



ATIS-0100504.1998(R2006)

**Packet-Switched Data Communication Services –  
Performance Parameters, Measurements Methods, and  
Objectives**



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Revision and consolidation  
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American National Standard  
for Telecommunications –

**Packet-Switched Data  
Communication Service –  
Performance Parameters,  
Measurements Methods, and Objectives**

Secretariat

**Alliance for Telecommunications Industry Solutions**

Approved January 6, 1998

**American National Standards Institute, Inc.**

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**Foreword** (This foreword is not part of American National Standard T1.504-1998.)

This standard defines a set of performance parameters for packet-switched data communication services. The parameters may be used to specify or measure the performance of any virtual connection (or section of a virtual connection) delimited by the interfaces described in two ITU-T Recommendations: X.25 (*Interface between data terminal equipment (DTE) and data circuit-terminating equipment (DCE) for terminals operating in the packet mode and connected to public data Networks by dedicated circuit*), and X.75 (*Packet-switched signalling systems between public networks providing data transmission services*). The parameters address both virtual call and permanent virtual circuit services. They describe performance relative to three primary data communication functions: access, user information transfer, and disengagement. Each function is considered with respect to three general performance criteria: speed, accuracy, and dependability. The defined parameters are of two types: primary parameters and availability parameters. The primary parameters provide a relatively detailed description of performance that encompasses each of the three functions and criteria. The availability parameters are derived from observations of the primary parameter values and provide a more macroscopic, longer-term view of performance. The parameters are defined on the basis of standard interface events to facilitate their measurement with stand-alone test equipment.

This standard evolved from, and is complementary to, a number of other national and international standards. In *American National Standard for Information systems - Data communications systems and services - User-oriented performance parameters*, ANSI X3.102-1992, parameters are defined that may be used to describe the performance of data communication services from the point of view of the end user in a protocol-independent manner. Many of the parameters defined in this T1 standard are protocol-specific counterparts to the parameters in ANSI X3.102. In *American National Standard for Information systems - Data communication systems and services - Measurement methods for user-oriented performance evaluation*, ANSI X3.141-1987 (R1992), measurement methods for the parameters described in ANSI X3.102 are defined. In *General quality of service parameters for communications via Public Data Networks*, ITU-T Recommendation X.140, protocol-independent quality-of-service parameters are defined that are similar to those defined in ANSI X3.102. Finally, packet switched service performance parameters, apportionment boundaries and worst-case performance values for international packet switched services are defined in ITU-T Recommendations X.134, X.135, X.136, and X.137.

This standard contains five annexes. Annexes A through E are informative and are not considered part of the standard.

Suggestions for improvement of this standard are welcome. They should be sent to the Alliance for Telecommunications Industry Solutions, 1200 G Street, NW, Suite 500, Washington, DC 20005

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American National Standard  
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# Packet-Switched Data Communication Service – Performance Parameters, Measurement Methods, and Objectives

## 1 Introduction

The purpose of this standard is to define a set of parameters that may be used in specifying and measuring the performance of packet switched data communication services provided in accordance with ITU-T Recommendations X.25 and X.75.

The defined parameters are applicable to switched virtual-call services. Some parameters pertain to permanent virtual-circuit services as well. They describe four general service performance characteristics: speed, accuracy, dependability, and availability.

The parameters defined in this standard may be used to specify or measure the performance of any entity delimited by the boundaries defined in ITU-T Recommendations X.25 and X.75. This standard specifies the boundaries, described in ITU-T Recommendations X.25 and X.75, at which performance can be measured. The specification of these boundaries is not intended to imply any specific allocation of performance responsibility.

The organization of this standard is summarized in figure 1. For comparability and completeness, packet switched network performance is considered in the context of the 3x3 performance matrix defined in *American National Standard for Information systems - Data communication systems and services - User-oriented performance parameters*, ANSI X3.102-1992. Three protocol-independent data communication functions are identified in the matrix: access, user information transfer, and disengagement. Each function is considered with respect to three general performance concerns (or "performance criteria"): speed, accuracy, and dependability. A two-state model provides a basis for describing service availability.

Clause 2 presents normative references.

Clause 3 defines sections of a virtual connection whose boundaries are associated with the interfaces described in ITU-T Recommendations X.25 and X.75, and defines a set of packet layer reference events (PEs) that provide a basis for performance parameter definition.

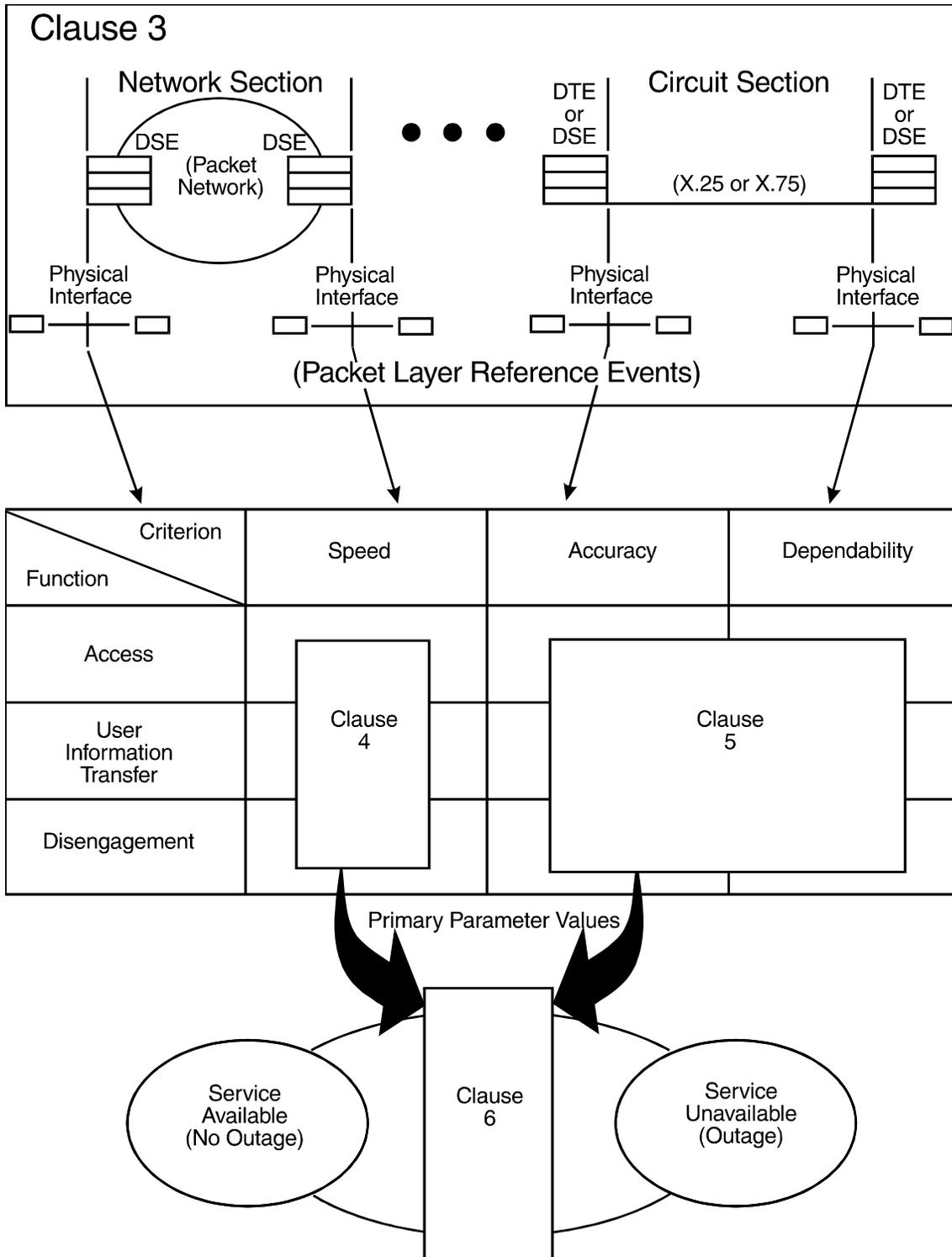
Clause 4 defines protocol-specific speed-of-service parameters associated with each of the three data communication functions.

Clause 5 defines protocol-specific accuracy and dependability parameters associated with each function.

Clause 6 specifies the availability function and defines availability parameters.

Clause 7 defines measurement methods to be used in assessing and comparing the performance of packet-switched data communications services.

Clause 8 specifies worst-case design objectives for the packet-switched performance parameters defined in clauses 4, 5, and 6.



**Figure 1 - Packet-switched service performance description**

## 2 Normative references

The following standards contain provisions which, through reference in this text, constitute provisions of this American National Standard. At the time of publication, the editions indicated were valid. All standards are subject to revision, and parties to agreements based on this American National Standard are encouraged to investigate the possibility of applying the most recent editions of the standards indicated below.

ANSI X3.102-1992, *Information systems - Data communication systems and services - User-oriented performance parameters*

ANSI X3.141-1987 (R1992), *Information systems - Data communication systems and services - Measurement methods for user-oriented performance evaluation*

ITU-T Recommendation X.25 (10/96), *Interface between data terminal equipment (DTE) and data circuit-terminating equipment (DCE) for terminals operating in the packet mode and connected to public data networks by dedicated circuit*<sup>1)</sup>

ITU-T Recommendation X.75 (10/96), *Packet-switched signalling systems between public networks providing data transmission services*<sup>1)</sup>

ITU-T Recommendation X.140 (09/92), *General quality of service parameters for communication via public data networks*<sup>1)</sup>

## 3 Performance model

This clause of this standard defines four basic virtual connection sections for which performance objectives may be specified: the access circuit section, the internetwork circuit section, the access network section, and the transit network section. These four basic virtual connection sections are described in 3.1 and shown in figure 2. They represent a set of building blocks with which any end-to-end virtual connection can be modeled.

The specific packet layer reference events used in defining performance parameters, as described in ITU-T Recommendations X.25 and X.75, are defined in 3.2 and shown in figure 3.

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<sup>1)</sup> Available from the American National Standards Institute, 11 West 42nd Street, New York, NY 10036.

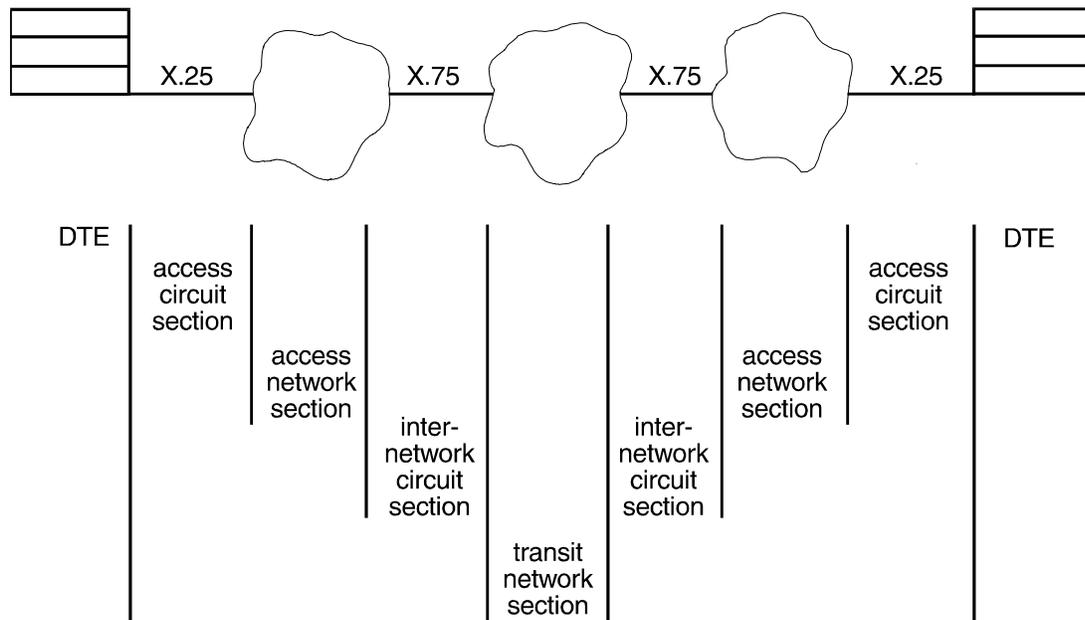


Figure 2 - Sections of a national virtual connection

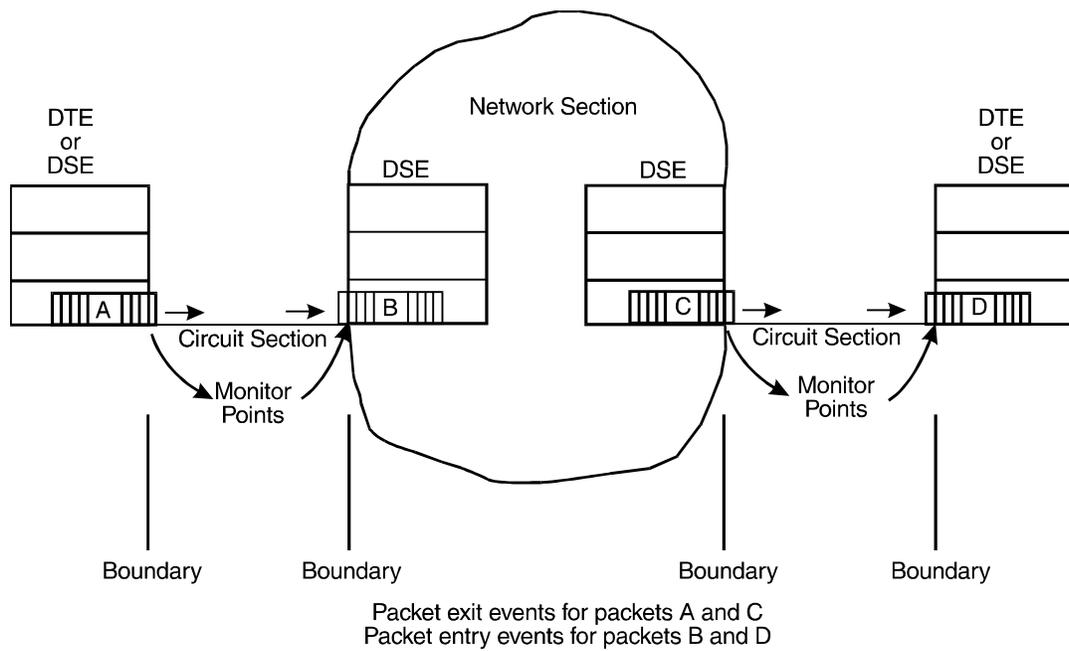


Figure 3 - Example packet layer reference events

### 3.1 Virtual connection sections

In the context of this standard, the following definitions apply:

#### 3.1.1 Circuit section

Either an access circuit section or an internetwork circuit section.

##### 3.1.1.1 Access circuit section

The physical circuit or set of circuits connecting a Data Terminal Equipment (DTE) to the local Data Switching Exchange (DSE). It does not include any parts of the DTE or DSE. It is assumed that the procedures of ITU-T Recommendation X.25 are used on an access circuit section.

##### 3.1.1.2 Internetwork circuit section

The physical circuit or set of circuits connecting a DSE in one network with a DSE in a different network. It does not include any parts of either DSE. It is assumed that the procedures of ITU-T Recommendation X.75 are used on an internetwork circuit section.

#### 3.1.2 Network section

The network components that provide a virtual connection between two circuit sections. A network section may be either an access network section or a transit network section.

##### 3.1.2.1 Access network section

A network section connected to (at least) one access circuit section.

##### 3.1.2.2 Transit network section

A network section between two internetwork circuit sections.

#### 3.1.3 Basic section of a virtual connection

A general term for an access circuit section, an internetwork circuit section, an access network section, or a transit network section.

#### 3.1.4 Section boundary

The boundary that separates a network section from the adjacent circuit section, or separates an access circuit section from the adjacent DTE. (Also called *Boundary*.)

NOTE - Figure 2 illustrates the definitions and delimitations of the virtual connection sections.

### 3.2 Packet layer reference events

In the context of this standard, the following definitions apply:

#### 3.2.1 Packet layer reference event

The event that occurs when a packet crossing a section boundary changes the state of the packet layer interface.

NOTE - The relevant state transitions are those defined explicitly or implicitly in ITU-T Recommendations X.25 and X.75.

Two classes of packet layer reference events are defined:

### **3.2.1.1 Packet entry event**

A packet layer reference event that corresponds to a packet entering a network section (from a circuit section) or a packet entering a DTE (from an access circuit section).

### **3.2.1.2 Packet exit event**

A packet layer reference event that corresponds to a packet exiting a network section (to a circuit section) or a packet exiting a DTE (to an access circuit section).

The time of occurrence of a packet entry event is defined to coincide with the time at which the last bit of the closing flag of the frame carrying the packet crosses the boundary out of the circuit section. The time of occurrence of a packet exit event is defined to coincide with the time at which the first bit of the address field of the frame carrying the packet crosses the boundary into the circuit section. If retransmissions occur, the packet exit event occurs with the first transmission and the packet entry event occurs with the last transmission.

Figure 3 illustrates these terms.

A single packet crossing a boundary between two adjacent virtual connection sections may change more than one aspect of the packet layer interface state, and consequently more than one packet layer reference event may be created. Particular reference events are specified by identifying:

- 1) The relevant boundary
- 2) The type of packet transferred
- 3) The event class (packet entry or packet exit)
- 4) The particular aspect of the state that is changed by the event.

### **3.2.2 Performance-significant reference events**

The performance-significant reference events are the packet layer reference events useful in defining performance parameters. Table 1 lists performance-significant X.25 packet layer reference events associated with the boundaries of access circuit sections. Table 2 lists performance-significant X.75 packet layer reference events associated with the boundaries of internetwork circuit sections. These events and their reference numbers are used in the performance parameter definitions specified in clauses 4 to 6.

The entries in tables 1 and 2 describe the type of packet transferred and the resulting state of the packet layer interface. With the exception of the diagnostic and registration categories, all packet types identified in ITU-T Recommendations X.25 and X.75 are addressed in the tables.

The states identified in the tables differ from those defined in ITU-T Recommendations X.25 and X.75 in two respects:

- 1) Call collision states are omitted, since their specification is not required for performance parameter definition.
- 2) Several new ancillary states are defined, consistent with the existing X.25 and X.75 protocol specifications, to provide a basis for more detailed performance description.

Three ancillary X.25 states and three ancillary X.75 states are defined in this standard to permit more accurate description of flow control effects. The new X.25 states are DCE flow controlled, DTE flow controlled, and DTE and DCE flow controlled. The new X.75 states are STE X flow controlled, STE Y flow controlled, and STE X and STE Y flow controlled. A state diagram for the ancillary X.25 flow control states is shown in figure 4. A state diagram for the ancillary X.75 flow control states is shown in figure 5. In each case, the new states are numbered d4-d6.

Three ancillary state variables are defined:

- 1) *lwl* (*lower edge of the window on the transmit side*). This variable contains the latest P(R) received either in a data packet, a Receive Ready (RR), or a Receive Not Ready (RNR). The value may be implicitly represented using the upper window edge (and the window size).
- 2) *npr* (*next data packet to be received*). This variable contains the P(S) of the next data packet to be received.
- 3) *ric* (*received interrupt count*). Because only one unacknowledged interrupt packet can exist in a particular direction, the interface records the reception of an interrupt across the circuit section. This variable is used to record such events. The variable is cleared when the interrupt confirmation is transmitted.

If the state resulting from packet transfer is not the one listed in the relevant table or the state remains unchanged as a result of the packet transaction, the reference event does not occur. Aspects of the state other than those listed in these tables may change during packet entry or exit, but those events are not viewed as Performance-significant reference events.

When the tables list more than one aspect of the state that might be changed as a result of a particular packet's entry or exit, each of those changes represents a distinct packet layer reference event that can be used in defining different performance parameters. For example, in table 1, event 9a would be used when the correct receipt of the data is relevant, and 9b would be used when the receipt of the acknowledgment is relevant. Event 26b would be used in association with permanent virtual circuits (PVCS) and 26a with other logical channels.

**Table 1 - X.25 packet layer reference events and resulting interface states**

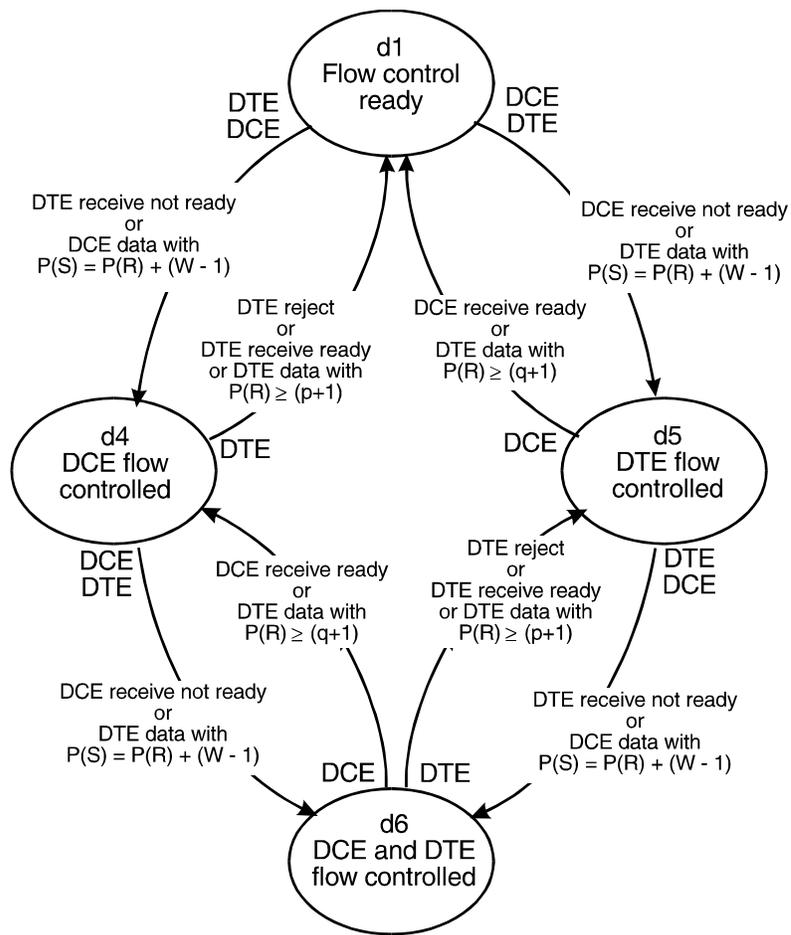
Number	Packet Type	Resulting Interface State
1	Incoming Call	p3 (DCE Waiting)
2	Call Request	p2 (DTE Waiting)
3	Call Connected	p4 (Data Transfer)
4	Call Accepted	p4
5	Clear Indication	p7 (DCE Clear Indication)
6	Clear Request	p6 (DTE Clear Request)
7	DCE Clear Confirmation	p1 (Ready)
8	DTE Clear Confirmation	p1
9 a	DCE Data	npr becomes P(S) + 1
b	DCE Data	lwt becomes P(R)
c	DCE Data	d1 (Flow Control Ready)
10 a	DTE Data	npr becomes P(S) + 1
b	DTE Data	lwt becomes P(R)
c	DTE Data	d1 (Flow Control Ready)
11	DCE Interrupt	ric becomes 1
12	DTE Interrupt	ric becomes 1
13	DCE Interrupt Confirmation	ric becomes 0
14	DTE Interrupt Confirmation	ric becomes 0
15 a	DCE RR	lwt becomes P(R)
b	DCE RR	d1
16 a	DTE RR	lwt becomes P(R)
b	DTE RR	d1
17 a	DCE RNR	lwt becomes P(R)
b	DCE RNR	d5 (DTE Flow Controlled)
c	DCE RNR	d6 (DTE + DCE Flow Controlled)
18 a	DTE RNR	lwt becomes P(R)
b	DTE RNR	d4 (DCE Flow Controlled)
c	DTE RNR	d6
19	DTE REJ	npr becomes P(R)*
20	Reset Indication	d3 (DCE Reset Indication)
21	Reset Request	d2 (DTE Reset Request)
22	DCE Reset Confirmation	d1
23	DTE Reset Confirmation	d1
24	Restart Indication	r3 (DCE Restart Indication)
25	Restart Request	d2 (DTE Restart Request)
26 a	DCE Restart Confirmation	p1
b	DCE Restart Confirmation	d1
27 a	DTE Restart Confirmation	p1
b	DTE Restart Confirmation	d1
**		

\* This is npr from the perspective of the DTE.

\*\* Diagnostic packets are for information only and they do not change the perceived state. Reference events for registration request and confirmation packets are left for further study

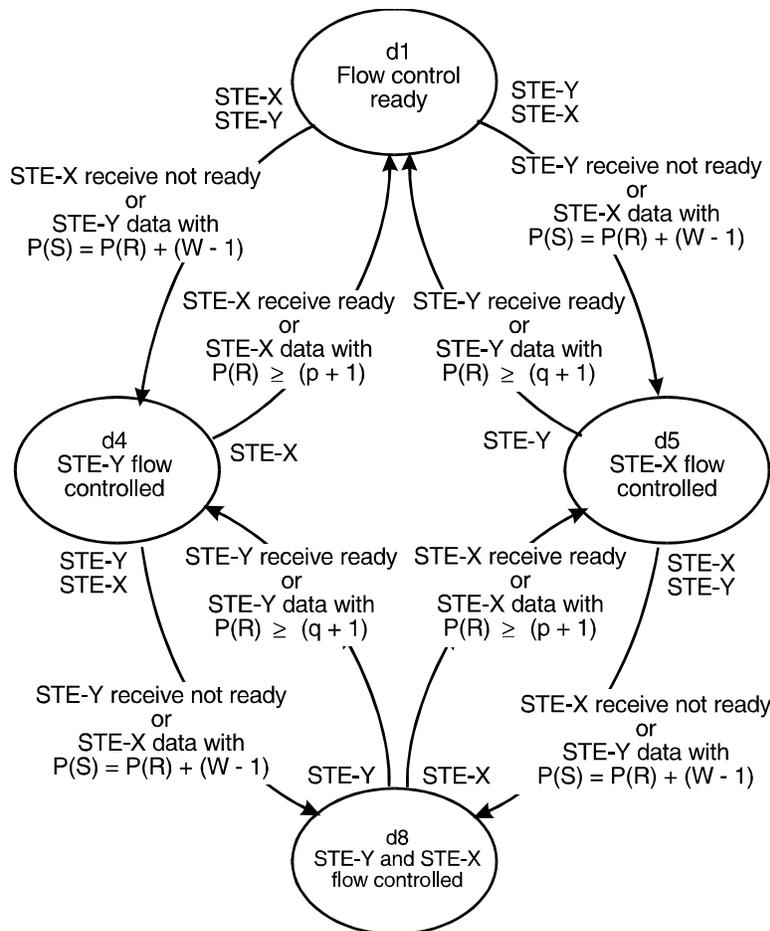
**Table 2 - X.75 packet layer reference events and resulting interface states**

<b>Number</b>	<b>Packet Type</b>	<b>Resulting Interface State</b>
1	Call Request	p2 or p3 (STE Call Request)
2	Call Connected	p4 (Data Transfer)
3	Clear Request	p6 or p7 (STE Clear Request)
4	Clear Confirmation	p1 (Ready)
5 a	Data	npr becomes P(S) + 1
b	Data	lwt becomes P(R)
c	Data	d1 (Flow Control Ready)
6 a	Interrupt	i2 or i3 (STE Interrupt Request)
b	Interrupt	i4 (STE X and Y Interrupt Request)
7 a	Interrupt Confirmation	i1 (No Interrupt Request)
b	Interrupt Confirmation	i2 or i3
8 a	RR	lwt becomes P(R)
b	RR	d1
9 a	RNR	lwt becomes P(R)
b	RNR	d4 or d5 (STE Flow Controlled)
c	RNR	d6 (STE X and Y Flow Controlled)
10	Reset Request	d2 or d3 (STE Reset Request)
11	Reset Confirmation	d1
12	Restart Request	r2 or r3 (STE Restart Request)
13 a	Restart Confirmation	p1
b	Restart Confirmation	d1



NOTE - Variables p and q represent the send sequence numbers of the last DTE data and DCE data packets transferred across the DTE/DCE interface, respectively.

Figure 4 - Diagram of DTE/DCE flow control states



NOTE - Variables p and q represent the send sequence numbers of the last STE-X data and STE-Y data packets transferred across the STE-X/STE-Y interface, respectively.

**Figure 5 - Diagram of STE-X/STE-Y flow control states**

## 4 Speed of service parameters

This clause specifies delay and throughput parameters that can be measured between any pair of boundaries delimiting a virtual connection section or collection of sections. Performance-related definitions for data terminal equipment are not specified, but the parameters defined in this clause may be employed in such definitions to assist users in establishing relationships between network performance and quality of service.

### 4.1 Call set-up delay

Call set-up delay refers to successful call set-up attempts and applies only to the virtual call capability of packet switched networks.

Call set-up delay observed at a single section boundary,  $B_i$ , is defined first. Call set-up delay between a pair of section boundaries ( $B_i$ ,  $B_j$ ) is then defined based on the former definition. In the former case, the call set-up delay includes the delay for all virtual connection sections on the called user side of  $B_i$  and the

called user response time. In the latter case, the call set-up delay includes only the delays between  $B_i$  and  $B_j$ .

#### 4.1.1 Definition of call set-up delay at a single section boundary

Call set-up delay at a section boundary,  $B_i$ , is defined using two packet layer reference events (PEs) specified in clause 3. It is the period of time that starts when either a call request or an incoming call packet creates a PE at  $B_i$ , and ends when the corresponding call connected or call accepted packet, accepting the virtual call, returns and creates its PE at  $B_i$ .

Call set-up delay at a section boundary =  $\{t_2 - t_1\}$  where

$t_1$  = Time of occurrence for the first PE

$t_2$  = Time of occurrence for the second PE

The two PEs can occur at any single section boundary within a virtual connection. The identities of the packets depend on the boundary of interest, as shown in figure 6. The first packet is the call request packet and the second packet is the corresponding call connected packet at every boundary except the two boundaries that delimit the access circuit section associated with the called DTE. The first packet is the incoming call packet and the second packet is the call accepted packet at the latter two boundaries. The specific PEs that are used in measuring call set-up delay at each section boundary are identified in table 3.

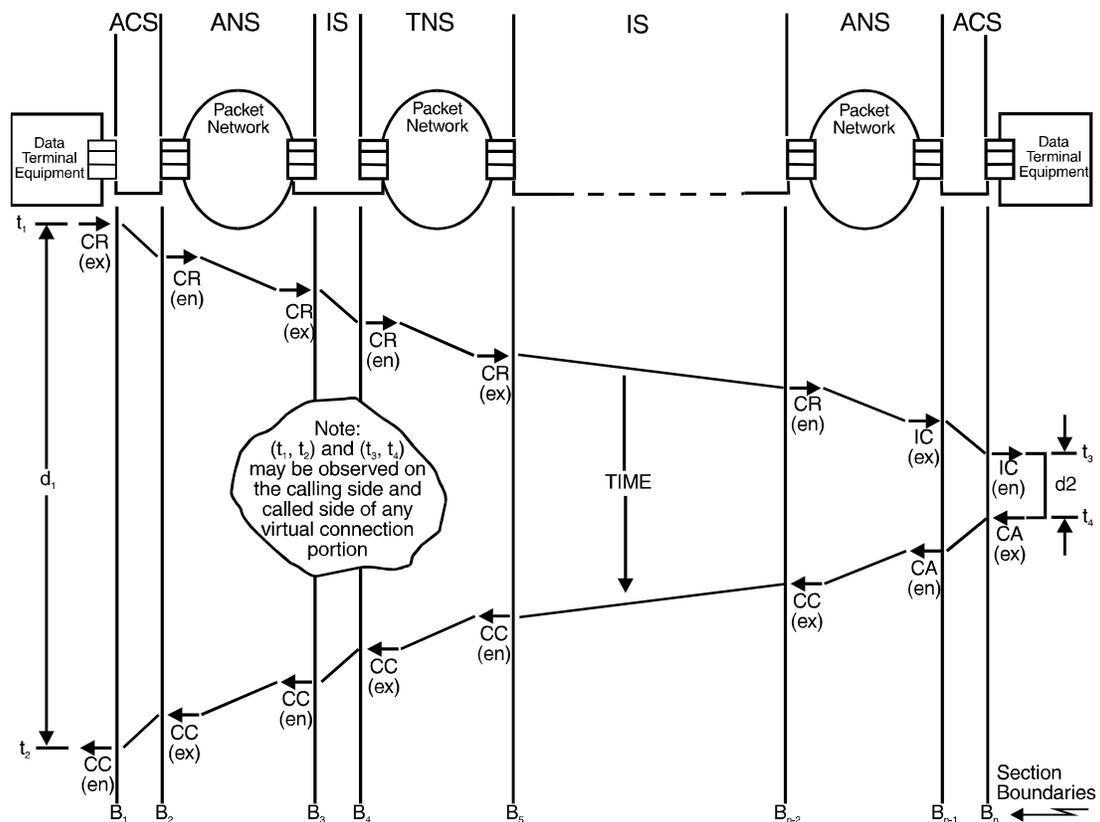


Figure 6 - Call set-up delay events

**Table 3 - Packet layer reference events (PEs) used in measuring call set-up delay**

Circuit Section	Starting PE	Ending PE
Access		
Calling DTE	2 (X.25)	3 (X.25)
Called DTE	1 (X.25)	4 (X.25)
Internetwork	1 (X.75)	2 (X.75)

NOTE - Refer to tables 1 and 2 for PE descriptions.

#### 4.1.2 Definition of call set-up delay between two section boundaries

For a particular virtual call, call set-up delay can be measured at one boundary,  $B_i$ , and at another boundary,  $B_j$ , further from the calling DTE. The difference in the values obtained is the call set-up delay contributed by the virtual connection section(s) between the two boundaries.

Call set-up delay between two section boundaries =  $(d_1 - d_2)$  where

$d_1$  = Call set-up delay measured at  $B_i$

$d_2$  = Call set-up delay measured at  $B_j$ .

The end-to-end call set-up delay is the call set-up delay between DTE boundaries (for example,  $B_1$  and  $B_n$  in figure 6). This end-to-end delay excludes the called user response time. The access circuit section call set-up delay is the call set-up delay between the boundaries delimiting an access circuit section (for example,  $B_1$  and  $B_2$  in figure 6). The internetwork circuit section call set-up delay is the call set-up delay between the boundaries delimiting an internetwork circuit section (for example,  $B_3$  and  $B_4$  in figure 6). The transit network section call set-up delay is the call set-up delay between the boundaries delimiting a transit network section (for example,  $B_4$  and  $B_5$  in figure 6). Under conditions of independence, the call set-up delay for a set of concatenated virtual connection sections can be calculated directly by adding the individual section delays.

#### 4.2 Data packet transfer delay

This delay refers to successful transfer of data packets and applies to both the virtual call and the permanent virtual circuit capabilities of packet switched networks. It is defined only between pairs of section boundaries.

Data packet transfer delay is the period of time that starts when a data packet creates a PE at a particular boundary,  $B_i$ , and ends when this same packet creates a later PE at another boundary,  $B_j$ . The specific PEs used in measuring data packet transfer delay at each section boundary are identified in table 4.

**Table 4 - Packet layer reference events (PEs) used in measuring data packet transfer delay and throughput**

Circuit Section	Starting/Ending PE
Access	
Source	10a (X.25)
Destination	9a (X.25)
Internetwork	5a (X.75)

NOTE - Refer to tables 1 and 2 for PE descriptions.

Data packet transfer delay =  $\{t_2 - t_1\}$  where

$t_1$  = Time of occurrence for the first PE

$t_2$  = Time of occurrence for the second PE.

The end-to-end data packet transfer delay is the one-way delay between DTE boundaries (for example,  $B_1$  and  $B_n$  in figure 6). The access circuit section data packet transfer delay is the delay between the boundaries delimiting an access circuit section (for example,  $B_1$  and  $B_2$  in figure 6). The internetwork circuit section data packet transfer delay is the delay between the boundaries delimiting an internetwork circuit section (for example,  $B_3$  and  $B_4$  in figure 6). The transit network section data packet transfer delay is the delay between the boundaries delimiting a transit network section (for example,  $B_4$  and  $B_5$  in figure 6). Under the condition of independence, the data packet transfer delay for a set of concatenated virtual connection sections can be calculated directly by adding the individual section delays.

### 4.3 Throughput parameters

The three throughput parameters, throughput, steady-state throughput, and throughput capacity, are defined as follows:

#### 4.3.1 Definition of throughput

Throughput for a virtual connection section is the number of user data bits successfully transferred in one direction across that section per unit time.<sup>2)</sup> Successful transfer means that no user data bits are lost, added, or inverted in transfer.

Assume that:

- 1) Data packet  $A_0$  is the final packet of a complete packet sequence crossing input boundary  $B_i$ .
- 2) Subsequently,  $k$  sequential data packets ( $A_1, A_2, \dots, A_k$ ) forming the next complete packet sequence cross the input boundary  $B_i$  immediately following  $A_0$ .
- 3) Data packet  $\hat{A}_0$  is the final packet of the first complete packet sequence when it crosses output boundary  $B_j$ .
- 4) Packets  $\hat{A}_1, \hat{A}_2, \dots, \hat{A}_m$  comprise the second complete packet sequence when it crosses output boundary  $B_j$ .

<sup>2)</sup> User data bits are the bits of the user data field in data packets of the X.25 or X.75 packet layer (this includes protocol information and all data above the packet layer). Framing, routing, bit stuffing, error control, and other protocol fields introduced by all protocols at or below the packet layer are excluded.

The PEs used in measuring throughput are the same as those used in measuring data packet transfer delay, as identified in table 4.

Let:

$t_1$  = Time of occurrence for the PE created by  $A_0$  at  $B_i$

$t_2$  = Time of occurrence for the PE created by  $A_k$  at  $B_i$

$t_3$  = Time of occurrence for the PE created by  $\hat{A}_0$  at  $B_j$

$t_4$  = Time of occurrence for the PE created by  $\hat{A}_m$  at  $B_j$

$f(A_r)$  = Number of user data bits in packet  $A_r$

Then a throughput measurement of size  $k$  is defined as follows:<sup>3)</sup>

$$\text{Throughput Measurement} = \frac{\sum_{r=1}^k f(A_r)}{\text{MAX}[(t_2 - t_1), (t_4 - t_3)]}$$

Clause 5 defines conditions under which a transfer of consecutive data packets is considered to be unsuccessful. Only successful throughput measurements should be included in the assessment of throughput performance.

#### 4.3.2 Definition of steady-state throughput

The steady-state throughput for a virtual connection is the value to which a throughput measurement converges as the duration of the observation period increases with statistically constant load<sup>4)</sup> on the virtual connection.

Assuming successful transfer, steady-state throughput is the same when measured at every pair of section boundaries of the virtual connection. Thus, assuming no user data bits are lost, added, or inverted in transfer, a steady-state throughput measurement can be made at any single section boundary within a virtual connection:

$$\text{Steady-state Throughput Measurement} = \frac{\sum_{r=1}^k f(A_r)}{(t_2 - t_1)},$$

where  $t_1$ ,  $t_2$ , and  $f(A_r)$  are defined above and  $k$  is sufficiently large.

Alternatively, the above equation can be used to calculate steady-state throughput with different definitions for  $t_1$  and  $t_2$ . Times  $t_1$  and  $t_2$  can be chosen in advance of the measurement. In this case, let  $(A_1, A_2, \dots, A_k)$  be the set of all virtual connection data packets crossing boundary  $B$  (creating PEs in one direction) at or following time  $t_1$  but before time  $t_2$ . Then the above equation still measures steady-state throughput.

#### 4.3.3 Definition of throughput capacity

Let  $B_i$  and  $B_j$  be two virtual connection section boundaries. Assume steady-state throughput is to be estimated with data packets flowing from  $B_i$  to  $B_j$ . Assume there is a statistically constant load,  $L$ , on the virtual connection section between  $B_i$  and  $B_j$ . The load,  $L$ , does not include the load on the virtual

<sup>3)</sup> In the case where throughput is measured by observation of exit events, a correction factor can be applied to the throughput measurement formula which will account for any difference in the size of  $A_0$  and  $A_k$  or  $\hat{A}_0$  and  $\hat{A}_m$ . See 7.4.3.

<sup>4)</sup> The phrase *statistically constant load* means a load that fluctuates, within limits set by the measurement precision objectives, about a stationary level.

connection being tested. Then the throughput capacity of that section under load,  $L$ , is defined as the steady state throughput maximized over all offered combinations of virtual connection parameter settings and choices for the performance and loading outside  $B_i$  and  $B_j$ . Measurement of throughput capacity for a section between boundaries  $B_i$  and  $B_j$  is accomplished in the same way as measurement of steady-state throughput. However, measurement of throughput capacity requires that the components outside of  $B_i$  and  $B_j$  have significantly higher throughput capacity under their respective loads than the throughput capacity being measured.

For the given statistically constant load,  $L$ , between  $B_i$  and  $B_j$ , and for a given set of testing arrangements, any measured steady-state throughput is a lower bound for the throughput capacity. To improve the estimate, the experiment may be repeated with different testing arrangements outside of  $B_i$  and  $B_j$ .

The end-to-end throughput capacity is the throughput capacity between DTE boundaries (for example,  $B_1$  and  $B_n$  in figure 6). The access circuit section throughput capacity is the throughput capacity between the boundaries delimiting an access circuit section (for example,  $B_1$  and  $B_2$  in figure 6). The internetwork circuit section throughput capacity is the throughput capacity between the boundaries delimiting an internetwork circuit section (for example,  $B_3$  and  $B_4$  in figure 6). The transit network section throughput capacity is the throughput capacity between the boundaries delimiting a transit section (for example,  $B_4$  and  $B_5$  in figure 6).

An upper bound for the throughput capacity of a set of concatenated virtual connection sections can be derived from the individual section throughput capacities as follows. If a section between boundaries  $B_i$  and  $B_j$  has throughput capacity,  $T_1$ , under load,  $L_1$ , and a section between boundaries  $B_k$  and  $B_m$  has throughput capacity  $T_2$  under load  $L_2$ , and those sections are concatenated so that  $B_j = B_k$ , with  $L_1$ , and  $L_2$  unchanged, then the resulting section's throughput capacity ( $T$ ) is bounded above by:

$$\text{MIN}[T_1, T_2] \geq T.$$

#### 4.4 Clear indication delay

Clear indication delay refers to successful call-clearing attempts and applies only to the virtual call capability of packet-switched networks. It is defined only between a pair of section boundaries.

Clear indication delay is the period of time that starts when either a clear request packet or a clear indication packet creates a PE at a boundary,  $B_i$ , and ends when the corresponding clear request or clear indication packet creates a later PE at another boundary,  $B_j$ . The specific PEs used in measuring clear indication delay at each section boundary are identified in table 5.

Clear Indication Delay =  $\{t_2 - t_1\}$ , where

$t_1$  = Time of occurrence for the first PE

$t_2$  = Time of occurrence for the second PE.

**Table 5 - Packet layer reference events (PEs) used in measuring clear indication delay**

Circuit Section	Starting/Ending PE
Access	
Clearing DTE	6 (X.25)
Cleared DTE	5 (X.25)
Internetwork	3 (X.75)

NOTE - Refer to tables 1 and 2 for PE descriptions.

The end-to-end clear indication delay is the one-way delay between DTE boundaries (for example,  $B_1$  and  $B_n$  in figure 6). The access circuit section clear indication delay is the delay between the boundaries delimiting an access circuit section (for example,  $B_1$  and  $B_2$  in figure 6). The internetwork section clear indication delay is the delay between the boundaries delimiting an internetwork section (for example,  $B_3$  and  $B_4$  in figure 6). The transit network section clear indication delay is the delay between the boundaries delimiting a transit network section (for example,  $B_4$  and  $B_5$  in figure 6). Under the condition of independence, the clear indication delay for a set of concatenated virtual connection sections can be calculated directly by adding the individual section delays.

## 5 Accuracy and dependability parameters

This clause of the standard specifies accuracy and dependability parameters that can be measured between any pair of boundaries delimiting a section or collection of sections. Performance definitions for data terminal equipment are not specified, but the parameters defined in this clause of the standard may be employed in such definitions to assist users in establishing relationships between network performance and quality of service.

### 5.1 Access parameters

Two access parameters, call set-up error probability and call set-up failure probability, are defined in this subclause.

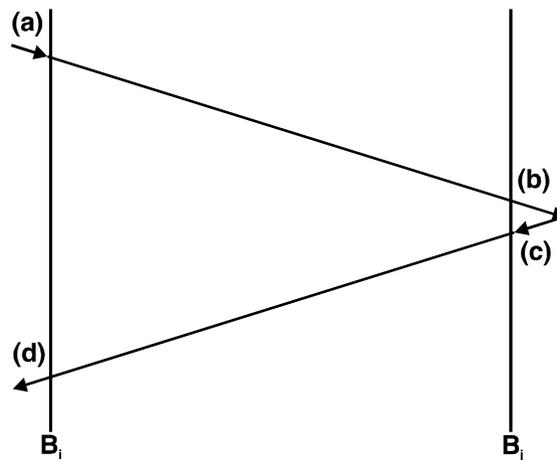
Call set-up error and call set-up failure are defined between pairs of section boundaries ( $B_i, B_j$ ).  $B_j$  is one of the set of boundaries to which the call attempt can properly be routed. Figure 7 identifies the sequence of four particular events that occur at these boundaries during a successful call set-up. A call set-up attempt over this section is an occurrence of event (a). A successful call set-up attempt over this section is a sequential occurrence of corresponding events (a, b, c, and d) within a 200-second time-out period.<sup>5)</sup> Call set-up errors and call set-up failures within this section are defined below. Any other unsuccessful call set-up attempt is caused by problems outside the section and is excluded from the measurement.

<sup>5)</sup> This period corresponds to timer T21 in ITU-T Recommendation X.25.

Interface	Boundary			
	$B_i$		$B_j$	
	Event (a)	Event (b)	Event (c)	Event (d)
X.25	2	3	1	4
X.75	1	2	1	2

NOTE - Refer to tables 1 and 2 for PE descriptions.

**(a) Packet layer reference events (PEs)**



**(b) Event sequence**

**Figure 7 - Packet layer reference events occurring during successful call set-up**

**5.1.1 Call set-up error probability**

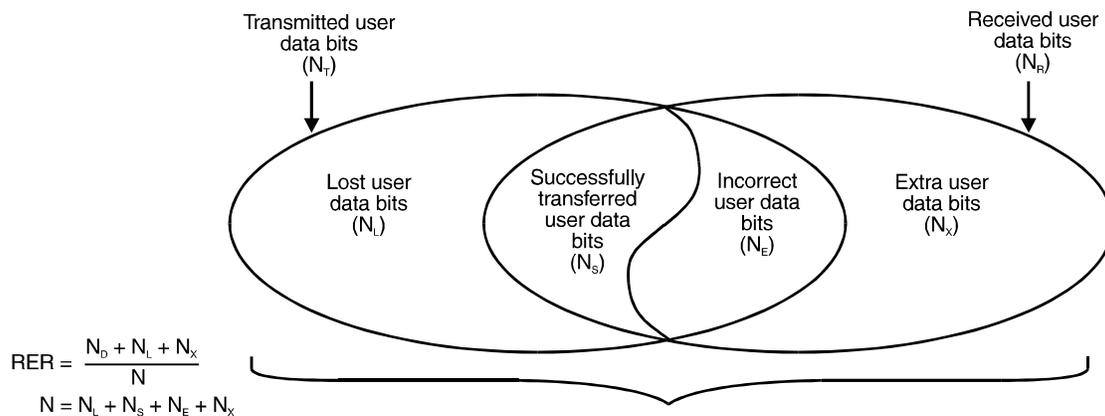
Call set-up error probability applies to virtual call services. It does not apply to permanent virtual circuit establishment. This parameter is used to measure the accuracy of the general user function of access in public packet switched services conforming to ITU-T Recommendations X.25 and X.75.

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

With reference to figure 7, a call set-up error is defined to occur on any call attempt in which event (d) occurs, but event (c) does not occur within a 200-second time-out period.

Call set-up error is essentially the case of a "wrong number." It occurs when the network responds to a valid call request by erroneously establishing a virtual call to a destination DTE other than the one designated in the call request, and does not correct the error prior to entry to the data transfer state. It may be caused, for example, by network operator administrative or maintenance actions.

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



**Figure 8 - Components of residual error ratio**

Call set-up error probability does not apply to the fast select mode of data transfer.

The optional user facilities in X.25 (including: hunt group, call redirection, call deflection subscription, call deflection selection, call redirection or deflection notification, called line address modified notification, and network user identifier) are assumed not to be used in the calculation of this parameter.

The specific PEs used in measuring call set-up error probability at each section boundary are those identified in figure 7.

### 5.1.2 Call set-up failure probability

Call set-up failure probability applies only to virtual call services. This parameter is used to measure the dependability of the general user function of access in public packet switched services conforming to ITU-T Recommendations X.25 and X.75.

#### 5.1.2.1 Definition of call set-up failure probability

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

With reference to figure 7, call set-up failure is defined to occur on any call attempt in which either one of the following outcomes is observed within a 200-second time-out period:<sup>6</sup>

- 1) Both events (b) and (d) do not occur.
- 2) Events (b) and (c) occur, but event (d) does not.

Call attempts that are cleared by the section as a result of incorrect performance or nonperformance on the part of an entity outside the section are excluded. The specific PEs used in measuring call set-up failure probability at each section boundary are those identified in figure 7.

<sup>6</sup> ITU-T Recommendation X.96 discusses limits on the frequency at which a DTE can repeat call attempts to a given destination.

### 5.1.2.2 Excluded call attempts

A call set-up attempt can also fail as a result of user blocking. Such failures are excluded from network performance measurement. Examples of user blocking include the following:

- 1) Either the originating or the called user issues a clear request to reject the call set-up attempt.
- 2) The called user delays excessively in generating the call accepted packet during the connection period, with the result that a connection is not established before the time-out.
- 3) All logical channels at the called DTE are in use.

## 5.2 User information transfer parameters

This subclause defines five user information transfer parameters: residual error ratio, reset stimulus probability, reset probability, premature disconnect stimulus probability, and premature disconnect probability. These parameters describe impairments observed during the data transfer state of a virtual call or permanent virtual circuit.

### 5.2.1 Residual error ratio

Residual error ratio<sup>7)</sup> applies to both virtual call and permanent virtual circuit services. This parameter is used to measure the accuracy of the general function of user information transfer in public packet-switched services conforming to ITU-T Recommendations X.25 and X.75.

#### 5.2.1.1 Definition of residual error ratio

Residual error ratio is the ratio of total incorrect, lost, and extra (for instance, duplicate) user data bits to total user data bits either transmitted or received.

Relationships among the quantities identified above are defined in figure 8. Incorrect user data bits are user data bits that are inverted in transfer between the section boundaries, that is, bits whose binary value observed at the section boundary on the destination side of a virtual connection section is the opposite of that observed at the section boundary on the source side. Lost user data bits are user data bits that are transferred into a virtual connection section at one section boundary, but are not transferred out of the virtual connection section at the other within 200 seconds of non-flow-controlled transmission. bits lost in association with a reset or premature disconnect are excluded in calculating residual error ratio. Extra user data bits are user data bits that are transferred out of a virtual connection section at one section boundary, but were not previously transferred into the virtual connection section at the other. Extra user data bits include duplicated user data bits and misdelivered user data bits.

The specific PEs used in measuring residual error ratio at each section boundary are identified in table 6. Only user data bits in data packets that create the specified PEs are counted in calculating residual error ratio estimates. In practice, it is not possible in all cases to distinguish lost, errored, and extra bit occurrences without detailed knowledge of the problems within the boundaries.

---

<sup>7)</sup> ITU-T has used the term *residual error rate*.

**Table 6 - Packet layer reference events (PEs) used in measuring residual error ratio**

Circuit Section	Starting/Ending PE
Access	
Source	10a (X.25)
Destination	9a (X.25)
Internetwork	5a (X.75)

NOTE - Refer to tables 1 and 2 for PE descriptions.

### 5.2.1.2 Components of residual error ratio

In some applications, it may be important to specify probability limits for the individual failure outcomes illustrated in figure 8 in addition to the overall residual error ratio. The general user information error, user information loss, and extra user information delivery probabilities defined in ITU-T Recommendation X.140 may be specialized to the corresponding user-data-bit-oriented measures as follows:

- 1) User data bit error probability  $P_1(E)$  is the ratio of total incorrect user data bits ( $N_E$ ) to total successfully transferred user data bits plus incorrect user data bits ( $N_S + N_E$ ) in a population of interest.
- 2) User data bit loss probability  $P_1(L)$  is the ratio of total lost user data bits ( $N_L$ ) to total transmitted user data bits ( $N_T$ ) in a population of interest.
- 3) Extra user data bit delivery probability  $P_1(X)$  is the ratio of total (unrequested) extra user data bits ( $N_X$ ) to total received user data bits ( $N_R$ ) in a population of interest.

The denominators of these ratios are chosen to ensure that each defined probability is properly normalized; that is, each failure outcome is expressed in proportion to the total number of opportunities for that outcome to occur. The mathematical relationship between residual error ratio (RER) and the three user data bits transfer failure probabilities defined in this subclause are as follows:

$$RER = \frac{[P_1(E)][N_E + N_S] + [P_1(L)][N_T] + [P_1(X)][N_R]}{N}$$

### 5.2.2 Reset parameters

Reset stimulus probability and reset probability are related parameters used to describe the dependability of the general function of user information transfer in public packet-switched services conforming to ITU-T Recommendations X.25 and X.75.

#### 5.2.2.1 Definition of reset stimulus probability

A reset stimulus is observed at a single section boundary. It is any event or combination of events that according to the protocol should result in a reset (or, in the case of a PVC, a reset or restart) being generated by the recipient.<sup>8)</sup> An example of a reset stimulus is a DTE transmitting a reject packet when the packet retransmission facility has not been subscribed.

The reset stimulus probability of a section at a boundary is the probability of a reset stimulus generated within that section and transferred across the boundary per virtual connection second.

<sup>8)</sup> For the purpose of performance parameter definition, it is assumed that the reset stimuli for an X.25 DTE are equivalent to the reset stimuli for an X.25 DCE.

### 5.2.2.2 Definition of reset probability

A reset event is defined to have been generated within a section when, in the absence of an external reset stimulus, two packets exit the section - one at each boundary - creating any one of the pairs of packet layer reference events listed in table 7.

**Table 7 - Pairs of PEs resulting from reset events**

<b>Boundaries of Section</b>	<b>Pair of PEs</b>
X.25 - X.25	[20(X.25), 20(X.25)]
X.25 - X.75	[20(X.25), 10(X.75)]
X.75 - X.75	[10(X.75), 10(X.75)]
<i>Additional PE Pairs Resulting from Reset Events on PVCS*</i>	
X.25 - X.25	[20(X.25), 24(X.25)]
X.25 - X.75	[20(X.25), 12(X.75)] or [24(X.25), 10(X.75)]
X.75 - X.75	[10(X.75), 12(X.75)]
NOTE - Refer to tables 1 and 2 for PE descriptions.	
*The restart packets associated with events 24(X.25) and 12(X.75) are not uniquely associated with the PVC.	

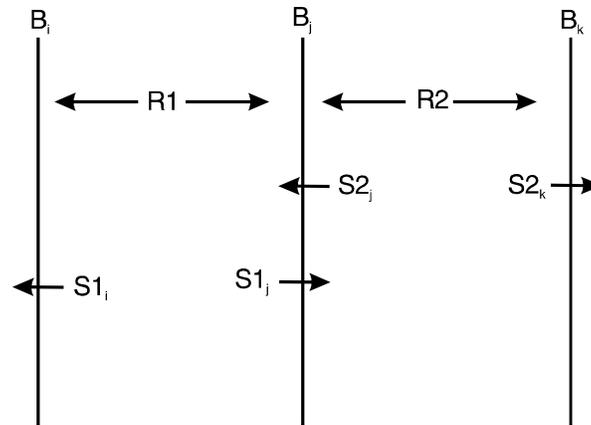
The reset probability for a virtual connection section is the probability, in any given second, that a reset event is generated within that section.

Reset events generated within a section may be estimated by counting the number of reset request and reset indication packets exiting the section during a measurement period; subtracting the number of reset request and reset indication packets entering the section during the same period; dividing the difference by 2; and then subtracting from the result any reset stimuli that enter the section during the period.

NOTE - Reset events may be associated with a loss of packets.

The specific PEs used in measuring reset probability at each section boundary are identified in table 7.

The reset stimulus and reset probabilities for a set of concatenated virtual connection sections may be estimated from the individual section probabilities as follows. Assume between boundaries ( $B_i$ ,  $B_j$ ) the reset probability is  $R1$  and the reset stimulus probabilities are  $S1_i$ ,  $S1_j$ . Assume between boundaries ( $B_j$ ,  $B_k$ ) the reset probability is  $R2$  and the reset stimulus probabilities are  $S2_j$ ,  $S2_k$ . Then on a VC passing through  $B_j$  the reset probability between  $B_j$  and  $B_k$  is approximately  $(R1 + R2 + S1_j + S2_j)$ . The reset stimulus probability at  $B_i$  is  $S1_i$  and the reset stimulus probability at  $B_k$  is  $S2_k$ . See figure 9.



**Figure 9 - Reset stimulus and reset probabilities**

### 5.2.3 Premature disconnect parameters

Premature disconnect stimulus probability and premature disconnect probability are related parameters used to describe the dependability of user information transfer in public packet switched networks conforming to ITU-T Recommendations X.25 and X.75. These parameters apply only to the virtual call capabilities of virtual call services.

#### 5.2.3.1 Definition of premature disconnect stimulus probability

A premature disconnect stimulus is observed at a single section boundary. It is any event or combination of events that according to the protocol should result in a clear or restart being generated by the recipient.<sup>9</sup> An example of a premature disconnect stimulus is the transmission of an incorrect packet type into a virtual connection section.

The premature disconnect stimulus probability of a section at a boundary is the probability of a premature disconnect stimulus generated within that section and transferred across the boundary per virtual connection second.

#### 5.2.3.2 Definition of premature disconnect probability

A premature disconnect event is defined to have been generated within a section when, in the absence of an external premature disconnect stimulus, two packets exit the section - one at each boundary - creating any one of the pairs of packet layer reference events listed in table 8.

<sup>9)</sup> For the purpose of performance parameter definition, it is assumed that the premature disconnect stimuli for an X.25 DTE are equivalent to the premature disconnect stimuli for an X.25 DCE.

**Table 8 - Pairs of PEs resulting from premature disconnect events**

<b>Boundaries of Section</b>	<b>Pair of PEs*</b>
X.25 - X.25	[5(X.25), 5(X.25)] or [5(X.25), 24(X.25)]
X.25 - X.75	[5(X.25), 3(X.75)] or [5(X.25), 12(X.75)] or [24(X.25), 3(X.75)]
X.75 - X.75	[3(X.75), 3(X.75)] or [13(X.75), 12(X.75)]

NOTE - Refer to tables 1 and 2 for PE descriptions.

\*The restart packets associated with events 24(X.25) and 12(X.75) are not uniquely associated with the virtual call.

The premature disconnect probability for a virtual connection section is the probability, in any given second, that a virtual call experiences a premature disconnect event generated within that section.

Premature disconnect events generated within a section may be estimated by counting the number of clear request or clear indication packets exiting the section during a measurement period; subtracting the number of clear request and clear indication packets entering the section during the same period; dividing the difference by two; and then subtracting from the result any premature disconnect stimuli that enter the section during that period.<sup>10)</sup>

The specific PEs used in measuring premature disconnect probability at each section boundary are identified in table 8.

The premature disconnect stimulus and premature disconnect probabilities for a set of concatenated virtual connection sections may be estimated from the individual section probabilities in a manner analogous to that described in 5.2.2.2.

### **5.3 Disengagement performance - call clear failure probability**

Call clear failure probability applies only to virtual call services. This parameter is used to measure the accuracy and dependability of the general function of disengagement in public packet-switched services conforming to Recommendations X.25 and X.75.

#### **5.3.1 Definition of call clear failure probability**

Call clear failure is defined with reference to events at the boundaries of a virtual connection section ( $B_i$ ,  $B_j$ ). A call clear attempt occurs when a call clear request or clear indication packet enters the section creating a packet layer reference event at  $B_i$ . A call clear failure occurs when no corresponding clear indication packet layer reference event occurs at  $B_j$  within 180 seconds. The relevant PEs are listed in table 9.

<sup>10)</sup> Premature disconnect events may be associated with a loss of packets.

**Table 9 - Packet layer reference events (PEs) used in measuring call clear failure probability**

Circuit Section	Packet Layer Reference Event	
	Starting PE	Ending PE
Access		
Clearing DTE	6(X.25)	—
Cleared DTE	—	5(X.25) (does not occur)
Internetwork	3(X.75)	3(X.75) (does not occur)

NOTE - Refer to tables 1 and 2 for the PE descriptions.

Call clear failure probability for a virtual connection section is the ratio of call clear failures to call clear attempts in a population of interest.

### 5.3.2 Local clear confirmation

The failure of a section to respond to a clear request or clear indication packet with a clear confirmation packet is not addressed in this standard. Recovery mechanisms for such occurrences are defined in both the protocols of ITU-T Recommendations X.25 and X.75.

## 6 Availability

This clause specifies availability parameters for virtual connection sections as defined in clause 3. Performance objectives for data terminal equipment are not specified, but the parameters defined in this clause of the standard may be employed in such a specification to assist users in establishing quantitative relationships between network performance and quality of service.

A two-state model provides a basis for describing overall service availability. A specified availability function compares the values for a set of "supported" primary parameters with corresponding outage thresholds to classify the service as "available" (no service outage) or "unavailable" (service outage). This clause specifies the availability function and defines the availability parameters that characterize the resulting binary random process.

Two availability parameters are defined in this clause: service availability and mean time between service outages. Each parameter can be applied to any basic section of a virtual connection. This generality makes the parameters useful in performance allocation and concatenation.

### 6.1 Availability function

Eight performance parameters from clauses 3 and 4 are used in computing the availability of a virtual connection: throughput capacity (clause 3), call set-up failure probability (clause 4), call set-up error probability (clause 4), residual error ratio (clause 4), reset probability (clause 4), reset stimulus probability (clause 4), premature disconnect probability (clause 4), and premature disconnect stimulus probability (clause 4). Five particular linear combinations of these parameters are called the availability decision parameters. Each decision parameter is associated with an outage threshold. These decision parameters and their outage thresholds are listed in table 10.

Performance is considered independently with respect to each availability decision parameter. If the value of the parameter is equal to or better than the defined outage threshold, performance relative to that

parameter is defined to be acceptable. If the value of the parameter is worse than the threshold, performance relative to that parameter is defined to be unacceptable.

The packet layer reference events that are used in defining the decision parameters do not occur if a data link layer at a section boundary is unavailable. During a continuous time interval, the data link layer of a circuit section is defined to be available for packet layer service if and only if:

- 1) The link is in the information transfer phase for at least 99% of the time interval.
- 2) All continuous periods when the link is not in the information transfer phase are less than 1 second in length.
- 3) All continuous busy (flow-controlled) conditions are less than 10 seconds in length.

Otherwise the data link layer is considered unavailable for providing packet layer service.

The data link layer of a circuit section can be unavailable for any of the following reasons:

- 1) A nonfunctional physical circuit
- 2) A data link layer controller either unable or unwilling to establish the information transfer phase
- 3) A data link layer controller either unable or unwilling to clear a busy condition.

A virtual connection section is defined to be available (or to be in the available state) if:

- 1) The performance is acceptable relative to all decision parameters
- 2) Both data link layers at the boundaries of the section are available.

The virtual connection section is defined to be unavailable (or in the unavailable state) if:

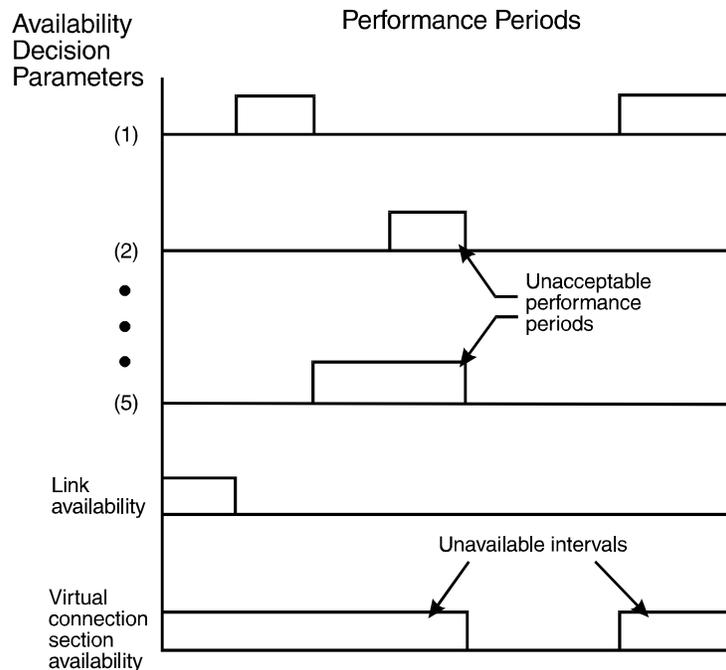
- 1) The performance of one or more of the five decision parameters is unacceptable, or
- 2) One or both of the data link layers at the boundaries of the section are unavailable due to causes inside the section. (Data link layer unavailability due to causes outside the section are excluded, that is, failures of link controllers or physical circuits outside the section in question.)

**Table 10 - Outage criteria for the availability decision parameters**

Availability Decision Parameters		Criteria*
Call set-up failure probability (cfp)	}	$cfp + cep > C_1$
Call set-up error probability (cep)		
Throughput capacity (tc)		$tc < C_2$
Residual error ratio (rer)		$rer > C_3$
Reset probability (rp)	}	$rsp_1 + rp + rsp_2 > C_4$
Reset stimulus probability (rsp 1, rsp 2)		
Premature disconnect probability (pdp)	}	$pdsp_1 + pdp + pdsp_2 > C_5$
Premature disconnect stimulus probability (pdsp <sub>1</sub> , pdsp <sub>2</sub> )		

\* The threshold constants  $C_1 - C_5$  are for further study. ITU-T Recommendation X.137 assigns provisional values to these constants as follows:  
 $C_1 = 0.9, C_2 = 80, C_3 = 10^{-3}, C_4 = 0.015, C_5 = 0.01.$

The intervals during which a virtual connection section is unavailable are identified by superimposing the unacceptable performance periods for all decision parameters as illustrated in figure 10.



**Figure 10 - Determination of availability states**

In order to exclude transient impairments from being considered as periods of unavailability, a single test of the availability state must exceed 5 minutes. In order to reduce the probability of state transitions during a test of the current availability state, that test should be less than 20 minutes.

## 6.2 Availability parameters

Two availability parameters are defined in this subclause: service availability and mean time between service outages.

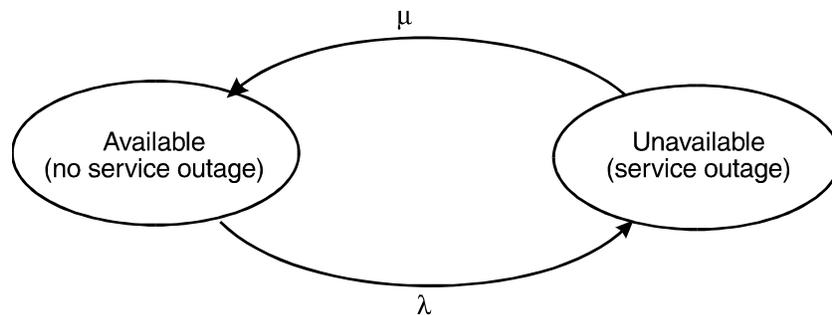
### 6.2.1 Definition of service availability

Service availability applies to both virtual call and permanent virtual circuit services. The service availability for a virtual connection section is the long-term percentage of scheduled service time in which that section is available.

Scheduled service time for a virtual connection section is the time during which the network provider has agreed to make that section available for service. The normal objective for scheduled service is 24 hours per day, 7 days a week.<sup>11)</sup>

### 6.2.2 Definition of mean time between service outages

Mean time between service outages applies to both virtual call and permanent virtual circuit services. The mean time between service outages for a virtual connection section is the average duration of any continuous interval during which the virtual connection section is available. Consecutive intervals of scheduled service time are concatenated.



**a. State diagram**

$$MTBSO = \frac{1}{\lambda} \qquad MTTSR = \frac{1}{\mu}$$

$$A = 100 \left[ \frac{MTBSO}{MTBSO + MTTSR} \right] = 100 \left[ \frac{\mu}{\lambda + \mu} \right]$$

$$U = 100 - A = 100 \left[ \frac{MTTSR}{MTBSO + MTTSR} \right] = 100 \left[ \frac{\lambda}{\lambda + \mu} \right] g$$

**b. Parameter relationships**

**Figure 11 - Basic availability model and parameters**

<sup>11)</sup> Other scheduled service times may be specified in some networks.

### 6.2.3 Related parameters

Four other parameters are commonly used in describing availability performance. These are generally defined as follows:

- 1) Mean time to service restoral (MTTSR) is the average duration of unavailable service time intervals.
- 2) Failure rate ( $\lambda$ ) is the average number of transitions from the available state to the unavailable state per unit available time.
- 3) Restoral rate ( $\mu$ ) is the average number of transitions from the unavailable state to the available state per unit unavailable time.
- 4) Unavailability (U) is the long-term ratio of unavailable service time to scheduled service time, expressed as a percentage.

Under the exponential distribution assumption of failure and restoration, the mathematical values for any of these parameters may be estimated from the values for service availability (A) and mean time between service outages as summarized in figure 11.

## 7 Requirements and example procedures for performance measurement

This clause provides information on measurement objectives, test architectures, test procedures, and reporting requirements for measurement of data communication performance. Subclause 7.1 describes typical measurement objectives. Subclause 7.2 describes test architectures and test equipment configurations used to make the measurements. Subclause 7.3 provides example data extraction and data reduction test procedures for each of four types of performance trials (access, data transfer, disengagement, and availability) used to estimate the performance parameters and gives acceptable approximations for estimating the residual error ratio and availability parameters. Subclause 7.4 identifies commonly used statistics for each parameter and describes measurement conditions that must be reported to ensure proper interpretation of the measurement results.

### 7.1 Measurement objectives

Three common objectives of performance measurement are discussed in this subclause. These are all presented in the context of single-variable statistics. Other objectives, even within such a context, can exist and these examples are not meant to be all-encompassing.

This subclause, by its discussion of various test architectures and test procedures, is designed to facilitate the data collection phase of appropriate experimental designs and to acquaint those responsible for test planning with various issues that could influence the choice of the design.

As there is always a cost versus sample size trade-off that must be resolved when attempting to estimate a given parameter, and as the cost of taking an observation can have a considerable effect on the final estimate precision achieved, this standard does not recommend a minimum number of observations. Additionally, in certain situations, estimating the parameter with great precision may not be as critical as in others.

#### 7.1.1 Parameter estimation

Estimates of means, variances, percentiles, modes, maxima, and minima are all examples of parameter estimation. In reporting any such estimated parameters it is always recommended that a measure of the precision of the estimate be included. This may take the form of some estimate of the variance of the estimate, or of a confidence interval about the estimate.

## 7.1.2 Hypothesis testing

Examples of uses of hypothesis test are given in 7.1.2.1 through 7.1.2.3 together with appropriate references to other T1 documents. The test may be either a fixed sample size or a sequential test (see Lehmann, et al.).

### 7.1.2.1 Acceptance testing

Acceptance of packet service may depend on demonstrating a given level of performance. This level of performance can be demonstrated by obtaining measurements of the specified performance parameters and performing a hypothesis test to determine whether the performance levels are acceptable.

Typically, these are one-sided hypothesis tests. An example test for data packet transfer delay is as follows: the null hypothesis ( $H_0$ ) is that the delay is acceptable, as it will be in most cases, while the alternative hypothesis ( $H_a$ ) is that the delay is too great and therefore unacceptable.

- 1)  $H_0$ : Data packet transfer delay mean is  $\leq x$  milliseconds;
- 2)  $H_a$ : Data packet transfer delay mean is  $> x$  milliseconds;
- 3)  $H_0$  and  $H_a$  will be reversed in cases where a higher parameter value is better (e.g., service availability).

### 7.1.2.2 Maintenance

Given that packet service has been accepted at a particular level of performance, service providers may want to establish maintenance limits. Such limits are thresholds of performance at which the provider takes action to restore performance that has degraded to less than acceptable levels. In the following test of hypotheses, the  $x$  refers to the value given in the acceptance test of 7.1.2.1, and rejecting the null hypothesis (no maintenance needed) would be grounds for performing maintenance.

- $H_0$ : Data packet transfer delay mean is  $\leq x+y$  milliseconds;  
 $H_a$ : Data packet transfer delay mean is  $> x+y$  milliseconds;

where ( $x > y > 0$ ).

### 7.1.2.3 Conformance of data to a particular distribution

In certain instances it is important to determine whether or not a set of measurements conforms reasonably well to a particular distribution. This type of test is important in determining whether an assumption about the distribution of a certain type of measurement is correct. In the test given below, the null hypothesis is that the distribution is uniform on the closed interval from 0 to 10.

- 1)  $H_0$ : The distribution of data packet transfer delay is uniform (0, 10);
- 2)  $H_a$ : Data packet transfer is not uniform (0, 10).

## 7.1.3 Pairwise and multiple comparisons

Pairwise and multiple comparisons are useful in assessing the effect of a factor or combination of factors on observed performance. A series of pairwise comparisons of means is not equivalent to a simultaneous comparison of all the means. Thus the conclusions that mean A is not significantly different from mean B, and mean C is not significantly different from mean B, do not necessarily lead one to the conclusion that means A and C are not significantly different from one another, and certainly not at the same level of significance. The theory of multiple comparisons may be found in statistical textbooks.

## 7.2 Measurement architectures

Measurements of packet-switched networks require a *source*, a *sink*, and one or more *monitors*. A source transmits call set-up requests, data packets, or call-clearing requests through the sections under test. A sink receives and acknowledges call processing or data packets from the sections under test. The monitor records (or records and time-stamps) the relevant reference events. The monitor function(s) should be placed as near as possible to the boundaries of the sections under test. Distance between the monitor functions and the appropriate boundaries must be compensated for in the calculation of the performance of the sections.

Sources and sinks can be either *controlled* or *noncontrolled*. Controlled sources and sinks are under the control of the test program and must respond quickly to events generated within the sections under test. Examples of controlled sources or sinks are stand-alone test equipment, special software within network equipment (e.g., within packet switches), and special programs within customer applications. Noncontrolled sources or sinks are sources and sinks not under the direct control of a test program. Noncontrolled sources and sinks may not always respond quickly to network events. The most important examples of noncontrolled sources and sinks are live customer applications, generating and receiving traffic according to application needs.

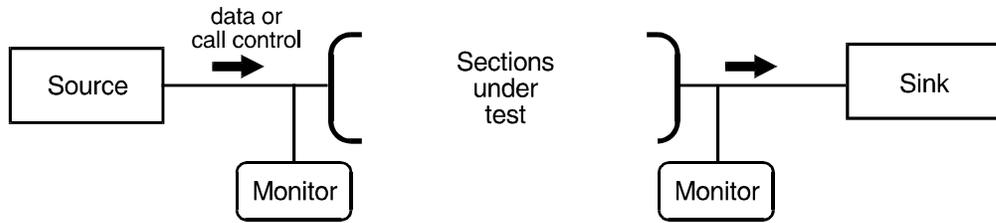
A monitor function can be provided by stand-alone test equipment "T"-connected at the appropriate X.25 or X.75 interface. Alternatively, a monitor function can reside in the test device that provides the source or sink function. Network equipment (such as packet switches) and customer equipment (e.g., DTEs) can also be programmed to record reference events and serve the monitor function.

Various combinations of monitors and controlled and noncontrolled sources and sinks can be used to measure packet network performance. Figure 12 illustrates some of these possibilities. The architectures are identified by specifying whether the source and sink are controlled (C) or noncontrolled (N), and whether the two section boundaries are monitored (M) or unmonitored (U). The notation (C,M/N,U) indicates a controlled source, a noncontrolled sink, monitoring at the source boundary, and no monitoring at the sink boundary. When both the source and the sink are controlled and there are time-synchronized monitor functions at both boundaries (C,M/C,M), all of the parameters defined in this standard can be measured without further assumptions. Other test architectures are limited because they cannot be used to measure all of the parameters. Several methods for synchronizing clocks in remote test sets are described in Howe, S. L.; Kamas, G. et al; and Levine, J. et al.

Subclauses 7.2.1 through 7.2.10 list the primary performance parameters and identify the test architectures that can be used to measure these parameters.<sup>12)</sup> In some cases the test architectures can be used to measure a parameter only if certain additional assumptions described in the relevant subclause are true.

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<sup>12)</sup> A loopback may be available as an alternative measurement architecture as described in 7.2.9.

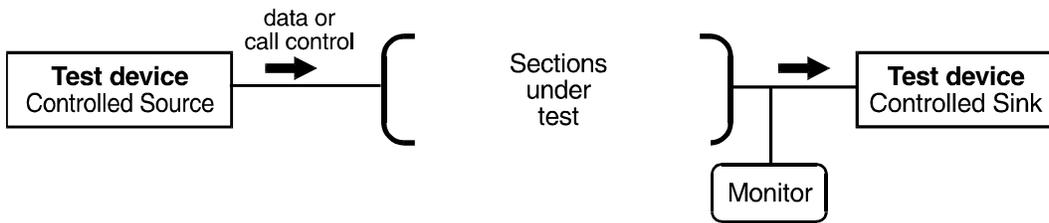


(a) Generic test architecture

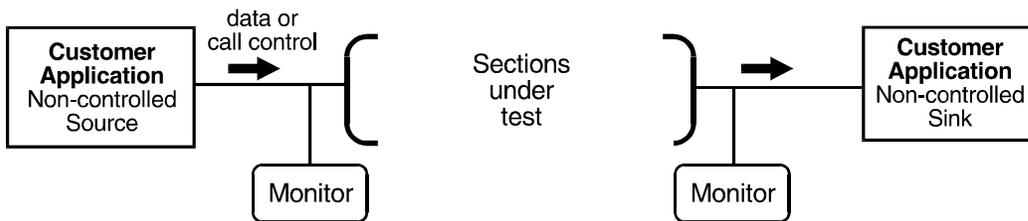


(Source and sink devices combined with monitors)

(b) C, M/C, M architecture



(c) C, U/C, M architecture



(d) N, M/N, M architecture

**Figure 12 - Measurement architectures**

### **7.2.1 Call set-up delay**

Call set-up delay (CSD) can best be measured with monitors at both section boundaries. If the sink is known to accept call set-up requests with constant or insignificant delay and if the probability of call set-up error events is insignificant, CSD can be measured without a sink-side monitor by subtracting the known sink delay from the one-sided measurement.

### **7.2.2 Data packet transfer delay and clear indication delay**

Both data packet transfer delay (DPTD) and clear indication delay (CID) require a sink-side monitor time synchronized with either a source-side monitor or with the source itself.

### **7.2.3 Throughput capacity**

Throughput capacity (TC) measurements require controlled sources and sinks rapidly transmitting and acknowledging data packets. In effect, the TC of the source and sink must be greater than or equal to the TC of the sections under test.

### **7.2.4 Call set-up error probability**

Call set-up error probability (CSEP) can only be measured if there is monitoring at both section boundaries. Without this monitoring, the events identified in figure 7 cannot be observed.

### **7.2.5 Call set-up failure probability**

Call set-up failure probability (CSFP) can best be measured with monitors at both boundaries. The sink device must be fast enough so that it does not significantly contribute to the probability of timing out. If the sink can be relied upon to accept every call set-up request, CSFP can be measured without a monitor at the sink side.

### **7.2.6 Residual error ratio**

Residual error ratio (RER) requires monitoring at both boundaries or a controlled source transmitting a known sequence of user data. Both of these architectures allow the transmitted and received user data to be compared.

### **7.2.7 Reset and premature disconnect performance**

Reset stimulus probability (RSP) and premature disconnect stimulus probability (PDSP) can be measured by a single monitor at a single boundary. Reset probability (RP) and premature disconnect probability (PDP) require monitors at both boundaries. These allow for distinguishing the resets and clears that exit the sections under test from the resets and clears stimulated at the distant boundary.

### **7.2.8 Call clearing failure probability**

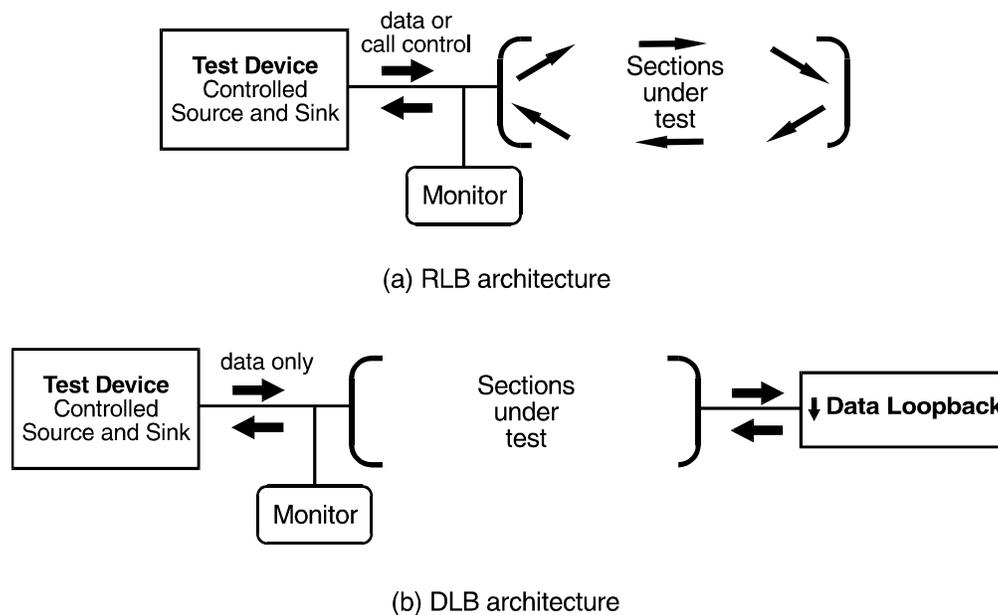
Call clearing failure probability (CCFP) can be measured with monitors at both boundaries. If the transmission of the clear request by a controlled source is reasonably well-synchronized with the sink-side monitor, the monitor can anticipate that clearing and observe call clearing failures.

## 7.2.9 Loopbacks

*Loopbacks* provide an alternative measurement architecture allowing a single test device to serve as both a source and a sink. Figure 13 illustrates the two possibilities for using loopbacks in packet-switched network performance measurements.<sup>13)</sup>

*Routing loopbacks* (RLB) are established by the packet network(s) by routing virtual circuits through one or more switching functions (or through multiple networks) back to the originating interface. The result is a virtual circuit that originates on one logical channel and terminates on a different logical channel on the same test device. A monitor at the section boundary can then be used to measure all of the primary performance parameters. If the routing loopback is significantly different from an ordinary virtual connection through the sections (e.g., in the number of switches or distance traversed), the performance calculations must compensate for those differences.

*Data loopbacks* (DLB) can be used to measure DPTD, TC, and RER. A DLB can be provided by special software within network equipment, by stand-alone test equipment, or by special test programs within customer applications. A DLB device terminates the physical, link, and packet layer of a virtual connection, removes the user data from the incoming data packets, and returns that data through the same virtual connection in new outgoing data packets. The DLB should rapidly acknowledge data packets and return the user data without significant delay or error. If sections under test have symmetric delays and residual errors, DPTD and RER are half of what is calculated by comparing outgoing and incoming data packets at a source-side monitor. TC measurements made at a monitored boundary yield the smaller value of TC for the two directions.



**Figure 13 – Loopback architectures**

<sup>13)</sup> Because of the asymmetry of the X.25 protocols, bit-by-bit, frame-by-frame, and packet-by-packet loopbacks are not appropriate for measuring packet-switched networks.

## 7.2.10 Summary

The possible combinations of controlled and noncontrolled sources and sinks and monitored and unmonitored boundaries yield 12 different measurement architectures. Routing loopbacks and data loopbacks create two more possible architectures. In table 11, the 14 architectures are listed with an indication of their ability to measure each primary parameter.

**Table 11 – Summary of measurement architectures**

Measurement architecture	Primary parameters											
	CSD	DPTD	TC	CID	CSEP	CSFP	RER	RSP	RP	PDSP	PDP	CCFP
C,M/C,M	Y	Y <sup>1)</sup>	Y	Y <sup>1)</sup>	Y	Y	Y	Y	Y	Y	Y	Y
N,M/C,M	Y	Y <sup>1)</sup>	N	Y <sup>1)</sup>	Y	Y	Y	Y	Y	Y	Y	Y
C,M/N,M	Y <sup>2)</sup>	Y <sup>1)</sup>	N	Y <sup>1)</sup>	Y	Y <sup>2)</sup>	Y	Y	Y	Y	Y	Y
N,M/N,M	Y <sup>2)</sup>	Y <sup>1)</sup>	N	Y <sup>1)</sup>	Y	Y <sup>2)</sup>	Y	Y	Y	Y	Y	Y
C,U/C,M	N	Y <sup>3)</sup>	Y	Y <sup>3)</sup>	N <sup>4)</sup>	N <sup>4)</sup>	Y <sup>5)</sup>	Y <sup>6)</sup>	N <sup>7)</sup>	Y <sup>6)</sup>	N <sup>7)</sup>	Y <sup>3)</sup>
C,U/N,M	N	Y <sup>3)</sup>	N	Y <sup>3)</sup>	N <sup>4)</sup>	N <sup>4)</sup>	Y <sup>5)</sup>	Y <sup>6)</sup>	N <sup>7)</sup>	Y <sup>6)</sup>	N <sup>7)</sup>	Y <sup>3)</sup>
C,M/C,U	Y <sup>8)</sup>	N	Y	N	N	Y <sup>9)</sup>	N	Y <sup>6)</sup>	N <sup>7)</sup>	Y <sup>6)</sup>	N <sup>7)</sup>	N
N,M/C,U	Y <sup>8)</sup>	N	N	N	N	Y <sup>9)</sup>	N	Y <sup>6)</sup>	N <sup>7)</sup>	Y <sup>6)</sup>	N <sup>7)</sup>	N
N,M/N,U	N <sup>10)</sup>	N	N	N	N	Y <sup>9)</sup>	N	Y <sup>6)</sup>	N <sup>7)</sup>	Y <sup>6)</sup>	N <sup>7)</sup>	N
C,M/N,U	N <sup>10)</sup>	N	N	N	N	Y <sup>9)</sup>	N	Y <sup>6)</sup>	N <sup>7)</sup>	Y <sup>6)</sup>	N <sup>7)</sup>	N
N,U/N,M	N	N	N	N	N	N	N	Y <sup>6)</sup>	N <sup>7)</sup>	Y <sup>6)</sup>	N <sup>7)</sup>	N
N,U/C,M	N	N	N	N	N	N	N	Y <sup>6)</sup>	N <sup>7)</sup>	Y <sup>6)</sup>	N <sup>7)</sup>	N
RLB	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y
DLB	–	Y <sup>11)</sup>	Y <sup>13)</sup>	–	–	–	Y <sup>12)</sup>	Y <sup>6)</sup>	N <sup>7)</sup>	Y <sup>6)</sup>	N <sup>7)</sup>	–

1) Assumes the two monitors are time synchronized.  
2) Assumes the noncontrolled sink is reasonably fast in responding.  
3) Assumes the source's creation of packets is time synchronized with the sink-side monitor.  
4) Cannot observe event (d), figure 7.  
5) Assumes the user data created by the source is known in advance.  
6) At the monitored boundary only.  
7) Cannot distinguish between events originating within the sections and events caused by stimuli at the distant boundary.  
8) Assumes there are no call set-up error events and the sink accepts the call with known or insignificant delay.  
9) Assumes no unsuccessful call set-up attempts due to the sink devices.  
10) Cannot exclude delays due to the sink devices.  
11) Assumes DPTD is equal in both directions, and the loopback introduces a known or insignificant delay.  
12) Assumes RER is equal in both directions.  
13) Measures the lesser TC of the two directions.

CSD – Call Set-Up Delay	RER – Residual Error Ratio
DPTD – Data Packet Transfer Delay	RSP – Reset Stimulus Probability
TC – Throughput Capacity	RP – Reset Probability
CID – Clear Indication Delay	PDSP – Premature Disconnect Stimulus Probability
CSEP – Call Set-Up Error Probability	PDP – Premature Disconnect Probability
CSFP – Call Set-Up Failure Probability	CCFP – Call Clear Failure Probability

### 7.3 Test procedures

Values for the primary parameters defined in this standard are obtained using three types of performance trials: access, data transfer, and disengagement. Values for call set-up delay, call set-up error probability, and call set-up failure probability are obtained using access trials. Values for data packet transfer delay, throughput capacity, residual error ratio, reset stimulus probability, reset probability, premature disconnect stimulus probability, and premature disconnect probability are obtained using data transfer trials. Clear indication delay and call clear failure probability values are obtained using disengagement trials. The availability parameters – service availability and mean time between service outages – are measured using a combination of the three types of performance trials.

Each performance trial consists of two procedures:

- 1) Data extraction: Packet layer reference events associated with the trial are created (or observed), time stamped, and recorded at the appropriate section boundaries;
- 2) Data reduction: The recorded reference event histories are processed consistent with the performance parameter definitions to determine the trial's outcome.

Subclauses 7.3.1 through 7.3.5 describe specific examples of performance trials and data extraction and data reduction procedures. Other equivalent (or superior) procedures can be designed and used.

#### 7.3.1 General assumptions and constraints

General assumptions and constraints that underlie the example test procedures are specified in 7.3.1.1 through 7.3.1.5.

##### 7.3.1.1 Test devices

The example procedures use the (C,M/C,M) test mode described in 7.2. The procedures assume that the test devices conform fully to the protocols employed at the monitored interfaces, i.e., in addition to generating the packet-level reference events appropriate for the particular trial, the test devices should respond correctly to the events generated by the section under test. The test devices should correctly record and accurately time stamp every reference event on the relevant logical channels. All such reference events should be recorded and time-stamped regardless of whether the event was expected. The complete record of reference events will be used in the determination of the trial's outcome.

##### 7.3.1.2 Trial sequences

Each of the example data extraction procedures defines an elementary sequence of steps in which a single access trial, one or more data transfer trials, or a single disengagement trial is conducted. To satisfy any of the measurement objectives discussed in 7.1, multiple trials must be completed. The trials described here may be conducted in any arbitrary sequence linked together using the following procedures. These linkages facilitate repeated trials for a single parameter and economical testing of several parameters.

Two procedures may be used to achieve the initial states of the access, data transfer, and disengagement trials. There are many possible final states of the  $b_i$  and  $B_j$  boundaries at the end of these trials. A generic approach is taken that ensures the attainment of the desired initial states regardless of the current state of the  $B_i$  and  $B_j$  boundaries. The procedures are written with reference to achieving a state at a particular boundary.

1. *Procedure for establishing state p1 at a boundary:* The following steps will create state p1 at a boundary:
  - a) Issue a SABM/SABME if the data link layer is not available and up (r1);
  - b) At an X.25/X.75 boundary, issue a DTE/STE clear request on the chosen LCN;

- c) Wait for confirmation of clearing.
2. *Procedure for establishing state p4/d1 at a boundary:* The following steps will create the state p4/d1 at a boundary:
- a) Get to state p1 (see the preceding procedure);<sup>14)</sup>
  - b) At the  $B_i$  boundary, issue a call request with the chosen LCN, called, and calling addresses;
  - c) At remote boundary  $B_j$ , wait for an incoming call or call request packet<sup>15)</sup> and issue the corresponding call accepted or call connected packet;
  - d) Wait for the corresponding call connected packet at the  $B_j$  boundary.

### 7.3.1.3 Trial failures and recovery

None of the example data extraction or data reduction procedures include detailed recovery algorithms for recovering from failed trials (e.g., failed call set-up attempts, resets, restarts, or residual errors). The trial linkages assume nothing about the states of the interfaces following a trial, so recovery routines are not absolutely necessary. However, all of the example procedures (data extraction and data reduction) could be made more robust by implementing failure-recovery mechanisms.

### 7.3.1.4 Matching packet level reference events

The example data reduction procedures all require matching of corresponding packet level reference events at the two test boundaries. In general, any reasonable method for matching packet events is sufficient. If the clocks in the test devices are synchronized, timing information can be used in event matching. If the section under test preserves data packet sequence numbers, that information can be used in the method that matches data packets. Data packet matching may also be done by comparing user data fields. If the data extraction procedures also include failure recovery mechanisms, the data reduction procedures must be more sophisticated in their ability to match packets (recognizing and compensating for lost, extra, and errored data packets).

### 7.3.1.5 Testing during service outages

By definition, the performance of the primary parameters is not to be evaluated during service outages. In each of the example data reduction procedures a decision is made as to whether the service was available during the monitored test. If the service was available, the trial results are included in the cumulative statistics used in evaluating the primary parameters. If the service was unavailable, the trial results may be used only in measuring availability performance.

Determining the availability state in the data reduction phase is a difficult problem. A single trial failure is insufficient to declare unavailability, so in the absence of other evidence, the service is assumed to be available. However, if this trial was contained in a 5-minute interval where one or more primary parameters performed worse than their decision criterion (see table 10), the service should be declared unavailable. Thus the availability decision performed in each individual data reduction procedure should take into consideration all trial results within plus or minus 5 minutes. Decisions about service availability can be delayed until all of the individual trials have been analyzed. If this approach is used, the primary performance parameter cumulative counters should be corrected retroactively wherever outages are discovered.

<sup>14)</sup> A channel allowed for incoming calls must be cleared at the called DTE.

<sup>15)</sup> The virtual connection may be established on an LCN other than the one originally chosen at the called DTE.

### 7.3.2 Example access trial

Values for call set-up delay (CSD), call set-up error probability (CSEP), and call set-up failure probability (CSFP) can be obtained using the example procedures specified in 7.3.2.1 and 7.3.2.2.

#### 7.3.2.1 Access trial data extraction

Figure 14 illustrates the access trial data extraction procedure. Boundaries  $B_i$  and  $B_j$  are at X.25 or X.75 interfaces bounding the set of virtual connection sections under test. Process A is a controlled source and Process B is a controlled sink.

Logical channels of boundaries  $B_i$  and  $B_j$  must initially be in state p1. Process A will transmit a call request (or incoming call) packet and wait for the corresponding call accepted (or call connected) packet. Process A should wait no less than the 200-second call set-up failure threshold defined in 5.1.

Timing of Process A and Process B must be sufficiently well-synchronized so that Process B will be ready to receive the corresponding call request (or incoming call) packet. Process B will receive that packet and respond with the appropriate call accepted (or call connected) packet. The time required for this response will be subtracted from CSD calculations; however, this response interval should be as small as possible to avoid substantially increasing the probability of exceeding the 200-second call set-up failure threshold.

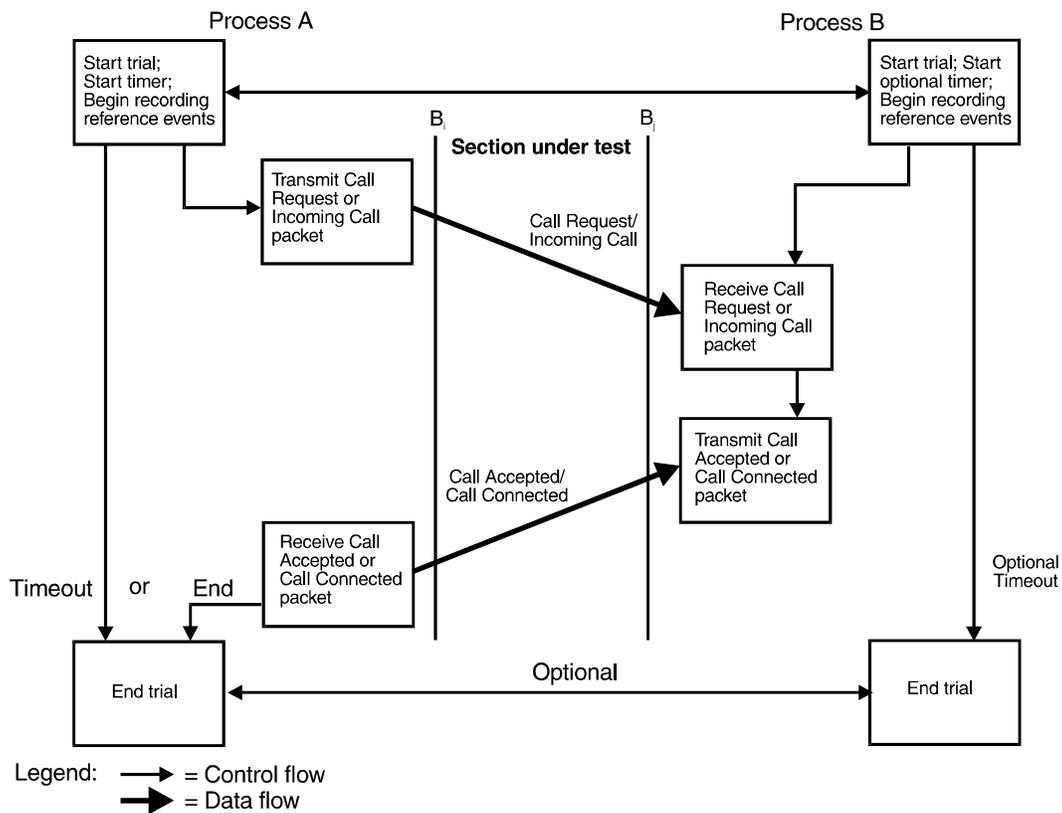


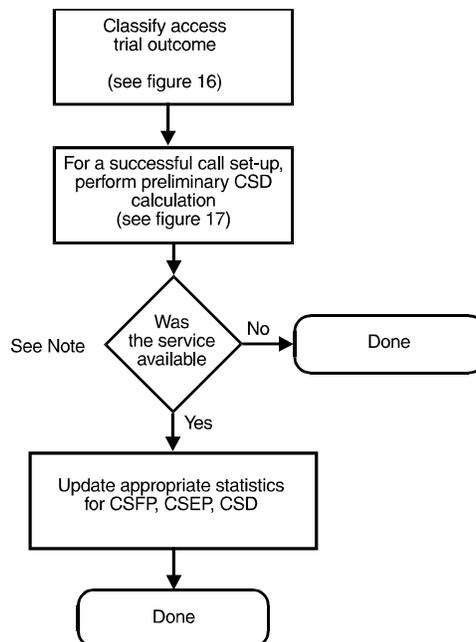
Figure 14 – Example access trial data extraction procedure

### 7.3.2.2 Access trial data reduction

Figure 15 illustrates the access trial data reduction procedure. Each access trial is classified as illustrated in figure 16. Figure 17 illustrates the steps needed to calculate CSD for successful call set-up attempts. The preliminary information derived in these two steps is used together with other trial results to determine if the service was available (see 7.3.1.5). If the service was available during this trial, cumulative call set-up statistics may be updated. Cumulative counters can be kept for total call set-up errors, total call set-up failures, and total call set-up delay (during successful attempts). Estimates of CSEP, CSFP, and mean CSD can then be created by dividing these cumulative counters by a cumulative count of call set-up attempts.

Figure 16 illustrates a method of using the record of packet level reference events (identified as events A, B, C, and D in this diagram) to determine if the call set-up attempt was successful or unsuccessful. Event D is said to have occurred only if the call accepted (or call connected) packet was received at  $B_i$  within the 200-second call set-up failure threshold. Otherwise it is assumed not to have occurred. If this process classifies any access trials as “Unsuccessful Trial - Cause Outside Portion Boundaries”, this is indicative of defective test devices that must be corrected.

Calculating CSD (see figure 17) first requires matching packets recorded at boundary  $B_i$  with packets recorded at boundary  $B_j$  (see 7.3.1.4). The exact timer values used in calculating CSD depend on the location of the boundaries  $B_i$  and  $B_j$  and the direction the packets are moving across those boundaries (see 3.2.1.2).



NOTE - If the appropriate statistical considerations have been addressed, this decision may be used in estimating the availability performance (see also 7.3.5).

**Figure 15 – Example access trial data reduction procedure**

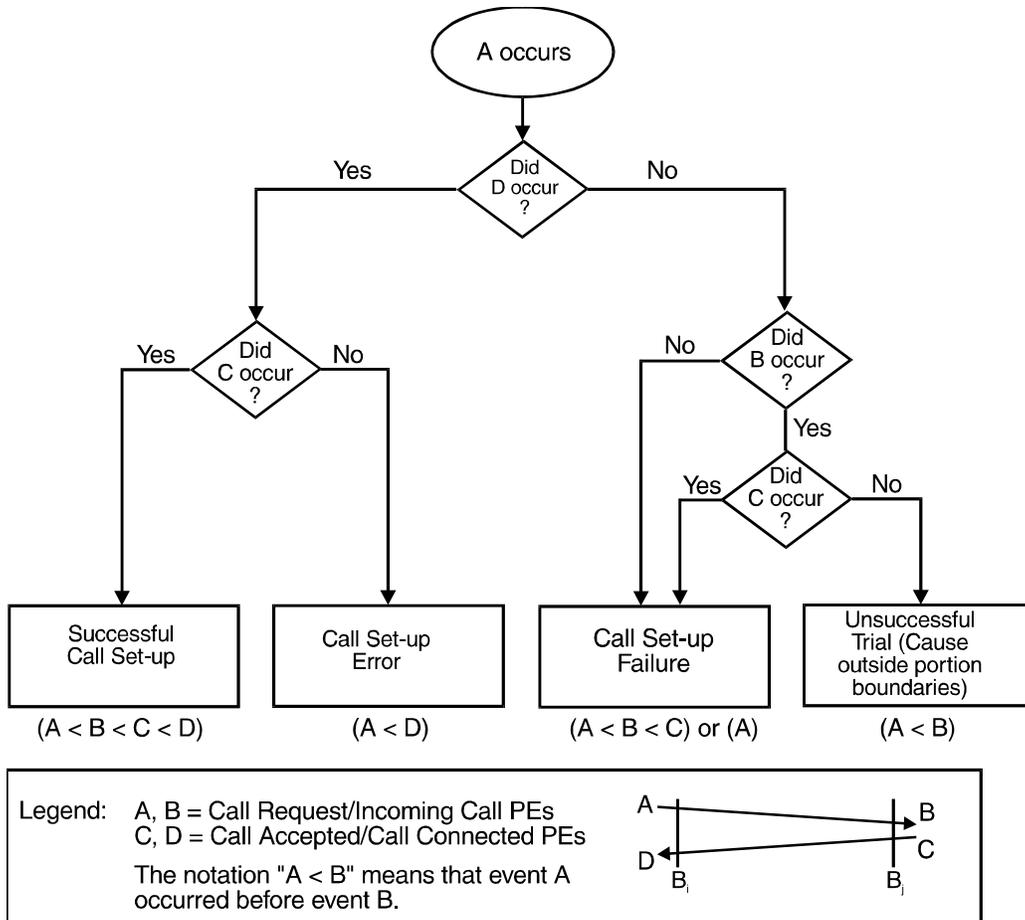
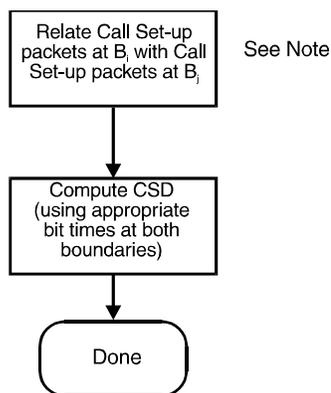


Figure 16 – Classifying an access trial outcome



NOTE - Any reasonable method is appropriate.

Figure 17 – CSD calculation

### 7.3.3 Example data transfer trial

Values for data packet transfer delay (DPTD), throughput capacity (TC), residual error ratio (RER), reset stimulus probability (RSP), reset probability (RP), premature disconnect stimulus probability (PDSP), and premature disconnect probability (PDP) can be obtained using the example procedures specified in 7.3.3.1 and 7.3.3.2.

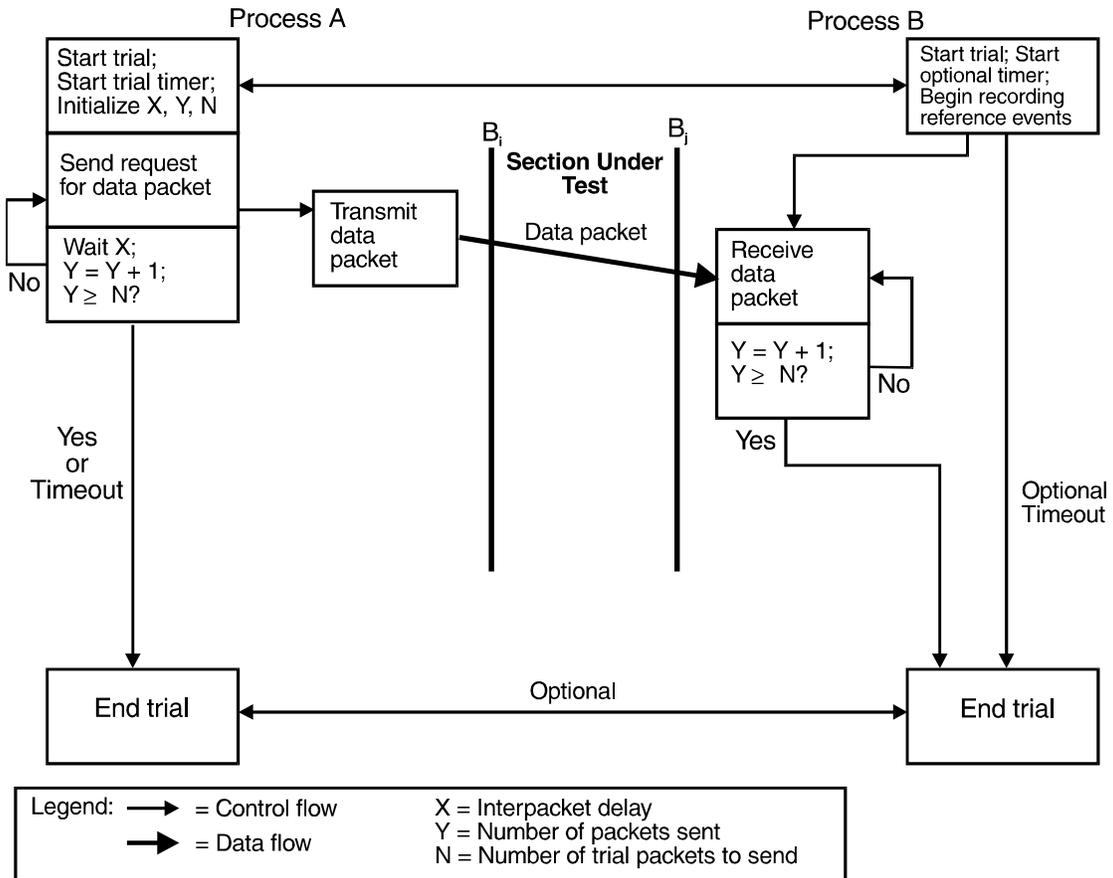


Figure 18 – Example data transfer trial extraction procedure

#### 7.3.3.1 Data transfer trial data extraction

Figure 18 illustrates the data transfer trial data extraction procedure. Boundaries  $B_i$  and  $B_j$  are at X.25 or X.75 interfaces bounding the set of virtual connection sections under test. Process A is a controlled source and Process B is a controlled sink.

Logical channels at boundaries  $B_i$  and  $B_j$  must be in state p4/d1. Process A will transmit N data packets. If throughput capacity (TC) is being tested, Process A should not delay the transmission of successive data packets except in response to window closings (in the figure:  $X=0$ ).

Process B should be ready to receive and record the reception of the corresponding data packets. **(Warning:** If packet splitting or recombination occurs in the sections under test, the expected number of

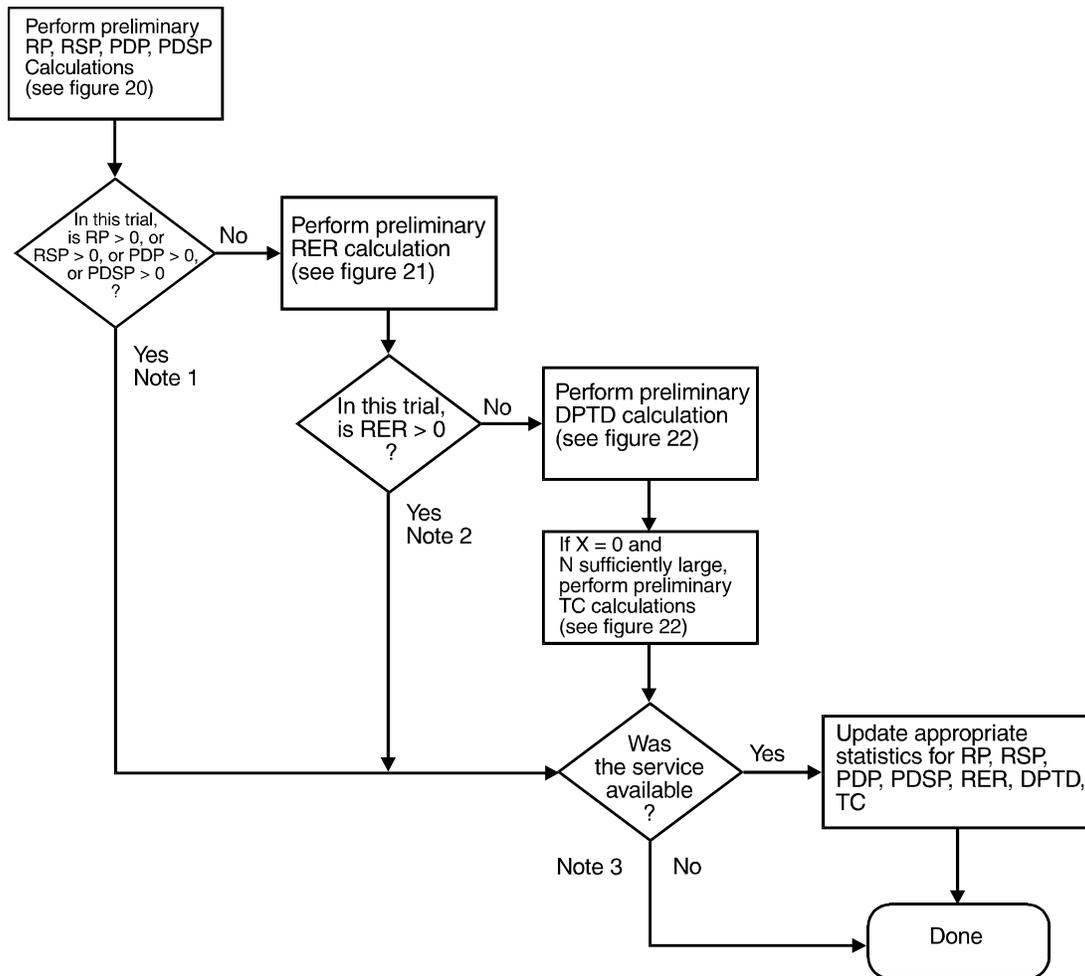
data packets at B, may be different from the data packets transmitted by Process A.) If TC is being tested, Process B should not delay in acknowledging frames and packets received. Clocks in Process A and Process B must be sufficiently well-synchronized so that the difference between the two clocks is an insignificant fraction of the likely DPTD value. Synchronization accuracy of one or two milliseconds is usually adequate (see Levine, J. et al).

Using a combination of timers and control mechanisms, the total time allowed for Process B to receive the last data packet from Process A should be at least the 200-second residual error ratio threshold defined in 5.2.1.1.

If the trial is being used to evaluate RP, RSP, PDP, and PDSP, both Process A and Process B should respond to resets, reset stimuli, premature disconnects, and premature disconnect stimuli as specified in ITU-T Recommendations X.25 and X.75. Process A should reestablish the virtual connection if it is prematurely disconnected. After a reset or connection re-establishment, Process A should resume transmission of data packets.<sup>16)</sup>

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<sup>16)</sup> User data bits lost or errored in conjunction with resets or clears are not counted as residual errors. User data bits retransmitted by Process A in order to recover from a reset or clear are not counted as residual errors.



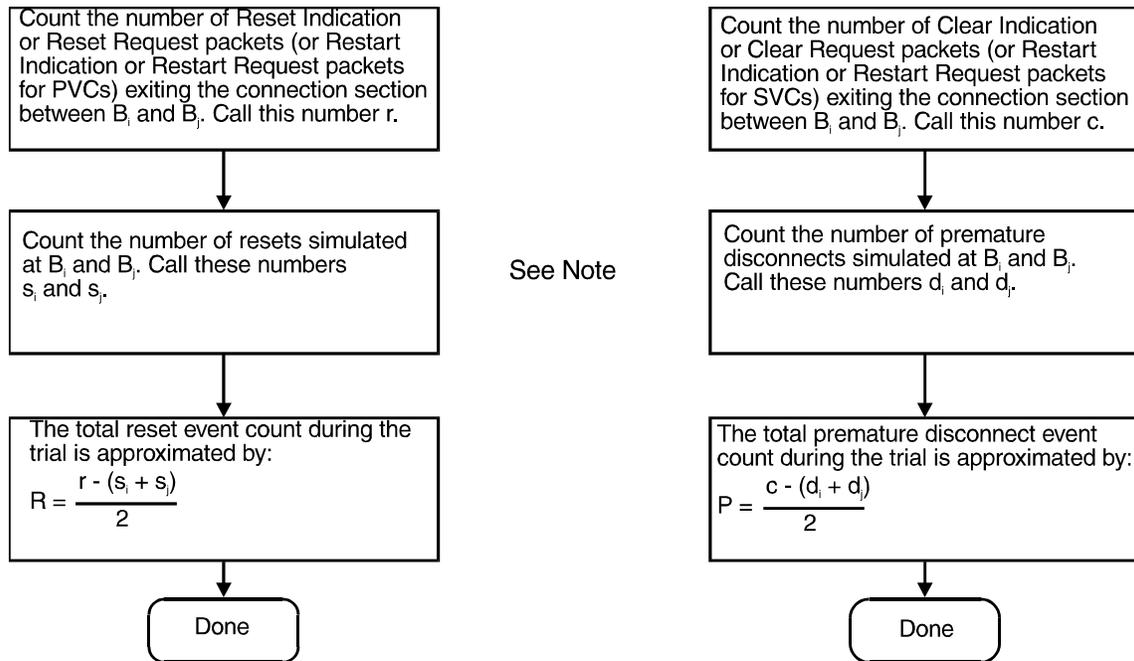
## NOTES

- 1 - More sophisticated procedures may recover from resets and premature disconnects, and permit residual error rate, data packet transfer delay and throughput capacity calculations.
- 2 - More sophisticated procedures may recover from residual errors and permit data packet transfer delay and throughput capacity calculations.
- 3 - This decision may be used in estimating service availability if the appropriate statistical considerations have been met.

**Figure 19 – Example data transfer trial data reduction procedure**

### 7.3.3.2 Data transfer trial data reduction

Figure 19 illustrates the data transfer trial data reduction procedure. Using the complete record of PEs, resets and premature disconnects are counted as illustrated in figure 20. The procedure of figure 21 is used to estimate the RER for trials in which no resets or premature disconnects occur. Figure 22 illustrates the steps needed to calculate DPTD and TC for trials in which the RER estimate is zero. The preliminary information derived in these three steps is used together with other trial results to determine if the service was available during this trial (see 7.3.1.5). If the service was available, cumulative data transfer statistics may be updated.



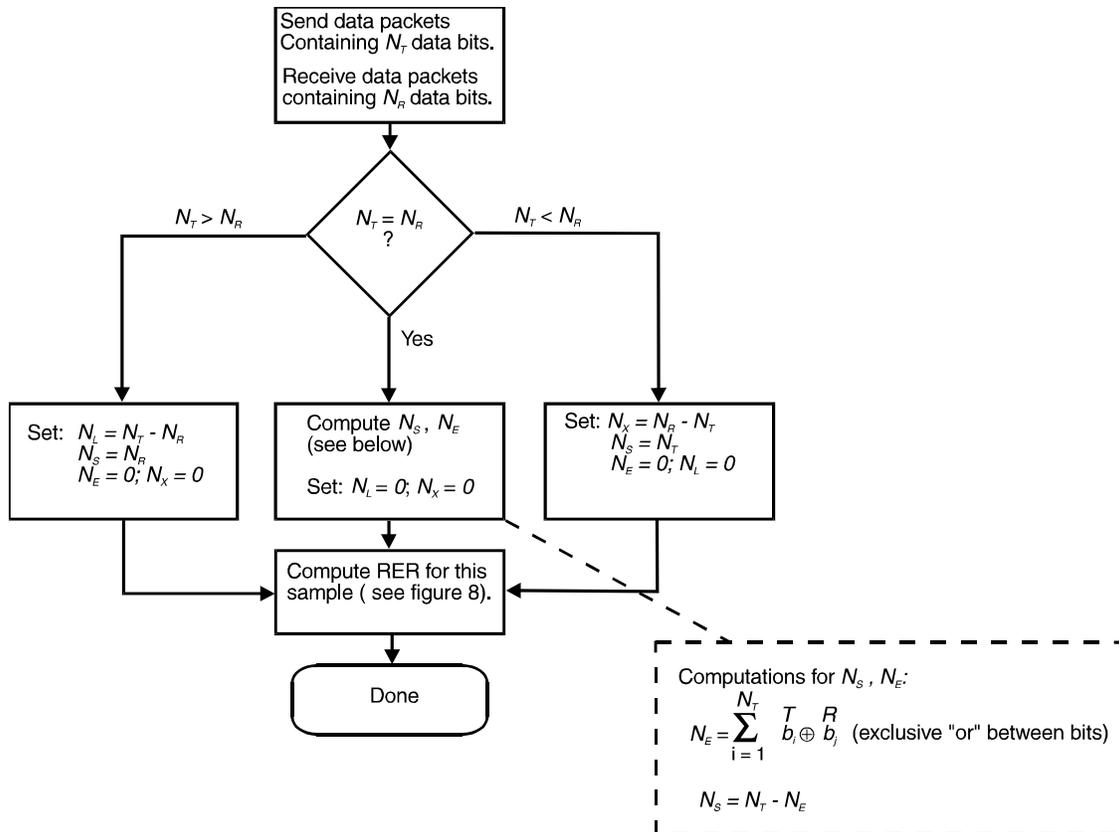
NOTE - These assume that sections outside of B<sub>i</sub> and B<sub>j</sub> will not unilaterally initiate resets, restarts, or clears, but will respond appropriately to reset and premature disconnect stimuli.

**Figure 20 – RP, RSP, PDP, and PDSP calculations (approximate method)**

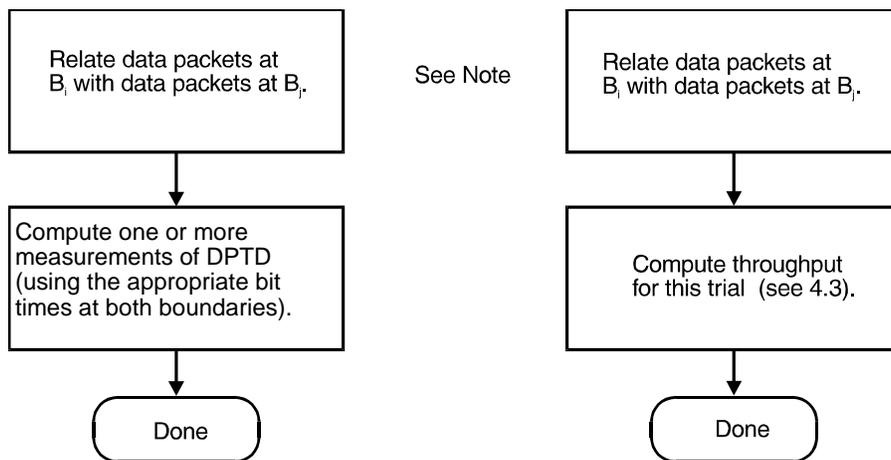
Cumulative counters can be kept for total reset events, total reset stimuli, total premature disconnect events, and total premature disconnect stimuli. Estimates of RP, RSP, PDP, PDSP can then be created by dividing those counters by a cumulative count of the time during which data transfer was tested. Long-term cumulative counters can also be kept for  $N_T$ ,  $N_R$ ,  $N_L$ ,  $N_E$ ,  $N_X$ , and  $N_S$ . Then RER can be estimated using the equations in figure 8. For successful data transfer attempts, total data packet transfer delay and total data packets transferred can be accumulated. The ratio of these two numbers is an estimate of mean DPTD. For successful TC trials, total bits transferred and total time in those trials (as defined in 4.3) can be accumulated. Dividing these numbers yields an estimate of TC.

Figure 20 illustrates a method of using the record of packet level reference events to evaluate the resets and premature disconnects that occurred during the trial. The equations presented depend on the fact that a reset (premature disconnect) event between B<sub>i</sub> and B<sub>j</sub> will cause two reset (or clear) packets to exit the section(s) under test while a reset (or premature disconnect) stimulus will cause one reset (or clear) packet to enter the sections and one to exit.

Figure 21 illustrates an acceptable approximation for calculating RER. The approximation is based on the assumption that in a single trial only one type of residual error can occur: i.e., lost bits and errored bits do not occur in the same trial, errored bits and extra bits do not occur in the same trial, and lost bits and extra bits do not occur in the same trial. For purposes of estimating RER, this approximation is assumed to be sufficiently accurate. Other, more sophisticated methods of comparing transmitted bits with received bits may yield a more accurate estimate of RER. User information received at B<sub>j</sub> more than 200 seconds after it was transmitted at B<sub>i</sub> is defined to be lost information  $N_L$  (see 5.2.1.1).



**Figure 21 – Residual error ratio calculation (approximate method)**



NOTE - Any reasonable method is appropriate.

**Figure 22 – DPTD and TC calculation**

Calculating DPTD and TC (see figure 22) first requires matching packets recorded at boundary  $B_i$  with packets recorded at boundary  $B_j$  (see 7.3.1.4). The exact timer values used in calculating DPTD and TC depend on the location of the boundaries  $B_i$  and  $B_j$ , and the direction the packets are moving across those boundaries (see 3.2.1.2).

### 7.3.4 Example disengagement trial

Clear indication delay (CID) and call clear failure probability (CCFP) values can be obtained using the example procedures specified in 7.3.4.1 and 7.3.4.2.

#### 7.3.4.1 Disengagement trial data extraction

Figure 23 illustrates the disengagement trial data extraction procedure. Boundaries  $B_i$  and  $B_j$  are at X.25 or X.75 interfaces bounding the set of virtual connection sections under test. Process A is a controlled source and Process B is a controlled sink.

Logical channels at  $B_i$  and  $B_j$  must initially be in state p4/d1. Process A will transmit a clear request (or clear indication) packet. Process B should be ready to receive and record the reception of the corresponding clear request (or clear indication) packet. Clocks in Process A and Process B must be sufficiently well-synchronized so that the difference between the two clocks is an insignificant fraction of the likely CID value. Synchronization accuracy of one or two milliseconds is usually sufficient.

Using a combination of timers and control mechanisms, the total time allowed for Process B to receive the call-clearing packet from Process A should be at least the 180-second call clear failure threshold defined in 5.3.1.

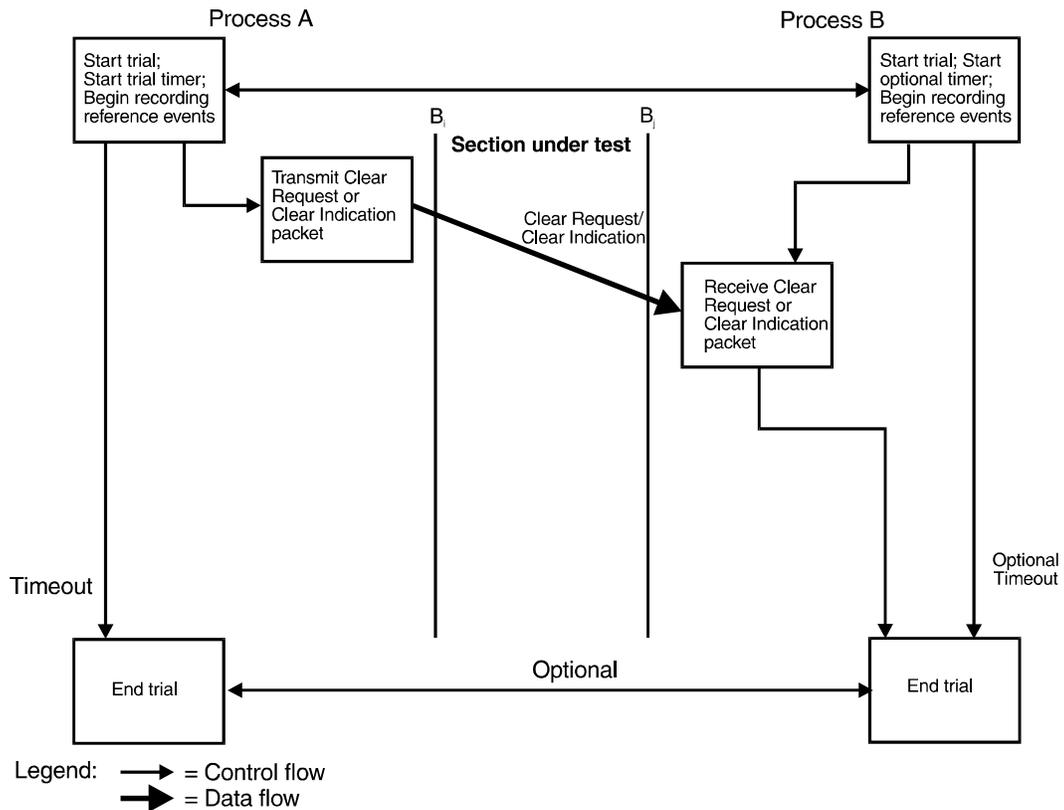
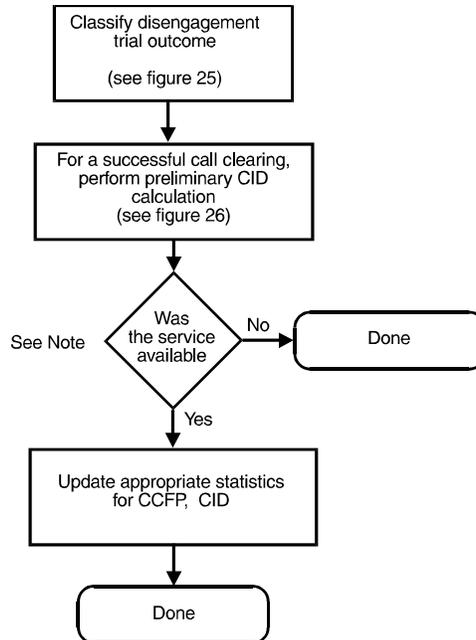


Figure 23 – Example disengagement trial data extraction procedure

### 7.3.4.2 Disengagement trial data reduction

Figure 24 illustrates the disengagement trial data reduction procedure. Each disengagement trial is classified as illustrated in figure 25. Figure 26 illustrates the steps needed to calculate CID for successful call clearings. The preliminary information derived in these two steps is used together with other trial results to determine if the service was available (see 7.3.1.5). If the service was available during this trial, cumulative call-clearing statistics may be updated. Cumulative counters can be kept for total call-clearing failures and total clear indication delay (during successful attempts). Estimates of CCFP and mean CID can then be created by dividing these cumulative counters by a cumulative count of call-clearing attempts.

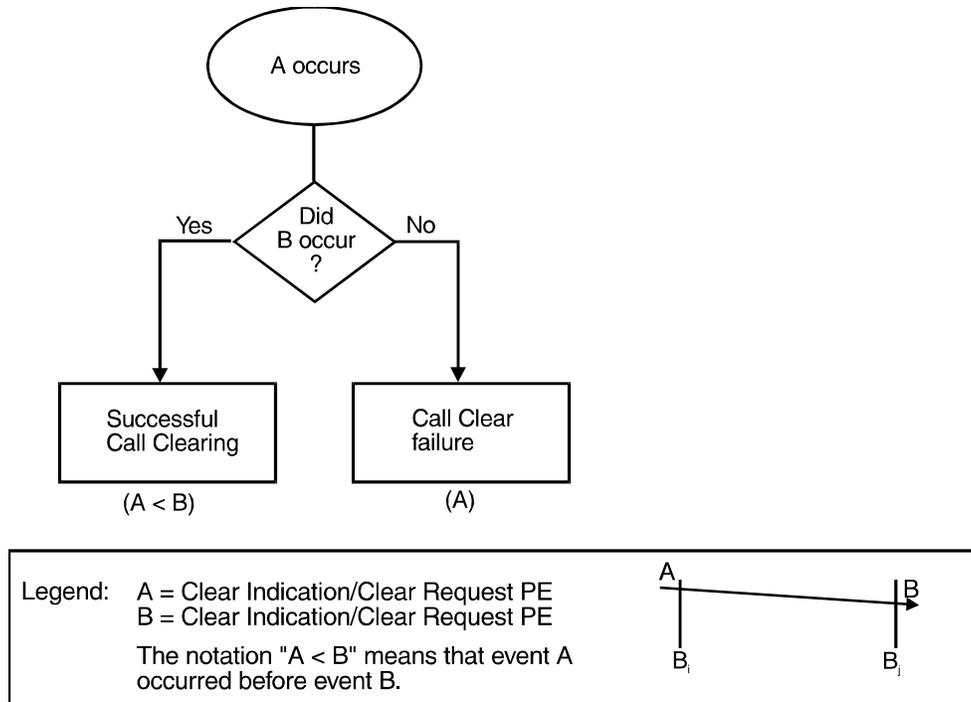


NOTE - If the appropriate statistical considerations have been addressed, this decision may be used in estimating the availability performance (see also 7.3.5).

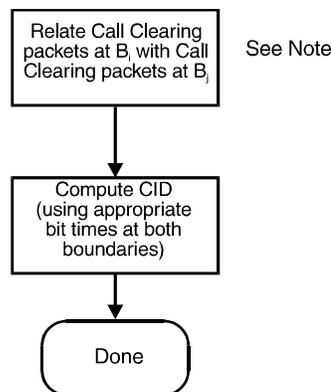
**Figure 24 – Example disengagement trial data reduction procedure**

Figure 25 illustrates a method of using the record of packet level reference events (identified as events A and B in this figure) to determine if the call clearing was successful or unsuccessful. Event B is said to have occurred only if the clear indication (or clear request) packet was received at  $B_j$  within the 180-second call clear failure threshold defined in 5.3.1. Otherwise it is assumed not to have occurred and the clear attempt is classified as a failure.

Calculating CID (see figure 26) first requires matching packets recorded at boundary  $B_i$  with packets recorded at boundary  $B_j$  (see 7.3.1.4). The exact timer values used in calculating CID depend on the location of the boundaries  $B_i$  and  $B_j$ , and the direction the packets are moving across those boundaries (see 3.2.1.2).



**Figure 25 – Classifying a disengagement trial outcome**



NOTE - Any reasonable method is appropriate.

**Figure 26 – CID calculation**

### 7.3.5 Estimating availability parameters

The availability parameters, service availability (SA) and mean time between service outages (MTBSO), can be measured for a given virtual connection section by an appropriate sequence of availability trials.

### 7.3.5.1 Minimal availability trial

The following test and its decision criteria are defined to be the minimum criteria necessary to sample the availability state of a section.

#### 7.3.5.1.1 Minimal availability trial data extraction

Figure 27 illustrates a procedure for conducting a minimal availability trial across a section. The trial is divided into two phases: access (Phase I) and user information transfer (Phase II). The trial requires controlled sources and sinks and monitors at each section boundary.

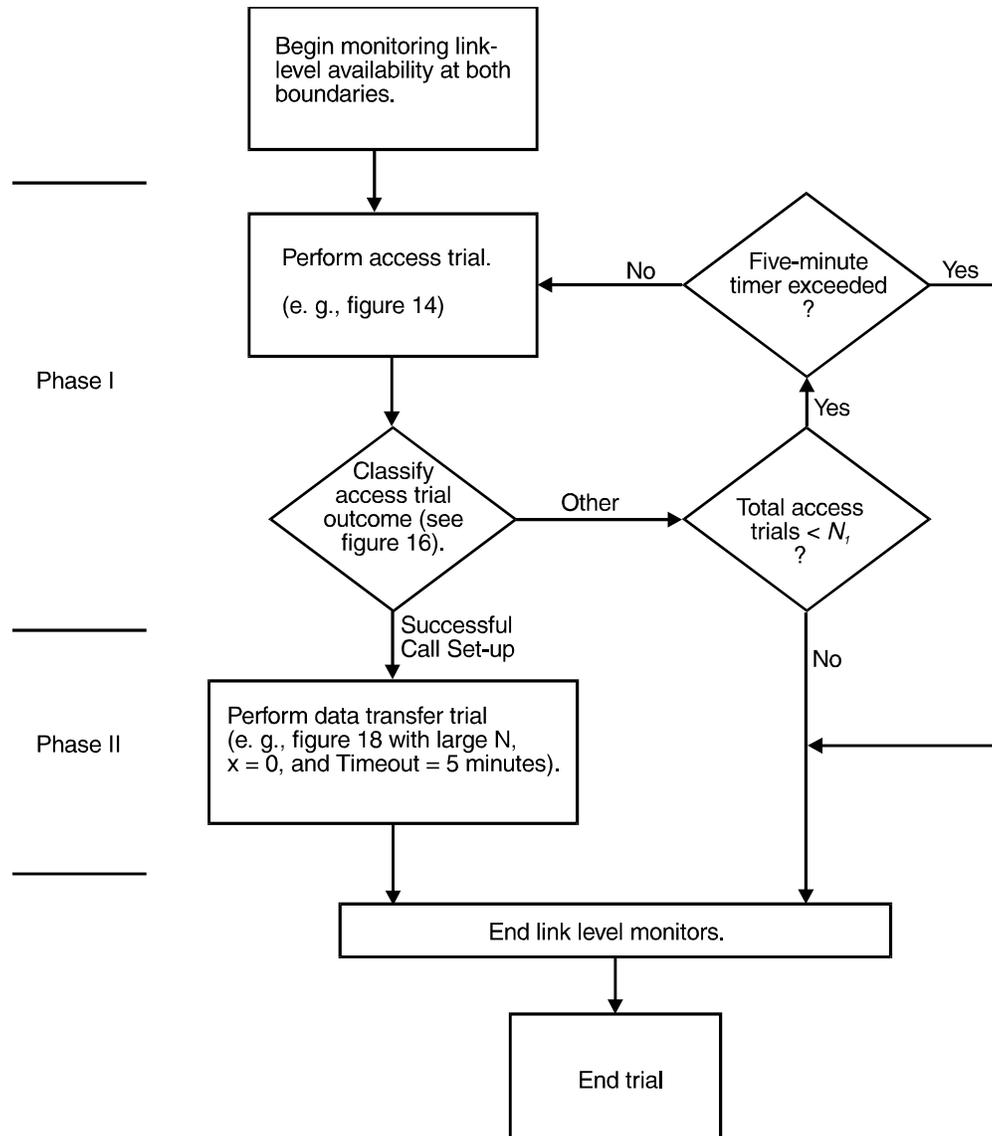


Figure 27 – Minimal availability trial data extraction procedure

In Phase I, no more than  $N_1$ <sup>17)</sup> successive call set-up attempts (access trials) are performed. Phase I is terminated when any of the following occur: a call set-up attempt succeeds,  $N_1$  successive call set-up attempts are unsuccessful, or the duration of Phase I exceeds 5 minutes (300 seconds). Each call set-up attempt may be performed according to the procedure described in 7.3.2.1 and illustrated in figure 14. Phase I is performed only in the case of a switched virtual call.

Phase II of a minimal availability trial attempts to maintain a virtual connection across the tested section for 5 minutes and maintain an average throughput that exceeds  $N_2$  bit/s during that interval. Packet transmission may be performed according to the procedure described in 7.3.3.1 and illustrated in figure 18. The values of  $N$  and  $X$  should be such that the procedure attempts to achieve throughput considerably greater than  $N_2$  bits per second. In the case of a switched virtual call, Phase II is performed only if a call set-up attempt in Phase I is successful. In the case of a permanent virtual circuit, only Phase II is performed.

### 7.3.5.1.2 Minimal availability trial data reduction

Figure 28 illustrates a procedure for reducing performance data recorded in a minimal availability trial. The procedure implements the availability decision criteria defined in clause 6.

Criteria defined in 6.1 are used to determine availability of the data link layers. Three cases are distinguished: (i) the data link layer at both section boundaries is available, (ii) the data link layer at either section boundary is unavailable due to causes inside the section, and (iii) the data link layer at one or both boundaries is unavailable due to causes outside the section and the data link layer at neither boundary is unavailable due to causes inside the section. In case (i), availability of the section is determined by results of Phases I and II of the trial. In case (ii), the section is declared unavailable; in case (iii), the trial is excluded.

Processing of Phase I results is illustrated in the upper part of figure 28. The outcome of each attempt is determined according to the procedure described in 7.3.2.2. If all attempts result in either call set-up error or call set-up failure, the virtual circuit section is considered to be unavailable for the duration of the trial. If any call set-up attempt is unsuccessful due to causes outside the portion boundaries, (e.g., because test equipment malfunctions), the trial is excluded and is not used to determine availability parameters.

If any access trial is successful and no access trial is unsuccessful due to causes outside the section, availability of the section is determined by results in Phase II.

Processing of Phase II results is illustrated in the lower part of figure 28. The reduction procedure implements the following decision criteria based on 6.1:

- 1) If the observed number of reset events plus the number of reset stimuli during Phase II is greater than  $N_3$ , the section is unavailable;
- 2) If the observed number of call cleared events due to premature disconnects or premature disconnect stimuli is greater than  $N_6$  (in the case of a switched virtual call), the section is unavailable;
- 3) If the measured residual error ratio during Phase II is greater than  $N_5$ , the section is unavailable;
- 4) If the observed throughput during Phase II is less than  $N_4$  bit/s, the section is unavailable.

If the section is not unavailable according to any of the preceding four criteria, the section is considered to be available during the trial.

<sup>17)</sup> The numbers  $N_1$ ,  $N_2$ ,  $N_3$ ,  $N_4$ ,  $N_5$ , and  $N_6$  used in 7.3.5 depend on the particular values chosen for the availability decision parameters in table 10 and are for further study.

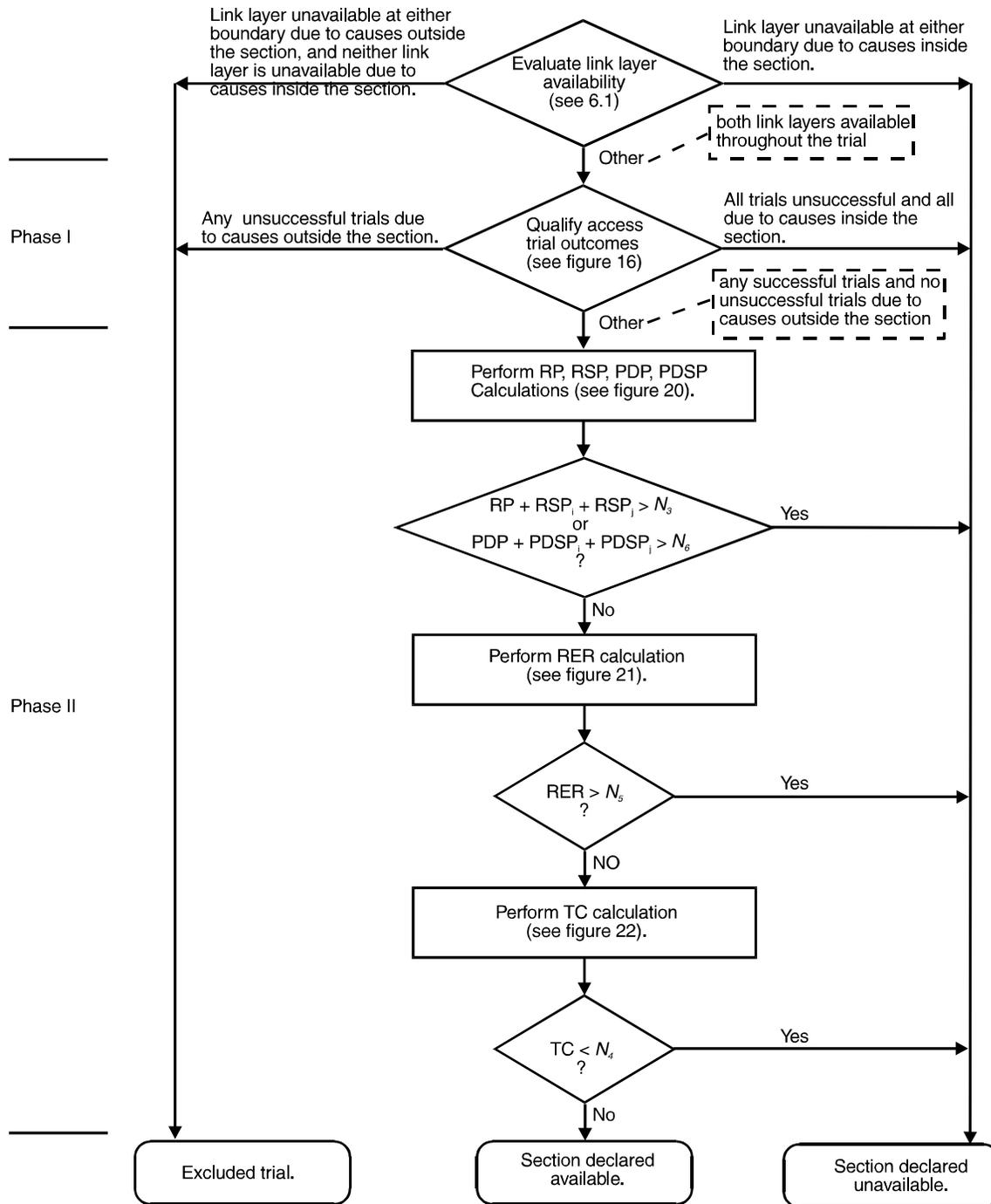
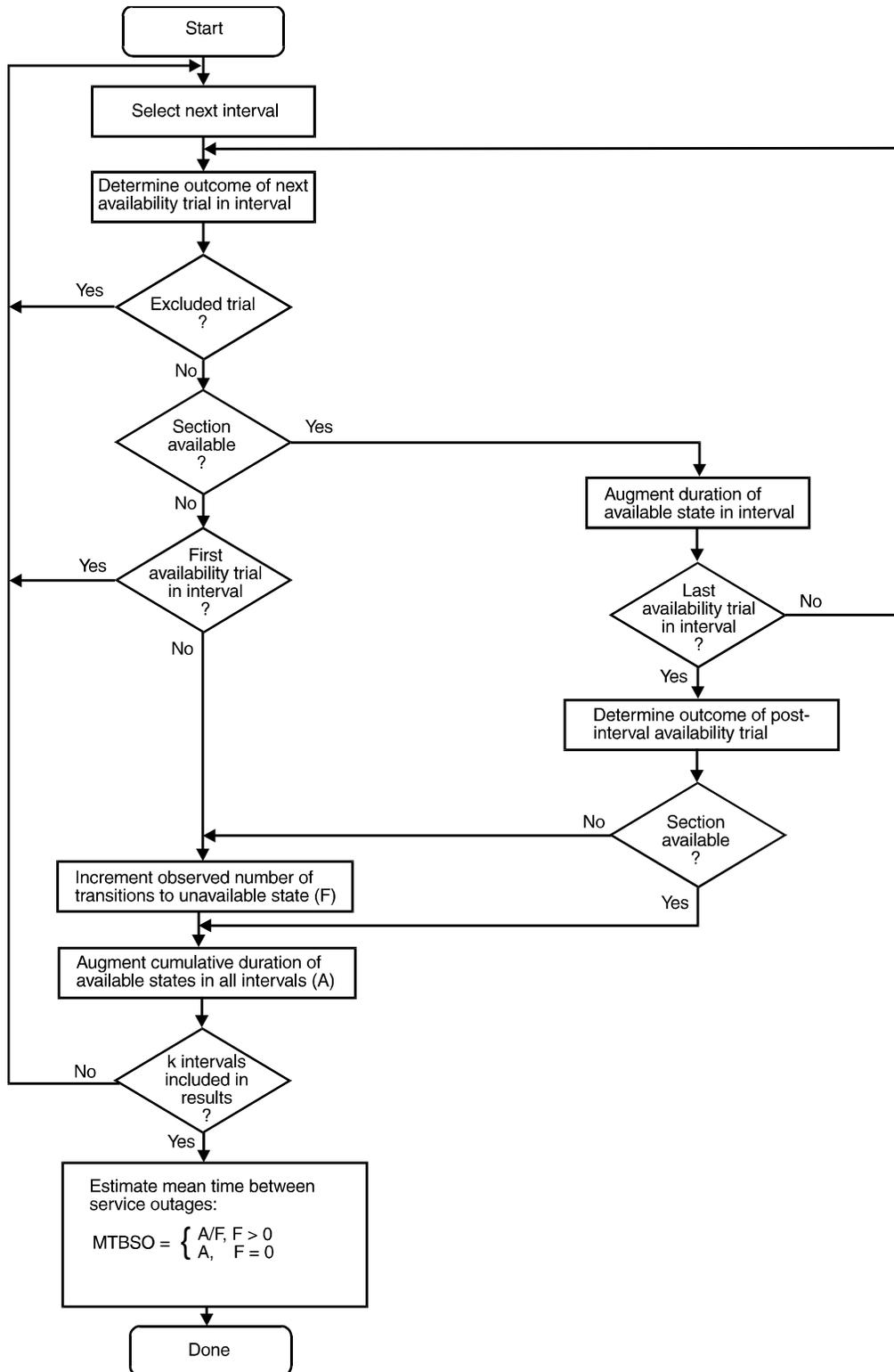


Figure 28 – Minimal availability trial data reduction procedure

7.3.5.2 Estimating service availability

A sufficient estimate of service availability for a virtual circuit section can be computed by performing a sequence of minimal availability trials as described in this clause.

A sequence of not less than 300 availability trials is conducted across the section over a long measurement period (e.g., 6 months). Because of the expected durations of service outages, successive trials should be separated by at least 7 hours (this serves to keep availability trials uncorrelated). The trials should be uniformly distributed across the scheduled service time. A trial whose outcome is excluded may be replaced by a trial conducted immediately following the excluded trial. The estimate of service availability is 100 times the number of trials in which the section is declared available divided by the total number of trials whose outcomes are not excluded.



**Figure 29 – Data reduction procedure for estimating mean time between service outages**

### 7.3.5.3 Estimating mean time between service outages

A sufficient estimate of the mean time between service outages for a virtual connection section can be computed by performing a sequence of availability trials.

Select  $k$  disjoint time intervals each not less than 30 minutes and not more than 3 hours. The total amount of time in the  $k$  intervals should exceed three times the a priori estimate of mean time between service outages. Consecutive availability trials are conducted across the section for the duration of each interval. An additional "post-interval" availability trial is conducted immediately following the last trial in an interval. An estimate of the mean time between service outages is obtained from the measured time ( $A$ ) in the available state and the observed number ( $F$ ) of transitions from the available state to the unavailable state.

Figure 29 illustrates a procedure for reducing the collected performance data and estimating the mean time between service outages. The procedure implements the following specifications:

- 1) If the outcome of any trial in an interval is excluded, discard all trials in the interval;
- 2) If the section is unavailable during the first trial in an interval, assume that the transition to the unavailable state occurred before the interval began and discard all trials in the interval;
- 3) If the section is available during the first trial in an interval and is unavailable during any subsequent trial in the interval, increment the observed number ( $F$ ) of transitions to the unavailable state by one. Augment the cumulative duration ( $A$ ) of available states by the duration of all trials in the interval that precede the first unavailability outcome. Discard all trials in the interval that follows the first unavailability outcome;
- 4) If the section is available during all trials in an interval, augment the cumulative duration ( $A$ ) of available states by the duration of these trials. If the section is unavailable during the post-interval trial, increment the number ( $F$ ) of transitions to the unavailable state.

After the results of all  $k$  intervals have been processed, the mean time between outages is estimated by the ratio  $A/F$  if  $F > 0$  and by  $A$  if  $F = 0$ .

The estimate of mean time between service outages assumes that, if an outage begins during an availability trial, either that trial or the following trial will determine that the section is unavailable. This is a reasonable assumption since service outages, in contrast to transient failures, will last more than 5 minutes.

Discarding the remainder of the interval following an unavailability outcome is both practical and statistically justifiable. The virtual connection section must return to the available state before any more available time can be accumulated and before any more transitions to the unavailable state can be observed. First, the expected time to restore service may be large with respect to the remaining time in the interval. It can be inappropriate and counterproductive to continue testing a failed or congested network section. Second, if transitions to the unavailable state are statistically independent, then discarding the remainder of the interval, which may include time in the available state, will not bias the result.<sup>18)</sup> Intervals should be short with respect to the sum of the expected time to restore service and the expected time between service outages. Thus each interval should be no longer than 3 hours. A minimum recommended interval length is 30 minutes, using 5-minute availability trials.

An outage that begins during the first availability trial of an interval may or may not result in an unavailable outcome. According to the estimation procedure, if an unavailable outcome occurs, the interval is discarded, the state transition is missed, and the mean time between service outages is overestimated. The post-interval trial identifies any state transition that occurs during the last trial of the interval. It also counts certain transitions that occurred outside of the interval. These transitions are counted with the same

<sup>18)</sup> If outages tend to be clustered, discarding trials following a transition to the unavailable state will tend to overestimate the mean time between service outages. If outages tend to be negatively clustered, discarding trials following a transition to the unavailable state will tend to underestimate the mean time between service outages.

probability as the probability that transitions during the first trial in an interval are missed. Thus, the two sources of bias tend to cancel out.

#### 7.4 Relevant statistics and measurement conditions

Table 12(a-c) provides a reference for the calculation of relevant statistics for the performance parameters. For each parameter an equation is given for calculating a single observation  $x$ . Most of these equations use variable names defined in clauses 4, 5, and 6.

For each parameter an equation is given for converting multiple observations,  $x_i$ , into a sample mean  $\bar{x}$ . In those cases where single observations do not depend on either the length of the observations or the numbers of bits transmitted, the sample mean is an arithmetic mean. In the other cases (TC, RER, RSP, RP, PDSP, PDP, MTBSO) the sample means are calculated with each observation weighted appropriately either by the length of the observation or the number of bits transmitted.

For three of the parameters (CSD, DPTD, and CID) a formula is given for estimating the variance in the distribution of the parameter. Four other parameters (CSEP, CSFP, CCFP, service availability) are assumed to be binomially distributed and as such the variance and the sample variance contain no additional information about service performance. For the remaining seven parameters whose individual observations depend on the length of observation or on the number of bits transmitted during the observation (TC, RER, RSP, RP, PDSP, PDP, MTBSO), no formula is given for computing a sample variance. The sample variance in these cases depends greatly on the size of the individual observations. In order to evaluate the variability in these parameters, a single fixed observation size must be chosen for each parameter.

The first step in developing a set of TC observations is choosing an optimal (or nearly optimal) set of user controllable factor levels  $\gamma$ . Packet layer window size, data packet length, throughput class, D-bit usage, and inter-packet gaps must all be chosen to maximize the possible throughput. Subclause 7.4.1 gives guidance on how these factors can improve throughput. Subclause 7.4.1 also lists other parameters that may be adjusted to improve throughput.

Each observation of TC is based on the optimal set of factor levels  $\gamma$ . The mean of these observations, weighted by the duration of each observation, yields a single estimate of throughput capacity.

To properly interpret measured performance values, the relevant measurement conditions must be known. Table 13 identifies general factors that may influence the values for each of the performance parameters defined in this standard. Measurements provided in accordance with this standard should contain or reference a specification of the relevant factor levels existing during the measurement. The effects of the specified factors on throughput are described in 7.4.1. Guidelines for multiple test reporting are given in 7.4.2, and a throughput measurement correction factor is given in 7.4.3.

#### 7.4.1 Factors

##### 7.4.1.1 Signaling rates

The signaling rate on a circuit section (usually measured in bits per second) provides an upper bound on the throughput capacity for that section. In general, faster signaling rates yield higher throughput. Call set-up delay, data packet transfer delay and clear indication delay are also affected by the signaling rate. A higher signaling rate on a circuit section generally results in lower delay values.

##### 7.4.1.2 Data link layer window size

As the data link layer underlies all of the logical channels above it, it affects call set-up delay and clear indication delay as well as data packet transfer delay and throughput. Usually, larger data link layer window sizes decrease delay and increase throughput.

#### **7.4.1.3 Packet layer window size**

Larger packet layer window sizes may increase throughput and decrease delay.

#### **7.4.1.4 Packet length**

The length of the data packets can affect data packet transfer delay, throughput, and possibly the residual error ratio. Longer data packets have greater delay associated with them, but greater throughput due to increased efficiency. Longer data packets have a theoretically larger probability of having errors in the user data field that could escape detection by the sixteen bit cyclic redundancy check of the link layer and thus a corresponding increase in the probability of residual errors.

#### **7.4.1.5 Other virtual connections**

The existence of active virtual connections other than the one under test on the same data link may increase the load on that link. Hence, a large number of performance parameters (call set-up delay, data packet transfer delay, throughput, call set-up failure probability, reset probability, premature disconnect probability, clear indication delay, service availability, and mean time between service outages) may be affected due to the underlying link layer contention and the increased probability of link layer failure.

#### **7.4.1.6 Time of day**

Because time of day generally influences network loading, this factor affects performance in a manner similar to the existence of other virtual connections (7.4.1.5).

#### **7.4.1.7 Throughput class**

This factor may affect the data packet transfer delay and throughput on some networks. It should be set to the maximum allowable when measuring throughput capacity.

#### **7.4.1.8 D bit usage**

Setting the D bit to 1 in a data packet transfers responsibility for window updating to the receiving DTE and can thus decrease throughput.

#### **7.4.1.9 Inter-packet gaps**

The rules governing the time intervals between successive data packets in data packet transfer delay trials (other than those required by the network for flow control purposes) should be specified. Increasing inter-packet gaps tends to decrease delay and throughput.

### **7.4.2 Reporting multiple tests made between different locations**

Full characterization of network performance can require multiple test measurements that are made between different node locations such as cities. In general, where tests are conducted between multiple locations this should be reported. If the measured data are significantly affected by this test conduct, then the geographic test arrangements should be considered a factor and incorporated into the appropriate statistical procedures used to estimate the performance parameters specified in this standard.

**Table 12 – Calculation of statistics for performance parameters**  
**a) Speed of service parameters – Delay**

Parameter	One Observation	Sample mean	Estimate of variance
Call set-up delay	$d_1 - d_2 = x^{1)}$	$\frac{1}{N} \sum_{i=1}^N x_i = \bar{x}$	$\frac{1}{N-1} \sum_{i=1}^N (x_i - \bar{x})^2$
Data packet transfer delay	$t_2 - t_1 = x^{2)}$	$\frac{1}{N} \sum_{i=1}^N x_i = \bar{x}$	$\frac{1}{N-1} \sum_{i=1}^N (x_i - \bar{x})^2$
Clear indication delay	$t_2 - t_1 = x^{3)}$	$\frac{1}{N} \sum_{i=1}^N x_i = \bar{x}$	$\frac{1}{N-1} \sum_{i=1}^N (x_i - \bar{x})^2$
1) $d_1, d_2$ defined in 4.1.2. 2) $t_1, t_2$ defined in 4.2. 3) $t_1, t_2$ defined in 4.4.			

**b) Speed of service parameter – Throughput capacity**

Parameter	One observation	Weights	Sample mean TC estimate
Throughput capacity	$T(\gamma) = \frac{B}{t} = x^{1)}$	$\frac{t_i}{\sum_{j=1}^M t_j} = w_i$	$\sum_{i=1}^M w_i x_i = \bar{x}$
1) $T(\gamma)$ is the steady state throughput measured with optimal factor levels, $\gamma$ ; $B$ is the number of user data bits transferred; $t$ is the time interval for the transfer (see 4.3.3).			

(continued)

**Table 12 (concluded)**  
**c) Accuracy, dependability, and availability parameters**

Parameter	One observation	Weights	Sample mean
Call set-up error probability	$0, 1 = x^1)$	—	$\frac{1}{N} \sum_{i=1}^N x_i = \bar{x}$
Call set-up failure probability	$0, 1 = x^2)$	—	$\frac{1}{N} \sum_{i=1}^N x_i = \bar{x}$
Residual error ratio	$\frac{N_E + N_L + N_X}{N_T} = x^3)$	$\frac{N_{T_i}}{\sum_{j=1}^M N_{T_j}} = w_i$	$\sum_{i=1}^M w_i x_i = \bar{x}$
Reset stimulus probability (for a single boundary)	$\frac{s}{N_{vc-s}} = x^4)$	$\frac{N_{vc-s_i}}{\sum_{j=1}^M N_{vc-s_j}} = w_i$	$\sum_{i=1}^M w_i x_i = \bar{x}$
Reset probability	$\frac{R}{N_{vc-s}} = x^5)$	$\frac{N_{vc-s_i}}{\sum_{j=1}^M N_{vc-s_j}} = w_i$	$\sum_{i=1}^M w_i x_i = \bar{x}$
Premature disconnect stimulus probability (for a single boundary)	$\frac{d}{N_{vc-s}} = x^6)$	$\frac{N_{vc-s_i}}{\sum_{j=1}^M N_{vc-s_j}} = w_i$	$\sum_{i=1}^M w_i x_i = \bar{x}$
Premature disconnect probability	$\frac{P}{N_{vc-s}} = x^7)$	$\frac{N_{vc-s_i}}{\sum_{j=1}^M N_{vc-s_j}} = w_i$	$\sum_{i=1}^M w_i x_i = \bar{x}$
Call clear failure probability	$0, 1 = x^8)$	—	$\frac{1}{N} \sum_{i=1}^N x_i = \bar{x}$
Service availability	$0, 100 = x^9)$	—	$\frac{1}{N} \sum_{i=1}^N x_i = \bar{x}$
Mean time between service outages	$\frac{A}{F} = x^{10)}$	$\frac{F_i}{\sum_{j=1}^M F_j} = w_i$	$\sum_{i=1}^M w_i x_i = \bar{x}$

1) x is 1 if a call set-up error occurs in the call set-up (see 5.1.1).

2) x is 1 if a call set-up failure occurs in the call set-up (see 5.1.2).

3) Outcome totals as specified in figure 8 (see 5.2.1).

4) s is the number of reset stimuli observed at the boundary (see 5.2.2.1);  $N_{vc-s}$  is the number of virtual circuit-seconds observed.

5) R is the number of reset events observed (see 5.2.2.2);  $N_{vc-s}$  is the number of virtual circuit-seconds observed.

6) d is the number of premature disconnect stimuli observed at the boundary (see 5.2.3.1);  $N_{vc-s}$  is the number of virtual circuit-seconds observed.

7) P is the number of premature disconnect events observed (see 5.2.3.2);  $N_{vc-s}$  is the number of virtual circuit-seconds observed.

8) x is 1 if a call clear failure occurs in the call clearing (see 5.3.1).

9) x is 100 if the observation determines the service is available (see 6.2.1).

10) A is the cumulative duration of available states; F is the number of transitions to unavailable state observed (see 6.2.2).

**Table 13 – Factors that may influence the performance parameter values**

Parameters →	C S D	D P T D	T C	C I D	C S E P	C S F P	R E R	R S P	R P	P D S P	P D P	C C F P	S A	M T B S O
Factors														
1. Signalling rate	X	X	X	X										
2. Data link layer window size	X	X	X	X										
3. Packet layer window size		X	X											
4. Packet length		X	X				X							
5. Other virtual connections	X	X	X	X		X			X		X		X	X
6. Time of day	X	X	X	X		X			X		X		X	X
7. Throughput class		X	X											
8. D bit usage			X											
9. Inter-packet gaps		X	X											
CSD – Call set-up delay DPTD – Data packet transfer delay TC – Throughput capacity CID – Clear indication delay CSEP – Call set-up error probability CSFP – Call set-up failure probability RER – Residual error ratio					RSP – Reset stimulus probability RP – Reset probability PDSP – Premature disconnect stimulus probability PDP – Premature disconnect probability CCFP – Call clear failure probability SA – Service availability MTBSO – Mean time between service outages									

### 7.4.3 Throughput measurement correction factor

In the case where throughput is measured by observation of exit events, a correction factor can be applied to the throughput measurement formula which will account for any difference in the size of  $A_0$  and  $A_k$  or  $\hat{A}_0$  and  $\hat{A}_m$ .<sup>19)</sup>

The correction factor in the case where exit events are observed at  $B_i$  is as follows:

<u>Condition</u>	<u>Correction</u>
$f(A_0) = f(A_k)$	No correction
$f(A_0) > f(A_k)$	Subtract the insertion time at boundary $B_i$ of the excess user data bits $[f(A_0) - f(A_k)]$ in $A_0$ from $t_2 - t_1$
$f(A_0) < f(A_k)$	Add the insertion time at boundary $B_i$ of the deficient user data bits $[f(A_k) - f(A_0)]$ in $\hat{A}_0$ to $t_2 - t_1$

The analogous table for the case where exit events are observed at  $B_j$  is as follows:

<u>Condition</u>	<u>Correction</u>
$f(\hat{A}_0) = f(\hat{A}_m)$	No correction
$f(\hat{A}_0) > f(\hat{A}_m)$	Subtract the insertion time at boundary $B_j$ of the excess user data bits $[f(\hat{A}_0) - f(\hat{A}_m)]$ in $\hat{A}_0$ from $t_4 - t_3$ .
$f(\hat{A}_0) < f(\hat{A}_m)$	Add the insertion time at boundary $B_j$ of the deficient user data bits $[f(\hat{A}_m) - f(\hat{A}_0)]$ in $\hat{A}_0$ to $t_4 - t_3$ .

The error introduced by not applying the correction factor will normally be very small.

## 8 Performance specifications

This clause specifies worst-case design objectives for the packet-switched service performance parameters defined in clauses 4 (speed of service), 5 (accuracy and dependability), and 6 (availability): call set-up delay, data packet transfer delay, throughput capacity, clear indication delay, call set-up error probability, residual error ratio, reset stimulus probability, reset probability, premature disconnect stimulus probability, premature disconnect probability, call clear failure probability, service availability, and mean time between service outages.

Subclause 8.1 clarifies the intended use and applicability of the design objectives specified in this standard and identifies some general assumptions that underlie them. Subclause 8.2 extends the performance model of clause 3 to define general portion types used for performance specification and allocation. Subclause 8.3 specifies the worst-case design objectives. Annex B provides example methods for estimation of end-to-end performance values.

### 8.1 Intended use, applicability, and general assumptions

The design objectives specified in this standard are intended to be used as worst-case limits in the planning of national packet-switched services. The intended users of this standard include packet-switched service providers, equipment manufacturers, and end users (i.e., customers of packet-switched service or equipment providers). This standard may be used (1) by service providers in the planning, development, and assessment of packet-switched services that meet user performance needs, (2) by equipment manufacturers as minimum performance requirements and metrics that will impact equipment design, and (3) by users as a common basis for evaluation of system performance and as information relevant to the

<sup>19)</sup> Notation used here is defined in 4.3.1.

matching of user needs with alternative provider offerings, including preparation of performance requirements in procurement. In the context of this standard, the term “worst-case” means that the design objectives should be met by all portions of any end-to-end virtual connection configured and used in accordance with the conditions and assumptions identified in this subclause.

As further described in 8.2, the access and transit network design objectives specified in 8.3 should be met in any end-to-end packet-switched service that is offered through the interconnection of two or more separate network service providers. Rules for combining portion values to estimate the end-to-end performance of concatenated portions are provided in annex B.

Two sets of values are specified for access portions depending on their interconnection to the transit portion. One set of values applies when the access portion is nominally interconnected to the transit portion. The other set applies when the access and transit portion providers cooperatively plan the interconnection. These two sets of values are called nominal interconnection values and cooperatively planned interconnection values, respectively. An example of a cooperatively planned interconnection is one that reduces the number of switches between the NI and the INI.

All values specified in this standard are based on (and only apply under) the following general assumptions:

- 1) Values for primary parameters exclude performance observed during periods of unavailability (see clause 6);
- 2) For each connection portion type, the ten worst-performing days in a calendar year are excluded;
- 3) The values apply to packet-switched services provided using terrestrial transmission systems.

NOTE – Satellite transmission systems can form parts of or provide the sole transmission means for packet-switched data networks. Satellite links may appear in both the access and transit portions of the end-to-end virtual connection. The ITU-T Recommendation X.25 and X.75 options (e.g., modulo and window size) and the internal protocols used in the connection should accommodate the inherent propagation delays of satellite links. Due to the built-in multipoint connectivity capability of satellite transmission systems, the topology of packet-switched networks may be simplified by reducing the number of switches in the network. For planning purposes, the one-way propagation delay for a satellite link is considered to be 270 milliseconds.

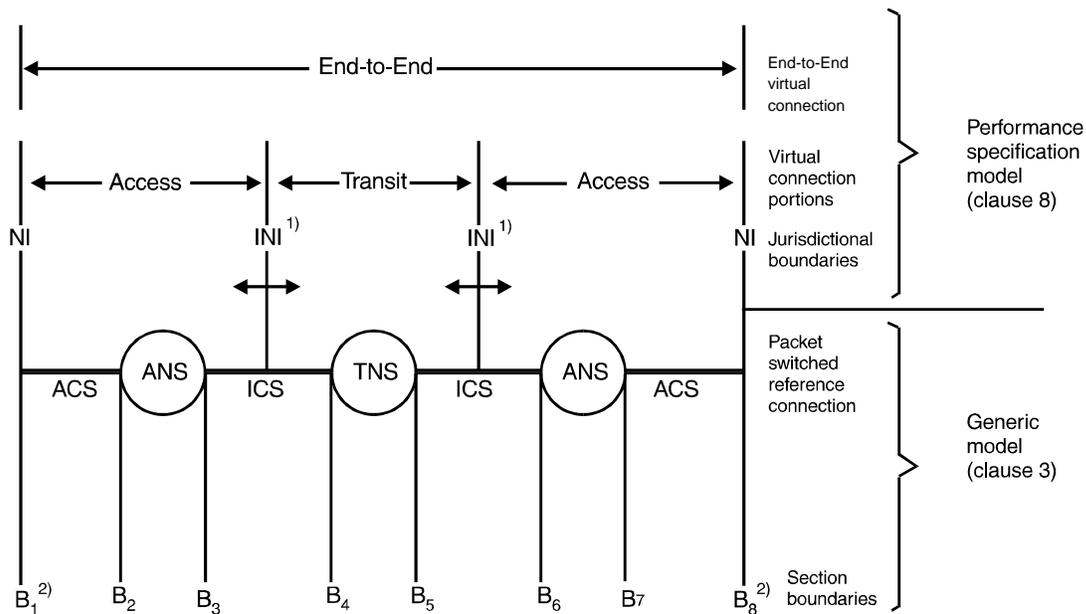
The design objectives specified in this standard may not be achievable in all network configurations existing at the time of its publication. They represent agreements among service providers and end users applicable to the planned evolution of packet-switched services and their interconnection. The actual values achieved in a virtual connection portion will depend on many factors, including the traffic expected and actually offered, the internal network topology, and the signalling rates on the access and internetwork circuit sections. Variation away from the worst-case value for each factor can improve the performance.

## **8.2 Performance specification and apportionment model**

This subclause adapts the generic performance model defined in clause 3 to provide a jurisdictional basis for the specification and apportionment of packet-switched service performance. The generic model partitions an end-to-end virtual connection into basic sections and defines a set of performance-significant reference events that may be observed at the section boundaries. The performance specification model adds jurisdictional boundaries, defines associated access and transit connection portions, and adapts the generic reference event definitions for application at the jurisdictional boundaries. The jurisdictional boundaries and associated portion types are defined in 8.2.1. The reference event definitions applicable at the jurisdictional boundaries are presented in 8.2.2.

Figure 30 illustrates the jurisdictional boundaries, NI and INI, and associated access and transit virtual connection portions. A representative end-to-end virtual connection is divided into two access portions and an intervening transit portion. Although other jurisdictional arrangements are possible, an access portion generally comprises facilities provided by an “exchange carrier” and a transit portion generally comprises facilities provided by an “interexchange carrier”. Worst-case design objectives are specified in 8.3 for the access and transit portions. Example methods for calculating end-to-end values for concatenations of access and transit portions are described in annex B.

The measurement methods defined in clause 7 may be applied in direct measurements at the defined jurisdictional boundaries, or may be used to estimate the performance provided at jurisdictional boundaries on the basis of observations made at the adjacent section boundaries taking account of known characteristics of the access or transit circuits that connect equipment in separate jurisdictions.<sup>20)</sup> When performance values are based on such estimates, the corresponding estimation errors should also be reported.



Legend:

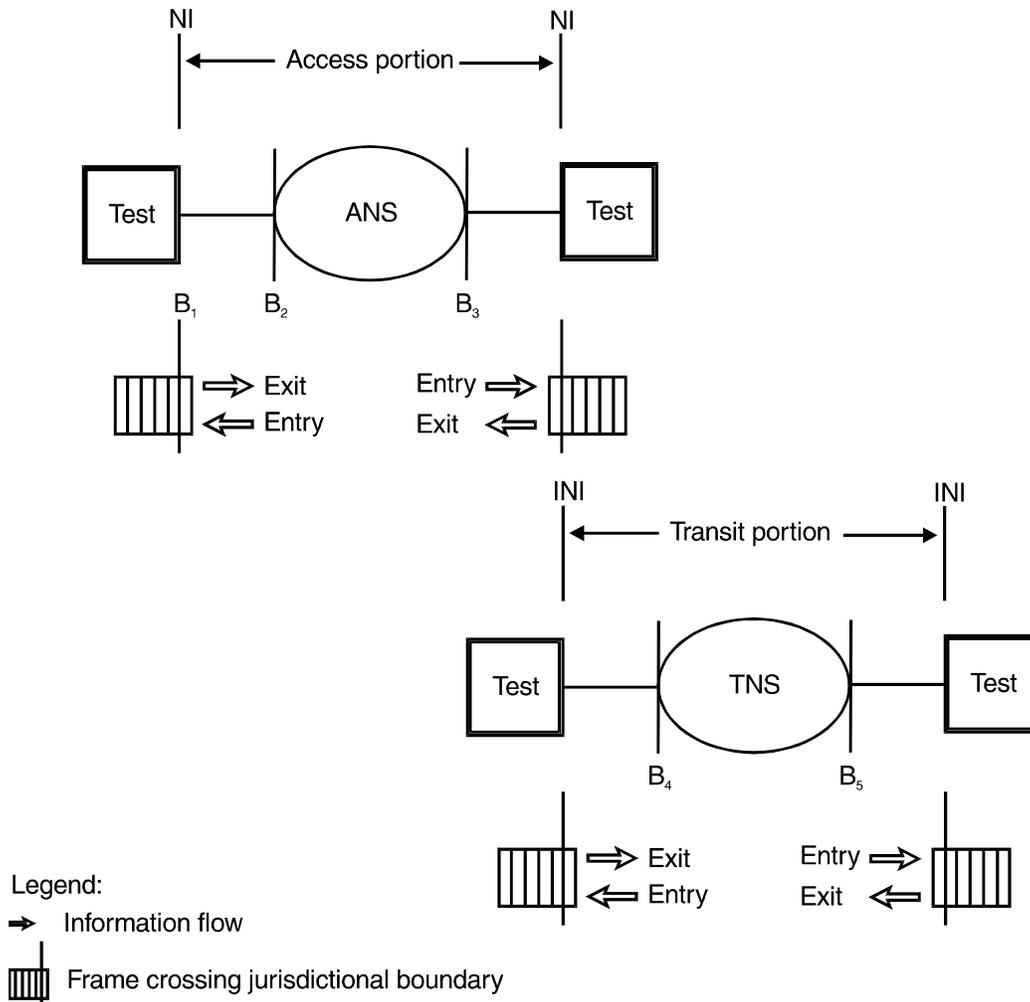
- |   |                               |
|---|-------------------------------|
| ACS = Access circuit section                              | INI = Internetwork Interface  |
| ANS = Access network section                              | NI = Network Interface        |
| B <sub>i</sub> = Section boundary I                       | TNS = Transit network section |
| ICS = Internetwork circuit section<br>(assumed 56 kbit/s) |                               |

<sup>1)</sup> Location of the INI depends on arrangements between the service providers.

<sup>2)</sup> Section boundaries, B<sub>1</sub> and B<sub>8</sub> may be coincident with the NI or may be created at the NI by an attached DTE.

**Figure 30 - National packet-switched service performance allocation model**

<sup>20)</sup> Direct measurement of performance at the defined jurisdictional boundaries will not always be practical. Section boundaries serve as practical surrogates for the jurisdictional boundaries in many cases.



**Figure 31 - Definition of reference events at the NI and INI boundaries**

### 8.2.1 Definitions

In the context of this standard, the following definitions apply:

**8.2.1.1 network interface (NI):** The NI is the jurisdictional boundary between the customer's and the network provider's equipment. This standard uses the NI boundaries to allocate packet-switched service performance responsibilities between the customers and the network service providers. Section boundaries  $B_1$  and  $B_8$ , which separate the DTE and network functions, are considered to be coincident with the NIs for measurement purposes.<sup>21)</sup>

**8.2.1.2 internetwork interface (INI):** The INI is the jurisdictional boundary between the access provider's and the transit network provider's equipment. In connections involving four or more networks, INIs are also jurisdictional boundaries between transit network providers. This standard uses the INI boundaries to

<sup>21)</sup> As indicated in figure 30, section boundaries  $B_1$  and  $B_8$  may be coincident with the NI or may be created at the NI by an attached DTE.

allocate packet-switched service performance responsibilities between the access network provider and the transit network provider (see 8.1 and annex A for a discussion of relevant performance concepts).

**8.2.1.3 access portion:** The access portion is the portion of an end-to-end packet-switched service connection between adjacent NI and INI interfaces. An access portion carries user information and signalling information between the network and internetwork interfaces. An access portion typically includes an access circuit section and an access network section.

**8.2.1.4 transit portion:** The transit portion is the portion of an end-to-end packet-switched service connection between the two internetwork interfaces. A transit portion carries user information and signalling information between two INIs. A transit portion typically includes a transit network section.

## 8.2.2 Reference event concepts

Reference event concepts apply as defined in 3.2 with the following additional concepts that apply in the context of clause 8 only. These extensions of the concepts defined in 3.2 are required to clarify application of the performance parameters at the jurisdictional boundaries.

Reference events are defined with respect to hypothetical test equipment (see figure 31). The hypothetical test equipment is physically outside of the portion under evaluation and performs all of the interface functions of the DTE or STE that it replaces. The performance of this hypothetical test equipment is assumed to be nominal (i.e., not degraded or exceptional). The rationale for placing the hypothetical test equipment at the NI and INI is to eliminate the performance degradations of circuits and equipment not part of the portion under evaluation.

A *packet layer reference event* is the event that occurs when a packet crossing a jurisdictional boundary changes the state of the packet layer interface of the hypothetical test equipment connected at the boundary.

As shown in figure 31, two classes of packet layer reference events are defined:

- 1) *Packet entry event:* A packet layer reference event that corresponds to a packet entering the hypothetical test equipment at an NI or INI boundary;
- 2) *Packet exit event:* A packet layer reference event that corresponds to a packet exiting the hypothetical test equipment at an NI or INI boundary.

The time of occurrence of a packet entry event is defined to coincide with the time at which the last bit of the closing flag of the frame carrying the packet crosses the relevant jurisdictional boundary. The time of occurrence of a packet exit event is defined to coincide with the time at which the first bit of the address field of the frame carrying the packet crosses the relevant jurisdictional boundary. If retransmissions occur, the packet entry event occurs with the last transmission and the packet exit event occurs with the first transmission.

## 8.3 Worst-case design objectives

This subclause specifies the worst-case design objectives for the packet-switched performance parameters identified above. Specific assumptions are noted with respect to each specification table.

### 8.3.1 Call set-up delay

Table 14 specifies the worst-case design objectives for call set-up delay for each of the two virtual connection portion types defined in 8.2. All values are based on (and only apply under) the following assumptions:

- 1) A basic call, in which none of the optional user facilities defined in ITU-T Recommendation X.25 are used and no called user data is sent;
- 2) Data link layer windows of entities outside the portion being specified are open (not flow controlled).

The defined objectives are limits on the means.

In table 14, the value  $X$  depends on the signalling rate of the access circuit.

Table 15 presents the  $X$  values for typical access circuit signaling rates.<sup>22)</sup> Other  $X$  values may be computed using the formula:

$$X = \frac{400}{R} \text{ ms,}$$

where  $R$  is the signalling rate in kilobits per second.<sup>23)</sup>

**Table 14 – Call set-up delay: Worst-case design objectives**

	Virtual connection portion		
	Access		Transit
	Nominal interconnection	Cooperatively planned interconnection	
<b>Mean (ms)<sup>1)</sup></b>	800 + $X$	450 + $X$	550
<sup>1)</sup> It is expected that 95 percent of measured call set-up delays will be within a factor of 2 of the mean. NOTE - See explanatory note in 8.1.			

**Table 15 – Values of  $X$  for table 14**

R (kbit/s)	X (ms)
2.4	167.0
4.8	83.4
9.6	41.7
48.0	8.34
56.0	7.15
64.0	6.25

<sup>22)</sup> These  $X$  values do not represent the delay performance of the access circuit, since these values do not include propagation delays, multiplexing delays, or the effects of retransmission.

<sup>23)</sup> The formula assumes that the transfer of each call set-up packet (i.e., both the Call Request Packet and the corresponding Call Accepted Packet) across an access circuit involves the transmission of 25 octets: 5 octets of frame level overhead, a 5-octet packet header, and 15 octets of DTE address information.

### 8.3.2 Data packet transfer delay

Table 16 specifies the worst-case design objectives for data packet transfer delay for each of the two virtual connection portion types defined in 8.2. All values are based on (and only apply under) the following assumptions:

- 1) A user data field length of 128 octets;
- 2) Data link and packet layer windows on the receiving side of the portion being specified are open.

The defined objectives are limits on the means.

In table 16, the value  $Y$  depends on the signalling rate of the access circuit. Table 17 presents the  $Y$  values for typical access circuit signalling rates.<sup>24)</sup> Other  $Y$  values may be computed using the formula:

$$Y = \frac{1088}{R} \text{ ms,}$$

where  $R$  is the signalling rate in kilobits per second.<sup>25)</sup>

**Table 16 – Data packet transfer delay: Worst-case design objectives**

	Virtual connection portion		
	Access		Transit
	Nominal interconnection	Cooperatively planned interconnection	
<b>Mean (ms)<sup>1)</sup></b>	600 + $Y$	200 + $Y$	300
<sup>1)</sup> It is expected that 95 percent of measured data packet transfer delays will be within a factor of 2 of the mean. NOTE - See explanatory note in 8.1.			

**Table 17 – Values of  $Y$  for table 16**

R (kbit/s)	Y (ms)
2.4	454.0
4.8	227.0
9.6	114.0
48.0	22.7
56.0	19.5
64.0	17.0

<sup>24)</sup> These  $Y$  values do not represent the delay performance of the access circuit, since these values do not include propagation delays, multiplexing delays, or the effects of retransmission.

<sup>25)</sup> The formula assumes that the transfer of a data packet across an access circuit involves the transmission of 136 octets: 5 octets of frame level overhead, a 3-octet packet header, and 128 octets of user information.

### 8.3.3 Throughput capacity

Table 18 specifies the worst-case design objectives for throughput capacity for each of the two virtual connection portion types defined in 8.2. All values are based on (and only apply under) the following assumptions:

- 1) no additional traffic on the access links;
- 2) 9600-bit/s signalling rates on the access links (applicability of the specified throughput capacity values to lower access link signalling rates is for further study);
- 3) a user data field length of 128 octets and requested throughput class corresponding to 9600 bit/s (NOTE – The throughput class finally applying to the call may be lower than the requested throughput class);
- 4) sufficiently large packet layer window sizes and data link layer window sizes on the access links;
- 5) D bit not used ( $D = 0$ );<sup>26)</sup>
- 6) values apply to either direction of transfer;
- 7) no resets or premature disconnects during the observation period;
- 8) throughput capacity sample sizes of 400 packets or more or at least 2 minutes.

The throughput capacity values defined here will not necessarily be achieved concurrently with the delay values defined in table 16.

**Table 18 – Throughput capacity: Worst-case design objectives**

	Virtual connection portion		
	Access		Transit
	Nominal interconnection	Cooperatively planned interconnection	
bits/s	4800	4800	4800

### 8.3.4 Clear indication delay

Table 19 specifies the worst-case design objectives for clear indication delay for each of the two virtual connection portion types defined in 8.2. All values are based on (and only apply under) the following assumptions:

- 1) Data link layer windows on the receiving side of the portion being specified are open;
- 2) The extended format of the clear request packet is not used.

The defined objectives are limits on the means.

<sup>26)</sup> An “internal D bit” or other end-to-end packet acknowledgment may be used to enhance transfer reliability. Such mechanisms will tend to reduce throughput capacity.

In table 19, the value  $Z$  depends on the signalling rate of the access circuit. Table 20 presents the  $Z$  values for typical access circuit signalling rates.<sup>27)</sup> Other  $Z$  values may be computed using the formula:

$$Z = \frac{80}{R} \text{ ms ,}$$

where  $R$  is the signalling rate in kilobits per second.<sup>28)</sup>

**Table 19 – Clear indication delay: Worst-case design objectives**

	Virtual connection portion		
	Access		Transit
	Nominal interconnection	Cooperatively planned interconnection	
<b>Mean (ms)<sup>1)</sup></b>	560 + $Z$	250 + $Z$	300
<sup>1)</sup> It is expected that 95 percent of measured clear indication delays will be within a factor of 2 of the mean. NOTE - See explanatory note in 8.1.			

**Table 20 – Values of  $Z$  for table 19**

R (kbit/s)	Z (ms)
2.4	33.4
4.8	16.7
9.6	8.34
48.0	1.67
56.0	1.43
64.0	1.25

### 8.3.5 Call set-up error probability

Table 21 specifies the worst-case design objectives for call set-up error probability for each of the two virtual connection portion types defined in 8.2.

<sup>27)</sup> These  $Z$  values do not represent the delay performance of the access circuit, since these values do not include propagation delays, multiplexing delays, or the effects of retransmission.

<sup>28)</sup> The formula assumes that the transfer of each call clearing packet across an access circuit involves the transmission of 10 octets: 5 octets of frame level overhead and 5 octets of packet header information.

**Table 21 – Call set-up error probability: Worst-case design objectives**

	Virtual connection portion		
	Access		Transit
	Nominal interconnection	Cooperatively planned interconnection	
<b>Probability</b>	$1 \times 10^{-5}$	$1 \times 10^{-5}$	$1 \times 10^{-5}$

### 8.3.6 Call set-up failure probability

Table 22 specifies the worst-case design objectives for call set-up failure probability for each of the two virtual connection portion types defined in 8.2.

**Table 22 – Call set-up failure probability: Worst-case design objectives**

	Virtual connection portion		
	Access		Transit
	Nominal interconnection	Cooperatively planned interconnection	
<b>Probability</b>	$5 \times 10^{-3}$	$5 \times 10^{-3}$	$5 \times 10^{-3}$

### 8.3.7 Residual error ratio

Table 23 specifies the worst-case design objectives for residual error ratio for each of the two virtual connection portion types defined in 8.2. All values are based on (and only apply under) the assumption that the user information field of the data packet is 128 octets.

**Table 23 – Residual error ratio: Worst-case design objectives**

	Virtual connection portion		
	Access		Transit
	Nominal interconnection	Cooperatively planned interconnection	
<b>Probability</b>	$1 \times 10^{-9}$	$1 \times 10^{-9}$	$1 \times 10^{-9}$

### 8.3.8 Reset stimulus probability

Table 24 specifies the worst-case design objectives for reset stimulus probability for each of the two virtual connection portion types defined in 8.2.

**Table 24 – Reset stimulus probability: Worst-case design objectives**

	Virtual connection portion		
	Access		Transit
	Nominal interconnection	Cooperatively planned interconnection	
<b>Reset stimuli per VC-sec<sup>1)</sup></b>	$1 \times 10^{-6}$	$1 \times 10^{-6}$	$1 \times 10^{-6}$
<sup>1)</sup> The probability is the reset stimuli per virtual circuit (VC) second.			

**8.3.9 Reset probability**

Table 25 specifies the worst-case design objectives for reset probability for each of the two virtual connection portion types defined in 8.2.

**Table 25 – Reset probability: Worst-case design objectives**

	Virtual connection portion		
	Access		Transit
	Nominal interconnection	Cooperatively planned interconnection	
<b>Resets per VC-sec<sup>1)</sup></b>	$5 \times 10^{-6}$	$5 \times 10^{-6}$	$5 \times 10^{-6}$
<sup>1)</sup> The probability is the resets per virtual circuit (VC) second.			

**8.3.10 Premature disconnect stimulus probability**

Table 26 specifies the worst-case design objectives for premature disconnect stimulus probability for each of the two virtual connection portion types defined in 8.2.

**Table 26 – Premature disconnect stimulus probability: Worst-case design objectives**

	Virtual connection portion		
	Access		Transit
	Nominal interconnection	Cooperatively planned interconnection	
<b>Premature disconnect stimuli per VC-sec<sup>1)</sup></b>	$1 \times 10^{-7}$	$1 \times 10^{-7}$	$1 \times 10^{-7}$
<sup>1)</sup> The probability is the premature disconnect stimuli per virtual circuit (VC) second.			

### 8.3.11 Premature disconnect probability

Table 27 specifies the worst-case design objectives for premature disconnect probability for each of the two virtual connection portion types defined in 8.2.

**Table 27 – Premature disconnect probability: Worst-case design objectives**

	Virtual connection portion		
	Access		Transit
	Nominal interconnection	Cooperatively planned interconnection	
Premature disconnects per VC-sec. <sup>1)</sup>	$3 \times 10^{-6}$	$3 \times 10^{-6}$	$3 \times 10^{-6}$
<sup>1)</sup> The probability is the premature disconnects per virtual circuit (VC) second.			

### 8.3.12 Call clear failure probability

Table 28 specifies the worst-case design objectives for call clear failure probability for each of the two virtual connection portion types defined in 8.2. All values are based on (and only apply under) the following assumptions:

- 1) Data link layer windows on the receiving side of the portion being specified are open;
- 2) The extended format of the clear request packet is not used.

**Table 28 – Call clear failure probability: Worst-case design objectives**

	Virtual connection portion		
	Access		Transit
	Nominal interconnection	Cooperatively planned interconnection	
Probability	$1 \times 10^{-5}$	$1 \times 10^{-5}$	$1 \times 10^{-5}$

### 8.3.13 Service availability

Table 29 specifies the worst-case design objectives for service availability for each of the two virtual connection portion types defined in 8.2. All values are based on (and only apply under) the assumption that the threshold constants defined in table 10 have the following values:  $C_1 = 0.9$ ,  $C_2 = 80$ ,  $C_3 = 10^{-3}$ ,  $C_4 = 0.015$ , and  $C_5 = 0.01$ .

**Table 29 – Service availability: Worst-case design objectives**

	Virtual connection portion		
	Access		Transit
	Nominal interconnection	Cooperatively planned interconnection	
<b>Percent</b>	99.7	99.7	99.7

**8.3.14 Mean time between service outages**

Table 30 specifies the worst-case design objectives for mean time between service outages (MTBSO) for each of the two virtual connection portion types defined in 8.2. All values are based on (and only apply under) the assumption that the threshold constants defined in table 10 have the following values:  $C_1 = 0.9$ ,  $C_2 = 80$ ,  $C_3 = 10^{-3}$ ,  $C_4 = 0.015$ , and  $C_5 = 0.01$ .

**Table 30 – Mean time between service outages (MTBSO): Worst-case design objectives**

	Virtual connection portion		
	Access		Transit
	Nominal interconnection	Cooperatively planned interconnection	
<b>Hours</b>	1300	1400	1300

## Annex A (informative)

### Statistical formulas

Various statistical formulas are given below for the sake of completeness and to eliminate the confusion that often results from comparison of formulas for sample (estimated) parameters and distributional (population) parameters.

#### A.1 Sample mean

As the arithmetical average of a set of observations, the observed mean is used as an estimate of the true mean of the underlying distribution. Under rather general conditions the mean,

$$\bar{x} = \frac{\sum_{k=1}^n x_k}{n}$$

is the Maximum Likelihood Estimator of the true distributional mean.

A mean should always be reported in conjunction with either a sample variance, a 95th percentile, a 99th percentile, or a 95% confidence interval about it.

#### A.2 Sample variance

The sample variance,  $s(X)$ , or  $s$ , of a sequence of observations, is defined to be:

$$\frac{\sum_{k=1}^n (x_k - \bar{x})^2}{n - 1},$$

and is used to obtain estimates of the precision of the estimated mean,  $\bar{x}$ .

The use of the factor  $1/(n-1)$  assures that the estimator will be unbiased in the sense that  $E(s(X)) = V$ , where  $V$  is the true variance of the underlying distribution.

#### A.3 Xth percentile

The  $x$ th percentile ( $0 < x < 100$ ) of a cumulative continuous distribution,  $F$ , is any number  $Y$  satisfying the equation  $F(Y) = x/100$ . If  $F$  is also strictly increasing,  $Y$  is unique. For discrete distributions, most percentiles are not unique. In this case, any  $Y$  for which  $F(Y)$  is minimal, subject to the condition that  $F(Y) \geq x$ , is an  $x$ th percentile.

For example, the 95th percentile of a set of measurements would be any number  $Y$  for which

- 1) at least 95% of the measurements fall below  $Y$ ; and
- 2) the number of such measurements below  $Y$  is minimal.

#### A.4 Minimum

The minimum of a set of measurements is the least value attained by any measurement in the set.

#### A.5 Maximum

The maximum of a set of measurements is the greatest value attained by any measurement in the set.

## Annex B (informative)

### Methods for estimating performance of concatenated access and transit virtual connection portions

This annex provides practical methods for estimating the performance of a set of concatenated access and transit network virtual connection portion types defined in 8.2. Clause B.1 provides general concatenation methods for the packet-switched parameters specified in 8.3. Clause B.2 provides representative examples that illustrate use of the methods described in B.1.

#### B.1 General methods

The following general methods can be used to calculate the approximate concatenated performance of a set of access and transit portions. Although alternative network models and statistical assumptions are possible, the methods presented in this annex provide one practical way of estimating end-to-end performance from the performance of the individual network portions.

##### B.1.1 Call set-up delay, data packet transfer delay, clear indication delay

The approximate mean delay,  $D$ , of concatenated access and transit portions can be computed as:

$$D = d_1 + d_2 + \dots + d_N - (N - 1) \times F,$$

where  $d_i$  is the mean delay associated with portion  $i$  ( $i = 1, 2, \dots, N$ );  $N$  is the total number of access and transit portions that compose the concatenated virtual connection; and the value of the correction factor  $F$  depends on the parameter of interest and is approximately 7.15 ms for call set-up delay, 19.5 ms for data packet transfer delay, and 1.43 ms for clear indication delay.<sup>29</sup> The factor  $(N - 1) \times F$  is an approximate correction for overlap in the times of occurrence of the relevant entry and exit reference events when access and transit delays are concatenated at  $(N - 1)$  INI boundaries (see 8.2.2). The methods used for estimating delays for concatenated network portions assume that the effect of retransmission on mean delay is negligible.

##### B.1.2 Throughput capacity

An upper bound for the concatenated performance of a set of access and transit portions for the throughput capacity can be estimated from the individual portion throughput capacities using the methods described in 4.3.3, where references to boundaries  $B_i$ ,  $B_j$ , and  $B_K$  are understood to be the relevant NI and INI boundaries.<sup>30)</sup>

##### B.1.3 Call set-up error probability, call set-up failure probability, residual error ratio, call clear failure probability

Assuming that the accuracy and dependability performance of the individual network portions are statistically independent, then a very close approximation to the concatenated performance of a set of access and transit network portions can be obtained for the call set-up error probability, call set-up failure probability, residual error ratio, and call clear failure probability parameters by simply summing the corresponding values for the individual network portions. This is justified because the expected probabilities are small.

<sup>29)</sup> The value of  $F$  used in B.1.1 was found by referencing the relevant table in 8.3 (table 15 for call set-delay, table 17 for data packet transfer delay, table 20 for clear indication delay) for the appropriate  $R$  value (the provisional design objectives specified in 8.3 assume a 56-kbit/s internetwork circuit signaling rate, which is applicable at an INI boundary).

<sup>30)</sup> Additional information supporting the practicality of this upper bound can be found in C.3.2 of ITU-T Recommendation X.135.

#### B.1.4 Reset stimulus probability, reset probability, disconnect stimulus probability, disconnect probability

The concatenated performance of a set of access and transit network portions can be estimated for the reset stimulus probability, reset probability, disconnect stimulus probability, and disconnect probability parameters using the methods described in 5.2.2.2 and 5.2.3.2, where references to boundaries  $B_i$ ,  $B_j$ , and  $B_k$  are understood to be the relevant NI and INI boundaries.

#### B.1.5 Service availability, mean time between service outages (MTBSO)

Assuming that the service availability performance values associated with the individual network portions are statistically independent, then the concatenated service availability performance of a set of access and transit portions can be estimated by multiplying the percents of time each of the network portions is available.

The concatenated MTBSO performance of a set of access and transit portions can be estimated by assuming that the times between service outages in each individual network portion are statistically independent and exponentially distributed. Under these assumptions, the end-to-end MTBSO performance design objective,  $T$ , can be calculated using the following formula:

$$T = [T_1^{-1} + T_2^{-1} + \dots + T_i^{-1} + \dots + T_N^{-1}]^{-1},$$

where  $T$  will be in hours if the MTBSO for each of the  $N$  network portions,  $T_i$  ( $i = 1, 2, \dots, N$ ), is expressed in hours.

## B.2 Illustrative examples

This clause provides examples that illustrate use of the general performance concatenation methods described in clause B.1. The examples estimate the concatenated performance of two access portions both nominally interconnected with an intervening transit portion for selected parameters specified in 8.3.

### B.2.1 Call set-up delay

Referring to the design objectives specified in table 14, the relevant values for this example are:  $800 + X$  ms (access portion) and 600 ms (transit portion). Substituting these values in the formula in B.1.1 with  $N = 3$  (two access portions and one transit portion, which results in the overall virtual connection crossing two INI boundaries) and  $F = 7.15$  ms (for call set-up delay) gives  $D = (800 + X) + (600) + (800 + X) - 2 \times (7.15) = 2186 + 2 \times X$  ms.

### B.2.2 Throughput capacity

Referring to the methods described in 4.3.3 (see B.1.3), the following formula may be used to estimate the maximum overall throughput capacity of  $N$  concatenated access and transit portions:  $TC \leq \text{MIN}[TC_1, TC_2, \dots, TC_i, \dots, TC_N]$ , where  $TC_i$  is the throughput capacity for portion  $i$  ( $i = 1, 2, \dots, N$ ). For this example, there are  $N = 3$  portions, so  $i = 1, 2, 3$ . Referring to the design objectives (provisionally) specified in table 18, the relevant values are:  $TC_1 = TC_3 = 4800$  b/s (access portion) and  $TC_2 = 4800$  b/s (transit portion). Substituting these values in the above formula gives  $TC \leq \text{MIN}[4800, 4800, 4800] = 4800$  b/s.

### B.2.3 Call set-up error probability

Referring to the design objectives specified in table 21, the relevant values for this example are:  $1.0 \times 10^{-5}$  (access portion) and  $1.0 \times 10^{-5}$  (transit portion). Summing these values for two access portions and one transit portion yields a practical estimate of the concatenated performance (see B.1.4):  $1.0 \times 10^{-5} + 1.0 \times 10^{-5} + 1.0 \times 10^{-5} = 3.0 \times 10^{-5}$ .

### B.2.4 Reset stimulus probability and reset probability

Referring to the design objectives specified in table 24 and table 25, the relevant values for the reset stimulus probability are  $1.0 \times 10^{-6}$  (access portion) and  $1.0 \times 10^{-6}$  (transit portion) and for the reset probability are  $1.0 \times 10^{-5}$  (access portion) and  $1.0 \times 10^{-5}$  (transit portion). With reference to figure B.1, the methods described in 5.2.2.2 (see B.1.5) can be used to estimate the concatenated performance. In this example, the reset stimulus probability for the set of concatenated portions is  $S_{11} = S_{34} = 1.0 \times 10^{-6}$ . In this example, the reset probability for the set of concatenated portions can be estimated as  $(R_1 + S_{1_1} + S_{1_2} + R_2 + S_{2_2} + S_{2_3} + R_3 + S_{3_3} + S_{3_4}) = 1 \times 10^{-5} + 1 \times 10^{-6} + 1 \times 10^{-6} + 1 \times 10^{-5} + 1 \times 10^{-6} + 1 \times 10^{-6} + 1 \times 10^{-5} + 1 \times 10^{-6} + 1 \times 10^{-6} + 1 \times 10^{-6} = 3.4 \times 10^{-5}$ .

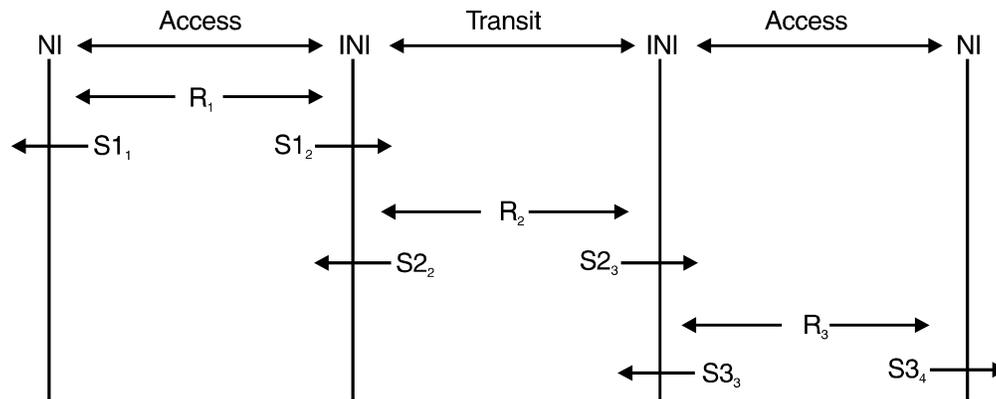


Figure B.1 - Reset stimulus and reset probabilities (B.2.4)

### B.2.5 Service availability

Referring to the design objectives specified in table 29, the relevant values for the service availability are 99.7% (access portion) and 99.7% (transit portion). A practical estimate of the concatenated performance for this example can be made by multiplying the service availabilities specified for the individual network portions (see B.1.5):  $99.7 \times 99.7 \times 99.7 = 99.1\%$ .

### B.2.6 Mean time between service outages (MTBSO)

Referring to the design objectives specified in table 30, the relevant values for the MTBSO are 1300 hours (access portion) and 1300 hours (transit portion). A practical estimate of the concatenated performance for this example can be made by substituting these values into the formula in B.1.5:  $T = [1300^{-1} + 1300^{-1} + 1300^{-1}]^{-1} = 433$  hours.

## **Annex C** (informative)

### **Sampling estimation of availability parameters**

#### **C.1 Test of availability**

The definition of availability requires that observed performance for all five decision parameters be compared with outage thresholds. A single success of the following test is defined to be sufficient for declaring the virtual connection section available. A single failure of a section to meet any of the six individual decision criteria is defined to be sufficient for declaring the virtual section unavailability of the section.

The minimal availability test can be initiated in either direction across the section by equipment and components outside of the section. The test is divided into two phases: access and user information transfer. The access phase is used in conjunction with switched virtual calls only.

In those situations where greater flexibility in managing the Type I and Type II errors is required, an alternative non-minimal test (the SPRT methodology described in C.4) should be used.

##### **C.1.1 A minimal test of availability**

Phase I: Attempt 4 consecutive call set-ups across A.

Phase II: (If the test did not fail in Phase I) To ensure that the availability test does not fail as a result of insufficient data input, attempt to maintain a virtual connection across A for 5 minutes. Attempt to maintain an average throughput significantly greater than 80 bit/s (e.g. at least 150 bit/s) during that interval.

There are six criteria for deciding if the test has failed or succeeded:

- 1) The test fails in Phase I if all four call set-up attempts result in either call set-up error or call set-up failure (switched virtual calls only).

A statistical analysis in this Phase I test is presented in C.1.2. As an alternative the SPRT methodology presented in C.4 may be used in place of the above Phase I test. The SPRT methodology provides more flexibility in controlling Type I and Type II errors.

- 2) The test fails in Phase II if the total reset events plus reset stimuli is five or greater.
- 3) The test fails in Phase II if the throughput is less than 80 bit/s.
- 4) The test fails in Phase II if the residual error ratio is greater than  $10^{-3}$ .
- 5) The test fails in Phase II if the call and subsequent reestablishments of that call are cleared two or more times due to premature disconnects and/or premature disconnect stimuli (switched virtual calls only).
- 6) The test fails in Phase I or Phase II if a data link layer at a section boundary is unavailable during a 5-minute interval due to causes inside A.

If the test passes all six decision criteria, the test is successful and the virtual connection section A is considered to be available during the test. If any of the decision criteria are failed, the virtual connection section A is considered to have been unavailable for the duration of the test.

Because many performance parameters must be supported simultaneously in order for A to be considered available, during normal operation (without a testing procedure like the one described above) it is not possible to prove the section is available (e.g., it may not be possible to observe both access and user information transfer simultaneously). Therefore during normal operation, if the section is correctly performing the currently requested function, the section is assumed to be available.

Service availability and mean time between service outage values can be estimated on the basis of this minimal test (availability performance samples). Such estimation is more practical than measurement based on continuous service observation.

### C.1.2 The statistical basis for the Phase I test with N = 4

By definition the service is unavailable if the probability of call set-up error plus the probability of call set-up failure is greater than 0.9:

$$cfp + cep > 0.9$$

Therefore we take the following as the null hypotheses,  $H_0$ , and the alternative hypothesis,  $H_a$ :

$$H_0: cep + cfp < 0.9$$

$$H_a: cep + cfp > 0.9$$

Using the X.137 minimal availability test, the probability of Type I and Type II errors are given below:

$$\Pr(\text{Type I error}) < z^4 \approx 0.24 \text{ (for } z = 0.7\text{)}$$

$$\Pr(\text{Type II error}) < 1 - (0.9)^4 \approx 0.35$$

Table C.1 presents the probabilities of various events given the actual level of call set-up failure and error probability:

**Table C.1 - Error performance of the minimal test**

Actual cep + cfp	Probability of correctly identifying the available state	Probability of correctly identifying the unavailable state	Probability of identifying the available state as unavailable Pr (Type I error)	Probability of identifying the unavailable estate as available Pr (Type II error)
0.1	0.9999	NA	0.0001	NA
0.2	0.998	NA	0.002	NA
0.3	0.992	NA	0.008	NA
0.4	0.974	NA	0.026	NA
0.5	0.937	NA	0.063	NA
0.6	0.87	NA	0.13	NA
0.7	0.76	NA	0.24	NA
0.8	0.59	NA	0.41	NA
> 0.9	NA	> 0.65	NA	< 0.35
0.95	NA	0.81	NA	0.19
0.99	NA	0.96	NA	0.04
0.999	NA	0.996	NA	0.004

NA Not applicable

Table C.1 shows the extent to which this test protects against calling an available state unavailable. Also with more than 65% probability, the test will correctly identify the unavailable state.

The SPRT methodology of clause C.4 below should be used as an alternative, non-minimal test for those situations where greater flexibility in managing the Type I and Type II errors is required.

## C.2 Procedures for estimating service availability

A sufficient estimate of the service availability percentage can be computed as follows. Based on an *a priori* estimate of the service availability, choose a sample size “s”, not less than 300. Choose “s” testing times during scheduled service time and distribute them across a long measurement period (e.g. 6 months). Because of the expected duration of service outages, choose no two testing times closer together than 7 hours (this serves to keep the observations uncorrelated). The testing times should be uniformly distributed across the scheduled service time. At each predetermined testing time, perform the availability test described above. If the test fails, the section is declared unavailable for that sample. Otherwise, the section is declared available. The estimate of the service availability percentage is the number of times the section was declared available multiplied by 100 and divided by the total number of samples.

## C.3 Procedures for estimating mean time between service outages

A sufficient estimate of the mean time between service outage parameter can be computed by conducting consecutive availability performance samples and by counting the observed changes from the available state to the unavailable state.

Prior to performing any tests, choose  $k$  disjoint intervals of time each not less than 30 minutes nor more than 3 hours. The total amount of time in the  $k$  intervals should exceed 3 times the *a priori* estimate of mean time between service outages. For the duration of each pre-defined interval conduct consecutive availability performance samples. The amount of time observed in the available state will be added to a cumulative counter called “A”. The number of observed transitions from the available state to the unavailable state will be accumulated in a counter called “F”.<sup>31)</sup>

For each pre-defined interval:

- 1) If all of the consecutive availability samples succeed, then add the total length of the interval to A. Do not change the cumulative value of F.
- 2) If the first availability sample succeeds and any subsequent sample in the interval fails, increase F by one. Add to A the total length of all availability samples prior to the first failure. Following the first failed availability sample, the remaining time in the interval may be discarded without testing its availability.
- 3) If the first availability sample fails, assume that the state transition occurred before the interval began. Add nothing to the count of observed availability time, A. Add nothing to the cumulative count of observed state changes, F. The remaining time in the interval may be discarded without testing its availability.

After the results of every pre-defined interval have been accumulated, the ratio,  $A/F$ , is an estimate of the mean time between service outages. A statistically more precise estimate can be obtained by increasing the number of observed intervals,  $k$ .

The estimate of mean time between service outages assumes that, if an outage begins during an availability performance sample, either this sample or the following sample will decide that the section is

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<sup>31)</sup> Each counter is initially set to zero.

unavailable. This is a reasonable assumption since service outages, in contrast to transient failures, will last more than 5 minutes.

Discarding the remainder of the interval following a failed availability sample is both practical and statistically justifiable. The virtual connection section must return to the available state before any more available time can be accumulated and before any more transitions to the unavailable state can be observed. First, the expected time to restore service may be large with respect to the remaining time in the interval. It can be inappropriate and counterproductive to continue testing a failed or congested network section. Second, if transitions to the unavailable state are statistically independent, then discarding the remainder of the interval, which may include time in the available state and a proportional number of transitions back into the unavailable state, will not bias the result.<sup>32</sup> The only consequence of discontinuing the test is the loss of testing time. To minimize that loss, the test intervals should be short with respect to the sum of the expected time to restore service and the expected time between service outages. Thus each test should be no longer than 3 hours.

There are two sources of bias in the estimation procedure described above. First, if an outage begins during the last availability sample of the interval, that transition may or may not cause the sample to fail. If it does not fail, the state transition is missed and the mean time between service outages is overestimated. Second, a state transition to the unavailable state during the first availability sample of the interval may or may not cause that sample to fail. According to the estimation procedure, if the sample does fail, the interval will be discarded, the state transition missed, and the mean time between service outages overestimated. These edge effects can be minimized by increasing the length of each interval, consequently increasing the number of availability samples, and thus decreasing the effect of the first and last sample outcomes as a proportion of the total sampled outcomes. A minimum recommended interval length is 30 minutes and size 5 minute availability samples.

Alternatively, both biases can be corrected by replacing the first instruction above with:

“If all of the consecutive availability samples succeed, then add the total length of the interval to A. Take one additional availability sample immediately following the interval. If that sample fails, increase F by one. If that sample succeeds, do not change F. The length of the additional sample has no effect on A.”

This modification identifies any state transitions that occurred during the last sample of the interval and eliminates the first source of bias. It also counts certain transitions that occurred outside of the interval. These transitions are counted with the same probability as the probability that the second source of bias inappropriately discards transitions. Thus, this modified procedure corrects both sources of bias. Using this modification, the mean time between service outages can be more accurately estimated.

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<sup>32)</sup> If outages tend to be clustered, discontinuing a test following a transition to the unavailable state will tend to overestimate the mean time between service outages. If outages tend to be negatively clustered, discontinuing a test following a transition to the unavailable state will tend to underestimate the mean time between service outages.

## C.4 SPRT methodology

### C.4.1 SPRT test procedure

For Phase I, perform a SPRT<sup>33)</sup> of the following pair of hypotheses, utilizing an appropriate value of  $z$  ( $z < 0.9$ ). This test will use successive call set-up attempts across the section under test, A. If the SPRT decides that  $H_0$  is true, then proceed to Phase II of the Minimal Test, otherwise if the SPRT decides that  $H_a$  is true then terminate the test and conclude that the service is unavailable due to the sum of call failure probability and call error probability exceeding the service outage threshold of 0.9.

$H_0$ :  $cep + cfp < z$  (service outage criterion is not met)

$H_a$ :  $cep + cfp > 0.9$  (service outage criterion is met)

### C.4.2 SPRT methodology

The hypotheses used are based on the provisional criteria specified in this Recommendation according to which the sum of the call failure probability and call error probability exceeding 0.9 (i.e.  $cfp + cep > 0.9$ ) determines a service outage. Implicit in this criteria is the assertion that one can in fact distinguish between  $cfp + cep > 0.9$  and  $cfp + cep < 0.9$ . However, the best that one can really do is to distinguish between  $cfp + cep > 0.9$  and  $cfp + cep > z$  ( $0 < z < 0.9$ ).

The sequential probability ratio test (SPRT) methodology controls both the Type I and Type II errors simultaneously, and, for deciding between two alternatives, is the most powerful statistical tool available<sup>34)</sup>. For simplicity this annex uses the same probability of a wrong decision for both Type I and Type II errors<sup>35)</sup>. The assumption of a binomial distribution for the success or failure of an individual call set-up attempt is made in this subclause.

The material below discusses the hypotheses to be tested, the decision rule, the upper and lower decision points, the expected number of call set-up attempts, and the least number of successful or failed attempts to end the SPRT.

### C.4.3 Hypotheses

The SPRT uses the following pair of hypotheses, where  $H_0$  corresponds to the outage threshold not being exceeded, and  $H_a$  corresponds to the outage threshold being exceeded.

$H_0$ :  $cep + cfp < z$  (service outage criterion is not met)

$H_a$ :  $cep + cfp > 0.9$  (service outage criterion is met)

### C.4.4 Decision rule and upper and lower decision points

The SPRT reaches a decision based on observed performance being greater or less than a particular value. These values depend on the number of observations taken ( $n$ ) and are denoted by  $UD(n)$  and  $LD(n)$  respectively. Formulas for  $LD(n)$  and  $UD(n)$  are given below after the decision rule.

<sup>33)</sup> The SPRT methodology is given in C.4.2.

<sup>34)</sup> See George G. Roussas, *A First Course in Mathematical Statistics*, by Addison-Wesley.

<sup>35)</sup> In the formulas below, error = Pr (Type I error) = Pr (Type II error). Values of error from 0.01 to 0.10 are commonly used.

**C.4.4.1 Decision rule**

If, upon making  $n$  attempts, the number of failed attempts is greater than  $UD(n)$  then the criterion for service outage is met.

$$\left\{ \sum_{j=1}^n x_j \geq UD(n) \right\}$$

If, upon making  $n$  attempts, the number of failed attempts is less than  $LD(n)$  then the criterion for service outage is not met.

$$\left\{ \sum_{j=1}^n x_j \leq LD(n) \right\}$$

Keep attempting calls until a decision is reached.

Formulas for  $UD(n)$  and  $LD(n)$

$$UD(n) = \frac{\left( \log \left( \frac{(1 - \text{error})}{(\text{error})} \right) - n * \log \left( \frac{(1 - 0.9)}{(1 - z)} \right) \right)}{\log \left( \frac{0.9 * (1 - z)}{z * (1 - 0.9)} \right)}$$

$$LD(n) = \frac{\left( \log \left( \frac{(\text{error})}{(1 - \text{error})} \right) - n * \log \left( \frac{(1 - 0.9)}{(1 - z)} \right) \right)}{\log \left( \frac{0.9 * (1 - z)}{z * (1 - 0.9)} \right)}$$

**C.4.5 Expected number of call set-up attempts**

The expected number of call set-up attempts until the SPRT reaches a decision is useful in determining the length and cost of the test. Under  $H_0$  and  $H_a$  respectively, the expected number of call set-up attempts are  $E_0(N)$  and  $E_a(N)$ . Asymptotic approximations for them are as follows, and are based on the use of a binomial probability for the sum of call set-up error and call set-up failure. Calculations resulting in entries in table C.3 greater than 100 were made up using these approximations. The rest of table C.3 was constructed using iterative matrix techniques yielding more precise values.

$$E_0(N) \approx \frac{\left( (1 - 2 * \text{error}) * \log\left(\frac{\text{error}}{(1 - \text{error})}\right) \right)}{z * \log\left(\frac{0.9 * (1 - z)}{z * (1 - 0.9)}\right) + \log\left(\frac{(1 - 0.9)}{(1 - z)}\right)}$$

$$E_a(N) \approx \frac{\left( (1 - 2 * \text{error}) * \log\left(\frac{(1 - \text{error})}{\text{error}}\right) \right)}{0.9 * \log\left(\frac{0.9 * (1 - z)}{z * (1 - 0.9)}\right) + \log\left(\frac{(1 - 0.9)}{(1 - z)}\right)}$$

**C.4.6 Least number of failures or successes to end a SPRT**

The quantities L and U represent the least number of call set-up attempts required by the SPRT to decide if H<sub>0</sub> or H<sub>a</sub> respectively are true. If all L call set-up attempts are successful, then the outage criterion is not met, while if all U of the call set-up attempts fail, then the outage criterion is met. The SPRT test will often continue after the values of U or L are reached, but they are the least values at which a decision in favor of H<sub>a</sub> or H<sub>0</sub> respectively could be taken. Tabulated values of U, L, E<sub>0</sub>(N) are provided in tables C.2 and C.3.

**Table C.2 - U/L – Minimum number of call set-up attempts**

Percent error			
z	10%	5%	1%
0.85	39/6	52/8	81/12
0.80	19/4	25/5	40/7
0.75	13/3	17/4	26/6
0.70	9/2	12/3	19/5
0.65	7/2	10/3	15/4
0.60	6/2	8/3	12/4
0.55	5/2	6/2	10/4
0.50	4/2	6/2	8/3
0.45	4/2	5/2	7/3
0.40	3/2	4/2	6/3
0.35	3/2	4/2	5/3
0.30	2/2	3/2	5/3
0.25	2/2	3/2	4/3
0.20	2/2	2/2	4/3
0.15	2/2	2/2	3/3
0.10	1/1	2/2	3/3

**NOTES**

- 1 - U is the least number of successive call set-up errors or failures needed to terminate the SPRT (deciding in favor of H<sub>a</sub>).
- 2 - L is the least number of successive call set-up successes needed to terminate the SPRT (deciding in favor of H<sub>0</sub>).
- 3 - z is the cutoff value in the null hypothesis (H<sub>0</sub>).
- 4 - The column headings represent the specified error rates. Due to approximations used in the SPRT, the error rates are bounded above by error/(1 – error). The differences are small over the range of error rates being considered.

**Table C.3 -  $E_a(N) / E_0(N)$  — Expected number of call set-up attempts**

Percent error			
$z$	10%	5%	1%
0.85	161.3/143.7	243.2/216.6	413.3/368.1
0.80	51.5/45.3	74.5/65.1	122.7/101.4
0.75	27.4/22.3	39.3/32.5	63.9/52.2
0.70	17.1/14.4	24.5/20.1	40.1/32.3
0.65	12.1/10.2	17.3/13.9	27.9/22.2
0.60	9.2/7.4	13.3/10.8	21.0/16.3
0.55	7.4/6.1	10.0/7.7	16.5/13.0
0.50	5.8/4.9	8.6/6.5	13.0/10.1
0.45	5.4/4.3	7.0/5.4	10.9/8.4
0.40	4.0/3.7	5.6/4.8	8.8/7.2
0.35	3.9/3.4	5.5/4.3	7.0/5.7
0.30	2.6/2.8	4.1/3.7	6.5/5.2
0.25	2.6/2.6	3.7/3.3	5.4/4.6
0.20	2.4/2.5	2.7/2.8	5.0/4.1
0.15	2.4/2.3	2.5/2.7	3.7/3.7
0.10	1.0/1.0	2.4/2.4	3.7/3.7

**NOTES**

1 -  $E_a(N)$  is the expected number of trials needed to terminate the SPRT when the “network” is unavailable.

2 -  $E_0(N)$  is the expected number of trials needed to terminate the SPRT when the “network” is available.

3 -  $z$  is the cutoff value in the null hypothesis ( $H_0$ ).

4 - The column headings represent the specified error rates. Due to approximations used in the SPRT, the error rates are bounded above by  $\text{error}/(1 - \text{error})$ . The differences are small over the range of error rates being considered.

**Annex D**  
(informative)

**Bibliography**

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<sup>36)</sup> Available from Superintendent of Documents, U.S. Government Printing Office, Washington, DC 20402.

**Annex E**  
(informative)

**Acronyms**

CCFP	Call Clearing Failure Probability
CID	Clear Indication Delay
CSD	Call Set-up Delay
CSEP	Call Set-up Error Probability
CSFP	Call Set-up Failure Probability
DCE	Data Circuit-Terminating Equipment
DLB	Data Loopback
DPTD	Data Packet Transfer Delay
DSE	Data Switching Exchange
DTE	Data Terminal Equipment
INI	Internetwork Interface
LCN	Logical Channel Number
MTBSO	Mean Time Between Service Outages
MTTSR	Mean Time To Service Restoral
NI	Network Interface
PE	Packet Layer Reference Event
PVC	Permanent Virtual Circuit
RER	Residual Error Ratio
RLB	Routing Loopback
SA	Service Availability
SABM	Set Asynchronous Balanced Mode
SABME	Set Asynchronous Balanced Mode Extended
SPRT	Sequential Probability Ratio Test
STE	Signaling Terminal Equipment
VC	Virtual Circuit
cep	call error probability
cfp	call failure probability
lwl	lower edge of the window on the transmit side
lwt	lower edge of the window on the receive side
npr	next data packet to be received
pdp	premature disconnect probability
pdsp	premature disconnect stimulus probability
rer	residual error ratio
ric	received interrupt count
rp	reset probability
rsp	reset stimulus probability
tc	throughput capacity