Telecommunication Standardization Assembly (WTSA), which meets every four years, establishes the topics for study by the ITU-T study groups which, in turn, produce Recommendations on these topics.

The approval of ITU-T Recommendations is covered by the procedure laid down in WTSA Resolution 1.

In some areas of information technology which fall within ITU-T's purview, the necessary standards are prepared on a collaborative basis with ISO and IEC.

NOTE

In this Recommendation, the expression "Administration" is used for conciseness to indicate both a telecommunication administration and a recognized operating agency.

Compliance with this Recommendation is voluntary. However, the Recommendation may contain certain mandatory provisions (to ensure e.g. interoperability or applicability) and compliance with the Recommendation is achieved when all of these mandatory provisions are met. The words "shall" or some other obligatory language such as "must" and the negative equivalents are used to express requirements. The use of such words does not suggest that compliance with the Recommendation is required of any party.

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As of the date of approval of this Recommendation, ITU had not received notice of intellectual property, protected by patents, which may be required to implement this Recommendation. However, implementors are cautioned that this may not represent the latest information and are therefore strongly urged to consult the TSB patent database.

CONTENTS

ITU-T Recommendation Y.1530

Call processing performance for voice service in hybrid IP networks

1 Scope

This Recommendation defines performance parameters and objectives for point-to-point call processing in voice service provided by Hybrid IP networks. Call processing delay parameters for ISDN are defined in ITU-T Rec. I.352. Call processing accuracy and dependability parameters for ISDN are defined in ITU-T Rec. I.359. The parameters defined in this Recommendation are general in that no particular Hybrid IP network call processing protocol is assumed. The protocols used at different signalling interfaces traversed in processing a particular call may differ, and may include (for example) SIP, H.323, and BICC. Call processing protocol events defined in ISDN Recommendations are used for illustration in some cases. Call processing performance measurement architectures are presented in Appendix II.

The parameters defined in this Recommendation are applicable only when the network is available. The availability of Hybrid IP networks will be addressed in a separate Recommendation.

Table 1 illustrates the application of the three generic performance criteria defined in ITU-T Rec. I.350 to the call processing functions defined for Hybrid IP networks.

Table 1/Y.1530 – Performance parameters for call processing functions

Call processing function	Speed	Accuracy	Dependability
1) Connection set-up	Connection set-up delay Connection post selection delay (Note 1) Connection answer signal delay (Note 1)	Connection set-up error probability	Connection set-up failure probability
2) Connection disengagement	Connection disconnect delay Connection release delay (Note 2)	Connection premature disconnect probability	Connection clearing failure probability
NOTE 1 – These parameters are defined for traffic engineering; objectives are not set for network performance. NOTE 2 – This parameter has local significance only; objectives will not be set.			

2 References

The following ITU-T Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published. The reference to a document within this Recommendation does not give it, as a stand-alone document, the status of a Recommendation.

[1] ITU-T Recommendation I.350 (1993), *General aspects of quality of service and network performance in digital networks, including ISDNs*.

[2] ITU-T Recommendation I.352 (1993), *Network performance objectives for connection processing delays in an ISDN*.

[3] ITU-T Recommendation I.359 (1999), *Accuracy and dependability of ISDN 64 kbit/s circuit-mode connection types*.

[4] ITU-T Recommendation Q.931 (1998), *ISDN user-network interface layer3 specification for basic call control*.

[5] ITU-T Recommendation Q.764 (1999), *Signalling System No. 7 – ISDN user part signalling procedures*.

[6] ITU-T Recommendation Y.1540 (2002), *Internet protocol data communication service – IP packet transfer and availability performance parameters*.

[7] ITU-T Recommendation E.671 (2000), *Post-selection delay in PSTN/ISDN networks using Internet telephony for a portion of the connection*.

[8] ITU-T Recommendation H.323 (2003), *Packet-based multimedia communications systems*.

[9] ITU-T Recommendation H.225.0 (2003), *Call signalling protocols and media stream packetization for packet-based multimedia communication systems*.

[10] ITU-T Recommendation H.245 (2003), *Control protocol for multimedia communication*.

[11] ITU-T Recommendation H.450.1 (1998), *Generic functional protocol for the support of supplementary services in H.323*.

3 Definitions

This Recommendation defines the following terms:

3.1 end-to-end call processing performance: Network performance for call processing from UNI to UNI.

3.2 hybrid IP networks: Networks in which end-to-end services are provided through interworking between IP-based networks and networks employing other technologies, e.g., PSTN, ISDN, xDSL and CATV.

4 Abbreviations

This Recommendation uses the following abbreviations:

ACM Address Complete Message

ANM Answer Message

BICC Bearer Independent Call Control

CASD Connection Answer Signal Delay

CCFP Connection Clearing Failure Probability

CDD Connection Disconnect Delay

CPDP Connection Premature Disconnect Probability

CPDS Connection Premature Disconnect Stimulus

CPE Customer Premises Equipment

CPSD Connection Post Selection Delay

CRD Connection Release Delay

CSD Connection Set-up Delay

CSEP Connection Set-up Error Probability

CSFP	Connection Set-up Failure Probability
FFS	For Further Study
HRX	Hypothetical Reference Connection
IAM	Initial Address Message
ISDN	Integrated Services Digital Network
ITU-T	International Telecommunication Union – Telecommunication Standardization Sector
IWF	InterWorking Function
MP	Measurement Point
MPT	Measurement Point Terminal
RE	Reference Event
REL	Release
SIP	Session Initiation Protocol
UNI	User-Network Interface
VoIP	Voice over Internet Protocol

5 Reference model

Within the context of this Recommendation, a reference model is intended to establish the context for the definition of performance parameters and the specification of objectives. This reference model provides a baseline reference connection that includes measurement points at which performance-significant reference events are observed and values for call processing performance parameters are determined, and a specification of the traffic loading conditions under which objectives are to be met. This reference model consists of a reference configuration, a set of performance-significant reference events, and time intervals for establishing the conditions under which objectives are to be met.

5.1 Reference configuration

The general reference configuration used in describing the call processing performance of voice service provided by Hybrid IP networks is illustrated in Figure 1.

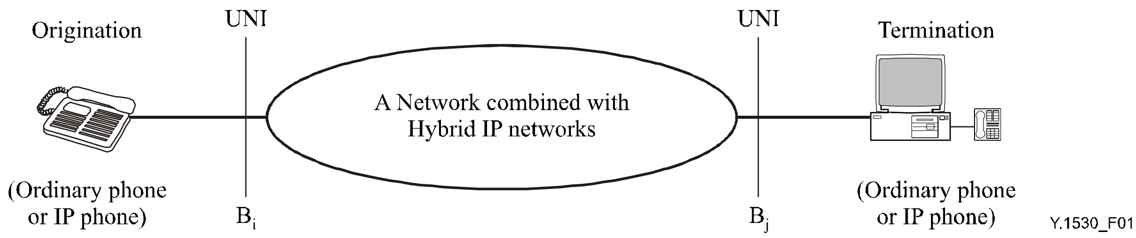


Figure 1/Y.1530 – General reference configuration

A reference configuration for ISDN/IP networking under TIPHON scenario 2 is described in Appendix I. A reference architecture for BICC is illustrated in Appendix III.

5.2 Reference events

The reference events in ISDN can be uniquely specified by ITU-T Recs I.352 and I.359 based on signalling protocols in ITU-T Recs Q.931 and Q.764, but the call processing reference events in IP networks depend on the method used for IP telephony, such as H.323, BICC and SIP.

Therefore, the general model of reference events in this Recommendation should be based on a model composed of pairs of portion boundaries (B_i, B_j) in Figure 1 and exit and entry events at (B_i, B_j) uniquely.

5.3 Performance objectives for delay parameters

End-to-end delay parameters are defined using reference events, therefore, performance objectives applied to this set of parameters reflect the Quality of Service provided by a provider's network.

The performance objectives for the set of delay parameters can be expressed in terms of:

- mean delay; and
- 95%-tile delay.

However, this approach does not take into account the impact of traffic load on call processing performance parameters (i.e., speed, dependability and accuracy). An alternative worst-case objective is addressed in the next subclause. Assumptions regarding the call set-up attempt rate are stated in Annex A.

5.4 Alternate specification of objectives

In general, while the value taken by a particular call processing performance parameter is sensitive to the traffic load that exists on each portion of the network through which a call is being processed, the worst-case objectives specified in this Recommendation are meant to apply to any one-hour of time.

NOTE – To allow for the impacts of random, short-term traffic load fluctuations during any hour, the probability that a particular call processing performance parameter value, V, fails to meet its objective, O, may be specified. That is:

$$\text{Prob}\ \{V > O\} \le X$$

where:

V = actual parameter value

O = objective

The exact conditions or assumptions under which these worst-case objectives may apply are for further study. Use of an hour or possibly 15-minute interval of time would facilitate support by existing operations systems. It is recognized that the objective would be developed taking into consideration peak traffic factors such as "busy hour".

6 Call processing delays

The model for determining call processing delays (see Figure 2) allows for single portion and between portion definitions using the reference events. Parameters involving round trip delays (e.g., connection establishment) are defined by portion boundaries (i.e., B_i and B_j below). Unidirectional delay parameters (e.g., connection disconnect delay) are defined only between two portion boundaries. This approach ensures that performance is attributable to specific portions.

For each parameter, performance-significant Reference Event (RE) tables are provided containing those REs used in the parameter's definition. These REs are listed by relevant boundary. The boundaries are described according to Figure 1.

NOTE – The model and parameter definition regarding call processing delays for SIP and stream control transmission protocol (SCTP) is FFS.

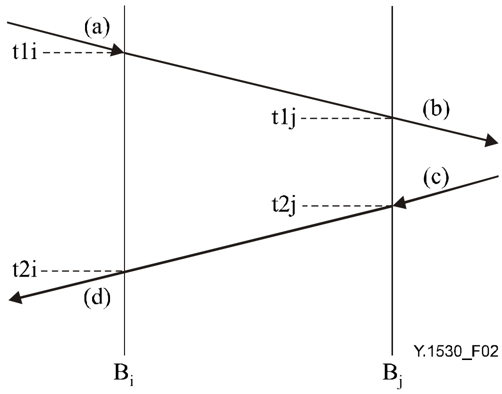


Figure 2/Y.1530 – Call processing delays

6.1 Call establishment delays

The boundaries (B_i and B_j) from Figure 3 for call establishment delays are measurement points. Figure 3 illustrates aspects of call establishment processing that are observable at measurement points. (a) illustrates set-up at origination and (b) illustrates set-up at termination. (c) illustrates alerting at termination and (d) illustrates alerting at origination. (e) illustrates connect at termination and (f) illustrates connect at origination.

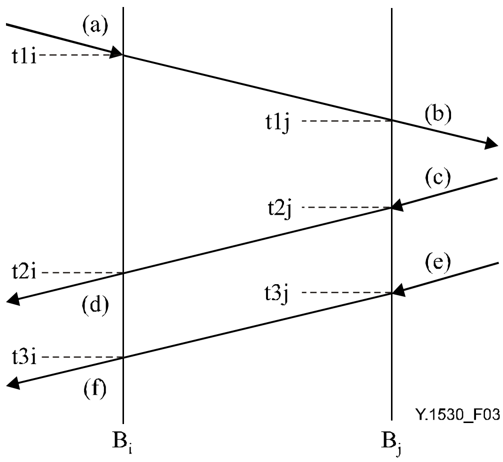


Figure 3/Y.1530 – Call establishment delays

6.1.1 Connection set-up delay (CSD)

Using the notation in Figure 3, the following difference is formed for end-to-end connection set-up delay:

$$CSD = (t3i - t1i) - (t3j - t1j)$$

The worst-case delay objective for CSD, applying to any one-hour period, may be specified as:

Table 2/Y.1530 – Connection Set-up Delay objectives*

Statistic	Objective
Mean	7500 ms
95% – ile	8450 ms

6.1.2 Connection post selection delay (CPSD)

Using the notation in Figure 3, the following difference is formed for connection post selection delay:

$$CPSD = (t2i - t1i) - (t2j - t1j)$$

The worst-case delay objective for CPSD, applying to any one-hour period, may be specified as:

Table 3/Y.1530 – Connection Post Selection Delay objectives*

Statistic	Objective
Mean	7500 ms
95% – ile[a)]	8450 ms

6.1.3 Connection answer signal delay (CASD)

Using the notation in Figure 3, the following difference is formed for end-to-end connection answer signal delay:

$$CASD = t3i - t3j$$

This double ended measure is optional.

The worst-case delay objective for CASD, applying to any one-hour period, may be specified as:

Table 4/Y.1530 – Connection Answer Signal Delay objectives*

Statistic	Objective
Mean	FFS
95% – ile[a)]	FFS

6.2 Call disengagement delays

Figure 4 illustrates aspects of call disconnect processing that are observable at the measurement points.

The boundaries (B_i and B_j) from Figure 4 for call disconnect delays are measurement points. (a) illustrates release at user i (origination). (b) illustrates clearing at user i (origination) and (c) illustrates clearing at user j (termination).

* Provisional values; the actual target values are for further study.

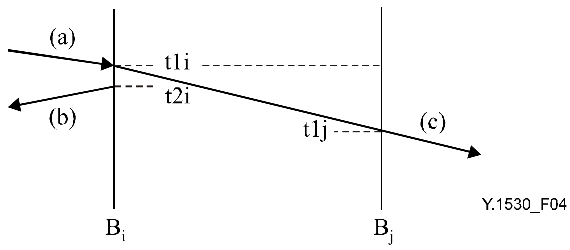


Figure 4/Y.1530 – Call disconnect processing delay

6.2.1 Connection disconnect delay (CDD)

Using the notation in Figure 4, the following difference is formed for connection disconnect delay:

$$CDD = t1j - t1i$$

The worst-case delay objective for CDD, applying to any one-hour period, may be specified as:

Table 5/Y.1530 – Connection Disconnect Delay objectives*

Statistic	Objective
Mean	3500 ms
95% – ile	FFS

6.2.2 Connection Release Delay (CRD)

Using the notation in Figure 4, the following difference is formed for connection release delay:

$$CRD = t2i - t1i$$

As this parameter does not have end-to-end significance, no objectives will be established.

7 Accuracy and dependability parameters and objectives

7.1 Access parameters

Two circuit mode access parameters, connection set-up error probability and connection set-up failure probability, are defined in 7.1.1 and 7.1.2.

Connection set-up error and connection set-up failure are defined between pairs of portion boundaries (B_i, B_j). B_j is one of the set of boundaries to which the connection set-up attempt can properly be routed. Figure 5 identifies the sequence of four particular events that occur at these boundaries during a successful connection set-up. A connection set-up attempt over this portion is a sequential occurrence of corresponding events [(a), (b), (c), (d)] prior to expiration of timer T. Connection set-up errors and connection set-up failures within this portion are defined below. Any other unsuccessful connection set-up attempt is caused by problems outside the portion and is excluded from the measurement.

NOTE – The model and definition regarding access parameters for SIP and stream control transmission protocol (SCTP) is FFS.

* Provisional values; the actual target values are for further study.

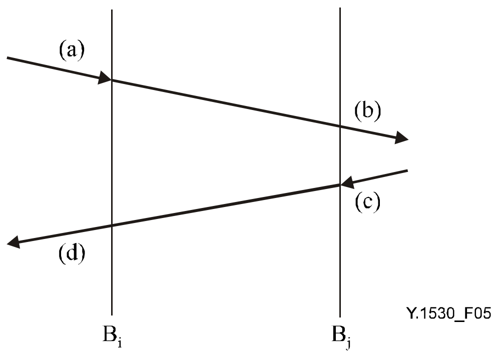


Figure 5/Y.1530 – Circuit mode reference events occurring during successful connection set-up

7.1.1 Connection set-up error probability (CSEP)

Connection set-up error probability is the ratio of total connection set-up attempts that result in connection set-up error to the total connection set-up attempts in a population of interest.

With reference to Figure 5, a connection set-up error is defined to occur on any connection set-up attempt in which event (d) occurs, but event (c) does not occur at an appropriate boundary prior to expiration of timer T.

Connection set-up error is essentially the case of a network-caused "wrong number". It occurs when the network responds to a valid connection request by erroneously establishing a connection to a destination terminal equipment (TE) other than the one designated in the connection 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.

Connection set-up error is distinguished from successful connection set-up by the fact that the intended called user is not contacted and not committed to the session during the connection set-up attempt.

The worst-case objective for CSEP, applying to any one-hour period, may be specified as:

Table 6/Y.1530 – Connection set-up error probability objectives*

Statistic	Objective
Mean	FFS

7.1.2 Connection set-up failure probability (CSFP)

Connection set-up failure probability is the ratio of total connection set-up attempts that result in connection set-up failure to the total connection set-up attempts in a population of interest.

With reference to Figure 5, connection set-up failure is defined to occur on any connection set-up attempt in which either one of the following outcomes is observed prior to expiration of timer T/:

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

Connection set-up attempts that are cleared by the portion as a result of incorrect performance or non-performance on the part of an entity outside the portion are excluded.

* Provisional values; the actual target values are for further study.

A connection set-up 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 set-up 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 connection period, with the result that a connection is not established before the time-out;
- all channels at the called TE are in use.

The worst-case objective for CSFP, applying to any one-hour period, may be specified as:

Table 7/Y.1530 – Connection set-up failure probability objectives*

Statistic	Objective
Mean	FFS

7.2 Disengagement parameters

This clause defines two disengagement parameters: connection premature disconnect probability and connection clearing failure probability.

7.2.1 Connection premature disconnect parameters

7.2.1.1 Connection premature disconnect probability (CPDP)

The premature disconnect probability for a portion is the probability, in any given second, that a connection experiences a premature disconnect generated within that portion.

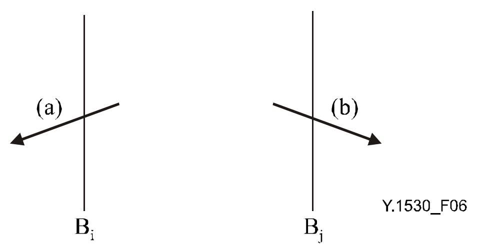


Figure 6/Y.1530 – Circuit mode reference events (REs) defining premature disconnect at B_i and B_j

Referring to Figure 6, a premature disconnect is defined to have been generated within a portion when, in the absence of an external disconnect event (e.g., a premature disconnect stimulus or a disconnect request), or when a premature disconnect stimulus occurs within the portion and is transferred across a boundary of the portion.

* Provisional values; the actual target values are for further study.

The worst-case objective for CPDP, applying to any one-hour period, may be specified as:

Table 8/Y.1530 – Connection premature disconnect probability objectives*

Statistic	Objective
Mean	FFS

7.2.1.2 Connection premature disconnect stimulus (CPDS)

The complete definition of premature disconnect stimulus is for further study. However, ten consecutive severely errored seconds is an instance of a premature disconnect stimulus.

7.2.2 Connection clearing failure probability (CCFP)

Connection clearing failure probability is the ratio of total connection clearing failures to the total connection clearing attempts in a population of interest.

Connection clearing failure is defined with reference to events at the boundaries of a portion (B_i, B_j). For example, in the ISDN context a connection clearing attempt occurs when a DISConnect or RELease message enters the portion creating a reference event at B_i. A connection clearing failure occurs when no corresponding connection clearing reference event occurs at B_j within T_{ccf} seconds.

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

The worst-case objective for CCFP, applying to any one-hour period, may be specified as:

Table 9/Y.1530 – Connection clearing failure probability objectives*

Statistic	Objective
Mean	FFS

8 Security

This Recommendation does not specify a protocol. Hence, there are a few areas where security issues may arise, and all are associated with implementation of the performance parameters in measurement systems.

Measurement systems that assess the performance of networks according to the parameter definitions defined in this Recommendation should limit the measurement call and traffic to appropriate levels to avoid abuse. Administrations or Operators should agree on acceptable levels of measurement call and traffic in advance.

Systems that monitor user calls and traffic for the purpose of measurement must maintain the confidentiality of user information.

Systems that attempt to make measurements may employ techniques to determine if additional calls and traffic have been inserted by an attacker appearing to be part of the population of interest.

* Provisional values; the actual target values are for further study.

Annex A

Call set-up timers

While call processing performance parameters in this Recommendation are the same as N-ISDN parameters in ITU-T Rec. I.352, call attempts for voice service in hybrid IP networks may be different from call attempts for N-ISDNs. User terminals for N-ISDNs make point-to-point 64-kbit/s calls, and call processing equipment is seldom overloaded. On the other hand, intelligent user terminals for hybrid IP networks can make many call attempts at a high rate, and these call attempts may degrade the call processing performance by overloading call processing equipment.

Call processing performance, such as connection establishment delay, is determined through measurements defined at the UNI. If TEs make call attempts at a high rate, the call processing performance of call attempts on the UNI will strongly depend on the rate of call attempts from TEs. Thus, specifying conditions for the call attempt rate of the TEs should minimize the impact on call processing performance.

1) User terminals do not make any new calls in the interval between a connection request and its complete/incomplete response.
2) The intervals between calls made by a user terminal shall be more than a time interval value T1.
3) Calls considered in this Recommendation are initiated at times when the network is available.
4) The intervals between an incomplete answer and the next call request shall be more than a time interval value T2.

Appendix I

Reference model for TIPHON senario 2

Figure I.1 illustrates a general reference configuration. This reference configuration is based on TIPHON Scenario 2 [1]. Table I.1 shows TIPHON inter-network connection cases and scenarios.

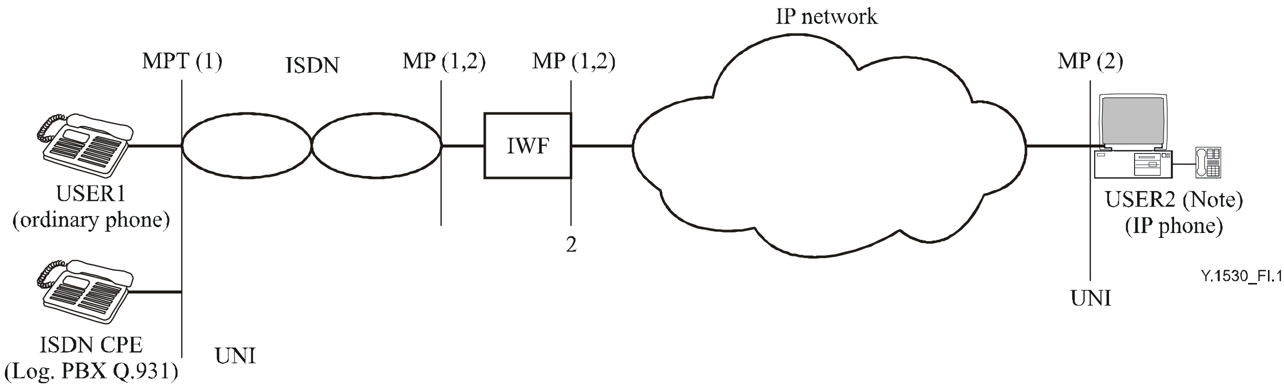


Figure I.1/Y.1530 – General reference configuration

Table I.1/Y.1530 – Inter-network connection cases

Case	Origination	Backbone	Termination	Scenario
1	IP	IP	IP	0
2	IP	IP	SCN	1
3	IP	SCN	SCN	1
4	IP	SCN	IP	4
5	SCN	IP	SCN	3
6	SCN	IP	IP	2
7	SCN	SCN	IP	2
8	SCN	SCN	SCN	(Note)
SCN Switched Circuit Network NOTE – Case 8 is outside the scope of TIPHON.				

Two Measurement Points T (MPT) are located at an User-Network Interface (UNI) and an interworking function (IWF) interface of ISDN. Two Measurement Points (MP) are located at an UNI and an IWF interface of IP networks. The measurement points in Figure I.1 are identified according to the method established below:

- MPT (1) represents the MPT associated with user 1.
- MP (1,2) represents the MPT associated with IWF.
- MP (1,2) represents the MP associated with IWF.
- MP (2) represents the MP associated with user 2.

An IWF performs all protocol conversions and data adaptations. An IWF device may be used to connect two networks (i.e., a network adaptor) or terminal to a network (i.e., a terminal adaptor). The IWF provides the following mechanisms:

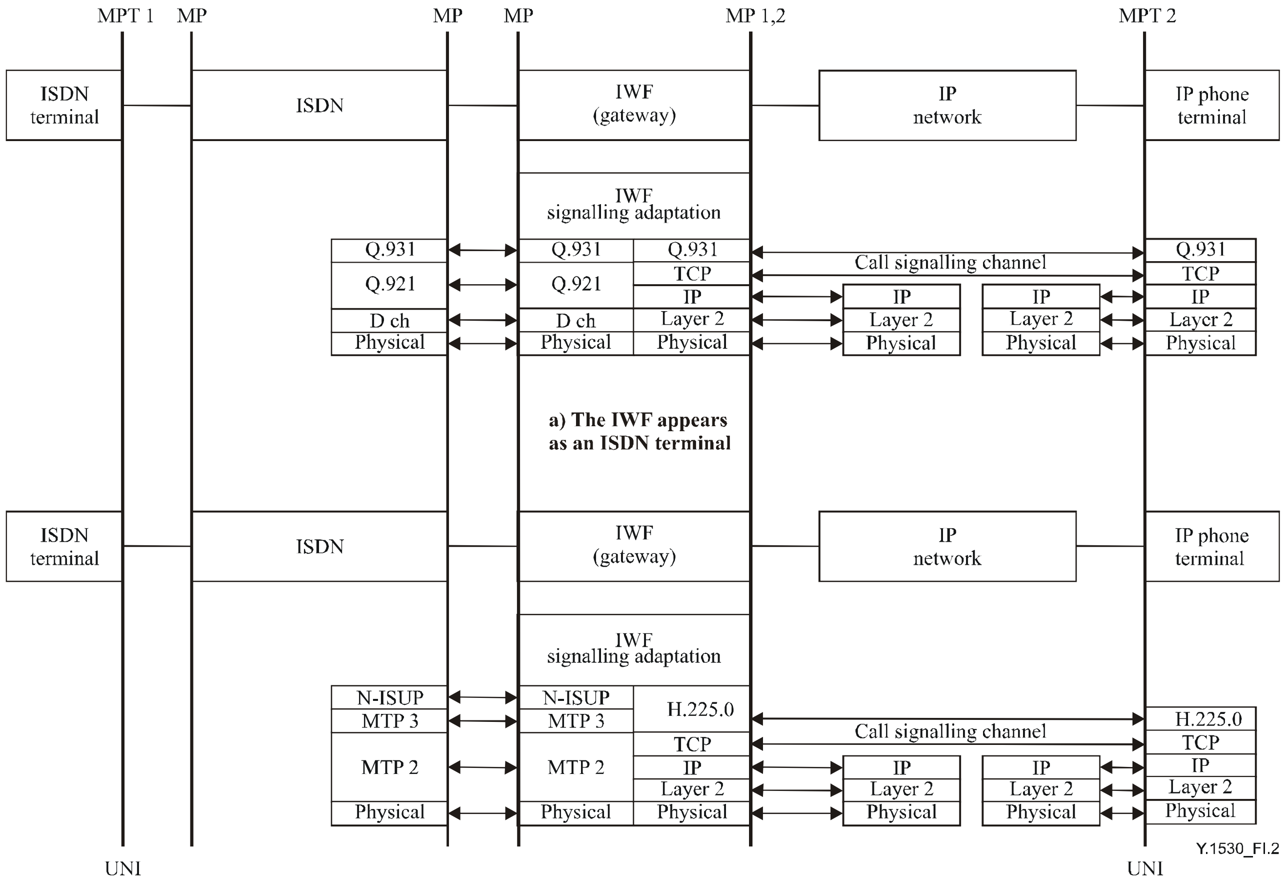
- Signalling adaptation: This consists of the processing and translation of incoming signalling messages. It mainly concerns the call set-up and clearing phases.
- Media control: This consists of identifying, processing, and translating service-specific control events that may be generated by the user or the terminal.
- Media adaptation: This consists of adapting the voice data to the data transfer channel of the downstream network.

An IWF consists of gateway and gatekeeper. The gateway is responsible for providing all translations necessary for transmission formats and control procedures between the IP supported portion and the ISDN. The gatekeeper is an optional equipment that provides call control services to the terminals.

Figure I.2-a shows the IWF protocol stacks in the control plane. Figure I.2-b shows the protocol stack needed with the gateway connected as an ISDN node.

Figure I.3 shows call establishment processing delay for voice service in TIPHON scenario 2, and CSD = $D_{1,2} - D_2$, CPSD = $D_{A1} - D_2$, CASD = $D_{2,1}$.

Figure I.4 shows call disconnect processing delay for voice service in TIPHON scenario 2, and CDD = $D_{1,c2}$, CRD = $D_{1,c}$.



b) The IWF appears as a network node

Figure I.2/Y.1530 – Functional components, protocols and measuring boundaries

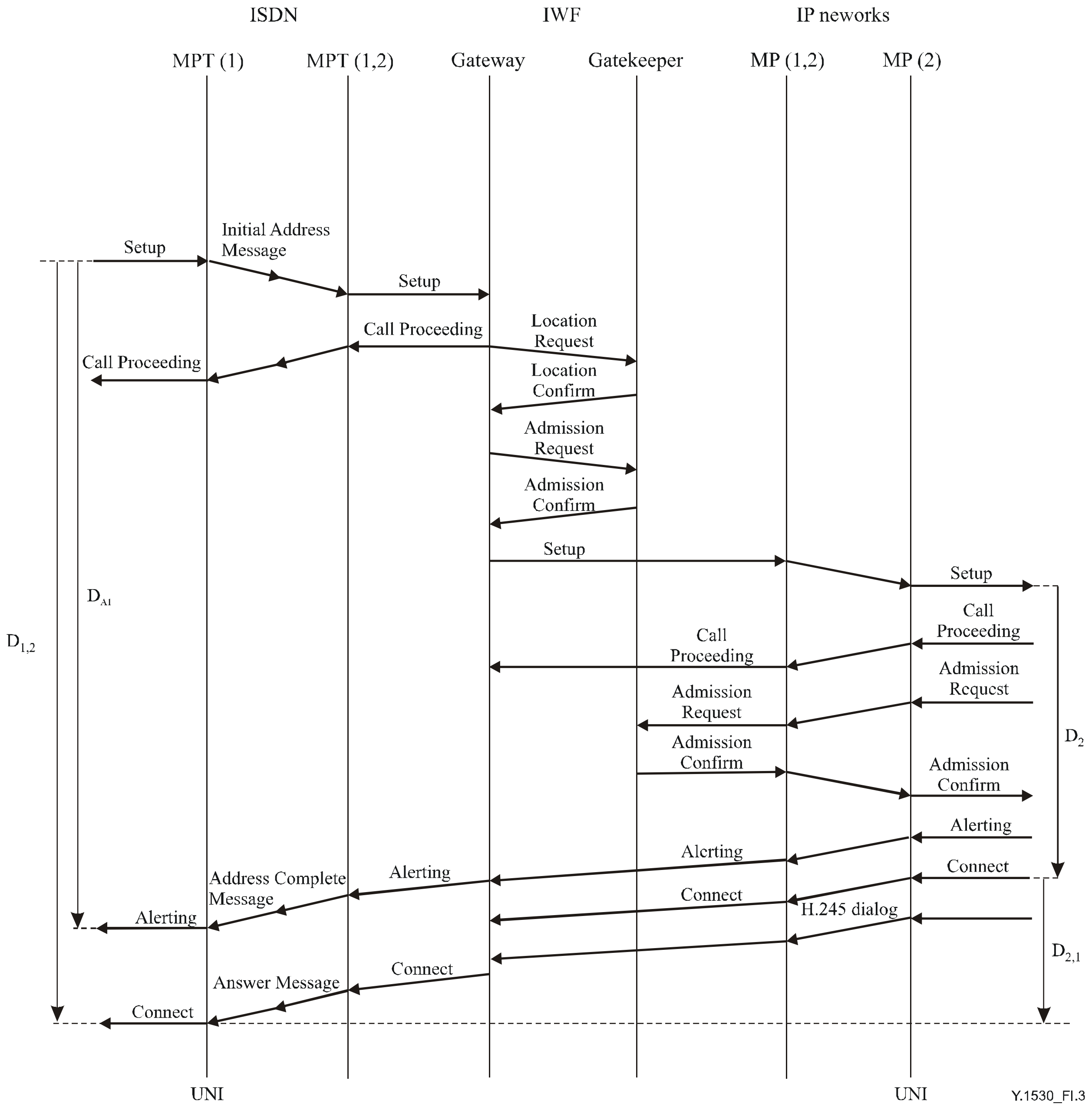


Figure I.3/Y.1530 – Call establishment processing delay for voice service in hybrid IP networks

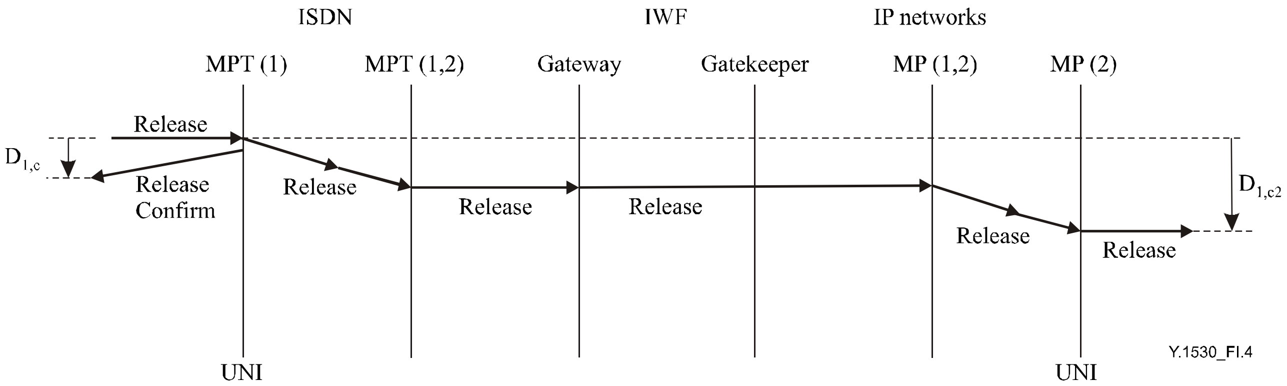


Figure I.4/Y.1530 – Call disconnect processing delays

Appendix II

Measurement methods for estimating VoIP signalling performance

II.1 Performance parameters to be measured

The following are the speed of service, accuracy and dependability parameters associated with the transfer of user information:

- Connection Set-up Delay (CSD).
- Connection Answer Signal Delay (CASD).
- Connection Set-up Failure Probability (CSFP).
- Connection Release Delay (CRD).
- Premature Disconnect Probability (PDP).

II.2 Measurement architectures

This clause provides an overview of the various architectures, which may be used to measure performance parameters for VoIP networks.

A general measurement methodology involves setting up a call to a data sink and generating a known and sufficient quantity of traffic. The protocol and user information signals transferred across the user/network (DTE/DCE) interfaces are observed in real-time and a chronological event history is compiled. This history can then be analysed to provide a measurement of the performance parameters. The measurements of packet-switched networks require a source, a sink and one or more monitors. In the case of measurements over IP connections, the connection, for example a TCP session or RTP session, is to be established and the network is to be configured prior to performing the measurement trial. The source transmits packets over the transport channel. The sink simply receives the packets and performs functions like call processing. The function of the monitor is to record the relevant reference events. The monitor function should be placed as near as possible to the boundaries of the portions under test.

Sources and sinks can either be controlled or non-controlled. Controlled sources and sinks are under the control of the test program and must respond quickly to events exiting the portions under test. Examples of controlled sources or sinks are stand-alone test equipment, special software within network equipment and special programs within customer applications. Non-controlled sources and sinks may not always respond quickly to network events. The most important examples of non-controlled sources and sinks are live customer applications, generating and receiving traffic according to their own needs.

A monitor function can be provided by stand-alone test equipment connected at the appropriate IP interface. The monitor function can also reside in the test device that provides the source or sink function.

Various combinations of monitors and controlled and non-controlled sources and sinks can be used to measure VoIP network performance. Figure II.1 shows some of these combinations. The architectures are identified by specifying whether the source and sink are controlled (C) or uncontrolled (N), and whether the two portion boundaries are monitored (M) or unmonitored (U).

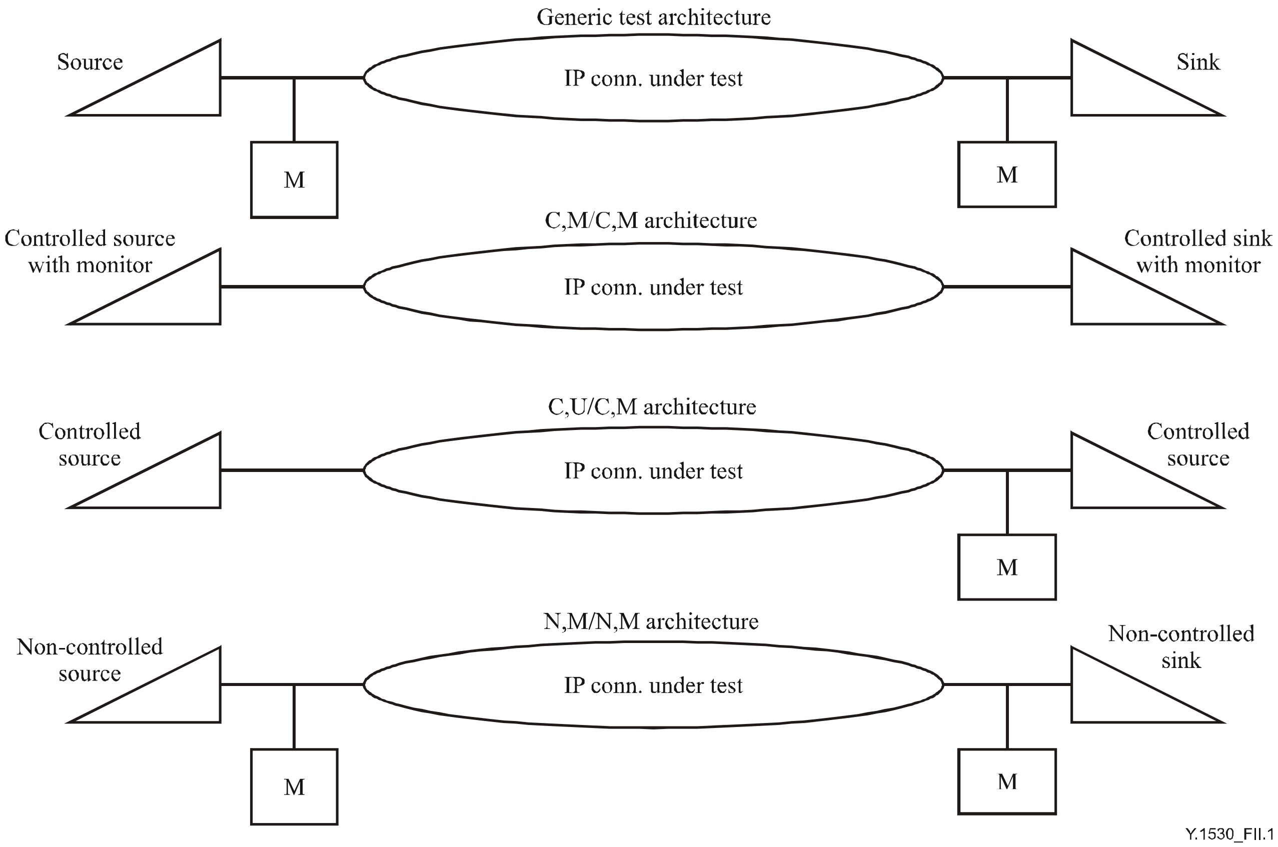


Figure II.1/Y.1530 – Example test architectures

The possible combinations of controlled and non-controlled sources and sinks and monitored and unmonitored boundaries yield different architectures. Table II.1 shows these architectures with an indication of their ability to measure each parameter.

Table II.1/Y.1530 – Summary of measurement architectures

Measurement / Architecture	Performance parameters				
	CSD	**CASD**	**CSFP**	**CRD**	**PDP**
CM/CM	Y	Y	Y	Y	Y
NM/CM	Y	Y	Y	Y	Y
CM/NM	Y	Y	Y	Y	Y
NM/NM	Y1)	Y1)	Y1)	Y1)	Y1)
CU/CM	Y	Y	Y	Y	Y
CU/NM	Y2)	Y2)	Y2)	Y2)	Y2)
CM/CU	Y	Y	Y	Y	N
NM/CU	Y	Y	Y	Y	N
NM/NU	Y	N	N	N	N
CM/NU	Y	N	N	N	N

Table II.1/Y.1530 – Summary of measurement architectures

Architecture \ Measurement	Performance parameters				
	CSD	CASD	CSFP	CRD	PDP
NU/NM	N	N	N	N	N
NU/CM	N	N	N	N	N

1) Assumes the two monitors are synchronized.

2) Assumes the user data created by source is known in advance, i.e., assumes that a predetermined Call SeQ. number or equivalent is used at the source, which is known to the sink.

II.3 Measurement methods using CM/CM architecture

Each performance measurement trial consists of two procedures:

- *Data Extraction*: Packet-layer reference events associated with the trial are created (or observed), time stamped and recorded at the appropriate section boundaries.
- *Data Reduction*: The recorded reference event histories are processed, consistent with the performance parameter definition, to determine the trial's outcome.

The transport connection should be active and operational on the network portion under test, i.e., the IP service should be in the available state during the entire length of each test. Under certain conditions, the service may be considered unavailable after the performance test is done. This is in the event that any defined minimum performance objectives are not met. Such a criterion may be based on the service availability definition in this Recommendation.

Matching packet level reference events

The data reduction procedure requires that all packet level reference events be matched at the two test boundaries. Several alternative methods can be used to do this. A specific pattern of user information data can be used. The data reduction procedure should also be sophisticated enough to recognize lost, errored or extra packets.

II.4 Access function VoIP signalling measurement methods

II.4.1 Data extraction procedure

The data extraction procedure is shown in Figure II.2.

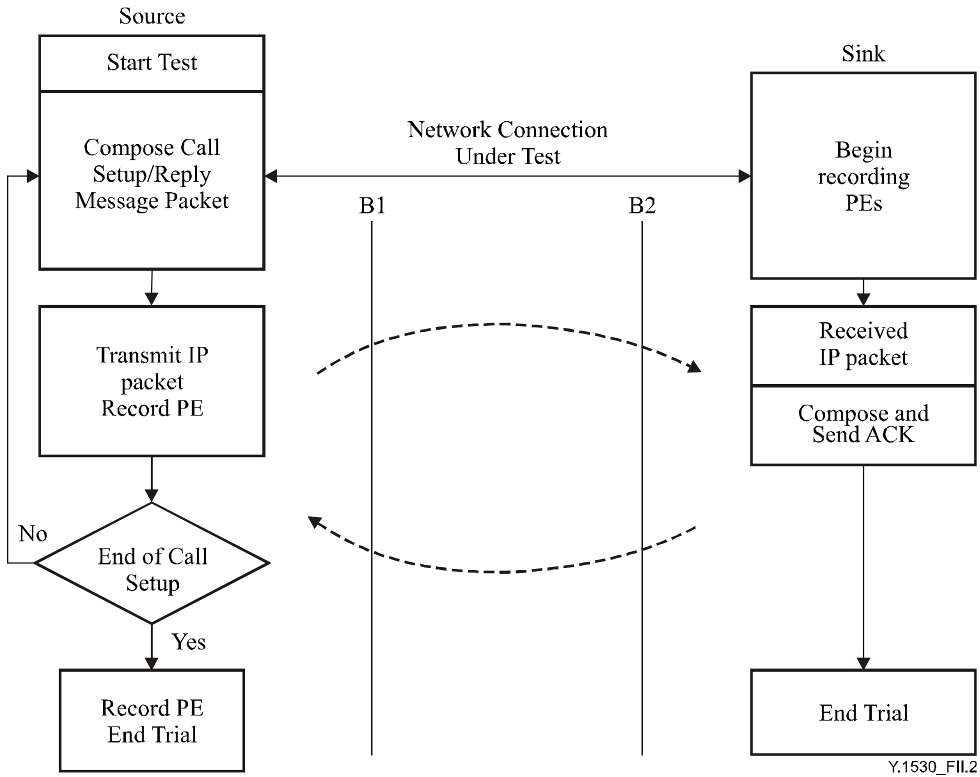


Figure II.2/Y.1530 – Call establishment trial data extraction procedure

II.4.2 Data reduction procedure

The data reduction procedure is used to analyse the data collected from the data extraction procedure to calculate the relevant performance parameters for the information transfer over the IP connection. Since it is necessary to ensure that the service availability criterion is satisfied before any performance can be meaningfully evaluated, the data reduction procedure should include necessary computation to help determine if the service is considered available. If service was deemed available, the call performance statistics can be updated. Figure II.3 shows the steps for the data reduction procedure.

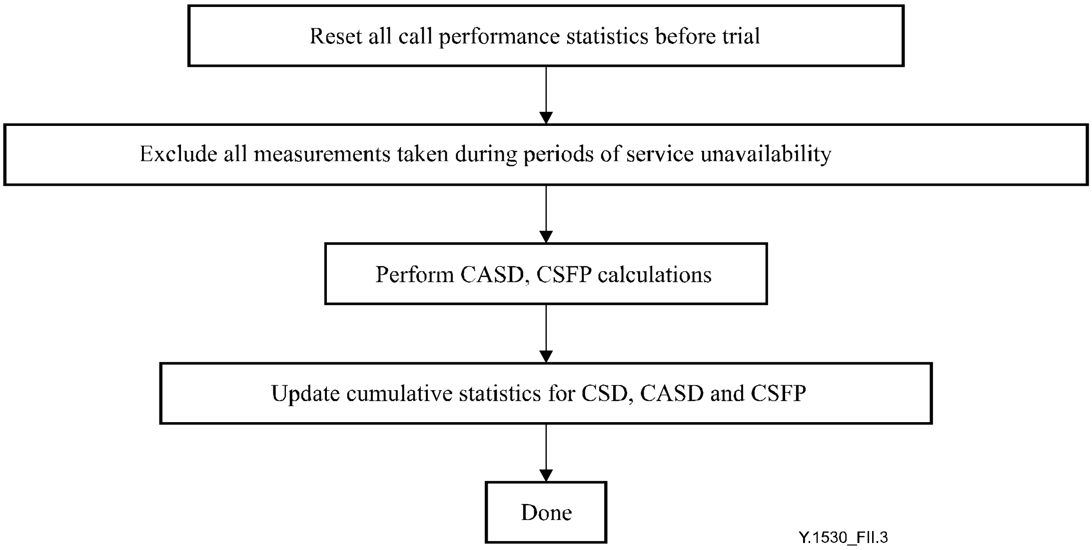


Figure II.3/Y.1530 – Call establishment trial data reduction procedure

Figure II.3 shows the procedures for calculation of CASD. Figures II.4 and II.5 show the procedures for the calculation of CASD and CSD, respectively.

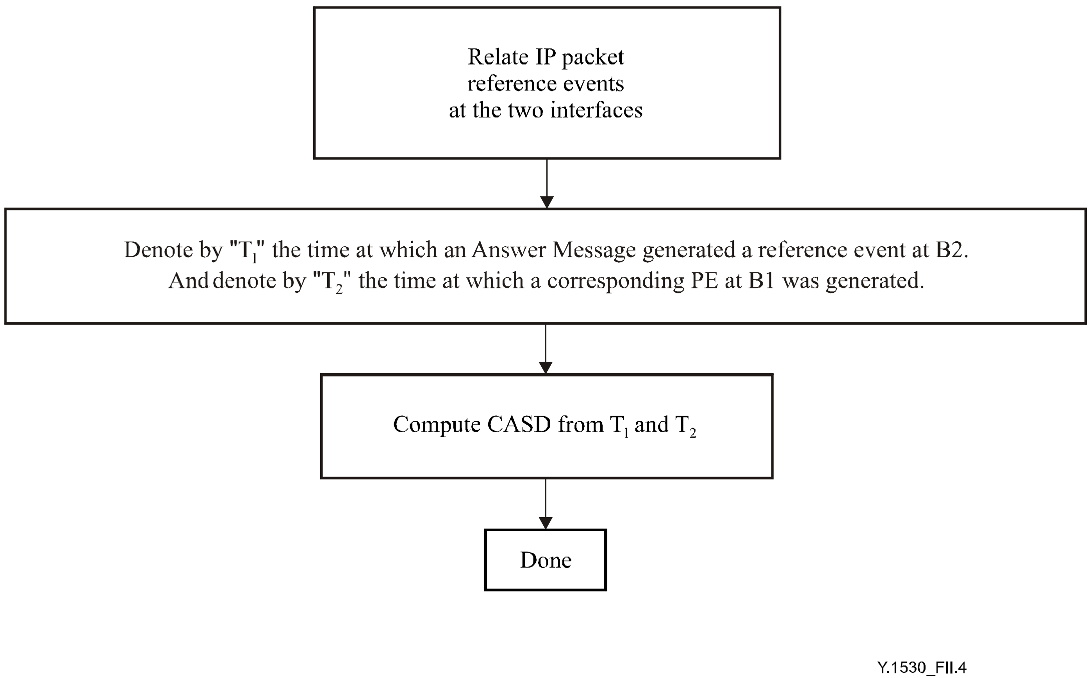


Figure II.4/Y.1530 – Connection answer signal delay calculations

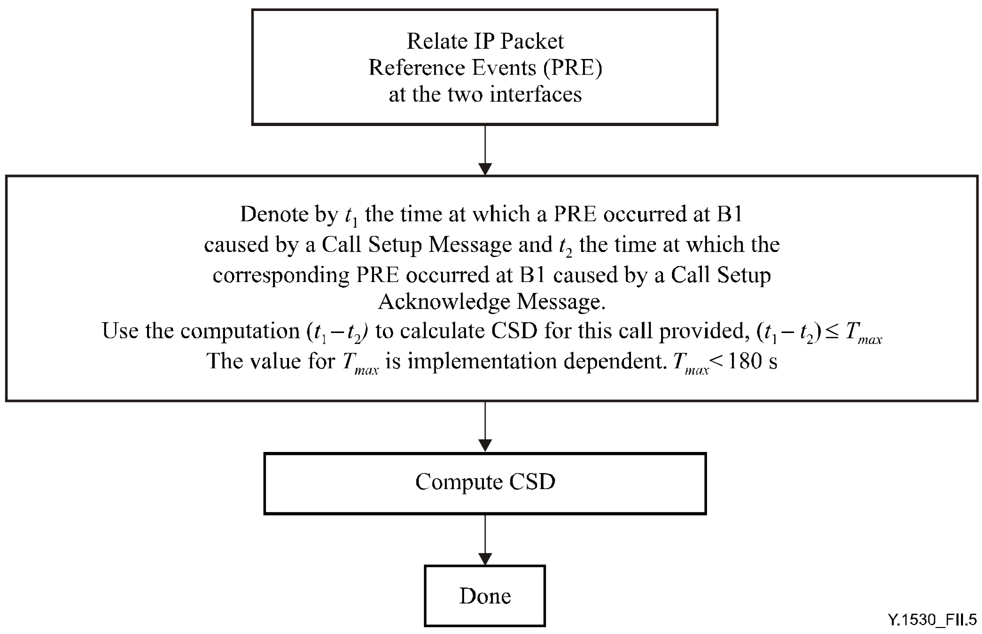


Figure II.5/Y.1530 – VoIP connection set-up delay calculations

II.5 Disengagement function VoIP measurement methods

II.5.1 Data extraction procedure

The data extraction procedure is shown in Figure II.6.

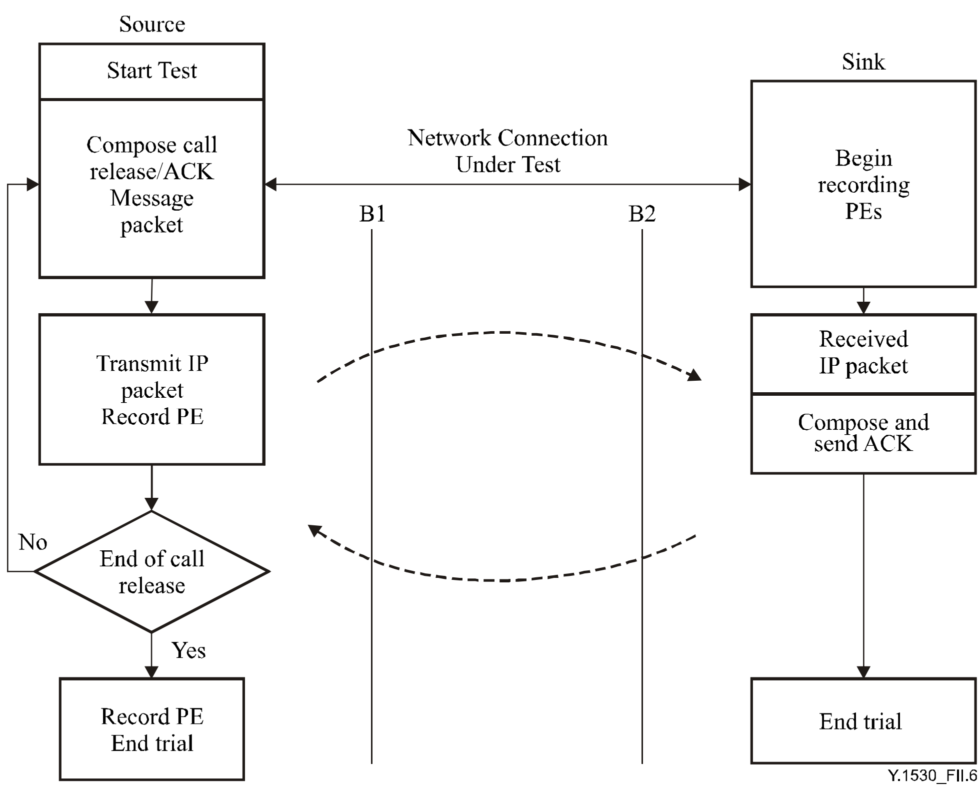


Figure II.6/Y.1530 – Call release trial data extraction procedure

II.5.2 Data Reduction Procedure

The data reduction procedure is used to analyse the data collected from the data extraction procedure to calculate the relevant performance parameters for the information transfer over the IP connection. Figure II.7 below shows the steps for the data reduction procedure.

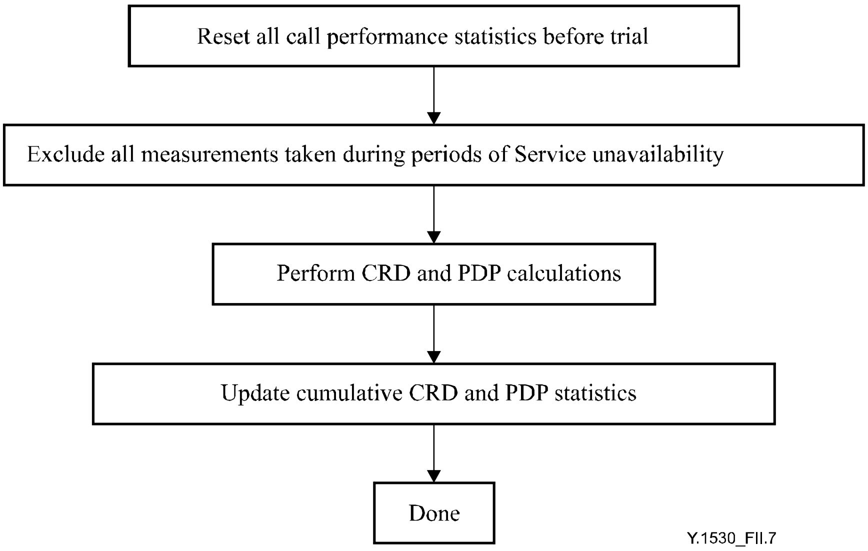


Figure II.7/Y.1530 – Call release trial data reduction procedure

Figure II.8 shows the procedure for the calculation of CRD.

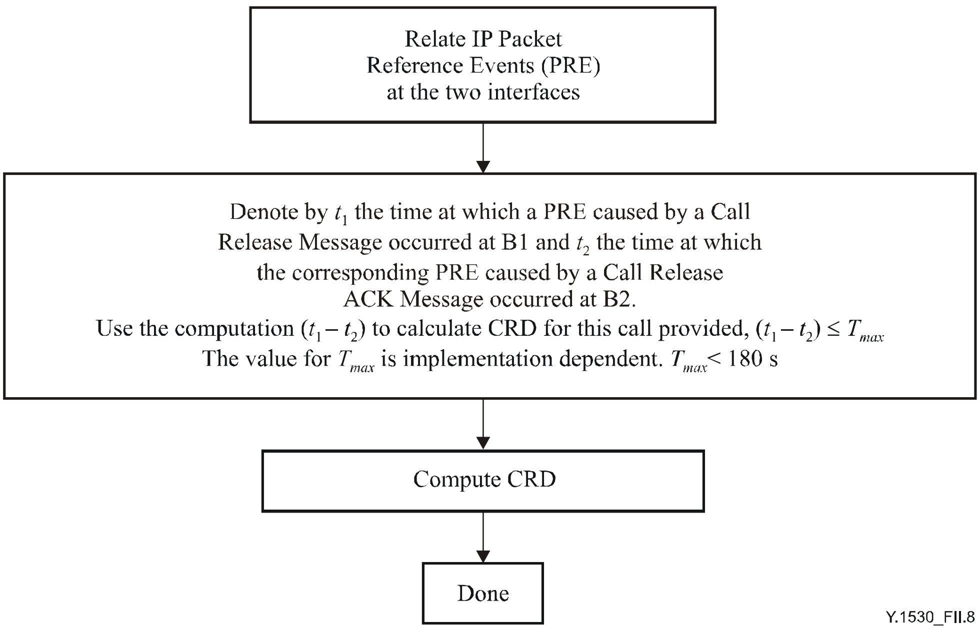


Figure II.8/Y.1530 – VoIP connection release delay calculations

Appendix III

BICC architecture

An instance of the Bearer Independent Call Control (BICC) architecture is shown in Figure III.1. It consists of ATM or IP backbone networks and Media gateways (MG) connected to network edge nodes. The role of a MG is to map TDM time slots to ATM cells or IP packets and vice versa. MGs can be either line (local loop) or trunk MGs. Media gateway controllers (MGC) perform call control functions and interact with one or more MGs under their control.

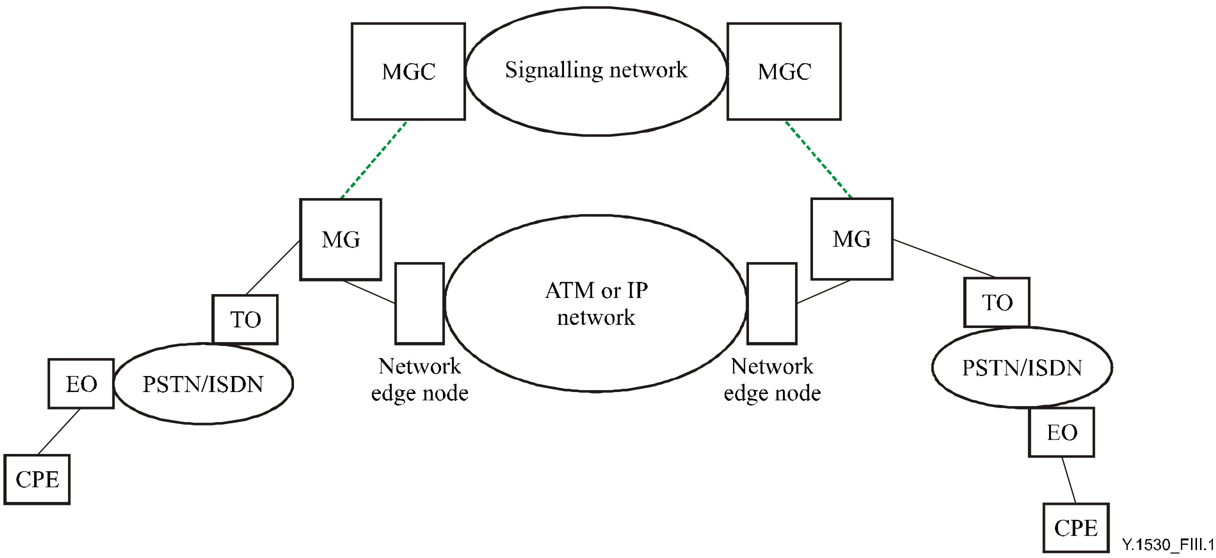


Figure III.1/Y.1530 – An instance of BICC architecture

When MGs are connected to an ATM network, AAL type 1 or AAL type 2 may be used to transport voice services. When MGs are connected to an IP network, RTP over UDP and IP is used. MGs interface to the PSTN with TDM 64 kbit/s channels.

So far BICC supports ATM networks with AAL type 1 and type 2 and IP networks with RTP/UDP/IP for voice transport.

Current BICC capabilities will remain for some time when the network evolves to the multi-service MPLS network.

In this architecture, UNIs for call processing are defined as UNIs on PSTN/ISDN.

Appendix IV

New call processing QoS class structure

The new call processing QoS class structure with four classes for voice service in IP Hybrid networks to meet varying services is proposed. Call processing QoS classes should be more about definition, requirements, protocols, and so on.

Call processing QoS Class 1 specifies that all objectives of call processing performance parameters are equal to those specified in ITU-T Recs I.352 and I.359 about ISDN. This class may be easy to be acceptable for users and for carriers that design some internetworking.

Call processing QoS Class 2 specifies that objectives of connection set-up delay and connection set-up failure probability are more than those specified in ISDN. Other parameters of call processing performance are equal to those of ISDN specified because the control of these parameters would be difficult and users' concern would be low. This class may be used for some IP telephone networks with internetworking.

Call processing QoS Class E specifies that objectives of connection set-up delay and connection set-up failure probability are less than those specified in ISDN. Other parameters of call processing performance are equal to those of ISDN specified because the control of these parameters would be difficult and users' concern would be low. This class may supply high or excellent call processing performance for special services. This class should not be used for emergency services defined in ITU-T Rec. Y.1271.

Call processing QoS Class U is the unbound (or unspecified) Class that all objectives of call processing performance parameters are the unbound (or unspecified). This class may be used for best effort IP call processing networks with low cost.

Table IV.1 illustrates the new call processing QoS class structure with four classes for voice service in IP Hybrid networks.

Table IV.1/Y.1530 – Call processing QoS class definitions and performance objectives

<table>
<tr><th rowspan="2">Call processing QoS class</th><th colspan="3">Connection set-up parameters</th><th colspan="2">Connection disengagement parameters</th></tr>
<tr><th>Connection set-up delay</th><th>Connection set-up error probability</th><th>Connection set-up failure probability</th><th>Connection disconnect delay</th><th>CPDP
CCFP</th></tr>
<tr><td>QoS Class E
(High or Excellent priority level)</td><td>Mean < 7500 ms
95%ile < 8450 ms
(FFS)</td><td rowspan="3">Default
(FFS)</td><td>Mean < A
(FFS)</td><td rowspan="3">Defaults
Mean = 3500 ms
95%ile = 4250 ms
[I.352]</td><td rowspan="3">Defaults
(FFS)</td></tr>
<tr><td>QoS Class 1
(Ordinary telephone level)</td><td>Mean = 7500 ms
95%ile = 8450 ms
[I.352]</td><td>Mean = A
(Value A is FFS)
[I.359]</td></tr>
<tr><td>QoS Class 2
(IP telephone level)</td><td>Mean > 7500 ms
95%ile > 8450 ms
(FFS)</td><td>Mean > A
(FFS)</td></tr>
<tr><td>QoS Class U
(Best effort level)</td><td>U</td><td>U</td><td>U</td><td>U</td><td>U</td></tr>
<tr><td colspan="6">CPDP Connection Premature Disconnect Probability
CCFP Connection Clearing Failure Probability
U Unspecified or Unbound
FFS For further study</td></tr>
</table>

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