INTERNATIONAL TELECOMMUNICATION UNION

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THE INTERNATIONAL
TELEGRAPH AND TELEPHONE
CONSULTATIVE COMMITTEE

BLUE BOOK

VOLUME VIII – FASCICLE VIII.3

DATA COMMUNICATION NETWORKS

TRANSMISSION, SIGNALLING AND SWITCHING, NETWORK ASPECTS, MAINTENANCE AND ADMINISTRATIVE ARRANGEMENTS

RECOMMENDATIONS X.40-X.181

IXTH PLENARY ASSEMBLY
MELBOURNE, 14-25 NOVEMBER 1988

Geneva 1989

COVERING NOTE

GENERAL SECRETARIAT INTERNATIONAL TELECOMMUNICATION UNION

CORRIGENDUM No. 1

Geneva, June 1990

FASCICLE VIII.3

IXth PLENARY ASSEMBLY OF THE CCITT
MELBOURNE, 1988

Please insert the following corrections to the texts of Recommendations X.134, X.135, X.136, X.137, X.140, X.141 in Fascicle VIII.3 of Volume VIII of the CCITT Blue Book.

1. *Recommendation X.134*

Section: Considerata, last line

Currently: et
Should be: and

Figure: 1/X.134

Currently: Funtion
Should be: Function

Section: 1.7.2, 3rd line

Currently: Each of
Should be: Each end of

Figure: 4/X.134

Currently: paquets
Should be: packets

Figure: 5/X.134, note

Currently: paquets
Should be: packets

Table: 1/X.134, line 2

Currently: Cal
Should be: Call

2. *Recommendation X.135*

Section: 4.4, 3rd paragraph, 1st and 4th lines

Currently: worse-case
Should be: worst-case

Section: Supplement No. 1, 2nd paragraph, 2nd line

Currently: presentative
Should be: representative

Section: Supplement No. 1, 2nd paragraph, 2nd line

Currently: of
Should be: or of

3. *Recommendation X.136*

Section: 3.3.1, 3rd line

Currently: of of
Should be: of

Section: 3.3.2, 1st paragraph, 2nd line

Currently: packet
Should be: packets

Section: 3.3.2, 3rd paragraph, 4th line

Currently: substracting
Should be: subtracting

Section: 4.1, 1st paragraph, 2nd line

Currently: clean
Should be: clear

Section: 4.1, 2nd paragraph, 2nd line

Currently: atempts
Should be: attempts

4. *Recommendation X.137*

Title

Currently: Recommandation
Should be: Recommendation

Section: 1.2, first line

Currently: devides
Should be: divides

Section: 1.3, last line

Currently: or
Should be: of

Section: 3.2, 1st paragraph, 3rd line

Currently: schedules
Should be: scheduled

Section: 3.3, 1st line

Currently: conribution
Should be: contribution

Annex A: title

Currently: ANNEXE
Should be: ANNEX

Annex B: Section B.1, 1st paragraph, last line

Currently: connectin
Should be: connection

Annex B: Section B.2, equation

Currently: heures
Should be: hours

5. *Recommendation X.140*

Annex B: 4th paragraph, 1st line

Currently: matix
Should be: matrix

6. *Recommendation X.141*

Section: 3.2, 2nd paragraph, 3rd line

Currently: in of situations
Should be: in situations

Section: 3.2, 5th paragraph, 1st line

Currently: adaption
Should be: adaptation

Section: 3.3.3, 2nd paragraph, 1st line

Currently: frame
Should be: frames

Section: 4.2, 3rd paragraph, 2nd line

Currently: insure
Should be: ensure

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Data communication networks

SECTION 4 – *Network aspects*

PRELIMINARY NOTES

1 The Questions entrusted to each Study Group for the Study Period 1989-1992 can be found in Contribution No. 1 to that Study Group.

2 In this Fascicle, the expression "Administration" is used for shortness to indicate both a telecommunication Administration and a recognized private operating agency.

3 The status of annexes and appendices attached to the Series X Recommendations should be interpreted as follows (except where specified):

- an *annex* to a Recommendation forms an integral part of the Recommendation;
- an *appendix* to a Recommendation does not form part of the Recommendation and only provides some complementary explanation or information specific to that Recommendation.

FASCICLE VIII.3

PART I

Recommendations X.40 to X.181

DATA COMMUNICATION NETWORKS

TRANSMISSION, SIGNALLING AND SWITCHING, NETWORK ASPECTS, MAINTENANCE, ADMINISTRATIVE ARRANGEMENTS

SECTION 3

TRANSMISSION, SIGNALLING AND SWITCHING

Recommendation X.40

STANDARDIZATION OF FREQUENCY-SHIFT MODULATED TRANSMISSION SYSTEMS FOR THE PROVISION OF TELEGRAPH AND DATA CHANNELS BY FREQUENCY DIVISION OF A GROUP

(Geneva, 1972)

The CCITT,

considering

(a) that some Administrations are planning the introduction of public data networks;

(b) that, to facilitate interworking between some networks, it is desirable to standardize the characteristics of transmission systems for the provision of channels for certain maximum modulation rates;

(c) that interest has been expressed in deriving channels by frequency division of a group;

(d) that Recommendation X.1 defines the user classes of service for public data networks;

(e) that Recommendation X.1 includes user classes 3, 4 and 5 which correspond to maximum user data signalling rates of 600 bit/s, 2400 bit/s and 9600 bit/s, the transmission channels for which can be economically provided by frequency division of a group;

Note – In the case of synchronously operated terminals a method of keeping synchronism between the subscribers is necessary. This implies the need for a method to provide bit sequence independency in accordance with Recommendation X.2, e.g. a scrambler. This is provided external to this system but forms part of the network.

(f) that, for the present, no interest has been shown in providing separate channels for 600 bauds;

(g) that standardization of channels for modulation rates less than 600 bauds, for example 200 bauds, is the subject of other Recommendations (e.g. R.38 A and R.38 B);

(h) that there could be economic advantages in providing 2400- and 9600-baud channels (and possibly, in due course, 600-baud) in the one system;

unanimously declares the following view:

1 A group will be used as a bearer circuit.

2 The nominal modulation rates should be standardized at 2400 bauds and 9600 bauds.

3 For the 2400-baud channels the nominal mean frequencies are: $(110 - 4n)$ kHz, where $n = 1, 2, \ldots, 12$ (Figure 1/X.40).

For the 9600-baud channels the nominal mean frequencies are 96 kHz for channel 1 and 72 kHz for channel 2 (Figure 1/X.40).

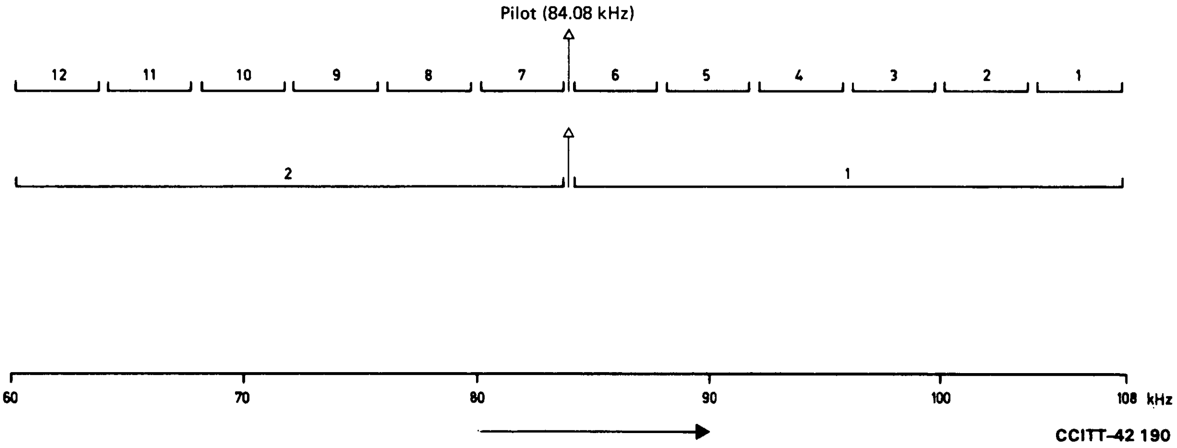

FIGURE 1/X.40

Division of the primary group into data channels for 2400 bauds and 9600 bauds

The mean frequency F_0 is defined as the half-sum of the characteristic frequencies corresponding to the start polarity (F_A) and the stop polarity (F_Z).

4 The mean frequencies at the sending end should not deviate by more than ± 20 Hz both for 2400-baud channels and 9600-baud channels.

5 The difference between the two characteristic frequencies in the same channel is fixed at:

2 kHz in the case of 2400-baud channels,

8 kHz in the case of 9600-baud channels.

6 The maximum tolerance of this difference is ± 10% both for 2400-baud channels and 9600-baud channels.

7 The total average power transmitted to the primary group is limited to −4 dBm0 (400 μW at a point of zero relative level). This sets, for the average power of a derived channel, the limit of

−15 dBm0 for the 2400-baud channels,

−7 dBm0 for the 9600-baud channels,

in a fully equipped system. Note Recommendation H.52, § a) 2, which says:

"In order to limit cross-modulation effects in wideband systems, the power level of any individual spectral component in the band 60-108 kHz should not exceed −10 dBm0 (except for the environment of the pilot for which a separate Recommendation exists)."

"With regard to its effect on non-telephone type signals, a discrete component is defined as a signal of sinusoidal form with a minimum duration of about 100 ms."

To meet this requirement at 9600 bauds, a data scrambler may be used external to the system.

8 The in-service levels of the permanent "start" polarity and permanent "stop" polarity signals must not differ by more than 1.5 dB and the higher of these two signal levels must comply with those of § 7 above.

9 The "start" polarity frequency is the lower of the two characteristic frequencies in the primary basic group and the "stop" polarity frequency is the higher one.

10 In the case of 9600 bauds where scramblers are used external to the system to comply with § 7 above, it will also be necessary to drop the continuous −7 dBm0 "start" polarity signal to −10 dBm0 in the absence of channel modulator control.

11 The receiving equipment should operate satisfactorily when the receiving level falls to 6 dB below the nominal level. The receiving equipment should have been restored to start polarity when the receiving level has fallen to 12 dB below the nominal level.

The alarm-control level is left to the choice of each Administration.

12 The maximum degree of isochronous distortion on standardized text is provisionally fixed at 8% in the whole receiver level range (± 6 dB from nominal level) for closed circuit measurements.

13 Systems should be designed in such a manner that the combined use of 6 channels for 2400 bauds and 1 channel for 9600 bauds is possible.

14 As an optional facility it should be possible to replace any 2400-baud channel, in particular channels No. 1 and No. 12, by a channel translating equipment which enables the insertion of a VFT system according to Recommendations R.35, R.35 *bis*, R.36, R.37, R.38 A or R.38 B.

Recommendation X.50

FUNDAMENTAL PARAMETERS OF A MULTIPLEXING SCHEME FOR THE INTERNATIONAL INTERFACE BETWEEN SYNCHRONOUS DATA NETWORKS

(Geneva, 1972; amended at Geneva, 1976 and 1980)

The establishment in various countries of public synchronous data networks creates a need to standardize a preferred multiplexing scheme to be used on international links between these countries.

The CCITT,

considering

that the resolution of the fundamental parameters of a multiplexing scheme is urgently needed for the interworking of data networks using different envelope structures;

unanimously declares the following view:

1 Division 1

1.1 This Recommendation sets out the fundamental parameters of a multiplexing scheme for interworking of networks that make use of the following structures:

a) 8-bit envelope (see Explanatory Note 1 below);

b) four 8-bit envelopes grouping (see Explanatory Note 2 below);

c) 10-bit envelope (see Explanatory Note 3 below), in the case where at least one of the networks is structured according to a) or b).

1.2 For interworking between two networks both of which utilize the 10-bit envelope structure as identified in § 1.1 c) above, Recommendation X.51 will apply.

1.3 Paragraph 2 of this Recommendation deals with the basic multiplexing parameters which shall be used in any application of this Recommendation.

1.4 Paragraph 3 of this Recommendation, in addition to § 2, applies to the interworking between two networks both of which utilize the 8-bit envelope structure, as identified in § 1.1 a) above.

1.5 Paragraph 4 of this Recommendation, in addition to § 2, applies to the interworking of networks as identified in § 1.1 above in cases other than those described in §§ 1.2 and 1.4 above with due regard to the transit situations.

1.6 The use of the status bit, besides that indicated in this Recommendation, should comply with Recommendations X.21 and X.21 *bis*, together with Recommendation X.71 for connections using decentralized signalling and with Recommendation X.60 for connections using common channel signalling.

2 Division 2

2.1 The multiplex gross bit rate of 64 kbit/s should be standardized for international links and framing information for the channels should be contained within the 64 kbit/s capability.

2.2 For the basic multiplexing of information bearer channels, the following applies:

i) structures suitable both for handling homogeneous (with respect to bearer rates) mixes of bearer channels and structures suitable for handling heterogeneous mixes of bearer channels are required;

ii) the signal elements of each individual channel should be assembled in 8-bit envelopes;

iii) an 8-bit envelope interleaved structure should be used;

iv) for the multiplex signal framing a distributed framing pattern should be used, employing the framing bits of consecutive 8-bit envelopes but taking into account the requirements for service digits (housekeeping digits);

v) these interleaved 8-bit envelopes will appear on the 64 kbit/s bearer as follows:

- 12.8 kbit/s channels will repeat every 5th 8-bit envelope;
- 6.4 kbit/s channels will repeat every 10th 8-bit envelope;
- 3.2 kbit/s channels will repeat every 20th 8-bit envelope;
- 800 bit/s channels will repeat every 80th 8-bit envelope.

2.3 The following multiplexing structure is recommended:

i) the multiplexing structure will comprise 80 8-bit envelopes;

ii) this structure will allow the multiplexing of channels at the bearer rates indicated in § 2.2 v) above;

iii) within each 12.8 kbit/s channel, only a homogeneous mixture of subrate channels will be allowed;

iv) a 72-bit long framing pattern is recommended. This pattern is part of the 80-bit pattern which is generated according to the primitive polynomial:

$$1 + x^4 + x^7$$

of the 2^7 Galois field with the forcing configuration

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and which is reproduced in Table 1/X.50, showing 8 bits ("A" to "H") reserved for housekeeping;

v) the first F bit, indicated as "A" in Table 1/X.50 is used to convey to the distant end alarm indications detected at the local end corresponding to:

- absence of incoming pulses;
- loss of frame alignment;

vi) the "A" bit shall be assigned such that:

"A" equals 1 means no alarm;

"A" equals 0 means alarm;

vii) the other F bits indicated as "B", "C", "D", "E", "F", "G", and "H" in Table 1/X.50 are reserved to convey further international housekeeping information. The exact use of the remaining housekeeping bits is under study. Pending the resolution of the housekeeping requirements, these bits are provisionally fixed to:

"B" equals 1, "C" equals 1, "D" equals 0,

"E" equals 0, "F" equals 1, "G" equals 1, "H" equals 0.

TABLE 1/X.50

1st bit

A	1	0	0	0	1	1	1	1	1	B	1	0	0	0	0	1	1	1	0
C	1	1	1	0	0	1	0	1	1	D	0	1	0	0	1	0	0	0	0
E	0	1	0	0	0	1	0	0	1	F	0	0	0	1	0	1	1	1	0
G	0	1	1	0	1	1	0	0	0	H	0	1	1	0	0	1	1	0	1

Forcing configuration

2.4 For frame synchronization, the following general requirements are recommended:

i) the frame synchronization method should be insensitive as far as possible to bit errors, error bursts and short bursts of the alarm indication signal (AIS) generated by transmission equipment;

ii) when a slip occurs in the transmission equipment, a fast frame alignment should be possible.

2.5 In addition to § 2.4 above, the frame synchronization method should offer the following performances:

i) the frame alignment recovery time, after a slip and in absence of bit errors should be less than 120 envelopes, with 95% probability;

ii) the time from the start of a disturbance as defined in § 2.4 i) to any action affecting the data channels shall be [including the transmission of alarm to the distant end defined in § 2.3 v) and vi)] greater than x (x in the range 1-20 ms);

iii) a random error rate of 1 in 10^4 shall not cause any frame alignment recovery action.

3 Division 3

3.1 For interworking between two networks, both of which utilize the 8-bit envelope structure, as identified in § 1.1 a) above, each individual channel should be assembled into single 8-bit envelopes. As an alternative to the multiplexing structure recommended in § 2.3 above, other structures may be used by bilateral agreement. One of the preferred structures is described below:

i) the multiplexing structure will comprise 20 8-bit envelopes;

ii) the structure will allow the multiplexing of channels at the bearer rates 12.8 kbit/s, 6.4 kbit/s and 3.2 kbit/s indicated in § 2.2 v) above;

iii) within each 12.8 kbit/s channel only a homogeneous mixture of subrate channels will be allowed;

iv) a 19-bit long framing pattern is recommended. The pattern is part of the 20-bit pattern which is generated to the primitive polynomial:

$$1 + x^2 + x^5$$

of the Galois 2^5 field with the forcing configuration

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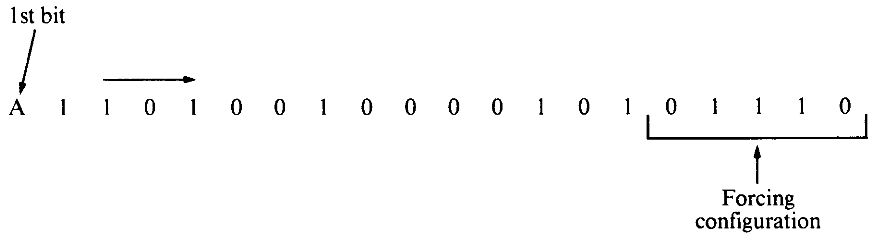
and is reproduced in Table 2/X.50;

v) the first F bit indicated as “A” in Table 2/X.50, is used as stated in § 2.3 v) above;

vi) the sense of “A” will be in accordance with § 2.3 vi) above.

TABLE 2/X.50



3.2 For frame synchronization, the general requirements and the performances should be as recommended in §§ 2.4 and 2.5 above.

4 Division 4

For the interworking of networks as identified in § 1.1 above, in cases other than those described in §§ 1.2 and 1.4, the following shall apply.

4.1 A network using the 10-bit envelope structure shall interwork with other networks, as identified in § 1.1 a) and b) above, by offering the same characteristics as a network using the four 8-bit envelopes grouping. Therefore in the following, the term "network providing the four 8-bit envelopes grouping" will cover the case of a network using either four 8-bit envelopes grouping as identified in § 1.1 b) or the 10-bit envelope structure, as identified in § 1.1 c).

4.2 When either end of an international connection, carrying point-to-point or switched service, terminates in a network providing four 8-bit envelopes grouping, the use of the four 8-bit envelopes grouping may be required on the international connection carrying point-to-point or switched services. This is subject to further study.

4.3 The alignment of the four 8-bit envelopes grouping shall be subject to the following conditions:

i) the method of alignment shall allow switched and non-switched point-to-point data circuits to be conveyed;

ii) alignment shall be monitored and maintained at both ends of an international connection in the two networks providing the four 8-bit envelopes grouping by means of a pattern(s) on bit S_D of the four 8-bit envelopes grouping throughout the data phase of the call;

iii) transit exchanges shall not overwrite the S_D bit once they have through connected;

iv) alignment shall be established at both ends of an international connection in the two networks providing the four 8-bit envelopes grouping prior to the through connection of the originating exchange.

Note 1 – The interexchange signalling procedures are expected to allow iv) above without unacceptable delay in call set-up.

Note 2 – Problems caused by imitation of the S_D pattern(s) should be studied further.

4.4 The specific strategy of the four 8-bit envelopes grouping alignment would be the subject of further study.

4.5 In the case where one of the networks is not providing the four 8-bit envelopes grouping, there is no relationship between any four 8-bit envelopes grouping and any character structure outgoing from that network. Moreover this network will not originate any alignment information for the outgoing four 8-bit envelopes grouping.

4.6 When links containing a four 8-bit envelopes grouping are connected in tandem to provide a trunk, the four 8-bit envelopes grouping alignment information shall be maintained across the connection.

4.7 In transit switching, the four 8-bit envelopes grouping alignment information shall be maintained once the transit exchange has through connected.

4.8 When links containing single 8-bit envelopes are connected in tandem to provide a trunk, the information and status bits of the 8-bit envelopes shall be transparently conveyed across the connection.

5 Division 5

To regulate transmission of 64-kbit/s streams, the following will apply:

i) In networks where the 64-kbit/s transmission paths have an octet structure throughout (i.e. 8-bit time slots are generally available), the 8-bit envelopes of the data multiplex signal should be aligned with the octet structure. This relationship will be established across 64-kbit/s interfaces by means of the 8-kHz timing signal.

ii) In networks where an octet structure is not utilized throughout (i.e. 8-bit time slots are not generally available on 64-kbit/s transmission paths), the 8-bit envelopes need not necessarily be aligned with the octet structure. On 64-kbit/s interfaces the 8-kHz timing signal may not be used; in that case the framing information is conveyed across this interface fully within the multiplex.

iii) Whether on an international link the 8-bit envelopes have to be aligned with the octet structure is provisionally subject to bilateral agreement but further study is required.

EXPLANATORY NOTES

Note 1 – 8-bit envelope

In an 8-bit envelope, bit 1 is reserved for framing purposes, bits 2-7 are information bits of the channel, and bit 8 is a status bit (see Figure 1/X.50).

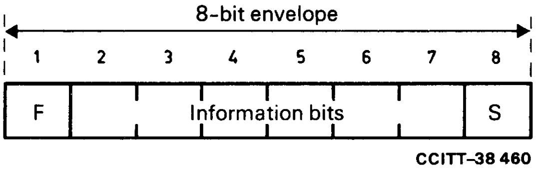


FIGURE 1/X.50

The addition of the framing and the status bits results in a 33% increase in bit rate, so that bearer channel rates are:

12.8 kbit/s for the 9.6 kbit/s data signalling rate;

6.4 kbit/s for the 4.8 kbit/s data signalling rate;

3.2 kbit/s for the 2.4 kbit/s data signalling rate;

800 bit/s for the 600 bit/s data signalling rate.

The status bit is associated with each envelope and, in conjunction with the information bits, conveys call control information.

Note 2 – Four 8-bit envelopes grouping

A group of four 8-bit envelopes is assembled on a single channel as a 32-bit group providing 24 information bits. This gives the possibility of accommodating three 8-bit characters, e.g. P, Q, R, as in Table 3/X.50.

TABLE 3/X.50

F	P1	P2	P3	P4	P5	P6	S_A	8-bit envelope A
F	P7	P8	Q1	Q2	Q3	Q4	S_B	8-bit envelope B
F	Q5	Q6	Q7	Q8	R1	R2	S_C	8-bit envelope C
F	R3	R4	R5	R6	R7	R8	S_D	8-bit envelope D

Status bit S_D is used to provide the alignment information of the four 8-bit envelopes grouping.

Status bits S_A, S_B and S_C in conjunction with the 24 information bits convey call control information.

When the three 8-bit characters P, Q and R are accommodated as above described, status bits S_A, S_B and S_C are respectively associated with those characters.

The four 8-bit envelopes grouping is applied on a per channel basis. For example, for the 12.8-kbit/s bearer rate, the four 8-bit envelopes group recurs after twenty 8-bit envelopes of the multiplexed stream, as in Figure 2/X.50.

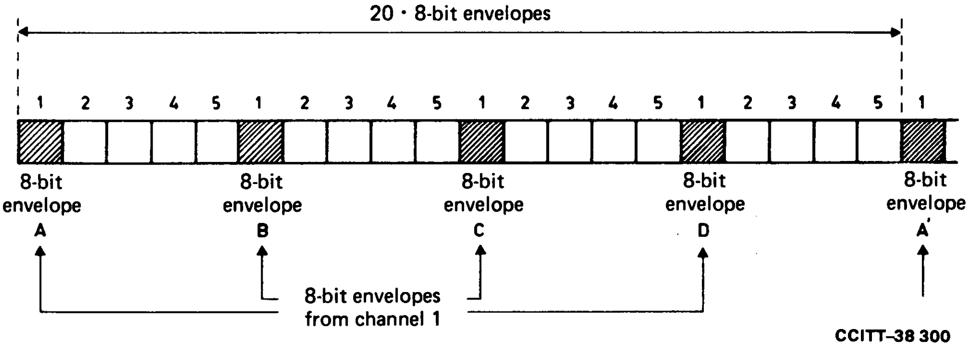


FIGURE 2/X.50

Note 3 – 10-bit envelope

In a 10-bit envelope, bit 1 is a status bit, bit 2 is reserved for envelope alignment purposes and bits 3-10 are information bits of the channel (see Figure 3/X.50).

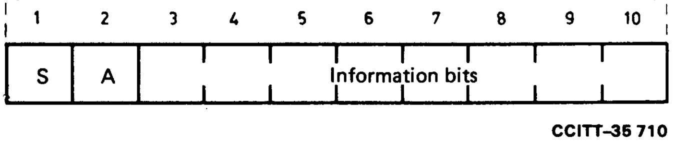


FIGURE 3/X.50

The addition of the envelope alignment and the status bits results in a 25% increase in bit rate, so that bearer channel rates are:

12.0 kbit/s for the 9.6 kbit/s data signalling rate;
6.0 kbit/s for the 4.8 kbit/s data signalling rate;
3.0 kbit/s for the 2.4 kbit/s data signalling rate;
750 bit/s for the 600 bit/s data signalling rate.

The status bit is associated with each envelope and, in conjunction with the associated 8-bit byte information bits, conveys call control information.

Recommendation X.50 *bis*

FUNDAMENTAL PARAMETERS OF A 48-kbit/s USER DATA SIGNALLING RATE TRANSMISSION SCHEME FOR THE INTERNATIONAL INTERFACE BETWEEN SYNCHRONOUS DATA NETWORKS

(Geneva, 1980)

1 General

1.1 This Recommendation sets out the fundamental parameters of a transmission scheme that should be used for 48-kbit/s data signalling rate for interworking of networks that make use of the following structures:

a) 8-bit envelope (see Explanatory Notes 1 and 2 of Recommendation X.50);
b) 10-bit envelope (see Explanatory Note 3 of Recommendation X.50); in the case where at least one of the networks is structured according to a).

1.2 For interworking between two networks both of which utilize the 10-bit envelope structure as identified in § 1.1 b) above, Recommendation X.51 *bis* will apply.

1.3 Paragraph 2 of this Recommendation deals with the basic parameters which shall be used in any application of this Recommendation and, in particular, for interworking between two networks, both of which utilize the 8-bit envelope structure.

1.4 Paragraph 3 of this Recommendation, in addition to § 2, applies to the interworking of networks with different envelope structures.

1.5 The use of the status bit, in addition to the indication given in this Recommendation, should comply with Recommendation X.21 and X.21 *bis*, together with Recommendation X.71 for connections using decentralized signalling and with Recommendation X.60 for connections using common channel signalling.

2 Transmission scheme

2.1 The gross bit rate of 64 kbit/s should be standardized for international links.

2.2 The signal element of the 64-kbit/s channel should be assembled in 8-bit envelopes in which bit 1 is the F bit, bits 2-7 are information bits and bit 8 is the status bit S.

2.3 The use and the value to be assigned to the F bits of the 8-bit envelopes are under study.

3 Interworking of networks with different envelope structures

The problem of interworking of networks with different envelope structures should be further studied taking into account the recommendations in § 4 of Recommendation X.50.

Recommendation X.51

FUNDAMENTAL PARAMETERS OF A MULTIPLEXING SCHEME FOR THE INTERNATIONAL INTERFACE BETWEEN SYNCHRONOUS DATA NETWORKS USING 10-bit ENVELOPE STRUCTURE

(Geneva, 1976; amended at Malaga-Torremolinos, 1984)

The CCITT,

considering

(a) that Recommendation X.50 sets out the fundamental parameters for a multiplexing scheme for the interworking of networks where at least one makes use of the 8-bit envelope structure or of the four 8-bit envelopes grouping,

(b) that there is a requirement for a multiplexing scheme for the interworking between two networks where both use 10-bit envelope structure,

unanimously declares the view

that the following fundamental parameters shall be used between networks using the 10-bit envelope structure.

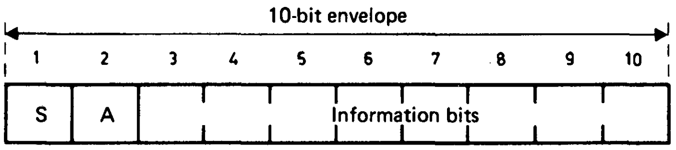
1 Gross bit rate

For transmission on the international link the multiplexed bit stream shall have a gross bit rate of 64 kbit/s. The fundamental multiplex structure shall have a gross bit rate of 60 kbit/s and shall utilize padding techniques for transmission on the 64 kbit/s international bearer.

2 Fundamental multiplex

For the fundamental multiplexing of information bearer channels, the following applies:

2.1 The signal elements of each individual channel shall be assembled in 10-bit envelopes, in which bit 1 is a status bit (see Note), bit 2 is an envelope alignment bit, and bits 3-10 are information bits, as in Figure 1/X.51.



CCITT-35 710

FIGURE 1/X.51

The addition of the status and the envelope alignment bits results in a 25% increase in bit rate, so that the bearer channel rates are:

12.0 kbit/s for the 9.6-kbit/s data signalling rate;
6.0 kbit/s for the 4.8-kbit/s data signalling rate;
3.0 kbit/s for the 2.4-kbit/s data signalling rate;
750 bit/s for the 600-bit/s data signalling rate.

Note – A status bit S bit is associated with each envelope and in conjunction with the associated 8-bit data byte conveys call control information (cf. Recommendations X.21, X.21 *bis*, X.60, X.71 and X.50).

2.2 A 10-bit envelope interleaved structure shall be used.

2.3 These interleaved envelopes will appear on the 60 kbit/s fundamental multiplex as follows:

- 12.0-kbit/s channels will repeat every 5th envelope;
- 6.0-kbit/s channels will repeat every 10th envelope;
- 3.0-kbit/s channels will repeat every 20th envelope;
- 750-bit/s channels will repeat every 80th envelope.

2.4 Both structures suitable for handling homogeneous (with respect to bearer rates) mixes of bearer channels and structures suitable for handling heterogeneous mixes of bearer channels are required, with the constraint that the division of any 12-kbit/s bearer channels of the multiplex shall be homogenous providing either two 6-kbit/s, four 3-kbit/s or sixteen 750-bit/s bearer channels.

3 Method of framing

3.1 *Overall structure*

The residual 4-kbit/s capacity obtained by carrying the fundamental 60-kbit/s multiplex on the 64-kbit/s bearer shall be distributed so that a padding bit is inserted after each group of 15 bits from the fundamental multiplex (see also Figure 2/X.51).

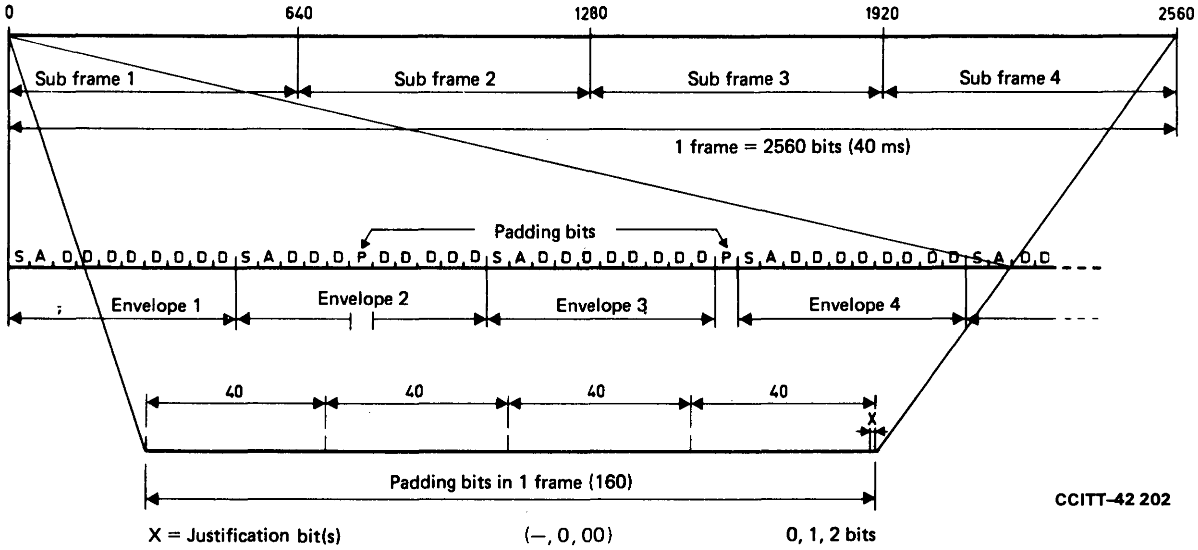


FIGURE 2/X.51

Multiplex frame structure

The frame length shall be 2560 bits in the case of a synchronized bearer, i.e. 2400 bits or 240 envelopes from the fundamental multiplex interleaved with 160 padding bits.

When justification is used (for national purposes) in the case of a non-synchronized bearer the last padding bit in the frame can be deleted or an extra padding bit added when needed, resulting in a variable frame length of 2560 ± 1 bit. (This can allow a maximum speed tolerance of approximately ± 4 parts in 10^4.)

The padding bits shall contain the framing pattern, justification service digits and housekeeping signalling (alarms, etc.).

3.2 *Framing*

3.2.1 *Frame alignment patterns*

The frame alignment method is based on the use of 4 equidistantly distributed frame alignment patterns written into the padding bits, dividing the frame into 4 subframes. Each subframe alignment pattern starts with the 14-bit pattern:

11111001101010

followed by a 2-bit subframe identifier unique to the subframe, i.e.:

SF1 = 00, SF2 = 01, SF3 = 10, SF4 = 11.

3.2.2 *Framing strategy*

3.2.2.1 *Loss of frame alignment*

The criterion for loss of frame alignment shall be three consecutive frame alignment patterns including subframe identifier in error.

The frame alignment shall also be considered lost if the first received frame alignment pattern including subframe identifier after reframing is in error.

3.2.2.2 *Reframing*

The criterion for reframing shall be the detection of one valid frame alignment pattern.

3.2.2.3 *Reframing procedure*

After loss of frame alignment:

- the outgoing envelopes shall be set to all ones,
- the state shall be signalled to the distant end, and
- a parallel hunt for a valid frame alignment pattern shall be started.

After a valid frame alignment pattern is found:

- the two following padding bits shall be accepted as subframe identifiers and be used to set the frame and subframe counter(s) as applicable,
- the blocking of the outgoing data channels shall be removed, and
- the signalling of out of frame alarm to the distant end shall be terminated.

4 **Justification**

The 64-kbit/s bearer carrying the 10-bit envelope multiplex normally shall be locked to the data stream and therefore justification on international links is not required. However, justification could be required for national purposes. To achieve this, plus minus justification shall be used in which four repeated justification service signals occupy the 3 bits immediately following each subframe identifier. The last padding bit of the frame is used as a justification digit.

The repeated justification service signals are:

010 no justification (i.e. one padding bit at end of frame),

100 one justification bit has been added (i.e. two padding bits at end of frame),

001 the justification bit has been deleted (i.e. no padding bit at end of frame).

In evaluating the signals in one frame a majority decision of the four received signals is used. In case of no majority, no justification shall be assumed.

If framing is lost, no justification shall be assumed before reframing has occurred.

5 Housekeeping signals and functions

The padding bits not used for framing and justification shall be available for housekeeping information signals, for both international and national use. The definition and allocation of some of the available housekeeping bits is left for further study. The following allocation is recommended.

5.1 *International housekeeping bits*

Eight bits A, B, C, D, E, F, G, and H (cf. Recommendation X.50) are allocated for international housekeeping signals.

The bit A is used to convey to the distant end alarm indications detected at the local end corresponding to:

- absence of incoming pulses,
- loss of frame alignment,

and the bit A shall be assigned such that:

- A equals 1 means no alarm,
- A equals 0 means alarm.

The other bits B, C, D, E, F, G and H are reserved to convey further international housekeeping signals. The exact use is under study. Pending the result of the study these bits shall be set to binary 1.

5.2 *Cyclic error-control*

A cyclic error-control (cf. Recommendation V.41) to be used end-to-end on the international 64-kbit/s link is recommended but not mandatory. The multiplex frame (2560 bits) is divided modulo 2 by the polynomial $x^{16} + x^{12} + x^5 + 1$ and the resulting reminder (16 bits), the check bits, are sent in the next frame, 4 bits in each subframe. An error is detected at the receiving end by comparing the check bits generated locally, by dividing the received multiplex frame with the same polynomial, and the check bits received in the following frame. The error detection shall be blocked in the out-of-frame state.

5.3 *National housekeeping signals*

A total of 48 housekeeping bits, 12 in each subframe, remains for national housekeeping signals, of which the following are foreseen:

Network status	1-4 bits
Multiplex channel allocation (depending on number of speed classes and coding)	5-10 bits
Internal and external alarms	1-4 bits

These signals could possibly be extended for international use. Housekeeping bits not used in one network shall be set to binary 1.

6 Allocation and use of padding bits (40 bits) in one subframe (640 bits) for framing, justification and housekeeping

The allocation of padding bits in one subframe numbered P1 to P40 is described below and shown in Figure 3/X.51.

P1-P4	International housekeeping bits A, B, C, and D (cf. Recommendation X.50)	
P5-P8	Error check bits	4 bits
P9-P20	National housekeeping bits	12 bits
P21-P34	Framing pattern Code 11111001101010	14 bits
P35-P36	Subframe identifier Code 00, 01, 10 or 11	2 bits

For P37-P40 two alternatives exist:

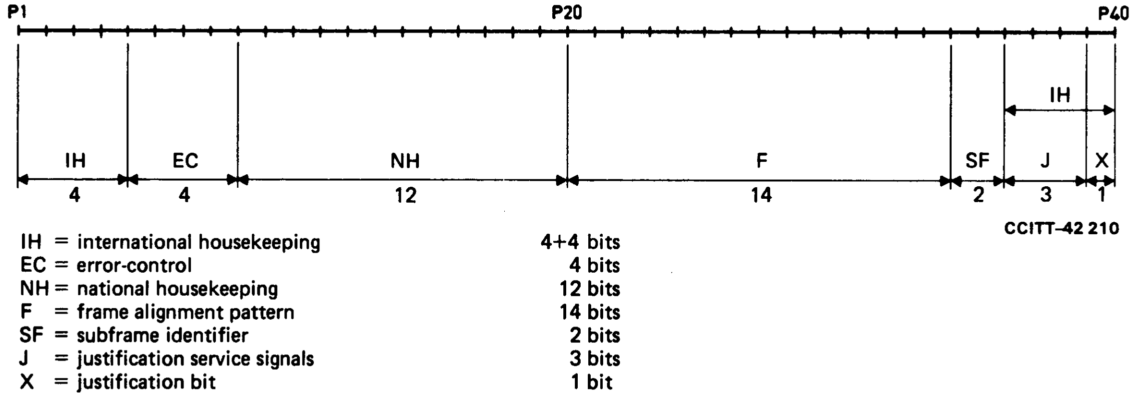
I – Synchronous transmission bearer

P37-P40	International housekeeping bits E, F, G and H (cf. Recommendation X.50)	

II – Asynchronous transmission bearer

P37-P39	Justification service signals . . . . Code 001, 010, 100	3 bits
P40(P41)	Justification bit(s) 0, 1, 2 bit(s) Code –, 0, 00	

Only the justification bit(s) in the last subframe (SF4) is used for justification.



IH = international housekeeping	4+4 bits
EC = error-control	4 bits
NH = national housekeeping	12 bits
F = frame alignment pattern	14 bits
SF = subframe identifier	2 bits
J = justification service signals	3 bits
X = justification bit	1 bit

FIGURE 3/X.51

Allocation of padding bits in one subframe (40 bits)

7 Transmission of the 48 kbit/s user data signalling rate

Generally, Recommendation X.51 *bis* applies.

Optionally, on bilateral agreement, the scheme described in this Recommendation may be applied also to transmit the 48 kbit/s user data signalling rate. Using this option, the bearer channel rate of the fundamental multiplex described in § 2 becomes 60 kbit/s, permitting only one single channel to be conveyed.

FUNDAMENTAL PARAMETERS OF A 48-kbit/s USER DATA SIGNALLING RATE TRANSMISSION SCHEME FOR THE INTERNATIONAL INTERFACE BETWEEN SYNCHRONOUS DATA NETWORKS USING 10-bit ENVELOPE STRUCTURE

(Geneva, 1980)

The CCITT,

considering

that there is a requirement for a 48-kbit/s user data signalling rate transmission scheme for the interworking between two networks where both use 10-bit envelope structure,

unanimously declares the view

that the following fundamental parameters shall be used in the transmission scheme to carry the 48-kbit/s user data signalling rate between networks using the 10-bit envelope structure.

1 Transmission scheme

1.1 The gross bit rate of 64 kbit/s should be standardized for international links.

1.2 The signal elements of the 48-kbit/s channel shall be assembled in 10-bit envelopes, in which bit 1 is a status bit, bit 2 is an envelope alignment bit, and bits 3-10 are user data information bits as in Figure 1/X.51 *bis.*

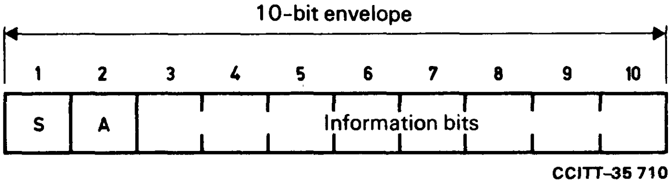
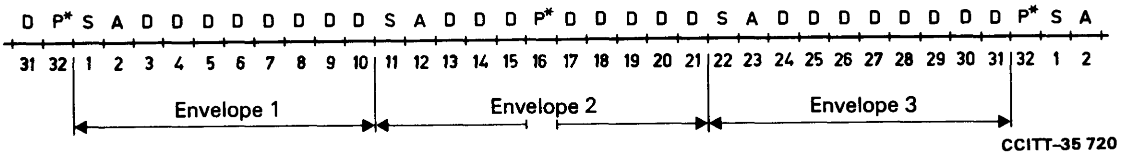


FIGURE 1/X.51 *bis*

1.3 The basic transmission scheme consists of consecutive 10-bit envelopes interleaved with padding bits occurring every 16th bit. Looking at a group of 32 consecutive bits of the 64-kbit/s bit stream containing 3 envelopes with 24 user data bits D, and numbering the bits starting with the S bit of envelope 1, the padding bits P shall be inserted in the bit positions 16 and 32 as in Figure 2/X.51 *bis.*



* Padding bits

FIGURE 2/X.51 *bis*

1.4 The padding bits shall carry a simple framing pattern that shall be used to identify the envelopes, within the 64-kbit/s stream.

A tentative proposal for such a simple framing pattern would be the following:

i) the padding bit in the position 16 of Figure 2/X.51 *bis* is set to binary 0;

ii) the padding bit in the position 32 of Figure 2/X.51 *bis* is set to binary 1.

Note – Other more complex framing patterns, which allow the use of padding bits for such functions as, for example, housekeeping signalling or justification in the national network, are for further study.

1.5 The framing strategy is for further study.

1.6 The use of the framing pattern to monitor the error rate in the transmission path, which will be optional, is for further study.

1.7 The envelope alignment bit shall carry a pattern of alternating binary 0 and binary 1 in consecutive envelopes, i.e. the pattern on the A bits in Figure 2/X.51 *bis* can be either 010 or 101.

Note – Other patterns on the A bits, e.g. "all zeros" or "all ones" could be used for alarm signals from the distant end and this is for further study.

1.8 The use of the status bit should comply with Recommendations X.21 and X.21 *bis*, together with Recommendation X.71 for connections using decentralized signalling, and with Recommendation X.60 for connections using common channel signalling.

Recommendation X.52

METHOD OF ENCODING ANISOCHRONOUS SIGNALS INTO A SYNCHRONOUS USER BEARER[1]

(Geneva, 1980)

The CCITT,

considering that

(a) Recommendation X.1 defines the user classes of service in public data networks;

(b) Recommendation X.2 defines the international user facilities in public data networks;

(c) Recommendations X.21 and X.21 *bis* define the interface between data terminal equipment (DTE) and data circuit-terminating equipment (DCE) for synchronous operation;

(d) Recommendations X.50 and X.51 define the multiplexing scheme for the international interface between synchronous data networks;

(e) Recommendations X.60, X.61 and X.71 define the signalling system on international circuits between synchronous data networks;

(f) some circuits implementing synchronous data networks also will connect to those networks DTEs operating in user classes of service 1 and 2;

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[1] This Recommendation is only valid for interworking between synchronous data networks. For the interworking between anisochronous data networks the Series R Recommendations will apply.

unanimously declares the following view:

1 Scope

1.1 In the case where two synchronous data networks offer service for DTEs in user classes of service 1 and 2, the transfer of the anisochronous signals between the networks shall be performed using a synchronous user channel of 600 bit/s in the standardized multiplexing schemes given in Recommendations X.50 and X.51 if one or both of the networks nationally use the synchronous user channel of 600 bit/s.

1.2 In the case where two synchronous data networks offer service for DTEs in user classes of service 1 and 2 but do not provide the 600-bit/s rate, the transfer of the anisochronous signals between those two networks shall be performed using a synchronous user channel of 2400 bit/s in the standardized multiplexing schemes given in Recommendations X.50 and X.51.

1.3 The method of encoding signals from DTEs in user classes of service 1 and 2 into the synchronous bearer shall be independent of the multiplexing scheme used.

1.4 The method of encoding shall be as defined in this Recommendation.

2 Encoding method

The encoding method implies that characters generated by DTEs in user classes of service 1 and 2 in accordance with Recommendation X.1 are transferred on international links as characters on a synchronous user channel, i.e. the transfer of characters on a synchronous user channel shall include the start signal as well as the stop signal with the following convention:

start polarity = binary zero;

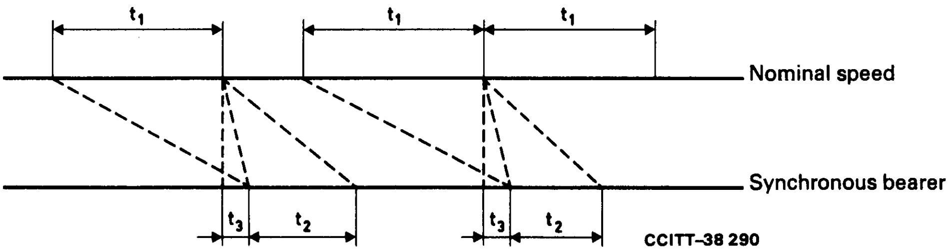
stop polarity = binary one.

Between any two characters on synchronous user channel the value of the bits shall be binary one.

The encoder and decoder shall be implemented in such a way that continuous start polarity (as well as continuous stop polarity) generated by a DTE can be transferred.

On the multiplexed link there need not be any relation between characters and envelopes.

The encoder shall be implemented in such a way that the time delay between the reception of a character at nominal speed and the start of sending the character on a synchronous user channel is less than 1 bit at the data signalling rate of the synchronous user channel used.



t_1 One character at nominal speed

t_2 One character at synchronous signalling rate

t_3 Time delay < 1.67 ms

FIGURE 1/X.52

ANNEX A

(to Recommendation X.52)

Location of the encoder

The location of the encoder, e.g. in the DCE in question or at a control point in the network, is a national matter. The location however will have no impact on the method described in this Recommendation.

When discussing the location of the encoder for harmonization reasons one should bear in mind that:

- in the case of a DCE located encoder, no special features for handling asynchronous signals are needed in network components such as concentrators and multiplexers and that all maintenance functions, subscriber line signalling scheme, local network modems, etc. implemented for the synchronous user classes of service can be used without any changes;
- if the encoder is placed at a central point, the data signalling rate on the local loop can be kept at the lowest possible rate allowing the use of a simple 2-wire modem and the sharing of conversion equipment at the central point by a number of subscribers.

ANNEX B

(to Recommendation X.52)

Higher data signalling rates

In the case where asynchronous DTEs operating at higher data signalling rates than given in Recommendation X.1 are connected to synchronous data networks, the same principle for encoding as given in the text of this Recommendation could be used and the relationship between data signalling rate and bearer channel rate shall be as shown in Table B-1/X.52.

TABLE B-1/X.52

Data signalling rate	Bearer channel rate
600 bit/s	2400 bit/s
1200 bit/s	2400 bit/s
2400 bit/s	4800 bit/s
4800 bit/s	9600 bit/s

Recommendation X.53

NUMBERING OF CHANNELS ON INTERNATIONAL MULTIPLEX LINKS AT 64 kbit/s

(Geneva, 1980; amended at Malaga-Torremolinos, 1984)

The CCITT,

considering that

Recommendations X.50 and X.51 define multiplexing schemes for international links at 64 kbit/s.

unanimously declares

the following view on the numbering of the tributary channels.

Tributary data channels conveyed within a 64-kbit/s multiplex link according to Recommendations X.50 and X.51, should be identified, for operational and maintenance purposes, by the following label:

i) One decimal digit D_1 indicating the multiplexing structure.

   $D_1 = 1$ for the 80 8-bit envelope structure (Division 2 of Recommendation X.50).

   $D_1 = 2$ for the 20 8-bit envelope structure (Division 3 of Recommendation X.50).

   Note – This applies to multiplexing structures defined in Recommendation X.50 only.

ii) One decimal digit D_2 indicating the channel rate.

   $D_2 = 3, 4, 5, 6$ for the rates of 600, 2400, 4800, 9600 and 48 000 bit/s respectively.

   Note – Digits 1 and 2 are reserved for user classes of service 1 and 2.

iii) Two decimal digits, D_3 and D_4, indicating the position "n" assigned in the frame with respect to the first envelope of the channel considered; $n \leqslant 80$ for the 80 envelopes frames defined in Recommendation X.50 (Division 2) and Recommendation X.51; $n \leqslant 20$ for the 20 envelopes frame defined in Recommendation X.50 (Division 3).

Recommendation X.54

ALLOCATION OF CHANNELS ON INTERNATIONAL MULTIPLEX LINKS AT 64 kbit/s

(Geneva, 1980; amended at Malaga-Torremolinos, 1984)

The CCITT,

considering that

Recommendations X.50 and X.51 define multiplexing schemes for international links at 64 kbit/s,

unanimously declares

the following view on the allocation of the tributary channels.

On international links carrying data channels multiplexed at 64 kbit/s according to Recommendations X.50 and X.51, the allocation of tributary channels at rates of 0.6, 2.4, 4.8 and 9.6 kbit/s within the multiplex frame, should be chosen, by bilateral agreement, among the configurations listed in Table 1/X.54.

Note 1 – If, on bilateral agreement, a single 48 kbit/s channel is transmitted, optionally permissible within an X.51 multiplex frame, this configuration is numbered 00.

Note 2 – The phase number i ($i = 1, \dots, 5$) corresponds to the set of envelopes $i + 5j$ ($j = 0, \dots, 15$ for 80 envelope frames; $j = 0, \dots, 3$ for 20 envelope frames) of each frame. Each phase contains either one 9.6-kbit/s or two 4.8-kbit/s or four 2.4-kbit/s or sixteen 0.6-kbit/s channels.

TABLE 1/X.54

Allocation of tributary channels in the 64-kbit/s multiplex frame

Configuration number	Phase number				
	1	2	3	4	5
00	48				
01	9.6	9.6	9.6	9.6	9.6
02	9.6	9.6	9.6	9.6	4.8
03	9.6	9.6	9.6	9.6	2.4
04	9.6	9.6	9.6	9.6	0.6
05	9.6	9.6	9.6	4.8	4.8
06	9.6	9.6	9.6	4.8	2.4
07	9.6	9.6	9.6	4.8	0.6
08	9.6	9.6	9.6	2.4	2.4
09	9.6	9.6	9.6	2.4	0.6
10	9.6	9.6	9.6	0.6	0.6
11	9.6	9.6	4.8	4.8	4.8
12	9.6	9.6	4.8	4.8	2.4
13	9.6	9.6	4.8	4.8	0.6
14	9.6	9.6	4.8	2.4	2.4
15	9.6	9.6	4.8	2.4	0.6
16	9.6	9.6	4.8	0.6	0.6
17	9.6	9.6	2.4	2.4	2.4
18	9.6	9.6	2.4	2.4	0.6
19	9.6	9.6	2.4	0.6	0.6
20	9.6	9.6	0.6	0.6	0.6
21	9.6	4.8	4.8	4.8	4.8
22	9.6	4.8	4.8	4.8	2.4
23	9.6	4.8	4.8	4.8	0.6
24	9.6	4.8	4.8	2.4	2.4
25	9.6	4.8	4.8	2.4	0.6
26	9.6	4.8	4.8	0.6	0.6
27	9.6	4.8	2.4	2.4	2.4
28	9.6	4.8	2.4	2.4	0.6
29	9.6	4.8	2.4	0.6	0.6
30	9.6	4.8	0.6	0.6	0.6
31	9.6	2.4	2.4	2.4	2.4
32	9.6	2.4	2.4	2.4	0.6
33	9.6	2.4	2.4	0.6	0.6
34	9.6	2.4	0.6	0.6	0.6
35	9.6	0.6	0.6	0.6	0.6
36	4.8	4.8	4.8	4.8	4.8
37	4.8	4.8	4.8	4.8	2.4
38	4.8	4.8	4.8	4.8	0.6
39	4.8	4.8	4.8	2.4	2.4
40	4.8	4.8	4.8	2.4	0.6
41	4.8	4.8	4.8	0.6	0.6
42	4.8	4.8	2.4	2.4	2.4
43	4.8	4.8	2.4	2.4	0.6
44	4.8	4.8	2.4	0.6	0.6
45	4.8	4.8	0.6	0.6	0.6
46	4.8	2.4	2.4	2.4	2.4
47	4.8	2.4	2.4	2.4	0.6
48	4.8	2.4	2.4	0.6	0.6
49	4.8	2.4	0.6	0.6	0.6
50	4.8	0.6	0.6	0.6	0.6
51	2.4	2.4	2.4	2.4	2.4
52	2.4	2.4	2.4	2.4	0.6
53	2.4	2.4	2.4	0.6	0.6
54	2.4	2.4	0.6	0.6	0.6
55	2.4	0.6	0.6	0.6	0.6
56	0.6	0.6	0.6	0.6	0.6

Recommendation X.55

INTERFACE BETWEEN SYNCHRONOUS DATA NETWORKS USING A 6 + 2 ENVELOPE STRUCTURE AND SINGLE CHANNEL PER CARRIER (SCPC) SATELLITE CHANNELS

(Malaga-Torremolinos, 1984)

The CCITT,

considering

(a) that the bearer rate recognized by the CCITT is 64 kbit/s;

(b) that 64 kbit/s satellite channels on TDMA systems are not yet operational;

(c) that 64 kbit/s channels on SCPC systems are under investigation;

(d) that for an interim period only 48, 50 or 56 kbit/s channels via a satellite will be available in many cases;

(e) that there is a requirement to interface transmission systems using an 8-bit envelope structure with such satellite systems,

recommends

that the fundamental parameters for a multiplexing scheme using an 8-bit envelope structure for transmission via a 56-kbit/s SCPC satellite channel should be as described in this Recommendation.

Note – In some cases, it may be necessary to adapt between 64 kbit/s and 56 kbit/s bearer rates.

1 Gross bit rate

For transmission on the international digital satellite link, the multiplexed bit stream shall have a gross bit rate of 56 kbit/s. On the tributary, each transmitted and received tributary data stream has the 8-bit envelope structure as recommended by Recommendation X.50.

The adaptation to the SCPC 56-kbit/s channel is achieved:

- by suppressing the F bits (framing bits);
- by using one out of 7 S bits (status bits) for framing.

2 Fundamental multiplex

2.1 *System capacity*

The capacity is the same as recommended in Recommendation X.50.

Note – Further study is required to accommodate user classes 7 and 11.

2.2 *Multiplex structure*

The multiplex is based on envelopes of 7 bits.

In a 7-bit envelope, bits 1-6 are information bits of the tributary channel and bit 7 is reserved for framing and signalling purposes (as described in Recommendation X.50) (see Figure 1/X.55).

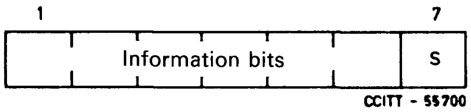


FIGURE 1/X.55

7-bit envelope

2.3 *Framing pattern*

One bit out of 7 S bits is used for framing purposes. The 72-bit framing pattern and the housekeeping bits ABCDEFGH as defined in Recommendation X.50 shall be used.

The resulting framing scheme is shown in Figure 2/X.55.

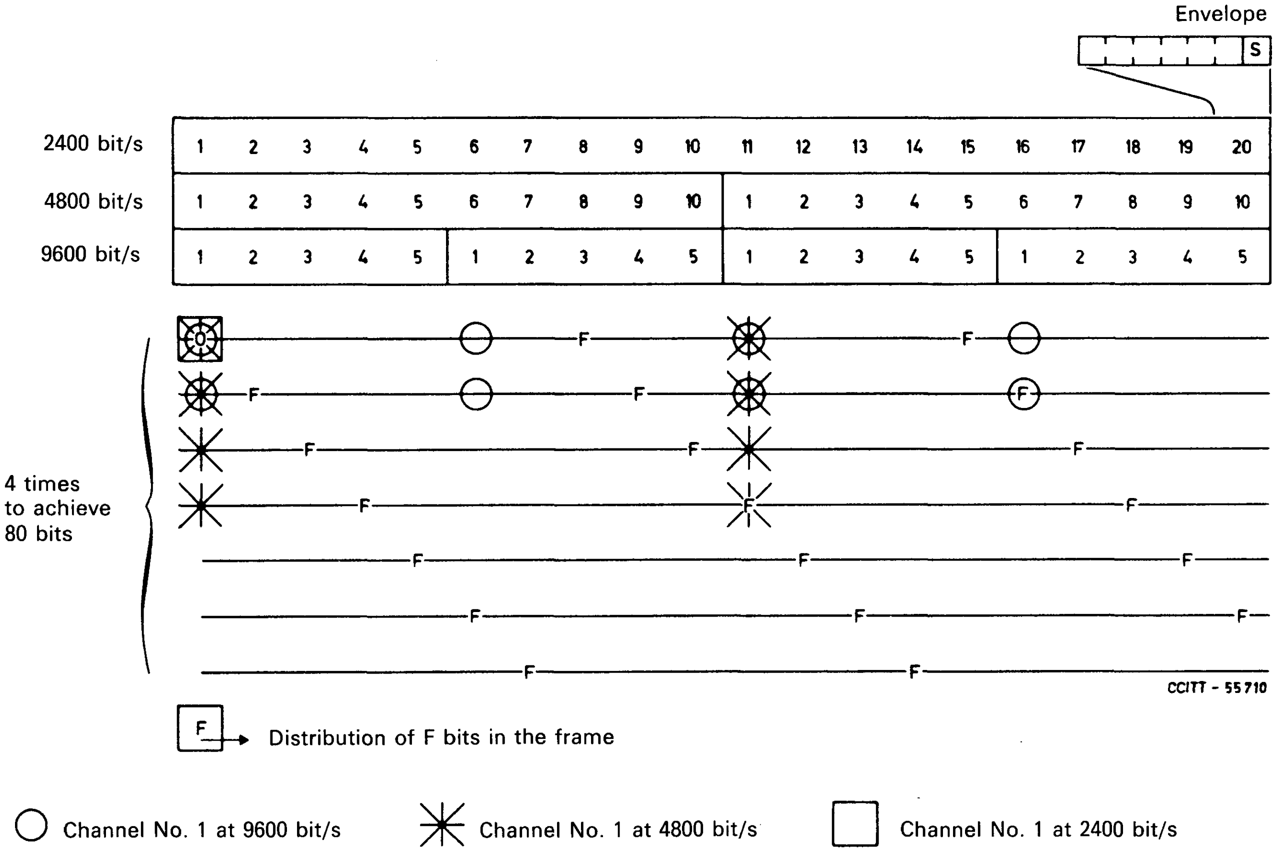


FIGURE 2/X.55

Framing scheme

2.4 *Framing strategy*

2.4.1 *Loss of frame alignment*

The criterion for loss of frame alignment shall be the reception of 8 consecutive erroneous F bits.

2.4.2 *Reframing*

The criterion for reframing shall be the detection of 8 consecutive F bits.

2.4.3 *Alarm and consequent action*

When a loss of alignment is achieved:

- the outgoing signals shall be set to all ones;
- the state shall be signalled to the distant end as recommended in Recommendation X.50 via the housekeeping bit A.

3 Adaptation between 56 kbit/s and 64 kbit/s bearers (when used)

The 6 + 2 envelope structure of the 64 kbit/s bearer is described in Recommendation X.50.

3.1 *Insertion and deletion of F bits*

3.2 *Sharing of S bits between framing and signalling mechanism*

a) At the transmit part:
One S bit out of 7 S bits is suppressed to be replaced by an F bit.

b) At the received part:
The F bit is suppressed and replaced by the last value of the S bit of the tributary channel.

Note – The mechanism, described above, is chosen due to the fact that the information on the S bit changes very slowly. The process will only bring a delay of 6 bits for the signal signalling transition when the corresponding S bit is affected.

For each channel, only one out of 7 S bits is affected.

Recommendation X.56

INTERFACE BETWEEN SYNCHRONOUS DATA NETWORKS USING AN 8 + 2 ENVELOPE STRUCTURE AND SINGLE CHANNEL PER CARRIER (SCPC) SATELLITE CHANNELS

(Malaga-Torremolinos, 1984)

The CCITT,

considering

(a) that the bearer rate recognized by the CCITT is 64 kbit/s;

(b) that 64 kbit/s satellite channels on TDMA systems are not yet operational;

(c) that 64 kbit/s channels on SCPC systems are under investigation;

(d) that for an interim period, only 48, 50 or 56 kbit/s channels via a satellite will be available in many cases;

(e) that there is a requirement of a multiplexing scheme for the interworking between two networks where both use 10-bit envelope structure but transmission is at a gross bit rate of 56 kbit/s, typically via an SCPC satellite system with forward error correction.

recommends

that the fundamental parameters for a multiplexing scheme using a 10-bit envelope structure for transmission via a 56-kbit/s SCPC satellite channel should be as described in this Recommendation.

1 Gross bit rate

For transmission on the international digital satellite link, the multiplexed bit stream shall have a gross bit rate of 56 kbit/s. The fundamental multiplex structure shall have a gross bit rate of 54 kbit/s and shall utilize padding techniques for transmission on the 56 kbit/s bearer channel. On the tributary channel interface, each transmitted and received tributary channel data stream has the 10-bit envelope structure as recommended in Recommendation X.51. The adaptation to the SCPC 56-kbit/s channel is achieved by suppressing the A bit of each envelope within the multiplex system.

2 Fundamental multiplex

For the fundamental multiplexing of information bearer channels, the following applies:

2.1 The signal elements of each individual channel shall be assembled in 9-bit envelopes, in which bit 1 is a status bit (S bit) (see Note), and bits 2-9 are information bits, as in Figure 1/X.56.

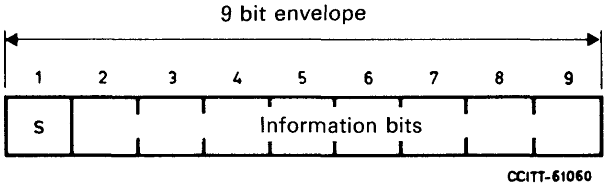


FIGURE 1/X.56

The addition of the status bit results in a 12½% increase in bit rate, so that the bearer channel rates are:

10.8 kbit/s for the 9.6 kbit/s data signalling rate;

5.4 kbit/s for the 4.8 kbit/s data signalling rate;

2.7 kbit/s for the 2.4 kbit/s data signalling rate;

675 bit/s for the 600 bit/s data signalling rate.

Note – A status bit (S bit) is associated with each envelope and in conjunction with the associated 8-bit data byte conveys call control information (cf. Recommendations X.21, X.21 *bis*, X.60, X.71 and X.50).

2.2 A 9-bit envelope interleaved structure shall be used.

2.3 These interleaved envelopes will appear on the 54 kbit/s fundamental multiplex as follows:

- 10.8 kbit/s channels will repeat every 5th envelope;
- 5.4 kbit/s channels will repeat every 10th envelope;
- 2.7 kbit/s channels will repeat every 20th envelope;
- 675 bit/s channels will repeat every 80th envelope.

2.4 Both structures suitable for handling homogeneous (with respect to bearer rates) mixes of bearer channels and structures suitable for handling heterogeneous mixes of bearer channels are required, with the constraint that the division of any 10.8 kbit/s bearer channel of the multiplex shall be homogeneous providing either two 5.4 kbit/s, four 2.7 kbit/s or sixteen 675 bit/s bearer channels.

3 Method of framing

3.1 *Overall structure*

The residual 2 kbit/s capacity obtained by carrying the fundamental 54 kbit/s multiplex on the 56 kbit/s bearer shall be distributed so that a padding bit is inserted after each group of 27 bits from the fundamental multiplex (see also Figure 2/X.56).

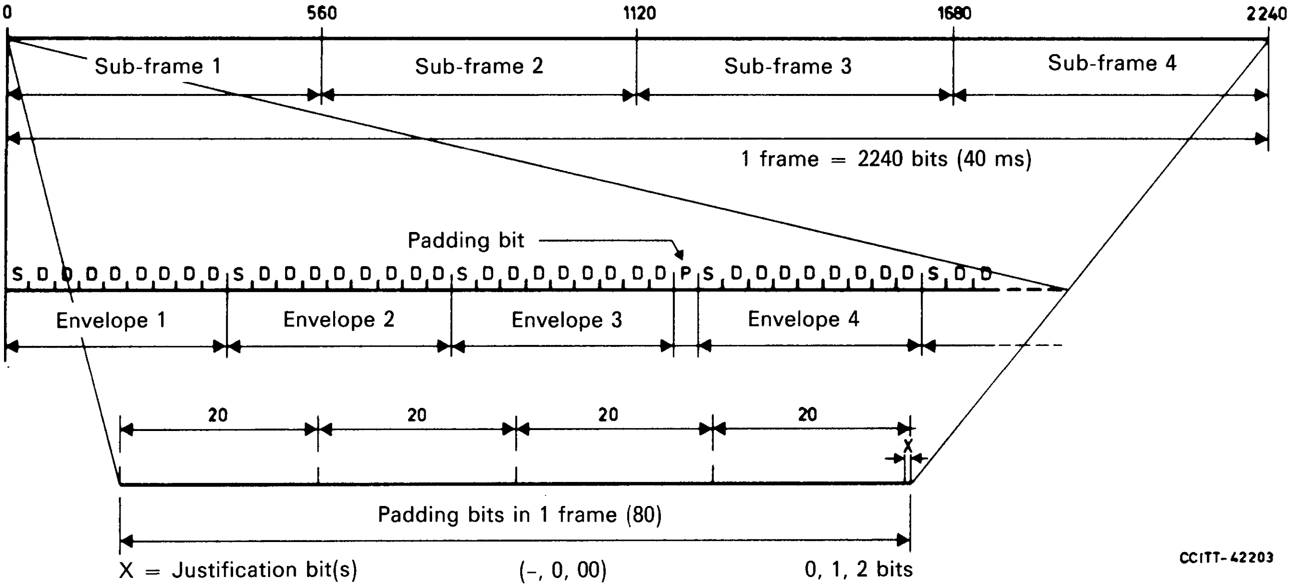


FIGURE 2/X.56

Multiplex frame structure

The frame length shall be 2240 bits in the case of a synchronized bearer, i.e. 2160 bits or 240 envelopes from the fundamental multiplex interleaved with 80 padding units.

When justification is used (for national purposes) in the case of a non-synchronized bearer, the last padding bit in the frame can be deleted or an extra padding bit added when needed, resulting in a variable frame length of 2240 ± 1 bit. (This can allow a maximum speed tolerance of approximately ± 4.5 parts in 10^4.)

The padding bits shall contain the framing pattern, justification service digits and housekeeping signalling (alarms, etc.).

3.2 *Framing*

3.2.1 *Frame alignment patterns*

The frame alignment method is based on the use of 4 equidistantly distributed frame alignment patterns written into the padding bits, dividing the frame into 4 sub-frames. Each sub-frame alignment pattern starts with the 14-bit pattern:

11111001101010

followed by a 2-bit sub-frame identifier unique to the sub-frame, i.e.:

SF1 = 00, SF2 = 01, SF3 = 10, SF4 = 11.

3.2.2 *Framing strategy*

3.2.2.1 *Loss of frame alignment*

The criterion for loss of frame alignment shall be three consecutive frame alignment patterns including the sub-frame identifier in error.

The frame alignment shall also be considered lost if the first received frame alignment pattern including sub-frame identifier after reframing is in error.

3.2.2.2 *Reframing*

The criterion for reframing shall be the detection of one valid frame alignment pattern.

3.2.2.3 *Reframing procedure*

After loss of frame alignment:

- the outgoing envelopes shall be set to all ones;
- the state shall be signalled to the distant end; and
- a parallel hunt for a valid frame alignment pattern shall be started.

After a valid frame alignment pattern is found:

- the two following padding bits shall be accepted as sub-frame identifiers and be used to set the frame and sub-frame counter(s) as applicable;
- the blocking of the outgoing data channels shall be removed; and
- the signalling of out-of-frame alarm to the distant end shall be terminated.

4 Justification

The 56 kbit/s bearer carrying the 9-bit envelope multiplex normally shall be locked to the data stream, and therefore justification on international links is not required. However, justification could be required for national purposes. To achieve this, plus/minus justification shall be used in which four repeated justification service signals occupy the 3 bits immediately following each sub-frame identifier. The last padding bit of the frame is used as a justification digit.

The repeated justification service signals are:

010 no justification (i.e. one padding bit at end of frame);

100 one justification bit has been added (i.e. two padding bits at end of frame);

001 the justification bit has been deleted (i.e. no padding bit at end of frame).

In evaluating the signals in one frame, a majority decision of the four received signals is used. In case of no majority, no justification shall be assumed.

If framing is lost, no justification shall be assumed before reframing has occurred.

5 Housekeeping signals and functions

The padding bits not used for framing and justification shall be available for housekeeping information signals, for both international and national use. The definition and allocation of the available housekeeping bits is left for further study.

6 Allocation and use of padding bits (20 bits) in one sub-frame (560 bits) for framing, justification and housekeeping

The allocation of padding bits in one sub-frame numbered P1 to P20 is described below and shown in Figure 3/X.56.

P1-P14	Framing pattern: 14 bits Code 11111001101010
P15-P16	Sub-frame identifier: 2 bits Code 00, 01, 10 or 11

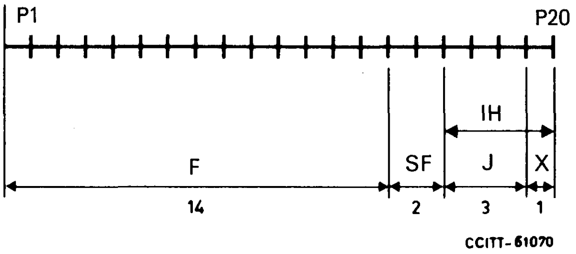
For P17-P20, two alternatives exist:

a) Synchronous transmission bearer

P17-P20	International housekeeping bits A, B, C and D (cf. Recommendation X.50)

b) Asynchronous transmission bearer

P17-P19	Justification service signals: 3 bits Code 001, 010, 100
P20	In the first three sub-frames (SF1, SF2, SF3), may remain as housekeeping bits as above. Their use is for further study.
P20(P21)	In the last sub-frame (SF4), is used for justification: Justification bit(s) ... 0, 1, 2 bit(s) Code –, 0, 00



IH	= International housekeeping	4 bits
F	= Frame alignment pattern	14 bits
SF	= Sub-frame identifier	2 bits
J	= Justification service signals	3 bits
X	= Justification or housekeeping bit (depending on sub-frame)	1 bit

FIGURE 3/X.56

Allocation of padding bits in one sub-frame (20 bits)

Recommendation X.57

METHOD OF TRANSMITTING A SINGLE LOWER SPEED DATA CHANNEL ON A 64 kbit/s DATA STREAM

(Malaga-Torremolinos, 1984)

The CCITT,

considering

(a) that the Recommendation X.1 defines the synchronous user classes of services to be provided by a public data network;

(b) that Recommendation X.50 defines some envelope data formats which can be used in public data networks;

(c) that the use of a full 64 kbit/s data stream for the transmission of a single lower speed synchronous data channel may be economically attractive in some applications, when a low cost 64 kbit/s channel is available;

(d) that for customer access to circuit switched data services in the ISDN, a method for adaption of synchronous user data rates to a 64 kbit/s bearer rate is defined in Recommendation X.30, but that in certain applications as mentioned under (c) a simpler method is preferred,

unanimously declares

that for networks using a 6 + 2 envelope structure, a single low speed synchronous data channel (i.e. 600, 2400, 4800, 9600 bit/s) shall be transmitted through a 64 kbit/s data stream by using the following method:

The 6 + 2 envelopes related to the low speed channel shall be repeated as many times as required to reach the 64 kbit/s speed.

The receive end can recover the original data signal by taking out of the received 64 kbit/s stream an envelope for every envelope period of the chosen data rate.

In the networks which transmit the 8 kHz envelope timing, no alignment circuit is required in the receiver; in the networks not transmitting the 8 kHz timing, the alignment is obtained by transmitting an alignment pattern in the framing bit position of each envelope. When the framing bit is not used for alignment, it is available to transmit housekeeping information or, when this is not required, it is set to zero. Further study is needed for the case of 8 + 2 envelope structure.

Recommendation X.58

FUNDAMENTAL PARAMETERS OF A MULTIPLEXING SCHEME FOR THE INTERNATIONAL INTERFACE BETWEEN SYNCHRONOUS NON-SWITCHED DATA NETWORKS USING NO ENVELOPE STRUCTURE

(Melbourne, 1988)

The CCITT

considering

(a) that Recommendation X.50 sets out the fundamental parameters for a multiplexing scheme for the interworking of networks where at least one makes use of the 8-bit envelope structure or of the four 8-bit envelopes grouping;

(b) that there is a requirement for a multiplexing scheme for the interworking between two networks where both use non-envelope structured data,

unanimously declares the view

that the following fundamental parameters shall be used between networks using no envelope structures.

1 Gross bit rate

For transmission on the international link the aggregate bit stream shall have a gross bit rate of 64 kbit/s.

2 Tributary channel bit rates

The following tributary channel bit rates are supported:

- 2.4 kbit/s
- 4.8 kbit/s
- 9.6 kbit/s
- 19.2 kbit/s

Other bit rates are not excluded.

3 Multiplex scheme

The multiplex scheme is shown in Figure 1/X.58. The frame length is 640 bit. The frame duration is 10 ms. Tributary channel data is grouped in octets and appears in slots An through Fn. Slots Sn contain synchronization octets. Slots Tn contain service octets.

160 bits = 20 octets

S1	A1	B1	C1	D1	E1	F1	B2	A2	D2	C2	F2	E2	A3	B3	C3	D3	E3	F3	T1
S2	B4	A4	D4	C4	F4	E4	A1	B1	C1	D1	E1	F1	B2	A2	D2	C2	F2	E2	T2
S3	A3	B3	C3	D3	E3	F3	B4	A4	D4	C4	F4	E4	A1	B1	C1	D1	E1	F1	T3
S4	B2	A2	D2	C2	F2	E2	A3	B3	C3	D3	E3	F3	B4	A4	D4	C4	F4	E4	T4

FIGURE 1/X.58

Multiplex scheme

3.1 *Data octets*

One frame contains 72 data octets. Thus the multiplex stream can cupport
24 channels of 2.4 kbit/s, or
12 channels of 4.8 kbit/s, or
6 channels of 9.6 kbit/s, or
3 channels of 19.2 kbit/s, or
combinations thereof.

The allocation of individual octets to a tributary channel is detailed below.

3.1.1 *2.4 kbit/s*

2.4 kbit/s tributary channels employ 1 out of 24 data octets. A 2.4 kbit/s channel will thus be allocated to all slots with the same identifier, i.e. identification letter and identification digit (e.g. A1).

3.1.2 *4.8 kbit/s*

4.8 kbit/s tributary channels employ 1 out of 12 data octets. A 4.8 kbit/s channel will thus be allocated to all slots with the same identification letter in the range A-F and two different identification digits 1 and 3 or 2 and 4 (e.g. B1 and B3).

3.1.3 *9.6 kbit/s*

9.6 kbit/s tributary channels employ 1 out of 6 data octets. A 9.6 kbit/s channel will thus be allocated to all slots with the same identification letter in the range A-F and four different identification digits 1, 2, 3 and 4 (e.g. D1, D2, D3 and D4).

3.1.4 *19.2 kbit/s*

19.2 kbit/s tributary channels employ 1 out of 3 data octets. A 19.2 kbit/s channel will thus be allocated to all slots with two different identification letters from the range A-F: A and D or B and E or C and F, and four different identification digits 1, 2, 3 and 4 (e.g. C1, F1, C2, F2, C3, F3, C4 and F4).

3.1.5 *Other bit rates*

For other bit rates no allocation of octets to a tributary channel is specified. From the scheme in Figure 1/X.58 it can be derived that any bit rate *n* times 2.4 kbit/s where *n* is 1 through 24 can be supported. Detailed allocation schemes should be specified by bilateral agreement.

3.2 *Synchronization octets*

One frame contains 4 synchronization octets. These contain fixed bit patterns as follows:

S1 = 27 = 00100111

S2 = 1B = 00011011

S3 = 05 = 00000101

S4 = 35 = 00110101

3.3 *Service octets*

The octets T1 through T4 are available for housekeeping information signals.

The following allocation is recommended but not mandatory.

Octet T1 contains eight bits, A, B, C, D, E, F, G and H (cf. Recommendation X.50). Bit A is the first bit transmitted.

The bit A is used to convey to the distant end alarm indications detected at the local end corresponding to:

- absence of incoming pulses,
- loss of frame alignment,

and that bit A shall be assigned such that:

- A equals 1 means no alarm,
- A equals 0 means alarm.

The other bits B, C, D, C, F, G and H are reserved to convey further international housekeeping signals. The exact use is under study. Pending the result of the study these bits shall be set to binary 1.

The octets T2 through T4 are reserved for national use and shall be set to binary 1 on an international link.

4 **Frame synchronization**

Frame synchronization is obtained by the receiving multiplexer during normal operation. No interaction between multiplexers at both ends of the link is required for this purpose.

Recommendation X.60

COMMON CHANNEL SIGNALLING FOR CIRCUIT SWITCHED DATA APPLICATIONS

(Geneva, 1980)

The CCITT,

considering

(a) that public networks providing circuit-switched data transmission services are being established in various countries;

(b) that common channel signalling offers advantages when used for interexchange signalling in digital circuit-switched telecommunication networks;

(c) that a need has been established for a standardized common channel signalling system, known as CCITT Signalling System No. 7, for use in international and national applications in single service and multiservices digital networks;

(d) that Signalling System No. 7 has been defined with a functional structure clearly separating:

- the Message Transfer Part, common for all services and applications, and
- User Parts for different services and applications, and in particular, the Data User Part for circuit-switched data applications,

unanimously declares the view

that common channel signalling for international circuit-switched data applications be in accordance with Signalling System No. 7, i.e.:

i) that the data service call control related elements of the signalling system be as specified for the Data User Part in Recommendation X.61,

ii) that the message transfer related elements of the signalling system be as specified for the Message Transfer Part in Recommendations Q.701-Q.707.

Note 1 – Signalling System No. 7 including the Data User Part offers a basis for definition of common channel signalling for national data applications.

Note 2 – The implications of the use of the signalling system in multiservices networks and Integrated Service Digital Networks (ISDN) providing circuit-switched data services have not yet been fully studied.

Recommendation X.61[1]

SIGNALLING SYSTEM No. 7 – DATA USER PART

(Former Recommendation X.60, Geneva, 1976
amended at Geneva, 1980, and Malaga-Torremolinos, 1984)

TABLE OF CONTENTS

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[1] This Recommendation appears in the Series Q Recommendations as Recommendation Q.741.

1 Functional description of the signalling system

1.1 *General*

Use of Signalling System No. 7 for call control or for facility registration and cancellation signalling for circuit-switched data transmission services requires:

- application of Data User Part (DUP) functions, in combination with
- application of an appropriate set of Message Transfer Part (MTP) functions.

A general description of the signalling system is given in Recommendation Q.701. That Recommendation also defines the division of functions and the requirements of interaction between the Message Transfer Part and the Data User Part.

1.2 *Data User Part*

The Data User Part specified in this Recommendation defines the necessary call control, and facility registration and cancellation related elements for international common channel signalling by use of Signalling System No. 7 for circuit-switched data transmission services. As regards call control and signalling procedures for international user facilities and network utilities, refer to Recommendation X.300.

The signalling system meets all requirements defined by CCITT concerning service features, user facilities and network utilities for circuit-switched data transmission services.

It can be used to control switching of various types of data circuits, including satellite circuits, to be used in worldwide circuit-switched data connections. It is designed for both-way operation of data circuits.

The signalling system is suitable for national circuit-switched data applications. Most data signalling message types and signals specified for international use are also required in typical national data applications. In addition to these, national data applications typically require additional types of signals; such requirements that have been identified are already provided for. The system provides ample spare capacity to cater for further additions of new message types and signals should such a need arise.

The label structures specified for data signalling messages require that all exchanges using the signalling system are allocated codes from code plans established for the purpose of unambiguous identification of signalling points, see Recommendations Q.701 and Q.704. The principles to apply to the international signalling network are specified in Recommendation Q.708.

1.3 *Message Transfer Part*

The Message Transfer Part of Signalling System No. 7 is specified in separate Recommendations. An overview description of the Message Transfer Part is contained in Recommendation Q.701.

The Message Transfer Part defines a range of functions by which different signalling modes and different signalling network configurations may be realized. Any application of Signalling System No. 7 requires that an appropriate selection of these functions is applied depending on the intended use of the system and the characteristics of the telecommunications network concerned.

2 General function of data signalling messages, signals, indicators, codes and conditions

This paragraph describes general functions of data signalling messages, signals, indicators, codes and conditions which are used to set up a call, to control user facilities and to control and supervise a circuit. The requirements relating to the use of the signalling messages and their signalling information content are specified in §§ 3, 4, and 5.

2.1 *Signalling messages*

2.1.1 *Call and circuit related messages*

Call and circuit related messages are used to set up and clear a call or control and supervise the circuit state.

2.1.1.1 *Address message*

A message sent in the forward direction, containing signalling information required to route and connect the call to the called user. This message contains address information, class of service information, etc., and may also contain additional information such as, for example, calling line identity.

2.1.1.2 *Calling line identity message*

A message sent in the forward direction, containing the calling line identity or the originating network identity. This message is sent subsequently to an address message, which does not contain the calling line identity, when requested by the destination network.

2.1.1.3 *Call accepted message*

A message sent in the backward direction, containing information to indicate that connection of the call is allowed by the destination exchange. It may also contain additional information such as, for example, called line identity.

2.1.1.4 *Call rejected message*

A message sent in the backward direction containing a signal to indicate the cause of the failure of the call set-up as the response to the address message and initiating clearing of the call. The call rejected message will be sent as either the first response, or the second response after sending the call accepted message when the call fails to be completed at the destination exchange, e.g. because no call accepted signal was received from the called user.

2.1.1.5 *Clear message*

A message sent in either direction, containing information about the clearing of the call.

2.1.1.6 *Circuit state message*

A message sent in either direction, containing signals to control and supervise a circuit.

2.1.2 *Facility registration and cancellation related messages*

Facility registration and cancellation related messages are used to exchange information between originating and destination exchanges to register and cancel information related to user facilities. The exchange of this type of message is generally not associated with a call between two users.

2.1.2.1 *Facility registration/cancellation request message*

A message sent in the forward direction to register or cancel a user facility. This message contains information which identifies the user requesting facility registration or cancellation and information relating to the facility concerned.

2.1.2.2 *Facility registration/cancellation request accepted message*

A message sent in the backward direction, containing information that registration or cancellation is completed or accepted at the destination exchange.

2.1.2.3 *Facility registration/cancellation request rejected message*

A message sent in the backward direction, containing information that the registration or cancellation is not completed or accepted at the destination exchange with information indicating a reject cause.

2.2 *Service information*

The service information provides the highest level of discrimination between different sets of signalling messages. It contains the following components.

2.2.1 *Service indicator*

Information used to identify the User Part to which the signalling message belongs.

2.2.2 *National indicator*

Information used for discrimination between international and national messages. In case of national messages, it may for example also be used for discrimination between different label alternatives for national use.

2.3 *Signalling information transferred in the signalling messages*

2.3.1 *Label components*

In the case of call and circuit related messages, the label is used for message routing and, in general, for identification of the data circuit selected for the call. In the case of facility registration and cancellation messages, the label only provides a message routing function. The standard label structure consists of the following components.

2.3.1.1 *Destination point code*

Information identifying the signalling point to which the message is to be routed.

2.3.1.2 *Originating point code*

Information identifying the signalling point from which the message has been originated.

2.3.1.3 *Bearer identification code*

Information identifying the 64 kbit/s bearer among those interconnecting the destination point and originating point.

2.3.1.4 *Timeslot code*

Information identifying the submultiplexed circuit at a lower bit rate on the 64 kbit/s bearer which is itself identified by the bearer identification code.

2.3.2 *Message format identifiers*

2.3.2.1 *Heading*

Information discriminating, as applicable, between different groups of individual types of messages within the set of messages identified by the service information. The heading is split into two levels. The first level discriminates between different message groups. The second level either discriminates between different message types or contains a signal.

2.3.2.2 *Field length indicator*

Information associated with and indicating the length of a variable length field.

2.3.2.3 *Field indicator*

Information associated with and indicating the presence or absence of an optional field.

2.3.3 *Basic call set-up address information*

2.3.3.1 *Address signal*

A signal containing an element of a Data Country Code (DCC), Data Network Identification Code (DNIC) or a data number.

2.3.3.2 *Destination address*

Information sent in the forward direction consisting of a number of address signals indicating the complete data number of the called user.

2.3.4 *Basic call set-up indicators*

2.3.4.1 *National/international call indicator*

Information (for national use only) sent in the forward direction indicating whether the call is a national or international call. In the destination network, it may for example be used in connection with user facilities requiring separate handling of international calls.

2.3.4.2 *DCC/DNIC indicator*

Information (for national use only) sent in either direction, associated with a data number, indicating whether the DCC/DNIC is included in that data number.

2.3.4.3 *Alternative routing indicator*

Information sent in the forward direction indicating that the call has been subjected to an alternative routing and which may be used to prevent the call being set up over an alternative route more than once.

Note – This signal is provided provisionally, and will be subject to change when the necessary network capabilities for routing have been determined.

2.3.4.4 *User class indicator*

Information sent in the forward direction, indicating the user class of service of the calling user. This indicator may be used to determine the type of interexchange data circuit to be selected and to verify that the calling and called users belong to the same user class.

2.3.5 *Basic call set-up response signals*

2.3.5.1 *Call accepted signal*

A signal sent in the backward direction, indicating that the call can be completed. At the originating exchange, it results in preparing for data path through-connect and charging.

2.3.5.2 *Transit through-connect signal*

A signal sent in the backward direction, specifically provided for interworking with decentralized signalling (see Recommendation X.80), indicating that the call can be completed and that through-connection of transit exchanges using decentralized signalling may take place.

2.3.5.3 *Network failure signal*

A signal sent in the backward direction indicating that the call cannot be completed because of a temporary fault condition within the network, e.g. expiry of a time-out or line fault. At the originating exchange it results in sending a *no connection* call progress signal to the calling user and clearing the call.

2.3.5.4 *Number busy signal*

A signal sent in the backward direction, indicating that the call cannot be completed because the called user's access line to the exchange is engaged in another call. At the originating exchange it results in sending a *number busy* call progress signal to the calling user and clearing the call.

2.3.5.5 *Access barred signal*

A signal sent in the backward direction, indicating that the call cannot be completed because a user facility prevents connection of the call to the called user, e.g. as a result of failure of a closed user group validation check. At the originating exchange, it results in sending an *access barred* call progress signal to the calling user and clearing the call.

2.3.5.6 *Changed number signal*

A signal sent in the backward direction, indicating that the call cannot be completed because the called number has been changed recently. At the originating exchange, it results in sending a *changed number* call progress signal to the calling user and clearing the call.

2.3.5.7 *Not obtainable signal*

A signal sent in the backward direction, indicating that the call cannot be completed because the called number is not in use or assigned. At the originating exchange, it results in sending a *not obtainable* call progress signal to the calling user and clearing the call.

2.3.5.8 *Out of order signal*

A signal sent in the backward direction, indicating that the call cannot be completed because either the called user's terminal or the called user's access line is out of service or faulty. At the originating exchange, it results in sending an *out of order* call progress signal to the calling user and clearing the call.

2.3.5.9 *Controlled not ready signal*

A signal sent in the backward direction, indicating that the call cannot be completed because the called user's terminal is in a *controlled not ready* condition. At the originating exchange, it results in sending a *controlled not ready* call progress signal to the calling user and clearing the call.

2.3.5.10 *Uncontrolled not ready signal*

A signal sent in the backward direction, indicating that the called user's terminal is in an *uncontrolled not ready* condition. At the originating exchange, it results in sending an *uncontrolled not ready* call progress signal to the calling user and clearing the call.

2.3.5.11 *DCE power off signal*

A signal sent in the backward direction, indicating that the called user's DCE is switched off. At the originating exchange, it results in sending a *DCE power off* call progress signal to the calling user and clearing the call.

2.3.5.12 *Network fault in local loop signal*

A signal sent in the backward direction, indicating that a fault has been detected in the local access connection for the called user. At the originating exchange, it results in sending a *network fault in local loop* call progress signal to the calling user and clearing the call.

2.3.5.13 *Call information service signal*

A signal sent in the backward direction, indicating that the called terminal is not available for reasons which have been indicated to the information service, and which are not covered by another specific signal. At the originating exchange, it results in sending a *call information service* call progress signal to the calling user and clearing the call.

2.3.5.14 *Incompatible user class of service signal*

A signal sent in the backward direction, indicating that the called user's terminal is incompatible with the characteristics of the calling user's terminal, e.g. different user class of service. At the originating exchange, it results in sending an *incompatible user class of service* call progress signal to the calling user and clearing the call.

2.3.5.15 *Network congestion signal*

A signal sent in the backward direction, indicating that the call cannot be completed because of temporary congestion or temporary fault conditions encountered on the route to the called customer. At the originating exchange this signal results in sending a *network congestion* call progress signal to the calling user and clearing the call.

2.3.5.16 *Degraded service signal*

A signal sent in the backward direction, indicating that a part of the network, due to faulty conditions, has a very much reduced grade of service, which is likely to persist for some time. At the originating exchange, it results in sending a *long-term network congestion* call progress signal to the calling user and clearing the call.

2.3.5.17 *Charge/no charge indicator*

Information (for national use only) sent in the backward direction that may be used to indicate that the call should not be charged at the originating exchange.

2.3.6 *Basic call clearing and circuit state signals*

2.3.6.1 *Circuit released signal*

A signal sent in either direction indicating that the interexchange data circuit has been released.

2.3.6.2 *Circuit released acknowledgement signal*

A signal sent in either direction in response to the *circuit released* signal and indicating that the interexchange data circuit has been released.

2.3.6.3 *Reset circuit signal*

A signal sent to return the interexchange data circuit to the idle state at both ends in situations where, due to memory mutilation or other causes, the state of the circuit is ambiguous.

2.3.6.4 *Blocking signal*

A signal sent for maintenance purposes indicating to the exchange at the other end of the interexchange data circuit that the circuit has to be blocked for outgoing calls.

2.3.6.5 *Unblocking signal*

A signal sent to cancel the blocked condition at the exchange at the other end of the interexchange data circuit caused by an earlier *blocking* signal.

2.3.6.6 *Blocking acknowledgement signal*

A signal sent in response to a *blocking* signal indicating that the interexchange data circuit has been blocked.

2.3.6.7 *Unblocking acknowledgement signal*

A signal sent in response to an *unblocking* signal indicating that the interexchange data circuit has been unblocked.

2.3.7 *Additional signals relating to the closed user group facilities*

2.3.7.1 *Closed user group call indicator*

Information sent in the forward direction and in some circumstances in the backward direction, indicating whether or not the call involves a closed user group, whether an interlock code is included in the message and whether or not outgoing access is allowed for the calling user.

2.3.7.2 *Interlock code*

Information sent in the forward direction, and in some circumstances, in the backward direction, identifying a closed user group to which the calling user belongs.

2.3.8 *Additional signals relating to the bilateral closed user group and the bilateral closed user group with outgoing access facilities*

2.3.8.1 *Bilateral closed user group call indicator*

Information sent in the forward direction, indicating whether or not the call is a call within a bilateral closed user group.

2.3.8.2 *Registration request signal*

A signal sent in the forward direction, indicating that facility registration is required.

2.3.8.3 *Cancellation request signal*

A signal sent in the forward direction, indicating that facility cancellation is required.

2.3.8.4 *Registration completion signal*

A signal sent in the backward direction, indicating that facility registration is completed at the destination exchange. At the originating exchange, it results in sending a *registration/cancellation confirmed* call progress signal to the calling user.

2.3.8.5 *Registration accepted signal*

A signal sent in the backward direction, indicating that facility registration is accepted at the destination exchange. At the originating exchange it results in sending a *registration/cancellation confirmed* call progress signal to the calling user.

2.3.8.6 *Cancellation completed signal*

A signal sent in the backward direction, indicating that facility cancellation is completed at the destination exchange. At the originating exchange it results in sending a *registration/cancellation confirmed* call progress signal to the calling user.

2.3.8.7 *Local index*

Information sent in the forward direction and backward direction at bilateral closed user group registration. It indexes the subscriber file to identify the particular bilateral closed user group at the originating or destination exchange.

2.3.9 *Additional signals relating to the calling line identification facility*

2.3.9.1 *Calling line identification request indicator*

Information sent in the backward direction, indicating whether or not the calling line identity should be sent forward.

2.3.9.2 *Calling line identity indicator*

Information sent in the forward direction, indicating whether, and what form of, a calling line identity is included in the message.

2.3.9.3 *Calling line identity*

Information sent in the forward direction, consisting of a number of address signals indicating the (international) data number of the calling user.

2.3.10 *Additional signals relating to the called line identification facility*

2.3.10.1 *Called line identification request indicator*

Information sent in the forward direction, indicating whether or not the called line identity should be returned.

2.3.10.2 *Called line identity indicator*

Information sent in the backward direction, indicating whether, and what form of, the called line identity is included in the message.

2.3.10.3 *Called line identity*

Information sent in the backward direction, consisting of a number of address signals indicating the (international) data number of the called user.

2.3.11 *Additional signals relating to redirection of calls facility*

2.3.11.1 *Redirection request signal*

A signal (for national use only) sent in the backward direction, indicating that the called user has requested redirection of calls to another address.

2.3.11.2 *Redirection address indicator*

Information (for national use only) sent in the backward direction, indicating that a redirection address is included in the message.

2.3.11.3 *Redirection address*

Information (for national use only) sent in the backward direction, consisting of a number of address signals, indicating the data number to which the call is to be redirected.

2.3.11.4 *Redirected call indicator*

Information (for national use only) sent in the forward direction, indicating that the call is a redirected call. This indicator is used to prevent a further redirection, if the user at the new address has also requested redirection of calls.

2.3.11.5 *Redirected call signal*

A signal sent in the backward direction, indicating that the call has been redirected to an address other than the destination address selected by the calling user. At the originating exchange, it results in sending a *redirected* call progress signal.

2.3.12 *Additional signals relating to the connect when free and waiting allowed facilities*

2.3.12.1 *Connect when free signal*

A signal sent in the backward direction, indicating that the called user, having the *connect when free* facility, is busy and that the call has been placed in a queue. At the originating exchange it results in sending a *connect when free* call progress signal to the calling user if he has the *waiting allowed* facility or, if not, in sending the *number busy* call progress signal and clearing the call.

2.3.13 *Additional signals relating to the reverse charging and reverse charge acceptance facilities*

2.3.13.1 *Reverse charging request indicator*

Information sent in the forward direction, indicating that reverse charging is requested by the calling user.

2.3.13.2 *Reverse charge acceptance not subscribed signal*

A signal sent in the backward direction, indicating rejection of the call because the called user does not subscribe to the *reverse charge acceptance* facility. At the originating exchange it results in sending a *reverse charge acceptance not subscribed* call progress signal to the calling user.

2.3.14 *Additional signals relating to manual answer*

2.3.14.1 *Terminal called*

A signal sent in the backward direction, indicating that the called user operates with manual answer. At the originating exchange it results in sending a *terminal called* call progress signal to the calling user.

2.3.15 *Additional signals relating to the RPOA selection facilities*

2.3.15.1 *RPOA selection indicator*

Information (for national use only) sent in the forward direction, indicating whether or not the calling user requires selection of an RPOA for international call routing at the international gateway. When RPOA selection is required, it also indicates that a RPOA transit network identity is included in the message.

2.3.15.2 *RPOA transit network identity*

Information (for national use only) sent in the forward direction, identifying the requested RPOA transit network by its DNIC.

2.3.15.3 *RPOA out of order signal*

A signal (for national use only) sent in the backward direction, indicating that the call cannot be completed, because the selected RPOA transit network is not available for service. At the originating exchange, it results in sending an *RPOA out of order* call progress signal to the calling user.

2.3.16 *Additional signals relating to the network identification utilities*

2.3.16.1 *Network identity*

Information sent in either direction, identifying an originating, a transit or destination network by its DNIC.

2.3.16.2 *Originating network identification request indicator*

Information sent in the backward direction, indicating whether or not the originating network identity should be sent forward.

2.4 *Data channel signalling conditions*

These are interexchange data channel conditions employed in the call set-up and clear-down procedures. The conditions defined in § 2.4 are based on the characteristics of the relevant DTE/DCE interfaces for the circuit-switched service. The implications of other possible new DTE/DCE interfaces on these conditions have not yet been determined.

2.4.1 *Trunk free condition*

A condition transmitted in the forward or backward interexchange data channels when the circuit is free or under release at the sending exchange.

2.4.2 *Trunk seized condition*

A condition transmitted in the forward interexchange data channel when the circuit is seized but not through-connected.

2.4.3 *Call accepted condition*

A condition appearing in the backward interexchange data channel, indicating that all the succeeding exchanges involved in the connection have connected through. This condition is sent by the called user and corresponds to the *call accepted* state at the DTE/DCE interface.

2.4.4 *Clear request condition*

A condition, appearing in the forward and backward interexchange data channels, sent by the user when requesting to clear the call.

3 Formats and codes

3.1 *Basic format characteristics*

3.1.1 *General*

The data signalling messages are carried on the signalling data link by means of Signal Units (SU), the format of which is described in the specifications of the Message Transfer Part (MTP), see Recommendation Q.703.

The data signalling messages are divided into two categories, call and circuit related messages and facility registration and cancellation related messages. The Service Indicator (SI) included in each signal unit identifies to which category the message belongs.

The signalling information of each message constitutes the Signalling Information Field (SIF) of the corresponding SU signal unit and consists of an integral number of octets. It basically contains the label, the heading code and one or more signals and/or indicators.

3.1.2 *Service information octet*

3.1.2.1 *Format*

The service information octet comprises the service indicator and the subservice field.

The service indicator is used to associate signalling information with a particular User Part and is only used with message signal units (see Recommendation Q.703).

The information in the subservice field permits a distinction to be made between national and international signalling messages. In national applications when this discrimination is not required, possibly for certain national User Parts only, the subservice field can be used independently for different User Parts.

The format of the service information octet is shown in Figure 1/X.61.

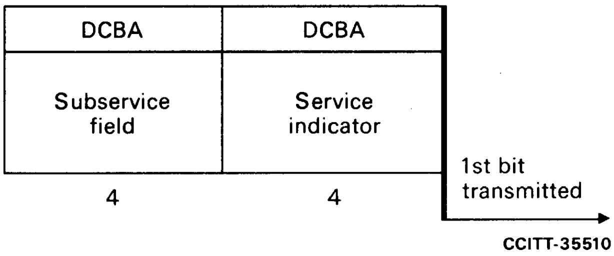


FIGURE 1/X.61

Service information octet

3.1.2.2 *Service indicator*

The service indicator will be coded as follows:

Bits: DCBA

0 1 1 0 call and circuit related messages
0 1 1 1 facility registration and cancellation messages.

The use of other service indicator codes is specified in Recommendation Q.704.

3.1.2.3 *Subservice field*

The subservice field is coded as shown in Table 1/X.61.

TABLE 1/X.61

Bits	B A	Spare
	D C	National indicator
	0 0	International message
	0 1	Spare (for international use)
	1 0	National message
	1 1	Reserved for national use

Note – Bits A and B are spare for possible needs that may require a common solution for all international User Parts and MTP level 3. Each bit is coded 0.

3.1.3 *Format principles*

The user generated information in the signalling information field is, in general, divided into a number of subfields which may be of either fixed or variable length. The first field is the label field, see § 3.2. Following the label field is a heading code H0 which, possibly together with a following subheader H1, identifies the structure of the message. Other fields may be mandatory or optional on a per individual message basis, the presence or absence of optional fields being indicated by field indicators. Each field indicated below is mandatory unless explicitly indicated as optional.

3.1.4 *Order of bit transmission*

Within each defined subfield the information is transmitted least significant bit first.

3.1.5 *Coding of spare bits*

Each spare bit is coded 0 unless otherwise indicated.

3.1.6 *Indicators for national use only*

A number of indicators specified are indicated as for national use only. In international use the corresponding bits are coded 0 and are, as regards their interpretation, equivalent to spare bits.

3.2 *Label*

3.2.1 *General*

The label is an item of information which forms part of every signalling message and is used by the message routing function at MTP Level 3 to select the appropriate signalling route and by the User Part function to identify the particular transaction (e.g. the call) to which the message pertains.

In general, the label information encompasses an explicit or implicit indication of the message source and destination and, depending on the application, various forms of transaction identification.

For call and circuit related messages the transaction is conveniently identified by including the corresponding circuit identity in the label. In the following, two such label structures are specified:

- a basic label structure which, consistent with the standard telephone label structure (Recommendation Q.723), is designed to meet the requirements for identification of data circuits derived from standard data multiplexers (see Recommendations X.50 and X.51);
- an alternative label structure, identical to the standard telephone label structure, which may be used in applications where the data circuits use full 64 kbit/s digital circuits without submultiplexing.

For facility registration and cancellation related messages the specified label structure is equivalent to the standard routing label of the MTP, see Recommendation Q.704.

Note – The indication (48) below the label field in Figures 5/X.61 to 11/X.61 refers to the basic label, which is described in § 3.2.2, but is intended to show that other label lengths are possible.

3.2.2 *Basic label for call and circuit related messages*

3.2.2.1 *General*

The basic label has a length of 48 bits and is placed at the beginning of the signalling information field. The format is as shown in Figure 2/X.61.

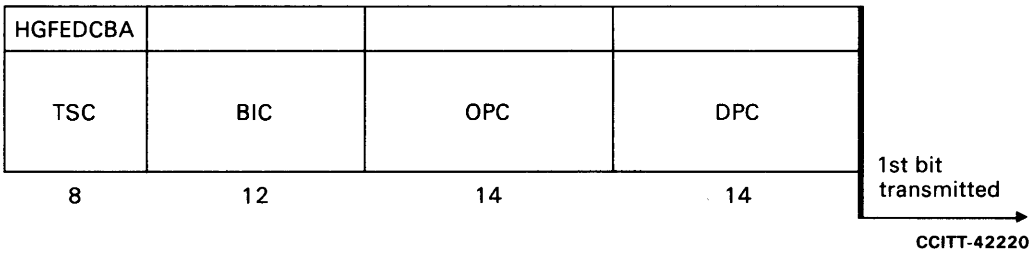


FIGURE 2/X.61
Basic label for data call and circuit related messages

The general function of the label components is defined in § 3.2.1. The portion of the basic label that consists of the Destination Point Code (DPC) and Originating Point Code (OPC) fields and the four least significant bits of the Bearer Identification Code (BIC) field corresponds to the standard routing label specified in Recommendation Q.704.

3.2.2.2 *Destination and originating point codes*

The standard structure requires that each data switching exchange in its role as a signalling point is allocated a code from a code plan established for the purpose of unambiguous identification of signalling points.

Separate code plans will be used for the international signalling network and for different national signalling networks.

The principles of code allocation and the codes for the international signalling network are specified in Recommendation Q.708.

The destination point code will be the code applicable to the data switching exchange to which the message is to be delivered. The originating point code will be the code applicable to the data switching exchange from which the message is sent.

3.2.2.3 *Bearer identification code*

The allocation of bearer identification codes to individual bearers is determined by bilateral agreement and/or in accordance with applicable predetermined rules.

For bearers which form part of a 2.048 Mbit/s PCM system according to Recommendation G.734, the bearer identification code contains in the 5 least significant bits a binary representation of the actual number of the time slot which is assigned to the bearer. The remaining bits of the bearer identification code are used where necessary, to identify one among several systems, interconnecting the originating point and destination point.

For bearers which form part of a 8.448 Mbit/s PCM system the bearer identification code will be coded in accordance with the scheme specified for the circuit identification code for the corresponding case in Recommendation Q.723.

3.2.2.4 *Time slot code*

The coding of the time slot code (TSC) is as follows (bit numbering as in Figure 2/X.61):

a) In the case where the data circuit is derived from the data multiplex carried by the bearer, identified by the bearer identification code:
   - bits ABCD will contain, in pure binary representation, the channel number of the circuit within the 12.8 kbit/s (Recommendation X.50) or 12 kbit/s (Recommendation X.51) phase; the channel number being in the range (see Recommendations X.50, X.51, X.53 and X.54):

     0-15 for 600 bit/s circuits
     0- 3 for 2400 bit/s circuits
     0- 1 for 4800 bit/s circuits
     0 for 9600 bit/s circuits
   - bits EFG will contain, in pure binary representation, the number of the 12.8 kbit/s or 12 kbit/s phase, the phase number being in the range 0-4;
   - bit H will be coded 0.

b) In the case where the data circuit uses the full 64 kbit/s bearer rate, the time slot code will be 01110000.

3.2.3 *Alternative label for call and circuit related messages*

In applications where all data circuits use full 64 kbit/s digital circuits, a label structure as shown in Figure 3/X.61 may be used in mutual agreement.

This label structure is equivalent to the standard telephone label structure specified in Recommendation Q.704. The destination point code (DPC) and originating point code (OPC) fields are as in the basic label structure and the Circuit Identification Code (CIC) is as the bearer identification code field in the basic label structure (see § 3.2.2).

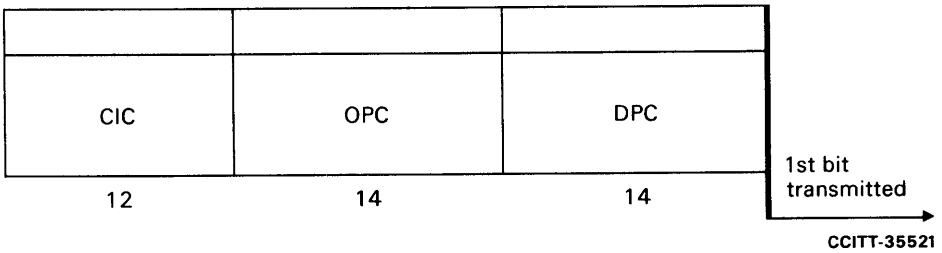


FIGURE 3/X.61

Alternative label for data and circuit related messages

3.2.4 *Standard label for facility registration and cancellation messages*

Facility registration and cancellation messages will have a label in accordance with Figure 4/X.61.

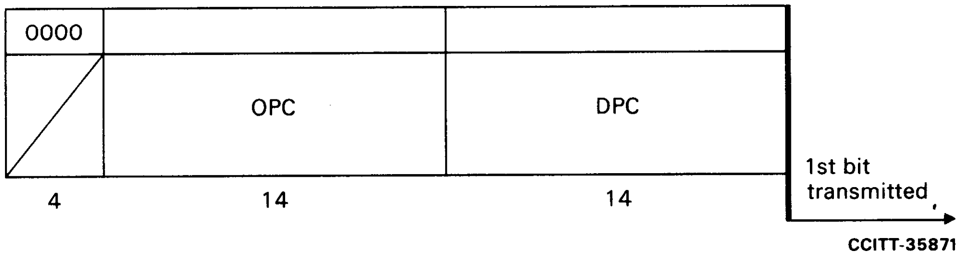


FIGURE 4/X.61
Standard label for facility registration and cancellation messages

This label structure is equivalent to the standard routing label specified for the MTP (see Recommendation Q.704). The destination point code (DPC) and originating point code (OPC) fields are used as for the basic label, see § 3.2.2.

3.2.5 *Modified label*

In cases where the data transmission service is provided by public data networks comprising few exchanges and signalling relations, it may be attractive to use shorter labels than those specified in §§ 3.2.2 to 3.2.4. In such applications a modified label, having the same order and function, but possibly different sizes, of subfields may be used in mutual agreement. In such a case the label used for MTP Level 3 messages should be modified accordingly. Also, in some national applications it may be necessary to use an extended modified label.

3.3 *Formats and codes for call and circuit related messages*

3.3.1 *Heading*

The different heading codes (H0) for the call and circuit control messages are allocated as shown in Table 2/X.61.

TABLE 2/X.61

0000	Spare
0001	Address message
0010	Calling line identification messages
0011	Spare
0100	Call accepted messages
0101	Call rejected messages
0110	Clear messages
0111	Circuit state messages
1000 to 1111	Spare

3.3.2 *Address message*

3.3.2.1 The format of the address message is as shown in Figures 5/X.61, and 5 *bis*/X.61.

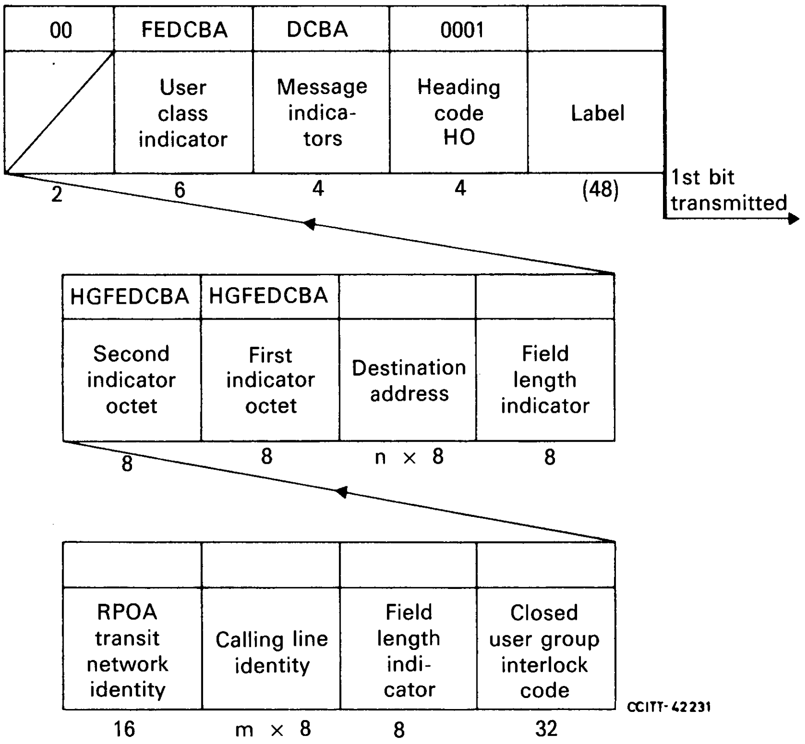


FIGURE 5/X.61

Address message

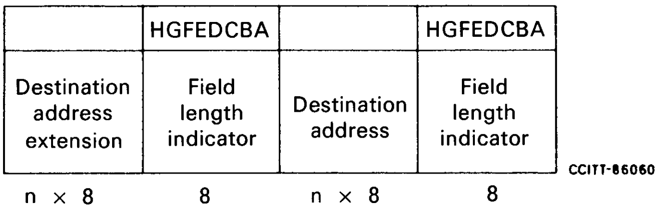


FIGURE 5 *bis*/X.61

Address message (destination address fields when address extension is used)

The fields, subfields and codes are as follows:

3.3.2.2 *Label*

See § 3.2.

3.3.2.3 *Heading code H0*

See § 3.3.1.

3.3.2.4 *Message indicator*

The coding is shown in Table 3/X.61.

TABLE 3/X.61

Bit A	Field indicator of first indicator octet
0	First indicator octet not included
1	First indicator octet included
B	DCC/DNIC indicator (national use only, see § 3.1.6)
0	DCC/DNIC included in destination address
1	DCC/DNIC not included in destination address
C	National/international call indicator (national use only, § 3.1.6)
0	International call
1	National call
D	Alternative routing indicator
0	No alternative routing made
1	Alternative routing made

3.3.2.5 *User class indicator*

The coding is shown in Table 4/X.61.

TABLE 4/X.61

Bits F E D C B A		
0 0 0 0 0 0 to 1 0 0 0 0 0	Spare	
1 0 0 0 0 1 to 1 0 0 1 1 0	Asynchronous user classes, as applicable; bitsABC coded as bits b1, b2, b3 of first user class character in Recommendation X.71	
1 0 0 1 1 1 to 1 0 1 1 1 1	Spare	
1 1 0 0 0 0	600 bit/s (user class 3)	Synchronous user classes corresponding to second user class charter in Recommendation X.71
1 1 0 0 0 1	2 400 bit/s (user class 4)	
1 1 0 0 1 0	4 800 bit/s (user class 5)	
1 1 0 0 1 1	9 600 bit/s (user class 6)	
1 1 0 1 0 0	48 000 bit/s (user class 7)	
1 1 0 1 0 1 to 1 1 1 0 1 1	Spare	
1 1 1 1 0 0 to 1 1 1 1 1 1	Reserved for national use	

3.3.2.6 *Spare bits*

3.3.2.7 *Field length indicator for destination address/address and sub-address*

The coding is shown in Table 5/X.61.

TABLE 5/X.61

Bits B A	00	The following bits of the field length indicator contain the number of digits of address and sub-address information
	01	The following bits of field length indicator contain the number of digits of address information; sub-address information follows in the address extension field
	10	Reserved
	11	Reserved
H G F E D C		A code expressing in pure binary representation the number of address signals (or address and sub-address signals) in the destination address. The maximum number of digits is limited to 32 (see Note)

Note – The maximum length of 32 decimal digits is derived from the provisional maximum length of the OSI Network Service Access Point (NSAP) address defined in Recommendation X.213.

3.3.2.8 *Destination address/address and sub-address field*

This field is divided into an even number of semi-octets. The decimal value of each destination address/address and sub-address digit is expressed in pure binary representation of an address/address and sub-address signal. The digits are sent in descending order with the most significant digit first. In case of an odd number of address/address and sub-address signals, a 4-bit 0000 filler code is included in the last semi-octet of the field.

3.3.2.9 *Field length indicator for address extension*

This field is an optional field that is included if bit A of the field length indicator for the destination address is equal to 1.

This field contains a code expressing in pure binary representation the number of sub-address signals in the destination address.

The coding is shown in Table 6/X.61.

TABLE 6/X.61

Bits B A	Reserved, coded 00
H G F E D C	The maximum number of digits is limited to 32 (see Note).

Note – See the Note to Table 5/X.61.

3.3.2.10 *Address extension for destination address*

This field is an optional field that is included if bit A of the field length indicator for the destination address is equal to 1.

This field is divided into an even number of semi-octets. The decimal value of each destination sub-address digit is expressed in pure binary representation of a sub-address signal. The digits are sent in descending order with the most significant digit first. In case of an odd number of sub-address signals a 4-bit filler code (0000) is included in the last semi-octet of the field.

3.3.2.11 *First indicator octet*

This is an optional field that is included if indicated in bit A of the message indicators. The coding is shown in Table 7/X.61.

TABLE 7/X.61

Bits B A	Calling line identity indicator
0 0	Calling line identity not included
0 1	Calling line identity without DCC/DNIC included (national use only)
1 0	DCC/DNIC only included
1 1	Calling line identity with DCC/DNIC included
DC	CUG call indicator
0 0	Ordinary call
0 1	Spare
1 0	CUG call, outgoing access allowed
1 1	CUG call, outgoing access not allowed
E	BCUG call indicator
0	Ordinary call
1	BCUG call
F	Reserved for charging information indicator; coded 0
G	Reserved for an additional routing information indicator; coded 0
H	Field indicator of the second indicator octet
0	Second indicator not included
1	Second indicator included

3.3.2.12 *Second indicator octet*

This is an optional field that is included if indicated in bit H of the first indicator octet. The coding is shown in Table 8/X.61.

TABLE 8/X.61

Bit A	Redirected call indicator (national use only, see § 3.1.6)
0	Ordinary call
1	Redirected call
B	RPOA selection indicator (national use only, see § 3.1.6)
0	No RPOA code included
1	RPOA code included
C	Reverse charging request indicator
0	No reverse charging request
1	Reverse charging request
D	Called line identification request indicator
0	No called line identification requested
1	Called line identification requested
E F G	Spare
H	Reserved for field indicator for third indicator octet; coded 0

3.3.2.13 *Closed user group interlock code*

This is an optional field that is included only when indicated in bits CD in the first indicator octet. The format of the interlock code is in accordance with Figure 6/X.61.

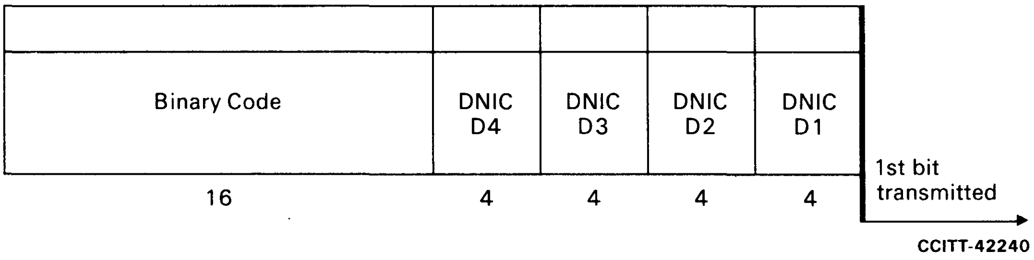


FIGURE 6/X.61
Format of interlock code

Each of the first four semi-octets contains a decimal digit, the value of which is expressed in pure binary representation, of the DNIC (or DCC plus one digit) of the coordinating Administration of the closed user group concerned (see Recommendation X.300). The 16-bit binary code is the code assigned to the closed user group concerned.

3.3.2.14 *Field length indicator for calling line identity*

This field is an optional field that is included only when the calling line identity is included. The coding is shown in Table 9/X.61.

TABLE 9/X.61

Bits B A	00	The following bits of the field length indicator contain the number of digits of the calling line identity (address and sub-address information)
	01	The following bits of the field length indicator contain the number of digits of the calling line identity (address information). Sub-address information follows in the calling line identity address extension field
	00	Reserved
	11	Reserved
H G F E D C	00	A code expressing in pure binary representation the number of addres signals (or address and sub-address signals) of the calling line identity The maximum number of digits is limited to 32 (see Note)

Note – See the Note to Table 5/X.61.

3.3.2.15 *Calling line identity*

This field is an optional field that is included only if indicated in bits AB of the first indicator octet. The field is divided into an even number of semi-octets. It contains a number of decimal digits of the national or international data number (address/address and sub-address) of the calling line or of the DNIC of the originating network. The coding of each digit, their order of transmission and the use of a filler code is as specified in § 3.3.2.8.

3.3.2.16 *Field length indicator for address extension for calling line identity*

This field is an optional field that is included if bit A of the field length indicator for the calling line identity is equal to 1.

This field contains a code expressing in pure binary representation the number of sub-address signals in the calling line identity.

The coding is shown in Table 6/X.61.

3.3.2.17 *Address extension for calling line identity*

This field is an optional field that is included if bit A of the field length indicator for the calling line identity is equal to 1.

This field is divided into an even number of semi-octets. The decimal value of each calling line identity sub-address digit is expressed in pure binary representation of an address signal. The digits are sent in descending order with the most significant digit first. In case of an odd number of sub-address signals a 4-bit 0000 filler code is included in the last semi-octet of the field.

3.3.2.18 *RPOA transit network identity*

This is an optional field that is included only if indicated in bit B in the second indicator octet. This field is divided into four semi-octets, each of which contains a decimal digit of the applicable DNIC. The coding and order of transmission of these digits is as specified in § 3.3.2.8.

3.3.3 *Call accepted message*

3.3.3.1 The format of the call accepted message is as shown in Figure 7/X.61.

The fields, subfields and codes are as follows:

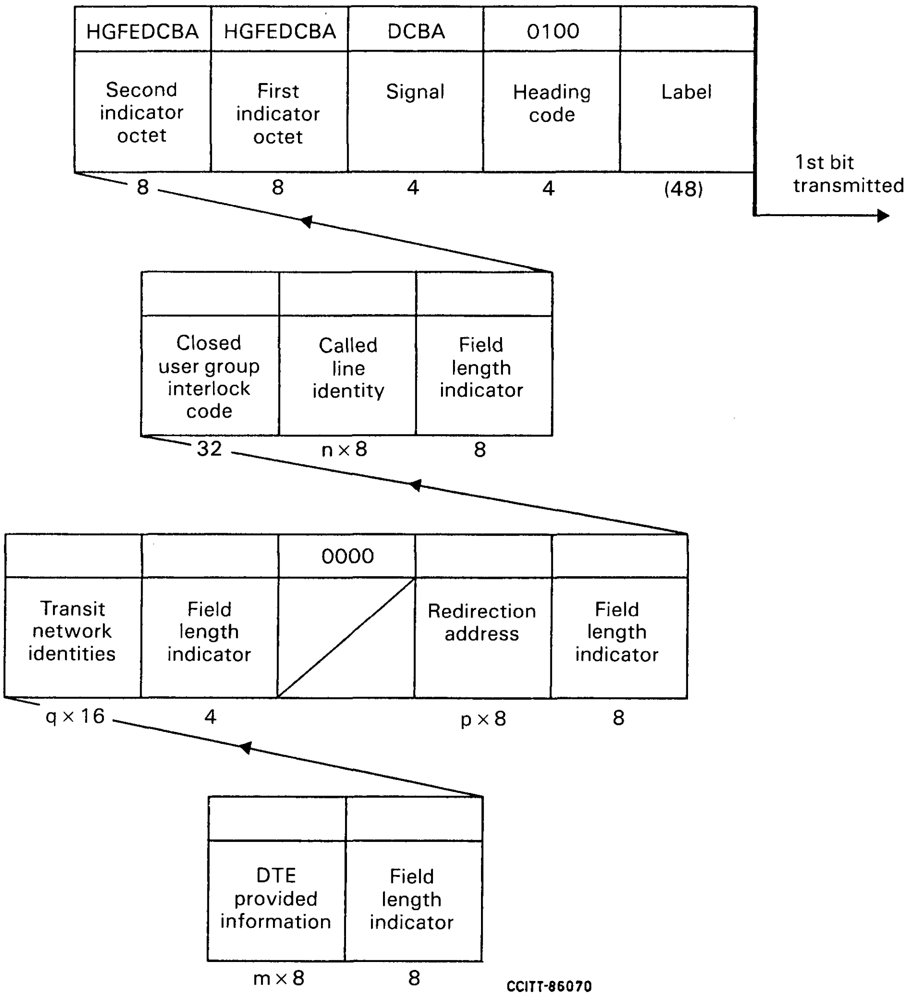


FIGURE 7/X.61

Call accepted message

3.3.3.2 *Label*

See § 3.2.

3.3.3.3 *Heading code H0*

See § 3.3.1.

3.3.3.4 *Signal*

The signal information is coded as shown in Table 10/X.61 (corresponding call progress signal digits, as applicable, are indicated within brackets).

TABLE 10/X.61

Bits DCBA	
0 0 0 0	Reserved for call progress signal code 00
0 0 0 1	Terminal called (01)
0 0 1 0	Redirected call (02)
0 0 1 1	Connected when free (03)
0 1 0 0 to 1 0 0 1	Spare
1 0 1 0	Call accepted
1 0 1 1	Transit through connect
1 1 0 0	Redirection request
1 1 0 1 to 1 1 1 1	Spare

3.3.3.5 *First indicator octet*

The coding is shown in Table 11/X.61.

TABLE 11/X.61

Bits B A	Called line identity indicator
0 0	Called line identity not included
0 1	Called line identity without DCC/DNIC included (national use only)
1 0	DCC/DNIC only included
1 1	Called line identity with DCC/DNIC included
C	Charge/no charge indicator (national use only, see § 3.1.6)
0	Normal charging
1	No charging
D	Calling line identity request indicator
0	Calling line identification not requested
1	Calling line identification requested
E	Originating network identification request indicator
0	Originating network identification not requested
1	Originating network identification requested
F	Transit network identity indicator
0	No transit network identity included
1	One or more transit network identity(ies) included
G	DTE provided information indicator
0	No DTE provided information
1	DTE provided information
H	Field indicator of the second indicator octet
0	Second indicator octet not included
1	Second indicator octet included

3.3.3.6 *Second indicator octet*

An optional field that is included if indicated in bit H of the first indicator octet. The coding is shown in Table 12/X.61.

TABLE 12/X.61

Bits B A	Redirection address indicator (national use only, see § 3.1.6)
0 0	Redirection address not included
0 1	Redirection address without DCC/DNIC included
1 0	Spare
1 1	Redirection address with DCC/DNIC
D C	CUG call indicator (national use only, see § 3.1.6) [a]
0 0	Ordinary call
0 1	Spare
1 0	CUG call, outgoing access allowed
1 1	CUG call, outgoing access not allowed
E,F,G	Spare
H	Reserved for field indicator of a third indicator octet; code 0

[a] Note that CUG information is only applicable to CUG calls that are redirected, see Recommendation X.300.

3.3.3.7 *Spare bits*

Included only when the called line identity is included.

3.3.3.8 *Field length indicator for called line identity*

An optional field that is included only when the called line identity is included. The coding is as specified in Table 9/X.61.

3.3.3.9 *Called line identity*

This field is an optional field that is included only if indicated in bits AB in the first indicator octet. This field is divided into an even number of semi-octets. It contains a number of decimal digits of the national or international data number (address/address and sub-address) of the called line or of the DNIC of the destination network. The coding of each digit, their order of transmission and the use of a filler code is as specified in § 3.3.2.8.

3.3.3.10 *Field length indicator for address extension for called line identity*

This field is an optional field that is included if bit A of the field length indicator for the called line identity is equal to 1.

This field contains a code expressing in pure binary representation the number of sub-address signals in the called line identity.

The coding is shown in Table 6/X.61.

3.3.3.11 *Address extension for called line identity*

This field is an optional field that is included if bit A of the field length indicator for the called line identity is equal to 1.

This field is divided into an even number of semi-octets. The decimal value of each called line identity sub-address digit is expressed in pure binary representation of a sub-address signal. The digits are sent in descending order with the most significant digit first. In case of an odd number of sub-address signals a 4-bit 0000 filler code is included in the last semi-octet of the field.

3.3.3.12 *Closed user group interlock code*

An optional field that is included only if indicated in bits CD of the second indicator octet. The format and code of the interlock code is as specified in § 3.3.2.13.

3.3.3.13 *Field length indicator for redirection address*

An optional field that is included only when a redirection address is included. The coding is as specified in Table 9/X.61.

3.3.3.14 *Redirection address*

An optional field that is included only if indicated in bits AB of the second indicator octet. This field is divided into an even number of semi-octets. It contains a number of decimal digits of the address/address and sub-address towards which the call has to be redirected. The coding of each digit, their order of transmission and the use of a filler code is as specified in § 3.3.2.8.

3.3.3.15 *Field length indicator for address extension for redirection address*

This field is an optional field that is included if bit A of the field length indicator for the redirection address is equal to 1.

This field contains a code expressing in pure binary representation the number of sub-address signals in the redirection address.

The coding is shown in Table 6/X.61.

3.3.3.16 *Address extension for redirection address*

This field is an optional field that is included if bit A of the field length indicator for the redirection address to equal to 1.

This field is divided into an even number of semi-octets. The decimal value of each redirection sub-address digit is expressed in a pure binary representation of a sub-address signal. The digits are sent in descending order with the most significant digit first. In case of an odd number of sub-address signals a 4-bit 0000 filler code is included in the last semi-octet of the field.

3.3.3.17 *Field length indicator*

An optional field that is included when at least one transit network identity is included. It is a code expressing in pure binary representation the number of transit network identities, i.e. the number of 16-bit subfields in the transit network identity field.

3.3.3.18 *Transit network identities*

An optional field that is included only when indicated in bit F of the first indicator octet. This field contains one or more 16-bit subfields, each divided into 4 semi-octets. The coding of each digit and their order of transmission is as specified in § 3.3.2.8.

3.3.3.19 *Field length indicator for DTE-provided information*

An optional field that is included when indicated by bit G in the first indicator octet. It is a code expressing a pure binary representation the number of characters of the DTE-provided information.

3.3.3.20 *DTE provided information*

An optional field that is included only when indicated by bit G of the first indicator octet. This field contains the characters of the DTE-provided information.

3.3.4 *Call rejected message*

3.3.4.1 The format of the call rejected message is as shown in Figure 8/X.61.

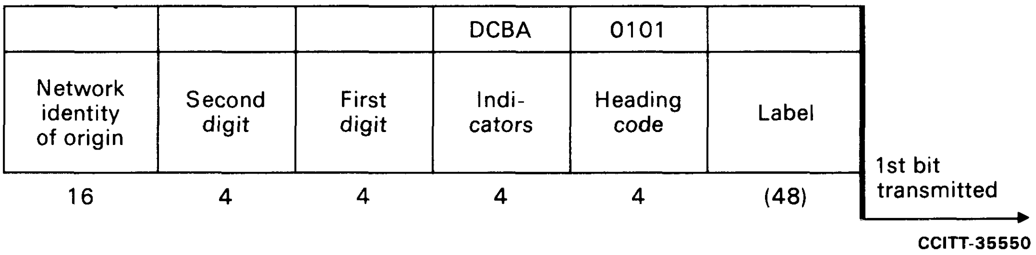


FIGURE 8/X.61
Call rejected message

The fields, subfields and codes are as follows:

3.3.4.2 *Label*

See § 3.2.

3.3.4.3 *Heading code*

See § 3.3.1.

3.3.4.4 *Indicators*

The coding is shown in Table 13/X.61.

TABLE 13/X.61

Bit A	Reserved for a field indicator of a possible optional field for extended call progress information
B	Field indicator of network identity of origin
0	Network identity of origin not included
1	Network identity of origin included
C	DTE provided information indication
0	No DTE provided information
1	DTE provided information
D	Reserved for a possible indication that immediate clearing should not take place: code 0.

3.3.4.5 *First and second digit*

Each of the two fields contains a decimal digit expressed in pure binary representation. The combination of the two decimal digits expresses the signal indicating the cause for call rejection. The values of the decimal digits are as shown in Table 14/X.61. This coding should be consistent with the corresponding coding of DTE/DCE interface call progress signals, see Recommendation X.21.

Note 1 – An interexchange signal not corresponding to a specific DTE/DCE interface call progress signal will, as required, be coded by over-decadic combination of the two digits.

Note 2 – Some of the call progress signal code groups specified in Recommendation X.21 correspond to other message types than the call rejected message.

Note 3 – The reverse charge acceptance not subscribed signal has not yet been assigned a code.

TABLE 14/X.61

Digits 20	Network failure
21	Number busy
41	Access barred
42	Changed number
43	Not obtainable
44	Out of order
45	Controlled not ready
46	Uncontrolled not ready
47	DCE power off
48	Invalid facility request [a]
49	Network fault in local loop
51	Call information service
52	Incompatible user class of service
61	Network congestion
71	Degraded service
72	RPOA out of order

[a] Applicable to the facility registration/cancellation request rejected message only, see § 3.4.4.4.

3.3.4.6 *Field length indicator for DTE-provided information*

An optional field that is included when indicated by bit C in the indicator. It is a code expressing in pure binary representation the number of characters of the DTE-provided information.

3.3.4.7 *DTE-provided information*

An optional field that is included only when indicated by bit C in the indicator. This field contains the characters of the DTE-provided information.

3.3.5 *Clear message*

3.3.5.1 The format of the clear message is as shown in Figure 9/X.61.

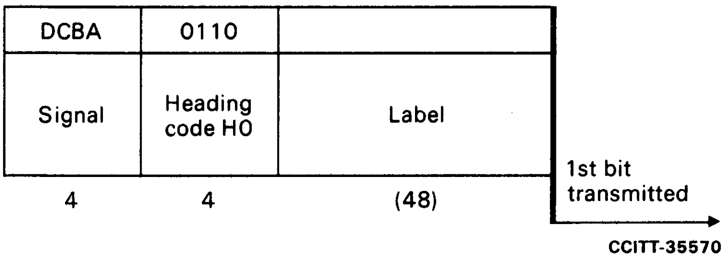


FIGURE 9/X.61
Clear message

The fields and codes are as follows:

3.3.5.2 *Label*

See § 3.2.

3.3.5.3 *Heading code H0*

See § 3.3.1.

3.3.5.4 *Signal*

The coding is shown in Table 15/X.61.

TABLE 15/X.61

Bits DCBA	
0 0 0 0	Spare
0 0 0 1	Spare
0 0 1 0	Circuit released (forward)
0 0 1 1	Circuit released acknowledgement (forward)
0 1 0 0 to 1 0 0 1	Spare
1 0 1 0	Circuit released (backward)
1 0 1 1	Circuit released acknowledgement (backward))
1 1 0 0 to 1 1 1 1	Spare

3.3.6 *Circuit state message*

3.3.6.1 The format of the circuit state message is as shown in Figure 10/X.61.

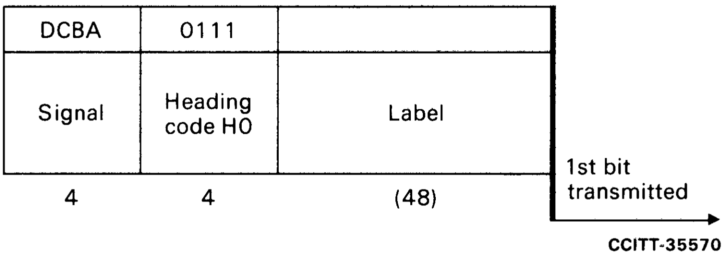


FIGURE 10/X.61
Circuit state message

The fields and codes are as follows:

3.3.6.2 *Label*

See § 3.2.

3.3.6.3 *Heading code H0*

See § 3.3.1.

3.3.6.4 *Signal*

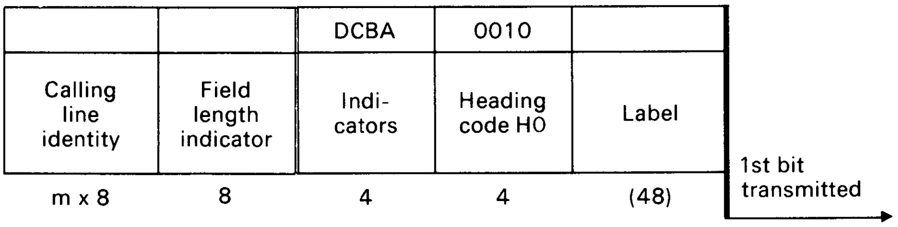
The coding is shown in Table 16/X.61.

TABLE 16/X.61

Bits DCBA	
0 0 0 0	Spare
0 0 0 1	Spare
0 0 1 0	Blocking
0 0 1 1	Blocking acknowledgement
0 1 0 0	Unblocking
0 1 0 1	Unblocking acknowledgement
0 1 1 0	Spare
0 1 1 1	Reset circuit
1 0 0 0 to 1 1 1 1	Spare

3.3.7 *Calling line identity message*

3.3.7.1 The format of the calling line identity message is as shown in Figure 11/X.61.



CCITT-86080

FIGURE 11/X.61
Calling line identity message

The fields, subfields and codes are as follows:

3.3.7.2 *Label*

See § 3.2.

3.3.7.3 *Heading code H0*

See § 3.3.1.

3.3.7.4 *Indicators*

The coding is shown in Table 17/X.61.

TABLE 17/X.61

Bits	B A	Calling line identity indicator
	0 0	Calling line identity not included [a]
	0 1	Calling line identity without DCC/DNIC included (national use only)
	1 0	DCC/DNIC only included
	1 1	Calling line identity with DCC/DNIC included
	C, D	Spare

[a] As presently defined, this message always includes the calling line identity.

3.3.7.5 *Field length indicator for calling line identity*

This field is an optional[1] field that is included only when the calling line identity is included. The coding is shown in Table 9/X.61.

3.3.7.6 *Calling line identity*

This field is an optional field that is included only if indicated in bits AB of the indicator field.

See also § 3.3.2.15.

3.3.7.7 *Field length indicator for address extension for calling line identity*

See § 3.3.2.16.

3.3.7.8 *Address extension for calling line identity*

See § 3.3.2.17.

3.4 *Formats and codes for facility registration and cancellation messages*

3.4.1 *Heading*

The different heading codes (H0) for the facility registration and cancellation messages are shown in Table 18/X.61.

TABLE 18/X.61

0000	Spare
0001	Facility registration/cancellation request message
0010	Facility registration/cancellation accepted messages
0011	Facility registration/cancellation rejected messages
0100 to 1111	Spare

3.4.2 *Facility registration/cancellation request message*

3.4.2.1 The format of the facility registration/cancellation request message is as shown in Figure 12/X.61.

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[1] As presently defined, this message always includes the calling line identity.

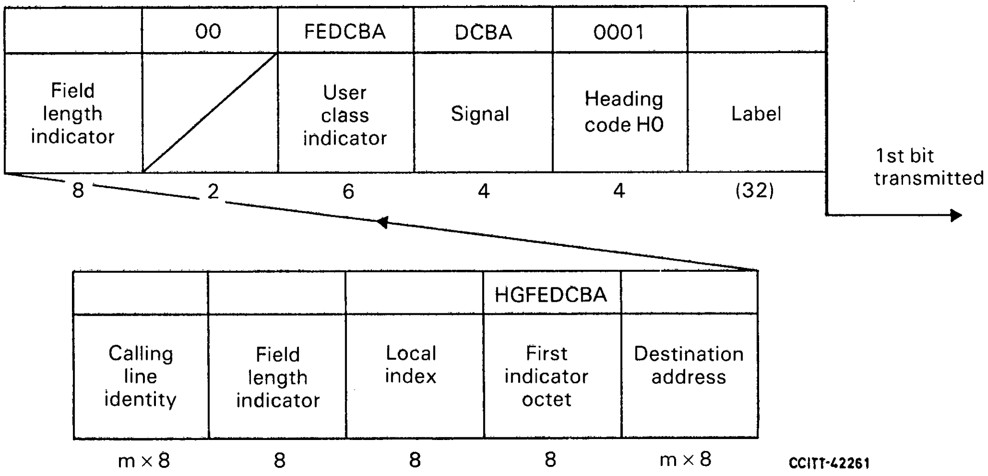


FIGURE 12/X.61

Facility registration/cancellation request message

The fields, subfields and codes are as follows:

3.4.2.2 *Label*

See § 3.2.

3.4.2.3 *Heading code H0*

See § 3.4.1.

3.4.2.4 *Signal*

The coding is shown in Table 19/X.61.

TABLE 19/X.61

Bits DCBA	
0 0 0 0	Spare
0 0 0 1	Registration request
0 0 1 0	Cancellation request
0 0 1 1 to 1 1 1 1	Spare

3.4.2.5 *User class indicator*

See § 3.3.2.5.

3.4.2.6 *Spare bits*

3.4.2.7 *Field length indicator for destination address*

A code expressing in pure binary representation the number of address signals in the destination address.

3.4.2.8 *Destination address*

This field is divided into an even number of semi-octets. The decimal value of each destination address digit is expressed in a pure binary representation of an address signal. The digits are sent in descending order with the most significant digit first. In case of an odd number of address signals a 4-bit 0000 filler code is included in the last semi-octet of the field.

3.4.2.9 *First indicator octet*

The coding is shown in Table 20/X.61.

TABLE 20/X.61

Bits	B A	Calling line identity indicator
	0 0	Calling line identity not included
	0 1	Calling line identity without DCC/DNIC included (national use only)
	1 0	DCC/DNIC only included
	1 1	Calling line identity with DCC/DNIC included
	C	BCUG indicator
	0	No BCUG registration/cancellation
	1	BCUG registration/cancellation
	D	Field indicator for local index
	0	Local index not included
	1	Local index included
	E,F,G	Spare
	H	Reserve for field indicator for second indicator octet; coded 0

3.4.2.10 *Local index*

This is an optional field that is included only if indicated in bit D in the first indicator octet. In the case of registration request it is the local index assigned by the user requesting registration. In the case of cancellation request it is the local index assigned by the remote user in the BCUG being cancelled.

3.4.2.11 *Field length indicator for the calling line identity*

This field is an optional field that is included only when the calling line identity is included. It is a code expressing in pure binary representation the number of address signals in the calling line identity included.

3.4.2.12 *Calling line identity*

This field is an optional field that is included only if indicated in bits AB in the first indicator octet. The code is as specified in § 3.3.2.15.

3.4.3 *Facility registration/cancellation request accepted message*

3.4.3.1 The format of the facility registration/cancellation request message is as shown in Figure 13/X.61.

	HGFEDCBA			DCBA	0010	
Local index	First indicator octet	Destination address	Field length indicator	Signal	Heading code H0	Label
8	8	n×8	8	4	4	(32)

1st bit transmitted →

CCITT-42271

FIGURE 13/X.61

Facility registration/cancellation request accepted message

The fields, subfields and codes are as follows:

3.4.3.2 *Label*

See § 3.2.

3.4.3.3 *Heading code H0*

See § 3.4.1.

3.4.3.4 *Signal*

The coding is shown in Table 21/X.61.

TABLE 21/X.61

Bits 0000 0001 0010 0011	Spare Registration completion Registration accepted Cancellation completed
0100 to 1111	Spare

3.4.3.5 *Field length indicator for the destination address*

See § 3.4.2.7.

3.4.3.6 *Destination address*

See § 3.4.2.8.

3.4.3.7 *First indicator octet*

The coding is shown in Table 22/X.61.

TABLE 22/X.61

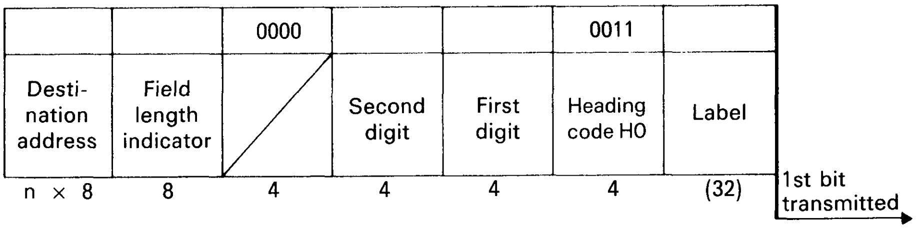
Bits A	Spare
B	BCUG indicator
0	No BCUG registration/cancellation
1	BCUG registration/cancellation
C	Field indicator for local index
0	Local index not included
1	Local index included
D-G	Spare
H	Reserved for field indicator for second indicator octet; coded 0

3.4.3.8 *Local index*

This is an optional field that is included only if indicated in bit C of the first indicator octet. It contains the local index of the user at the exchange from which the message is originated.

3.4.4 *Facility registration/cancellation request rejected message*

3.4.4.1 The format of the facility registration/cancellation request rejected message is as shown in Figure 14/X.61.



CCITT-86090

FIGURE 14/X.61

Facility registration/cancellation request rejected message

The fields, subfields and codes are as follows:

3.4.4.2 *Label field*

See § 3.2.

3.4.4.3 *Heading code*

See § 3.4.1.

3.4.4.4 *First and second digit*

Each of the two fields contains a decimal digit expressed in pure binary representation. The combination of the two decimal digits expresses the signal indicating the cause for facility registration/cancellation request rejection. The values of the decimal digits are as specified in § 3.3.4.5 for the relevant signals.

3.4.4.5 *Spare bits*

3.4.4.6 *Field length indicator*

See § 3.4.2.7.

3.4.4.7 *Destination address*

See § 3.4.2.8.

3.5 *Data channel signalling conditions*

The following conditions are those appearing in the interexchange data channels that in certain phases of a call have to be transmitted and/or detected in an exchange.

The coding of the presently specified data channel conditions is determined by the codes of the corresponding DTE/DCE interface states consistent with Recommendation X.21.

The data channel signalling conditions will be coded as follows (data bits/status bit):

a) *trunk free* condition: 0 . . . 0/0 (see Notes 1 and 3),

b) *trunk seized* condition: 1 . . . 1/0,

c) *call accepted* condition: 1 . . . 1/1,

d) *call request* condition: 0 . . . 0/0.

The above codes imply that the code 0 of the status bit on an interexchange data channel results in the OFF condition at the DTE/DCE interface consistent with Recommendation X.21, and that the code 1 results in the ON condition.

Note 1 – The code to be used for the *trunk free* condition in networks that cannot support bit sequence independence is for further study.

Note 2 – The implications for the data channel conditions, and their codes, of potential ISDN applications and/or of possible new DTE/DCE interfaces are a subject for further study.

Note 3 – As a national option, the data bits in the even positions of each envelope may be permanently inverted both at the transmitting and at the receiving ends of the interexchange data channels. Such inversion implies that the above specified codes (as well as information transferred during the data phase) will appear on the data channel correspondingly inverted. This option enables the *trunk free* condition in the case of the 8-bit envelope to be the same as the idle pattern for telephone channels as generated by a digital exchange complying with the standards related to the A-law.

4 Basic call control and signalling procedures

4.1 *General*

4.1.1 The call control procedures specified in § 4 are based on the requirements of the circuit-switched data transmission service as presently defined in the Series X Recommendations. In particular, the requirements specified for exchange through-connection and data channel conditions are dependent on the characteristics of the present DTE/DCE interfaces for the circuit-switched service. Also, the implications of ISDN applications of common channel signalling for circuit-switched data transmission services have not yet been fully determined.

4.1.2 The basic call control procedure is divided into two phases: call set-up and call clear-down, which are separated from one another by the data phase. A combination of messages on the signalling link and exchanges of conditions in the interexchange data channels are used to establish and terminate the different phases of the call.

4.1.3 The procedures specified in this § 4 in principle only relate to basic calls, i.e. calls not involving any user facilities. The additional requirements to be met in the cases of calls involving user facilities and network utilities are specified in § 5 and Recommendation X.300.

4.1.4 The interexchange data channel signalling conditions and the connect-through procedures specified ensure that the conditions in the network are compatible with the conditions and procedures for the present DTE/DCE interfaces.

4.1.5 Link-by-link transfer of signalling information assembled in messages is used and address information is signalled with all the elements of an address contained in one message. The network numbering is specified in Recommendation X.121. The network routing to apply is defined in Recommendation X.110.

4.1.6 Requirements of interworking with decentralized signalling are specified in Recommendation X.80.

4.2 *Overall call set-up and clear-down procedures*

The overall call set-up and clear-down procedures are outlined hereunder. The detailed signalling and switching procedures are covered in §§ 4.3 and 4.4 respectively. These procedures are illustrated in Tables 23/X.61 and 24/X.61.

TABLE 23/X.61

Call set-up and clear-down procedure for successful basic call

Originating exchange	Interexchange data circuit		Interexchange signalling link		Transit exchange	Interexchange signalling link		Interexchange data circuit		Destination exchange
	⟶	⟵	⟶	⟵		⟶	⟵	⟶	⟵	
Trunk free condition	TF	TF			Trunk free condition			TF	TF	Trunk free condition
Selection information received	|	|						|	|	
Determine routing	|	|						|	|	
Free circuit seized	|	|						|	|	
Trunk seized sent	TS	|						|	|	
Address message sent	|	|	AM					|	|	
	|	|			Address message received			|	|	
	|	|			Routing determined			|	|	
	|	|			Free circuit seized			|	|	
	|	|			Data path connected			TS	|	
	|	|			Address message sent	AM		|	|	
	|	|						|	|	Address message received
	|	|						|	|	Called user determined
	|	|						|	|	State and validity checked [a]
	|	|						|	|	User called
	|	|						|	CA	Call accepted received
	|	|						|	|	Data path connected
	|	|					CAM	|	|	Call accepted message sent [a]
	|	|			Call accepted message received			|	|	
	|	|		CAM	Call accepted message sent			|	|	

[a] Alternatively, the call accepted message can be sent before called user response.

Data circuit conditions

TF	Trunk free condition	RD	Ready for data condition
TS	Trunk seized condition	CR	Clear request condition
CA	Call accepted condition	| |	Contiguous transmission of previous condition

Messages on signalling link

AM	Address message
CAM	Call accepted message
CRM	Call rejected message
CLM	Clear message

TABLE 23/X.61 *(continued)*

Originating exchange	Interexchange data circuit		Interexchange signalling link		Transit exchange	Interexchange signalling link		Interexchange data circuit		Destination exchange
	⟶	⟵	⟶	⟵		⟶	⟵	⟶	⟵	
Call accepted message received										
Call accepted condition received		CA								
Data path connected										
Charging started										
Ready for data received by calling user	RD	RD						RD	RD	Ready for data received by called user
Data phase	Data	Data			Data phase			Data	Data	Data phase
Clear request received from calling user	CR							CR		Clear request received from called user [b]
Data path released					Clear message received					
Clear message sent	TF		CLM		Data path released					Clear request received
Trunk free sent		CR		CLM	Clear message sent on both sides	CLM			CR	from called user
Clear confirmation					Free trunk sent on		CLM			Data path released
sent to calling user					both sides			TF		Clear message sent
		TF			(Incoming) circuit free					Free trunk sent
									TF	Clear confirmation
										sent to called user
					Clear message received					Clear message received
					(Outgoing) circuit free					(Incoming) circuit free

[b] Optionally, remote user clear request condition may also be used as a release criterion.

TABLE 24/X.61

Call set-up and clear-down procedure for basic call with number busy

Originating exchange	Interexchange data circuit		Interexchange signalling link		Transit exchange	Interexchange signalling link		Interexchange data circuit		Destination exchange
	⟶	⟵	⟶	⟵		⟶	⟵	⟶	⟵	
Trunk free condition	TF	TF			Trunk free condition			TF	TF	Trunk free condition
Call set up as for successful call	TS		AM							
					Call set up as for successful call	AM		TS		
							CRM			Address message received Called user busy Call released Call rejected message sent
				CRM	Call rejected message received Call released Call rejected message sent Clear message sent Trunk free sent (Outgoing) circuit free	CLM		TF		Clear message sent (Incoming) circuit free
Call rejected message received Call released Calling user cleared Clear message sent Trunk free sent (Outgoing) circuit free	TF	TF	CLM							

Note – For legends Table 23/X.61.

4.2.1 *Call set-up*

4.2.1.1 When the originating exchange has received the complete selection information from the calling user and has determined that the call is to be routed to another exchange, it seizes a free interexchange data circuit and sends an address message on the signalling link. The address message in principle contains all of the information that is required to route and connect the call to the called user and may also include the calling line identity and other information related to any user facilities and network utilities that are required.

4.2.1.2 A transit exchange, on receipt of an address message will analyse the destination address and the other routing information to determine the routing of the call. The transit exchange then seizes a free interexchange data circuit and sends an address message to the next exchange and connects through the data path. In the case of congestion at the transit exchange it may select an alternative route, or send a call rejected message to the preceding exchange indicating congestion and clearing of the call.

4.2.1.3 Upon receipt of an address message the destination exchange will analyse the destination address to determine to which user the call should be connected. It will also check the called user's line condition and perform various checks to verify whether or not the connection is allowed. These checks will include correspondance of user class and any checks associated with user facilities. In the case where the connection is allowed, the destination exchange will call the called user in accordance with the applicable DTE/DCE interface protocol. The called user will normally respond with a *call accepted* (or corresponding) signal. If the call cannot be completed due to, for instance, the called user being busy, a call rejected message indicating this is sent to the preceding exchange and clearing takes place.

4.2.1.4 At connection of the call the destination exchange normally sends a call accepted message to the preceding exchange. Depending on the circumstances the call accepted message may include information related to specific network conditions and any user facilities or network utilities involved (see § 5).

4.2.1.5 Upon receipt of a call accepted message a transit exchange sends the corresponding call accepted message to the preceding exchange. If it is an international transit exchange the applicable transit network identity (see § 5.11) will be included in the call accepted message.

4.2.1.6 When the originating exchange receives a call accepted message indicating that the call can be completed, it prepares to connect through the data path. The originating exchange then connects through and starts charging, as applicable. In certain cases, e.g. when certain user facilities are involved, data path through-connection is preceded by the sending of a call progress signal or other information to the calling user.

4.2.1.7 In the cases when the call cannot be completed, the originating exchange will send an appropriate call progress signal, indicating the cause of call rejection, to the calling user and clear the call.

4.2.2 *Call clear-down*

4.2.2.1 Normally the clearing action initiated by a user will propagate rapidly along the connection and initiate release at each exchange involved. When both users clear at approximately the same time, clearing will propagate from both ends.

4.2.2.2 When detecting a valid *clearing* signal from the local user, the originating or destination exchange will release the connection and send a clear message to the adjoining exchange. The *clearing* signals originated by a user will pass through the local exchange and will appear on the interexchange data circuits and at the distant local exchange until such time as the *clearing* signals are acted upon and the connection is released. The actions at the exchange releasing the connection, including the condition sent on the interexchange data circuits when released, are therefore specified to be consistent with the clearing procedures of the DTE/DCE interfaces.

4.2.2.3 Clearing may also be initiated by a data exchange during call set-up when the call cannot be connected due to a user or network condition.

4.2.2.4 After release of the connection the clearing procedure is completed for each interexchange data circuit individually. A data circuit is assumed to be free for a new call at an exchange when both the forward and backward clearing indications relating to that data circuit have been sent and received.

4.3 *Normal switching procedures*

4.3.1 *General*

4.3.1.1 The switching procedures specified hereunder define the actions to be performed at call set-up and clear-down and the sequencing of these actions in relation to the handling of signalling messages and data channel signalling conditions. The specified connect-through and release actions and the coding of the data channel signalling conditions (see § 3.4) are based on the requirement for consistency with the present DTE/DCE interface protocol for the circuit-switched service.

Note – The implications for the procedure specified hereunder of possible new DTE/DCE interfaces for the circuit-switched service are for further study.

4.3.1.2 The *trunk free* condition is sent on the free interexchange data channels. Also, at release of an interexchange data circuit, the *trunk free* condition is immediately applied to its transmit channel. Both directions of transmission must be through-connected at (approximately) the same time.

4.3.1.3 The signalling information content in the signalling messages is specified in § 4.4. The time-out supervisions to be performed in relation to interexchange signalling and the procedures to be followed in abnormal conditions are specified in § 4.5.

4.3.2 *Call set-up*

4.3.2.1 *Originating exchange*

The call set-up actions are illustrated by means of a Specification and Description Language (SDL) diagram (see Recommendation Z.101) in Figure 15/X.61.

After having seized an interexchange data circuit, the originating exchange applies the *trunk seized* condition to the forward data channel. The sending of the address message and the application of the *trunk seized* condition may be performed in parallel as independent actions. The originating exchange then waits for the reception of a call accepted message or call rejected message.

Upon receipt of a call accepted message the originating exchange prepares to connect through the data path. In the case where user facilities apply, call progress signals may be sent to the calling user as applicable. The originating exchange then monitors the backward interexchange data channel for the presence of the *call accepted* condition. When this condition is detected, indicating that all succeeding exchanges have connected through, the originating exchange connects through and initiates charging where applicable.

In the cases when a call rejected message is received, the appropriate call progress signal is sent to the calling user and clearing takes place. Receipt of a call rejected message may also occur after receipt of a previous call accepted message.

Note – In the case of Recommendation X.20 concerning start-stop terminals, the originating exchange monitors the backward interexchange data channel for the presence of the call accepted condition (1, ON). When this condition is detected, the through-connection signal character (ACK) is sent by the originating exchange to the calling and called users. Then, the originating exchange connects through and initiates charging.

4.3.2.2 *Transit exchange*

The call set-up actions are illustrated by means of an SDL diagram in Figure 16/X.61.

Having seized a free interexchange data circuit and sent an address message to the succeeding exchange, the transit exchange connects through the data path.

If a call accepted message is received from the succeeding exchange the transit exchange sends a corresponding message to the preceding exchange. If a call rejected message is received, the corresponding message is sent and clearing takes place. Receipt of a call rejected message may also occur subsequent to the receipt of a previous call accepted message.

4.3.2.3 *Destination exchange*

The call set-up actions are illustrated by means of an SDL diagram in Figure 17/X.61.

In the case where the call is to a user that is indicated as ready to receive a call, the destination exchange sends the *incoming call* (or corresponding) signal to the user. The destination exchange usually connects through the data path when:

- the *call accepted* (or corresponding) signal has been received from the user, and
- the transmission to the called user of any additional information, e.g. related to user facilities, has been completed in accordance with the applicable DTE/DCE interface protocol.

It is necessary to ensure that the *trunk seized* condition is present in the receive data channel of the interexchange data circuit before through-connection for consistency with the called user DTE/DCE interface protocol when this is in accordance with the present standards, e.g. Recommendation X.21, for the circuit-switched service, cf. Note to § 4.3.3.2.

In the case where the call can be connected a call accepted message is sent to the preceding exchange. This message may be sent either before or after the *call accepted* (or corresponding) signal has been received from the called user. Waiting for the receipt of the *call accepted* or corresponding signal has the advantage that sending of the call accepted message is based on a positive indication that the call has been accepted by the called user. Sending the call accepted message earlier, e.g. in conjunction with the sending of the *incoming call* (or corresponding) signal to the user, has the advantage that the call set-up time is reduced in the normal condition.

In the case where certain user facilities apply, see § 5 and Recommendation X.300, through-connection normally takes place in conjunction with the sending of a second call accepted message.

In the case where the call cannot be connected and completed, a call rejected message is sent to the preceding exchange and clearing takes place.

4.3.3 *Call clear-down*

4.3.3.1 *Originating exchange*

The clearing actions are illustrated by means of SDL diagrams in Figures 15/X.61 and 18/X.61. Release of the connection is initiated by one of the following criteria (see also the Note to § 4.3.3.2):

a) detection of a *clear request* condition from the calling user,

b) optionally, detection of a *clear request* condition from the called user on the backward channel of the interexchange data circuit,

c) receipt of a call rejected message, or

d) receipt of a backward clear message.

After release of the connection a clear message is sent to the succeeding exchange and the calling user is cleared in accordance with the applicable DTE/DCE interface protocol.

4.3.3.2 *Transit exchange*

The clearing actions are illustrated by means of SDL diagrams in Figures 16/X.61 and 18/X.61. Release of the connection is initiated by one of the following criteria:

a) failure to complete call set-up,

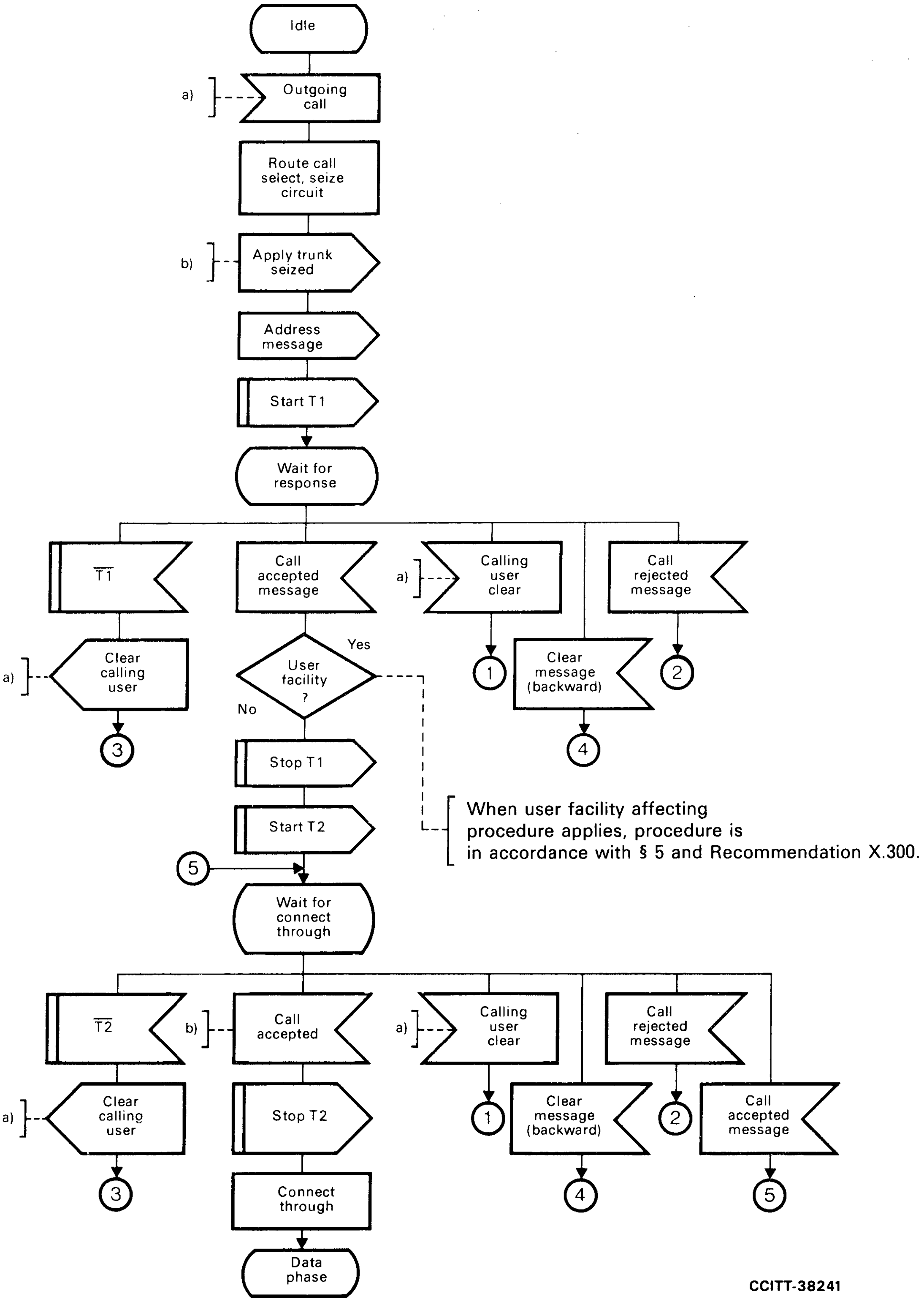
b) receipt of a call rejected message, or

c) receipt of a forward or backward clear message.

After release of the connection:

- a call rejected message is sent to the preceding exchange in the cases a) and b),
- a clear message is sent to the preceding exchange in the case c),
- a clear message is sent to the succeeding exchange in the cases b) or c).

Note – In the case where satellite data circuits are served by a terrestrial common channel signalling network, there is a probability that a clear message initiated by user clearing may arrive at the other end of the satellite circuit before all user data transmitted immediately before clearing has passed that end. Therefore, the action initiated by receipt of a clear message relating to a satellite circuit must be delayed by an appropriate time interval unless other release criteria have been met. The necessary arrangements to cater for such a situation are for further study.



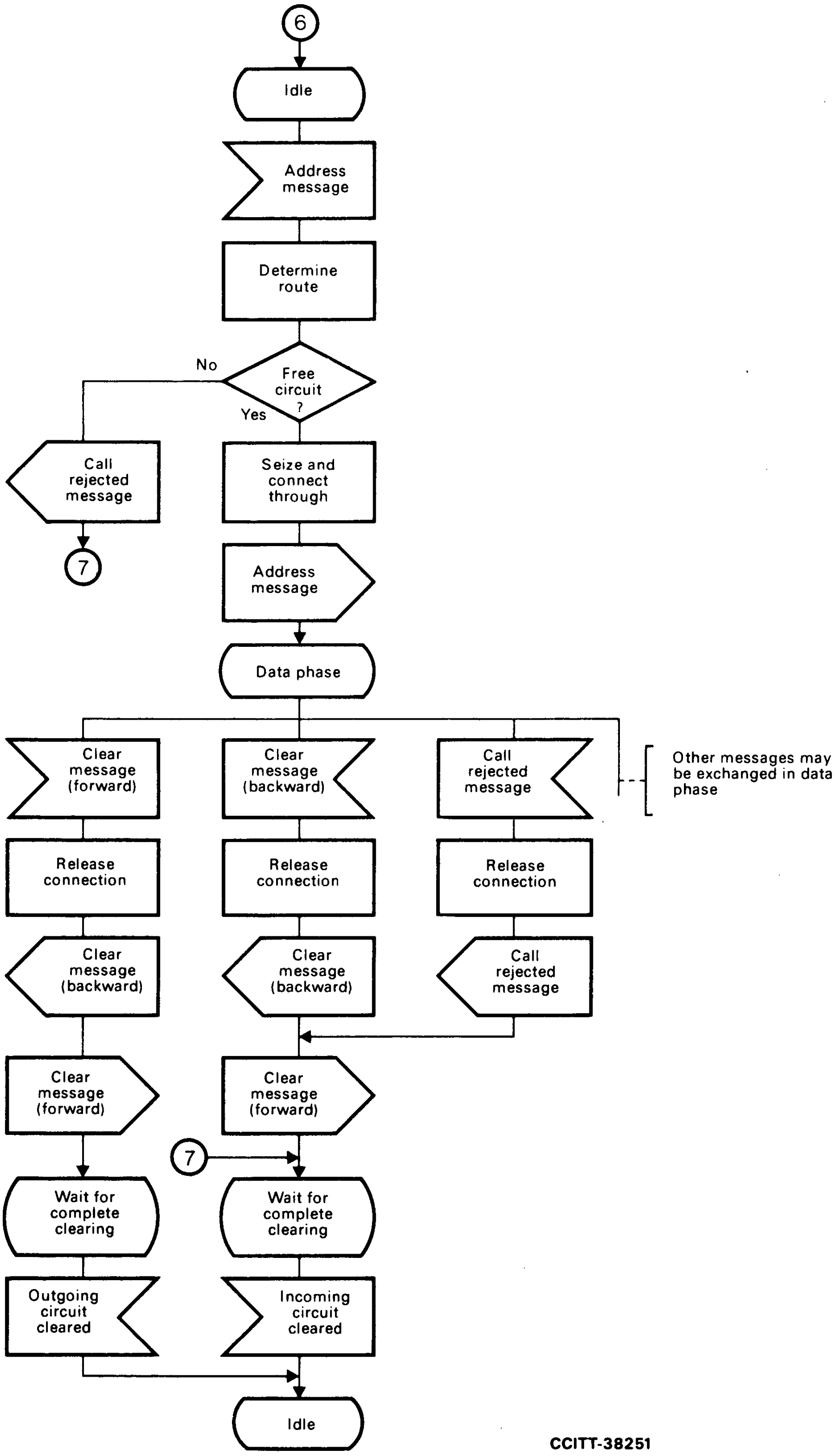
[a] In accordance with the applicable DTE/DCE interface protocol.

[b] In interexchange data channel.

Note – Connectors ① to ④ go to Figure 18/X.61 which also shows clearing in data phase. Time-outs T1 and T2 as in § 4.5.3.1.

FIGURE 15/X.61

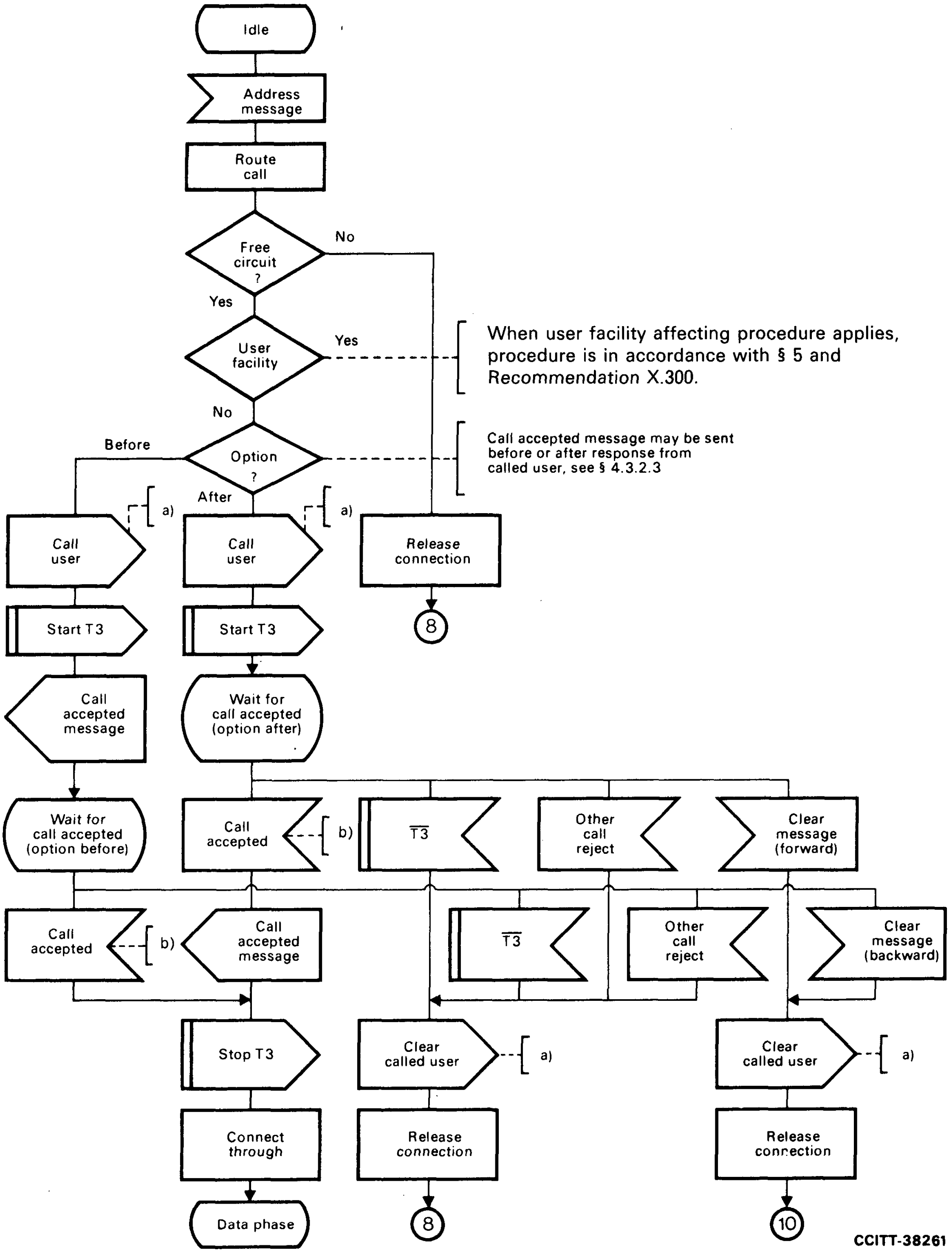
Call set-up at originating exchange



Note – Detailed clearing procedures are shown in Figure 18/X.61.

FIGURE 16/X.61

Call set-up at transit exchange

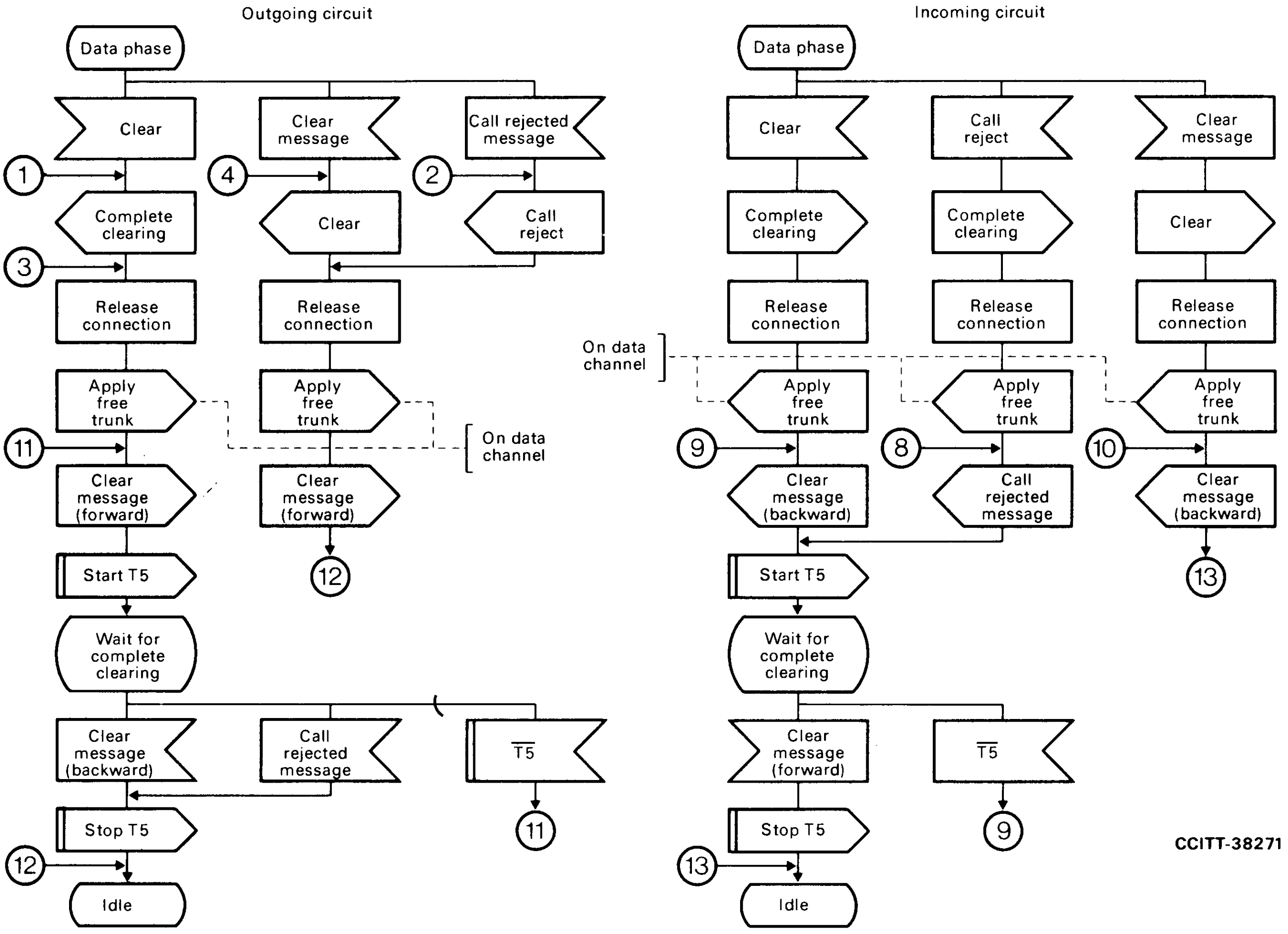


[a] In accordance with the applicable DTE/DCE interface protocol.
[b] Call accepted state, or corresponding, from called user.

Note – Connectors ⑧ to ⑩ go to Figure 18/X.61, which also shows clearing in data phase. Time-out T3 as in § 4.5.3.2.

FIGURE 17/X.61

Call set-up at destination exchange



Note – Connectors ① to ④ from Figure 15/X.61 and ⑧ to ⑩ from Figure 17/X.61. Time-out T5 and delayed maintenance alarm in case of ineffective release as in § 4.5.3.4.

FIGURE 18/X.61

Clearing of interexchange data circuits

4.3.3.3 *Destination exchange*

The clearing actions are illustrated by means of SDL diagrams in Figures 17/X.61 and 18/X.61. Release of the connection is initiated by one of the following criteria (see also the Note to § 4.3.3.2):

a) failure to complete call set-up,

b) detection of *clear request* condition from the called user,

c) optionally, detection of a *clear request* condition from the calling user on the forward channel of the interexchange data circuit, or

d) receipt of a forward clear message.

After release of the connection:

- a call rejected message is sent to the preceding exchange in the case a),
- a clear message is sent to the preceding exchange in the cases b), c) or d),
- the called user is cleared in accordance with the applicable DTE/DCE interface protocol in the cases b), c) and d).

4.4 *Detailed signalling procedures under normal conditions*

The signalling information content of the different signalling message types is specified in § 3. The general function of the different signalling information components is defined in § 2. In the following, requirements are detailed for the signalling information components that are involved in normal basic calls. The requirements for sending the messages and for the principal actions at their reception are specified in § 4.2.

4.4.1 *Address message*

In the international network the *destination* address will be the complete international data number of the called user in accordance with Recommendation X.121, i.e. including the DCC/DNIC.

The *DCC/DNIC* indicator is provided to cater for discrimination in national networks between cases where the destination address does or does not include the DCC/DNIC component. Depending on the national numbering and routing plans, this indicator may be necessary or useful for interpretation of the destination address and to determine the routing of the call. It may, for example, be used to identify an outgoing international call.

The *national/international call* indicator is provided to cater for discrimination in national networks between national and international calls. Depending on the implementation of user facilities and network functions that imply different handling of national and international calls, this indication may be a necessary or useful means for such differentiation. It may, for example, be used to determine whether a called line identity sent from the destination exchange should include the DNIC.

The *user class* indicator provides information about the user class of the calling user. At a transit exchange the user class information is used for selection of an appropriate type of data circuit. At the destination exchange the user class information is used to verify that the calling and the called users have compatible user classes of service.

The *alternative routing* indicator is set in the case where alternative routing is performed. It may be used to prevent the call being subjected to alternative routing more than once.

Provision is made for transfer in a national network of the calling line identity as part of the basic procedures, e.g. for call management purposes.

An address message may also contain additional signalling information relating to user facilities and network utilities the procedures of which are covered in § 5 and Recommendation X.300.

4.4.2 *Call accepted message*

The *call accepted* signal is used at the time of connection of the call to a called user having automatic answer. In some cases when a user facility applies, or when the called DTE operates with manual answer, another signal is used in the first call accepted message. In such cases the *call accepted* signal is used in a second call accepted message when the call is completed by receipt of a *call accepted* (or corresponding) signal from the called user. At receipt of the *call accepted* signal the originating exchange prepares normal connect through.

In some situations of interworking with decentralized signalling the *transit through-connect* signal will be used as specified in Recommendation X.80. It will normally be followed by a second call accepted message. At receipt of the *transit through-connect* signal the originating exchange waits for a further call accepted message.

In some cases when the called user has a user facility, an alternative signal to the *call accepted* signal and/or additional signalling information will be used as covered in § 5 and Recommendation X.300. Depending on the facility involved this may result in an alternative connect through procedure at the destination exchange.

Provision is made for transfer in a national network of the called line identity as part of the basic procedures, e.g. for call management purposes.

4.4.3 *Call rejected message*

This message contains a signal indicating the cause for call rejection. The signal to be used in a particular case and the applicable translation at the originating exchange to a *DTE/DCE* call progress signal is as defined in § 2.3.

Receipt of a call rejected message will initiate clearing (see § 4.3.2). In international operation the network identity of the exchange originating the signal will be included in the message.

4.4.4 *Clear message*

A clear message containing a *circuit released* signal will be sent after release of the connection in the case when a clear message or a call rejected message has not been received for the same call and circuit. In the case where a clear message or a call rejected message has been received for the same call and circuit, the clear message sent will contain a *circuit released acknowledgement* signal. The signal sent will in both cases be coded as forward and backward respectively depending on the direction of the call at call set-up.

An interexchange data circuit is considered to be free for a new call when a clear message or a call rejected message has been sent, or received, subsequent to receipt, or sending, of those types of messages.

4.4.5 *Other messages*

Other types of messages are also provided for control of user facilities and network utilities as covered in § 5 and Recommendation X.300. Also, some message types are used in abnormal conditions as specified in § 4.5.

4.4.6 *Head-on collision*

When both-way working is used on a group of circuits, head-on collision can occur, i.e. the exchanges at each end may seize the same interexchange data circuit at approximately the same time. Head-on collision is detected when after sending of an address message, an address message is received as the first "backward" message.

In international operation, it may be necessary to employ preventive measures to reduce the probability of head-on collisions and to take action to minimize their effects. The following describes the preferred method:

The available circuits are divided into two groups of circuits, one group where the near end has priority access and one group with priority access for the opposite end. When an outgoing call is to be set up, the group with near end priority is searched according to the FIFO principle (first in – first out). If no circuit is found within this group, the group with opposite end priority is searched according to the LIFO principle (last in – first out). The separation between the groups is dynamic: each time when a circuit is released, it is transferred to the group with near end priority if the last call originated in the near end and to the group with opposite end priority if the last call originated in the opposite end.

The compatibility between the method described above and methods described in Q.724 and in Recommendation X.71 requires further study.

The detailed arrangements, in case of head-on collision requires further study.

4.5 *Call handling in abnormal conditions*

4.5.1 *Sending of a second backward message at call set-up*

As specified in § 4.3.1.3, the call accepted message can be sent before receipt of a *call accepted* (or corresponding) signal from the called user. If subsequently a condition occurs, e.g. *call collision*, that prevents the call from being completed, a call rejected message indicating this condition is sent. In these cases sending of the call rejected message will clear the call. At the originating exchange, receipt of the call rejected message will result in sending the applicable call progress signal to the calling user.

In some interworking situations and with some user facilities, sending of a second call accepted message may apply in the normal condition.

4.5.2 *Blocking and unblocking sequences*

Sending of a *blocking* signal will have the effect of prohibiting outgoing calls from the distant end of the relevant circuit, but will in itself not prohibit incoming calls to the exchange. Sending of the *unblocking* signal will have the effect of cancelling the blocked condition effected by the *blocking* signal. Acknowledgement sequences are always required for both the *blocking* and *unblocking acknowledgement* signals respectively. The acknowledgement is not sent until the relevant action, i.e. blocking or unblocking of the circuit, has been taken.

Removal (restoration) of a circuit from (to) traffic from both ends thus requires completion of a blocking and blocking acknowledgement (unblocking and unblocking acknowledgement) signal sequence relating to both directions.

Blocking of a circuit may be made during a call. In such a case the blocking sequence will be completed but the progress of the call will not be affected. However, after clearing of the call by means of the normal clearing signal sequence, the *blocked* condition will prevent the circuit from being seized by a new call. A *blocked* condition may in some situations be cancelled by a *reset circuit* signal (see § 4.5.5).

Blocking and unblocking sequences may be initiated by automatic or manual actions.

4.5.3 *Time-out supervision*

At various stages in the call set-up and clear-down procedures it is necessary to wait for receipt of a signal or condition from an adjoining exchange or user. The duration of such periods has to be controlled by appropriate time-outs. See also Figures 15/X.61 to 18/X.61. The operation of some of the time-outs will be affected by certain user facilities, see § 5. The values specified for the time-outs in the following are provisional.

All time-outs related to call set-up are terminated in the case that clearing takes place before expiry of the time-out.

4.5.3.1 *Originating exchange*

The following time-outs are necessary at call set-up:

a) T1 = 10-20 s; the time between the sending of the address message and the receipt of a call accepted message. On expiry of the time-out T1 the originating exchange will send the *no connection* call progress signal to the calling user and clear the call.

b) T2 = 5-10 s; the time between the receipt of the first call accepted message and detection of the *call accepted* condition. On expiry of time-out T2 the originating exchange will send the *no connection* call progress signal to the calling user and clear the call.

   Note – The operation of time-out T2 is modified when certain user facilities apply (see § 5).

4.5.3.2 *Destination exchange*

The following time-outs are necessary at call set-up:

a) T3: value as specified for the relevant DTE/DCE interface; the time between the sending of the *incoming call* (or corresponding) signal to the called user and the receipt of the *call accepted* (or corresponding) signal from the called user. On expiry of time-out T3, as specified for the applicable DTE/DCE interface protocol, the destination exchange will send a call rejected message containing a *network failure* signal and thus clear the call.

b) T4 = 5-10 s; the time between the sending of the call accepted message and receipt of a calling line identity message (when identification is requested). On expiry of time-out T4, the destination exchange will send a call rejected message containing a *network failure* signal and thus clear the call.

4.5.3.3 *Transit exchange*

Because a transit exchange is through-connected without waiting for an external event, no time-out supervision is required at call set-up.

4.5.3.4 *Circuit supervision*

The following time-outs are required in all exchanges:

a) T5 = 5-10 s; the time between sending of the clear message, containing a *circuit released* signal, or a call rejected message and the receipt of a clear message or a call rejected message (relating to the same interexchange data circuit and clearing action). On expiry of time-out T5, a new clear message containing a *circuit released* signal will be sent. Should clearing continue to be ineffective, a maintenance alarm will be activated after an appropriate time interval, and the circuit is kept busy. No time-out will apply following sending of the clear message containing a *circuit released acknowledgement* signal.

b) T6 = 5-10 s; the time between sending of a *blocking* or *unblocking* signal and receipt of a *blocking acknowledgement* or *unblocking acknowledgement* signal (respectively). On expiry of time-out T6, the *blocking* or *unblocking* signal will be repeated. Should blocking or unblocking continue to be ineffective, a maintenance alarm will be activated after an appropriate time interval.

4.5.4 *Call clear-down before completion of call set-up*

In some circumstances of call clear-down in abnormal conditions, signalling information relating to the call may subsequently be received. With the exception in § 4.5.6 such information will in all cases be discarded.

In the case where user clearing is detected or a clear message is received during the call set-up, the call set-up process is terminated and normal clear-down is performed. If the originating exchange has seized an interexchange data circuit, a clear message will not be sent unless an address message has already been sent.

In some cases it may be required to clear a call for management purposes. This can be achieved by initiating at any exchange the clearing procedures. See also § 4.5.5.

4.5.5 *Circuit resetting in abnormal situations*

In the case where the state of an interexchange data circuit becomes ambiguous, due to for example memory mutilation or processor disturbances at an exchange (X), the reset circuit may be used by that exchange to align the state of the circuit at both ends. The *reset circuit* signal is always acknowledged by a *circuit released acknowledgement* signal.

When receiving a *reset circuit* signal an exchange (Y) will:

a) respond with a *circuit released acknowledgement* signal in the case where the circuit is indicated as free;

b) release the circuit and respond with a *circuit released acknowledgement* signal in the case where the circuit is busy;

c) respond with a *blocking* signal followed by a *circuit released acknowledgement* signal in the case where the circuit is unconnected but indicated as being blocked at exchange X by exchange Y;

d) release the circuit and respond with a *blocking* signal followed by a *circuit released acknowledgement* signal in the case where the circuit is busy and indicated as being blocked at exchange X by exchange Y;

e) cancel a *blocked* condition (for outgoing calls) indicated as initiated by the distant end and act as in a)-d) above, as applicable, in the case where such a *blocked* condition exists at exchange Y in addition to one of the conditions a)-d).

Note – If the exchange (X) sending the *reset circuit* signal wants to preserve a *blocked* condition at the other end (Y), it (X) has to send a *blocking* signal subsequent to the *reset circuit* signal.

After sending a *reset circuit* signal, the exchange (X) will regard the circuit as unavailable for traffic until a response is received (from Y) at which time the exchange will:

i) return the circuit to the *idle* condition when a *circuit released acknowledgement* signal is received;

ii) regard it as operational but blocked by the distant end (Y) for outgoing calls when a *blocking* signal is received.

Appropriate means to cover situations in which no response to a *reset circuit* signal is received should be provided.

Note – The possible provision of means for resetting a group of circuits by a single action is for further study.

4.5.6 *Receipt of unreasonable signalling information*

The Message Transfer Part of the signalling system will avoid mis-sequencing of or double delivery of messages with a high reliability. However, undetected errors at the signalling link level and exchange malfunctions may produce unreasonable signalling information in messages that are either ambiguous or inappropriate.

In order to resolve some possible ambiguities in the state of a circuit when unreasonable signals are received, the following will apply:

a) If a *circuit released* signal is received relating to an idle circuit, it will be acknowledged with a *circuit released acknowledgement* signal.

b) If a *circuit released acknowledgement* signal is received relating to an idle interexchange data circuit, it will be discarded.

c) If a *circuit released acknowledgement* signal is received relating to a busy interexchange data circuit for which a circuit released signal has not been sent, the circuit will be released and a circuit released signal will be sent.

d) If a *blocking* signal is received for a blocked intercxchange data circuit, a *blocking acknowledgement* signal will be sent.

e) If an *unblocking* signal is received for an unblocked interexchange data circuit, an *unblocking acknowledgement* signal will be sent.

Any other unreasonable signalling information received will be discarded (see, however, § 4.4.6). If the discarding of the information prevents a call from being completed; that call will eventually be cleared by the expiry of a time-out.

5 **Additional call control and signalling procedures**

5.1 *General*

Paragraph 5 refers to the call control and signalling procedures that apply, in addition to the basic procedures specified in § 4, where user facilities and network utilities are involved.

The principles and procedures for realization of international user facilities and network utilities are defined in Recommendation X.300, which thus gives the basis for the common channel signalling procedures. Therefore, the following is limited to an outline of the implications for the common channel signalling procedures of such user facilities and network utilities.

The additional signalling information components particular to user facilities and network utilities are indicated in § 2. The corresponding formats and codes are specified in § 3.

5.2 *Closed user group facilities*

Depending on whether a closed user group is involved, the address message may include a *closed user group call* indication and an *interlock code.*

In certain cases of redirection of a closed user group call, that closed user group information included in the address message will also be returned back, within a call accepted message, to the exchange controlling redirection.

5.3 *Bilateral closed user group facilities*

The signalling system is capable of supporting automatic user controlled procedures for registration and cancellation of bilateral closed user groups. Three types of messages:

- facility registration/cancellation request message,
- facility registration/cancellation request accepted message, and
- facility registration/cancellation request rejected message,

which may include a number of signalling indications relating to bilateral closed user groups, are provided for those procedures.

At call set-up within a bilateral closed user group, the address message will contain a *bilateral closed user group call* indication.

Note – Subject to further study, it may be necessary to include further information relating to this facility in the address message, see Recommendation X.300.

5.4 *Calling line identification*

The signalling system provides for transfer of the calling line identity:

a) in the address message, systematically or selectively, or

b) in a calling line identity message, on request from the destination exchange as indicated in the call accepted message.

5.5 *Called line identification*

The called line identity is transferred in the call accepted message on request from the originating exchange as indicated in the address message.

The *national/international* indicator included in the address message may be used by the destination exchange to determine whether the called line identity should be the national or the complete international data number of the called user.

5.6 *Redirection of calls*

The signalling system provides a number of signals that cater for the *redirection of calls* facility.

In the case where the call is released back to a controlling exchange at redirection, the call accepted message will contain the *redirection request* signal, a *redirection address* indication and the *redirection address.* The original forward connection is cleared from the controlling exchange.

The address message sent for a call that during redirection is set up towards the new number (i.e. the *redirection address*) will contain a *redirected call* indication.

When a redirected call has been connected to the *redirection address,* the call accepted message sent towards the originating exchange will contain the *redirected call* signal. The *redirected call* signal is equivalent to the *call accepted* signal but has also the additional function of sending a *call progress* signal to the called user.

5.7 *Connect when free and waiting allowed*

The call accepted message sent from the destination exchange, when a call to a busy user having the *connect when free* facility is put in a queue, will contain the *connect when free* signal. At the originating exchange this signal will among other actions inhibit time-out T2.

When the waiting call is connected to the called user, a second call accepted message, now containing the *call accepted* signal, will be sent.

5.8 *Reverse charging and reverse charge acceptance*

When a reverse charging request from a calling user is allowed by the originating network, the address message will contain a *reverse charging request* indication. In the case where reverse charging is rejected because the called user does not have the *reverse charge acceptance* facility, the call rejected message will contain the *reverse charge acceptance not subscribed* signal. Otherwise the call is accepted or rejected as an ordinary call.

Note – The principles for accounting of reverse charging calls have not yet been determined; thus the possible implications of special accounting arrangements for the switching or interexchange signalling procedures have not yet been determined.

5.9 *Manual answer*

The call accepted message sent from the destination exchange at connection of a call to a user who employs *manual answer,* will contain the *terminal called* signal. At receipt of the *terminal called* signal at the originating exchange, through-connection will be prepared but time-out T2 will be lengthened to 2-4 minutes.

When the called user responds by a *call accepted* signal, a second call accepted message, now containing the *call accepted* signal, will be sent.

5.10 *RPOA selection*

In the case where a calling user selects a particular RPOA, an address message sent in the originating network will contain an *RPOA selection* indication and the applicable *RPOA transit network identity.* If such a call is rejected because the selected RPOA transit network cannot handle the call, the call rejected message sent will contain the *RPOA out-of-order* signal.

5.11 *Network identification utilities*

The capability for *originating network identification* on request from the destination network is mandatory for international calls. When this utility is employed the call accepted message will contain an *originating network identification request* indication. The identity of the originating network is then sent in a calling line identity message.

The signalling system also provides for transfer of the identity of the originating network within the address message.

Destination network identification and transit network identification by means of transfer of the network identities in the call accepted message are mandatory for international calls.

6 Signalling performance and traffic characteristics in data applications

6.1 *Signalling reliability*

6.1.1 *General*

Recommendation Q.706 details the factors that influence the performance of the message transfer service provided by a signalling network that uses the Message Transfer Part of Signalling System No. 7. It also provides information that may be used to estimate that performance in particular applications.

6.1.2 *Unsuccessful calls due to signalling malfunctions*

Although the Message Transfer Part is designed to provide a high reliability for transfer of messages through a signalling network, certain irregularities in message transfer cannot be prevented in certain situations.

Loss of the message will in most cases result in an unsuccessful call. The proportion of lost messages will primarily depend on the reliability of equipment used to realize certain signalling functions. The requirements specified for such equipments in Recommendation Q.706 will ensure that the proportion of lost calls in typical applications is 1 in 10^5 or better.

In certain extreme conditions, it is also possible that the message transfer function delivers faulty messages with reasonable information or delivers messages out-of-sequence. The probability of such malfunctions is, however, negligible from the circuit-switched data service point of view, see Recommendation Q.706.

6.1.3 *Availability of signalling*

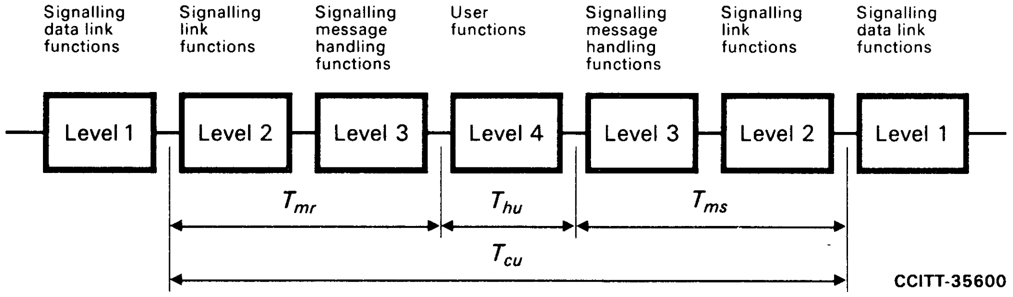
The availability of signalling primarily depends on the reliability of the equipment used to realize the signalling functions and the redundancy with which such equipment is provided.

No availability requirements for international signalling for the circuit-switched data service have yet been defined.

6.2 *Message transfer times*

6.2.1 *Functional reference points and signal transfer time components*

See Figure 19/X.61.



T_{cu}	Cross-office transfer time
T_{hu}	Data User Part handling time
T_{mr}	Message Transfer Part receiving time
T_{ms}	Message Transfer Part sending time

The definitions of these times are given in Recommendation Q.706.

FIGURE 19/X.61

Functional diagram of the signal transfer time

6.2.2 *Definitions*

6.2.2.1 **cross-office transfer time T_{cu}**

T_{cu} is the period which starts when the last bit of the signal unit leaves the incoming signalling data link and ends when the last bit of the signal unit enters the outgoing signalling data link for the first time. It also includes the queueing delay in the absence of disturbances but not the additional queueing delay caused by retransmission.

6.2.2.2 **data user part handling time, T_{hu}**

T_{hu} is the period which starts when the last bit of the message has entered the Data User Part and ends when the last bit of the derived message has left the Data User Part.

6.2.3 *Queueing delay*

An example of the queueing delays which may be expected in a particular case is shown in Appendix I to this Recommendation, see also § 6.3.

6.3 *Data signalling traffic models*

The characteristics of the signalling traffic generated for data call control will primarily depend on factors such as:

- the data traffic volume (call/s),
- the mix of different call types (international/national, successful/unsuccessful, etc.),
- the proportion of calls involving user facilities and network utilities and the mix of such facilities and utilities.

Appendix I contains two data signalling traffic models that indicate the mix of message types and lengths that result from particular sets of assumed conditions. The appendix also gives an example of the loading capacity of a signalling link for data call control signalling.

APPENDIX I

(to Recommendation X.61)

Examples of signalling traffic characteristics

I.1 *Signalling traffic models*

I.1.1 Tables I-1/X.61 and I-2/X.61 show two examples of mixes of data signalling message types and lengths. The models are simplified and do not fully reflect the possible variation of message lengths.

The following applies for both models:

- a mix of national and international calls is assumed with 8 and 12 digits in the data numbers respectively;
- the closed user group facility applies for 50% of the calls;
- the basic label specified in § 3.2.2.1 is used;
- the message length shown in the tables is the number of octets in the signalling information field of the corresponding signal unit; the overall length of the signal unit on the line is approximately 7 octets longer.

I.1.2 Table I-1/X.61 assumes that the calling line identity is always sent in the address message and that called line identification applies for 10% of the calls.

TABLE I-1/X.61

Example 1 of data signalling message mix

Message type	Messages/call	Message length (octets)
Address message	0.575	24
	0.425	18
Call accepted message	0.1	14
	0.9	8
Clear message	2	7

Message per call = 4
Average message length = 11 octets
Total amount of information per call = 576 bits.

I.1.3 Table I-2/X.61 assumes that the calling line identity is sent on request for 10% of the calls.

I.2 *Queueing delay and link loading*

Figure I-1/X.61 shows the mean value and standard deviation of message queueing delays for different signalling link loads.

The queueing delays shown in Figure I-1/X.61 assume:

- a message mix according to Table I-1/X.61,
- error-free operation of a signalling link using the basic error correction method.

The theoretical basis for calculation of the queueing delays and information about the performance of the signalling system under error conditions are included in Recommendation Q.706.

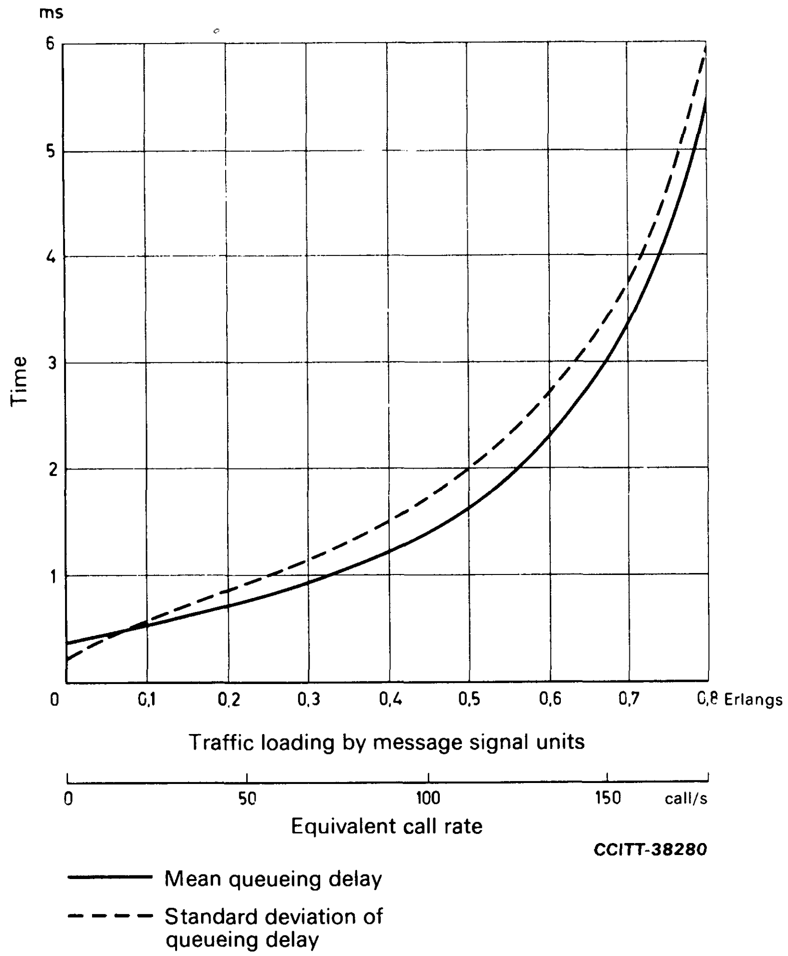
The equivalent call rate shown in the Figure I-1/X.61 assumes an even distribution of the calls in both directions of transmission.

TABLE I-2/X.61

Example 2 of data signalling message mix

Message type	Messages/call	Message length (octets)
Address message	0.575	18
	0.425	14
Call accepted message	1	8
Calling line identity message	0.1	14
Clear message	2	7

Messages per call = 4.1
Average message length = 9.7 octets
Total amount of information per call = 548 bits.



T_m 2.252 ms (144.1 bits and 64 kbit/s)
T_f 0.75 ms (48 bits and 64 kbit/s)
k_1 1.123
k_2 1.421

FIGURE I-1/X.61

Example of queueing delay as a function of link load

Recommendation X.70

TERMINAL AND TRANSIT CONTROL SIGNALLING SYSTEM FOR START-STOP SERVICES ON INTERNATIONAL CIRCUITS BETWEEN ANISOCHRONOUS DATA NETWORKS

(Geneva, 1972, amended at Geneva, 1976 and 1980;
Malaga-Torremolinos, 1984)

With the appearance of public data networks in various countries it becomes necessary to establish the appropriate international control signalling schemes for interworking in order to facilitate the introduction of such networks as much as possible. The main objective of public data networks is to offer to the user a great range of data signalling rates with a minimum of restrictions, very short call set-up and clear-down times and a variety of new service facilities. These requirements can be fulfilled only by a specially conceived signalling system which caters for all foreseeable needs and which is flexible enough to provide also for new, not yet defined services.

For these reasons, the CCITT,

unanimously recommends

for interworking between anisochronous data networks the control signalling scheme given below should be used on international circuits.

Note 1 – The start-stop user classes of service are specified in Recommendation X.1.

Note 2 – The signalling for synchronous user classes of service provided on anisochronous networks is the subject of further study.

Note 3 – The signalling on links between synchronous and anisochronous networks is the subject of further study.

Scope

This Recommendation defines a decentralized terminal and transit control signalling system for start-stop services on international circuits between anisochronous data networks.

1 General switching and signalling principles

1.1 The two classes, namely user class 1 and user class 2, which are considered applicable to anisochronous types of data network require a control signalling rate of 300 bit/s and 200 bit/s respectively.

Telex service based upon 50-baud trunks does not form part of this Recommendation[1].

1.2 Decentralized signalling will apply, the same channel being used for control signalling and data transmission.

1.3 Both terminal and transit operation will be required. Due to the inclusion of transit operation, link-by-link signalling control of calls will be adopted.

Onward selection from transit and incoming terminal centres should be arranged to overlap the receipt of *selection* signals, this in order to minimize call set-up times.

Selection signals will be transmitted by the originating country at automatic speed in a single block.

1.4 The numbering scheme that will be applied to networks accessed by this signalling system is defined in Recommendation X.121.

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[1] See Recommendation U.12 for telex and similar telegraph services.

The data network identification code (DNIC) (see Recommendation X.121), and network or service identification signals will be transmitted on both transit and terminal calls. However, the data country code (DCC) portion of the DNIC may be suppressed in the *selection* signals and only the network or service digit forwarded on terminal calls if requested by the incoming network.

1.5 Alternative routing will be permitted. The principle of high-usage circuits will be adopted, with overflow on to adequately provided routes between centres. Overflow on to higher speed circuits will not be permitted.

In order to prevent repeated alternative routing causing traffic to circulate back to the originating point, alternative routing will be restricted to once per call.

1.6 Both-way operation will be assumed and inverse order testing of circuits on both-way routes, or a close approximation to it by testing the route in small groups in fixed order always starting the search from the same position will be specified in order to minimize head-on collisions.

1.7 It is assumed that the gathering of information required for charging and accounting should normally be the responsibility of the originating Administration (see Recommendation D.10). Other arrangements for gathering information are for further study.

1.8 The grade of service to apply for the provision of circuits for links between public data networks of anisochronous type which carry traffic overflowed from other routes or from which overflow was not permitted would not be worse than one lost call in 50.

For high-usage direct links, circuits would be provided at a grade of service of not worse than one lost call in 10.

1.9 Sufficient switching equipment will be provided to ensure that congestion will not be signalled on more than 0.4% of calls in the busy hour, and only then when congestion has been positively identified.

1.10 The target setting-up time for the user classes of service applicable to these types of data networks will be one second.

2 Specific signalling characteristics

Notes applicable to § 2.

Note 1 – X denotes the international centre that originates the call under consideration on the international link concerned. Y denotes the international centre that receives the call under consideration on the international link.

Note 2 – Timings shown are within the centre concerned with no allowance being made for propagation and other delays, such as slow sending of selection signals from the originating terminal.

Note 3 – The times for permanent start polarity (A) and stop polarity (Z) are generally indicated in the following signal descriptions as integral multiples of a character (see Note 4).

Note 4 – For user class 1 the control signalling code (CSC) will employ 7-unit signalling characters with one parity bit, one start and two stop elements (see Table 8/X.70). The parity of the characters will be even and hence will be consistent with Recommendation X.4. The individual bits should be transmitted at the nominal modulation rate (300 bit/s) with the low order bit (i.e. b_1) first and completed by the parity bit (b_8).

The *end-of-selection* signal will be the International Alphabet (IA5) character 2/11(+). The reception confirmation will use IA5 character 2/10 (*). All other signals will be characters chosen from column 3 of IA5 (see Table 1/X.70). This choice helps ensure that the end of selection and reception confirmation signals are uniquely separable from the other signalling characters.

For user class 2 the CSC will employ 4-unit signalling characters with one parity bit, one start and two stop elements (see Table 8/X.70). The parity of the characters will be even with regard to elements of Z polarity. The individual bit should be transmitted at the nominal modulation rate (200 bit/s) with the low order bit (i.e. b_1) first and completed by the parity bit (b_5).

2.1 The signalling system between two data networks of anisochronous type is described in Table 1/X.70.

2.2 The incoming equipment may release the connection if the *calling* signal exceeds the specified maximum period (see Remarks column of Table 1/X.70). Start polarity will be maintained on the backward signalling path from centre Y to centre X.

2.3 The first forward path signal following the *calling* signal (class-of-traffic character) is distinctive from the first backward path signal to provide a guard against head-on collisions in the case of both-way operation.

A head-on collision is detected by the fact that centre X receives a first class-of-traffic character instead of the *reception-confirmation* signal or *reception-congestion* signal.

When a head-on collision is detected, the switching equipment at each end of the circuit should make another attempt to select a free circuit, either on the same group of circuits or on a group of overflow circuits, if facilities for alternative routing exist and there are no free circuits on the primary route. In the event of a further head-on collision on the second attempt, no further attempt will be made and the call will be cleared down. In the case of a transit centre, the *call progress* signal No. 20 followed immediately by the *clearing* signal will be returned to the preceding centre after the *reception-confirmation* signal and the *network* or *service identification* signals.

2.4 Failure to receive the *reception-confirmation* or *reception-congestion* signal within 4 seconds from the start of the calling signal or the reception of a spurious signal, as indicated by a character other than a first class-of-traffic characters the *reception-confirmation* signal or *reception-congestion* signal, should initiate the automatic *retest* signal on the circuit concerned.

In the case of failure to receive the correct *reception-confirmation* or *reception-congestion* signal, another attempt to select a circuit should be made (once only). In the case of transit calls, if the second attempt is unsuccessful, the *call progress* signal No. 20 followed immediately by the *clearing* signal, will be returned to the preceding centre after the *reception-confirmation* signal and the *network* or *service identification* signals.

2.5 *Selection* signals can be divided into two parts. The first part, designated as the *network selection* signals, contains information regarding network and user requirements and may be composed of one to nine (or possibly more) characters (see Tables 2/X.70, 3/X.70, 4/X.70, 4a/X.70, 5/X.70 and 5a/X.70). The second part comprises the *address* signals (the called network terminal number which is preceded by the DNIC always in the case of a transit call and also for terminal calls unless the destination country requests omission of the data country code portion, see Tables 6/X.70 and 6a/X.70).

The *network selection* signals used in the forward direction (see also Appendix II) are further subdivided and assembled as follows (see §§ 2.5.1 to 2.5.4 below) for signalling purposes.

Note that the term "user class of service" is abbreviated in the following to "user class".

2.5.1 *First class-of-traffic character* (see Table 2/X.70)

The *calling* signal is always followed by at least one class-of-traffic character. The bit functions of this character were so chosen that no further characters are needed for most connections.

If there is a need for indication of further requirements, a second class-of-traffic character (see § 2.5.3) may be used. Whether a second class-of-traffic or user class characters follow or not, will be indicated by the bits b_3 and b_4 of the first class-of-traffic character.

TABLE 1/X.70

Decentralized signalling between anisochronous data networks

Signal or function	Forward path (X towards Y)	Backward path (Y towards X)	Remarks
Free line	Start polarity (polarity A)	Start polarity (polarity A)	
Calling signal	Stop polarity (polarity Z) for a minimum period of one character and a maximum period of two characters followed immediately by *selection* signals		The equipment at centre Y should be ready to receive *selection* signals within one character period. The minimum and consequently the maximum periods will be lengthened at the request of the incoming country. (*Note* – The duration of the *calling* signal may require review in the light of false calling signals.)
Reception confirmation signal		Stop polarity followed by CSC character No. 14 (user class 2) or by IA5 character No. 2/10 (user class 1)	Stop polarity returned within three character periods after the end of receipt of the first class-of-traffic character. The return of CSC character No. 14 or IA5 character No. 2/10 shall be commenced within one to two character periods after the inversion to stop polarity. The *reception-confirmation* signal will have to be absorbed by the switching equipment of X and should not be able to go through that equipment to arrive at the preceding centre.
Selection signals	At least one (first class-of-traffic character only) or possibly several network selection signals depending on the network requirement (see Appendix I), the digits of the DNIC of the called network, the digits of the called terminal number, and an *end-of-selection* signal		These signals are transmitted immediately after the *calling* signal without awaiting the reception at X of the *reception-confirmation* signal. The selection signals are transmitted according to the control signalling code at the appropriate data signalling rate for the user class-of-service concerned and at automatic speed in a single block which includes an *end-of-selection* signal. For the user class 1 the *end-of-selection* signal will be IA5 character No. 2/11. For user class 2 the *end-of-selection* signal will be CSC character No. 11. The data country code (DCC) may be omitted on terminal calls at the request of the incoming country.
Network or service identification signals		CSC No. 12 followed by the data network identification code (DNIC) of the network	The character CSC No. 12 and DNIC follow the *reception-confirmation* signal at automatic speed within one to two character periods. These signals must go through centre X and arrive at the originating network. In all cases, the country or network identity shall consist of four decimal digits. The value of the fourth digit should, in the case when it is not explicity defined by the numbering plan, be at the discretion of the country in question within the limits allowed by the numbering plan.

TABLE 1/X.70 *(continued)*

Signal or function	Forward path (X towards Y)	Backward path (Y towards X)	Remarks
Reception-congestion signal		Stop polarity for a period of one or two characters followed by the *clearing* signal	This signal is returned within 0-5 character periods after the start of receipt of the *calling* signal when the selection signal cannot be received. This signal should be absorbed by centre X and not allowed to be received by a preceding exchange.
Call progress signals without clearing		CSC No. 11 followed by 2 digits (see Table 7d/X.70)	Examples would be *redirected call* or *terminal-called call progress* signal (for further details see Appendix III).
Call connected signal		One CSC character (see Table 7/X.70)	See § 2.14 of the text and for further details see Appendice III.
Strat of transit through-connect signal (STTC)		CSC No. 15 (see Table 7/X.70)	This signal always precedes the *transit through-connect* signal.
Tansit through-connect signal (TTC)		One CSC character (see Table 7b/X.70)	This signal will always be prefaced by the *start of transit through-connect* signal and will be returned preceding a *call progress* signal without clearing when this has to be sent. It will also be transmitted when the *calling* and/or *called line identification* is required (for further details see Appendix III).
Transit centres through-connected signal (TTD)	CSC No. 11 (see Table 6/X.70)		This signal will be transmitted within 40 to 120 ms after the receipt of the *transit through-connect* signal (TTC) when no calling line identification is required (for further details see Appendix III).
Called line identification (if applicable).		Combinations of the *called line identification* signals transmitted at automatic speed within 120 ms of the receipt of the TTD signal or the first character of the *calling line identification* signals	The *called line identification* signal consists of the data network or service identification code (DNIC) followed by the digits of the network terminal number and CSC No. 12. Where no identification is available, only No. 12 is sent (for further details see Appendix III).
Calling line identification (if applicable)	Combinations of the *calling line identification* signals transmitted at automatic speed within 40 to 120 ms of receipt of the *transit through-connect* signal (TTC)		The *calling line identification* signal consists of the data network or service identification code (DNIC) followed by the digits of the network terminal number and CSC No. 12. Where no identification is available only the DNIC and CSC No. 12 is sent (for further details, see Appendix III).

TABLE 1/X.70 *(concluded)*

Signal or function	Forward path (X towards Y)	Backward path (Y towards X)	Remarks
Originating through-connection signal	ACK character (combination 0/6 of IA5)		For definition see § 2.14 of the text and for further details see Appendix III).
Call progress signals with clearing		CSC No. 11 followed by 2 digits (see Table 7d/X.70), followed by the *clearing* signal	
Waiting signal	Stop polarity	Stop polarity	
Clearing signal	Inversion to start polarity in the direction of clearing. The minimum recognition time is 210 ms and the maximum time is 420 ms		The minimum period of start polarity on one signalling path which in itself ensures the complete release of the connection is 420 ms.
Clear confirmation signal	Inversion to continuous start polarity in the opposite direction after a minimum duration fo 210 ms of clearing signal and a maximum duration of 490 ms		The minimum and maximum periods for the release of the international circuit by an exchange are 210 ms and 490 ms respectively.
Incoming guard delay	Period of 390-420 ms measured from the moment when start polarity has been established on both signalling paths by: – either recognizing or transmitting the *clearing* signal on one signalling path, and – either transmitting or recognizing the *clear confirmation* signal on the other signalling path		A new incoming call shall not be accepted until this guard period has elapsed.
Outgoing guard delay	Period of 840 ms measured from the moment when start polarity has been established on both signalling paths by: – either recognizing or transmitting the *clearing* signal on one signalling path, and – either transmitting or recognizing the *clear confirmation* signal on the other signalling path		A new outgoing call shall not be originated until this guard period has elapsed.
Automatic retest signal	Stop polarity for 1-2 character periods followed by CSC No. 13 stop polarity for 4 seconds and then start polarity for a period of 56 seconds and the signal sequence is then repeated		See § 2.17 of the text.
Backward busy signal		Continuous stop polarity for a maximum period of 5 minutes	

Note – For the control signalling code (CSC) numbers mentioned, refer to Table 8/X.70.

TABLE 2/X.70

First CSC[a] character on the forward and backward paths

Combination				Condition signalled
b_4	b_3	b_2	b_1	
A	A			No further network selection signal follows[b]
A	Z			Second class-of-traffic character follows (see Table 4/X.70)[b]
Z	A			User class character follows (see Table 3/X.70)[b]
		A		Alternative routing not allowed[b]
		Z		Alternative routing allowed[b]
			A	Transit traffic[b]
			Z	Terminal traffic[b]
Z	Z	A	A	Retest signal[b]
Z	Z	A	Z	Reception confirmation for user class 2 only[c]
Z	Z	Z	A	Not allocated
Z	Z	Z	Z	Not allocated

[a] CSC = Control signalling code (see Table 8/X.70).
For user class 1 all characters are in column 3 ($b_5 = 1$, $b_6 = 1$, $b_7 = 0$) of International Alphabet No. 5. The eight bit (b_8) is chosen to give even parity over the character.

[b] First class-of-traffic character.

[c] For user class 2 only. The reception confirmation signal for user class 1 will be IA5 character No. 2/10.

2.5.2 *User class character (indication of speed and code* (see Table 3/X.70))

This character, if used, will follow the first class-of-traffic character and will be required when, for example, this information cannot be derived from the incoming line.

As eight user classes in Table 3/X.70 are not sufficient, a second user class character may be added by means of an escape character. Whether a second user class character follows or not, will be indicated by the bits b_1, b_2 and b_3 of the first user class character. Whether a second class-of-traffic character follows or not will be indicated by bit b_4 of the first user class character.

TABLE 3/X.70

First user class character a)

Combination				Condition signalled from X to Y b)
b_4	b_3	b_2	b_1	
A				No second class-of-traffic character follows
Z				A second class-of-traffic character follows (See Table 4/X.70)
	A	A	A	Reserve
	A	A	Z	300 bit/s (user class 1)
	A	Z	A	50 bit/s (user class 2)
	A	Z	Z	100 bit/s (user class 2)
	Z	A	A	110 bit/s (user class 2)
	Z	A	Z	134.5 bit/s (user class 2)
	Z	Z	A	200 bit/s (user class 2)
	Z	Z	Z	A second user class character follows c)

a) For user class 1 all characters are in column 3 ($b_5 = 1$, $b_6 = 1$, $b_7 = 0$) of International Alphabet No. 5. The eighth bit (b_8) is chosen to give even parity over the character.

b) The user class character may be omitted if, for example, the information can be derived from the incoming line.

c) For future extension.

2.5.3 *Second and subsequent class-of-traffic characters* (see Tables 4/X.70 and 4a/X.70)

These characters follow any user class characters required. The number of these class-of-traffic characters depends on the number of user facilities available.

The bit b_4 of the second or subsequent class-of-traffic characters indicate whether another class-of-traffic character follows or not.

TABLE 4/X.70

Second class-of-traffic character [a]

Combination				Condition signalled from X to Y
b_4	b_3	b_2	b_1	
A				No third class-of-traffic character follows
Z				Third class-of-traffic character follows (see Table 4a/X.70)
	A			No closed user group sequence follows
	Z			Closed user group sequence follows (see Table 5/X.70)
		A		Called line identification not required
		Z		Called line identification required
			A Z	Reserved for national use [b]

---

[a] For user class 1 all characters are in column 3 ($b_5 = 1$, $b_6 = 1$, $b_7 = 0$) of International Alphabet No. 5. The eighth bit (b_8) is chosen to give even parity over the character.

[b] On international circuits bit b_1 should be set to A polarity.

TABLE 4a/X.70

Third class-of-traffic character [a]

Combination				Condition signalled from X to Y
b_4	b_3	b_2	b_1	
A				No fourth class-of-traffic character follows
Z				Fourth class-of-traffic character follows [b]
	A			Redirection not allowed [c]
	Z			Redirection allowed [c]
		A		Not multiple address call [c]
		Z		Multiple address call [c]
			A Z	Not allocated

[a] For user class 1 all characters are in column 3 ($b_5 = 1$, $b_6 = 1$, $b_7 = 0$) of International Alphabet No. 5. The eighth bit (b_8) is chosen to give even parity over the character.

[b] Reserved for future needs.

[c] The international use of this signal requires further study.

2.5.4 *Closed user group characters* (see Tables 5/X.70 and 5a/X.70)

These characters are only used in conjunction with the second and possibly subsequent class-of-traffic characters which may follow.

The start of closed user group (CUG) character would precede the closed user group number which would be coded into a number of hexadecimal characters up to a maximum of four (see Table 5/X.70).

TABLE 5/X.70

Start of closed user group character [a] [b]

Combination				Condition signalled from X to Y
b_4	b_3	b_2	b_1	
A				Without outgoing access
Z				With outgoing access
	A			No DNIC [c] follows
	Z			DNIC [c] follows
		A	A	1 Number of hexadecimal CUG characters which follow
		A	Z	2
		Z	A	3
		Z	Z	4

---

[a] For user class 1 all characters are in column 3 ($b_5 = 1$, $b_6 = 1$, $b_7 = 0$) of International Alphabet No. 5. The eighth bit (b_8) is chosen to give even parity over the character.

[b] The start of closed user group character shall precede the DNIC of the representative user followed by the closed user group number which would be coded into a number of hexadecimal characters up to a maximum of four, as indicated. The closed user group number would be transmitted with the least significant bit of the least significant character first.

[c] On international circuits, bit_3, should be set to Z polarity.

TABLE 5a/X.70

Closed user group characters [a]

Combination				Condition signalled from X to Y
b_4	b_3	b_2	b_1	
A	A	A	A	0 Hexadecimal closed user group character
A	A	A	Z	1
A	A	Z	A	2
A	A	Z	Z	3
A	Z	A	A	4
A	Z	A	Z	5
A	Z	Z	A	6
A	Z	Z	Z	7
Z	A	A	A	8
Z	A	A	Z	9
Z	A	Z	A	A
Z	A	Z	Z	B
Z	Z	A	A	C
Z	Z	A	Z	D
Z	Z	Z	A	E
Z	Z	Z	Z	F

---

[a] For user class 1 all characters are in column 3 ($b_5 = 1$, $b_6 = 1$, $b_7 = 0$) of International Alphabet No. 5. The eighth bit (b_8) is chosen to give even parity over the character.

2.5.5 The numerical characters used for the second part of the selection signals are shown in Table 6/X.70. When the first class-of-traffic character indicates a terminal call, the incoming country can adopt the option not to receive the data country code portion of the DNIC. The complete selection block is terminated by an *end-of-selection* signal which is different for user classes 1 and 2. They are shown in Tables 6/X.70 and 6a/X.70.

TABLE 6/X.70

Miscellaneous forward path signals [a]

Combination				Condition signalled from X to Y	
b_4	b_3	b_2	b_1		
A	A	A	A	0	Digits for: – data network identification code (DNIC) – called network terminal number – calling line identification signals
A	A	A	Z	1	
A	A	Z	A	2	
A	A	Z	Z	3	
A	Z	A	A	4	
A	Z	A	Z	5	
A	Z	Z	A	6	
A	Z	Z	Z	7	
Z	A	A	A	8	
Z	A	A	Z	9	
Z	A	Z	A	End-of-selection signal for user class 2 only only [b] Transit centres through connected signal (TTD)	
Z	A	Z	Z	End-of-calling line identification signal [c]	
Z	Z	A	A	Not allocated	
Z	Z	A	Z		
Z	Z	Z	A		
Z	Z	Z	Z		

[a] For user class 1 all characters are in column 3 ($b_5 = 1$, $b_6 = 1$, $b_7 = 0$) of International Alphabet No. 5. The eighth bit (b_8) is chosen to give even parity over the character.

[b] For user class 2 only. The end-of-selection signal for user class 1 will be IA5-character No. 2/11.

[c] This signal follows the DNIC when the calling line identification is not available. See § 2.13.

TABLE 6a/X.70

Other forward path signal

IA5-character	Condition signalled from X to Y
0/6	Originating through-connection
2/11	End-of-selection signal for user class 1
2/15	Start of extended address

2.6 The incoming equipment should maintain start polarity on the backward signalling path by releasing the connection if the first received character is spurious, as indicated by a character other than a first valid class-of-traffic character. This procedure prevents the possibility of regarding a second *selection* signal as a first class-of-traffic character and provides a further safeguard against false calls.

In the case of receipt of a spurious signal as indicated by a parity error or by a character other than a valid selection signal (with the exception of the first class-of-traffic character), the incoming equipment should return the *call progress* signal No. 20 to the preceding centre, immediately followed by the *clearing* signal after the *reception-confirmation* signal and the *network* or *service identification* signals.

The incoming equipment may release the connection if all of the selection signals are not correctly received within a period of 15 seconds from the reception of the first class-of-traffic character. In this event, the *call progress* signal No. 20 is returned to the preceding centre, immediately followed by the *clearing* signal after the *reception-confirmation* signal and the *network* or *service identification* signals.

2.7 The address signal may consist of the international data number and an address extension.

The international data number may have a maximum number of 14 digits comprising the 4 digit data network identification code and a 10 digit maximum network terminal number. Alternatively the 14 digits can be considered as the 3 digit data country code followed by a national number of 11 digits maximum (see Recommendation X.121).

The possible address extension may either be included in the 14 digit international data number or may be separated from the international data number by a start-of-extension address-signal (2/15). In that case the extended address provisionally consists of up to 32 decimal digits. The network shall not be required to look at or operate on a network extension address. However some networks may look at the network address extension if they wish.

Note – The maximum length of 32 decimal digits is derived from the provisional maximum length of the OSI Network Service Access Point (NSAP) address defined in Recommendation X.213.

2.8 In the case of receipt of the *reception-congestion* signal at a transit centre, the *call progress* signal No. 61 should be returned to the preceding centre (after the *reception-confirmation* and *network* or *service identification* signals) and followed by the *clearing* signal.

2.9 The *network* or *service identification* signals shall be sent following the *reception-confirmation* signal in all cases. In all cases the country or network identity shall consist of four decimal digits. The value of the fourth digit should, in the case when it is not explicitly defined by the numbering plan, be at the discretion of the country in question within the limits allowed by the numbering plan.

If several transit networks are involved in setting up a call, the calling network will receive the network identifications one after the other. If a transit centre fails to receive the first character of the *network* or *service identification* signals within two seconds of the *reception-confirmation* signal, it will return to the preceding centre the *call progress* signal No. 20 (after the *reception-confirmation* and *network* or *service identification* signals), followed by the *clearing* signal.

The *network* or *service identification* signals could be useful for retracing the route followed by a call (for traffic statistics, international accounts, analysis of unsuccessful calls and the clearing of faults).

It is possible for a transit centre to receive backward path signals such as the *network* or *service identification* signals, *call connected* signal or *call progress* signals from subsequent centres, while the backward path signals originated locally are still being sent. It is necessary for the transit centre to ensure that the received signals are passed to the preceding centre without mutilations or loss.

2.10 The backward path signals indicating effective and ineffective call conditions are scheduled in Tables 7/X.70, 7a/X.70, 7b/X.70, 7c/X.70 and 7d/X.70.

TABLE 7/X.70

Miscellaneous backward path signal [a)]

Combination				Condition signalled from X to Y
b_4	b_3	b_2	b_1	
A	A	A	A	0 Digits for: – network or service identification signals – called line identification signals – call progress signals
A	A	A	Z	1
A	A	Z	A	2
A	A	Z	Z	3
A	Z	A	A	4
A	Z	A	Z	5
A	Z	Z	A	6
A	Z	Z	Z	7
Z	A	A	A	8
Z	A	A	Z	9
Z	A	Z	A	Start of call progress signal (see Table 7d/X.70)
Z	A	Z	Z	End-of-called line identification signal [b)] Start of network or service identification signal
Z	Z	A		Call connected signal
			A	Call metering
			Z	No call metering
Z	Z	Z	A	Start of transit through-connect signal (STTC)
Z	Z	Z	Z	Further backward path signal follows (see Table 7a/X.70)

[a)] For user class 1 all characters are in column 3 (b_5 = 1, b_6 = 1, b_7 = 0) of International Alphabet No. 5. The eighth bit (b_8) is chosen to give even parity over the character.

[b)] This signal is also used alone when the called line identification is not available.

TABLE 7a/X.70

Further miscellaneous backward path signals [a] [b]

Combination				Condition signalled from Y to X
b_4	b_3	b_2	b_1	
A				Reserved for national use
Z				
	A	A	A	Not allocated
	A	A	Z	
	A	Z	A	
	A	Z	Z	
	Z	A	A	
	Z	A	Z	
	Z	Z	A	
	Z	Z	Z	

[a] For user class 1 all characters are in column 3 ($b_5 = 1$, $b_6 = 1$, $b_7 = 0$) of International Alphabet No. 5. The eighth bit (b_8) is chosen to give even parity over the character.

[b] These signals follow combination ZZZZ in Table 7/X.70.

TABLE 7b/X.70

Transit through-connect signals [a) b)]

Combination				Condition signalled from Y to X
b_4	b_3	b_2	b_1	
A	A	A	A	Not allocated
A	A	A	Z	
A	A	Z	A	
A	A	Z	Z	
A	Z	A	A	
A	Z	A	Z	
A	Z	Z	A	
A	Z	Z	Z	
Z	A	A	A	
Z	A	A	Z	
Z	A	Z	A	
Z	A	Z	Z	
Z	Z			Transit through-connect signal (TTC)
		A		Calling line identification not required
		Z		Calling line identification required
			A	Call metering
			Z	No call metering

a) For user class 1 all characters are in column 3 (b_5 = 1, b_6 = 1, b_7 = 0) of International Alphaber No. 5. The eighth bit (b_8) chosen to give even parity over the character.

b) These signals follow the start of transit through-connect signal (STTC) in Table 7/X.70.

TABLE 7c/X.70

Other backward path signals

IA5-character	Condition signalled from Y to X
2/10	Reception-confirmation for user class 1

TABLE 7d/X.70

Call progress signals [a) g)]

Numerical code first/second digit	Category	Significance
01 02 03	Without clearing	Terminal called Redirected call Connect when free
20 21 22 23	With clearing, due to subscriber – short term [b)]	Network failure [c)] Number busy [d)] [d)]
41 42 43 44 45 46 47 48 49 51 52	With clearing, due to subscriber – long term [b)]	Access barred Changed number Not obtainable Out of order Controlled not ready Uncontrolled not ready DCE power off [d)] Network fault in local loop Call information service Incompatible user class service
61	With clearing, due to network – short term [b)]	Network congestion
71 72	With clearing, due to network – long term [b)]	Degradated service [e)]
81 82 83	With clearing, due to DTE-network procedure	Registration/cancellation confirmed [f)] [d)] [d)]

[a)] For user class 1 all characters are in column 3 ($b_5 = 1$, $b_6 = 1$, $b_7 = 0$) of International Alphabet No.5. The bit (b_8) is chosen to give even parity over the character.

[b)] "Short term" in this context approximates to the holding time of a call, whilst "long term" implies a condition that can persist for some hours or even days.

[c)] At the originating exchange, this results in sending a call progress signal "no connection" to the calling customer and clearing the call.

[d)] These signals are normally only utilized between first centre and subscriber and are not signalled on inter-network links.

[e)] Only utilized within national networks.

[f)] Not yet included. To be studied in relation to Recommendation X.300 on network call-control procedures.

[g)] A *call progress* signal without clearing should precede the *called line identification* signal. A *call progress* signal with clearing could precede or follow the *called line identification* signal.

2.11 If any *call progress* signal or the *call connected* signal are not received within 30 seconds from the end of the selection, then the *call progress* signal No. 20 will be returned to the preceding centre (after the *reception-confirmation* and *network* or *service identification* signals), followed by the *clearing* signal.

2.12 If the called station is not able to receive information immediately, the return of the *call connected* signal should be delayed accordingly. This point is left for further study.

2.13 In this type of signalling, originating and terminating national centres contain the identification of the calling or called subscribers respectively. These identifications may be exchanged within the network as an optional subscriber's feature.

If the *called line identification* has been requested but is not available, the terminating centre in this connection should send only the *end-of-line identification* signal (CSC-character No. 12). If the *calling line identification* has been requested but is not available, the originating centre in this connection should send only the data network identification code (DNIC) signals followed by the *end-of-line identification* signal (CSC-character No. 12).

2.14 The *call connected* signal confirms that the call is accepted by the called subscriber and, if applicable, the *calling line identification* has been completely received by the terminating centre and passed to the called subscriber and, when applicable, that the *called line identification* has been completely transmitted to the originating centre (see Appendix III).

The originating *through-connection* signal confirms that the *call connected* signal has been received by the originating centre and, when applicable, that a *call progress* signal without clearing has been completely received by the originating centre and passed to the calling subscriber or, when applicable, that the *called line identification* has been completely received by the originating centre and passed to the calling subscriber (see Appendix III).

The *call connected* signal is sent on the backward path by the terminating centre. The originating *through-connection* signal is sent by the originating centre both to calling and called subscribers.

The connection must be switched through in the originating centre within 20 ms after transmission of the originating *through-connection* signal (see Appendix III). This limit follows from the condition given in Recommendation X.20 for the beginning of data transmission.

The connection must be switched through in the terminating centre within 40 ms after transmission of the *call connected* signal (see Appendix III).

The connection must be switched through in the transit centres within 40 ms after transmission of the *call connected* or *transit through-connect* signal (see Appendix III).

If a transit centre has a character orientated switch, the connection may be switched through within 40 ms after transmission of the *call connected* signal for user class 2 service.

Complete network through-connection is assured when the originating *through-connection* signal is received by the data terminals.

2.15 If the terminating centre fails to receive the *transit centres through-connected* signal (TTD) or, if applicable, the first character of the *calling line identification* signals within 4 seconds after having sent the *transit through-connect* signal (TTC), it will return to the preceding centre the *call progress* signal No. 20 followed by the *clearing* signal.

2.16 The guard delays on clearing are measured from the moment when start polarity has been established on both signalling paths by:

- either recognizing or transmitting the *clearing* signal on one signalling path, and
- either transmitting or recognizing the *clear-confirmation* signal on the other signalling path.

For incoming calls this guard period shall be 390-420 ms. A new incoming call shall not be accepted until this guard period has elapsed. This is on the assumption that the terminating centre will be able to accept the first *selection* signal after a negligible period of stop polarity and will also be able to return the *reception-confirmation* signal within a negligible delay after the receipt of the first class-of-traffic character.

The guard period on clearing for outgoing calls should be a period of at least 840 ms. A new outgoing call shall not be originated until this guard period has elapsed.

If exchanges are able to distinguish between the different clearing conditions, shorter periods may be introduced accordingly.

2.17 The automatic *retest* signal will be initiated as indicated in § 2.4.

This signal transmitted over the forward signalling path is composed of a maximum of five successive cycles, each cycle incorporating:

- stop polarity for 1-2 character periods (see Note) followed by CSC No. 13, followed by stop polarity for a maximum period of 4 seconds;
- start polarity for a period of 56 seconds.

Note – The minimum and consequently the maximum periods will be lengthened at the request of the incoming country Y (see Remarks column of Table 1/X.70).

The circuit should be tested up to 5 times at nominal intervals of one minute and a check made to confirm the receipt of the *reception-confirmation* signal on the backward path in response to each test. If the *reception-confirmation* signal has not been received at the end of this first group of tests, the retest will continue with a further group of up to 5 tests at either 5- or 30-minute nominal intervals. If 5-minute intervals are used and the *reception-confirmation* signal has not been received at the end of this second group of tests, further retests will be made at 30-minute intervals. An alarm will be given at an appropriate time. However, this retest procedure may be discontinued at any stage at the discretion of the outgoing Administration.

If, however, during the above sequence of retests, the *reception-confirmation* signal is received, a *clearing* signal will be transmitted in place of the *retest* signal. Following a valid *clear-confirmation* signal, the incoming and the outgoing sides of the trunk circuit should not be returned to service until after expiry of the appropriate guard delay time. In order to cater for the possibility that a faulty circuit may be seized at both ends, the automatic retest equipment should be arranged to allow an incoming call to be received during the start polarity period. An Administration may, however, ignore such calls which occur during the incoming guard delay period.

The interval between the tests at the two ends of the trunk circuit should be made different by increasing the nominal interval by 20% at one end, to be sure that successive retests do not overlap at both ends. In general, the intercontinental transit centre having the higher DNIC should take the longer interval (i.e. 1.2, 6 and 36 minutes). Nevertheless, when this requirement would entail considerable difficulty, alternative arrangements may be adopted by agreement between the two Administrations or RPOAs concerned.

Where an exchange has knowledge of a transmission system failure it is desirable that the *retest* signals shall not be applied to the circuits affected.

In order to avoid simultaneous seizure of too many registers at the distant centre, it is advisable that the *retest* signals, which may be sent simultaneously on various circuits subjected to this test, should be out of phase with one another.

The use of a special first class-of-traffic character for retest permits the incoming centre to be informed about retests on its incoming circuits.

2.18 If at the receiving end, parity does not check during the establishment of the connection, provisionally the connection should be cleared down unless otherwise specified. However, the possibility of different actions remains open for further study.

TABLE 8/X.70

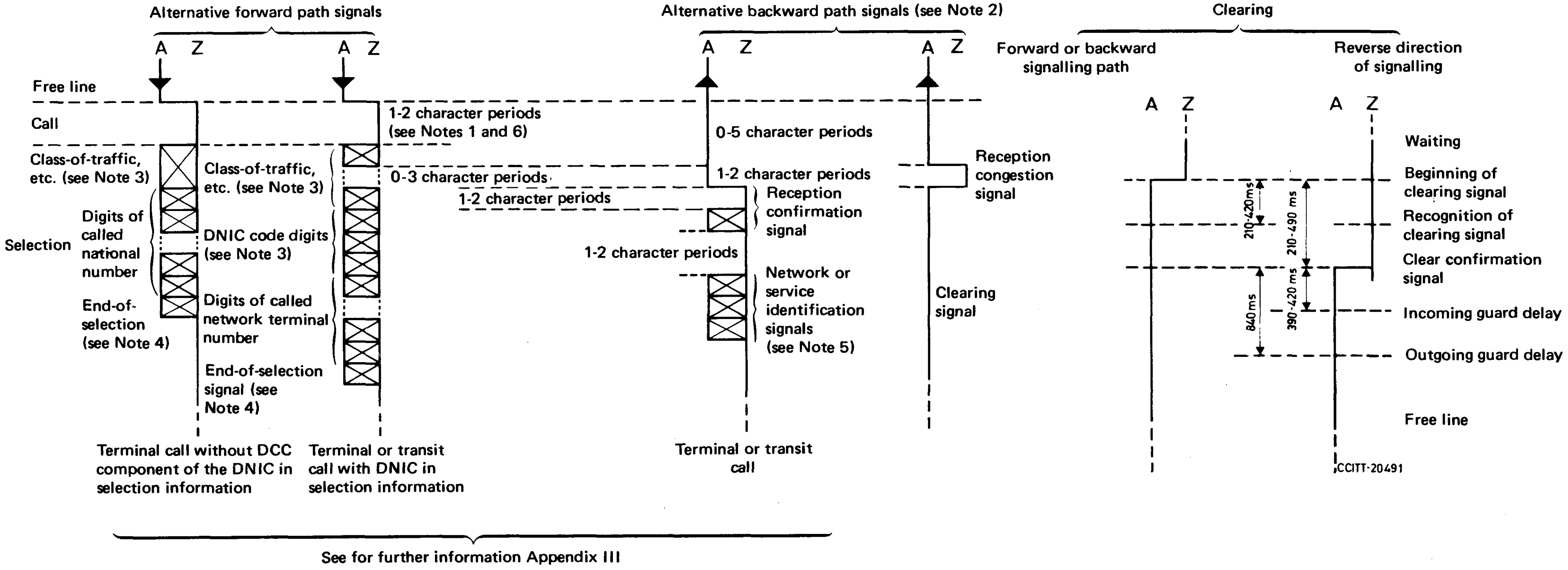
Control signalling code (CSC)

CSC character number	CSC character structure			
	b_4	b_3	b_2	b_1
1	A	A	A	A
2	A	A	A	Z
3	A	A	Z	A
4	A	A	Z	Z
5	A	Z	A	A
6	A	Z	A	Z
7	A	Z	Z	A
8	A	Z	Z	Z
9	Z	A	A	A
10	Z	A	A	Z
11	Z	A	Z	A
12	Z	A	Z	Z
13	Z	Z	A	A
14	Z	Z	A	Z
15	Z	Z	Z	A
16	Z	Z	Z	Z

Note 1 – The 7-unit code with one parity check bit, 1-unit start and 2-unit stop elements for user class 1 and the 4-unit code with 1 parity check bit, 1-unit start and 2-unit stop elements for user class 2 used in this control signalling system are listed in the table. As the bits b_5, b_6 and b_7 of the 7-unit code have a permanent pattern (1,1,0) only the bits b_1, b_2, b_3 and b_4 are shown.

Note 2 – The parity bit of the signal should correspond to even parity with regard to unit elements of Z polarity. The individual bits should be transmitted at the nominal data signalling rate of 200 bit/s (user class 2) and 300 bit/s (user class 1) with the low order bit (b_1) first and completed by the parity check bit (b_5 or b_8).

Note 3 – The transmitting part of the signalling device shall send the control character at the nominal modulation rate (300 bauds for user class 1 and 200 bauds for user class 2) of ±0,2% with a maximum degree of gross start-stop distortion of 5%. The receiving part of the signalling device shall have an effective net margin of not less than 40%.



Note 1 – Timings shown as character periods refer to the complete control character of user class (11 units at 300 bits/s) and user class 2 (8 units at 200 bit/s). Switching and propagation delays are not included.

Note 2 – *Forward path* signals may also appear in the backward path, indicating a head-on collision on both-way circuits.

Note 3 – *Network selection* signals (class-of-traffic, user class characters, etc.). See Tables 2-5/X.70. DNIC comprises 4 digits.

Note 4 – *Selection* signals will always be sent as a single block by the originating network with and *end-of-selection* signal in all cases.

Note 5 – The *network* or *service identification* signals comprise a distinctive character followed by the DNIC of the network concerned.

Note 6 – The minimum and consequently the maximum periods will be lengthened at the request of the incoming country Y.

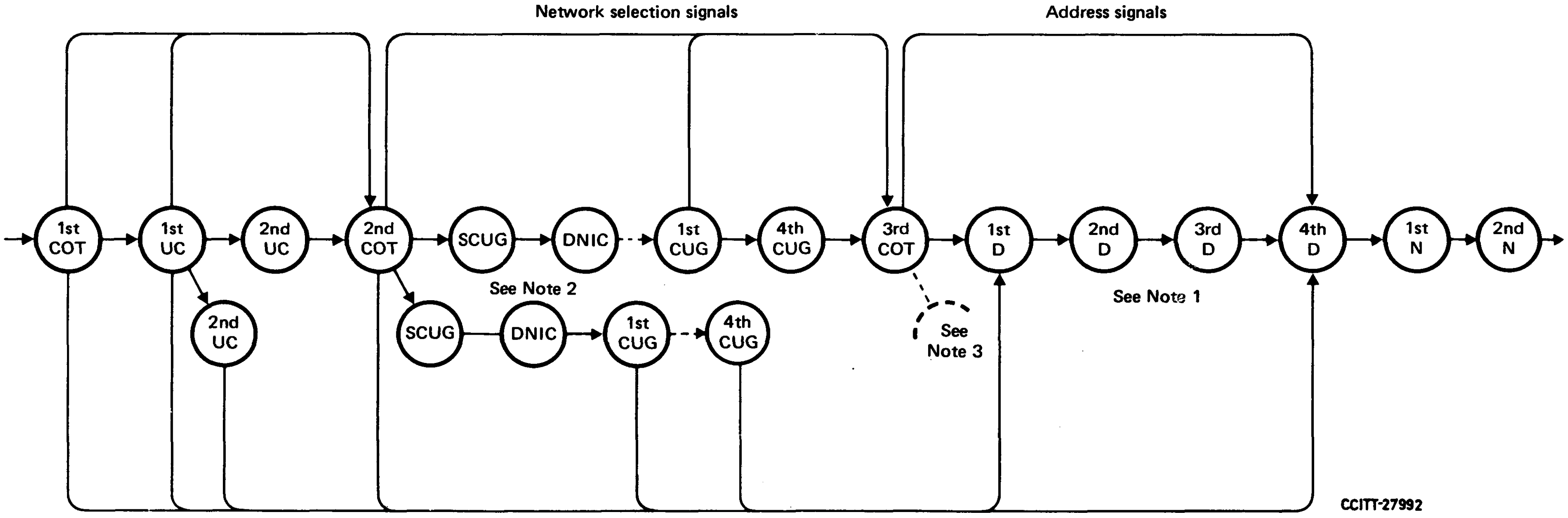
FIGURE 1/X.70

Decentralized signalling between data networks of anisochronous type

APPENDIX I

(to Recommendation X.70)

Possible sequences of network selection signals



COT	Class-of-traffic character
UC	User class character
SCUG	Start of closed user group character
DNIC	Data network identification code
CUG	Closed user group character
D	Data network (or service) identification code digit
N	Called number digit

Note 1 – The first three digits D_1, D_2 and D_3, form the data country code (DCC) component of the data network identification code (DNIC). The fourth digit (D_4) is the network or service digit of the DNIC.

Note 2 – The DNIC comprises four digits as defined in Note 1.

Note 3 – Reserved for future extension.

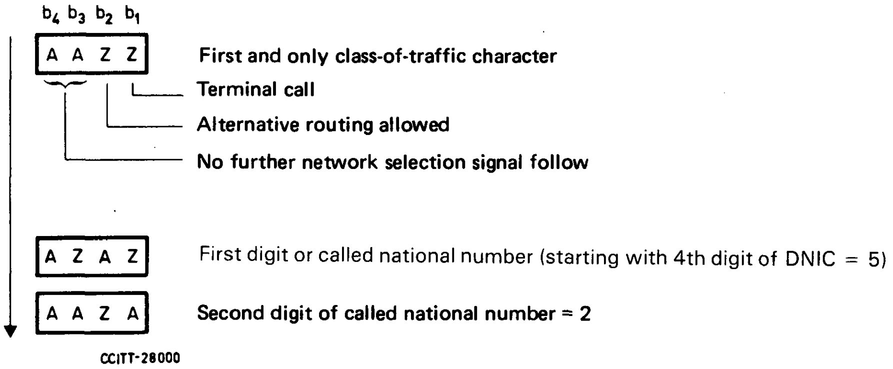
APPENDIX II

(to Recommendation X.70)

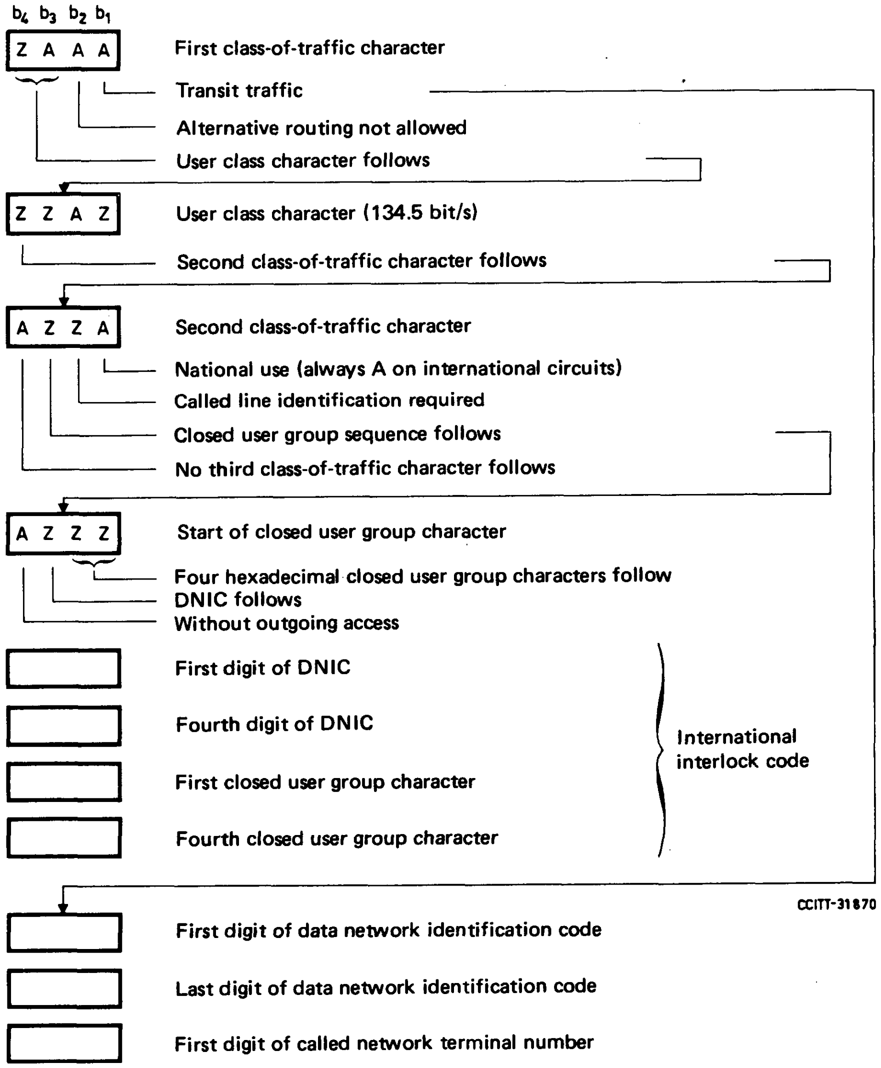
Examples of network selection signals

II.1 *First example* (minimum sequence of network selection signals)

This example shows a sequence of minimal length. The country of destination has indicated that it does not wish to receive the DCC component of the DNIC. (The preceding *calling* signal, start and stop elements, possible stuffing bits and the parity bit are not shown. The bits are shown in the order of b_4, b_3, b_2 and b_1).



II.2 *Second example* (a sequence of *network selection* signals including closed user group characters)

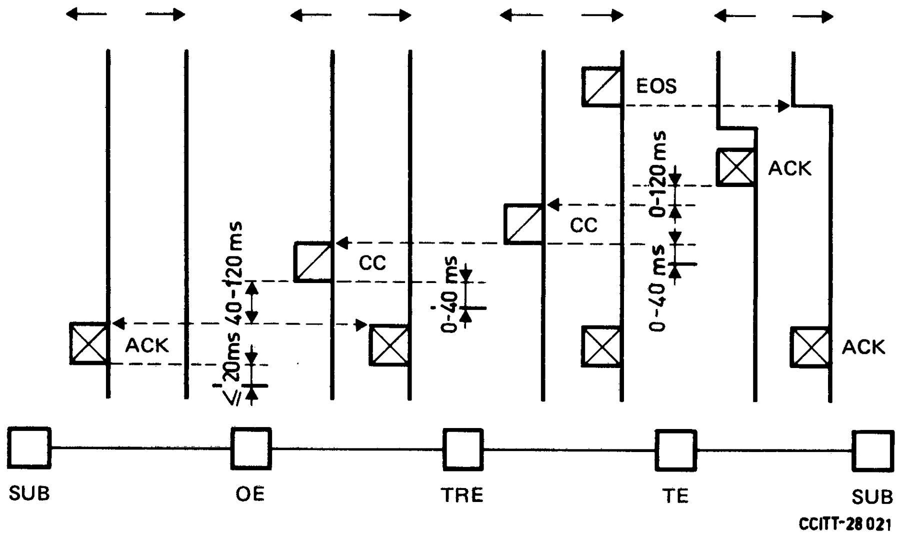


APPENDIX III (A)

(to Recommendation X.70)

Through-connection procedure

Called and calling line identification not required (no *connect-when-free* facility)



---→	Correlation line
——	Through-connection
⧄	CSC or IA5-character
☒	IA5-character

EOS	End-of-selection signal
CC	Call connected signal
SUB	Subscribers
OE	Originating exchange
TRE	Transit exchange
TE	Terminating exchange

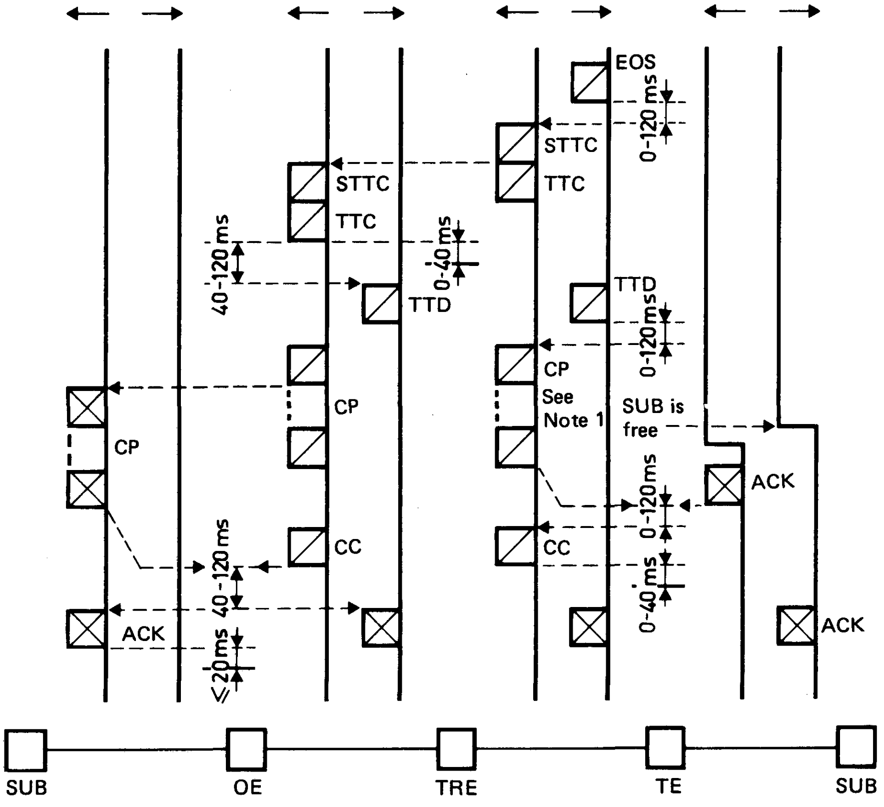
Note – Timings shown are worst case figures and design objectives should be to keep them as short as possible.

APPENDIX III (B)

(to Recommendation X.70)

Through-connection procedure

Called and calling line identification not required (subscriber is busy, *connect-when-free* facility)



CCITT-28031

– – – ➔	Correlation line
——	Through-connection
◪	CSC or IA5-character
☒	IA5-character
EOS	End-of-selection signal
STTC	Start-of-transit through-connect signal
TTC	Transit through-connect signal

TTD	Transit centres through-connected signal
CP	Call progress signals (without clearing)
CC	Call connected signal
SUB	Subscribers
OE	Originating exchange
TRE	Transit exchange
TE	Terminating exchange

Note 1 – *Call progress* signals comprise a distinctive character followed by a 2-digit number.

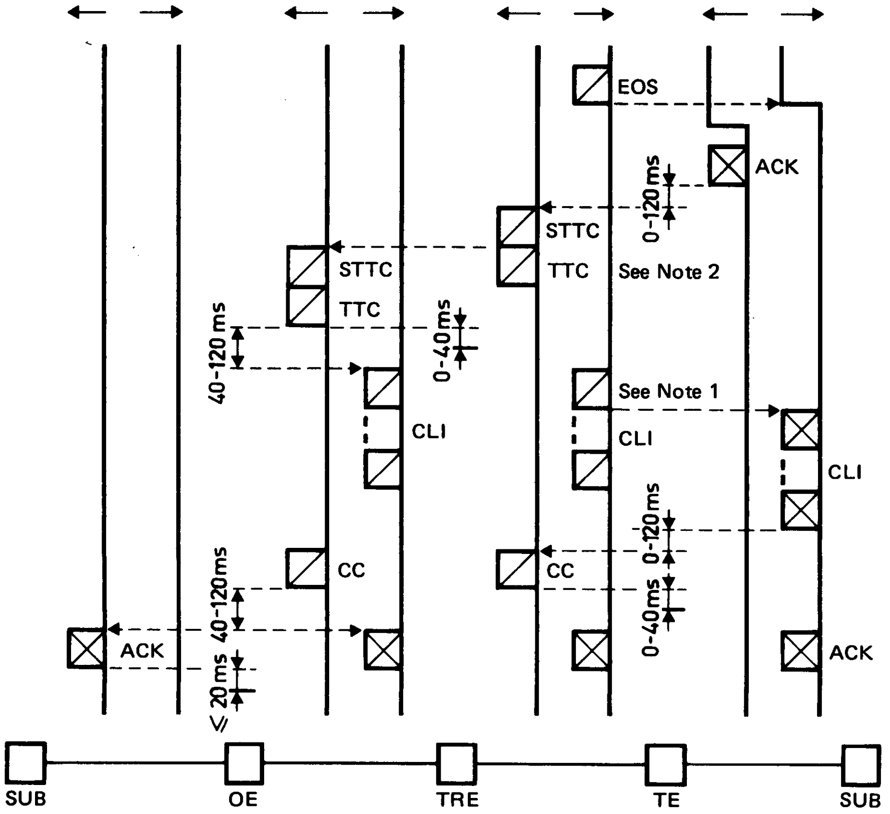
Note 2 – See Note in Appendix III (A).

APPENDIX III (C)

(to Recommendation X.70)

Through-connection procedure

Called line identification not required, calling line identification required (no *connect-when-free* facility)



CCITT-28042

- - - ▸	Correlation line
——	Through-connection
⧄	CSC or IA5-character
⊠	IA5-character
EOS	End-of-selection signal
STTC	Start-of-transit through-connect signal

TTC	Transit through-connect signal
CLI	Calling line identification signals
CC	Call connected signal
SUB	Subscribers
OE	Originating exchange
TRE	Transit exchange
TE	Terminating exchange

Note 1 – The *calling line identification* signal consists of the DNIC followed by the digits of the subscriber number and CSC No. 12. Where no identification is available, only the DNIC followed by CSC No. 12 is sent.

Note 2 – In this example, it is assumed that the *STTC* signal is sent after the receipt of the *call accepted* signal (ACK). However, some countries may decide to return this signal following a positive subscriber check state (not busy), at the same time as the subscriber call set-up is initiated.

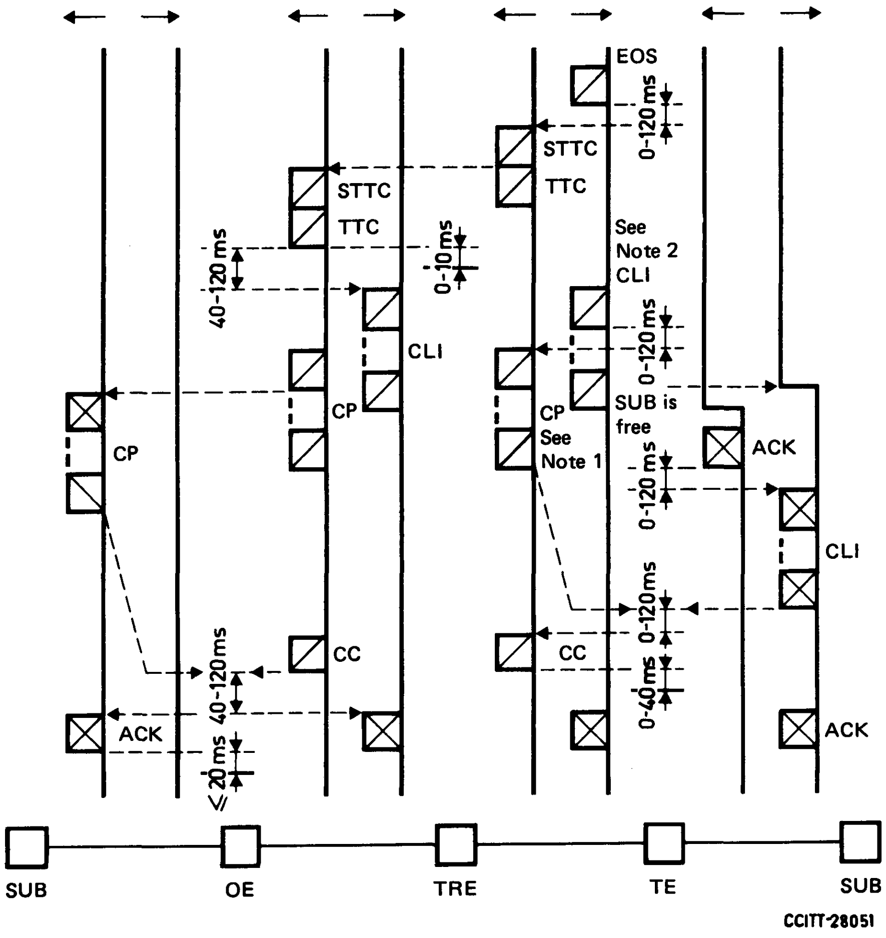
Note 3 – See Note in Appendix III (A).

APPENDIX III (D)

(to Recommendation X.70)

Through-connection procedure

Called line identification not required, calling line identification required (subscriber is busy, *connect-when-free* facility)



- - - ▸	Correlation line
———	Through-connection
▨	CSC or IA5-character
☒	IA5-character

EOS	End-of-selection signal
STTC	Start-of-transit through-connect signal
TTC	Transit through-connect signal
CLI	Calling line identification signals
CP	Call progress signals
CC	Call connected signal
SUB	Subscribers
OE	Originating exchange
TRE	Transit exchange
TE	Terminating exchange

Note 1 – *Call progress* signals comprise a distinctive character followed by a 2-digit number.

Note 2 – The *calling line identification* signal consists of the DNIC followed by the digits of the subscriber number and CSC No. 12. Where no identification is available, only the DNIC followed by CSC No. 12 is sent.

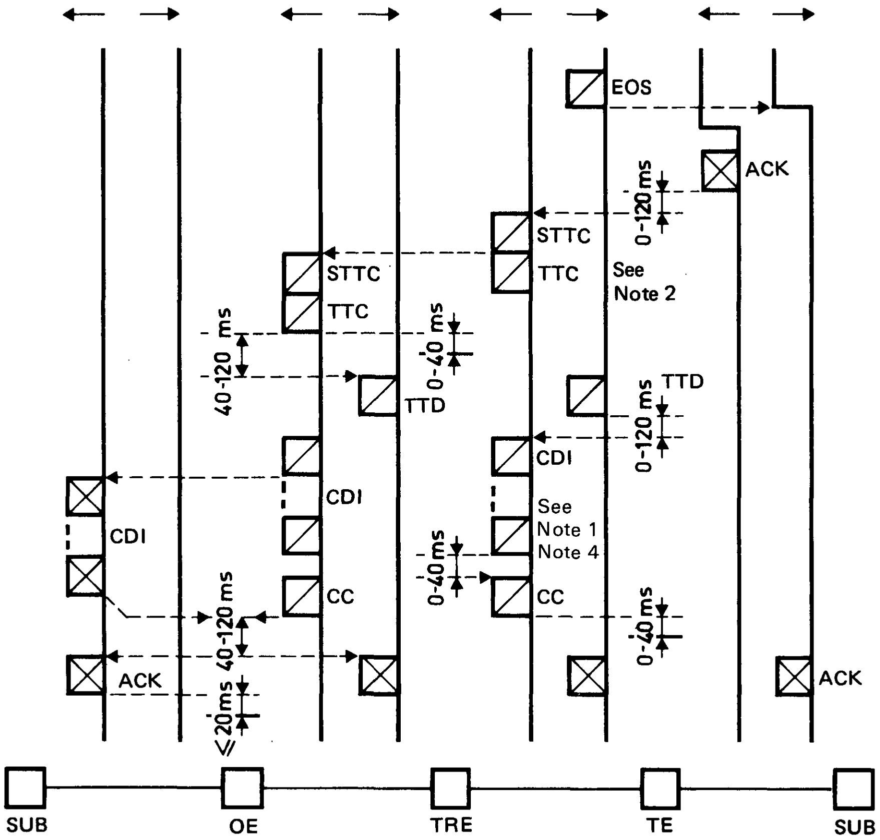
Note 3 – See Note in Appendix III (A).

APPENDIX III (E)

(to Recommendation X.70)

Through-connection procedure

Called line identification required, calling line identification not required (no *connect-when-free* facility)



CCITT-28 061

- - -▸	Correlation line	TTD	Transit centres through-connected signal
——	Through-connection	CDI	Called line identification signals
⧄	CSC or IA5-character	CC	Call connected signal
		SUB	Subscribers
⊠	IA5-character	OE	Originating exchange
EOS	End-of-selection signal	TRE	Transit exchange
STTC	Start-of-transit through-connect signal	TE	Terminating exchange
TTC	Transit through-connect signal		

Note 1 – The *called line identification* signal consists of the DNIC followed by the digits of the subscriber number and CSC No. 12. Where no identification is available, only CSC No. 12 is sent.

Note 2 – In this example, it is assumed that the *STTC* signal is sent after the receipt ot the *call accepted* signal (ACK). However, some countries may decide to return this signal following a positive subscriber check state (not busy), at the same time as the subscriber call set-up is initiated.

Note 3 – See Note in Appendix III (A).

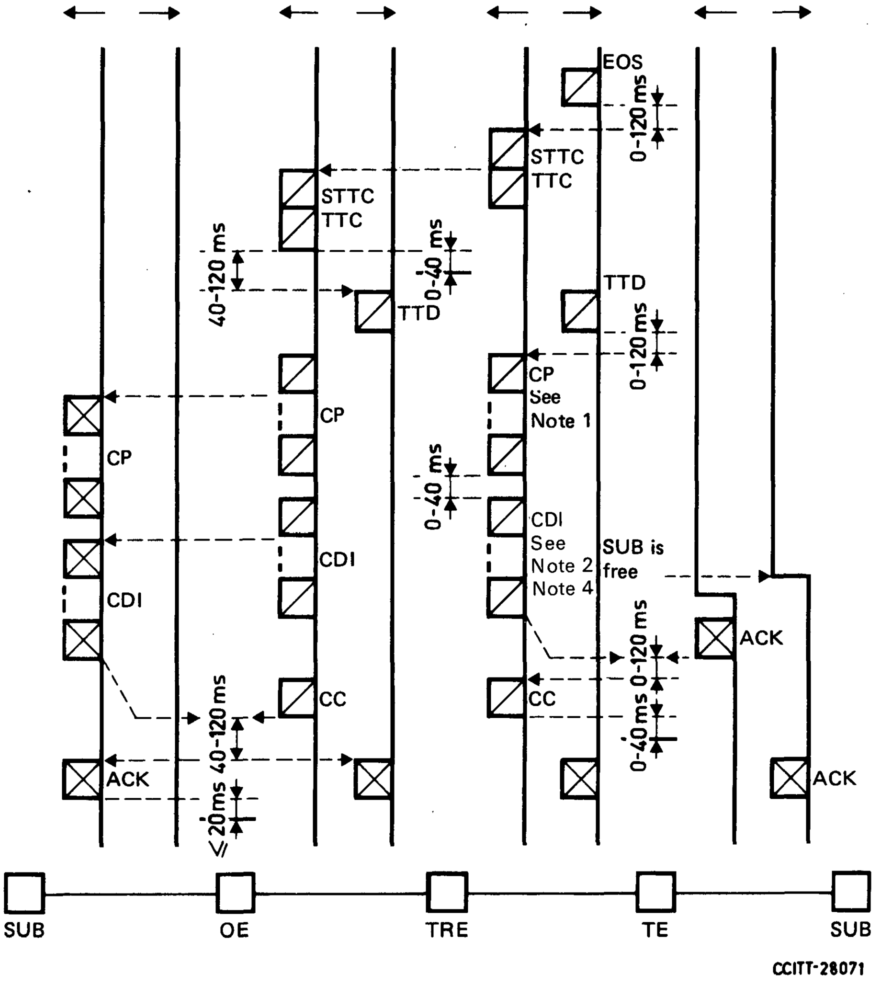
Note 4 – If the call is cleared the relevant *call progress* signal should be sent before or after the *called line identification* signal.

APPENDIX III (F)

(to Recommendation X.70)

Through-connection procedure

Called line identification required, calling line identification not required (subscriber is busy, *connect-when-free* facility)



- - - →	Correlation line	TTD	Transit centres through-connected signal
———	Through-connection	CP	Call progress signals
⧄	CSC or IA5-character	CDI	Called line identification signals
		CC	Call connected signal
⊠	IA5-character	SUB	Subscribers
EOS	End-of-selection signal	OE	Originating exchange
STTC	Start-of-transit through-connect signal	TRE	Transit exchange
TTC	Transit through-connect signal	TE	Terminating exchange

Note 1 – *Call progress* signals comprise a distinctive character followed by 2-digit number.

Note 2 – The *called line identification* signal consists of the DNIC followed by the digits of the subscriber number and CSC No. 12. Where no identification is available, only CSC No. 12 is sent.

Note 3 – See Note in Appendix III (A).

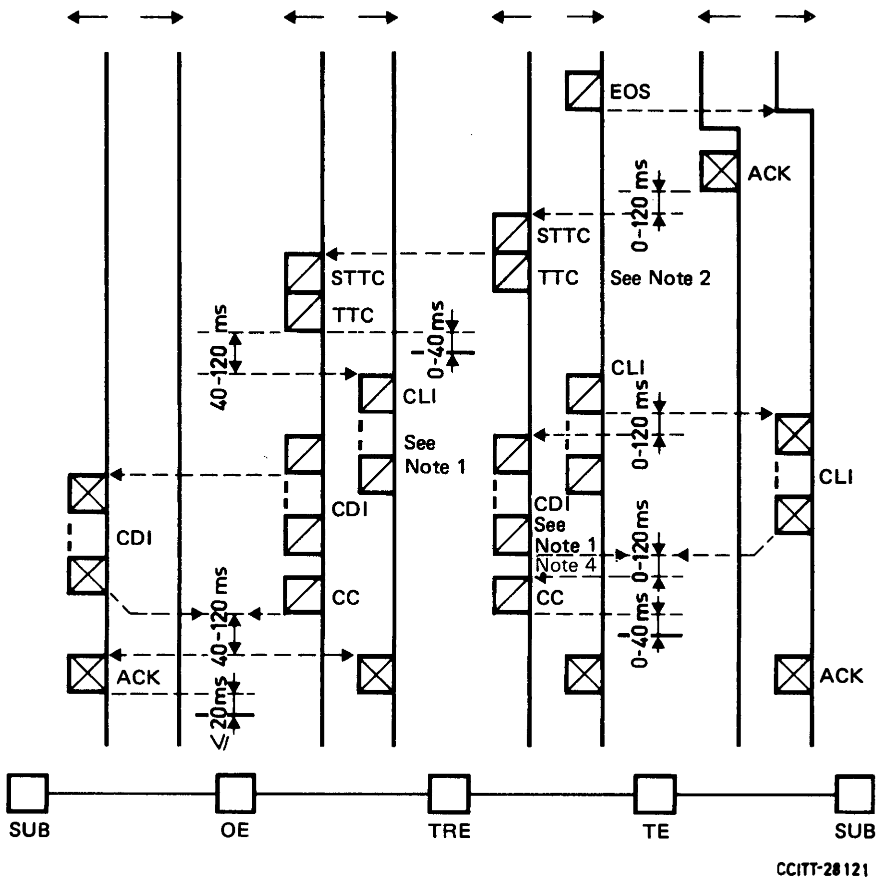
Note 4 – If the call is cleared after the sending of the *called line identification* signal, but before through-connection, a relevant *call progress* signal with clearing could be sent.

APPENDIX III (G)

(to Recommendation X.70)

Through-connection procedure

Called and calling line identification required (no *connect-when-free* facility)



- - - ▶ Correlation line

—— Through-connection

▨ CSC or IA5-character

☒ IA5-character

EOS	End-of-selection signal
STTC	Start-of-transit through-connect signal
TTC	Transit through-connect signal
CLI	Calling line identification signals
CDI	Called line identification signals
CC	Call connected signal
SUB	Subscribers
OE	Originating exchange
TRE	Transit exchange
TE	Terminating exchange

Note 1 – The *called line identification* signal consists of the DNIC followed by the digits of the subscriber number and CSC No. 12. Where no called line identification is available, only CSC No. 12 is sent.

The *calling line identification* signal consists of the DNIC followed by the digits of the subscriber number and CSC No. 12. Where no calling line identification is available, only the DNIC followed by CSC No. 12 is sent.

Note 2 – In this example, it is assumed that the *STTC* signal is sent after the receipt of the call accepted signal (ACK). However, some countries may decide to return this signal following a positive subscriber check state (not busy), at the same time as the subscriber call set-up is initiated.

Note 3 – See Note in Appendix III (A).

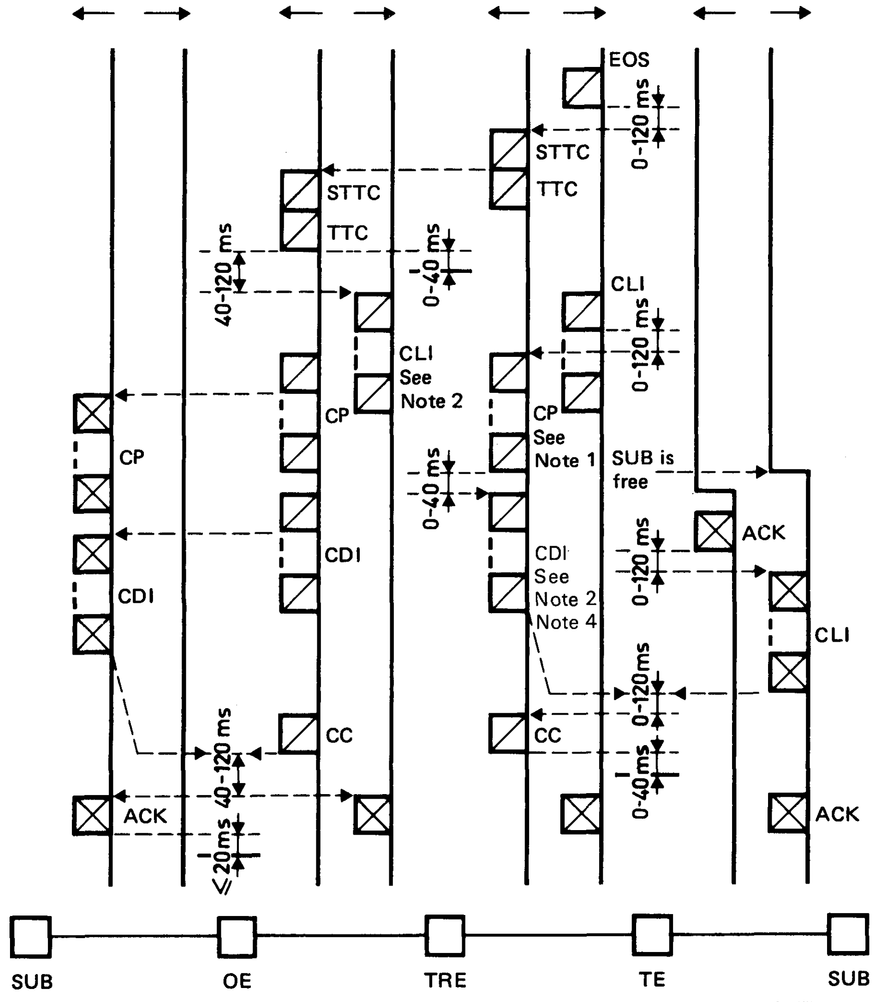
Note 4 – If the call is cleared, the relevant *call progress* signal should be sent before or after the *called line identification* signal.

APPENDIX III (H)

(to Recommendation X.70)

Through-connection procedure

Called and calling line identification required (subscriber is busy, *connect-when-free* facility)



CCITT-28 131

- - - ► Correlation line

——— Through-connection

⧄ CSC or IA5-character

⊠ IA5-character

EOS	End-of-selection signal
STTC	Start-of-transit through-connect signal
TTC	Transit through-connect signal
CLI	Calling line identification signals
CP	Call progress signals
CDI	Called line identification signals
CC	Call connected signal
SUB	Subscribers
OE	Originating exchange
TRE	Transit exchange
TE	Terminating exchange

Note 1 – *Call progress* signals comprise a distinctive character followed by a 2-digit number.

Note 2 – The *called line identification* signal consists of the DNIC followed by the digits of the subscriber number and CSC No. 12. Where no called line identification is available, only CSC No. 12 is sent.
The *calling line identification* signal consists of the DNIC followed by the digits of the subscriber number and CSC No. 12. Where no calling line identification is available only the DNIC followed by CSC No. 12 is sent.

Note 3 – See Note in Appendix III (A).

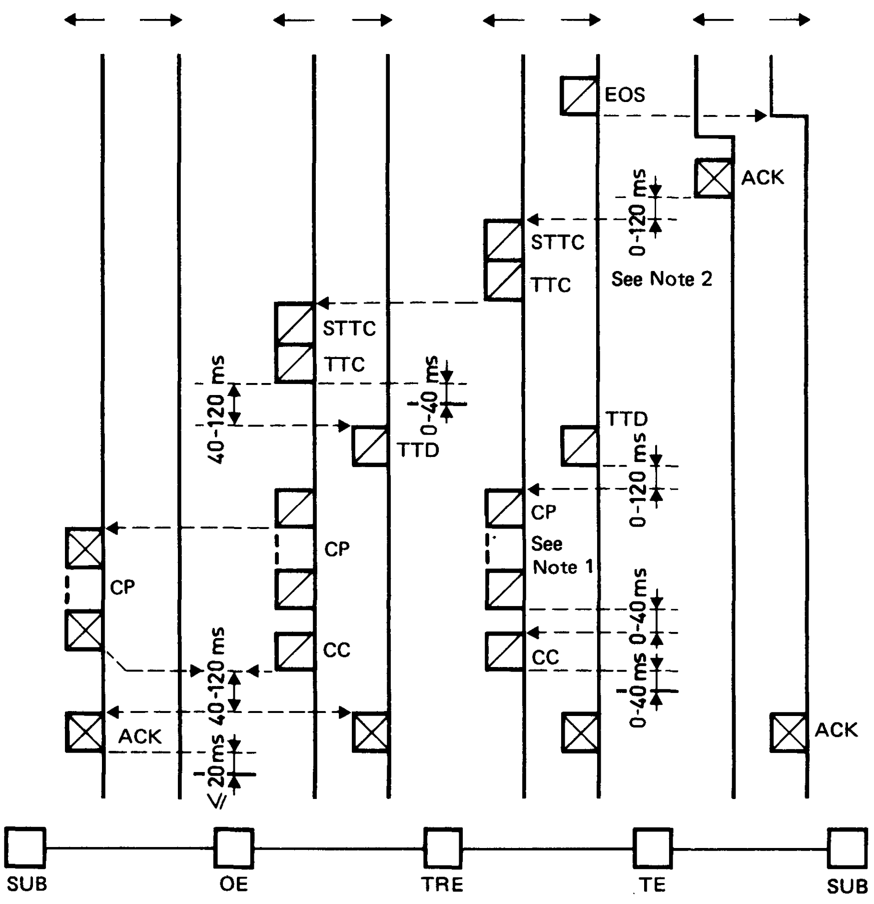
Note 4 – If the call is cleared after the sending of the *called line identification* signal, but before through-connection, a relevant *call progress* signal with clearing could be sent.

APPENDIX III (I)

(to Recommendation X.70)

Through-connection procedure

Called and calling line identification not required (call progress signal without clearing, e.g. redirected call)



CCITT-28101

– – – ▸ Correlation line

——— Through-connection

▨ CSC or IA5-character

☒ IA5-character

EOS	End-of-selection signal
STTC	Start-of-transit through-connect signal
TTC	Transit through-connect signal
TTD	Transit centres through-connected signal
CP	Call progress signal
CC	Call connected signal
SUB	Subscribers
OE	Originating exchange
TRE	Transit exchange
TE	Terminating exchange

Note 1 – *Call progress* signals comprise a distinctive character followed by a 2 digit number.

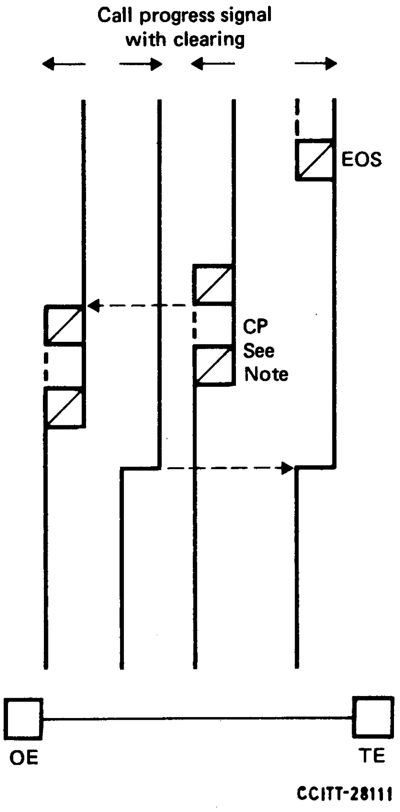
Note 2 – In this example, it is assumed that the *STTC* signal is sent after the receipt of the *call accepted* signal (ACK). However, some countries may decide to return this signal following a positive subscriber check state (not busy), at the same time as the subscriber call set-up is initiated.

Note 3 – See Note in Appendix III (A).

APPENDIX IV

(to Recommendation X.70)

Unsuccessful call



– – –→	Correlation line
⧄	CSC or IA5-character
EOS	End-of-selection signal
OE	Originating exchange
TE	Terminating exchange
CP	Call progress signal

Note – *Call progress* signals comprise a distinctive character followed by a 2-digit number.

Recommendation X.71

DECENTRALIZED TERMINAL AND TRANSIT CONTROL SIGNALLING SYSTEM ON INTERNATIONAL CIRCUITS BETWEEN SYNCHRONOUS DATA NETWORKS

(Geneva, 1976, amended at Geneva, 1980; Malaga-Torremolinos, 1984, and Melbourne, 1988)

With the appearance of public data networks in various countries it becomes necessary to establish the appropriate international control signalling schemes for interworking in order to facilitate the introduction of such networks as much as possible. The main objective of public data networks is to offer to the user a great range of data signalling rates with a minimum of restrictions, very short call set-up and clear-down times and a variety of new service facilities. These requirements can be fulfilled only by specially conceived signalling systems which cater for all foreseeable needs and which are flexible enough to provide also for new, not yet defined, services.

For these reasons, the CCITT

unanimously recommends

for interworking between synchronous data networks utilizing decentralized control signalling techniques the scheme given below should be used on international circuits.

Note 1 – The synchronous user classes of service are as specified in Recommendation X.1.

Note 2 – The signalling on links between synchronous and anisochronous networks is the subject of further study.

Note 3 – The interworking between common channel and decentralized signalling is the subject of Recommendation X.80.

Scope

This Recommendation defines a decentralized control signalling system for use in setting up terminal and transit calls on international circuits between synchronous data networks.

1 General switching and signalling principles

1.1 Signalling will be at bearer rates appropriate to the synchronous user classes of data only. It is expected that start-stop user classes of data, telex, etc., will be assembled and transmitted in accordance with Recommendation X.52.

1.2 The control signalling should employ bits transmitted at the maximum data signalling rate of the links provided.

1.3 Decentralized signalling will apply, the same channel being used for control signalling and data transmission.

1.4 Both terminal and transit operation will be required. Due to the inclusion of transit operation, link-by-link signalling control of calls will be adopted.

The data network identification code (DNIC) (see Recommendation X.121), and *network* or *service identification* signals will be transmitted on both transit and terminal calls. However, the data country code (DCC) portion of the DNIC may be suppressed and only the network or service digit forwarded on terminal calls if requested by the incoming network.

Onward selection from transit and incoming terminal centres should be arranged in order to commence as soon as possible.

Selection signals will be transmitted by the originating or transit country, or network, in a single block.

1.5 The numbering scheme that will be applied to networks accessed by this signalling system is defined in Recommendation X.121.

1.6 Alternative routing will be permitted. The principle of high-usage circuits will be adopted, with overflow onto adequately provided routes between centres.

Overflow onto higher speed circuits will not be permitted.

In order to prevent repeated alternative routing causing traffic to circulate back to the originating point, alternative routing will be restricted to once per call.

1.7 Both-way operation will be assumed and inverse order of testing of circuits on both-way routes or a close approximation to it by testing the route in small groups in fixed order always starting the search from the same position, will be specified in order to minimize head-on collisions.

1.8 It is assumed that the gathering of information required for charging and accounting should normally be the responsibility of the calling Administration (see Recommendation D.10). Other arrangements for gathering information are for further study.

1.9 The grade of service to apply for the provision of circuits for links between public data networks of the synchronous type which carry traffic overflowing from other routes, or from which overflow was not permitted, would not be worse than one call lost in 50.

For high-usage direct links circuits would be provided at a grade of service of not worse than one lost call in 10.

1.10 Sufficient switching equipment will be provided to ensure that congestion will not be signalled by return of a *reception-congestion* signal or absence of a *proceed-to-select* signal on more than 0.4% of calls in the busy hour and, in the first case, only then when congestion has been positively identified.

1.11 The target setting-up time for the user classes of service applicable to these types of data networks is for further study.

2 Specific signalling characteristics

Notes applicable to § 2.

Note 1 – X denotes the international centre which originates the call under consideration on the international link concerned. Y denotes the international centre which receives the call under consideration on the international link.

Note 2 – Timings shown are within the centre concerned, excluding propagation and other transmission delays.

Note 3 – The signalling plan will employ 8-bit signalling characters and continuous 0s[1] and 1s.

During the control signalling stage, the status bits are 0s. Upon the final through-connection in the originating exchange, the status bits on both signalling paths are 1s.

For the case of signalling characters, the parity of the characters will be odd, and hence will be consistent with Recommendation X.4 for links and connections using end-to-end synchronous operation, and with Recommendation X.21. For the case of signals being continuous 0s, or continuous 1s, parity is undefined inasmuch as no characters are employed. Moreover, character synchronization is not maintained over a period of continuous 0s or 1s, but must be re-established when further signalling characters are sent.

All groups of contiguous characters will be preceded by at least two repetitions of International Alphabet No. 5 (IA5) character 1/6 (SYN). The term "at least" means two SYN characters for the 600 bit/s user class. For the higher speed user classes, the number of SYN characters could be two or more but the total number of SYN characters should not unnecessarily prolong the setting-up time. If two signalling groups are combined to form one group of contiguous characters, the SYN characters may be omitted from within this group.

The *end-of-selection* signal will be the IA5 character 2/11 (+). The *call confirmation* and *proceed-to-select* signals will use IA5 character 2/10 (*).

---

[1] The impact of the all zeros pattern is left for further study.

Apart from the abovementioned signals (namely, continuous 0s, continuous 1s, 1/6, 2/10 and 2/11), all signals will be characters chosen from column 3 of IA5 (see Table 1/X.71). This choice helps ensure that the synchronization and other characters specified above are uniquely separable from the IA5 column 3 signalling characters.

An example of three successive signalling characters within five octets of one channel of the Recommendation X.50 multiplex structure is shown in Appendix V. In the Recommendation X.51 multiplex structure, the signalling characters will be aligned with the 8 + 2 envelope.

2.1 The signals between two data networks of synchronous type are described in Table 1/X.71. There are two protocols, the Call Confirmation Protocol (CCP) and the Proceed-To-Select Protocol (PTSP). The CCP is the basic method of this Recommendation and the PTSP is an option for an interim period at the discretion of the incoming network.

2.2 The incoming equipment may release the connection as follows:

2.2.1 *Call Confirmation Protocol*

If the *calling* signal exceeds the specified maximum period, but not before at least one call confirmation character has been transmitted.

2.2.2 *Proceed-to-select Protocol*

If the first selection signal is not received within 2 seconds after having sent the *proceed-to-select* signal.

2.3 A head-on collision is detected by the fact that exchange X receives *calling* signal (repetitions of 1s) followed by SYN characters, instead of *call confirmation* or *proceed-to-select* signal (SYN characters followed by repetitions of 2/10) or *reception-congestion* signal (repetitions of 1s followed by *clearing* signal).

When a head-on collision is detected, the switching equipment at each end of the circuit should make another attempt to select a free circuit, either on the same group of circuits or on a group of overflow circuits, if facilities for alternative routing exist and there are no free circuits on the primary route. In the event of a further head-on collision on the second attempt, no further attempt will be made and the call will be cleared down. In the case of a transit centre, the *call progress* signal No. 20 is returned to the preceding centre within a sequence of signals ordered as follows: *call confirmation* or *proceed-to-select, network* or *service identification,* the *call progress* signal and *clearing.*

2.4 Failure to receive *reception-congestion, call confirmation* or *proceed-to-select* signal within 4 seconds from the start of the *calling* signal, the reception of a spurious signal as indicated by a signal other than *reception-congestion, call confirmation* or *proceed-to-select* signal, or by a head-on collision, can initiate the automatic *retest* signal on the circuit concerned.

The need for an automatic *retest* signal may not be so great in a digital environment, its purpose being met by alternative methods. If an automatic *retest* signal is used, however, it will conform to § 2.16.

In the case of failure to receive *reception-congestion, call confirmation* or *proceed-to-select* signal, an attempt to select another circuit should be made (once only). In the case of transit calls, if the second attempt is unsuccessful, the *call progress* signal No. 20 is returned to the preceding centre within a sequence of signals ordered as follows: *call confirmation* or *proceed-to-select, network* or *service identification,* the *call progress* signal and *clearing.*

TABLE 1/X.71

Decentralized signalling between synchronous data networks

Signal or function	Forward path (X towards Y)	Backward path (Y towards X)	Remarks
Free line	S = 0, continuous repetitions of 0s	S = 0, continuous repetitions of 0s	
Calling signal	S = 0, continuous repetitions of 1s		
	For CC protocol, this signal is continuous for a minimum period of 10 ms or 16 information bits, whichever is the greater in time, and for a maximum period of 15 ms or 24 information bits, whichever is the greater in time [a]		The equipment at exchange Y should be ready to receive *selection* signals within a period of 10 ms or 16 information bits, whichever is the greater in time, from the start of the received *calling* signal.
	For PTS protocol, this signal is continuous until the *proceed-to-select* signal is received.		The *proceed-to-select* signal should be returned when the equipment is ready to receive *selection* signals.
Call confirmation signal (CC protocol)		S = 0, continuous repetitions of IA5 character 2/10 maintained until the first class-of-traffic character is recognized and always preceded by at least 2 SYN characters (1/6).	Returned within 10 ms or 16 information bits of receipt of the *calling* signal, whichever is the greater in time. The *call confirmation* signal shall be followed by the *network* or *service identification* signal within 50 ms of receipt of the first class-of-traffic character, followed by the *waiting* signal if no other characters follow contiguously. The *call confirmation* signal will have to be absorbed at centre X and should not be able to go through the equipment to arrive at the preceding centre.
Poceed-to-select signal (PTS protocol)		S = 0, continuous repetitions of IA5 character 2/10 maintained until the first class-of-traffic character is recognized and always preceded by at least 2 SYN characters (1/6).	Returned within 3 seconds from the start of the received *calling* signal. The *proceed-to-select* signal shall be followed by the *network* or *service identification* signal within 50 ms of receipt of the first class-of-traffic character, followed by the *waiting* signal if no other signalling characters follow contiguously. The *proceed-to-select* signal will have to be absorbed at centre X and should not be able to go through the equipment to arrive at the preceding centre.

TABLE 1/X.71 *(continued)*

Signal or function	Forward path (X towards Y)	Backward path (Y towards X)	Remarks
Selection signals	S = 0, at least one (first class-of-traffic character only) and possibly several network *selection* signals depending on the network requirement (see Appendix I), the digits of the DNIC of the called network, the digits of the called terminal number, and as end-of-selection character (2/11) and then followed by the *waiting* signal		The *selection* signals are transmitted at the maximum data signalling rate of the links provided. The data country code (DCC) may be omitted on terminal calls at the request of the incoming country. For CC protocol, these signals, preceded by at least two SYN characters, are transmitted immediately after the *calling* signal without awaiting the reception at X of the *call-confirmation* signal. For PTS protocol, these signals, preceded by at least two SYN characters, are transmitted immediately after reception at X of the *proceed-to-select* signal.
Network or service identification signals		S = 0, IA5 character 3/11 followed by the DNIC of the network followed by the *waiting* signal if no other signalling characters follow contiguously	The character 3/11 and DNIC follows the *call confirmation* of *PTS* signal. These signals, preceded by at least two SYN characters (1/6) when they follow a *waiting* signal, must go through centre X and arrive at the originating network.
Waiting signal	S = 0, repetitions of 1s for a period of at least 15 information bits		This signal must be sent if two groups of signalling characters cannot be combined to form one group of contiguous characters.
Reception congestion signal		S = 0, repetitions of 1s for a minimum period of 10 ms or 16 information bits, whichever is the greater in time, and for a maximum period of 24 information bits or 15 ms, whichever is the greater in time, followed by the *clearing* signal	It may be possible that this signal will be preceded by the *call confirmation* signal or a part of it. This signal is returned as soon as possible and the target time will be within 15 ms or 24 information bits of the start of the *calling* signal, whichever is the greater in time, when the *selection* signals cannot be received. This signal should be absorbed by X and not allowed to be received by a preceding centre. This signal should be provided in networks using the CC protocol and may be provided in networks using the PTS protocol.
Call progress signal without clearing (if required)		S = 0, one IA5 character 3/10 and then 2 characters according to Table 7d/X.71, followed by the *waiting* signal if no other signalling characters follow contiguously	These signals are preceded by at least two SYN characters (1/6) when they follow a *waiting* signal. Examples would be redirected-call or *terminal-called call progress* signals, which are followed by a return to the *waiting* signal.

TABLE 1/X.71 *(continued)*

Signal or function	Forward path (X towards Y)	Backward path (Y towards X)	Remarks
Call connected signal		S = 0, one IA5 character, 3/12-15, according to Table 7/X.71, followed by the *waiting* signal if no other signalling characters follow contiguously	See § 2.13 of the text and Appendix III. This signal is preceded by at least two SYN characters (1/6) if it follows a *waiting* signal.
Start-of-transit through-connect signal (STTC)		S = 0, one IA5 character 3/14 according to Table 7/X.71	This signal always precedes the transit through-connect signal.
Transit through-connect signal (TTC)		S = 0, one IA5 character, 3/12-3/15, according to Table 7b/X.71, followed by the waiting signal if no other signalling characters follow contiguously	This signal will always be prefaced by the *start-of-transit through-connect* signal and will be returned preceding a *call progress* signal without clearing when this has to be sent. It will also be transmitted when the calling and/or called line identification is required (for further details see Appendix III). The signal is returned at the same time as the subscriber call set-up procedure is initiated following a positive subscribers state check, e.g. not busy, no loss of synchronization, or when the subscriber is busy and connect when free facility is provided (see Appendix III(B) for example).
Transit centres through-connected signal (TTD)	S = 0, one IA5 character, 3/10 according to Table 6/X.71		This signal is returned by the originating exchange 30-50 ms following receipt of the *transit through-connect* signal. The signal is omitted and replaced by the calling line identity if it is requested.
Called line identification signal (if applicable)		S = 0, *called line identification* signal transmitted between 0 and 30 ms after the transit *centres through-connected* signal or the first character of the calling line identification is received	The *called line identification* signal consists of the DNIC followed by the digits of the network terminal number and then the end-of-identification character (3/11) (see § 2.12 of the text and Appendix III). If the called line identification is requested and is not available within the network receiving the request, a dummy identification consisting of the end-identification character (3/11) only is transmitted. This signal is preceded by at least two SYN characters (1/6) when it follows a *waiting* signal.

TABLE 1/X.71 *(continued)*

Signal or function	Forward path (X towards Y)	Backward path (Y towards X)	Remarks
Calling line identification signal (if applicable)	S = 0, *calling line identification* signal transmitted between 30 and 50 ms after the *transit through-connect* signal is received		The *calling line identification* signal consists of the DNIC followed by the digits of the network terminal number and then the end-of-identification character (3/11). If the calling line identification is requested and is not available within the network receiving the request, the DNIC followed by character 3/11 shall be transmitted. See Appendix III.
Terminating through-connection signal		Continuous repetitions of 1s (S = 1) from the called DTE received by the originating centre	This signal confirms through-connection in both directions of transmission in the destination centre (see § 2.13 of the text and Appendix III).
Originating through-connection signal	Continuous repetitions of 1s (S = 1) received by the called DTE from the calling DTE		This signal confirms through-connection in both directions of transmission in the originating centre (see § 2.13 of the text and Appendix III).
Call progress signals with clearing		S = 0, at least two SYN characters (1/6) followed by character 3/10 followed by 2 digits (see Table 7d/X.71) followed by the *clearing* signal	These signals are preceded by at least two SYN characters (1/6) when they follow a *waiting* signal.
Clearing signal	Continuous repetitions of 0s (S = 0) in the direction of clearing. The minimum recognition time is 16 bits and the maximum time is 60 ms		The minimum period of one signalling path which itself ensures the complete release of the connection is 60 ms.
Clear confirmation signal	Continuous repetitions of 0s (S = 0) in the opposite direction to clearing within 60 ms after reception of the *clearing* signal		The minimum and maximum periods for the release of the international circuit by a centre are 16 bits and 60 ms respectively.
Incoming guard delay	Period of 60-70 ms measured from the moment when continuous 0s (S = 0) has been established on both signalling paths by: – either recognizing or transmitting the *clearing* signal on one signalling path, and – either transmitting or recognizing the clear *confirmation* signal on the other signalling path		A new incoming call shall not be accepted until this guard period has elapsed.
Outgoing guard delay	Period of 130 ms measured from the moment when continuous 0s (S = 0) has been established on both signalling paths by: – either recognizing or transmitting the *clearing* signal on one signalling path, and – either transmitting or recognizing the *clear confirmation* signal on the other signalling path		A new outgoing call shall not be originated until this guard period has elaspsed.

TABLE 1/X.71 *(concluded)*

Signal or function	Forward path (X towards Y)	Backward path (Y towards X)	Remarks
Automatic retest signal	S = 0, continuous repetitions of 1s for a period of 4 seconds followed by continuous repetitions of 0s for a period of 56 seconds and the signal sequence is then repeated		See § 2.16 of the text.
Backward busy signal		S = 0, continuous repetitions of 1s for a maximum period of 5 minutes	

[a] The duration of the *calling* signal and return of the *call confirmation* signal is for further study in the light of experience.

Note 1 – The status bit may be OFF (= 0) or ON (= 1).

Note 2 – For call confirmation (CC) protocol and proceed-to-select (PTS) protocol see § 2.1.

2.5 *Selection* signals can be divided into two parts. The first part, designated the *network selection* signals, contains information regarding network and user requirements and may be composed of one to nine (or possibly more) characters (see Tables 2/X.71, 3/X.71, 3a/X.71, 4/X.71, 4a/X.71, 5/X.71 and 5a/X.71). The second part comprises the *address* signals (the called national terminal number which is preceded by the DNIC always in the case of a transit call and also for terminal calls unless the incoming destination country requests omission of the data country code portion, see Tables 6/X.71 and 6a/X.71).

The *network selection* signals used in the forward direction (see also Appendix II are further subdivided and assembled as follows (see §§ 2.5.1 to 2.5.4 below) for signalling purposes.

Note that the term "user class of service" is abbreviated in the following sections to "user class".

2.5.1 *First class-of-traffic character* (see Table 2/X.71)

The *calling* signal is always followed by at least one class-of-traffic character in addition to at least 2 SYN characters. The bit functions of the class-of-traffic character were chosen so that no further characters would be needed for most connections.

If there is a need for indication of further requirements, a second class-of-traffic character (see § 2.5.3 below) may be used. Whether the second class-of-traffic or user class characters follow or not, will be indicated by the bits b_3 and b_4 of the first class-of-traffic character.

2.5.2 *User class character (indication of speed and code)* (see Tables 3/X.71 and 3a/X.71)

This character, if used, will follow the first class-of-traffic character and will be required when, for example, this information cannot be derived from the incoming line.

As eight user classes in Table 3/X.71 are not sufficient, a second user class character may be added by means of an escape character. Whether a second user class character follows or not will be indicated by the bits b_1, b_2 and b_3 of the first user class character. Whether a second class-of-traffic character follows or not will be indicated by bit b_4 of the first user class character.

2.5.3 *Second and further class-of-traffic characters* (see Tables 4/X.71 and 4a/X.71)

These characters follow either the first class-of-traffic character or any user class characters required. The number of these class-of-traffic characters depends on the number of user facilities available.

The bit b_4 of the second or subsequent class-of-traffic characters indicate whether another class-of-traffic character follows or not.

TABLE 2/X.71

First class-of-traffic character [a)]

First four bits of character				Condition signalled from X to Y
b_4	b_3	b_2	b_1	
0	0			No further network selection signal follows
0	1			Second class-of-traffic character follows (Table 4/X.71)
1	0			User class character follows (Table 3/X.71)
		0		Alternative routing not allowed
		1		Alternative routing allowed
			0	Transit traffic
			1	Terminal traffic
1	1	0	0	Not allocated
1	1	0	1	
1	1	1	0	
1	1	1	1	

[a)] All characters are in column 3 ($b_5 = 1$, $b_6 = 1$, $b_7 = 0$) of International Alphabet No. 5. The eighth bit (b_8) is chosen to give odd parity over the character.

TABLE 3/X.71

First user class character [a)]

First four bits of character				Condition signalled from X to Y [b)]
b_4	b_3	b_2	b_1	
0				No second class-of-traffic character follows
1				A second class-of-traffic character follows (Table 4/X.71)
	0	0	0	Synchronous classes derived from line
	0	0	1	300 bit/s (user class 1)
	0	1	0	50 bit/s (user class 2)
	0	1	1	100 bit/s (user class 2)
	1	0	0	110 bit/s (user class 2)
	1	0	1	134.5 bit/s (user class 2)
	1	1	0	200 bit/s (user class 2)
	1	1	1	A second user class character follows (Table 3a/X.71)

[a)] All characters are in column 3 ($b_5 = 1$, $b_6 = 1$, $b_7 = 0$) of International Alphabet No. 5. The eighth bit (b_8) is chosen to give odd parity over the character.

[b)] The user class character(s) may be omitted if, for example, the information can be derived from the incoming line.

TABLE 3a/X.71

Second user class character [a]

First four bits of character				Condition signalled from X to Y [b]
b_4	b_3	b_2	b_1	
0	0	0	0	600 bit/s (user class 3)
0	0	0	1	2 400 bit/s (user class 4)
0	0	1	0	4 800 bit/s (user class 5)
0	0	1	1	9 600 bit/s (user class 6)
0	1	0	0	48 000 bit/s (user class 7)
0	1	0	1	Service (50 bit/s)
0	1	1	0	Telex (50 bit/s)
0	1	1	1	Gentex (50 bit/s)
1	0	0	0	TWX
1	0	0	1	Teletex (2400 bit/s)
1	0	1	0	Not allocated
1	0	1	1	
1	1	0	0	
1	1	0	1	
1	1	1	0	
1	1	1	1	

[a] All characters are in column 3 ($b_5 = 1$, $b_6 = 1$, $b_7 = 0$) of International Alphabet No. 5. The eighth bit (b_8) is chosen to give odd parity over the character.

[b] The user class character(s) may be omitted if, for example, the information can be derived from the incoming line.

TABLE 4/X.71

Second class-of-traffic character [a]

First four bits of character				Condition signalled from X to Y
b_4	b_3	b_2	b_1	
0				No third class-of-traffic character follows
1				Third class-of-traffic character follows (Table 4a/X.71)
	0			No closed user group sequence follows
	1			Closed user group sequence follows (Table 5/X.71)
		0		Called line identification not required
		1		Called line identification required
			0 1	Reserved for national use [b]

[a] All characters are in column 3 ($b_5 = 1$, $b_6 = 1$, $b_7 = 0$) of International Alphabet No. 5. The eighth bit (b_8) is chosen to give odd parity over the character.

[b] On international circuits bit b_1 should be set to zero.

TABLE 4a/X.71

Third class-of-traffic character [a]

First four bits of character				Condition signalled from X to Y
b_4	b_3	b_2	b_1	
0				No fourth class-of-traffic character follows
1				Fourth class-of-traffic character follows [c]
	0			Redirection not allowed [d]
	1			Redirection allowed [d]
		0		Not multiple address call [d]
		1		Multiple address call [d]
			0 1	Not allocated [b]

[a] All characters are in column 3 ($b_5 = 1$, $b_6 = 1$, $b_7 = 0$) of International Alphabet No. 5. The eighth bit (b_8) is chosen to give odd parity over the character.

[b] On international circuits bit b_1 should be set to zero.

[c] Reserved for future needs.

[d] The international use of this signal requires further study.

2.5.4 *Closed user group characters* (see Tables 5/X.71 and 5a/X.71)

These characters are only used in conjunction with the second and possibly subsequent class-of-traffic characters which may follow.

The start of closed user group character shall precede the closed user group number which should be coded into a number of hexadecimal characters up to a maximum of four (see Table 5/X.71).

2.5.5 The numerical characters used for the second part of the *selection* signals are shown in Tables 6/X.71 and 6a/X.71. When the first class-of-traffic character indicates a terminal call, the incoming country can adopt the option not to receive the data country code portion of the DNIC.

TABLE 5/X.71

Start of closed user group character [a] [b]

Combination				Condition signalled from X to Y
b_4	b_3	b_2	b_1	
0				Without outgoing access
1				With outgoing access
	0			No DNIC [c] follows
	1			DNIC follows [c]
		0	0	1 Number of hexadecimal closed user group characters which follow
		0	1	2
		1	0	3
		1	1	4

[a] All characters are in column 3 ($b_5 = 1$, $b_6 = 1$, $b_7 = 0$) of International Alphabet No. 5. The eighth bit (b_8) is chosen to give odd parity over the character.

[b] The start of closed user group character shall precede the DNIC of the representative user, followed by the closed user group number which should be coded into a number of hexadecimal characters up to a maximum of four, as indicated. The closed user group number shall be transmitted with the least significant bit of the least significant character first.

[c] On international circuits bit b_3 should be set to 1.

TABLE 5a/X.71

Closed user group characters [a]

Combination				Condition signalled from X to Y
b_4	b_3	b_2	b_1	
0	0	0	0	0 Hexadecimal closed user group character
0	0	0	1	1
0	0	1	0	2
0	0	1	1	3
0	1	0	0	4
0	1	0	1	5
0	1	1	0	6
0	1	1	1	7
1	0	0	0	8
1	0	0	1	9
1	0	1	0	A
1	0	1	1	B
1	1	0	0	C
1	1	0	1	D
1	1	1	0	E
1	1	1	1	F

[a] All characters are in column 3 ($b_5 = 1$, $b_6 = 1$, $b_7 = 0$) of International Alphabet No. 5. The eighth bit (b_8) is chosen to give odd parity over the character.

TABLE 6/X.71

Miscellaneous forward path signals [a]

First four bits of character				Condition signalled from X to Y	
b_4	b_3	b_2	b_1		
0	0	0	0	0	Digits for: – data network identification code (DNIC) – called network terminal number – calling line identification signal
0	0	0	1	1	
0	0	1	0	2	
0	0	1	1	3	
0	1	0	0	4	
0	1	0	1	5	
0	1	1	0	6	
0	1	1	1	7	
1	0	0	0	8	
1	0	0	1	9	
1	0	1	0	Transit centres through-connected (TTD)	
1	0	1	1	End-of-calling line identification signal [b]	
1	1	0	0	Not allocated	
1	1	0	1		
1	1	1	0		
1	1	1	1		

[a] All characters comprising these signals are in column 3 (b_5 = 1, b_6 = 1, b_7 = 0) of International Alphabet No. 5. The eighth bit (b_8) is chosen to give odd parity over the character.

[b] This signal follows the DNIC when the calling line identification is not available (see § 2.12).

TABLE 6a/X.71

Other forward path signals (with odd parity)

IA5-character	Condition signalled from X to Y
1/6	SYN
2/11	End-of-selection
2/15	Start of extended address

2.6 The incoming equipment should maintain continuous 0s on the backward signalling path if the received character is spurious as indicated by a character other than continuous 1s (calling signal). This procedure provides a safeguard against false calls.

In the case of receipt of a spurious signal as indicated by a parity error or by a character other than a *selection* signal (with the possible exception of SYN characters), the incoming equipment should return the *call progress* signal No. 20 to the preceding centre immediately followed by the *clearing* signal after the *call confirmation* or *proceed-to-select* signal and the *network* or *service identification* signals.

The incoming equipment may release the connection if all of the *selection* signals are not correctly received within a period of 2 seconds from the recognition of the *calling* signal for the CC Protocol or from the start of transmission of the *proceed-to-select* signal for the PTS Protocol. In this event, the *call progress* signal No. 20 is returned to the preceding centre immediately followed by the *clearing* signal after the *call confirmation* or *proceed-to-select* signal and the *network* or *service identification* signals.

2.7 The international data number may have a maximum number of 14 digits comprising the 4 digit data network identification code and a 10 digit maximum network terminal number. Alternatively, the 14 digits can be considered as the 3 digit data country code followed by a national number of 11 digits maximum (see Recommendation X.121).

The possible address extension may either be included in the 14 digit international data number or may be separated from the international data number by a start-of-extension-address signal (2/15). In that case the extended address consists of up to 40 decimal digits. The network shall not be required to look at or operate on a network extension address. However some networks may look at the network address extension if they wish.

2.8 In the case of receipt of the *reception congestion* signal at a transit centre, the *call progress* signal No. 61 should be returned to the preceding centre (after the *call confirmation* or *proceed-to-select* signal, *network* or *service identification* signal) followed by the *clearing* signal.

2.9 The *network* or *service identification* signals shall be sent following the *call confirmation* or *proceed-to-select* signal in all cases. In all cases the country or network identity shall consist of four decimal digits. The value of the fourth digit should, in the case when it is not specifically defined by the numbering plan, be at the discretion of the country in question within the limits allowed by the numbering plan.

If several transit networks are involved in setting up a call the calling network will receive the network identifications one after the other. If a transit centre fails to receive the first character of the *network* or *service identification* signals, within two seconds of the *call confirmation* signal, it will return to the preceding centre, the *call progress* signal No. 20 (after the *call confirmation* or *proceed-to-select* signal and the *network* or *service identification* signal followed by the *clearing* signal).

The *network* or *service identification* signals could be useful for retracing the route followed by a call (for traffic statistics, international accounts, analysis of unsuccessful calls and the clearing of faults).

It is possible for a transit centre to receive backward path signals such as *network* or *service identification* signals, *call connected* signal or *call progress* signals from subsequent centres, while the backward path signals originated locally are still being sent. It is necessary for the transit centre to ensure that the received signals are passed to the preceding centre without mutilations or loss.

2.10 The backward path signals indicating effective and ineffective call conditions are scheduled in Tables 7/X.71, 7a/X.71, 7b/X.71, 7c/X.71 and 7d/X.71.

2.11 If the *call progress, call connected* or alternatively *terminating through-connection* signals are not received within 15 seconds from the end of selection, then the *call progress* signal No. 20 will be returned to the preceding centre (after the *call confirmation* or *proceed-to-select* signal, *network* or *service identification* signal), followed by the *clearing* signal. The further action to be taken in the case of reception of *call progress* signals without clear is for further study.

2.12 In this type of signalling, originating and terminating national centres contain the identification of the calling or called subscribers respectively. These identifications may be exchanged within the network as an optional subscribers' feature.

If the called line identification has been requested but is not available, the terminating centre in the connection should send only the *end-of-line identification* signal (3/11).

If the calling line identification has been requested but is not available, the originating centre should send only the DNIC followed by the *end-of-line identification* signal (3/11).

2.13 The *call-connected* signal confirms that the call is accepted by the called subscriber and, if applicable, the calling line identification has been completely received by the terminating centre and passed to the called subscriber, and when applicable that the called line identification has been completely transmitted to the originating centre (see Appendix III).

The *terminating through-connection* signal confirms (by change of status bit from 0 to 1) that through-connection in both directions of transmission has been effected at the terminating exchange (see Appendix III).

The originating *through-connection* signal confirms that the *call connected* signal has been received by the originating centre and when applicable that the called line identification has been completely received by the originating centre and passed to the calling subscriber (see Appendix III).

The *call connected* signal is sent on the backward path by the terminating centre. The originating through-connection signal (change of status bit from 0 to 1) is sent by the originating centre both to the calling and called subscribers.

TABLE 7/X.71

Miscellaneous backward path signals [a)]

First four bits of character				Condition signalled from Y to X
b_4	b_3	b_2	b_1	
0	0	0	0	0 Digits for: – network or service identification signals – called line identification signal – call progress signal
0	0	0	1	1
0	0	1	0	2
0	0	1	1	3
0	1	0	0	4
0	1	0	1	5
0	1	1	0	6
0	1	1	1	7
1	0	0	0	8
1	0	0	1	9
1	0	1	0	Start of call progress signal (see Table 7d/X.71)
1	0	1	1	End-of-called-line identification signal [b)] Start of network or service identification signal
1	1	0		Call connected signal
			0	Call metering
			1	No call metering
1	1	1	0	Start of transit through-connect signal (STTC) [c)]
1	1	1	1	Further backward path signal follows (see Table 7a/X.71)

[a)] All characters are in column 3 ($b_5 = 1$, $b_6 = 1$, $b_7 = 0$) of International Alphabet No. 5. The eighth bit (b_8) is chosen to give odd parity over the character.

[b)] This signal is also used alone when the called line identification is not available.

[c)] This signal always precedes the *transit through-connect* signals detailed in Table 7b/X.71.

TABLE 7a/X.71

Further miscellaneous backward path signals [a] [b]

First four bits of character				Condition signalled from Y to X
b_4	b_3	b_2	b_1	
0				Reserved for national use
1				Reserved for national use
	0	0	0	Not allocated
	0	0	1	
	0	1	0	
	0	1	1	
	1	0	0	
	1	0	1	
	1	1	0	
	1	1	1	

[a] All characters are in column 3 (b_5 = 1, b_6 = 1, b_7 = 0) of International Alphabet No. 5. The eighth bit (b_8) is chosen to give odd parity over the character.

[b] These signals follow combination 1111 in Table 7/X.71.

TABLE 7b/X.71

Transit through-connect signal [a) b)]

First four bits of character				Condition signalled from Y to X
b_4	b_3	b_2	b_1	
0	0	0	0	Not allocated
0	0	0	1	
0	0	1	0	
0	0	1	1	
0	1	0	0	
0	1	0	1	
0	1	1	0	
0	1	1	1	
1	0	0	0	
1	0	0	1	
1	0	1	0	
1	0	1	1	
1	1			Transit through-connect (TTC)
		0		Calling line identification not required
		1		Calling line identification required
			0	Call metering
			1	No call metering

[a)] All characters are in column 3 ($b_5 = 1$, $b_6 = 1$, $b_7 = 0$) of International Alphabet No. 5. The eighth bit (b_8) is chosen to give odd parity over the character.

[b)] These signals follow the start of *transit through-connect* signal in Table 7/X.71.

TABLE 7c/X.71

Other backward path signals (with odd parity)

IA5-character	Condition signalled from Y to X
1/6	SYN
2/10	Call confirmation or proceed-to-select

TABLE 7d/X.71

Call progress signals [a) g)]

Numerical code first/second digit	Category	Significance
01	Without clearing	Terminal called
02		Redirected call
03		Connect when free
20	With clearing, due to short term condition [b)]	Network failure [f)]
21		Number busy
22		[c)]
23		[c)]
41	With clearing, due to long term condition [b)]	Access barred
42		Changed number
43		Not obtainable
44		Out of order
45		Controlled not ready
46		Uncontrolled not ready
47		DCE power off
48		[c)]
49		Network fault in local loop
51		Call information service
52		Incompatible user class of service
61	With clearing, due to network short term conditions [b)]	Network congestion
71	With clearing, due to network long term conditions [b)]	Degraded service
72		[e)]
81	With clearing, due to DTE-network procedure	Registration/cancellation confirmed [d)]
82		[c)]
83		[c)]

[a)] All characters comprising these signals are in column 3 ($b_5 = 1$, $b_6 = 1$, $b_7 = 0$) of International Alphabet No. 5. The eighth bit (b_8) is chosen to give odd parity over the character.

[b)] "Short term" in this context approximates to the holding time of a call whilst "long term" implies a condition that can persist for some hours or even days.

[c)] These signals are only utilized between the first exchange and the subscriber and are not signalled on inter-network links.

[d)] Not yet included. To be studied in relation to Recommendation X.300 on network call control procedures.

[e)] Only utilized within national networks.

[f)] At the originating exchange, this results in sending a *call progress* signal "no connection" to the calling customer, and clearing the call.

[g)] A *call progress* signal without clearing should precede the *called line identification* signal. A *call progress* signal with clearing could precede or follow the *called line identification* signal.

2.14 If the terminating centre fails to receive the *transit centres through-connected* signal (TTD) or, if applicable, the first character of the *calling line identification* signals within 4 seconds after having sent the *transit through-connect* signal (TTC), it will return to the preceding centre the *call progress* signal No. 20 followed by the *clearing* signal.

2.15 If the originating centre fails to receive the *terminating through-connect* signal within 10 s after reception of the TTC or CC signal (whichever occurs first) the *call progress* signal No. 20 will be sent to the calling terminal followed by the *clearing* signal. Upon receipt of a *call progress* signal without clearing within the time limit, the timer should be reset with a new time-out period specified in accordance with that in Recommendation X.21 (T3B in Table C-1/X.21). The timer should be stopped upon receiving the terminating through-connection signal.

Note – The time-out supervision for the *connect-when-free/waiting allowed* facility is for further study.

2.16 The guard delays on clearing are measured from the moment when continuous 0s (S = 0) has been established on both signalling paths by:

- either recognizing or transmitting the *clearing* signal on one signalling path, and
- either transmitting or recognizing the *clear confirmation* signal on the other signalling path.

For incoming calls this guard period shall be 60-70 ms.

A new incoming call shall not be accepted until this guard period has elapsed. This is on the assumption that the terminating centre will be able to send the *call confirmation* signal after a negligible period from receipt of the *calling* signal.

The guard period on clearing for outgoing calls should be a period of at least 130 ms. A new outgoing call shall not be originated until this guard period has elapsed.

If centres are able to distinguish between the different clearing conditions, shorter periods may be introduced accordingly.

2.17 The automatic *retest* signal will be initiated, as indicated in § 2.4.

This signal transmitted over the forward signalling path is composed of a maximum of five successive cycles, each cycle incorporating:

S = 0, continuous repetitions of 1s for a period of 4 seconds, followed by:

S = 0, continuous repetitions of 0s for a period of 56 seconds.

The circuit should be marked "unavailable" for outgoing traffic and tested up to 5 times at nominal intervals of one minute, and a check made to confirm the receipt of the *call confirmation* or *proceed-to-select* signal on the backward path in response to each test. If the *call confirmation* or *proceed-to-select* signal has not been received at the end of this first group of tests, the retest will continue with a further group of up to 5 tests at either 5- or 30-minute nominal intervals. If 5-minute intervals are used and the *call confirmation* or *proceed-to-select* signal has not been received at the end of this second group of tests, further retests will be made at 30-minute intervals. An alarm will be given at an appropriate time. However, this retest procedure may be discontinued at any stage at the discretion of the outgoing Administration.

If, however, during the above sequence of retests, the *call confirmation* or *proceed-to-select* signal is received, a *clearing* signal will be transmitted in place of the *retest* signal. Following a valid *clear confirmation* signal, the incoming and the outgoing sides of the trunk circuit should not be returned to service until after expiry of the appropriate guard delay time.

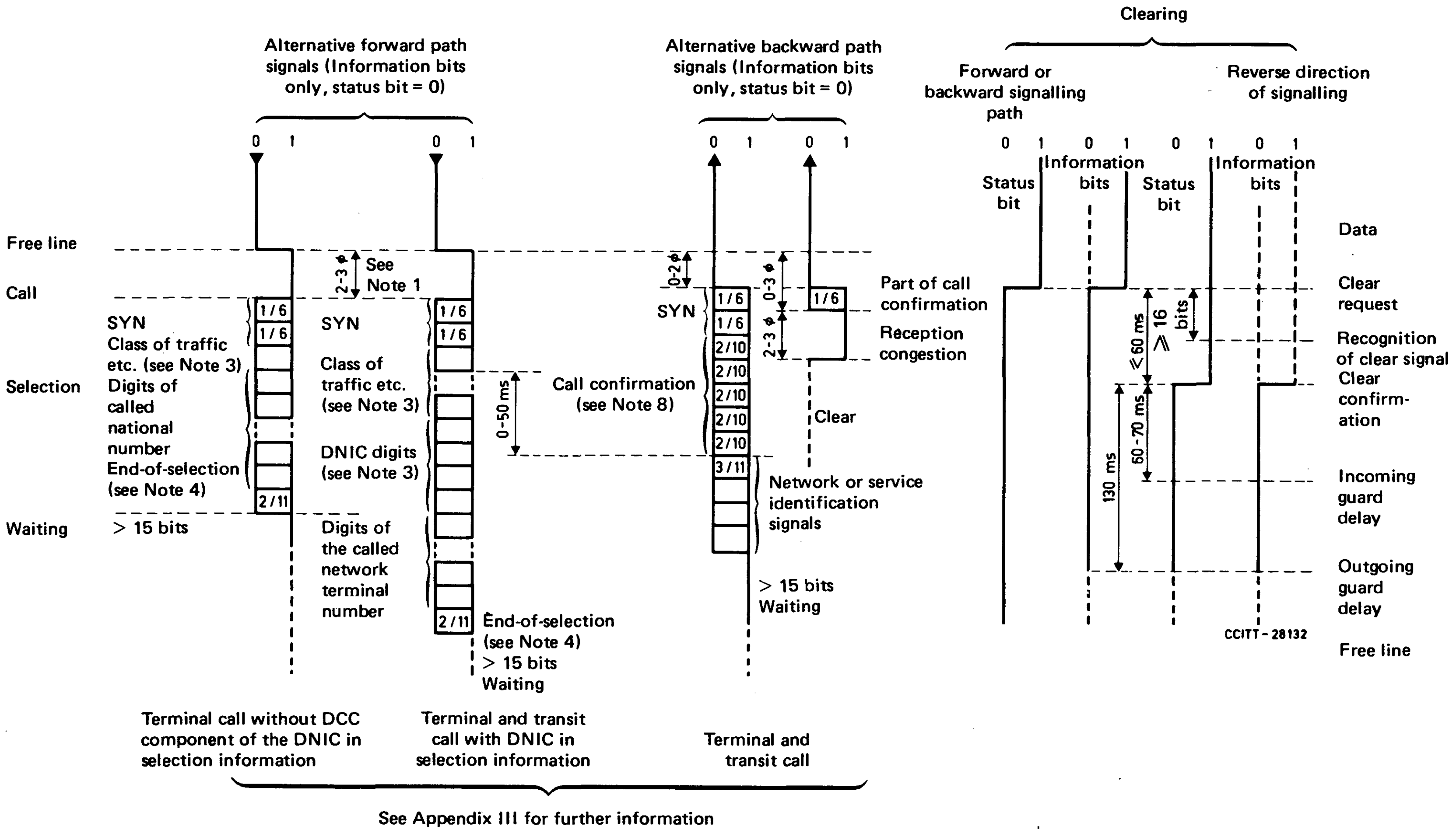
In order to cater for the possibility that a faulty circuit may be seized at both ends, the automatic retest equipment should be arranged to allow an incoming call to be received during continuous repetitions of 0s (S = 0). Administrations may, however, ignore such calls which occur during the incoming guard delay period.

The interval between the tests at the two ends of the trunk circuit should be made different by increasing the nominal interval by 20% at one end, to be sure that successive retests do not overlap at both ends. In general, the intercontinental transit centre having the higher DNIC should take the longer interval (i.e. 1.2, 6 and 36 minutes). Nevertheless, when this requirement would entail considerable difficulty, alternative arrangements may be adopted by agreement between the two Administrations or RPOAs concerned.

Where an exchange has knowledge of a transmission system failure, it is desirable that the *retest* signals shall not be applied to the circuits affected.

In order to avoid simultaneous seizure of too many registers at the distant centre, it is advisable that the *retest* signals, which may be sent simultaneously on various circuits subjected to this test, should be out of phase with one another.

2.18 If at the receiving end parity does not check, provisionally the connection should be cleared down unless otherwise specified. However, the possibility of different actions remains open for further study.



Note 1 – Timings may be shown in ms or in periods of information bits. The symbol ø indicates that the interval may be in multiples of 8 information bits or 5 ms whichever is greatest in time.

Note 2 – *Forward path* signals may also appear on the backward path, indicating a head-on collision on both-way circuits.

Note 3 – *Network selection* signals (class-of-traffic, user class characters, etc.). See Tables 2-5/X.71. DNIC comprises 4 digits.

Note 4 – *Selection* signals will be sent by the originating network in a single block always with an *end-of-selection* signal.

Note 5 – The *network identification* signal comprises character 3/11 followed by the DNIC of the network concerned.

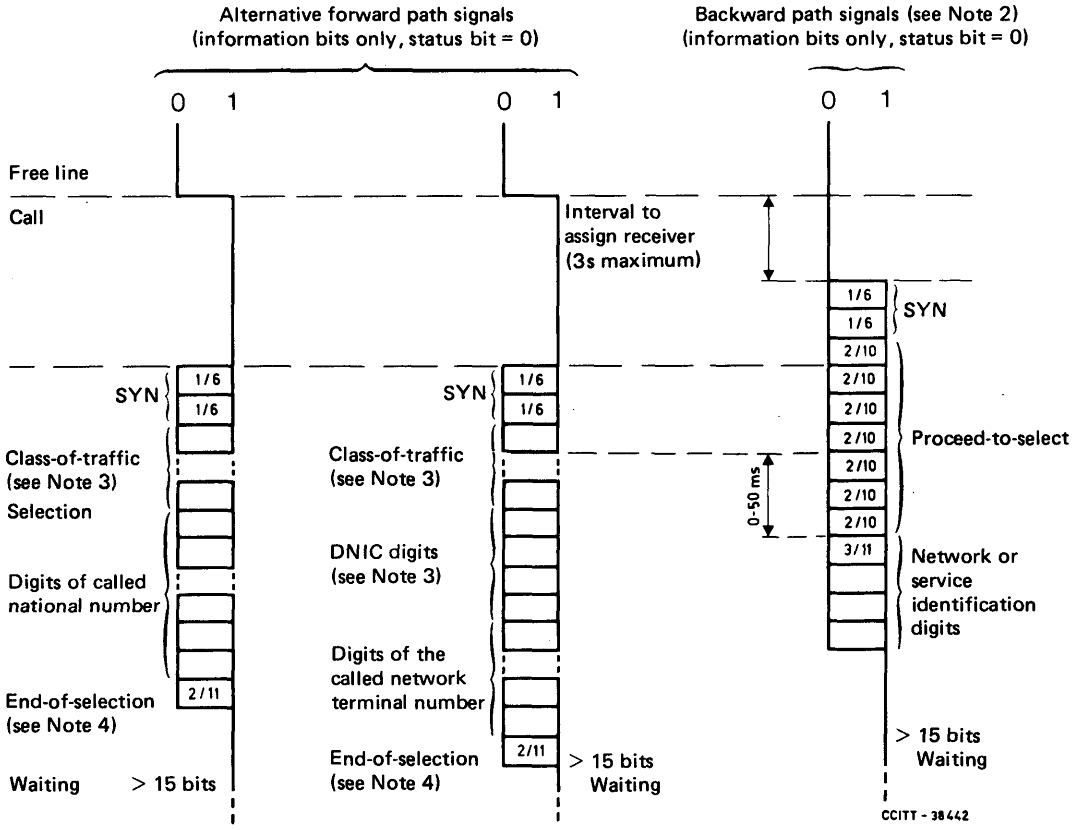
Note 6 – For further details on *call-connected* and *through-connection* signals and their timings see § 2.13 and Appendix III.

Note 7 – All characters shown are from the International Alphabet No. 5.

Note 8 – As an interim arrangement at the discretion of the incoming network, the transmission of the selection signals may be delayed until a *proceed-to-select* signal has been received. In this case the characters shown for the *call confirmation* signal will be used for the *proceed-to-select* signal.

FIGURE 1/X.71

Decentralized signalling between data networks of the synchronous type



Terminal call without DCC component of the DNIC in selection information

Terminal and transit call with DNIC in selection information

Terminal and transit call with DNIC in selection information

Note – Where reference is made, these are the Notes of Figure 1/X.71.

FIGURE 1a/X.71

Initial phases of calls when the proceed-to-select protocol is employed

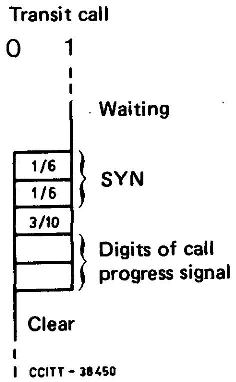


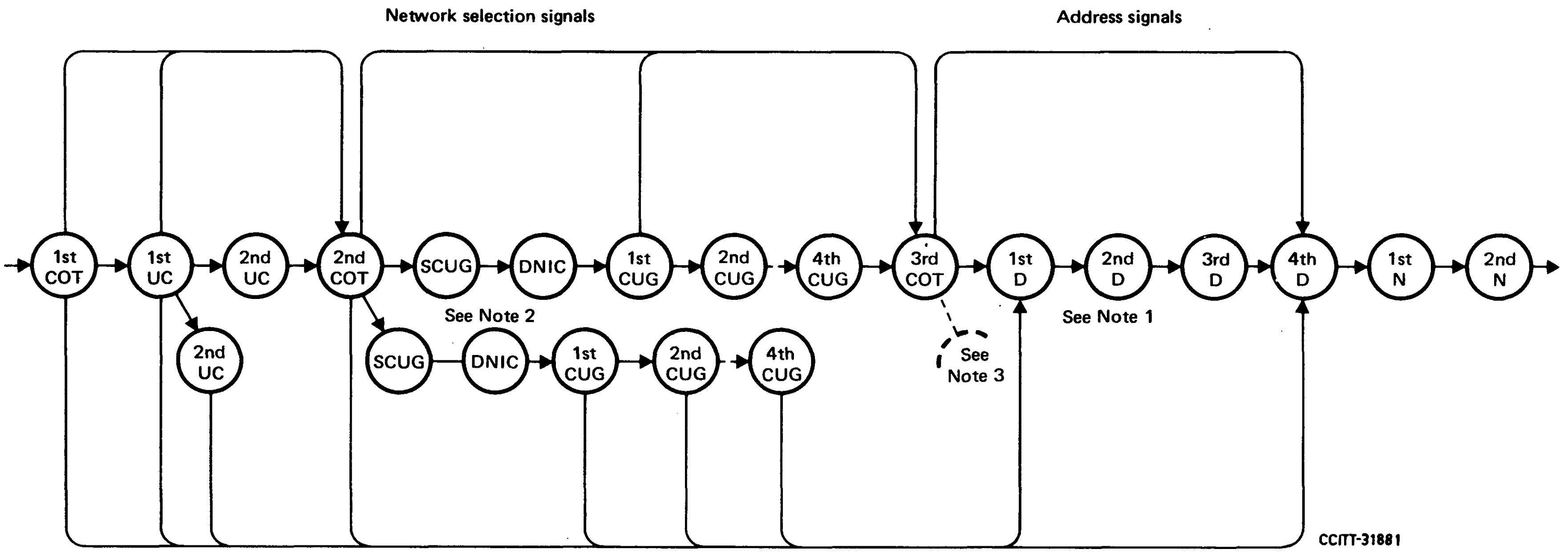
FIGURE 2/X.71

Call progress signal for a transit call

APPENDIX I

(to Recommendation X.71)

Possible sequences of network selection signals



COT	Class-of-traffic character
UC	User class character
CUG	Closed user group
D	Data network (or service) identification code digit
N	Called number digit
SCUG	Start of closed user group character
DNIC	Data network identification code

Note 1 – The first three D digits form the data country code (DCC) component of the data network identification code (DNIC). The fourth D digit is the network or service digit of the DNIC.

Note 2 – The DNIC comprises four digits as defined in Note 1.

Note 3 – Reserved for future extension.

APPENDIX II

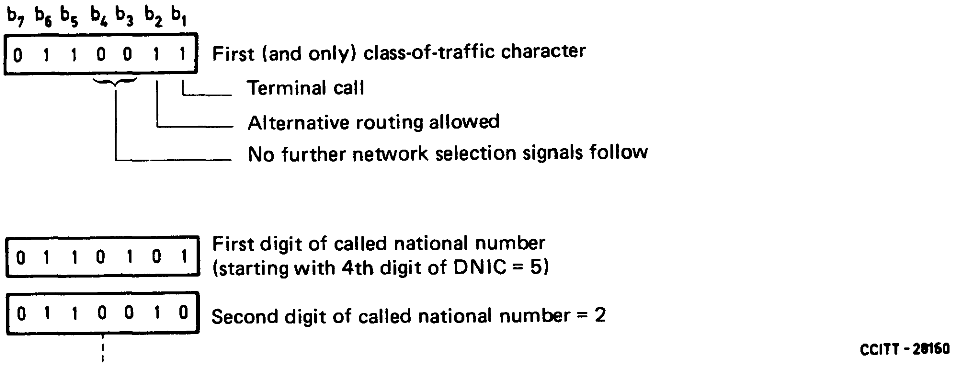
(to Recommendation X.71)

Examples of network selection signals

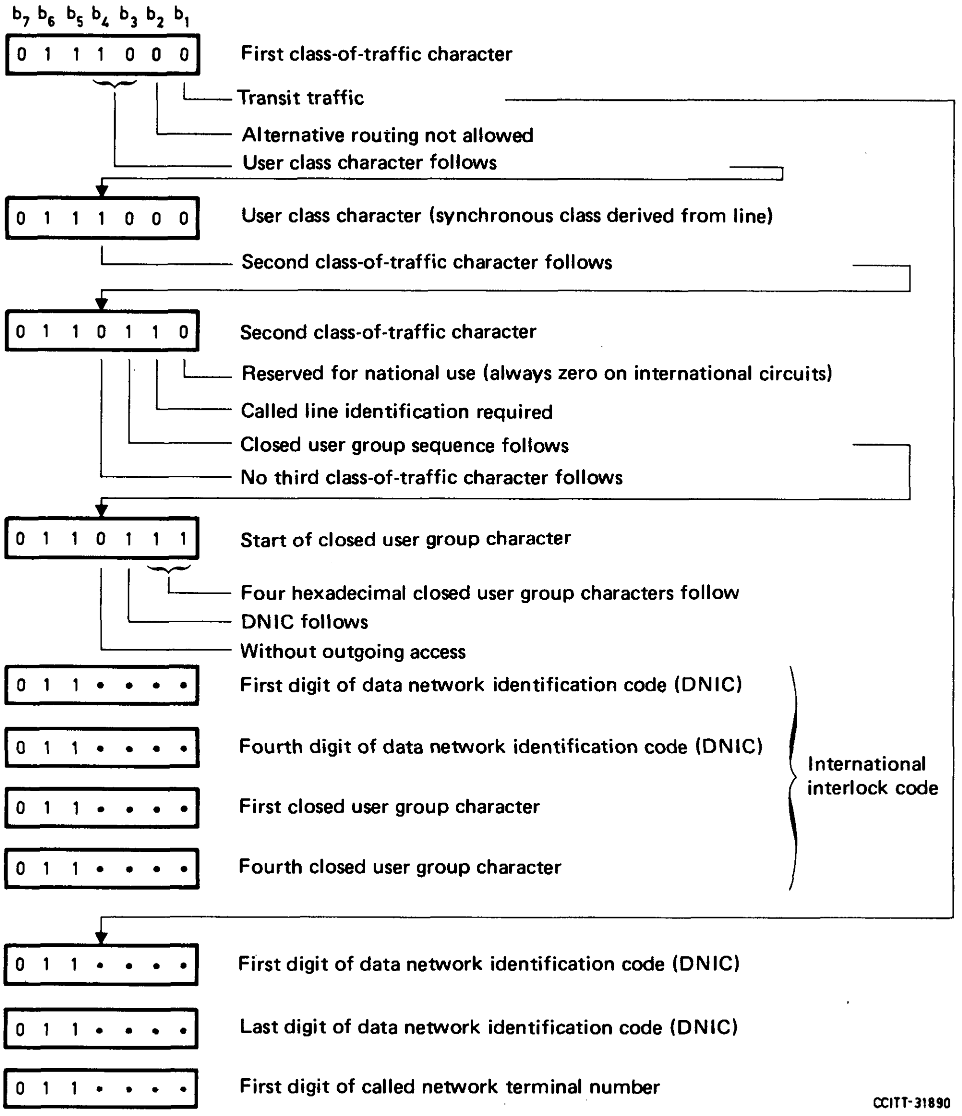
II.1 *First example* (minimum sequence of network selection signals)

This example shows a sequence of minimal length. (The remaining bits in each complete envelope and the preceding calling signal are not shown. The bits are shown in the order of b_7, b_6, b_5, b_4, b_3, b_2, b_1.)

In this example the country of destination has indicated that it does not wish to receive the DCC component of the DNIC.



II.2 *Second example* (a sequence of network selection including closed user group characters)

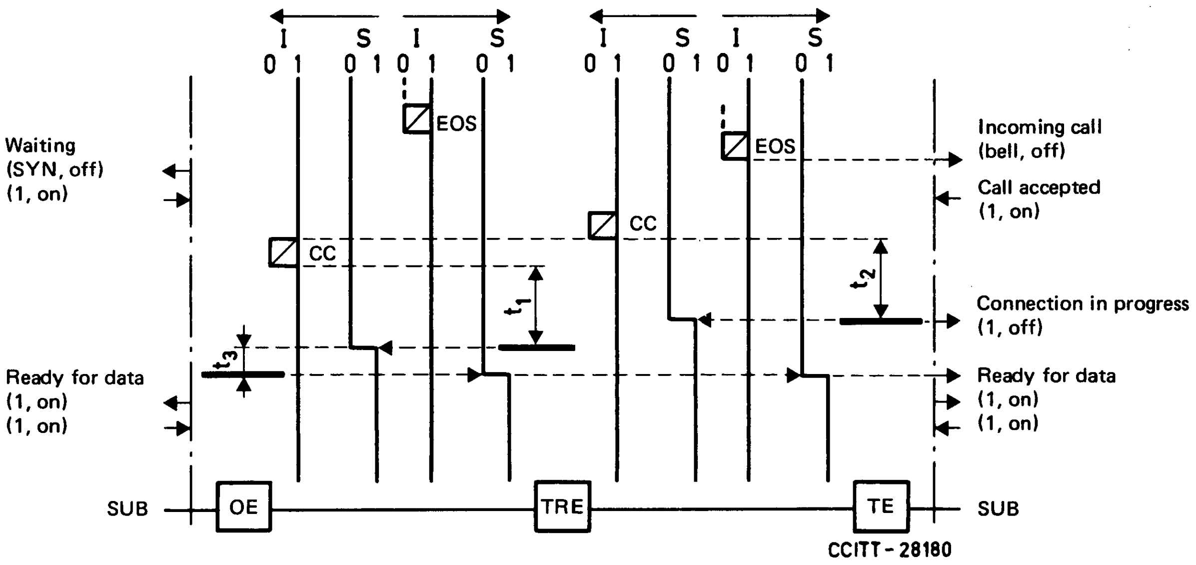


APPENDIX III (A)

(to Recommendation X.71)

Through-connection procedure

Called and calling line identification not required (No *connect-when-free* facility)



S	Status bit	CC	Call connected signal
– – – –	Correlation line	SUB	Subscriber
———	Through-connection	OE	Originating exchange
		TRE	Transit exchange
☒	IA5-character	TE	Terminating exchange
		I	Information bit
EOS	End-of-selection signal		

Note 1 – Where groups of signalling characters are not contiguous, the *waiting* signal (S = 0, repetitions of 1s for a period of at least 15 information bits) must be sent during the interim period.

Note 2 – $t_1 = 0\text{-}30$ ms, $t_2 = 0\text{-}40$ ms, $t_3 = 0\text{-}40$ ms.

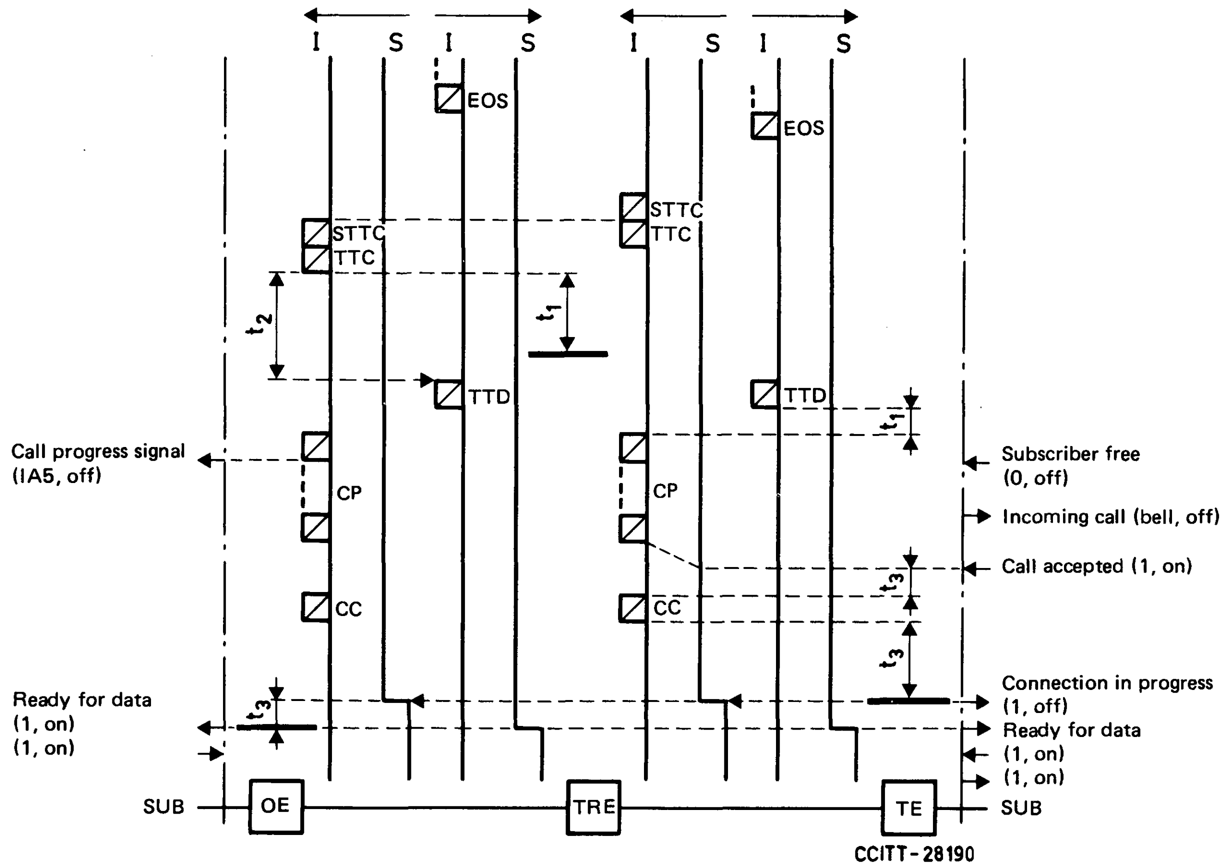
Note 3 – The timings given in Note 2 cover worst case conditions and exchange design should aim to keep them as short as possible.

APPENDIX III (B)

(to Recommendation X.71)

Through-connection procedure

Called and calling line identification not required (*Connect-when-free* facility, subscriber is busy)



S	Status bit
I	Information bit
– – – –	Correlation line
———	Through-connection
▨	IA5-character
EOS	End-of-selection signal
STTC	Start-of-transit through-connect signal

TTC	Transit through-connect signal
TTD	Transit centres through-connected signal
CP	Call progress signal
CC	Call connected signal
SUB	Subscriber
OE	Originating exchange
TRE	Transit exchange
TE	Terminating exchange

Note 1 – Where groups of signalling characters are not contiguous, the *waiting* signal (S = 0, repetitions of 1s for a period of at least 15 information bits) must be sent during the interim period.

Note 2 – $t_1 = 0\text{-}30$ ms, $t_2 = 30\text{-}50$ ms, $t_3 = 0\text{-}40$ ms.

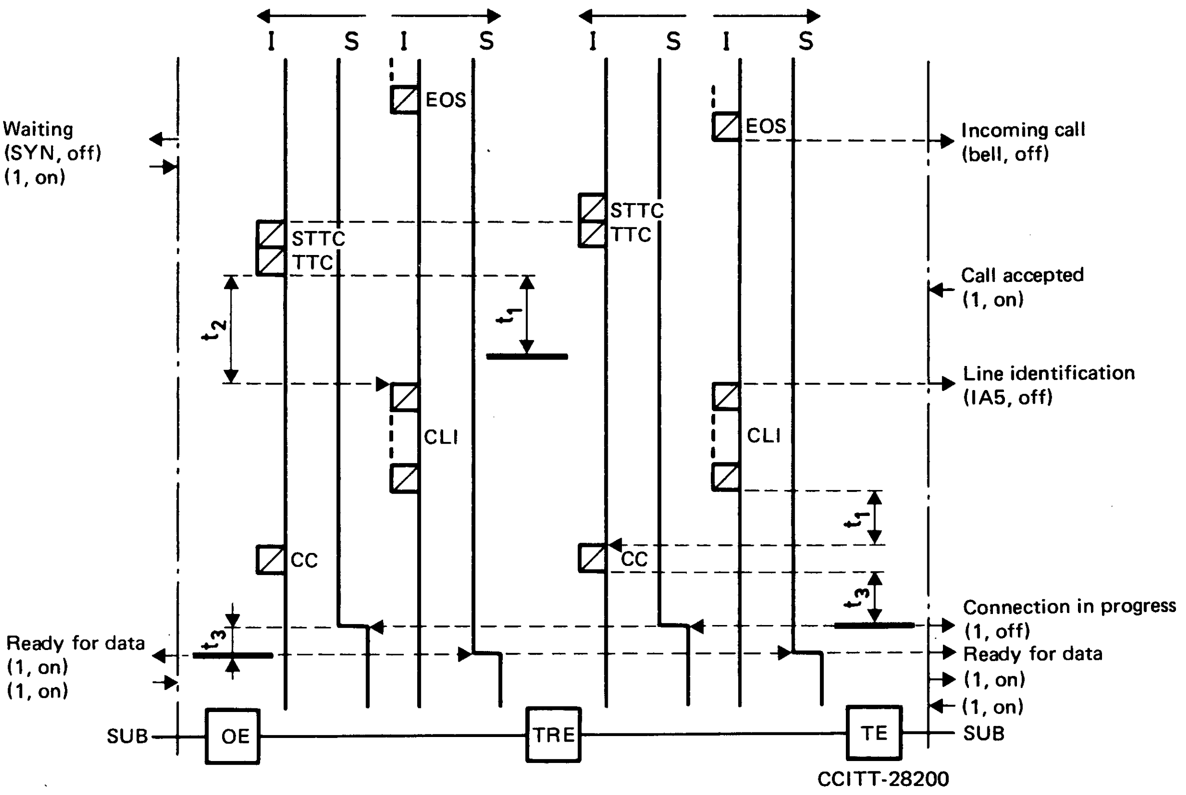
Note 3 – The timings given in Note 2 cover worst case conditions and exchange design should aim to keep them as short as possible.

APPENDIX III (C)

(to Recommendation X.71)

Through-connection procedure

Called line identification not required, calling line identification required (No *connect-when-free* facility)



S	Status bit
– – – –	Correlation line
———	Through-connection
⧄	IA5-character
EOS	End-of-selection signal
STTC	Start-of-transit through-connect signal
TTC	Transit through-connect signal

CLI	Calling line identification signals
CC	Call connected signal
SUB	Subscriber
OE	Originating exchange
TRE	Transit exchange
TE	Terminating exchange
I	Information bit

Note 1 – Where groups of signalling characters are not contiguous, the *waiting* signal ($S = 0$, repetitions of 1s for a period of at least 15 information bits) must be sent during the interim period.

Note 2 – $t_1 = 0\text{-}30$ ms, $t_2 = 30\text{-}50$ ms, $t_3 = 0\text{-}40$ ms.

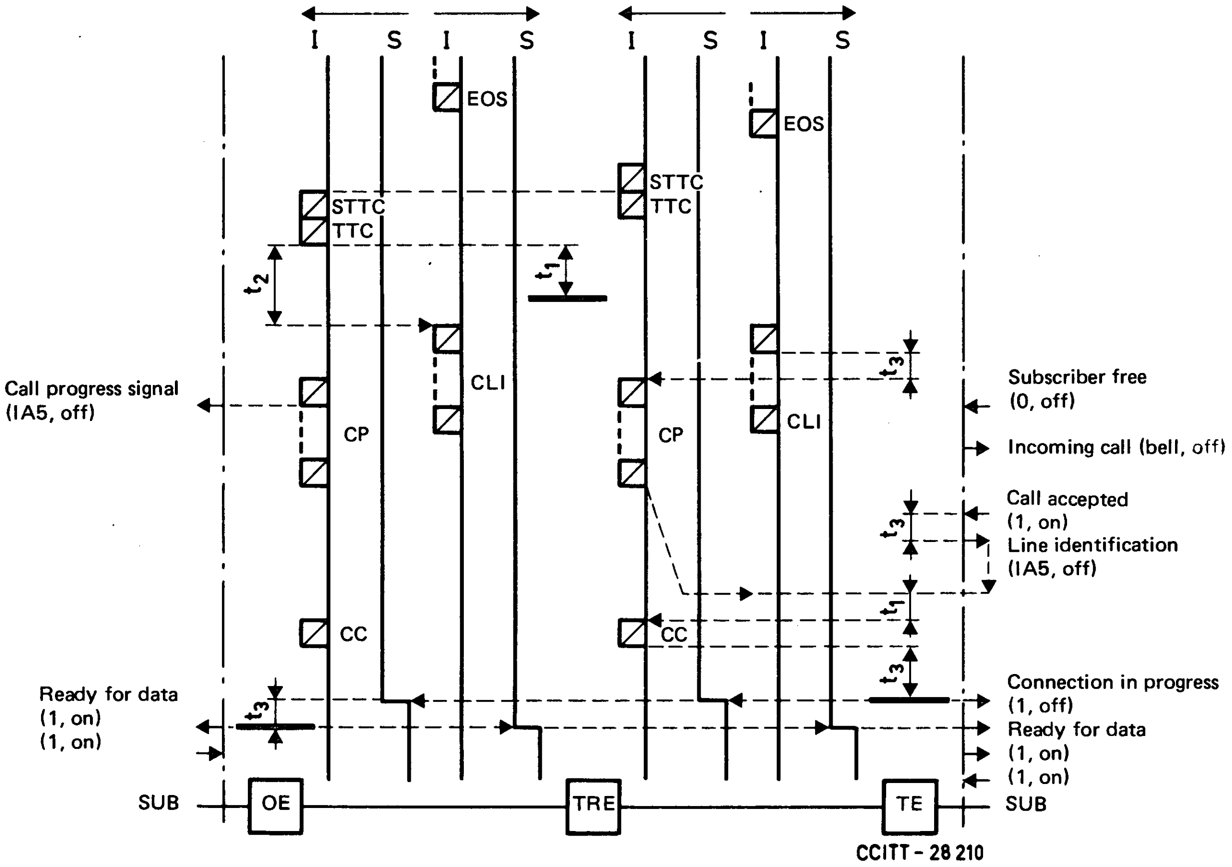
Note 3 – The timings given in Note 2 cover worst case conditions and exchange design should aim to keep them as short as possible.

APPENDIX III (D)

(to Recommendation X.71)

Through-connection procedure

Called line identification not required, calling line identification required (*Connect-when-free* facility, subscriber is busy)



S	Status bit	CLI	Calling line identification signals
— — — —	Correlation line	CP	Call progress signal
———	Through-connection	CC	Call connected signal
		SUB	Subscriber
▨	IA5-character	OE	Originating exchange
EOS	End-of-selection signal	TRE	Transit exchange
STTC	Start-of-transit through-connect signal	DE	Destination exchange
TTC	Transit through-connect signal	I	Information bit

Note 1 — Where groups of signalling characters are not contiguous, the *waiting* signal (S = 0, repetitions of 1s for a period of at least 15 information bits) must be sent during the interim period.

Note 2 – $t_1 = 0\text{-}30$ ms, $t_2 = 30\text{-}50$ ms, $t_3 = 0\text{-}40$ ms.

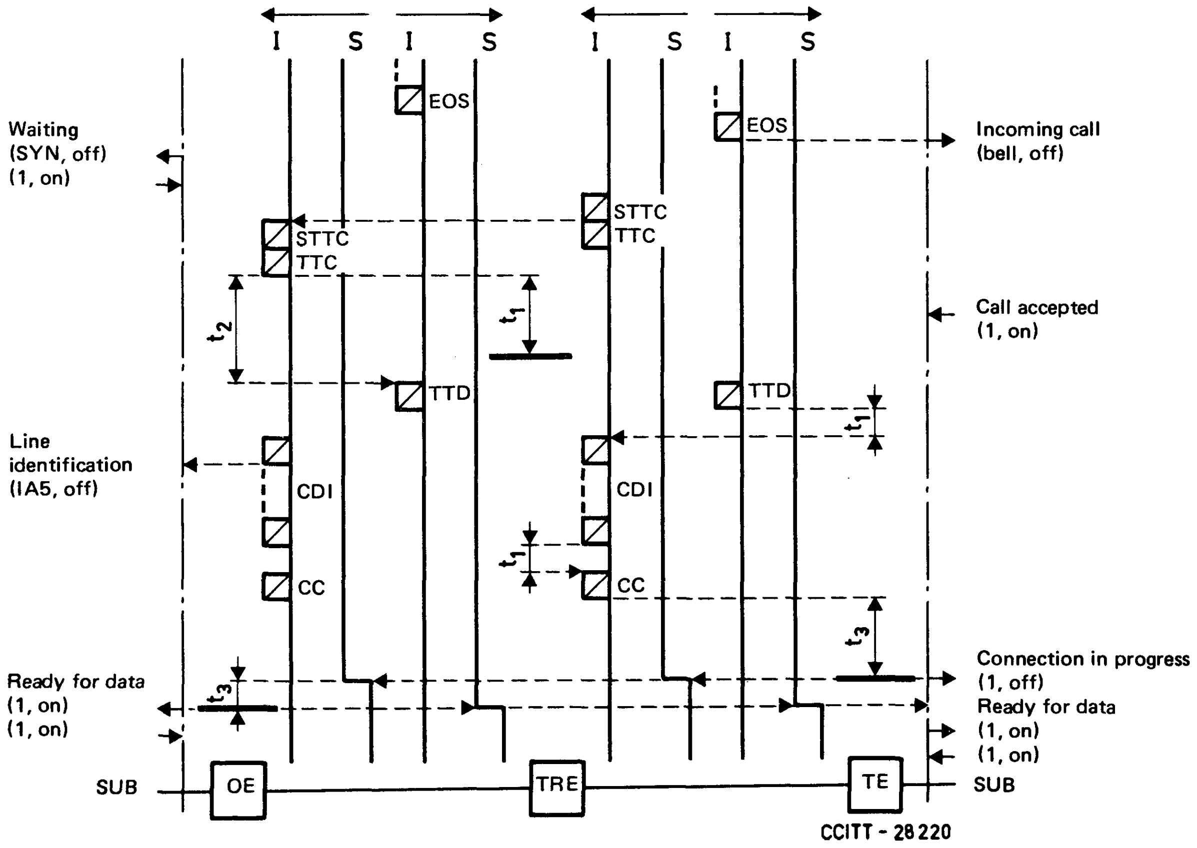
Note 3 – The timings given in Note 2 cover worst case conditions and exchange design should aim to keep them as short as possible.

APPENDIX III (E)

(to Recommendation X.71)

Through-connection procedure

Called line identification required, calling line identification not required (No *connect-when-free* facility)



S	Status bit	CDI	Called line identification signals
- - - -	Correlation line	CC	Call connected signal
———	Through-connection	SUB	Subscriber
⧄	IA5-character	OE	Originating exchange
		TRE	Transit exchange
EOS	End-of-selection signal	TE	Terminating exchange
STTC	Start-of-transit through-connect signal	I	Information bit
TTC	Transit through-connect signal		
TTD	Transit centres through-connected signal		

Note 1 – Where groups of signalling characters are not contiguous, the *waiting* signal (S = 0, repetitions of 1s for a period of at least 15 information bits) must be sent during the interim period.

Note 2 – t_1 = 0-30 ms, t_2 = 30-50 ms, t_3 = 0-40 ms.

Note 3 – The timings given in Note 2 cover worst case conditions and exchange design should aim to keep them as short as possible.

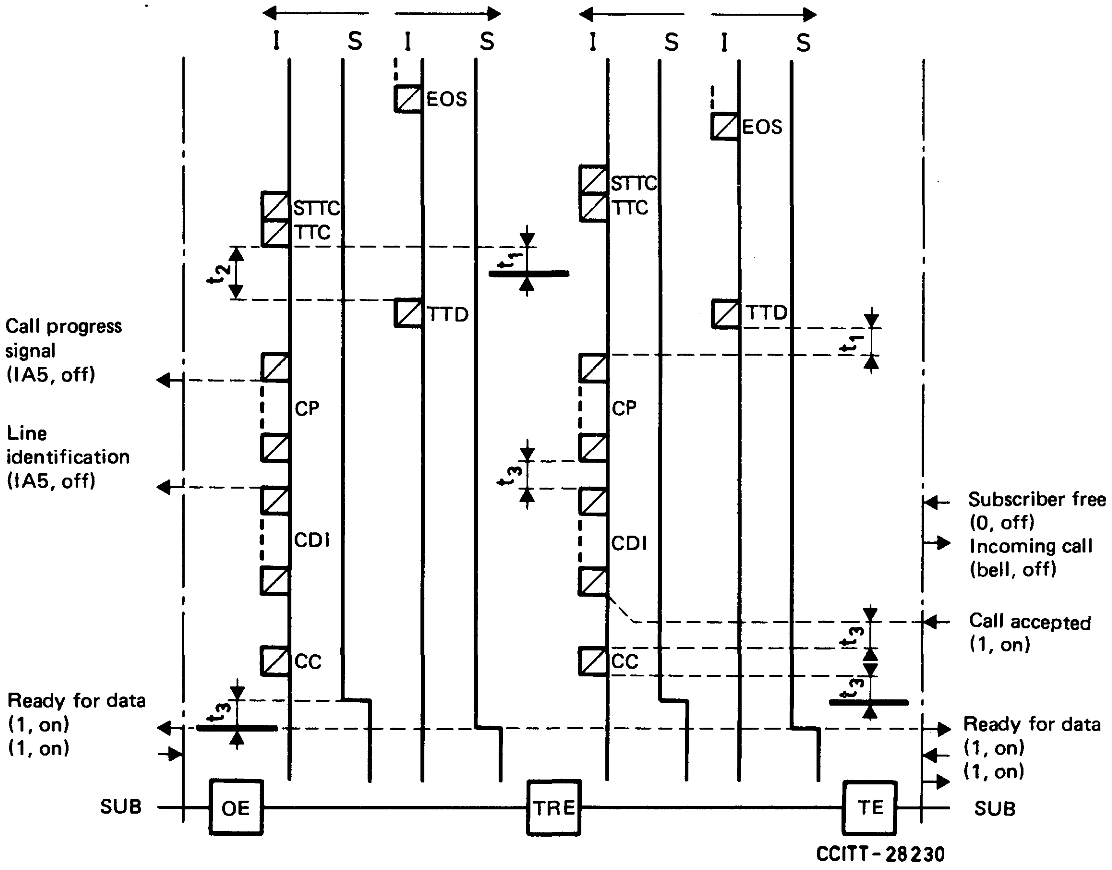
Note 4 – If the call is cleared, the relevant *call progress* signal should be send before or after the *called line identification* signal.

APPENDIX III (F)

(to Recommendation X.71)

Through-connection procedure

Called line identification required, calling line identification not required (*Connect-when-free* facility, subscriber is busy)



S	Status bit	TTD	Transit centres through-connected signal
– – – –	Correlation line	CP	Call progress signal
———	Through-connection	CC	Call connected signal
▨	IA5-character	SUB	Subscriber
		OE	Originating exchange
EOS	End-of-selection signal	TRE	Transit exchange
STTC	Start-of-transit through-connect signal	TE	Terminating exchange
TTC	Transit through-connect signal	I	Information bit

Note 1 – Where groups of signalling characters are not contiguous, the *waiting* signal (S = 0, repetitions of 1s for a period of at least 15 information bits) must be sent during the interim period.

Note 2 – $t_1 = 0\text{-}30$ ms, $t_2 = 30\text{-}50$ ms, $t_3 = 0\text{-}40$ ms.

Note 3 – The timings given in Note 2 cover worst case conditions and exchange design should aim to keep them as short as possible.

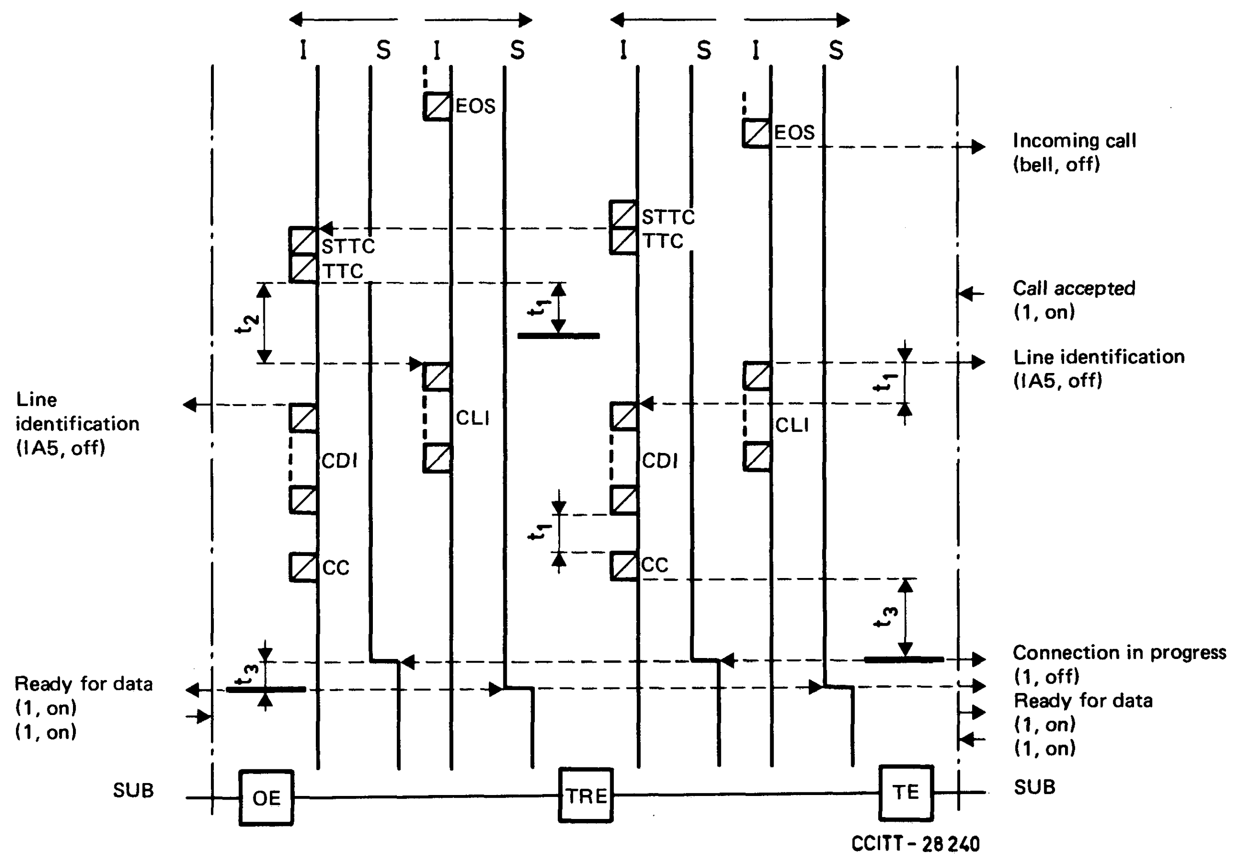
Note 4 – If the call is cleared after the sending of the *called line identification* signal, but before through-connection, a relevant *call progress* signal with clearing could be sent.

APPENDIX III (G)

(to Recommendation X.71)

Through-connection procedure

Called and calling line identification required (No *connect-when-free* facility)



S	Status bit	CDI	Called line identification signals
– – – –	Correlation line	CLI	Calling line identification signal
———	Through-connection	CC	Call connected signal
▨	IA5-character	SUB	Subscriber
		OE	Originating exchange
EOS	End-of-selection signal	TRE	Transit exchange
STTC	Start-of-transit through-connect signal	TE	Terminating exchange
TTC	Transit through-connect signal	I	Information bit

Note 1 – Where groups of signalling characters are not contiguous, the *waiting* signal (S = 0, repetitions of 1s for a period of at least 15 information bits) must be sent during the interim period.

Note 2 – $t_1 = 0\text{-}30$ ms, $t_2 = 30\text{-}50$ ms, $t_3 = 0\text{-}40$ ms.

Note 3 – The timings given in Note 2 cover worst case conditions and exchange design should aim to keep them as short as possible.

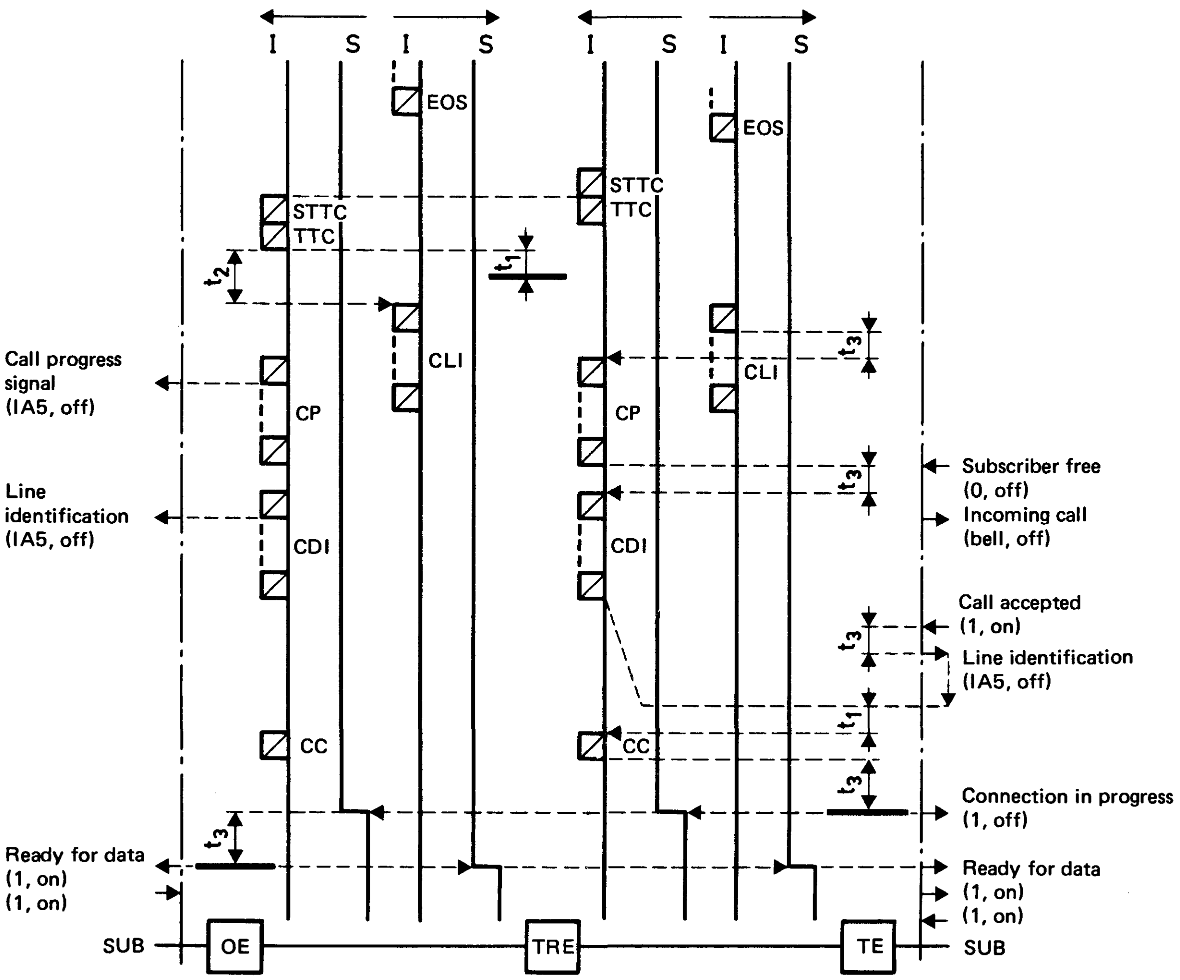
Note 4 – If the call is cleared, the relevant *call progress* signal should be send before or after the *called line identification* signal.

APPENDIX III (H)

(to Recommendation X.71)

Through-connection procedure

Called and calling line identification required (*Connect-when-free* facility, subscriber is busy)



CCITT-28250

S	Status bit
– – – –	Correlation line
———	Through-connection
⧄	IA5-character
EOS	End-of-selection signal
STTC	Start-of-transit through-connect signal
TTC	Transit through-connect signal
CLI	Calling line identification

CP	Call progress signal
CDI	Called line identification signal
CC	Call connected signal
SUB	Subscriber
OE	Originating exchange
TRE	Transit exchange
TE	Terminating exchange
I	Information bit

Note 1 – Where groups of signalling characters are not contiguous, the *waiting* signal (S = 0, repetitions of 1s for a period of at least 15 information bits) must be sent during the interim period.

Note 2 – $t_1 = 0\text{-}30$ ms, $t_2 = 30\text{-}50$ ms, $t_3 = 0\text{-}40$ ms.

Note 3 – The timings given in Note 2 cover worst case conditions and exchange design should aim to keep them as short as possible.

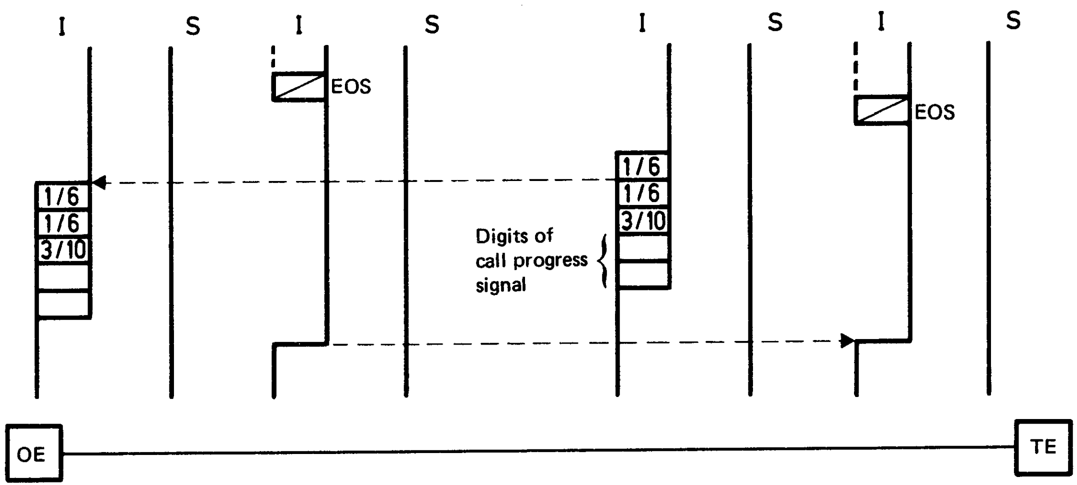
Note 4 – If the call is cleared after the sending of the *called line identification* signal, but before through-connection, a relevant *call progress* signal with clearing could be sent.

APPENDIX IV

(to Recommendation X.71)

Unsuccessful call

Call progress signal with clearing



CCITT - 28260

S	Status bit
I	Information bit
– – – –	Correlation line
⧄	IA5-character
EOS	End-of-selection signal
OE	Originating exchange
TE	Terminating exchange

Note – *Call progress* signals without clearing may be included to indicate such facilities as call redirection.

APPENDIX V

(to Recommendation X.71)

Format of signalling characters within the Recommendation X.50

An example of three successive signalling characters within five octets of one channel of the Recommendation X.50 multiplex structure.

				a_1	a_2	a_3	0
F	a_4	a_5	a_6	a_7	a_8	b_1	0
F	b_2	b_3	b_4	b_5	b_6	b_7	0
F	b_8	c_1	c_2	c_3	c_4	c_5	0
F	c_6	c_7	c_8				

Status bits are 0s.

$a_1 \dots a_8$ is a signalling character

$b_1 \dots b_8$ is a signalling character

$c_1 \dots c_8$ is a signalling character

The framing bits F will be assigned on the multiplexed stream according to Recommendation X.50. No alignment of signalling characters with the envelopes of the multiplex structure is assumed or required.

Recommendation X.75

PACKET-SWITCHED SIGNALLING SYSTEM BETWEEN PUBLIC NETWORKS PROVIDING DATA TRANSMISSION SERVICES

(provisional, Geneva, 1978; amended at Geneva, 1980, Malaga-Torremolinos, 1984, and Melbourne, 1988)

The establishment in various countries of public networks providing packet-switched data transmission services creates a need to standardize international interworking.

The CCITT,

considering

(a) that Recommendation X.1 includes specific user classes of service for data terminal equipments operating in the packet mode, Recommendation X.2 defines user facilities, Recommendations X.25, X.28, X.29, X.31 and X.32 define DTE/DCE interface characteristics and Recommendation X.96 defines *call progress* signals;

(b) that the logical links A1 and G1 in an international connection are defined in Recommendation X.92 for packet-switched data transmission services;

(c) that Recommendations X.300, X.301 and X.302 define the general principles and arrangements for interworking between public data networks, and between public data networks and other public networks;

(d) that Recommendations X.320, X.322, X.323 and X.325 provide descriptions of interworking cases among networks;

(e) that Recommendation X.180 defines the administrative arrangements for International Closed User Groups and that Recommendation X.181 defines the administrative arrangements for the provision of international Permanent Virtual Circuits;

(f) that the necessary elements of the signalling terminal (STE) interface Recommendation at the gateway/transit data switching exchange should be defined independently as:

Physical layer – the mechanical, electrical, functional and procedural characteristics to activate, maintain and deactivate the physical link at the signalling terminal interface;

Link layer – the link layer procedures for data interchange across the interface between the signalling terminals;

Packet layer – the packet format and signalling procedures for the exchange of packets containing control information and user data at the signalling terminal interface;

(g) that Recommendations X.134, X.135, X.136 and X.137 define the quality of service parameters in public networks providing packet-switched data transmission services;

(h) that Recommendations X.110, X.121, X.122, E.164 and E.166 describe the routing principles and numbering plans for public networks including ISDNs;

unanimously declares

(1) that the basic system structure of the signalling and data transfer procedures in terms of elements, should be as specified in the Introduction, *Basic system structure*;

(2) that the mechanical, electrical, functional and procedural characteristics to activate, maintain and deactivate the physical link at the signalling terminal interface should be as specified in § 1 below, *Physical layer – Characteristics of the signalling terminal/physical circuit interface*;

(3) that the link layer procedures which operate over the physical circuits and provide a mechanism for reliable transport of packets at the signalling terminal interface should be as specified in § 2 below, *Link layer procedures between signalling terminals*;

(4) that the packet signalling procedures for the exchange of call information and user data at the signalling terminal interface should be as specified in § 3 below, *Packet layer procedures between signalling terminals*;

(5) that the packet format for packets exchanged at the signalling terminal interface should be as specified in § 4 below, *Packet formats for virtual calls and permanent virtual circuits*;

(6) that the procedure and formats for user facilities and network utilities at the signalling terminal interface should be as specified in § 5 below, *Procedure and formats for user facilities and network utilities.*

INDEX

0 Introduction

0.1 *General*

This Recommendation defines the characteristics and operation of a signalling system for use on interconnecting links between various types of public networks to provide internetwork data transmission services. It permits the transfer of call control and network control information and user traffic.

The Recommendation applies to all links between packet-switched public data networks in different countries and also in a number of cases of international links with ISDNs as specified in Recommendation X.300. These include links between ISDNs and packet-switched public data networks and links between ISDNs providing packet-switched data transmission services as defined in Recommendation X.31. It may also be used on such links where the two public networks are in the same country.

Each internetwork link comprises two directly connected signalling terminals (STEs) each within a public network. Transmission facilities between the two STEs may comprise either one or a number of circuits. Each STE is associated with one end of one link and is part of an exchange or exchange function in the public network.

Certain parts of this Recommendation apply in only a limited range of interworking situations; these are clearly indicated in the text. Some concern links between public networks in the same country, and others concern links where at least one public network is not a packet-switched data network.

The protocol elements included in this Recommendation can be used to support the Network Layer Service for interworking situations.

0.2 *Elements*

The system is made up of communicating elements which function independently and are therefore defined separately. These elements are:

a) the physical circuits which comprise transmission facilities, and a set of mechanical, electrical, functional and procedural interface characteristics between the transmission facilities and the signalling terminals and which provide a mechanism for information transfer between two signalling terminals;

b) the link layer procedures which operate over the physical circuits and provide a mechanism for reliable transport of packets between the two signalling terminals independently of the particular types of physical circuit in use;

c) the packet layer procedures which use the link layer procedures and provide a mechanism for the exchange of call control information and user traffic between the two signalling terminals.

0.3 *Basic system structure*

The basic system structure of the signalling procedures, in terms of the elements, is shown in Figure 1/X.75.

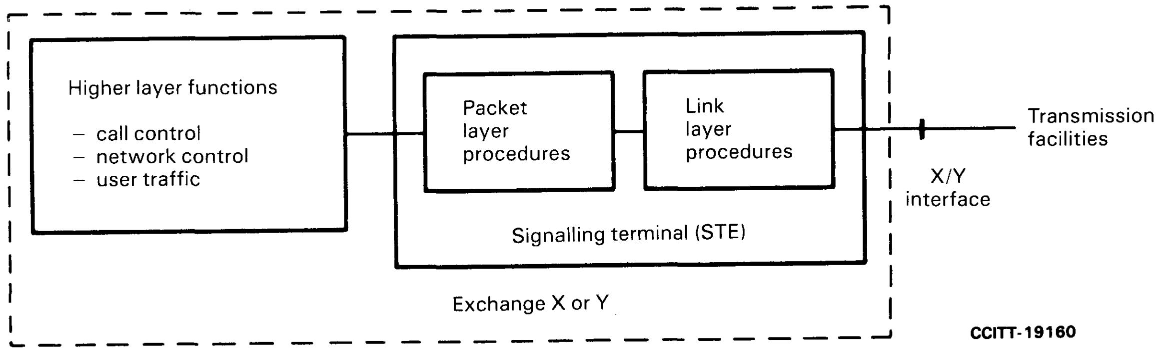


FIGURE 1/X.75

Basic system structure of signalling procedures

Note – Applicable to this Recommendation:

a) STE-X denotes the STE of the exchange under consideration on the link concerned;

b) STE-Y denotes the STE of the other exchange under consideration on the link;

c) the STE-X/STE-Y interface is abbreviated to the X/Y interface;

d) multiple X/Y interfaces may be used between two networks. In this case, each X/Y interface behaves according to the physical, link and packet layer formats and procedures within this Recommendation.

1 Physical layer – Characteristics of the signalling terminal/physical circuit interface

The characteristics of the signalling terminal/physical circuit interface, defined as the physical layer element, shall be in accordance with Recommendation G.703, for physical circuits having a bearer rate of 64 kbit/s and optionally, by bilateral agreement, 2048 Mbit/s (see Note). In addition, Administrations may use for digital circuits any other recognized rate (e.g. 1544 Mbit/s, see Note) by bilateral agreement.

However, for an interim period by bilateral agreement, any other recognized rates could be used for analogue circuits, in which case the characteristics of the signalling terminal/physical circuit interface shall be in accordance with the appropriate V-Series Recommendations.

Each physical circuit of the link must be capable of supporting duplex operation.

In the case of international interworking between packet-switched public data networks, the link is assumed to be data link A1 and/or data link G1 in terms of the hypothetical reference connections defined in Recommendation X.92.

Note – It is for further study whether modifications to the link layer procedures are required for data signalling rates higher than 64 kbit/s in order to support high throughput.

2 Link layer procedures between signalling terminals

2.1 *Scope and field of application*

2.1.1 In order to provide a mechanism for the reliable transport of packets between two signalling terminals, it is necessary to define a procedure which can accept and deliver packets to the packet layer when either single or multiple physical circuits are employed. A multiplicity of physical circuits is required if the effects of circuit failures are not to disrupt the packet layer operation.

2.1.2 The Single link procedure (SLP) described in §§ 2.2 to 2.4 is used for data interchange over a single physical circuit, conforming to the description given in § 1, between two STEs. When multiple physical circuits are employed in parallel this single link procedure is used independently on each circuit and the Multilink procedure (MLP) described in § 2.5 is used for data interchange over these multiple parallel links. In addition, when only a single physical circuit is employed, Administrations may agree bilaterally to use this multilink procedure over the one link.

2.1.3 Each transmission facility is duplex.

2.1.4 The single link procedure is based upon the Link access procedure (LAPB) described in § 2 of Recommendation X.25. The procedure uses the principle and terminology of the High level data link control (HDLC) procedure specified by the International Organization for Standardization (ISO).

The multilink procedure is based on the principle and terminology of the multilink procedure specified by ISO.

2.1.5 For each SLP employed, either extended mode (modulo 128) or non-extended mode (modulo 8) may be used. The choice of the mode employed for such link procedures is independent of all others and of the choice of mode for the corresponding packet layer procedures. All choices are matters for bilateral agreement.

2.2 *Frame structure*

2.2.1 All transmissions are in frames conforming to one of the formats of Tables 1/X.75 and 2/X.75. The flag preceding the address field is defined as the opening flag. The flag following the Frame checking sequence (FCS) field is defined as the closing flag.

2.2.2 *Flag sequence*

All frames shall start and end with the flag sequence consisting of one 0 bit followed by six contiguous 1 bits and one 0 bit. The STE shall only send complete and distinct eight-bit flag sequences when sending multiple flag sequences (see § 2.2.11). A single flag may be used as both the closing flag for one frame and the opening flag for the next frame.

2.2.3 *Address field*

The address field shall consist of one octet. The address field identifies the intended receiver of a command frame and the transmitter of a response frame. The coding of the address field is described in § 2.4.2 below.

2.2.4 *Control field*

The control field shall consist of one or two octets. The content of this field is described in § 2.3.2 below.

TABLE 1/X.75

Frame formats (modulo 8)

Bit order of transmission	12345678	12345678	1 to 8	16 to 1	12345678
	Flag	Address	Control	FCS	Flag
	F 01111110	A 8 bits	C 8 bits	FCS 16 bits	F 01111110

Bit order of transmission	12345678	12345678	1 to 8		16 to 1	12345678
	Flag	Address	Control	Information	FCS	Flag
	F 01111110	A 8 bits	C 8 bits	I N bits	FCS 16 bits	F 01111110

FCS frame checking sequence

$0 \leq N \leq N1 - 32$

TABLE 2/X.75

Frame formats (modulo 128)

Bit order of transmission	12345678	12345678	1 to a)	16 to 1	12345678
	Flag	Address	Control	FCS	Flag
	F	A	C	FCS	F
	01111110	8 bits	bits a)	16 bits	01111110

Bit order of transmission	12345678	12345678	1 to a)		16 to 1	12345678
	Flag	Address	Control	Information	FCS	Flag
	F	A	C	I	FCS	F
	01111110	8 bits	bits a)	N bits	16 bits	01111110

FCS frame checking sequence

$0 \leqslant N \leqslant N1 - 40$

a) Sixteen bits for frame formats that contain sequence numbers; 8 bits for frame formats that do not contain sequence numbers (see *Note*).

Note – For an interim period, frames that do not contain sequence numbers may alternatively have a 16 bit control field format as described in § 2.3.2.1.3.

2.2.5 *Information field*

The information field of a frame, when present, follows the control field (see § 2.2.4 above) and precedes the frame check sequence (see § 2.2.7 below). See § 2.3.4.9, § 2.5.2 and § 4 for the various codings and groupings of bits in the information field as used in this Recommendation.

See §§ 2.3.4.9 and 2.4.8.5 with regard to the maximum information field length.

2.2.6 *Transparency*

The STE, when transmitting, shall examine the frame content between the two flag sequences including the address, control, information and FCS fields and shall insert a 0 bit after all sequences of five contiguous 1 bits (including the last five bits of the FCS) to ensure that a flag sequence is not simulated. The STE, when receiving, shall examine the frame content and shall discard any 0 bit which directly follows five contiguous 1 bits.

2.2.7 *Frame checking sequence (FCS) field*

The notation used to describe the FCS is based on the property of cyclic codes that a code vector such as 1000000100001 can be represented by a polynomial $P(x) = x^{12} + x^5 + 1$. The elements of an *n*-element code word are thus the coefficients of a ploynomial of order $n - 1$. In this application, these coefficients can have the value 0 or 1 and the polynomial operations are performed modulo 2. The polynomial representing the content of a frame is generated using the first bit received after the frame opening flag as the coefficient of the highest order term.

The FCS field shall be a 16-bit sequence. It shall be the ones complement of the sum (modulo 2) of:

1) the remainder of $x^k (x^{15} + x^{14} + x^{13} + x^{12} + x^{11} + x^{10} + x^9 + x^8 + x^7 + x^6 + x^5 + x^4 + x^3 + + x^2 + x + 1)$ divided (modulo 2) by the generator polynomial $x^{16} + x^{12} + x^5 + 1$ where *k* is the number of bits in the frame existing between, but not including, the final bit of the opening flag and the first bit of the FCS, excluding bits inserted for transparency; and
2) the remainder of the division (modulo 2) by the generator polynomial $x^{16} + x^{12} + x^5 + 1$ of the product of x^{16} by the content of the frame existing between, but not including, the final bit of the opening flag and the first bit of the FCS, excluding bits inserted for transparency.

As a typical implementation, at the transmitter, the initial content of the register of the device computing the remainder of the division is preset to all 1s and is then modified by the division by the generator polynomial (as described above) on the address, control and information fields; the ones complement of the resulting remainder is transmitted as the 16-bit FCS.

As the receiver, the initial content of the register of the device computing the remainder is preset to all 1s. The final remainder, after multiplication by x^{16} and then division (modulo 2) by the generator polynomial $x^{16} + x^{12} + x^5 + 1$ of the serial incoming protected bits and FCS, will be 0001110100001111 (x^{15} through x^0, respectively) in the absence of transmission errors.

Note – Explanatory examples are given in Appendix 1 of Recommendation X.25.

2.2.8 *Order of bit transmission*

Addresses, commands, responses and sequence numbers shall be transmitted with the low order bit first (for example, the first bit of the sequence number that is transmitted shall have the weight 2°).

The order of transmitting bits within the information field is not specified under § 2. The FCS shall be transmitted to the line commencing with the coefficient of the highest term, which is found in bit position 16 of the FCS field (see Tables 1/X.75 and 2/X.75).

Note – In Tables 3/X.75, 4/X.75, 5/X.75, 6/X.75, 7/X.75, 8/X.75 and 10/X.75, bit 1 is defined as the low order bit.

2.2.9 *Invalid frames*

The definition of an invalid frame is described in § 2.3.5.3 below.

2.2.10 *Frame abortion*

Aborting a frame is performed by transmitting at least seven contiguous 1s (with no inserted 0s).

2.2.11 *Interframe time fill*

Interframe time fill is accomplished by transmitting contiguous flags between frames, i.e., multiple eight-bit flag sequences (see § 2.2.2).

2.2.12 *Link channel states*

A link channel as defined here is the means for transmission for one direction.

2.2.12.1 *Active channel state*

The incoming or outgoing channel is defined to be in an active condition when it is receiving or transmitting respectively, a frame, an abortion sequence or interframe time fill.

2.2.12.2 *Idle channel state*

The incoming or outgoing channel is defined to be in an idle condition when it is receiving or transmitting, respectively, a contiguous 1 state for a period of at least 15 bit times.

See § 2.3.5.5 for a description of STE action when an idle condition exists on its incoming channel for an excessive period of time.

2.3 *Elements of procedures*

2.3.1 The elements of procedures are defined in terms of actions that occur on receipt of frames.

A procedure is derived from these elements of procedures and is described in § 2.4 below. Together, §§ 2.2 and 2.3 form the general requirements for the proper management of the link.

2.3.2 *Control field formats and parameters*

2.3.2.1 *Control field formats*

The control field contains a command or a response, and sequence numbers where applicable.

Three types of control field formats (see Tables 3/X.75 and 4/X.75) are used to perform numbered information transfer (I format), numbered supervisory functions (S format) and unnumbered control functions (U format).

TABLE 3/X.75

Control field formats (modulo 8)

Control field bits	1	2	3	4	5	6	7	8
I format	0	N(S)			P	N(R)		
S format	1	0	S	S	P/F	N(R)		
U format	1	1	M	M	P/F	M	M	M

N(S) Transmitter send sequence number (bit 2 = low-order bit).

N(R) Transmitter receive sequence number (bit 6 = low-order bit).

S Supervisory function bit.

M Modifier function bit.

P/F Poll bit when issued as a command, final bit when issued as a response (1 = Poll/Final).

P Poll bit (1 = Poll).

TABLE 4/X.75

a) Control field formats (modulo 128)

Control field bits	1st Octet								2nd Octet							
	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
I format	0	N(S)							P	N(R)						
S format	1	0	S	S	X	X	X	X	P/F	N(R)						
U format	1	1	M	M	P/F	M	M	M								

b) Alternative U format, control field formats (modulo 128) (see Note)

Control field bits	1st Octet								2nd Octet							
	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
U format	1	1	M	M	U	M	M	M	P/F	X	X	X	X	X	X	X

N(S) Transmitter send sequence number (bit 2 = low-order bit).

N(R) Transmitter receive sequence number (bit 10 = low-order bit).

S Supervisory function bit.

M Modifier function bit.

X Reserved and set to 0.

U Unspecified.

P/F Poll bit when issued as a command, final bit when issued as a response (1 = Poll/Final).

P Poll bit (1 = Poll).

Note – For an interim period, as described in § 2.3.2.1.3, Administrations may bilaterally agree to use an unnumbered format that consists of a 2-octet control field.

2.3.2.1.1 *Information transfer format – I*

The I format is used to perform an information transfer. The functions of N(S), N(R) and P/F are independent: i.e., each I frame has a N(S), a N(R) which may or may not acknowledge additional I frames received by the STE, and a P bit.

2.3.2.1.2 *Supervisory format – S*

The S format is used to perform link supervisory control functions such as acknowledge I frames, request transmission of I frames, and to request a temporary suspension of transmission of I frames. The function of N(R) and P/F are independent; i.e., each supervisory frame has an N(R) which may or may not acknowledge additional I frames received by the STE, and a P/F bit that may be set to 0 or 1.

2.3.2.1.3 *Unnumbered format – U*

The U format is used to provide additional link control functions. This format contains no sequence number but does include a P/F bit that may be set to 0 to 1. The encoding of the unnumbered commands and responses is as defined in Tables 5/X.75 and 6/X.75. Unnumbered U frames make use of a single octet control field for both modulo 8 and extended modulo 128 operations. However, for an interim period and for extended modulo 128 operations only, some Administrations may choose after bilateral agreement, the 2 octet control field coding described in b) of Table 6/X.75.

2.3.2.2 *Control field parameters*

The various parameters associated with the control field formats are described below.

2.3.2.2.1 *Modulus*

Each I frame is sequentially numbered and may have the value 0 through modulus minus 1 (where “modulus” is the modulus of the sequence numbers). The modulus equals 8 or 128 and the sequence numbers cycle through the entire range.

2.3.2.2.2 *Send state variable V(S)*

The send state variable denotes the sequence number of the next in-sequence I frame to be transmitted. The send state variable can take on the value 0 through modulus minus 1. The value of the send state variable is incremented by 1 with each successive I frame transmission, but cannot exceed N(R) of the last received I or S format frame by more than the maximum number of outstanding I frames (k). The value of k is defined in § 2.4.8.6 below.

2.3.2.2.3 *Send sequence number N(S)*

Only I frames contain N(S), the send sequence number of transmitted frames. At the time of an in-sequence I frame is designated for transmission, the value of N(S) is set equal to the value of the send state variable.

2.3.2.2.4 *Receive state variable V(R)*

The receive state variable denotes the sequence number of the next in-sequence I frame expected to be received. This receive state variable can take on the values 0 through modulus minus 1. The value of the receive state variable is incremented by 1 by the receipt of an error free, in-sequence I frame whose send sequence number N(S) equals the receive state variable.

2.3.2.2.5 *Receive sequence number N(R)*

All I frames and supervisory frames contain N(R), the expected send sequence number of the next received I frame. At the time that a frame of the above types is designated for transmission, the value of N(R) is set equal to the current value of the receive state variable. N(R) indicates that the STE transmitting the N(R) has received correctly all I frames numbered up to and including [N(R) – 1].

2.3.2.2.6 *Poll/Final (P/F) bit*

All frames contain P/F the Poll/Final bit. In command frames the P/F bit is referred to as the P bit. In response frames it is referred to as the F bit.

2.3.3 *Functions of the Poll/Final bit*

The Poll bit set to 1 is used by the STE to solicit (poll) a response from the other STE. The Final bit set to 1 is used by the STE to indicate the response frame transmitted by the other STE as a result of the soliciting (poll) command.

The use of the P/F bit is described in § 2.4.3 below.

2.3.4 *Commands and responses*

The following commands and responses will be supported by the STE and are represented in Tables 5/X.75 and 6/X.75.

The supervisory function bit encoding 11, and those encodings of the modifier function bits in Tables 3/X.75 and 4/X.75 not identified in Tables 5/X.75 and 6/X.75, are identified as *undefined or not implemented* command and response control fields.

The commands and responses are as follows:

2.3.4.1 *Information (I) command*

The function of the information (I) command is to transfer across a data link sequentially numbered frames containing an information field.

2.3.4.2 *Receive ready (RR) command and response*

The Receive ready (RR) supervisory frame is used by the STE to:

1) indicate it is ready to receive an I frame;
2) acknowledge previously received I frames numbered up to and including [N(R) – 1].

An RR frame may be used to indicate the clearance of a busy condition that was reported by the earlier transmission of an RNR frame by that same STE. In addition to indicating the STE status, the RR command with the P bit set to 1 may be used by an STE to ask for the status of the other STE.

TABLE 5/X.75

Commands and responses (modulo 8)

Format	Command	Response	Encoding							
			1	2	3	4	5	6	7	8
Information transfer	I (information)		0	N(S)			P	N(R)		
Supervisory	RR (receive ready)	RR (receive ready)	1	0	0	0	P/F	N(R)		
	RNR (receive not ready)	RNR (receive not ready)	1	0	1	0	P/F	N(R)		
	REJ (reject)	REJ (reject)	1	0	0	1	P/F	N(R)		
Unnumbered	SABM (set asynchronous balanced mode)		1	1	1	1	P	1	0	0
	DISC (disconnect)		1	1	0	0	P	0	1	0
		FRMR (frame reject)	1	1	1	0	F	0	0	1
		UA (unnumbered acknowledgement)	1	1	0	0	F	1	1	0
		DM (disconnected mode)	1	1	1	1	F	0	0	0

TABLE 6/X.75

a) Commands and responses (modulo 128)

			1	2	3	4	5	6	7	8	9	10 to 16
Format	Command	Response	Encoding									
Information transfer	I (information)		0	N(S)							P	N(R)
Supervisory	RR (receive ready)	RR (receive ready)	1	0	0	0	0	0	0	0	P/F	N(R)
	RNR (receive not ready)	RNR (receive not ready)	1	0	1	0	0	0	0	0	P/F	N(R)
	REJ (reject)	REJ (reject)	1	0	0	1	0	0	0	0	P/F	N(R)
Unnumbered	SABME (set asynchronous balanced mode extended)		1	1	1	1	P	1	1	0		
	DISC (disconnect)		1	1	0	0	P	0	1	0		
		FRMR (frame reject)	1	1	1	0	F	0	0	1		
		UA (unnumbered acknowledgement)	1	1	0	0	F	1	1	0		
		DM (disconnected mode)	1	1	1	1	F	0	0	0		

b) Alternative unnumbered commands and responses (modulo 128) (see Note 2)

			1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
Format	Command	Response	Encoding (Note 1)															
Unnumbered	SABME (set asynchronous balanced mode extended)		1	1	1	1	U	1	1	0	P	0	0	0	0	0	0	0
	DISC (disconnect)		1	1	0	0	U	0	1	0	P	0	0	0	0	0	0	0
		FRMR (frame reject)	1	1	1	0	U	0	0	1	F	0	0	0	0	0	0	0
		UA (unnumbered acknowledgement)	1	1	0	0	U	1	1	0	F	0	0	0	0	0	0	0
		DM (disconnected mode)	1	1	1	1	U	0	0	0	F	0	0	0	0	0	0	0

Note 1 – Bit 5 of unnumbered format frames is unspecified in alternative *b)*.

Note 2 – For an interim period, as described in § 2.3.2.1.3, Administrations may bilaterally agree to use a format that consists of a 2-octet control field.

2.3.4.3 *Receive not ready (RNR) command and response*

The Receive not ready (RNR) supervisory frame is used by the STE to indicate a busy condition; i.e., temporary inability to accept additional incoming I frames. I frames numbered up to and including [N(R) – 1] are acknowledged. I frame N(R) and any subsequent I frames received, if any, are not acknowledged; the acceptance status of these I frames will be indicated in subsequent frames.

In addition to indicating the STE status, the RNR command with the P bit set to 1 may be used by an STE to ask for the status of the other STE.

2.3.4.4 *Reject (REJ) command and response*

The reject (REJ) supervisory frame is used by the STE to request retransmission of I frames starting with the frame numbered N(R). I frames numbered [N(R) – 1] and below are acknowledged. Additional I frames pending initial transmission may be transmitted following the retransmitted I frame(s).

Only one REJ exception condition for a given direction of information transfer may be established at any time. The REJ exception condition is cleared (reset) upon the receipt of an I frame with an N(S) equal to the N(R) of the REJ frame.

An REJ frame may be used to indicate the clearance of a busy condition that was reported by the earlier transmission of an RNR frame by that same STE. In addition to indicating the STE status, the REJ command with the P bit set to 1 may be used by an STE to ask for the status of the other STE.

2.3.4.5 *Set asynchronous balanced mode (SABM) command and set asynchronous balanced mode extended (SABME) command*

The SABM unnumbered command is used to place the addressed STE in the asynchronous balanced mode (AMB) information transfer phase, where all command/response control fields will be one octet in length.

The SABME unnumbered command is used to place the addressed STE in the asynchronous balanced mode information transfer phase, where numbered command/response control fields will be two octets in length and unnumbered command response fields will be one octet in length (see *Note*).

No information field is permitted with the SABM and SABME command. The transmission of a SABM/SABME command indicates the clearance of a busy condition that was reported by the earlier transmission of an RNR frame by that same STE. The STE confirms acceptance of SABM/SABME (modulo 8/ modulo 128) by the transmission at the first opportunity of an UA response. Upon acceptance of this command both the send state variable and the receive state variable are set to 0.

Previously transmitted I frames that are unacknowledged when this command is actioned remain unacknowledged.

Note – For an interim period, as described in § 2.3.2.1.3, Administrations may bilaterally agree to use a format that consists of a 2-octet control field.

2.3.4.6 *Disconnect (DISC) command*

The DISC unnumbered command is used to terminate the mode previously set. It is used to inform the STE receiving the DISC command that the STE sending the DISC command is suspending operation. No information field is permitted with the DISC command. Prior to actioning the DISC command, the addressed STE confirms the acceptance of DISC by the transmission of an UA response. The STE sending the DISC enters the disconnected phase when it receives the acknowledging UA response.

Previously transmitted I frames that are unacknowledged when this command is actioned remain unacknowledged.

2.3.4.7 *Unnumbered acknowledge (UA) response*

The UA unnumbered response is used by the STE to acknowledge the receipt and acceptance of the mode-setting commands. Received mode-setting commands are not activated until the UA response is transmitted. The transmission of an UA response indicates the clearance of a busy condition that was reported by the earlier transmission of an RNR frame by that same STE. No information field is permitted with the UA response.

2.3.4.8 *Disconnected mode (DM) response*

The DM unnumbered response is used to report a status where the STE is logically disconnected from the link, and is in the disconnected phase. The DM response is sent in this phase in response to the reception of a set mode command, to inform the STE that the STE is still in disconnected phase and cannot action a set mode command. No information field is permitted with the DM response.

An STE in a disconnected phase will monitor received commands and will react to an SABM/SABME command as outlined in § 2.4.4 below, and will respond with a DM response with the F bit set to 1 to any other command received with the P bit set to 1.

2.3.4.9 *Frame reject (FRMR) response*

The FRMR unnumbered response is used by the STE to report an error condition not recoverable by retransmission of the identical frame, i.e., at least one of the following conditions, which results from the receipt of a valid frame:

1) the receipt of a command or a response control field that is undefined or not implemented;
2) the receipt of an I frame with an information field which exceeds the maximum established length;
3) the receipt of an invalid N(R);
4) the receipt of a frame with an information field which is not permitted or the receipt of a supervisory or unnumbered frame with incorrect length;
5) the receipt of a supervisory frame with the F bit set to 1, except during a timer recovery condition as described in § 2.4.5.9 or except as a reply to a command sent with the P bit set to 1;
6) the receipt of an unexpected UA or DM response;
7) the receipt of an invalid N(S).

An invalid N(R) is defined as one which points to an I frame which has previously been transmitted and acknowledged or to an I frame which has not been transmitted and is not the next sequential I frame awaiting transmission. A valid N(R) must be within the range from the lowest send sequence number N(S) of the still unacknowledged frame(s) to the current STE send state variable included (or to the current internal variable x if the STE is in the timer recovery condition as described in § 2.4.5.9). This constraint applies even if the STE is in a frame rejection condition.

An invalid N(S) is defined as an N(S) which is equal to the last transmitted $N(R) + k$ and is equal to the received state variable V(R), where k is the maximum number of outstanding information frames (see § 2.4.8.6 below).

An information field which immediately follows the control field, and consists of 3 octets (modulo 8) or 5 octets (modulo 128), is returned with this response and provides the reason for the FRMR response. This format is given in Tables 7/X.75 and 8/X.75.

For condition 4) listed above, bits W and X should be set to 1.

For conditions 5), 6) and 7) listed above, bit W should be set to 1.

In all cases, the STE receiving the FRMR should examine the contents of the rejected frame control field for further clarification of the cause of the error before recording this error.

2.3.5 *Exception condition reporting and recovery*

The error recovery procedures which are available to effect recovery following the detection/occurrence of an exception condition at the link layer are described below. Exception conditions described are those situations which may occur as the result of transmission errors, STE malfunction or operational situations.

2.3.5.1 *Busy condition*

The busy condition results when an STE is temporarily unable to continue to receive I frames due to internal constraints, e.g., receive buffering limitations. In this case an RNR frame is transmitted from the busy STE. I frames pending transmission may be transmitted from the busy STE prior to or following the RNR frame. An indication that the busy condition has cleared is communicated by the transmission of an UA (only in response to a SABM/SABME command), RR, REJ or SABM/SABME (modulo 8/modulo 128) frame.

TABLE 7/X.75

FRMR information field format (modulo 8)

Information field bits

1 2 3 4 5 6 7 8	9	10 11 12	13	14 15 16	17	18	19	20	21	22	23	24
Rejected frame control field	0	V(S)	C/R	V(R)	W	X	Y	Z	0	0	0	0

Rejected frame control field is the control field of the received frame which caused the frame reject.

V(S) is the current send state variable value at the STE reporting the rejection condition (bit 10 = low-order bit).

C/R set to 1 indicates the rejected frame was a response.

C/R set to 0 indicates the rejected frame was a command.

V(R) is the current receive state variable value at the STE reporting the rejection condition (bit 14 = low-order bit).

W set to 1 indicates that the control field received and returned in bits 1 through 8 was invalid or not implemented.

X set to 1 indicates that the control field received and returned in bits 1 through 8 was considered invalid because the frame contained an information field which is not permitted with this frame or is a supervisory or unnumbered frame with incorrect length. Bit W must be set to 1 in conjunction with this bit.

Y set to 1 indicates that the information field received exceeded the maximum established capacity.

Z set to 1 indicates the control field received and returned in bits 1 through 8 contained an invalid N(R).

Bits 9 and 21 through 24 shall be set to 0.

TABLE 8/X.75

FRMR information field format (modulo 128)

Information field bits

1 to 16	17	18 to 24	25	26 to 32	33	34	35	36	37	38	39	40
Rejected frame control field	0	V(S)	C/R	V(R)	W	X	Y	Z	0	0	0	0

Rejected frame control field is the control field of the received frame which caused the frame reject. When the rejected frame is an unnumbered frame, the control field of the rejected frame is positioned in bit positions 1-8, with 9-16 set to 0. If, however, the interim solution mentioned in § 2.3.2.1.3 is adopted, the 2-octet control field will be placed in bit positions 1-16.

V(S) is the current send state variable value at the STE reporting the rejection condition (bit 18 = low order bit).

C/R set to 1 indicates the rejected frame was a response. C/R set to 0 indicates the rejected frame was a command.

V(R) is the current receive state variable value at the STE reporting the rejection condition (bit 26 = low-order bit).

W set to 1 indicates that the control field received and returned in bits 1 through 16 was invalid or not implemented.

X set to 1 indicates that the control field received and returned in bits 1 through 16 was considered invalid because the frame contained an information field which is not permitted with this frame or is a supervisory or unnumbered frame with incorrect length. Bit W must be set to 1 in conjunction with this bit.

Y set to 1 indicates that the information field received exceeded the maximum established capacity.

Z set to 1 indicates the control field received and returned in bits 1 through 16 contained an invalid N(R).

Bits 17 and 37 through 40 shall be set to 0.

2.3.5.2 *N(S) sequence error condition*

The information field of all I frames received whose N(S) does not equal the receive state variable V(R) will be discarded.

An N(S) sequence error exception condition occurs in the receiver when an I frame received contains an N(S) which is not equal to the receive state variable V(R) at the receiver. The receiver does not acknowledge (increment its receive state variable) the I frame causing the sequence error, or any I frame which may follow until an I frame with the correct N(S) is received.

An STE which receives one or more valid I frames having sequence errors or subsequent supervisory frames (RR, RNR and REJ) shall accept the control information contained in the N(R) field and the P/F bit to perform link control functions; e.g., to receive acknowledgement of previously transmitted I frames, and to cause the STE to respond (P bit sent to 1).

2.3.5.2.1 *REJ recovery*

The REJ frame is used by a receiving STE to initiate a recovery (retransmission) following the detection of an N(S) sequence error.

With respect to each direction of transmission on the link, only one *sent REJ* exception condition from an STE is established at a time. A *sent REJ* exception condition is cleared when the requested I frame is received.

An STE receiving REJ initiates sequential (re-)transmission of I frames starting with the I frame indicated by the N(R) obtained in the REJ frame.

The retransmitted frame(s) may contain an N(R) and a P bit that is updated from, and therefore different from, the ones contained in the originally transmitted I frame(s).

2.3.5.2.2 *Time-out recovery*

If an STE, due to a transmission error, does not receive (or receives and discards) a single I frame or the last I frames in a sequence of I frames, it will not detect an N(S) sequence error condition and therefore will not transmit an REJ frame. The STE which transmitted the unacknowledged I frame(s) shall, following the completion of a system specified time-out period (see §§ 2.4.5.9 and 2.4.8.1 below), take appropriate recovery action to determine at which I frame retransmission must begin. The retransmitted frames may contain an N(R) and a P bit that are updated from, and therefore different from, the ones contained in the originally transmitted I frames.

2.3.5.3 *Invalid frame condition*

Any frame which is invalid will be discarded, and no action is taken as the result of that frame. An invalid frame is defined as one which:

a) is not properly bounded by two flags;

b) in non-extended (modulo 8) operation, contains fewer than 32 bits between flags; in extended (modulo 128) operation, contains fewer than 40 bits between flags of frames that contain sequence numbers or 32 bits between flags of frames that do not contain sequence numbers.

   Note – Or fewer than 40 bits (modulo 128) if 2-octet control field is used as alternative b during the interim period (see § 2.3.2.1.3).

c) contains a Frame check sequence (FCS) error;

d) contains an address other than A or B (for single link operation) or other than C or D (for multilink operation).

For those networks that are octet aligned, a detection of non-octet alignment may be made at the link layer by adding a frame validity check that requires the number of bits between the opening flag and the closing flag, excluding bits inserted for transparency, to be an integral number of octets in length, or the frame considered invalid.

2.3.5.4 *Frame rejection condition*

A frame rejection condition is established upon the receipt of an error-free frame with one of the conditions listed in § 2.3.4.9 above.

This frame rejection exception condition is reported by sending an FRMR response for appropriate STE action.

Once an STE has established a frame rejection condition, no additional I or S format frames are accepted until the condition is reset except for examination of the P bit. The FRMR response may be repeated at each opportunity, as specified in § 2.4.7.3 until recovery is effected by the other STE or until the STE initiates its own recovery in case the other STE does not respond.

2.3.5.5 *Excessive idle channel state condition on incoming channel*

Upon detection of an idle channel state condition (see § 2.2.12.2 above) on the incoming channel, the STE shall wait for a period T3 (see § 2.4.8.3 below) without taking any specific action, waiting for detection of a return to the active channel state (i.e., detection of at least one flag sequence). After the period T3, the STE shall notify the MLP or the packet layer of the excessive idle channel state condition, but shall not take any action that would preclude the other STE from establishing the link by normal link set-up procedures.

The value of T3 is a system parameter and is agreed bilaterally.

2.4 *Description of the procedures*

2.4.1 *Extended and non-extended modes of operation*

Changing from non-extended operation to extended operation, or vice versa, requires bilateral agreement and is not supported dynamically.

Table 5/X.75 indicates the command and response control field formats used with the non-extended (modulo 8) service. The mode setting command employed to initialize (set up) or reset the non-extended mode is the SABM command. Table 6/X.75 indicates the command and response control field formats used with the extended (modulo 128) service. The mode setting command employed to initialize (set up) or reset the extended mode is the SABME command.

2.4.2 *Procedure for addressing*

Commands are sent with the remote STE address and responses are sent with the local STE address.

In order to allow differentiation between single link operation and multilink operation for diagnostic and/or maintenance reasons, different address pair encodings shall be assigned to links operating with the multilink procedure (MLP) compared to links operating with the single link procedure (SLP). These STE addresses are coded as follows;

	Address	1	2	3	4	5	6	7	8
Single link operation	A	1	1	0	0	0	0	0	0
	B	1	0	0	0	0	0	0	0
Multilink operation	C	1	1	1	1	0	0	0	0
	D	1	1	1	0	0	0	0	0

A and B, or C and D, are assigned by bilateral agreement between the Administrations.

2.4.3 *Procedure for the use of the P/F bit*

The STE receiving an SABM/SABME, DISC, supervisory command or I frame with the P bit set to 1 will set the F bit to 1 in the next response frame it transmits.

The response frame returned by the STE to an SABM/SABME or DISC command with the P bit set to 1 will be an UA or DM response with the F bit set to 1. The response frame returned by the STE to an I frame with the P bit set to 1, received during the information transfer phase, will be an RR, REJ, RNR or FRMR response with the F bit set to 1. The response frame returned by the STE to a supervisory command with the P bit set to 1, received during the information transfer phase, will be an RR, REJ, RNR or FRMR response with the F bit set to 1.

The response frame returned to an I frame or supervisory frame with the P bit set to 1, received in the disconnected phase, will be a DM with F bit set to 1.

The P bit may be used by the STE in conjunction with the time-out recovery condition (see § 2.4.5.9 below).

When not used the P/F bit is set to 0.

Note – Other use of the P bit by the STE is a subject for further study.

2.4.4 *Procedures for link set up and disconnection*

2.4.4.1 *Link set up*

The STE will indicate that it is able to set up the link by transmitting contiguous flags (active channel state).

Either STE may initialize the link by sending SABM/SABME (modulo 8/modulo 128) and starting Timer T1 in order to determine when too much time has elapsed waiting for a reply. The opposite STE upon receiving SABM/SABME correctly, sends UA and resets both its state variables to 0. If UA is received correctly, then the link is set up and the initiating STE resets both its state variables to 0 and stops Timer T1.

If, upon receipt of SABM/SABME correctly, the STE determines that it cannot enter the indicated phase, it sends the DM response.

When receiving the DM response, the STE which has transmitted an SABM/SABME stops its Timer T1 and does not enter the information transfer phase.

The STE sending SABM/SABME will ignore and discard any frames except SABM/SABME, DISC, UA and DM from the other STE.

Frames other than UA and DM in response to a received SABM/SABME will be sent only after the link is set up and if no outstanding SABM/SABME exists.

If an SABM/SABME or DISC command, UA or DM response is not received correctly, the result will be that the Timer T1 will run out in the STE which originally sent the SABM/SABME and that the STE may resend SABM/SABME and restart Timer T1.

After transmission of SABM/SABME N2 times by the STE, appropriate recovery action will be initiated.

The value of N2 is defined in § 2.4.8.4 below.

2.4.4.2 *Information transfer phase*

After having transmitted the UA response to the SABM/SABME command or having received the UA response to a transmitted SABM/SABME command, the STE will accept and transmit I and supervisory frames according to the procedures described in § 2.4.5 below.

When receiving an SABM/SABME (modulo 8/modulo 128) command while in the information transfer phase, the STE will conform to the resetting procedure described in § 2.4.7 below.

2.4.4.3 *Link disconnection*

During the information transfer phase, either STE shall indicate a request for disconnecting the link by transmitting a DISC command, and it shall start Timer T1 (see § 2.4.8.1 below).

The STE, on correctly receiving a DISC command, will send a UA response and enter the disconnected phase. The STE, on receiving a UA or DM response to a sent DISC command, stops its timer, and enters the disconnected phase. If a UA or DM response is not received correctly, this will result in the expiration of the Timer T1 in the STE which originally sent the DISC command. If Timer T1 runs out, this STE will retransmit a DISC command and restart Timer T1. This action will continue until a UA response or a DM response is correctly received or until recovery takes place at a higher layer after transmission of DISC N2 times. The value of N2 is defined in § 2.4.8.4 below.

2.4.4.4 *Disconnected phase*

2.4.4.4.1 After having received a DISC command and returned a UA response, or having received the UA response to a transmitted DISC command, the STE will enter the disconnected phase.

In the disconnected phase, the STE may initiate link set up. In the disconnected phase, the STE will react to the receipt of an SABM/SABME command as described in § 2.4.4.1 above and will transmit a DM response in answer to a received DISC command.

When receiving any other command frame (defined, or undefined or not implemented) with the P bit set to 1, the STE will transmit a DM response with the F bit set to 1. Other frames received in the disconnected phase will be ignored.

2.4.4.4.2 After recovery from an internal malfunction, the STE may either initiate a resetting procedure (see § 2.4.7 below) or disconnect the link (see § 2.4.4.3 above) prior to a link set up procedure (see § 2.4.4.1 above).

2.4.4.5 *Collision of unnumbered commands*

Collision situations shall be resolved in the following way.

2.4.4.5.1 If the sent and received unnumbered commands are the same, each STE shall send the UA response at the earliest possible opportunity. Each STE shall enter the indicated phase after receiving a UA response.

2.4.4.5.2 If the sent and received unnumbered commands are different, each STE shall enter the disconnected phase and issue a DM response at the earliest possible opportunity.

2.4.5 *Procedures for information transfer*

The procedures which apply to the transmission of I frames in each direction during the information transfer phase are described below.

In the following, "number one higher" is in reference to a continuously repeated sequence series, i.e., 7 is 1 higher than 6 and 0 is 1 higher than 7 for modulo 8 series, and 127 is 1 higher than 126 and 0 is 1 higher than 127 for modulo 128 series.

2.4.5.1 *Sending I frames*

When the STE has an I frame to transmit (i.e., an I frame not already transmitted, or having to be retransmitted as described in § 2.4.5.6 below), it will transmit it with an N(S) equal to its current send state variable V(S), and an N(R) equal to its current receive state variable V(R). At the end of the transmission of the I frame, it will increment its send state variable V(S) by 1.

If the Timer T1 is not running at the time of transmission of an I frame, it will be started.

If the send state variable V(S) is equal to the last value of N(R) received plus *k* (where *k* is the maximum number of outstanding I frames, see § 2.4.8.6) the STE will not transmit any new I frames, but may retransmit an I frame as described in § 2.4.5.6 or § 2.4.5.9 below.

When the STE is in a busy condition, it may still transmit I frames provided that the other STE is not busy. When in the frame rejection condition, the STE will stop transmitting I frames.

2.4.5.2 *Receiving an I frame*

2.4.5.2.1 When the STE is not in a busy condition and receives a valid I frame whose send sequence number N(S) is equal to the STE receive state variable V(R), the STE will accept the information field of this frame, increment by one its receive state variable V(R), and act as follows:

a) If the STE is still not in a busy condition:

   i) If an I frame is available for transmission by the STE, it may act as in § 2.4.5.1 above and acknowledge the received I frame by setting N(R) in the control field of the next transmitted I frame to the value of the STE receive state variable V(R). The STE may also acknowledge the received I frame by transmitting an RR with the N(R) equal to the value of the STE receive state variable V(R).

   ii) If no I frame is available for transmission by the STE, it will transmit an RR with the N(R) equal to the value of the STE receive state variable V(R).

b) If the STE is now in a busy condition, it will transmit an RNR frame with N(R) equal to the value of the STE receive state variable V(R) (see § 2.4.5.8).

2.4.5.2.2 When the STE is in a busy condition, it may ignore the information field contained in a received I frame.

2.4.5.3 *Reception of invalid frames*

When the STE receives an invalid frame (see § 2.3.5.3), this frame will be discarded.

2.4.5.4 *Reception of out of sequence I frames*

When the STE receives a valid I frame whose send sequence number is incorrect, i.e., not equal to the current STE receive state variable V(R), it will discard the information field of the frame and transmit a REJ frame with the N(R) set to one higher than the N(S) of the last correctly received I frame. The REJ frame will be a command frame with the P bit set to 1 if an acknowledged transfer of the retransmission request is required; otherwise the REJ frame may be either a command or a response frame. The STE will then discard the information field of all I frames received until the expected I frame is correctly received. When receiving the expected I frame, the STE will then acknowledge the I frame as described in § 2.4.5.2 above. The STE will use the N(R) and P bit information in the discard I frames, as described in § 2.3.5.2 above.

2.4.5.5 *Receiving acknowledgement*

When correctly receiving an I frame or a supervisory frame (RR, RNR or REJ), even in the busy condition except in the frame rejection condition, the STE will consider the N(R) contained in this frame as an acknowledgement for all I frames it has transmitted with an N(S) up to and including the received N(R) – 1. The STE will stop Timer T1 when it correctly receives an I frame or a supervisory frame with the N(R) higher than the last received N(R) (actually acknowledging some I frames), or an REJ frame with an N(R) equal to the last received N(R).

If Timer T1 has been reset and if there are outstanding I frames still unacknowledged, Timer T1 will be restarted. If Timer T1 then runs out, the STE will follow the retransmission procedure (in § 2.4.5.9 below) with respect to the unacknowledged I frames.

2.4.5.6 *Receiving an REJ frame*

When receiving an REJ frame, the STE will set its send state variable V(S) to the N(R) received in the REJ control field. It will transmit the corresponding I frame as soon as it is available or retransmit it in accordance with the procedures described in § 2.4.5.1 above. (Re)transmission will conform to the following procedure:

i) If the STE is transmitting a supervisory command or response when it receives the REJ frame, it will complete that transmission before commencing transmission of the requested I frame.

ii) If the STE is transmitting an unnumbered command or response when it receives the REJ frame, it will ignore the request for retransmission.

iii) If the STE is transmitting an I frame when the REJ frame is received, it may abort the I frame and commence transmission of the requested I frame immediately after abortion.

iv) If the STE is not transmitting any frame when the REJ frame is received, it will commence transmission of the requested I frame immediately.

In all cases, if other unacknowledged I frames have already been transmitted following the one indicated in the REJ frame, then those I frames will be retransmitted by the STE following the retransmission of the requested I frame. Other I frames not yet transmitted may be transmitted following the retransmitted I frames.

If the REJ frame was received from the other STE as a command with the P bit set to 1, the STE will transmit an RR, RNR or REJ response with the F bit set to 1 before transmitting or retransmitting the corresponding I frame.

2.4.5.7 *Receiving an RNR frame*

After receiving an RNR frame whose N(R) acknowledges all frames previously transmitted, the STE will stop Timer T1 and may then transmit an I frame, with the P bit set to 0, whose send sequence number is equal to the N(R) indicated in the RNR frame, restarting the Timer T1 as it does. After receiving an RNR frame whose N(R) indicates a previously transmitted frame, the STE will not transmit or retransmit any I frame, Timer T1 being already running. In either case, if the Timer T1 runs out before receipt of a busy clearance indication, the STE will follow the procedure described in § 2.4.5.9 below. In any case, the STE will not transmit any other I frames before receiving an RR or REJ frame or before the completion of a link resetting procedure.

2.4.5.8 *STE busy condition*

When the STE enters a busy condition, it will transmit an RNR frame at the earliest opportunity. The RNR frame will be a command frame with the P bit set to 1 if an acknowledged transfer of the busy condition indication is required; otherwise the RNR frame may be either a command or response frame. While in the busy condition, the STE will accept and process supervisory frames, will accept and process the contents of the N(R) fields of I frames, and will return an RNR response with the F bit set to 1 if it receives a supervisory command or I command frame with the P bit set to 1. To clear the busy condition, the STE will transmit either an REJ frame or an RR frame, with N(R) set to the current receive state variable V(R), depending on whether or not it discarded information fields of correctly received I frames. The REJ frame or the RR frame will be a command frame with the P bit set to 1 if an acknowledged transfer of the busy-to-non-busy transition is required, otherwise the REJ frame or the RR frame may be either a command or a response frame.

2.4.5.9 *Waiting acknowledgement*

If Timer T1 runs out waiting for the acknowledgement from the other STE for an I frame transmitted, the STE will enter the timer recovery condition, add one to its transmission attempt variable and set an internal variable "x" to the current value of its send state variable V(S). The STE will then restart Timer T1, set its send state variable to the last value of N(R) received from the other STE and retransmit the corresponding I frame with the P bit set to 1, or transmit an appropriate supervisory command frame (RR, RNR or REJ) with the P bit set to 1.

The timer recovery condition is cleared when the STE receives a valid supervisory frame with the F bit set to 1.

If, while in the timer recovery condition, the STE correctly receives a supervisory frame with the F bit set to 1 and with the N(R) within the range from its current send state variable V(S) to x included, it will clear the timer recovery condition (including stopping Timer T1) and set its send state variable V(S) to the value of the received N(R), and may then resume with I frame transmission or retransmission, as appropriate.

If, while in the timer recovery condition, the STE correctly receives an I frame or a supervisory frame with the P/F bit set to 0 and with a valid N(R) (see § 2.3.4.9) within the range from its current send state variable V(S) to x included, it will not clear the timer recovery condition. The value of the received N(R) may be used to update the send state variable V(S). However, the STE may decide to keep the last transmitted I frame in store (even if it is acknowledged) in order to be able to retransmit it when the P bit set to 1 when Timer T1 runs out at a later time.

If Timer T1 runs out in the timer recovery condition, the STE will add one to its transmission attempt variable, restart Timer T1, and either retransmit the I frame sent with the P bit set to 1 or transmit an appropriate supervisory command with the P bit set to 1.

If the transmission attempt variable is equal to N2, the STE will initiate a link resetting procedure as described in § 2.4.7.2 below. N2 is a system parameter (see § 2.4.8.4 below).

2.4.6 *Conditions for link resetting or link reinitialization (link set-up)*

2.4.6.1 When the STE receives, during the information transfer phase, a frame which is not invalid (see § 2.3.5.3) with one of the conditions listed in § 2.3.4.9 above, the STE will request the other STE to initiate a link resetting procedure by transmitting an FRMR response as described in § 2.4.7.3.

2.4.6.2 When the STE receives, during the information transfer phase, an FRMR response from the other STE, the STE will initiate the link resetting procedures as described in § 2.4.7.2.

2.4.7 *Procedure for link resetting*

2.4.7.1 The link resetting procedure is used to initialize both directions of information transfer according to the procedure described below. The link resetting procedure only applies during the information transfer phase.

2.4.7.2 The link resetting procedure indicates a clearance of the busy condition, if present.

The STE will initiate a link resetting by transmitting an SABM/SABME command to the other STE and starting its Timer T1 (see § 2.4.8.1 below). Upon reception of a UA response from the other STE, the STE will reset its send and receive state variables V(S) and V(R) to zero, will stop its Timer T1, and will remain in the information transfer phase. Upon reception of a DM response from the DTE as a denial to the link resetting request, the STE will stop its Timer T1 and will enter the disconnected phase.

If upon receipt of the SABM/SABME command correctly, the STE determines that it can continue in the information transfer phase, it will return a UA response, will reset its send and receive state variables V(S) and V(R) to zero, and will remain in the information transfer phase. If, upon receipt of the SABM/SABME command correctly, the STE determines that it cannot remain in the information transfer phase, it will return a DM response as a denial to the resetting request and will enter the disconnected phase.

The STE, having sent an SABM/SABME command, will ignore and discard any frames except an SABM/SABME or DISC command, UA or DM response received. The receipt of an SABM/SABME or DISC command from the other STE will result in a collision situation that is resolved per § 2.4.4.5 above. Frames other than the UA or DM response sent in response to a received SABM/SABME or DISC command will be sent only after the link is reset and if no outstanding SABM/SABME command exists.

After the STE sends the SABM/SABME command, if a UA or DM response is not received correctly, Timer T1 will run out. The STE will then resend the SABM/SABME command and will restart Timer T1. After N2 attempts to reset the link, the STE will initiate appropriate higher layer recovery action and will enter the disconnected phase. The value of N2 is defined in § 2.4.8.4 below.

2.4.7.3 The STE may ask the other STE to reset the link by transmitting an FRMR response (see § 2.4.6.1 above).

After transmitting an FRMR response, the STE will enter the frame rejection condition. The frame rejection condition is cleared when the STE receives or transmits an SABM/SABME or DISC command. Any other frame received while in the frame rejection condition will cause the STE to retransmit the FRMR response with the same information field as originally transmitted.

The STE may start Timer T1 on transmission of the FRMR response. If Timer T1 runs out before the frame rejection condition is cleared the STE may retransmit the FRMR response, and restart Timer T1. After N2 attempts to get the other STE to reset the link, the STE may reset the link itself as described in § 2.4.7.2 above. The value of N2 is defined in § 2.4.8.4 below.

In the frame rejection condition, I frames and supervisory frames will not be transmitted. Also, received I frames and supervisory frames will be discarded by the STE except for the observance of a P bit set to 1. When an additional FRMR response must be transmitted as a result of the receipt of a P bit set to 1 while Timer T1 is running, Timer T1 will continue to run.

Upon reception of a FRMR response (even during a frame rejection condition), the STE will initiate a resetting procedure by transmitting a SABM/SABME command as described in § 2.4.7.2 above.

2.4.8 *List of system parameters*

The system parameters are as follows:

2.4.8.1 *Timer T1*

The period of Timer T1, at the end of which transmission of a frame may be initiated, is a system parameter agreed for a period of time between the Administrations.

The period of Timer T1 will take into account whether the timer is started at the beginning or end of transmission of the frame in the STE.

The proper operation of the procedure requires that the transmitter's Timer T1 be greater than the maximum time between transmission of a frame (SABM/SABME, DISC, I for supervisory command, or DM or FRMR response) and the reception of the corresponding frame) returned as an answer to that frame (UA, DM or acknowledging frame). Therefore, the receiver STE should not delay the response or acknowledging frame returned to one of the above frames by more than a value T2, where T2 is a system parameter (see § 2.4.8.2).

The STE will not delay the response or acknowledging frame returned to one of the above frames by more than a period T2.

2.4.8.2 *Parameter T2*

The period of parameter T2 shall indicate the amount of time available at the STE before the acknowledging frame must be initiated in order to ensure its receipt by the other STE prior to Timer T1 running out at the STE (parameter T2 < Timer T1).

2.4.8.3 *Timer T3*

The STE shall support a Timer T3 system parameter, the value of which shall be made known to both STEs.

The period of Timer T3, at the end of which an indication of an observed excessively long idle channel state condition is passed to the packet layer or the MLP, shall be sufficiently greater than the period of the Timer T1 (i.e., T3 > T1) so that the expiration of T3 provides the desired level of assurance that the link channel is in a non-active, non-operational state, and is in need of link set up before normal link operation can resume.

2.4.8.4 *Maximum number of attempts to complete a transmission, N2*

The value of the maximum number N2 of transmission and retransmissions of a frame following the running out of Timer T1 is a system parameter agreed for a period of time between Administrations. The value of N2 can be different in STE-X and STE-Y.

2.4.8.5 *Maximum number of bits in an I frame, N1*

The maximum number of bits in an I frame (excluding flags and 0 bits inserted for transparency) is a system parameter which depends upon the maximum length of the information fields transferred across the X/Y interface.

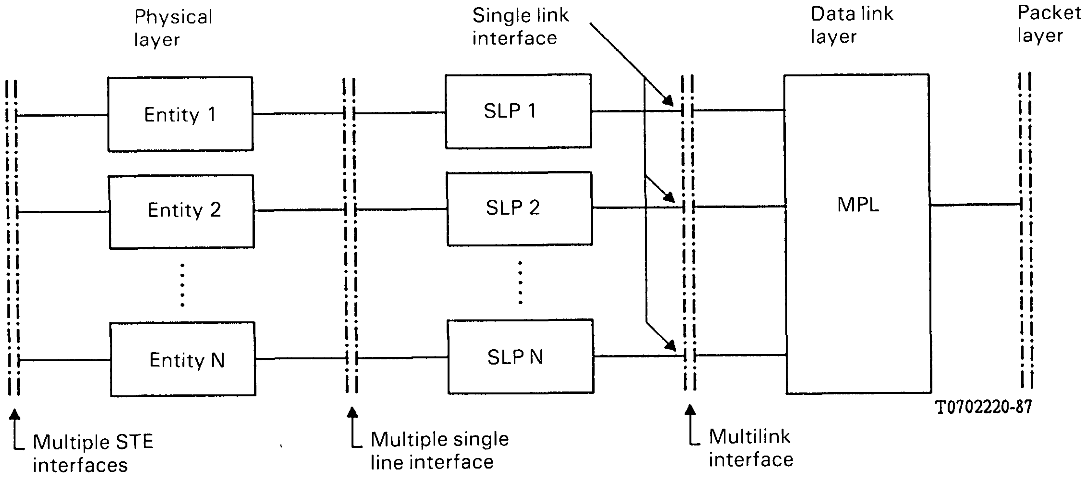
Note – When multilink procedures are used, N1 shall allow for the multilink control field (MLC). See § 2.5.2 below. Appendix II of Recommendation X.25 provides additional information on N1. The utility field has to be added.

2.4.8.6 *Maximum number of outstanding I frames, k*

The maximum number (k) of sequentially numbered I frames that the STE may have outstanding (i.e., unacknowledged) at any given time is a system parameter which can never exceed 7/127 (modulo 8/modulo 128). It shall be agreed for a period of time between Administrations and shall have the same value for both the STEs.

2.5 *Multilink procedures (MLP)*

The multilink procedure (MLP) exists as an added upper sublayer of the data link layer, operating between the packet layer and a multiplicity of single data link protocol functions (SLPs) in the data link layer (see Figure 2/X.75).



SLP Single link procedure
MLP Multilink procedure

FIGURE 2/X.75

Multilink functional organization

A multilink procedure (MLP) must perform the functions of distributing across the available SLPs, packets which are to be transmitted to the remote STE and of resequencing packets received from the remote STE for delivery to the packet layer.

Note 1 – In § 2.5.4.4 (MT1 expiry) and § 2.5.4.5 (retransmission), other mechanisms can be envisaged to achieve the same functions.

Note 2 – In § 2.5.5.4 (MN1), § 2.5.5.1 (MT1) and § 2.5.5.2 (MT2) other mechanisms can be envisaged to achieve the same functions.

2.5.1 *Field of application*

The optional multilink procedure (MLP) described below is used for data interchange over one or more single link procedures (SLPs), each conforming to the description in §§ 2.2, 2.3 and 2.4, in parallel between two STEs. The multilink procedure provides the following general features:

a) achieve economy and reliability of service by providing multiple SLPs between two STEs;

b) permit addition and deletion of SLPs without interrupting the service provided by the multiple SLPs;

c) optimize bandwidth utilization of a group of SLPs through load sharing;

d) achieve graceful degradation of service when an SLP(s) fails;

e) provide each multiple SLP group with a single logical data link layer appearance to the packet layer, and

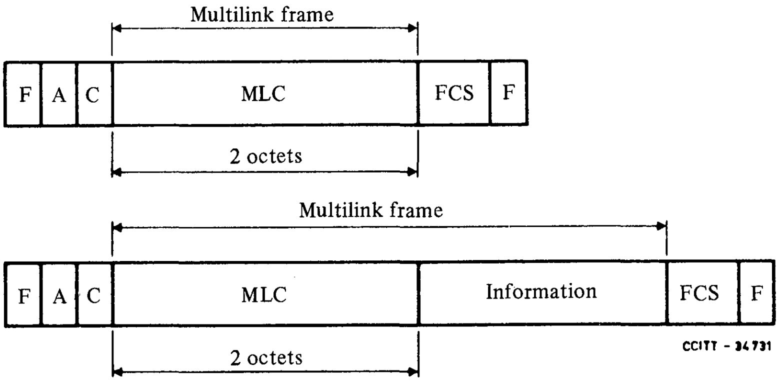
f) provide sequencing of the received packets prior to delivering them to the packet layer.

2.5.2 *Multilink frame structure*

All information transfers over an SLP are in multilink frames conforming to one of the formats shown in Table 9/X.75.

TABLE 9/X.75

Multilink frame formats



2.5.2.1 *Multilink control field*

The multilink control field (MLC) consists of two octets and its contents are described in § 2.5.3.

2.5.2.2 *Multilink information field*

The information field of a multilink frame, when present, follows the MLC. See § 2.5.3.2.3, § 2.5.3.2.4 and § 4 for the various codings and grouping of bits in the multilink information field.

2.5.3 *Multilink control field format and parameters*

2.5.3.1 *Multilink control field format*

The relationship shown in Table 10/X.75 exists between the order of bits delivered to/received from an SLP and the coding of the fields in the multilink control field.

2.5.3.2 *Multilink control field parameters*

The various paramaters associated with the multilink control field format are described below. See Table 10/X.75 and Figure 2/X.75.

2.5.3.2.1 *Void sequencing bit (V)*

The void sequencing bit (V) indicates if a received multilink frame shall be subjected to sequencing constraints. V set to 1 means sequencing shall not be required. V set to 0 means sequencing shall be required.

Note – For the purpose of this Recommendation, this bit shall be set to 0.

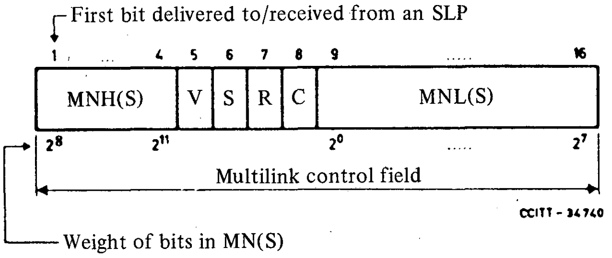
2.5.3.2.2 *Sequence check option bit (S)*

The sequence check option bit (S) is only significant when V is set to 1 (indicating that sequencing of received multilink frames shall not be required). S set to 1 shall mean no MN(S) number has been assigned. S set to 0 shall mean an MN(S) number has been assigned, so that although sequencing shall not be required, a duplicate multilink frame check may be made, as well as a missing multilink frame identified.

Note – For the purpose of this Recommendation, this bit shall be set to 0.

TABLE 10/X.75

Multilink control field format



MNH(S)	Bits 9-12 of 12-bit multilink send sequence number MN(S)
MNL(S)	Bits 1-8 of 12-bit multilink send sequence number MN(S)
V	Void sequencing bit
S	Sequence check option bit
R	MLP reset request bit
C	MLP reset confirmation bit

2.5.3.2.3 *MLP reset request bit (R)*

The MLP reset request bit (R) is used to request a multilink reset (see § 2.5.4.2). R set to 0 is used in normal communication; i.e., no request for a multilink reset. R set to 1 is used by the STE MLP to request the reset of the remote MLP state variables. In this R = 1 case, the multilink information field does not contain packet layer information, but may contain an optional 8-bit cause field that incorporates the reason for the reset.

Note – The encoding of the cause field is a subject for further study.

2.5.3.2.4 *MLP reset confirmation bit (C)*

The MLP reset confirmation bit (C) is used in reply to an R bit set to 1 (see § 2.5.3.2.3) to confirm the resetting of the multilink state variables (see § 2.5.4.2). C set to 0 is used in normal communication; i.e., no multilink reset request has been activated. C set to 1 is used by the STE MLP in reply to a multilink frame from the remote STE with R set to 1, and indicates that the MLP state variable reset process has been completed. In this C = 1 case, the multilink frame is used without an information field.

2.5.3.2.5 *Multilink send state variable MV(S)*

The multilink send state variable MV(S) denotes the sequence number of the next in-sequence multilink frame to be assigned to an SLP. The variable can take on the value 0 through 4095 (modulus 4096). The value of MV(S) is incremented by 1 with each successive multilink frame assignment.

2.5.3.2.6 *Multilink sequence number MN(S)*

Multilink frames contain the multilink sequence number MN(S). Prior to the assignment of an in-sequence multilink frame, the value of MN(S) is updated to equal the value of the multilink send state variable MV(S). The multilink sequence number is used to resequence and to detect missing and duplicate multilink frames at the receiver before the contents of a multilink frame information field is delivered to the packet layer.

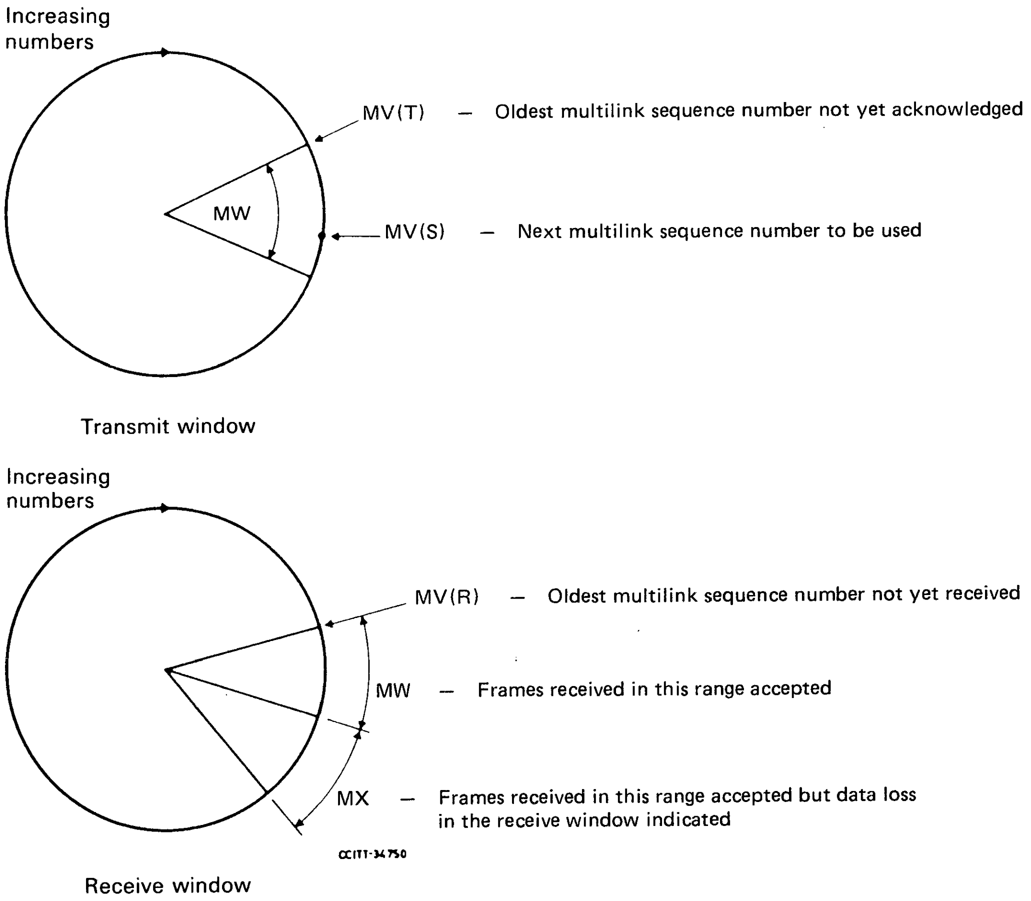


FIGURE 3/X.75

Parameters

2.5.3.2.7 *Transmitted multilink acknowledged state variable MV(T)*

MV(T) is the state variable at the transmitting STE denoting the oldest multilink frame which is awaiting an indication that a local SLP has received an acknowledgement from its remote SLP. This variable MV(T) can take on the value 0 through 4095 (modulus 4096). Some multilink frames with sequence numbers higher than MV(T) may already have been acknowledged.

2.5.3.2.8 *Multilink receive state variable MV(R)*

The multilink receive state variable MV(R) denotes the sequence number at the receiving STE of the next in-sequence multilink frame to be received and delivered to the packet layer. This variable MV(R) can take on the value 0 through 4095 (modulus 4096). The value of MV(R) is updated as described in § 2.5.4.4 below. Multilink frames with higher sequence numbers in the MLP receive window may already have been received.

2.5.3.2.9 *Multilink window size MW*

MW is the maximum number of sequentially numbered multilink frames that the STE may transfer to its SLPs beyond the lowest numbered multilink frame which has not as yet been acknowledged. MW is a system parameter which can never exceed (4095 – MX).

The value of MW shall be agreed between Administrations and shall have the same value for both STEs for a given direction of information transfer.

Note – Factors which will affect the value of parameter MW include, but are not limited to single link transmission and propagation delays, the number of links, the range of multilink frame lengths, and SLP parameters N2, T1 and *k*.

The MLP transmit window contains the sequence numbers MV(T) to [MV(T) + MW – 1] inclusive.

The MLP receive window contains the sequence numbers MV(R) to [MV(R) + MW – 1] inclusive. Any multilink frame received within this window shall be delivered to the packet layer when its MN(S) is the same as MV(R).

2.5.3.2.10 *Receive MLP window guard region MX*

MX is a system parameter which defines a guard region of multilink sequence numbers of fixed size beginning at [MV(R) + MW]. The range of MX shall be large enough for the receiving MLP to recognize the highest MN(S) outside of its receive window that it may legitimately receive after a multilink frame loss has occurred.

A multilink frame with sequence number MN(S) = Y received in this guard region indicates that those missing multilink frame(s) in the range MV(R) to [Y – MW] has(have) been lost. MV(R) is then updated to [Y – MW + 1].

Note – A number of methods may be selected in calculating a value for the guard region MX:

a) In a system where the transmission MLP assigns h_i in-sequence contiguous multilink frames at a time to the i^{th} SLP, MX should be greater than or equal to the sum of $[h_i + 1 - h_{\min}]$, where $h_{\min}$ equals the smallest h_i encountered. Where there are L SLPs in the multilink group, MX should be greater than or equal to:

$$\sum_{i=1}^{L} h_i + 1 - h_{\min}\,;\ \textit{or}$$

b) In a system where the transmitting MLP assigns on a rotation basis h in-sequence, contiguous multilink frames at a time to each SLP, MX at the receiving MLP should be greater than or equal to $[h(L - 1) + 1]$, where L is the number of SLPs in the multilink group; or

c) MX should be no larger than MW.

Additional methods of selecting MX values are for further study.

2.5.4 *Description of multilink procedure (MLP)*

The procedure below is presented from the perspective of the transmitter and receiver of multilink frames.

The arithmetic is performed modulo 4096.

2.5.4.1 *Initialization*

The STE will perform an MLP initialization by first resetting MV(S), MV(T), and MV(R) to zero and then initializing each of its SLPs. Upon successful initialization of at least one of the SLPs, the STE shall perform the multilink resetting procedure as described in § 2.5.4.2. An SLP initialization is performed according to § 2.4.4.1 of this Recommendation.

Note – An SLP that cannot be initialized should be declared out of service and appropriate recovery action should be taken.

2.5.4.2 *Multilink resetting procedure*

The multilink resetting procedure provides the mechanism for synchronizing the sending and receiving MLPs in both STEs when deemed necessary by either STE. Exact cases when a MLP reset procedure will be invoked are for further study. Following a successful multilink resetting procedure, the multilink sequence numbering in each direction begins with the value 0.

Appendix I provides examples for the multilink resetting procedures when initiated by either a single STE or by both STEs simultaneously.

A multilink frame with R = 1 is used to request multilink reset, and a multilink frame with C = 1 confirms that the multilink reset process has been completed. An MLP resets MV(S) and MV(T) to zero on transfer of a multilink frame with R = 1 and resets the MV(R) to zero on receipt of a multilink frame with R = 1.

When the MLP initiates the resetting procedure, it removes all of the unacknowledged multilink frames that are held in that MLP and its associated SLPs, and retains control of those frames. Hereafter, the initiating MLP does not transmit a multilink frame with R = C = 0 until the reset process is completed. (One method to remove multilink frames in the SLP is to disconnect the link of that SLP.) The initiating MLP then resets its multilink send state variable MV(S) and its transmitted multilink frame acknowledged state variable MV(T) to zero. The initiating MLP then transmits a multilink frame with R = 1 as a reset request on one of its SLPs and starts Timer MT3. The value of the MN(S) field in the R = 1 frame may be any value, since when R = 1 the MN(S) field is ignored by the receiving MLP. The initiating MLP continues to receive and process multilink frames from the remote MLP, in accordance with the procedures as described in § 2.5.4.4 until it receives a multilink frame with R = 1 from the remote MLP.

An MLP which has received a multilink frame with R = 1 (reset request) in the normal communication status from an initiating MLP starts the operation as described above; the MLP should receive no multilink frames with R = C = 0 until the reset process is completed. Any such frame received is discarded. When the MLP has already initiated its own multilink resetting procedure and has transferred the multilink frame with R = 1 to one of its SLPs for transmission, that MLP does not repeat the above operation upon receipt of a multilink frame with R = 1 from the remote MLP.

Receipt of a frame with R = 1 (reset request) causes the receiving MLP to deliver to the packet layer those packets already received and to identify those multilink frames transmitted but unacknowledged. The packet layer may be informed of the packet loss at the original value of MV(R) and at any subsequent value(s) of MV(R) for which there has been no multilink frame received up to and including the highest numbered multilink frame received. The receiving MLP then resets its multilink receive state variable MV(R) to zero.

After an MLP transmits a multilink frame with R = 1 on one of its SLPs, it shall receive confirmation of successful transfer from that SLP as one of the conditions before transmitting a multilink frame with C = 1; when the initiating MLP then received a multilink frame with R = 1, and has completed the variable resetting operation above, the initiating MLP transmits a multilink frame with C = 1 (reset confirmation) to the remote MLP. When an MLP has

1) received a multilink frame with R = 1,
2) sent a multilink frame with R = 1 on one of its SLPs, and
3) completed the variable resetting operation above,

that MLP then transmits a multilink frame with C = 1 (reset confirmation) to the initiating MLP as soon as possible, given that confirmation of the transfer of the R = 1 multilink frame has been received from that SLP. The C = 1 multilink frame is a reply to the multilink frame with R = 1. The value of the MN(S) field in the above C = 1 frame may be any value, since with C = 1 the MN(S) field is ignored by the receiving MLP. The multilink sequence number MN(S) received in each direction following multilink reset will begin with the value zero.

When an MLP uses the same or only one SLP to transmit the multilink frame with C = 1, the MLP can transmit the multilink frame with C = 1 immediately after the multilink frame with R = 1 without waiting for SLP indication of transfer completion. An MLP may use two different SLPs as long as one is used for transmitting the multilink frame with R = 1 and the other is used for transmitting the multilink frame with C = 1 following receipt of the SLP indication of successful transmission of the R = 1 multilink frame. A multilink frame with R = C = 1 is never used and will be discarded if received.

When an MLP receives the multilink frame with C = 1, the MLP stops its Timer MT3. The successful transmission of the multilink frame with C = 1 to the remote MLP and the reception of a multilink frame with C = 1 from the remote MLP completes the resetting procedure. The first multilink frame transmitted with R = C = 0 shall have a multilink sequence number MN(S) value of zero. (The originating MLP, having successfully delivered a multilink frame with C = 1 to the remote MLP, and having received a multilink frame with C = 1, could immediately transmit multilink frames with R = C = 0. However, to insure that the multilink frames with R = C = 0 are not discarded because they arrive at the remote MLP prior to the SLP acknowledgement of the reception of the C = 1 multilink frame, the MLP should use the same SLP as that which acknowledged receipt of the multilink frame with C = 1.)

When the initiating MLP receives a multilink frame with C = 1 without having received a multilink frame with R = 1, it will retransmit the multilink frame with R = 1 and restart its Timer MT3.

When an MLP additionally receives one or more multilink frames with R = 1 between receiving a multilink frame with R = 1 and transmitting a multilink frame with C = 1, the MLP shall discard the extra multilink frames with R = 1. When an MLP receives a multilink frame with C = 1, which is not a reply to a multilink frame with R = 1, the MLP shall discard the multilink frame with C = 1.

After an MLP transmits a multilink frame with C = 1 on one of its SLPs, the MLP may receive a multilink frame with R = 1 from the remote MLP. The MLP shall regard the multilink frame with R = 1 as a new reset request and shall start the multilink resetting procedure from the beginning.

When Timer MT3 runs out, the MLP restarts the multilink resetting procedure from the beginning. The value of Timer MT3 shall be large enough to include the transmission, retransmission and propagation delays in the SLPs, and the operation time of the MLP that receives a multilink frame with R = 1 and responds with a multilink frame with C = 1.

2.5.4.3 *Transmitting multilink frames*

2.5.4.3.1 *General*

The transmitting STE MLP shall be responsible for controlling the flow of packets from the packet level into multilink frames and then to the SLPs for transmission to the receiving STE MLP.

The functions of the transmitting STE MLP shall be to:

1) accept packets from the packer layer;
2) allocate multilink control fields, containing the appropriate sequence number MN(S), to the packets;
3) assure that MN(S) is not assigned outside the MLP transmit window (MW);
4) pass the resultant multilink frames to the SLPs for transmission;
5) accept indications of successful transmission acknowledgements from the SLPs;
6) monitor and recover from transmission failures or difficulties that occur at the SLP sublayer; and
7) accept flow control indications from the SLPs and take appropriate actions.

2.5.4.3.2 *Transmission of multilink frames*

When the transmitting MLP accepts a packet from the packet layer, it shall place the packet in a multilink frame, set the MN(S) equal to MV(S), assure that MN(S) is not assigned outside the transmit window (MW), set V, S, R and C to 0, and then increment MV(S) by 1.

In the following, incrementing send and receive state variables is in reference to a continuously repeated sequence series, i.e., 4095 is 1 higher than 4094, and 0 is 1 higher than 4095 for modulo 4096 series.

If the MN(S) is less than MV(T) + MW, and the remote STE has not indicated a busy condition on all available links, the transmitting MLP may then assign the new multilink frame to an available link. The transmitting MLP shall always assign the lowest MN(S) unassigned multilink frame first. Also, the transmitting

MLP may assign a multilink frame to more than one link. When the SLP successfully completes the transmission of a multilink frame(s) by receiving an acknowledgement from the remote SLP, it shall indicate this to the transmitting MLP. The transmitting MLP may then discard the acknowledged multilink frame(s). As the transmitting STE receives new indications of acknowledgements from the SLPs, MV(T) shall be advanced to denote the lowest numbered multilink frame not yet acknowledged.

Whenever an SLP indicates that it has attempted to transmit a multilink frame N2 times, the MLP will then assign the multilink frame to the same or one or more other links, unless the MN(S) has been acknowledged on some previous link. The MLP shall always assign the lowest MN(S) frame first.

Note 1 – If an MLP implementation is such that a multilink frame is transmitted on more than one link (e.g., to increase the probability of successful delivery) there is a possibility that one of these multilink frames (i.e., a duplicate) may be delivered to the remote MLP after an earlier one has been acknowledged [the earlier multilink frame would have resulted in the receiving remote MLP having incremented its MV(R) and the transmitting MLP having incremented its MV(T)]. To ensure that an old duplicate multilink frame is not mistaken for a new frame by the receiving remote MLP, it is required that the transmitting MLP shall never send a new multilink frame with MN(S) equal to MN(S)′ – MW – MX, where MN(S)′ is associated with a duplicate multilink frame that is being transmitted on other SLPs, until all SLPs have either successfully transferred the multilink frame or retransmitted the frame their maximum number of times. Alternatively, the incrementing of MV(T) may be withheld until all SLPs have either successfully transferred the multilink frame or retransmitted the frame their maximum number of times. These and other alternatives are for further study.

Flow control is achieved by the window size parameter MW, and through busy conditions being indicated by the remote SLPs.

The MLP will not assign a multilink frame with an MN(S) greater than MV(T) + MW – 1. At the point where the next multilink frame to be assigned has a MN(S) = MV(T) + MW, the MLP shall hold this and subsequent multilink frames until an indication of acknowledgement that advances MV(T) is received from the SLPs.

The remote MLP may exercise flow control of the MLP by indicating a busy condition over one or more remote STE SLPs. The number of SLPs made busy will determine the degree of MLP flow control realized. When the MLP receives an indication of a remote SLP busy condition from one or more of its SLPs, the MLP may reassign any unacknowledged multilink frames that were assigned to those SLPs. The MLP will assign the multilink frames containing the lowest MN(S) to an available SLP as specified above.

In the event of a circuit failure, an SLP reset or SLP disconnection, all multilink frames unacknowledged on an SLP link shall be retransmitted on an operational SLP(s) which is(are) not in the busy condition.

Note 2 – The action to be taken on the receipt of an RNR frame by the SLP whose unacknowledged multilink frames have been removed for further study.

Note 3 – The means of detecting transmitting MLP malfunctions (e.g., sending more than MW multilink frames) and the actions to be taken are for further study.

2.5.4.4 *Receiving multilink frames*

Any multilink frame less than two octets in length shall be discarded by the receiving STE.

Note 1 – The procedures to be followed by the receiving STE when V and/or S is equal to 1 are for further study.

When the STE receives multilink frames from one of its SLPs, the STE will compare the multilink sequence number MN(S) of the received multilink frame to its multilink receive state variable MV(R), and act on the frame as follows:

a) If the received MN(S) is equal to the current value of MV(R), i.e., is the next expected in-sequence multilink frame, the MLP delivers the packet to the packet layer.

b) If the MN(S) is greater than the current value of MV(R) but less than [MV(R) + MW + MX], the MLP keeps the received multilink frame until condition a) is met, or discards it if it is a duplicate.

c) If the MN(S) is other than that in a) and b) above, the multilink frame is discarded.

Note 2 – In case c above the recovery from the desynchronization greater than MX between the local and the remote MLP, i.e., the value of MN(S) assigned to new multilink frames at the remote MLP is higher than MV(R) + MW + MX at the local MLP, is for further study.

On receipt of a multilink frame, MV(R) is incremented in the following way:

i) If MN(S) is equal to the current value of MV(R), the MV(R) is incremented by the number of consecutive in-sequence multilink frame received. If additional multilink frames are awaiting delivery pending receipt of a multilink frame with MN(S) equal to MV(R), then Timer MT1 (see § 2.5.5.1) is restarted; otherwise MT1 is stopped.

ii) If MN(S) is greater than the current value of MV(R) but less than MV(R) + MW, MV(R) remains unchanged. Timer MT1 is started, if not already running.

iii) If MN(S) is $\geqslant$ MV(R) + MW but $<$ MV(R) + MW + MX, MV(R) is incremented to MN(S) − MW + 1 and then the packet layer may be informed of the packet loss at the original value of MV(R). As MV(R) is being incremented, if the multilink frame with MN(S) = MV(R) has not yet been received, the packet layer may be informed of the packet loss also; if the multilink frame with MN(S) = MV(R) has been received, it is delivered to the packet layer. After MV(R) reaches MN(S) − MW + 1, it may then be incremented further as above until the first unacknowledged MN(S) is encountered (see Figure 4/X.75).

iv) If the MN(S) is other than that in i), ii) and iii) above, MV(R) remains unchanged.

If Timer MT1 runs out, MV(R) is incremented to MN(S) of the next multilink frame awaiting delivery to the packet layer and then the packet layer may be informed of the packet loss at the original MV(R). The procedure follows i) and a) above as long as there are consecutive in-sequence multilink frames which have been received.

When flow control of the other MLP is desired, one or more SLP(s) may be made to indicate a busy condition. The number of remote SLPs made busy determines the degree of flow control realized.

If the MLP can exhaust its receive buffer capacity before resequencing can be completed, Timer MT2 (see § 2.5.5.2 below) may be implemented. Whenever a busy condition is indicated by the MLP on all its SLPs, and multilink frames at the MLP are awaiting resequencing, Timer MT2 shall be started. When the busy condition is cleared on one or more SLPs by the MLP, Timer MT2 shall be stopped.

If Timer MT2 runs out, the multilink frame with MN(S) = MV(R) is blocked and shall be considered lost. MV(R) shall be incremented to the next sequence number not yet received, and the packets contained in multilink frames with intervening multilink sequence numbers are delivered to the packet layer. Timer MT2 shall be restarted if the busy condition remains in effect on all SLPs and more multilink frames are awaiting resequencing.

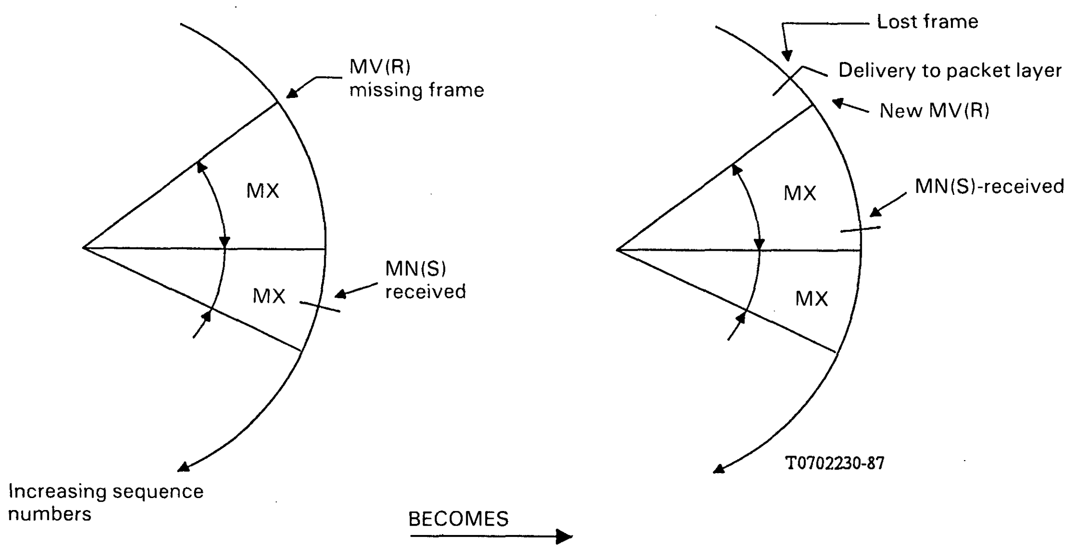


FIGURE 4/X.75

Detecting lost multilink frames

2.5.4.5 *Retransmission of multilink frames*

If an SLP has retransmitted a multilink frame MN1 times, the STE will then assign the multilink frame to the same or one or more other links, unless the MN(S) has been acknowledged on some previous link. The STE shall always reassign the lowest MN(S) frame first. The first SLP transmits the frame N2 times, regardless of the value of MN1.

Note – The procedures associated with the reassigning of multilink frames from a link of poor quality (e.g., before N2 transmissions) to other links are for further study.

2.5.4.6 *Taking an SLP out of service*

An SLP may be taken out of service for maintenance, traffic, or performance considerations.

An SLP is taken out of service by disconnecting at the physical layer or the data link layer. Any outstanding multilink frames will be treated as in § 2.5.4.1. The usual procedure would be to flow control the remote SLP by an RNR, and then to disconnect logically the local SLP (see § 2.4.4.3 above).

If Timer T1 has run out N2 times and the SLP resetting procedure is unsuccessful, then the SLP will enter the disconnected phase, taking the SLP out of service (see §§ 2.4.5.8 and 2.4.7.2 above).

Note – In the case when all SLPs are out of service, the recovery mechanism is based on initiating the MLP reset procedure. Additional recovery procedures are for further study.

2.5.5 *List of multilink system parameters*

2.5.5.1 *Lost-frame timer MT1*

Timer MT1 is used at a receiving STE to provide a means to identify during low traffic periods that the multilink frame with MN(S) equal to MV(R) is lost.

2.5.5.2 *Group busy timer MT2*

Timer MT2 is provided at a receiving STE to identify a "blocked" multilink frame condition (e.g., a buffer exhaust situation) that occurs before required resequencing can be accomplished. MT2 is started when all SLPs are busy and there are multilink frames awaiting resequencing. If MT2 runs out before the "blocked" multilink frame MV(R) is received, the "blocked" multilink frame(s) is(are) declared lost. MV(R) is incremented to the value of the next in-sequence multilink frame to be received, and any packets intervening multilink frames are delivered to the packet layer.

Note – MT2 may be set to infinity; e.g., when the receiving STE always has sufficient storage capacity.

2.5.5.3 *MLP reset confirmation timer MT3*

Timer MT3 is used by the MLP to provide a means of identifying that the remote MLP multilink frame with the C bit set to 1 that is expected following the transmission of the MLP multilink frame with R bit set to 1 has not been received.

2.5.5.4 *Retransmission attempts MN1*

MN1 has a value between zero and the smallest N2 over all SLPs inclusive. If a multilink frame is to be retransmitted at the SLP sublayer, MN1 retries indicates when action may be taken at the MLP sublayer.

3 Packet layer procedures between signalling terminals

General principles

Section 3 of this Recommendation relates to the transfer of packets at the STE-X/STE-Y (X/Y) interface. The procedures apply to packets which are successfully transferred across the X/Y interface.

Each packet to be transferred across the X/Y interface shall be contained within the link layer information field which will delimit its length, and only one packet shall be contained in the information field of an I frame.

Note – Some networks require the data field of packets to contain an integral number of octets. The arrangements for interworking with such networks is subject to bilateral agreement between Administrations. The transmission by a DTE of data fields not containing an integral number of octets to the network may cause a loss of data integrity.

To enable simultaneous virtual calls and/or permanent virtual circuits, logical channels are used. Each virtual call and permanent virtual circuit is assigned a logical channel group number (in the range 0 to 15 inclusive) and a logical channel number (in a range of 0 to 255 inclusive). For virtual calls, a logical channel group number and a logical channel number are assigned during the call set-up phase. The range of logical channel and logical channel groups that are available for assignment to virtual calls is agreed bilaterally for a period of time. For permanent virtual circuits using the static method, a logical channel group number and a logical channel number are assigned at the time of establishment (see Recommendation X.181). Procedures for a dynamic method are for further study.

The combination of logical channel number 0 and logical channel group number 0 will not be used for virtual calls and permanent virtual circuits.

In the case that multiple STE X/Y interfaces are used between two networks, virtual calls may be distributed over the available STEs. STE selection may be performed once by the originating and each transit network for a call request. The procedure for selecting the particular X/Y interface is network dependent. During the existence of a particular virtual call, each packet related to that call uses the STEs selected at call set-up.

For permanent virtual circuit, each packet related to that circuit uses the STEs selected at establishment time of the permanent virtual circuit. In the case that multiple X/Y interfaces are used between two networks, bilateral agreement is necessary selecting the specific STE X/Y interface to be used.

In the case that multiple STE X/Y interfaces are used between two networks, the networks may apply network utilities and their parameters either in common or independently to the STE X/Y interfaces.

For virtual calls, it is assumed that the gathering of information required for charging and accounting should normally be the responsibility of the calling Administration (see Recommendation D.10). Other arrangements for gathering information are for further study. For permanent virtual circuit, responsibility of gathering information required for charging and accounting should normally be the source Administration (see Recommendation X.181).

The group of logical channels to be assigned for permanent virtual circuits has to be agreed bilaterally between Administrations.

3.1 *Procedures for virtual call set-up and clearing*

Virtual calls will be set up and cleared according to the procedures described hereunder. The procedures for calls set-up and clearing are only applicable when a logical channel is in the *packet layer ready* state (r1). In all other *r* states these procedures are not applicable.

3.1.1 *Ready state*

If there is no call or call attempt in existence and if call set-up is possible, the logical channel is in the *ready* state (p1), within the *packet layer ready* state (r1).

3.1.2 *Call request packet*

An STE indicates a call request by transferring a *call request* packet which specifies a logical channel in the *ready* state (p1) across the X/Y interface. The logical channel selected by the calling STE is then in the STE *call request* state (p2/3). If this state persists for more than T31, the calling STE will clear the call. The value of T31 is 200 seconds (see Annex D).

Note – In the *call request* packet, bit 7 of the general format identifier (see § 4.1.1) may be used in conjunction with the delivery confirmation procedure (see § 3.3.4). The bit 7 is conveyed transparently through an STE.

3.1.3 *Call connected packet*

The called STE will indicate acceptance of the call by the called DTE by transferring across the X/Y interface a *call connected* packet specifying the same logical channel as that of the *call request* packet. This places the specified logical channel in the *flow control ready* state (d1) within the *data transfer* state (p4). The procedure applying to the *data transfer* state is specified in § 3.3 below.

Note – In the *call connected* packet, bit 7 of the general format identifier (see § 4.1.1) may be used in conjunction with the delivery confirmation procedure (see § 3.3.4). This bit 7 is conveyed transparently through an STE.

3.1.4 *Call collision*

Call collision occurs if STE-X receives a *call request* packet when the logical channel specified is in state p2 or if the STE-Y receives a *call request* packet when the logical channel specified is in state p3. In these cases, both calls shall be cleared. The clearing cause field shall be coded "Network congestion".

In order to reduce the occurrence of this situation, inverse order testing of logical channels will be used. The *call request* packet of one STE will use the logical channel in the *ready* state with the lowest number; the *call request* packet of the other STE will use the logical channel in the *ready* state with the highest number. Which STE will use the lowest number and which the highest number will be agreed bilaterally.

3.1.5 *Clear request packet*

An STE may request clearing of a logical channel in any state by transferring across the X/Y interface a *clear request* packet specifying the logical channel. If the STE *clear request* state persists for more than T33, the actions taken by the STE are given in Annex D. The value of T33 is 180 seconds.

The clearing cause field will be coded according to the reason for clearing. Each STE shall be capable of generating the distinct codes for all of the call progress signals specified in Recommendation X.96 for the packet-switched data transmission service.

3.1.6 *Clear confirmation packet*

When an STE-X or STE-Y (STE X/Y) has received a *clear request* packet, it will free the logical channel, whatever the state of the logical channel except the STE X/Y *clear request* state (p6 or p7 respectively), and transfer across the X/Y interface a *clear confirmation* packet specifying the same logical channel. The logical channel is placed in the *ready* state (p1) within the *packet layer ready* state (r1). The receipt of a *clear confirmation* packet cannot be interpreted as an indication of the remote DTE being cleared.

3.1.7 *Clear collision*

If a logical channel is in the STE X/Y *clear request* state (p6 or p7 respectively) and the STE X/Y receives a *clear request* packet specifying the same logical channel, this STE will consider the clearing completed and will not transmit a *clear confirmation* packet. This logical channel is now in the *ready* state (p1) within the *packet layer ready* state(r1).

3.2 *Procedures for permanent virtual circuit service*

Figures B-1/X.75 and B-3/X.75 show the state diagrams which give a definition of events at the packet layer X/Y interface for logical channels assigned for permanent virtual circuits.

For permanent virtual circuits there is no call set-up or clearing. The procedures for the control of packets between STEs while in the *data transfer* state are contained in § 3.3.

In case of momentary failure within the network, the STE will reset the permanent virtual circuit as described in § 3.4.2, with the cause "Network congestion", and then will continue to handle data traffic.

If the network has a temporary inability to handle data traffic, the STE shall reset the permanent virtual circuit with the cause "Network out of order". When the network is again able to handle data traffic, the STE should reset the permanent virtual circuit with the cause "Network operational".

3.3 *Procedure for data and interrupt transfer*

The data transfer procedure described below applies independently to each logical channel existing at the X/Y interface.

Normal network operation dictates that user data in *data* packets and interrupt data are all passed transparently, unaltered through the network. The order of bits within these packets is preserved. A packet sequence received by an STE is always delivered as a complete packet sequence.

3.3.1 *States for data transfer*

Data, interrupt, flow control and *reset* packets may be transmitted and received by an STE in the *data transfer* state (p4) of the *packet layer ready* state (r1) of a logical channel at the X/Y interface. Only in this state, do the flow control and reset procedures described in § 3.4 below apply to data transmission on that logical channel to and from the STE. In all other *r* or *p* states the data and interrupt transfer, flow control, and reset procedures are not applicable.

3.3.2 *Numbering of data packets*

Each data packet transmitted at the X/Y interface for each direction of transmission in a virtual call or permanent virtual circuit is sequentially numbered. This sequential numbering is performed regardless of the layer of data [value of the qualifier (Q) bit].

The sequence numbering scheme of the packets is performed modulo 8 or 128. This modulo is common to all logical channels at the X/Y interface. The packet sequence numbers cycle through the entire range 0 to 7 or 0 to127 respectively. The selection of modulo 8 or 128 is done by bilateral agreement.

Only *data* packets contain this sequence number called the packet send sequence number P(S).

The first data packet to be transmitted across the X/Y interface for a given direction of data transmission when the logical channel has just entered the *flow control ready* state (d1), has a packet send sequence number equal to 0.

If an STE receives the first *data* packet with a packet send sequence number not equal to 0 after entering the *flow control ready* state (d1), it will reset the virtual call or permanent virtual circuit indicating the cause "Network congestion".

3.3.3 *Data field length of data packets*

The standard maximum data field length is 128 octets (1024 bits) and is provided by all Administrations. In addition for virtual calls, optional maximum data field lengths may be provided on a per call basis by bilateral agreement between Administrations in conjunction with an optional network utility defined in § 5.3.5 (see Note). For permanent virtual circuits, optional maximum data field length may be provided on a "per permanent virtual circuit" basis by bilateral agreement between Administrations and could be selected at establishment time. The value selected, in conjunction with the window size selected in § 3.4.1.1 has to satisfy the throughput class agreed between networks and end users at establishment time for a specific permanent virtual circuit. The attainable throughput at the STE X/Y interface is limited by the line characteristics and the traffic characteristics of other logical channels at the STE X/Y interface.

The data field length may contain any number of bits from 0 up to the agreed maximum data field length.

If an STE receives a *data* packet having a data field exceeding the maximum data field length, it will reset the virtual call or the permanent virtual circuit indicating the cause "Network congestion".

Note – Optional maximum data field lengths may be selected from the following list: 16, 32, 64, 256, 512 and 1024 octets. Maximum data field lengths of 2048 and 4096 octets are for further study.

3.3.4 *Delivery confirmation, more data and qualifier bits*

The setting of the *Delivery confirmation* bit (or D bit) is used to indicate whether or not an end-to-end acknowledgement of delivery is required for data being transmitted, this information being provided by means of the packet receive sequence number P(R) (see § 3.4.1.2).

A packet sequencing method is provided to enable coherent transmission of data longer than the maximum data field length of *data* packets.

Each complete packet sequence consists of any number (including 0) of full *data* packets (full means that the data field contains the bit number of the maximum data field length) with $M = 1$ and $D = 0$, followed by one other packet of any length up to (and including) the maximum with either $M = 0$ and $D = 0$ or 1, or $M = 1$ and $D = 1$. If an STE receives a packet which is not full, and which has the D bit set to 0 but the M bit set to 1, it will reset the virtual call or the permanent virtual circuit; the resetting cause shall be "Network congestion".

A complete packet sequence may be one of two levels as indicated by the *Qualifier* bit (or Q bit).

The value of the Q bit should not change within a complete packet sequence. If an STE detects that the value of this bit has changed within a packet sequence, it may reset the virtual call or the permanent virtual circuit; the resetting cause shall be "Network congestion".

Note – The value of the Q bit in a *data* packet, which follows a *data* packet with either $M = 0$ or both the M and D bits set to 1, may be set independently of the value of the Q bit in the previous packet.

3.3.5 *Interrupt procedure*

The interrupt procedure allows a DTE to transmit data to the remote DTE, without following the flow control procedure applying to *data* packets between STEs (see § 3.4 below). The interrupt procedure can only apply in the *flow control ready* state (d1) within the *data transfer* state (p4).

The interrupt procedure has no effect on the transfer and flow control procedures applying to the *data* packets on the virtual call or the permanent virtual circuit.

If an STE receives an *interrupt* packet with a user data field longer than 32 octets, the STE should reset the virtual call or the permanent virtual circuit.

An STE conveys an interrupt by transferring across the X/Y interface an *interrupt* packet. The other STE will convey the interrupt confirmation by transferring an *interrupt confirmation* packet.

The receipt of an *interrupt confirmation* packet indicates that the interrupt has been confirmed by the remote DTE by means of a *DTE interrupt confirmation* packet.

An *interrupt* packet is conveyed across the X/Y interface at or before the point in the stream of *data* packet at which it was generated by the DTE.

An STE receiving a further *interrupt* packet in the time between receiving one *interrupt* packet and transferring the *interrupt confirmation*, may either discard this *interrupt* packet or reset the virtual call or the permanent virtual circuit.

3.4 *Procedures for flow control and for reset*

The procedures for flow control of *data* packets and for reset only apply to the *data transfer* state (p4) and are specified below.

3.4.1 *Procedure for flow control*

At the X/Y interface of each logical channel used for a virtual call or a permanent virtual circuit, the transmission of *data* packets is controlled separately for each direction and is based on authorizations from the receiver.

3.4.1.1 *Window description*

At the X/Y interface of each logical channel used for a virtual call or a permanent virtual circuit, a window is defined for each direction of data transmission as the ordered set of W consecutive packet send sequence numbers of the *data* packets authorized to cross the interface.

The lowest sequence number in the window is referred to as the lower window edge. When a virtual call or a permanent virtual circuit at the X/Y interface has just been established or reset, the window related to each direction of data transmission has a lower window edge equal to 0. The packet send sequence number of the first *data* packet not authorized to cross the interface is the value of the lower window edge plus W (modulo 8 or 128).

The maximum value of the window size for each direction of transmission at the X/Y interface is common to all the logical channels and is agreed for a period of time bilaterally. This value does not exceed 7 or 127 (modulo 8 or 128).

For a particular virtual call or a permanent virtual circuit two window sizes may be selected, one for each direction of transmission. These window sizes may be less than or equal to the above-mentioned maximum. For virtual calls, the two sizes are selected by reference to a utility (see § 5.3.4) in the network utility field of the *call request* packet and the *call connected* packet, and, in some cases, by reference also to a correspondence table relating window size to throughput class. This table is agreed for a period of time between Administrations. For permanent virtual circuits two window sizes are selected at the establishment time and agreed between Administrations. The values selected in conjunction with the data field length selected in § 3.3.3 has to satisfy the throughput class agreed between networks and end users at establishment time for a specific permanent virtual circuit. The attainable throughput at the STE X/Y interface is limited by the line characteristics and the traffic characteristics of other logical channels at the STE X/Y interface.

3.4.1.2 *Flow control principles*

A number modulo 8 or 128 referred to as a packet receive sequence number P(R), conveys across the X/Y interface information from the receiver for the transmission of *data* packets. When transmitted across the X/Y interface, a P(R) becomes the lower window edge. In this way, additional *data* packets may be authorized by the receiver to cross the X/Y interface.

When the sequence number P(S) of the next *data* packet to be transmitted by the STE is within the window, the STE is authorized to transmit this *data* packet to the other STE, which may then accept it. When the sequence number P(S) of the next *data* packet to be transmitted by the STE is outside the window, the STE shall not transmit a *data* packet to the other STE. Otherwise, the other STE will consider the receipt of this *data* packet as a procedure error and will reset the virtual call or the permanent virtual circuit.

The packet receive sequence number, P(R), is conveyed in *data, receive ready* (RR) and *receive not ready* (RNR) packets, and implies that the STE transmitting the P(R) has accepted at least all *data* packets numbered up to and including [P(R) − 1].

The value of a P(R) received by the STE must be within the range starting from the last P(R) received by the STE up to and including the packet send sequence number of the next *data* packet to be transmitted by the STE. Otherwise, the STE will consider the receipt of this P(R) as a procedure error and will reset the virtual call or the permanent virtual circuit.

When the D bit is set to 0 in a *data* packet [P(S) = p], the significance of the P(R) [i.e., P(R) ⩾ p + 1] corresponding to that *data* packet is a local updating of the window across the packet layer interface.

When the D bit is set to 1 in a *data* packet [P(S) = p], the significance of the P(R) received corresponding to the *data* packet [i.e., P(R) ⩾ p + 1] is an indication that a P(R) has been received from the remote DTE for all data bits in the *data* packet in which the D bit had originally been set to 1 [i.e., P(S) = p].

Note 1 – The STE is required to send a P(R) corresponding to a *data* packet with the D bit set to 1 as soon as possible after it receives the P(R) from the remote DTE. An *RNR* packet may be used in this case if necessary.

Note 2 – In the case where a P(R) for a *data* packet with the D bit set to 1 is outstanding, local updating of the window will be deferred for subsequent *data* packets with the D bit set to 0. Some STEs may also defer updating of the window for previous *data* packets (within the window) with the D bit set to 0.

3.4.1.3 *STE receive ready (RR) packet*

RR packets are used by the STE to indicate that it is ready to receive the W *data* packets within the window starting with P(R), where P(R) is indicated in the *RR* packet.

3.4.1.4 *STE receive not ready (RNR) packet*

RNR packets are used by the STE to indicate a temporary inability to accept additional *data* packets for the virtual call or the permanent virtual circuit. An STE receiving an *RNR* packet shall stop transmitting *data* packets on the indicated logical channel but the window is updated by the P(R) indicated in the *RNR* packet.

The receive not ready situation indicated by the transmission of an *RNR* packet is cleared by the transmission in the same direction of an *RR* packet or by a reset procedure being initiated.

The transmission of an *RR* after an *RNR* at the packet layer is not to be taken as a demand for retransmission of packets which have already been transmitted.

3.4.2 *Procedure for reset*

The reset procedure is used to reinitialize the virtual call or the permanent virtual circuit. The reset procedure only applies in the *data transfer* state (p4) of the X/Y interface. In any other state of the interface the reset procedure is not applicable.

There are three states within the *data transfer* state (p4). They are *flow control ready* (d1), *STE-X reset request* (d2) and *STE-Y reset request* (d3). When entering state p4, the logical channel is placed in state d1.

When a virtual call or a permanent virtual circuit at the X/Y interface has just been reset, the window related to each direction of data transmission has a lower window edge equal to 0, and the numbering of subsequent *data* packets to cross the X/Y interface for each direction of data transmission shall start from 0.

3.4.2.1 *Reset request packet*

The STE shall indicate a request for reset by transmitting a *reset request* packet specifying the logical channel. This places the logical channel in the *reset request* state (d2 or d3).

In this state, the STE will discard *data, interrupt, RR* and *RNR* packets.

3.4.2.2 *Reset collision*

Reset collision occurs when both STEs simultaneously transfer a *reset request* packet. In this case both STEs shall consider that resetting is complete and shall not transfer a *reset confirmation* packet. The logical channel is then in the *flow control ready* state (d1).

3.4.2.3 *Reset confirmation packet*

When the logical channel is in the *reset request* state, the requested STE will confirm reset by transmitting to the requesting STE a *reset confirmation* packet. This places the logical channel in the *flow control ready* state (d1).

The *reset confirmation* packet can only be interpreted universally as having local significance; however, within some Administrations' networks, reset confirmation may have end-to-end significance. If the *reset request* state persists for more than T32, the actions taken by the STE are given in Annex D. The value of T32 is 180 seconds.

3.4.2.4 *Effect of reset procedure on data and interrupt packets*

Data and *interrupt* packets, transmitted by an STE before a reset procedure is initiated at its X/Y interface, will either be delivered before the corresponding reset procedure is initiated at the remote DTE/DCE interface, or discarded.

The first *data* and *interrupt* packets transmitted by an STE after a reset procedure is completed at its interface will be the first packets delivered after the corresponding reset procedure is completed at the remote DTE/DCE interface.

Data and *interrupt* packets transmitted by an STE after a reset procedure has been initiated by the other STE will be discarded by the latter STE until the reset procedure has been completed at the X/Y interference.

3.5 *Procedure for restart*

The restart procedure is used to clear simultaneously all the virtual calls and/or reset all the permanent virtual circuits at the X/Y interface.

There are three states of the X/Y interface concerned with the restart procedure. They are *packet layer ready* (r1), *STE-X restart request* (r2) and *STE-Y restart request* (r3). When entering state r1, all logical channels are placed in state p1.

3.5.1 *Restart by the STE*

The STE may at any time request a restart by transfering across the X/Y interface a *restart request* packet. The interface for each logical channel is then in the *request* state (r2 or r3).

In this state of the X/Y interface, the STE will discard all packet types except *restart request* and *restart confirmation* packets.

On receipt of a *restart request* packet, an STE shall clear all virtual calls and reset all permanent virtual circuits and shall place logical channels used for virtual calls in the *ready* state (p1) and the logical channels used for permanent virtual circuits in the *flow control ready* state (d1). The STE shall return a *restart confirmation* packet unless a collision has occurred.

The *restart confirmation* packet can only be interpreted universally as having local significance. If the *restart request* state persists for more than T30, the actions taken by the STE are given in Annex D. The value of T30 is 180 seconds.

3.5.2 *Restart collision*

Restart collision can occur when both STEs simultaneously transfer *restart request* packets. Under these circumstances, both STEs will consider that the restart is completed and will not expect a *restart confirmation* packet, neither will they transfer a *restart confirmation* packet.

3.6 *Relationship between layers*

Changes of operational states of the physical and link layer of the X/Y interface do not implicitly change the state of each logical channel at the packet layer. Such changes, when they occur, are explicitly indicated at the packet layer by the use of restart, clear or reset procedures as appropriate.

However, in some cases of trouble at the link layer, it may be appropriate to initiate the restart procedure, and accept no more new virtual calls or no more *data* packets on permanent virtual circuits.

A failure on the physical and/or link layer is defined as a condition in which the STE cannot transmit and receive any frames because of abnormal conditions caused by, for instance, a line fault between STEs.

When a failure on the physical and/or link layer is detected, virtual calls will be cleared and permanent virtual circuits will be declared out of order. The STE will transmit to the remote end in the network:

1) a reset with the cause "Network out of order" and the appropriate diagnostic for each permanent virtual circuit; and
2) a clear with the cause "Network congestion" and the appropriate diagnostic for each existing virtual call.

During the failure:

1) the STE will clear any virtual call with the cause "Network congestion" and an appropriate diagnostic;
2) for any *data* or *interrupt* packet received from the remote DTE on a permanent virtual circuit, the STE will reset the permanent virtual circuit with the cause "Network out of order" and an appropriate diagnostic;
3) a *reset request* packet received from the remote end on a permanent virtual circuit will be confirmed to the remote end by either a *reset confirmation* or *reset request* packet.

The appropriate diagnostic value depends on whether the failure was unexpected or the result of planned maintenance action; the values are No. 115 and No. 122 respectively (see also Note 3 of Annex E).

When the failure is recovered on the physical and link layers, the restart procedure will be actioned with the cause "Network operation" and a reset with the cause "Network operational" will be transmitted to both ends of each permanent virtual circuit going through the X/Y interface.

In other out-of-order conditions on the physical and/or link layer, the STE will clear virtual calls and reset permanent virtual circuits.

4 Packet formats for virtual calls and permanent virtual circuits

4.1 *General*

The formats of Recommendation X.75 packets are based on the general structure of packets in Recommendation X.25. It is anticipated that modification in Recommendation X.25 control packet formats will also be adopted in this Recommendation.

The possible extension of packet formats by the addition of new fields is for further study.

Bits of an octet are numbered 8 to 1 where bit 1 is the low order bit and is transmitted first. Octets of a packet are consecutively numbered starting from 1 and are transmitted in this order.

4.1.1 *General format identifier*

The general format identifier field is a four-bit binary coded field which is provided to indicate the general format of the rest of the header. The general format identifier field is located in bit positions 8, 7, 6 and 5 of octet 1 and 5 is the low order bit (see Table 11/X.75).

TABLE 11/X.75

General format identifier

General format identifier		Octet 1 Bits 8	7	6	5
Data packets	Sequencing numbers scheme modulo 8	X	X	0	1
	Sequencing numbering scheme modulo 128	X	X	1	0
Call set-up packets	Sequencing numbering scheme modulo 8	0	X	0	1
	Sequencing numbering scheme modulo 128	0	X	1	0
Clearing, flow control, interrupt, reset and restart packets	Sequencing numbering scheme modulo 8	0	0	0	1
	Sequencing numbering scheme modulo 128	0	0	1	0
General format identifier extension		U	U	1	1
Reserved format for other applications		U	U	0	0

Note – A bit which is indicated as X may be set to either 0 or 1 as specified in the text and in Figures 3/X.75, 4/X.75, 7/X.75 and 8/X.75. A bit which is indicated as U is unspecified.

Bit 8 of the general format identifier is used for the qualifier (Q) in *data* packets and is set to 0 in all other packet types.

Bit 7 of the general format identifier is used in *data* and in *call set-up* packets in conjunctions with the *delivery confirmation* (D) procedure, and is set to 0 in all other packet types.

Bits 5 and 6 are encoded for four possible indications. Two of the codes are used to distinguish packets using modulo 8 sequence numbering scheme from packets using modulo 128 sequence numbering scheme. The third code is used to indicate an extension to an extended family of general format identifier codes and extended formats which are a subject for further study. The fourth code is unassigned.

4.1.2 *Logical channel group number*

The logical channel group number appears in every packet except in *restart* packets (see § 4.5 below) in bit positions 4, 3, 2 and 1 of octet 1. This field is binary coded and bit 1 is the low order bit of the logical channel group number.

For each logical channel, this number has local significance at the X/Y interface.

4.1.3 *Logical channel number*

The logical channel number appears in every packet except in *restart* packets (see § 4.5 below) in all bit positions of octet 2. This field is binary coded and bit 1 is the low order bit of the logical channel number.

For each logical channel, this number has local significance at the X/Y interface.

4.1.4 *Packet type identifier*

Each packet shall be identifier in octet 3 of the packet according to Table 12/X.75.

TABLE 12/X.75

Packet type identifier

Packet Type	Octet 3 Bits							
	8	7	6	5	4	3	2	1
Call set-up clearing								
Call request	0	0	0	0	1	0	1	1
Call connected	0	0	0	0	1	1	1	1
Clear request	0	0	0	1	0	0	1	1
Clear confirmation	0	0	0	1	0	1	1	1
Data and interrupt								
Data	X	X	X	X	X	X	X	0
Interrupt	0	0	1	0	0	0	1	1
Interrupt confirmation	0	0	1	0	0	1	1	1
Flow control and reset								
Receive ready (modulo 128)	0	0	0	0	0	0	0	1
Receive ready (modulo 8)	X	X	X	0	0	0	0	1
Ready not ready (modulo 128)	0	0	0	0	0	1	0	1
Ready not ready (modulo 8)	X	X	X	0	0	1	0	1
Reset request	0	0	0	1	1	0	1	1
Reset confirmation	0	0	0	1	1	1	1	1
Restart								
Restart request	1	1	1	1	1	0	1	1
Restart confirmation	1	1	1	1	1	1	1	1

Note – A bit which is indicated as X may be set to either 0 to 1 as specified in the text and in Figures 5/X.75 to 20/X.75.

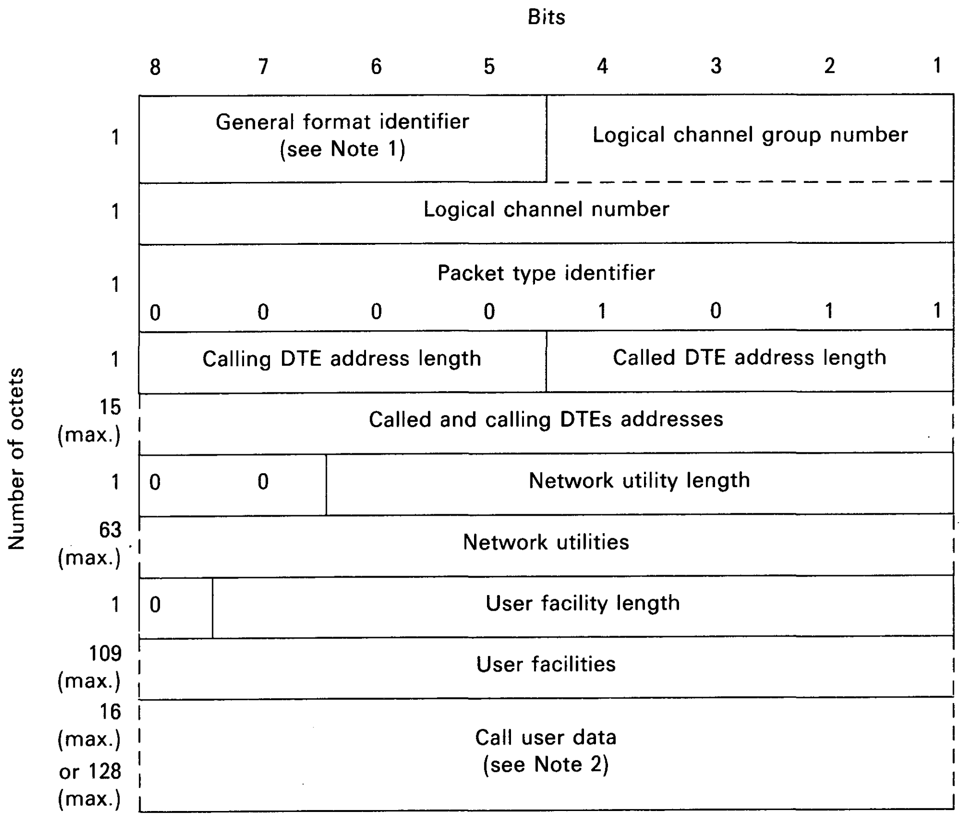
4.2 *Call set-up and clearing packets*

The following describes the nature of addresses present in the call set-up and clearing packets.

If the STE X/Y interface is between two PSPDNs, or between a PSPDN and an ISDN then the addresses will be in the international format given in Recommendation X.121, including escape digits where required. If the STE X/Y interface is between two ISDNs, then the addresses will be in the international format given in Recommendation E.164, including escape digits where required. Additional guidance is given in Recommendations X.31, X.122 and E.166.

4.2.1 *Call request packet*

Figure 5/X.75 illustrates the format of a *call request* packet. In this figure the user facility length field, user facilities field, and call user data field are as defined in Recommendation X.25.



Note 1 – Coded 0D01 (modulo 8) or 0D10 (modulo 128). D is the delivery confirmation bit.

Note 2 – More than 16 octets of call user data will only be present when the *fast select* optional user facility is requested.

FIGURE 5/X.75

Call request packet format

4.2.1.1 *General format identifier*

Bit 7 can be set to either 0 or 1.

4.2.1.2 *Address length field*

Octet 4 consists of field length indicators for the called and calling DTE address. Bits 4, 3, 2 and 1 indicate the length of the called DTE address in semi-octets. Bits 8, 7, 6 and 5 indicate the length of the calling DTE address in semi-octets. Each address length indicator is binary coded and bit 1 or 5 is the low order bit of the indicator.

4.2.1.3 *Address field*

Octet 5 and the following octets consist of the called DTE address followed by the calling DTE address as specified in § 4.2 above.

Each digit of an address is coded in a semi-octet in binary coded decimal with bit 5 or 1 being the low order bit of the digit.

Starting from the high order digit, the address is coded in octet 5 and consecutive octets with two digits per octet. In each octet, the higher order digit is coded in bits 8, 7, 6 and 5.

The address field shall be rounded up to an integral number of octets by inserting 0s in bits 4, 3, 2 and 1 of the last octet of the field when necessary.

4.2.1.4 *Network utility length field*

Bits 6 through 1 of the octet following the address field indicate the length of the network utility field in octets.

The network utility length field indicator is binary coded and bit 1 is the low order bit.

Bits 8 and 7 of this octet are unassigned and set to 0.

4.2.1.5 *Network utility field*

The network utility field contains an integral number of octets. The length of this field depends on the utilities present. The maximum length of this field is 63 octets.

The coding of the network utility field is defined in § 5 below.

4.2.1.6 *User facility length field*

Bits 7 through 1 of the octet following the network utility field indicate the length of the user facility field in octets. The user facility length indicator is binary coded and bit 1 is the low order bit.

Bit 8 of this octet is set to 0.

4.2.1.7 *User facility field*

The user facility field contains an integral number of octets. The length of this field depends on the facilities present. The maximum length of this field is 109 octets. The coding of the user facility field is dependent on the facilities being requested as defined in Recommendation X.25 (Table 29/X.25 and Annex G/X.25).

4.2.1.8 *Call user data field*

Following the user facility field, user data may be present. In the absence of the *fast select* optional user facility, the call user data field may contain any number of bits from 0 to 128 (16 octets). When the *fast select* optional facility is requested, the call user data may contain any number of bits from 0 to 1024 (128 octets). The contents of the field are passed unchanged.

Note – Some networks require the call user data field contains an integral number of octets (see § 3 Note).

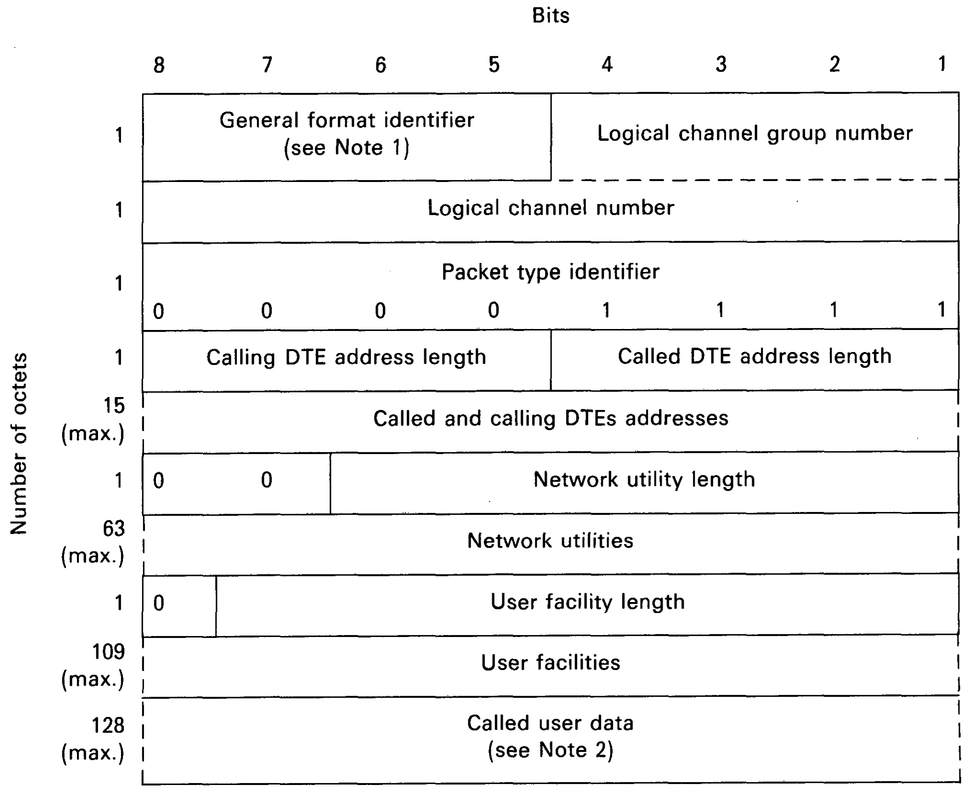
4.2.2 *Call connected packet*

Figure 6/X.75 illustrates the format of a *call connected* packet. Similarly to the *call request* packet, the *call connected* packet contains:

- an address length field,
- an address field,
- a network utility length field,
- a network utility field,
- a user facility length field,
- a user facility field, and
- a called user data field.

The coding of these fields is the same as that in *call request* packet (see § 4.2.1 above). Bit 7 of the general format identifier can be set to either 0 or 1. The address field may be empty. However, in the case of call redirection, the address field shall contain the address of the DTE to which the call was finally directed, and the utility field should contain the *called line address modified notification* utility (see § 5.3.10).

The called user data field may only be included for calls in which the *fast select* optional user facility has been requested with no restriction on response and may contain any number of bits from 0 up to 1024 (128 octets). The contents of the field are passed unchanged.



Note 1 – Coded 0D01 (modulo 8) or 0D10 (modulo 128). D is the delivery confirmation bit.

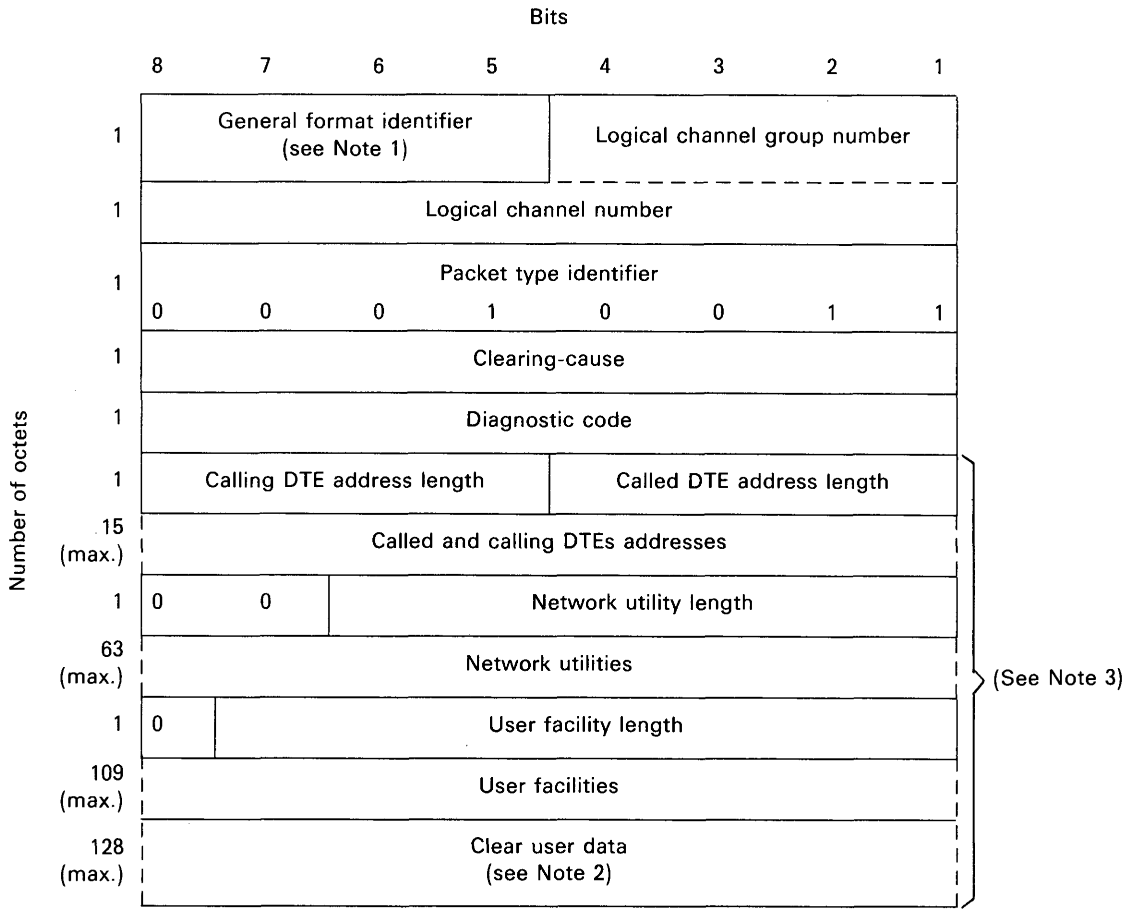
Note 2 – This field will only be included where the called user data is returned in response to a *call request* packet in which the *fast select* optional user facility has been requested with no restriction on response.

FIGURE 6/X.75

Call connected packet format

4.2.3 *Clear request packet*

Figure 7/X.75 illustrates the format of a *clear request* packet.



Note 1 – Coded 0001 (modulo 8) or 0010 (modulo 128).

Note 2 – This field will only be included where the clear user data is returned when *fast select* optional user facility has been requested.

Note 3 – Used only in the extended format (see § 4.2.3.3).

FIGURE 7/X.75

Clear request packet format

4.2.3.1 *Clearing cause field*

Octet 4 is the clearing cause field and contains the reason for the clearing of the call.

The coding of the clearing cause field in a *clear request* packet is given in Table 13/X.75.

An STE receiving a clearing cause other than that given in Table 13/X.75 will either pass this cause unchanged or change the cause to "Network congestion".

TABLE 13/X.75

Coding of clearing cause field in a clear request packet

Clearing cause	Octet 4							
	Bits							
	8	7	6	5	4	3	2	1
DTE originated	0	0	0	0	0	0	0	0
DTE originated (see Note 1)	1	X	X	X	X	X	X	X
Number busy	0	0	0	0	0	0	0	1
Out of order	0	0	0	0	1	0	0	1
Remote procedure error	0	0	0	1	0	0	0	1
Reverse charging acceptance not subscribed	0	0	0	1	1	0	0	1
Incompatible destination	0	0	1	0	0	0	0	1
Fast select acceptance not subscribed	0	0	1	0	1	0	0	1
Ship absent (see Note 2)	0	0	1	1	1	0	0	1
Invalid facility request	0	0	0	0	0	0	1	1
Access barred	0	0	0	0	1	0	1	1
Network congestion	0	0	0	0	0	1	0	1
Not obtainable	0	0	0	0	1	1	0	1
RPOA out of order (see Note 3)	0	0	0	1	0	1	0	1

Note 1 – When bit 8 is set to 1, the bits represented by Xs are those included by the remote DTE in the clearing or restarting clause field of the X.25 *clear* or *restart request* packet.

Note 2 – Used in conjunction with Mobile Maritime service.

Note 3 – May be received by the STE only if the optional *RPOA selection* utility was used by the STE.

4.2.3.2 *Diagnostic code field*

Octet 5 is the diagnostic code field and may contain additional information on the reason for the clearing of the call.

If the associated clearing cause field (octet 4) indicates any valid cause (see Table 13/X.75) except "Network congestion", the contents of this field will be passed unchanged. If the clearing cause field indicates "Network congestion" and the original clear or restart request was generated as the result of an event detected other than at the local STE-X/Y interface, then the value of the diagnostic code passed will be as shown in Table 14/X.75.

The diagnostic codes in *clear request* packets generated as the result of event detected at the local STE-X/Y interface are listed in Annex E.

TABLE 14/X.75

Diagnostic code mapping for clear request packet

Decimal value originally generated	Decimal value passed
0	same
1 to 111	114
112 to 127	same
128 to 255	113

4.2.3.3 *Extended format*

The following fields may follow the diagnostic code field in the extended format:

- an address length field,
- an address field,
- a network utility length field,
- a network utility field,
- a user facility length field,
- a user facility field, and
- a clear user data field.

4.2.3.3.1 *Address length field*

This single octet field consists of field length indicators for the called and calling DTE addresses. Bits 4, 3, 2 and 1 indicate the length of the called DTE address in semi-octets. Bits 8, 7, 6 and 5 indicate the length of the calling DTE address in semi-octets. Each address length indicator is binary coded and bit 1 or 5 is the low order bit of the indicator.

The address length field is always present when the network utility length field is present.

4.2.3.3.2 *Address field*

In the case that the clear request is issued, by a DTE to which a call has been redirected, as a direct response to the *call request* packet the address shall contain the address of the DTE to which the call was finally directed. Other use of this field is for further study.

Note – In the case of call redirection or call distribution within a hunt group, the utility field of *clear request* packet should include the *called line address modified notification* utility (see § 5.3.10).

4.2.3.3.3 *Network utility length field*

Bits 6 through 1 of the octet following the address field indicate the length of the network utility field in octets.

The network utility length field is binary coded and bit 1 is the low order bit.

Bits 8 and 7 of this octet is set to 0.

The network utility length field is always present when the user facility length is present.

4.2.3.3.4 *Network utility field*

The network utility field contains an integral number of octets. The length of this field depends on the utilities present. The maximum length of the field is 63 octets.

The coding of the network utility field is defined in § 5 below.

4.2.3.3.5 *User facility length field*

Bits 7 through 1 of the octet following the network utility field indicate the length of the user facility field in octets. The user facility length indicator is binary coded and bit 1 is the low order bit of the indicator.

Bit 8 of this octet is set to 0.

The user facility length field is always present when the user data field is present.

4.2.3.3.6 *User facility field*

The user facility field contains an integral number of octets. The length of this field depends on the facilities present. The maximum length of this field is 109 octets. The coding of the user facility field is dependent on the facilities being requested as defined in Recommendation X.25 (Table 29/X.25 and Annex G/X.25).

4.2.3.3.7 *Clear user data field*

For calls in which the *fast select* optional user facility has been requested, clear user data may be present, following the user facility field. The clear user data field may contain any number of bits from 0 up to 1024 (128 octets). The contents of the field are passed unchanged.

Note – Some networks require the clear user data field to contain an integral number of octets (see § 3 Note).

4.2.4 *Clear confirmation packet*

Figure 8/X.75 illustrates the format of the *clear confirmation* packet.

		Bits							
		8	7	6	5	4	3	2	1
	1	General format identifier (see Note)				Logical channel group number			
Octets	2	Logical channel number							
	3	Packet type identifier							
		0	0	0	1	0	1	1	1

Note – Coded 0001 (modulo 8) or 0010 (modulo 128).

FIGURE 8/X.75

Clear confirmation packet format

4.3 *Data and interrupt packets*

4.3.1 *Data packet*

Figures 9/X.75 and 10/X.75 illustrate the format of the *data* packets in the case of modulo 8 and modulo 128 respectively.

<table>
<tr><th></th><th colspan="8">Bits</th></tr>
<tr><th>Octets</th><th>8</th><th>7</th><th>6</th><th>5</th><th>4</th><th>3</th><th>2</th><th>1</th></tr>
<tr><td rowspan="2">1</td><td colspan="4">General format identifier</td><td colspan="4" rowspan="2">Logical channel group number</td></tr>
<tr><td>Q</td><td>D</td><td>0</td><td>1</td></tr>
<tr><td>2</td><td colspan="8">Logical channel number</td></tr>
<tr><td>3</td><td colspan="3">P(R)</td><td>M</td><td colspan="3">P(S)</td><td>0</td></tr>
<tr><td></td><td colspan="8">User data</td></tr>
</table>

D Delivery confirmation

M More data indication

Q Qualifier

FIGURE 9/X.75

Data packet format (modulo 8)

<table>
<tr><th></th><th colspan="8">Bits</th></tr>
<tr><th>Octets</th><th>8</th><th>7</th><th>6</th><th>5</th><th>4</th><th>3</th><th>2</th><th>1</th></tr>
<tr><td rowspan="2">1</td><td colspan="4">General format identifier</td><td colspan="4" rowspan="2">Logical channel group number</td></tr>
<tr><td>Q</td><td>D</td><td>1</td><td>0</td></tr>
<tr><td>2</td><td colspan="8">Logical channel number</td></tr>
<tr><td>3</td><td colspan="7">P(S)</td><td>0</td></tr>
<tr><td>4</td><td colspan="7">P(R)</td><td>M</td></tr>
<tr><td></td><td colspan="8">User data</td></tr>
</table>

D Delivery confirmation

M More data bit

Q Qualifier bit

FIGURE 10/X.75

Data packet format (modulo 128)

4.3.1.1 *Qualifier (Q) bit*

Bit 8 in octet 1 is used for the *qualifier* (Q) bit.

4.3.1.2 *Delivery confirmation (D) bit*

Bit 7 in octet 1 is the *delivery confirmation* (D) bit.

4.3.1.3 *Packet receive sequence number*

In Figure 9/X.75 bits 8, 7 and 6 of the octet 3 are used for indicating the packet receive sequence number P(R) is binary coded and bit 6 is the low order bit. In Figure 10/X.75, bits 2 through 8 of octet 4 are used for the packet send sequence number and bit 2 is the low order bit.

4.3.1.4 *More data bit*

In Figure 9/X.75, bit 5 in octet 3 is used for the *more data* mark (M bit). In Figure 10/X.75, bit 1 in octet 4 is used for the *more data* mark (M bit) (0 for no more data and 1 for more data).

4.3.1.5 *Packet send sequence number*

In Figure 9/X.75, bits 4, 3 and 2 of octet 3 are used for indicating the packet send sequence number P(S). P(S) is binary coded and bit 2 is the low order bit. In Figure 10/X.75, bits 2 through 8 of octet 3 are used for the packet send sequence number and bit 2 is the low order bit.

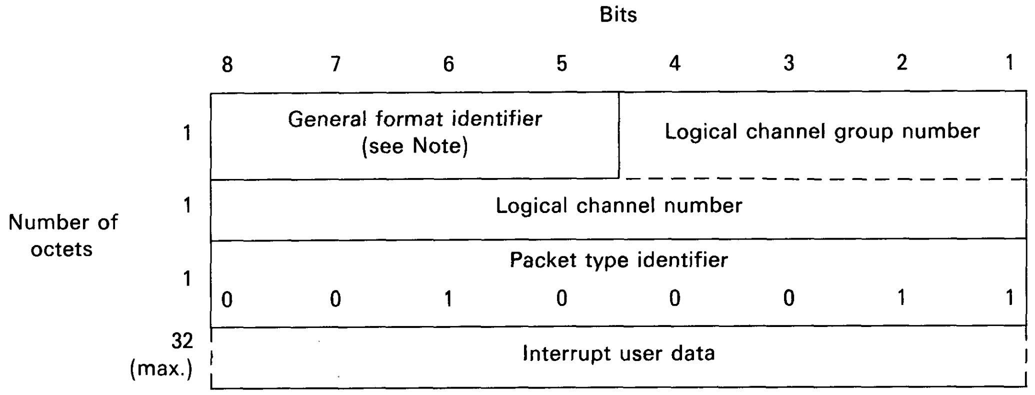
4.3.1.6 *User data field*

The bits following octet 3 (modulo 8) or octet 4 (modulo 128) contain user data.

Note – Some networks require the user data field to contain an integral number of octets (see § 3 Note).

4.3.2 *Interrupt packet*

Figure 11/X.75 illustrates the format of the *interrupt* packet.



Note – Coded 0001 (modulo 8) or 0010 (modulo 128).

FIGURE 11/X.75

Interrupt packet format

4.3.2.1 *Interrupt user data field*

Octet 4 and any following octets contain the interrupt user data. This field contains from 1 to 32 octets.

Note – Some networks require the interrupt user data field to contain an integral number of octets (see § 3 Note).

4.3.3 *Interrupt confirmation packet*

Figure 12/X.75 illustrates the format of the *interrupt confirmation* packet.

Bits

		8	7	6	5	4	3	2	1
Octets	1	General format identifier (see Note)				Logical channel group number			
	2	Logical channel number							
	3	Packet type identifier 0	0	1	0	0	1	1	1

Note – Coded 0001 (modulo 8) or 0010 (modulo 128).

FIGURE 12/X.75

Interrupt confirmation packet format

4.4 *Flow control and reset packets*

4.4.1 *Receive ready (RR) packet*

Figures 13/X.75 and 14/X.75 illustrate the format of *receive ready* packets in the case of modulo 8 and modulo 128 respectively.

Bits

		8	7	6	5	4	3	2	1
Octets	1	General format identifier 0	0	0	1	Logical channel group number			
	2	Logical channel number							
	3	P(R)			Packet type identifier 0	0	0	0	1

FIGURE 13/X.75

RR packet format (modulo 8)

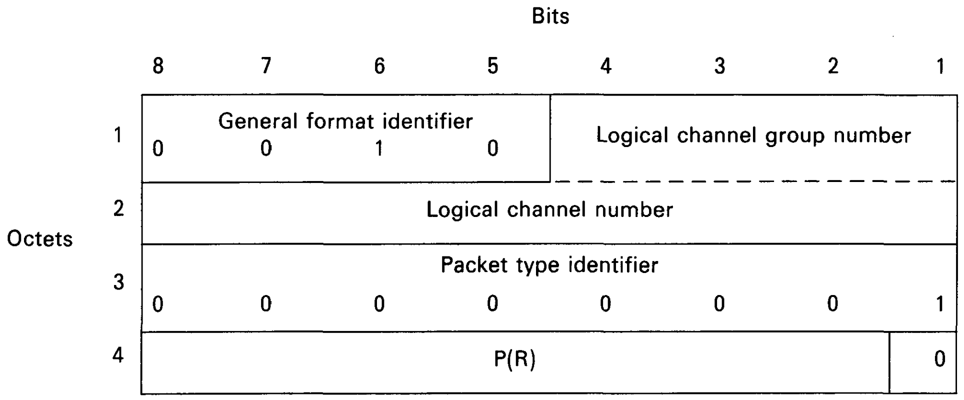


FIGURE 14/X.75

RNR packet format (modulo 128)

4.4.1.1 *Packet receive sequence number*

In Figure 13/X.75, bits 8, 7 and 6 of octet 3 are used for indicating the packet receive sequence number P(R). P(R) is binary coded and bit 6 is the low order bit. In Figure 14/X.75, bits 2 through 8 of octet 4 are used for the packet receive sequence number and bit 2 is the low order bit.

4.4.2 *Receive not ready (RNR) packet*

Figures 15/X.75 and 16/X.75 illustrate the format of *receive not ready* packets in the case of modulo 8 and modulo 128 respectively.

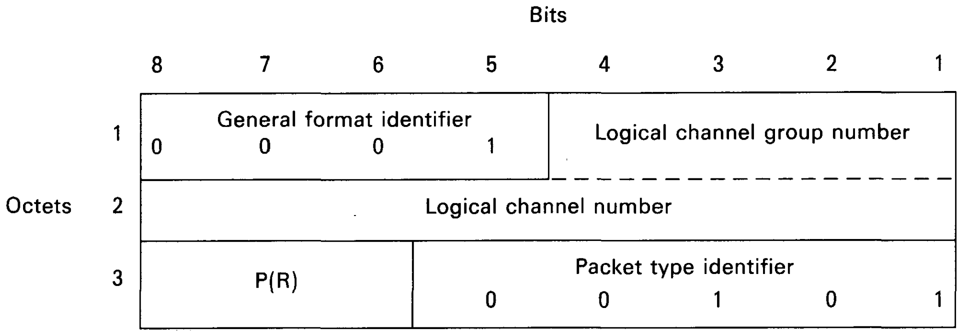


FIGURE 15/X.75

RNR packet format (modulo 8)

Bits

		8	7	6	5	4	3	2	1
	1	General format identifier				Logical channel group number			
		0	0	1	0				
Octets	2	Logical channel number							
	3	Packet type identifier							
		0	0	0	0	0	1	0	1
	4	P(R)							0

FIGURE 16/X.75

RNR packet format (modulo 128)

4.4.2.1 *Packet receive sequence number*

In Figure 15/X.75, bits 8, 7 and 6 of octet 3 are used for indicating the packet receive sequence number P(R). P(R) is binary coded and bit 6 is the low order bit. In Figure 16/X.75, bits 2 through 8 of octet 4 are used for the packet receive sequence number and bit 2 is the low order bit.

4.4.3 *Reset request packet*

Figure 17/X.75 illustrates the format of the *reset request* packet.

Bits

		8	7	6	5	4	3	2	1
	1	General format identifier (see Note)				Logical channel group number			
	2	Logical channel number							
Octets	3	Packet type identifier							
		0	0	0	1	1	0	1	1
	4	Resetting cause							
	5	Diagnostic code							

Note – Coded 0001 (modulo 8) or 0010 (modulo 128).

FIGURE 17/X.75

Reset request packet format

4.4.3.1 *Resetting cause field*

Octet 4 is the resetting cause field and contains the reason for the reset.

The coding of the resetting cause field in a *reset request* packet is given in Table 15/X.75.

An STE receiving a resetting cause other than that given in Table 15/X.75 will either pass this cause unchanged or change the cause to "Network congestion".

TABLE 15/X.75

Coding of resetting cause field in reset request packet

Resetting cause	Octet 4 Bits							
	8	7	6	5	4	3	2	1
DTE originated	0	0	0	0	0	0	0	0
DTE originated (see Note 1)	1	X	X	X	X	X	X	X
Out of order (see Note 2)	0	0	0	0	0	0	0	1
Remote procedure error	0	0	0	0	0	0	1	1
Network congestion	0	0	0	0	0	1	1	1
Remote DTE operational (see Note 2)	0	0	0	0	1	0	0	1
Network operational (see Note 3)	0	0	0	0	1	1	1	1
Incompatible destination	0	0	0	1	0	0	0	1
Network out of order (see Note 2)	0	0	0	1	1	1	0	1

Note 1 – When bit 8 is set to 1, the bits represented by Xs are those indicated by the remote DTE in the resetting cause field (virtual calls and permanent virtual circuits) or the restarting cause field (permanent virtual circuits) of the X.25 *reset* or *restart request* packets.

Note 2 – Applicable to permanent virtual circuits only.

Note 3 – If the STE receives a *reset request* packet with the cause "Network operational", it does not necessarily mean that the permanent virtual circuit is operational.

4.4.3.2 *Diagnostic code field*

Octet 5 is the diagnostic code field and may contain additional information on the reason for the reset.

If the associated resetting cause field (octet 4) indicates any valid cause (see Table 15/X.75) except "Network congestion", the contents of this field will be passed unchanged. If the resetting cause field indicates "Network congestion" and the original reset or restarting request was generates as the result of an event detected other than at the local STE-X/Y interface, then the value of the diagnostic code passed will be as shown in Table 16/X.75.

The diagnostic codes in *reset request* packets generated as the result of events detected at the local STE-X/Y interface are listed in Annex E.

4.4.4 *Reset confirmation packet*

Figure 18/X.75 illustrates the format of the *reset confirmation* packet.

4.5 *Restart packets*

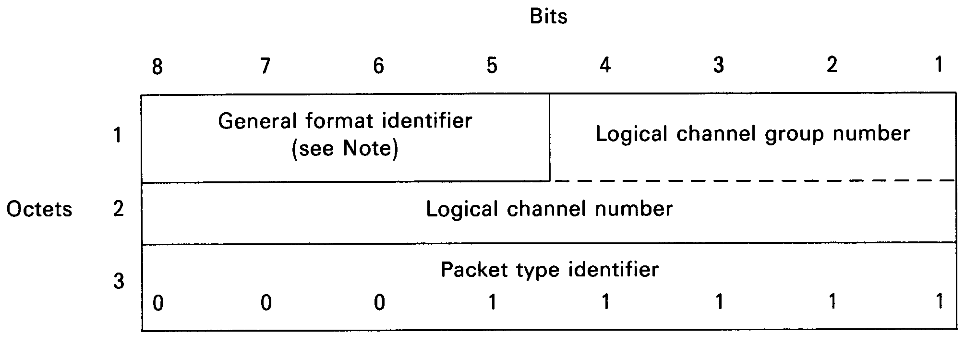
4.5.1 *Restart request packet*

Figure 19/X.75 illustrates the format of the *restart request* packet. Bits 4, 3, 2 and 1 of the first octet and all bits of the second octet are set to 0.

TABLE 16/X.75

Diagnostic codes mapping for reset request packet

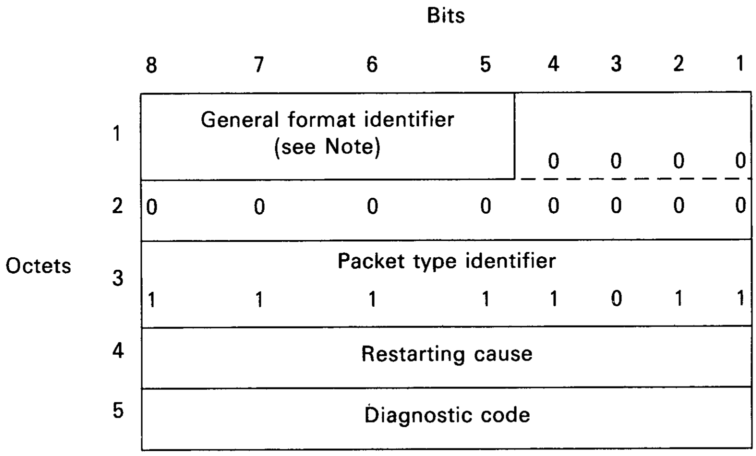
Decimal value originally generated	Decimal value passed
0	same
1 to 111	114
112 to 127	same
128 to 255	113



Note – Coded 0001 (modulo 8) or 0010 (modulo 128).

FIGURE 18/X.75

Reset confirmation packet format



Note – Coded 0001 (modulo 8) or 0010 (modulo 128).

FIGURE 19/X.75

Restart request packet format

4.5.1.1 *Restarting cause field*

Octet 4 is the restarting cause field and contains the reason for the restart.

The coding of the restarting cause field in the *restart request* packets is given in Table 17/X.75.

An STE receiving a restarting cause other than that given in Table 17/X.75 will either pass this cause unchanged or change the cause to "Network congestion".

TABLE 17/X.75

Coding of restarting cause field in restart request packet

Restarting cause	Octet 4 Bits 8	7	6	5	4	3	2	1
Network congestion	0	0	0	0	0	0	1	1
Network operational	0	0	0	0	0	1	1	1

4.5.1.2 *Diagnostic code field*

Octet 5 is the diagnostic code field and may contain additional information on the reason for the restart.

If the associated restarting cause field (octet 4) indicates any valid cause (see Table 17/X.75) except "Network congestion", the contents of this field will be passed unchanged in the resulting *clear* or *reset request* packet. If the restarting cause field indicates "Network congestion" then the value of the diagnostic code sent in the resulting *clear* or *reset request* packet will be as shown in Table 18/X.75.

TABLE 18/X.75

Diagnostic code mapping for restart request packet

Decimal value originally generated	Decimal value sent
0	same
1 to 111	114
112 to 127	same
128 to 255	113

The diagnostic codes in *restart request* packets generated as the result of events detected at the local STE-X/Y interface are listed in Annex E.

The bits of the diagnostic code field are all set to 0 when no specific reason for the restart is supplied.

4.5.2 *Restart confirmation packet*

Figure 20/X.75 illustrates the format of the *restart confirmation* packet. Bits 4, 3, 2 and 1 of the first octet and all bits of the second octet are set to 0.

Bits

		8	7	6	5	4	3	2	1
Octets	1	General format identifier (see Note)				0	0	0	0
	2	0	0	0	0	0	0	0	0
	3	Packet type identifier 1	1	1	1	1	1	1	1

Note – Coded 0001 (modulo 8) or 0010 (modulo 128).

FIGURE 20/X.75

Restart confirmation packet format

5 Procedures and formats for user facilities and network utilities

5.1 *Description of optional user facilities*

Signalling for CCITT specified DTE facilities and those user facilities (see Recommendation X.25) which do not require STE or transit network action is normally contained in the user facility field of X.75 packets. The contents of this field are conveyed transparently through an STE, which may examine and store them, but does not influence the progress of the call as a result.

Other user facilities which do require STE or transit network action are mapped into X.75 utilities and thus are not present in the X.75 facility field.

5.2 *Formats for optional user facilities*

The formats for optional user facilities are described in Recommendation X.25.

5.3 *Procedures for network utilities*

The network utility field is a network administrative signalling mechanism in the *call request, call connected* and *clear request* packets. The network utility field complements the user facility field and serves to separate user service signalling from network administrative signalling. The request for a service through an optional user facility may, in certain instances, require the use of a network utility.

There are three categories of network utilities:

- *International Mandatory network utilities:* These are the network utilities that must be supported by all international X.75 interworkings. International Mandatory means that every international STE must be capable of actioning the procedures for each network utility so classified. For some international mandatory utilities, not all calls need to signal the utility in the packet. International Mandatory utilities may also be used for national interworkings subject to bilateral agreements.

- *International Optional network utilities:* These are the network utilities that may be supported by international X.75 interworkings, subject to bilateral agreements. When an international optional utility has been bilaterally agreed for use, the procedures herein described in this utility are used. International optional utilities may also be used for national interworkings subject to bilateral agreements.
- *National network utilities:* These are the network utilities that may only be supported on links between networks in the same country, and are always subject to bilateral agreements.

The categorization of network utilities is given in Table 19/X.75. Utilities not listed in Table 19/X.75 are for further study and, therefore, no categorization is indicated.

TABLE 19/X.75

Categorization of network utilities

International mandatory network utilities	*Section*
Transit network identification	5.3.1
Call identifier	5.3.2
Throughput class indication	5.3.3
Window size indication	5.3.4
Packet size indication	5.3.5
Fast select indication	5.3.6
Closed user group indication	5.3.7
Closed user group with outgoing access indication	5.3.8
Called line address modified notification	5.3.10
Transit delay indication	5.3.13
International optional network utilities	
Reverse charging indication	5.3.9
Clearing network identification code	5.3.11
Transit delay selection	5.3.14
Tariffs	5.3.15
Network user identification	5.3.16
Utility marker	5.3.18
National network utilities	
RPOA selection	5.3.17

Several network utilities include the identification of a given network, If the given network is a public data network, it is identified by the first four digits (DNIC) of the international data number. However, if the given network is ISDN, it is identified by a four-digit field, the ISDN Network Identification Code (INIC), composed of:

0 + E.164 Country Code + National Network Digit(s)

where the number of National Network Digits depends on the size of E.164 Country Code, for a given country. National Network Digit(s) may be any value(s) agreed by the Administration within the given country. In order to identify additional ISDNs, some countries may also use the four-digit format composed of:

9 + E.164 Country Code + National Network Digit(s)

For national utilities, inclusion of E.164 country code is optional.

Alternate ways of ISDN network identification are for further study.

5.3.1 *Transit network identification (International Mandatory)*

The *transit network identification* is a network utility used to name a transit network controlling a portion of the (perhaps partially established) virtual circuit. A transit network is identified by its DNIC or INIC as specified in § 5.3 above.

A *transit network identification* is always present in the *call request* packet for each transit network controlling the virtual circuit up to this point of call set-up. When more than one transit network is identified, the order of identification in the network utility field identical to the order of traversal of transit networks following the path being established from the calling DTE to the destination network.

A *transit network identification* is always present for each transit network in the *call connected* packet, or the *clear request* packet issued as a direct response to the *call request* packet. The *transit network identification* utility is not present in a *clear request* packet issued either after receipt of the corresponding *call connected* packet or after transmission of the corresponding *call request* or *call connected* packets. When there is more than one transit network, the identification order in the network utility field is identical to the order of transversal of transit networks following the path established from the calling to called DTE.

5.3.2 *Call identifier (International Mandatory)*

The *call identifier* is a network utility which is always present in the *call request* packet. The *call identifier* parameter is established by the originating network and is an identifying name for each virtual circuit established. The *call identifier* when used in conjunction with the calling DTE address, uniquely identifies the virtual call. The uniqueness is only guaranteed over a period of time. The duration of this time is for further study.

The use of the *call identifier* in the *call connected* packet is for further study. The *call identifier* is not present in the *clear request* packet.

Note – The definition of the content of the *call identifier*, and further specification of the associated signalling mechanisms, require further study. Pending such further study, the content of a *call identifier* may or may not be significant for a given call, this is under the responsibility of the originating network. However, it is for further study whether a transit network can create a significant *call identifier*, in the case it would receive a *call identifier* which is not significant. When the *call identifier* is not significant, it would be coded as zero by the originating network.

5.3.3 *Throughput class indication (International Mandatory)*

The *throughput class indication* is a network utility that can be used by any STE for specifying the throughput classes applying to that call.

The STE associated with the virtual call originating network may request in the *throughput class indication* utility of the *call request* packet the throughput class values selected at the calling DTE/DCE interface. Any transit STE may also request throughout class values in the *throughput class indication* utility of the *call request* packet. If particular throughput classes are not explicitly requested, the STE is assumed to request the default throughput class values agreed between both Administrations.

Any STE, including the STEs associated with the virtual call originating and destination network, may reduce but must not raise the throughput class values requested for the call. In reducing the throughput class values, different criteria can be envisaged by the STE. The STE should consider the packet sizes, the window sizes and the throughput classes that it can support at a given time. The STE may also consider the STE resources available and the throughput classes requested for that call. The STEs associated with the virtual call originating and destination networks may also consider the flow control parameters used at the DTE/DCE interface.

Taking the above considerations into account, the throughput class any STE reduces down to may vary per individual call and may be higher or lower than or equal to the default throughput class values agreed between both Administrations.

When the called DTE has accepted the call, the STE associated with the virtual call destination network may confirm in the *throughput class indication* utility of the *call connected* packet the throughput class values that finally apply to the virtual call following the negotiation with the called DTE. Any transit STE may also confirm throughput class values in the *throughput class indication* utility of the *call connected* packet. The STE should not alter the throughput class values received in a *call connected* packet.

If particular throughput classes are not explicitly confirmed, STE-Y is assumed to confirm the lesser of the default throughput class values agreed between both Administrations and the throughput class value requested originally. If an STE detects that an explicitly confirmed throughput class value finally applying to the call is higher than the one requested, it should clear the call with an indication of "Network congestion".

The *throughput class indication* utility should not be present in the *clear request* packet. No indication of *throughput classes* should be present in the user facility field of the *call request, call connected* and *clear request* packets.

5.3.4 *Window size indication (International Mandatory)*

The *window size indication* is a network utility that can be used by any STE for negotiating the window sizes on a specified logical channel at the STE X/Y interface for each direction of transmission.

When using the *window size indication* utility in the *call request* packet, STE-X requests particular window sizes to be used at the STE X/Y interface for that call.

If particular window sizes are not explicitly requested, STE-X is assumed to request the default values for that call, that is either the standard value of 2 or other values agreed between both Administrations.

When using the *window size indication* utility in the *call connected* packet, STE-Y confirms the window sizes finally applying at the STE X/Y interface to that call.

If particular window sizes are not explicitly confirmed, STE-Y is assumed to confirm the default values as finally applying to that call.

Each finally applying value should be in the range from the value requested by STE-X or assumed as a default value to the standard value of 2 (both inclusive). If an STE detects that a value finally applying to that call is out of this range, it should clear the call with an indication of "Network congestion".

In altering the window size values, different criteria can be envisaged by the STE. The STE should consider the packet sizes, window sizes and the throughput classes that it can support at a given time. The STE may also consider the STE resources available and the throughput classes requested for that call. The STEs associated with the virtual call originating and destination networks may also consider the flow control parameters used at the DTE/DCE interface.

The *window size indication* utility should not be present in the *clear request* packet.

No indication of *window sizes* should be present in the user facility field of the *call request, call connected* and *clear request* packets.

5.3.5 *Packet size indication (International Mandatory)*

The *packet size indication* is a network utility that can be used by any STE for negotiating the maximum data field length of *data* packets on a specified logical channel at the STE X/Y interface for each direction of data transmission.

When using the *packet size indication* utility in the *call request* packet, STE-X requests the maximum data field lengths to be used at the STE X/Y interface for that call.

If particular data field lengths are not explicitly requested, STE-X is assumed to request default values for that call, that is either the standard value of 128 octets or other values agreed between both Administrations.

When using the *packet size indication* utility in the *call connected* packet, STE-Y confirms the data field lengths finally applying at the STE X/Y interface for that call.

If particular data field lengths are not explicitly confirmed, STE-Y is assumed to confirm the default values as finally applying to that call.

Each finally applying value should be in the range from the value requested by STE-X or assumed as a default value to the standard value of 128 octets (both inclusive). If an STE detects that a value finally applying to that call is out of this range, it should clear the call with an indication of "Network congestion".

In altering the data field length values, different criteria can be envisaged by the STE. The STE should consider the packet sizes, the window sizes and the throughput classes that it can support at a given time. The STE may also consider the STE resources available and the throughput classes requested for that call. The STEs associated with the virtual call originating and destination networks may also consider the flow control parameters used at the DTE/DCE interface.

The *packet size indication* utility should not be present in the *clear request* packet.

No indication of packet sizes should be present in the user facility field of the *call request*, *call connected* and *clear request* packets.

5.3.6 *Fast select indication (International Mandatory)*

The *fast select indication* is a network utility used for indicating that the *fast select* user facility applies to that call.

When using the *fast select indication* utility in the *call request* packet, the STE indicates that the *fast select* facility applies to that call, with the corresponding packet formats as described in § 4.

When restriction on response is indicated in such a *call request* packet, the corresponding STE is permitted to issue as a direct response to this packet a *clear request* packet with a clear user data field of up to 128 octets, and is not authorized to send a *call connected* packet.

When no restriction on response is indicated in such a *call request* packet, the corresponding STE is permitted to issue as a direct response to this packet a *call connected* packet with a called user data field of up to 128 octets or at any time a *clear request* packet with a clear user data field of up to 128 octets. If the call is connected, the originating STE is authorized to transmit a *clear request* packet with a clear user data field of up to 128 octets.

No indication of *fast select* should be present in the user facility field of the *call request*, *call connected* and *clear request* packets.

The *fast select indication* utility should not be present in the *call connected* and *clear request* packets.

All other procedures of a call in which the *fast select* facility has been indicated are the same as those of a virtual call.

5.3.7 *Closed user group indication (International Mandatory)*

The *closed user group indication* is a network utility used for enabling the establishment of virtual calls by DTEs which are members of international closed user groups.

When using the *closed user group indication* utility in the *call request* packet, the STE indicates that the international virtual call is requested on the basis of valid international closed user group membership. The network of the calling DTE supplies the relevant international interlock code.

The STE should not alter the *closed user group indication* received in a *call request* packet.

Only one of the *closed user group indication* and the *closed user group with outgoing access indication* utilities may be present in a *call request* packet.

No indication of *closed user group* should be present in the user facility field of the *call request, call connected* and *clear request* packets.

The *closed user group indication* utility should not be present in the *call connected* and *clear request* packets.

5.3.8 *Closed user group with outgoing access indication (International Mandatory)*

The *closed user group with outgoing access indication* is a network utility used for enabling the establishment of virtual calls by DTEs which are members of international closed user groups.

When using the *closed user group with outgoing access indication* utility in the *call request* packet, the STE indicates that the international virtual call is requested on the basis of valid international closed user group membership. In addition the STE signals an associated outgoing access capability. The network of the calling DTE supplies the relevant international interlock code.

The STE should not alter the *closed user group with outgoing access indication* received in a *call request* packet.

Only one of the *closed user group indication* and the *closed user group with outgoing access indication* utilities may be present in a *call request* packet.

No indication of *closed user group with outgoing access* should be present in the user facility field of the call *request call* connected and *clear request* packets.

The *closed user group with outgoing access* utility should not be present in the *call connected* and *clear request* packets.

5.3.9 *Reverse charging indication (International Optional)*

The *reverse charging indication* is a network utility used for enabling virtual calls to be established internationally, when the *reverse charging* facility applies.

When using the *reverse charging indication* utility in the *call request* packet, STE-X indicates a request for reverse charging to apply to the call.

In the absence of the *reverse charging indication* utility, STE-X is assumed not to request reverse charging for that call.

The *reverse charging indication* utility should not be present in the *call connected* and the *clear request* packets.

No indication of *reverse charging* should be present in the user facility field of the *call request, call connected* and *clear request* packets.

5.3.10 *Called line address modified notification (International Mandatory)*

The *called line address modified notification* is a network utility used for indicating the reasons for the called address in the packet being different from that specified in the *call request* packet.

The following reasons can be indicated with the use of the *called line address modified notification* utility:

i) call distribution within a hunt group;

ii) call redirection due to originally called DTE out of order;

iii) call redirection due to originally called DTE busy;

iv) call redirection due to prior request from the originally called DTE for systematic call redirection;

v) called DTE originated;

vi) call deflection by the originally called DTE.

Both the call distribution within a hunt group and the call redirection are limited to the network of the DTE originally called.

The *called line address modified notification* utility will be present in *call connected* packets where the called DTE address is different from that specified in the *call request* packets. It will also be present in the *clear request* packet where the call is cleared by a different DTE from the one orignally called as a direct response to *call request* packet.

The *called line address modified notification* utility should not be present in the *call request* packet as well as the *clear request* packet sent after the call is connected.

No indication of *called line address modified notification* should be present in the user facility field of the *call request, call connected* and *clear request* packets.

5.3.11 *Clearing network identification code (International Optional)*

The *clearing network identification code* is a network utility providing additional information on the origin of the *clear request* packet and is present only in the *clear request* packet issued after the call is connected.

The network originating the *clear request* is identified by the DNIC or INIC of that network as specified in § 5.3 above.

An STE receiving a *clearing network identification code* will pass this code unchanged whenever applicable.

5.3.12 *Traffic class indication (for further study)*

The *traffic class* utility indicates a service category for the virtual circuit being established. The *traffic class* signals service information (e.g., terminal, facsimile, maintenance) necessary for administering the call. Though their use is beyond the scope of this Recommendation, *traffic class* may have routing, tariff and other implications. The need for and definition of traffic classes are for further study.

5.3.13 *Transit delay indication (International Mandatory)*

The *transit delay indication* is a network utility that signals the accumulated expected nominal transit delay of a virtual circuit. It is included in the *call request* packet and *call connected* packet when a calling DTE has requested a transit delay in the *transit delay selection and indication* facility. The STE in the originating network will signal a value dependent on the characteristics of the originating network and on the characteristics of the outgoing link (e.g., link speed, satellite or cable).

Any outgoing STE in a transit network will add to the value received in the *transit delay indication* utility a value that depends on the characteristics of the network and the outgoing link.

The transit delay is defined as t_{3c} in Recommendation X.135, and is expressed in terms of a mean value. However, the detailed determination of the value is considered as a national matter. If the resulting value of the transit delay exceeds the maximum value that can be signalled in the utility parameter field, all bits of the utility parameter field will be set to "1".

The STE will signal the final value of the accumulated expected nominal transit delay transparently in the *call connected* packet.

For an interim period, when not all networks have yet implemented the transit delay signalling, an STE will not send the *transit delay indication* utility to a network that does not support it. This STE will signal, towards its own network, all 1's in the *transit delay indication* utility parameter field of the *call connected* packet.

No indication of *transit delay selection and indication* should be present in the user facility field of the *call request, call connected* and *clear request packets.*

5.3.14 *Transit delay selection (International Optional)*

The *transit delay selection* utility is a network utility that signals the transit delay requested by the calling DTE in the *transit delay selection and indication* facility. This utility will be signalled transparently from the originating network to the destination network in the call request packet. This utility may be used in conjunction with the *transit delay indication* utility for routing purposes.

The *transit delay selection* utility should not be present in *call connected* or *clear request* packets.

No indication of *transit delay selection and indication* should be present in the user facility field of the *call request*, *call connected* and *clear request* packets.

5.3.15 *Tariffs (International Optional)*

The *tariffs* utility is a network utility that is used to pass information from one network to one or more other networks participating in the call for the purpose of implementing billing, accounting, or tariff arrangements that may exist among the respective Administrations.

The *tariffs* utility may appear in the *call request*, *call connected*, and *clear request* packets. If this utility appears in the *call request* packet, the information it contains relates to the originating interface or network. If this utility appears in the *call connected* or *clear request* packet, the information it contains relates to the ultimate destination interface or network. The utility may appear in a *clear request* packet only if that packet is initiated by the destination DTE or DCE, in direct response to the call request.

The content of this utility is determined by the originating or destination network and does not depend on information passed to the network by a DTE.

Even if this utility is supported on the STE X/Y interface, it may not be present in a packet for a given virtual call if there is no need to exchange tariff-related information with that packet.

No more than one instance of this utility may appear in a packet.

5.3.16 *Network user identification (NUI) (International Optional)*

The *network user identification* utility is a network utility used to provide supplementary network user identification for billing, security or network management purposes.

The utility may be present in the *call request* packet. No indication of *network user identification* should be present in the user facility field of any packet.

Note – Whether the utility may be present in the *call connected* packet is for further study.

This utility provides a mechanism for distinguishing a standardized CCITT default format from a format not constrained by this Recommendation.

A network may support some or all format options of this utility.

A network receiving this utility determines whether it is the network responsible for verifying the value. If it is not the network responsible for verifying the value, the network forwards the utility to the next network. It is for further study whether a network may forward this utility to the next network if the NUI value has been verified.

The originating network (STE), in formulating the value/content of this utility, may make use of DTE/DCE interface subscription options, network default assumptions, and/or values passed by the DTE on a per-call basis.

5.3.17 *RPOA selection (National)*

RPOA selection is a network utility that may be used to name a RPOA transit network within the originating country through which a call is to be routed. In the case of international calls, this utility may indicate an international RPOA in the originating country.

This utility can be used to carry a RPOA transit network DNIC or INIC (see § 5.3 above) specified by the calling DTE. When more than one transit network is specified by the calling DTE, a sequence of *RPOA selection* utilities may be present in the *call request* packet. In this case, the order of identification of transit networks by the RPOA *selection* utilities is identical to the order specified by the calling DTE.

A network receiving a *call request* packet containing one or more *RPOA selection* utilities will route to the next requested network, removing the *RPOA selection* utility that names the next requested network. If it is not possible to route to the next requested network, the receiving network will clear the call.

The *RPOA selection* utility should not be present in the *call connected* and *clear request* packets. No indication of the *RPOA selection* should be present in the user facility field of the *call request* packet.

5.3.18 *Utility marker (International Optional)*

The *utility marker* is used to separate international and national X.75 utilities, as defined under § 5.3 from non-X.75 utilities that may be agreed bilaterally by the Administrations.

5.4 *Formats for network utilities*

5.4.1 *General*

The network utility field is present in all *call request* and *call connected* packets, and may be present in *clear request* packets, exchanged between STEs.

The utility field contains a number of utility elements. Each utility element consists of a utility code followed by a utility parameter.

If multiple instances of a utility parameter are required in the utility field, such as the RPOA selection or *transit network identification*, this information will be presented in multiple utility elements with an identical utility code.

The utility codes are divided into four classes, by the use of bits 7 and 8, in order to specify utility parameters consisting of 1, 2, 3 or a variable number of octets. The general class coding is shown in Table 20/X.75.

TABLE 20/X.75

General class coding for network utility field

	Utility code field Bits								
	8	7	6	5	4	3	2	1	
Class A	0	0	X	X	X	X	X	X	for single octet parameter field
Class B	0	1	X	X	X	X	X	X	for double octet parameter field
Class C	1	0	X	X	X	X	X	X	for triple octet parameter field
Class D	1	1	X	X	X	X	X	X	for variable length parameter field

Note – A bit which is indicated as X may be set to either 0 or 1 as discussed in the text.

For class D, the octet following the utility code indicates the length, in octets, of the utility parameter. The utility parameter length is binary encoded and bit 1 is the low order bit. The maximum length of utility parameter field for class D cannot exceed 61 octets due to the maximum length of the network utility field.

The utility code field is binary coded and, without extension, provides for a maximum of 64 utility codes for classes A, B and C and 63 utility codes for class D giving a total of 255 utility codes (see Figure 21/X.75).

Utility code 11111111 is reserved for extension of the utility code. The octet following this octet indicates an extended utility code having the format A, B, C or D as defined in Figure 21/X.75. Repetition of utility code 11111111 is permitted and thus additional extensions result.

The specific coding of the utility parameter field is dependent on the utility being requested.

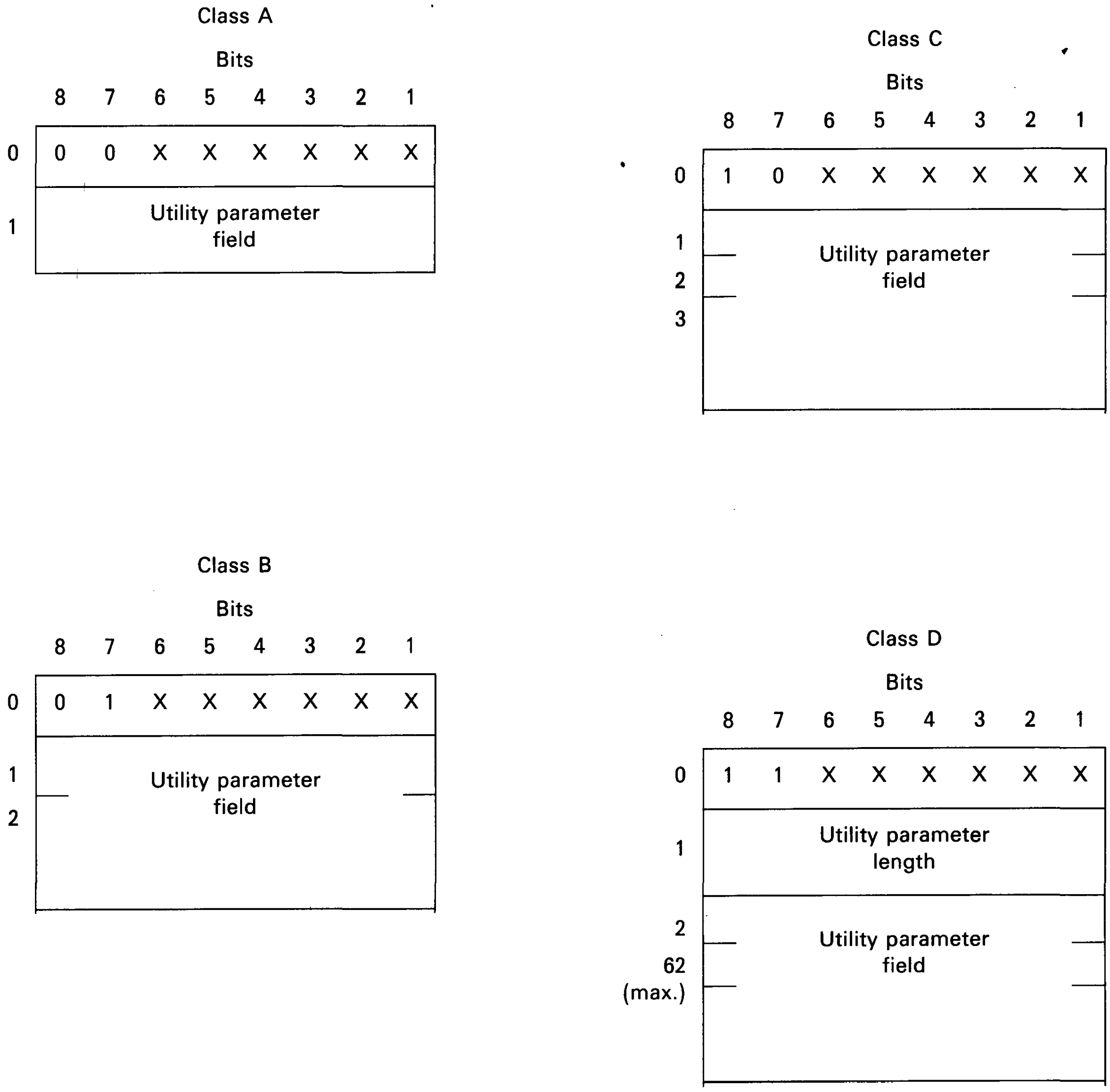


FIGURE 21/X.75

Utility code general formats

5.4.2 *Coding of utility code field*

The coding of the utility code field is given in Table 21/X.75.

Utility codings are the same for *call request*, *call connected* and *clear request* packets.

TABLE 21/X.75

Coding of the utility code field

Utility	Packet types in which it may be used			Utility code Bits							
	Call request	Call connected	Clear request	8	7	6	5	4	3	2	1
Transit network identification	X	X	X (See Note 1)	0	1	0	0	0	0	0	1
Call identifier	X	(See Note 2)		1	0	0	0	0	0	0	1
Throughput class indication	X	X		0	0	0	0	0	0	1	0
Window size indication	X	X		0	1	0	0	0	0	1	1
Packet size indication	X	X		0	1	0	0	0	0	1	0
Fast select and/or reverse charging indication	X			0	0	0	0	0	0	0	1
Closed user group indication	X			1	1	0	0	0	0	1	1
Closed user group with outgoing access indication	X			1	1	0	0	0	1	1	1
Called line address modified notification		X	X (See Note 1)	0	0	0	0	1	0	0	0
Clearing network identification code			X (See Note 3)	0	1	0	0	1	0	1	0
Traffic class indication	(See Note 4)			0	0	0	0	0	0	1	1
Transit delay indication	X	X		0	1	0	0	1	0	0	1
Transit delay selection	X			0	1	0	0	1	0	1	1
Tariffs	X	X	X (See Note 1)	0	0	0	0	0	1	1	1
NUI	X	(See Note 2)		1	1	0	0	0	1	1	0
RPOA selection	X			0	1	0	0	0	1	0	0
Utility marker	X	X	X	0	0	0	0	0	0	0	0

Note 1 – It is present in the *clear request* packet issued as a direct response to the *call request* packet.

Note 2 – The use of the *utility* in the *call connected* packet is for further study.

Note 3 – It is present only in the *clear request* packet issued after the call is connected.

Note 4 – The procedure is for further study.

5.4.3 *Coding of utility parameter field*

5.4.3.1 *Coding of transit network identification utility parameter*

Each of the four digits is coded in a semi-octet in binary coded decimal with bit 5 or 1 being the low order bit of the digit. The high order digit is coded into bits 8 to 5 of the first octet of the parameter.

5.4.3.2 *Coding of the call identifier utility parameter*

The call identifier consists of 24 bits of binary data.

5.4.3.3 *Coding of throughput class indication utility parameter*

The throughput class for transmission from the calling STE is indicated in bits 4, 3, 2 and 1. The throughput class for transmission from the called STE is indicated in bits 8, 7, 6 and 5.

The four bits indicating each throughput class are binary coded and correspond to throughput classes as indicated in Table 22/X.75.

TABLE 22/X.75

Coding of throughput classes

Bit: or Bit:	4 8	3 7	2 6	1 5	Throughput class (bit/s)
	0	0	0	0	Reserved
	0	0	0	1	Reserved
	0	0	1	0	Reserved
	0	0	1	1	75
	0	1	0	0	150
	0	1	0	1	300
	0	1	1	0	600
	0	1	1	1	1 200
	1	0	0	0	2 400
	1	0	0	1	4 800
	1	0	1	0	9 600
	1	0	1	1	19 200
	1	1	0	0	48 000
	1	1	0	1	64 000
	1	1	1	0	Reserved
	1	1	1	1	Reserved

5.4.3.4 *Coding of window size indication utility parameter*

The window size for the direction of transmission from the called STE is indicated in bits 7 to 1 of the first octet. The window size for the direction of transmission from the calling STE is indicated in bits 7 to 1 of the second octet. Bit 1 is the least significant bit. Bit 8 of each octet is unassigned and set to 0. Each window size value is binary encoded.

The range of window size values allowed at the STE X/Y interface is subject to a bilateral agreement between Administrations. Window sizes of 8 to 127 are only valid for calls which employ extended numbering.

5.4.3.5 *Coding of packet size indication utility parameter*

The maximum user data field length for the direction of transmission from the called STE is indicated in bits 4 to 1 of the first octet. The maximum user data field length for the direction of transmission from the calling STE is indicated in bits 4 to 1 of the second octet. Bits 8 to 5 of both octets are unassigned and set to 0.

The four bits indicating each maximum user data field length are binary encoded and express the logarithm to base 2 of the maximum number of octets of the data field of *data* packets. Bit 1 is the least significant bit.

The maximum user data field length values allowed at the STE X/Y interface are subject to a bilateral agreement between Administrations; however all Administrations will allow 128 octets.

5.4.3.6 *Coding of fast select and/or reverse charging indication utility parameter*

Bit: 8 7 6 5 4 3 2 1

Code: X Y U U U U U Z

U = Unassigned and set to 0,

X = 0 and Y = 0 or 1 for *fast select* not requested,

X = 1 and Y = 0 for *fast select* requested with no restriction on response,

X = 1 and Y = 1 for *fast select* requested with restriction on response,

Z = 0 for *reverse charging* not requested, and

Z = 1 for *reverse charging* requested.

5.4.3.7 *Coding of closed user group code and closed user group code with outgoing access*

5.4.3.7.1 *Utility parameter length*

Bit: 8 7 6 5 4 3 2 1

Code: 0 0 0 0 0 1 0 0

5.4.3.7.2 *Utility parameter*

The international interlock code is contained in the utility parameter field and consists of four octets.

The first two octets of the international interlock code consist of the four digits of DNIC or INIC as specified in § 5.3 above. Each digit is coded in a semi-octet in binary coded decimal with bit 5 or 1 being the low order bit of the digit. The high order digit is coded into bits 8 to 5 of the first octet of the parameter.

The remaining two octets contain the remaining 16 bits of the international interlock code, encoded with bit 8 of the third parameter octet as the high order bit.

5.4.3.8 *Coding of called line address modified notification utility parameter*

Bits: 8 7 6 5 4 3 2 1	
0 0 0 0 0 1 1 1	Call distribution within a hunt group
0 0 0 0 0 0 0 1	Call redirection due to originally called DTE, busy
0 0 0 0 1 0 0 1	Call redirection due to originally called DTE out of order
0 0 0 0 1 1 1 1	Call redirection due to prior request from originally called DTE for systematic call redirection
1 0 X X X X X X	Called DTE originated (see Note 1)
1 1 X X X X X X	Called deflection by the originally called DTE (see Note 2)

Note 1 – Each X may be independently set to 0 or 1 by the called DTE and is passed transparently.

Note 2 – The Xs are those set by the originally called DTE in the call forwarding selection facility.

5.4.3.9 *Coding of clearing network identification code parameter*

Each of the four digits of the DNIC or INIC of the clearing network are contained in the utility parameter field which consists of two octets. Each digit is coded in a semi-octet in binary coded decimal with bit 5 or 1 being the low order bit of the digit. The high order digit is coded into bits 8 to 5 of the first octet of the parameter.

5.4.3.10 *Coding of traffic class indication utility parameter*

The coding of the traffic class parameter is for further study.

5.4.3.11 *Coding of transit delay indication utility parameter*

This parameter is two octets. Transit delay is expressed provisionally in milliseconds, binary coded, with bit 8 of octet 1 being the high order bit and bit 1 of octet 2 being the low order bit.

5.4.3.12 *Coding of the transit delay selection utility parameter*

This parameter is two octets. Transit delay is expressed provisionally in milliseconds, binary coded, with bit 8 of octet 1 being the high order bit and bit 1 of octet 2 being the low order bit.

5.4.3.13 *Coding of the tariffs utility parameter*

The one octet parameter field consists of two subfields of 5 bits and 3 bits respectively:

Bit:	8	7	6	5	4	3	2	1
Code:	P	P	P	P	P	U	U	U

The interpretation of the first subfield which is called Primary tariff subfield is specified by Tables 23/X.75 and 24/X.75:

TABLE 23/X.75

Coding of primary tariff subfield

PPPPP 87654	Primary tariff subfield
00000	Subclass code 0
00001	Subclass code 1
.	.
.	.
.	.
11110	Subclass code 30
11111	Subclass code 31

TABLE 24/X.75

Interpretation of primary subclass codes

Primary subclass code(s)	Interface
0	X.25
1	Switched access X.28
2	Dedicated access X.28
3	X.32
4	X.75
5-15	[Reserved] (Note)
16-30	Reserved for national use
31	Unspecified or non-standard

Note – It is for further study whether a portion of the reserved range will be used to specify access interfaces associated with ISDN service.

The three bits of the second subfield (UUU) are used to designate a secondary, network-specific subclass code that has billing, accounting, or tariff significance. The origination/destination network can optionally use this subfield to specify one of up to seven subclass codes, with a significance set by the network providing the tariff class code value. If this secondary subfield is not utilized, it should be zero filled.

5.4.3.14 *Coding of network user identification utility parameter*

The octet following the utility code field indicates the length, in octets, of the utility parameter field. The next octet (the first octet of the parameter field) has one of two alternative formats:

a) CCITT Standardized Default Format:

Bit: 8 7 6 5 4 3 2 1
1 1 V R N F V E

Where V, R, NF, VE, and the remaining octets of parameter field for this case are specified below.

b) Format Not Constrainted by This Recommendation:

Bit: 8 7 6 5 4 3 2 1
Y Y X X X X X X

Where YY = 00, 01, or 10. Neither XXXXXX nor the remaining octets of the parameter field in this case are constrained by this Recommendation.

For the CCITT standardized default format (case a) above), all of the following apply:

V Bit: 6
0 NUI Value Unverified
1 (Reserved for "NUI Value Verified")

The use and coding of the R bit is for further study. Until this use is specified, this bit value is always to be set to 0.

The format option used for the NUI code proper is encoded in the NF bits:

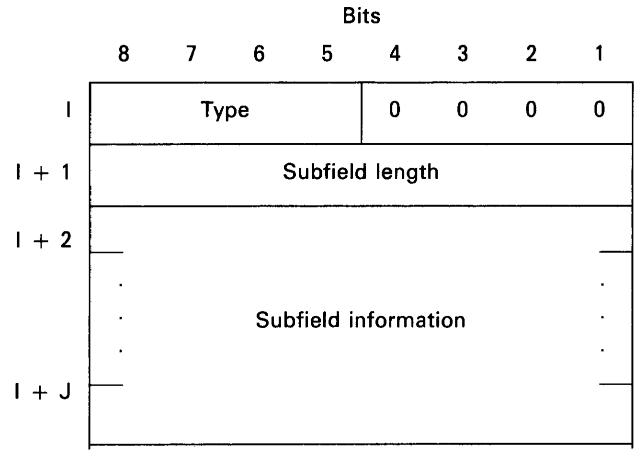
NF Bits: 4 3
0 0 First Subfield Conforms to ISO 7812/CCITT E.118
0 1 No Constraints on Following Octets
1 0 Subfield Format: No Content Constraints
1 1 [*Reserved*]

The verifying entity is encoded in the VE bits:

VE Bits: 2 1
0 0 Originating Network
0 1 Destination Network
1 0 First Transit Network (Note)
1 1 Other/Not Specified

Note – The use of international transit networks as verifying entities is for further study.

If NF = 01, the remaining octets of the parameter field are not constrained by this Recommendation. If NF = 00 or NF = 10, the remaining octets of the parameter field contain the NUI code proper and are divided into m subfields (m greater or equal to 1) and each subfield is defined as follows:



where I is the number of the initial octet of the subfield and (J – 1) is the number of octets of information in the subfield. The type semi-octet specifies the encoding format for the information of the subfield, as follows:

Bits 8	7	6	5	
1	1	0	1	BCD semi-octet
1	1	0	0	IA5 (T50) with bit 8 = 0
1	1	1	0	Reserved for national use
1	1	1	1	
Other				For future definition

Bits 4 through 1 of the first octet of each subfield are set to 0. Other values for this semi-octet are reserved for future use.

Subfield length is the number of semi-octets of information in the subfield, and is encoded in binary.

Note 1 – For Type = 1100 (IA5), subfield length must be an even value. For Type = 1101 (BCD), subfield length may be an even or odd value, although an integral number of octets will be assured by inserting zeros in bits 4, 3, 2 and 1 of the last octet of the subfield when necessary.

Note 2 – The need for a maximum value for the length of this utility parameter field, and the value of such a maximum, are for further study.

5.4.3.15 *Coding of RPOA selection utility*

The parameter field contains the DNIC or INIC (see § 5.3 above) for a requested RPOA transit network and is in the form of four decimal digits.

Each digit is coded in a semi-octet in binary coded decimal with bit 5 or 1 being the low order bit of the digit. The high order digit is coded into bits 8 to 5 of the first octet of the parameter.

5.4.3.16 *Coding of the utility marker utility parameter*

Bit: 8 7 6 5 4 3 2 1

Code: 0 0 0 0 0 0 0 0

ANNEX A

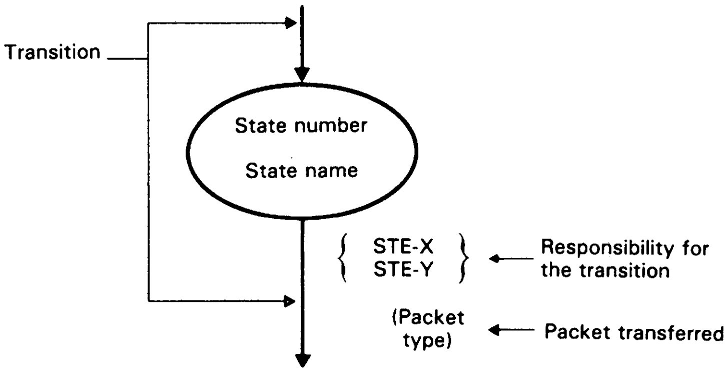
(to Recommendation X.75)

Definition of symbols for Annexes B, C and D

A.1 *General*

This annex contains the definitions for the symbols to be used in annexes B, C and D. Annex B defines the states of the X/Y interface and the transitions between states in the normal case, while Annex C contains the full definition of actions, if any, to be taken on the receipt of packets by an STE. Annex D describes the actions taken by the STE on time-outs, if any, in the packet layer.

A.2 *Symbol definition of the state diagrams*



CCITT-19172

Note 1 – Each state is represented by an ellipse wherein the state name and number are indicated.

Note 2 – Each state transition is represented by an arrow. The responsibility for the transition (STE-X or STE-Y) and the packet that has been transferred are indicated beside that arrow.

FIGURE A-1/X.75

Symbol definition of the state diagrams

A.3 *Order definition of the state diagrams*

For the sake of clarity, the normal procedure at the interface is described in a number of small state diagrams. In order to describe the normal procedure fully, it is necessary to allocate a priority to the different figures and to relate a higher order diagram with a lower one. This has been done by the following means:

- The figures are arranged in order of priority with Figure A-2/X.75 (*restart*) having the highest priority and subsequent figures having lower priority. Priority means that when a packet belonging to a higher order diagram is transferred, that diagram is applicable and the lower order one is not.
- The relation with a state in a lower order diagram is given by including that state inside an ellipse in the higher order diagram.

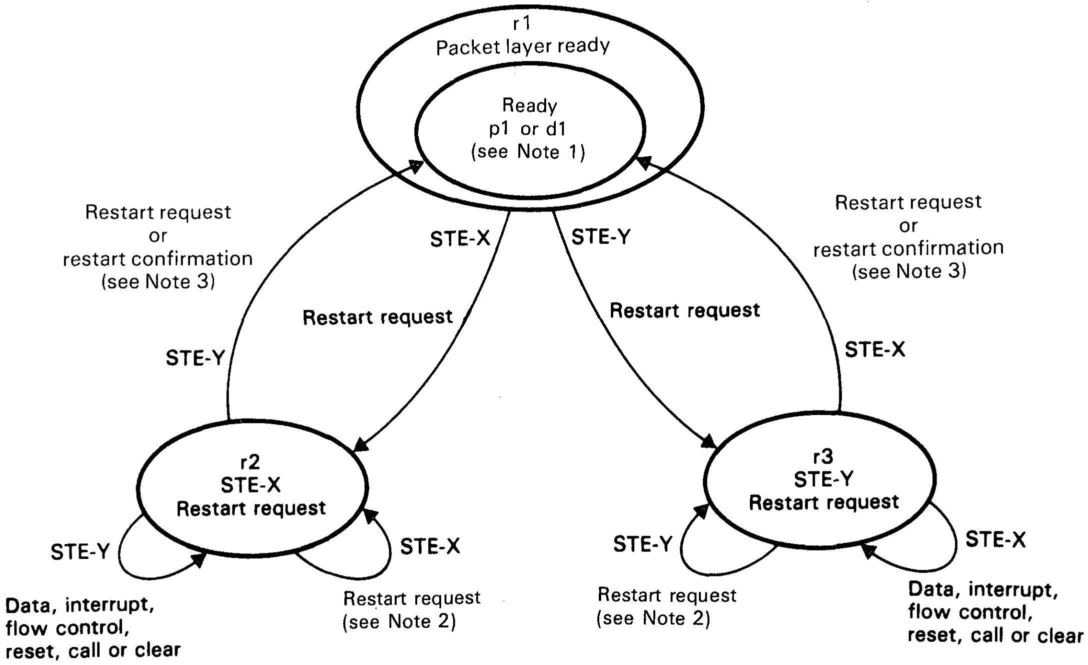
A.4 *Symbol definition of the action tables*

The entries given in Tables C-1/X.75 to C-5/X.75 and D-1/X.75 (see Annexes C and D) indicate the action, if any, to be taken by an STE on receipt of any kind of packet, and the state the STE enters, which is given in parenthesis, following the action taken.

ANNEX B

(to Recommendation X.75)

State diagrams for the packet layer interface between STEs for normal cases



CCITT-19180

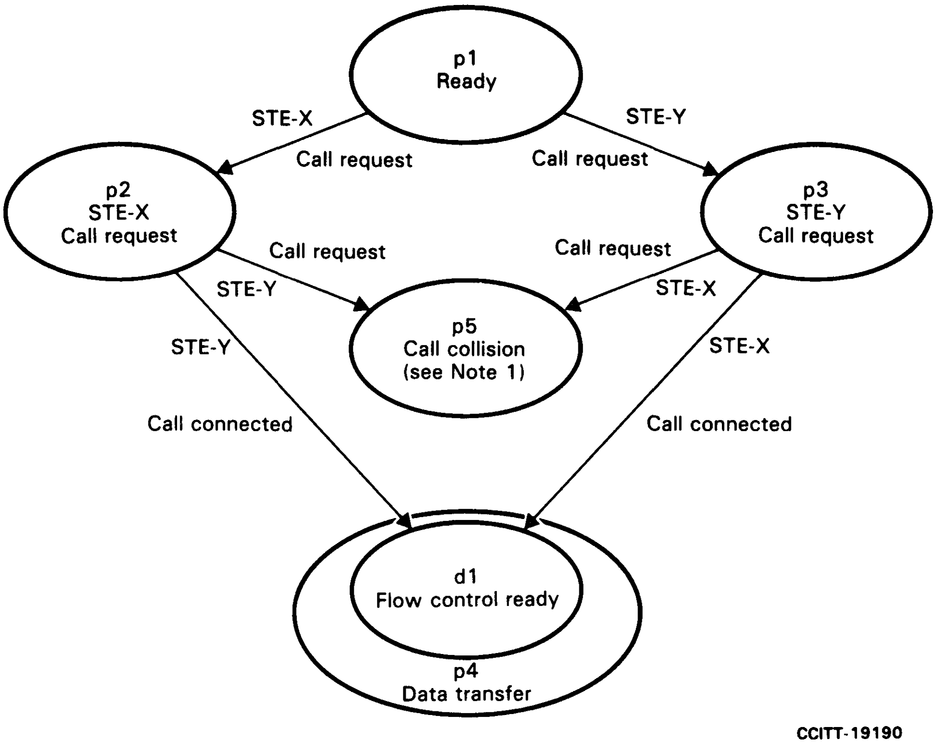
Note 1 – State p1 for virtual calls or state d1 for permanent virtual circuits.

Note 2 – This transition takes place after time-out T30 expires the first time.

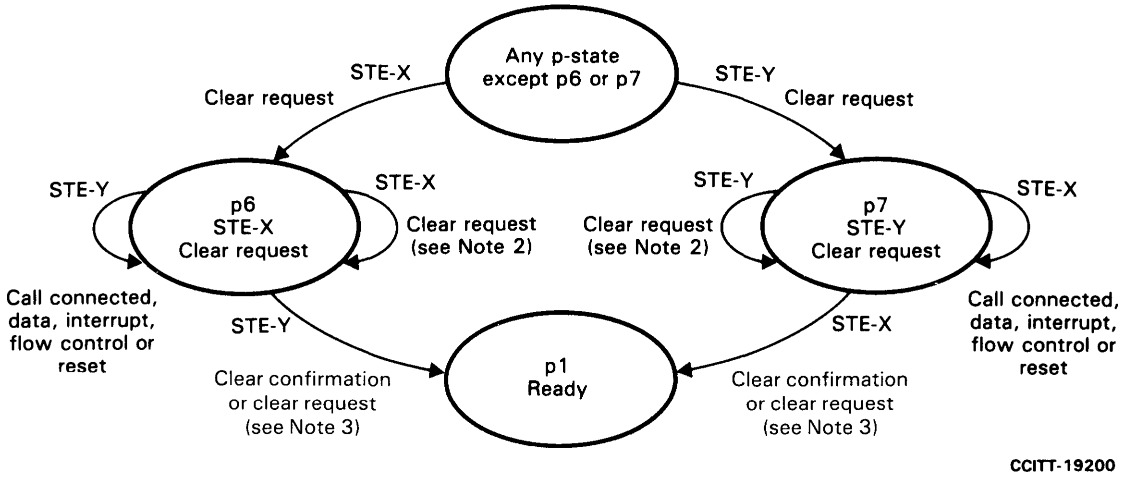
Note 3 – This transition takes place without the transmission of the packet after time-out T30 expires the second time.

FIGURE B-1/X.75

Diagram of states for the transfer of restart packets



a) Transfer of call establishment packets



b) Transfer of call clearing packets

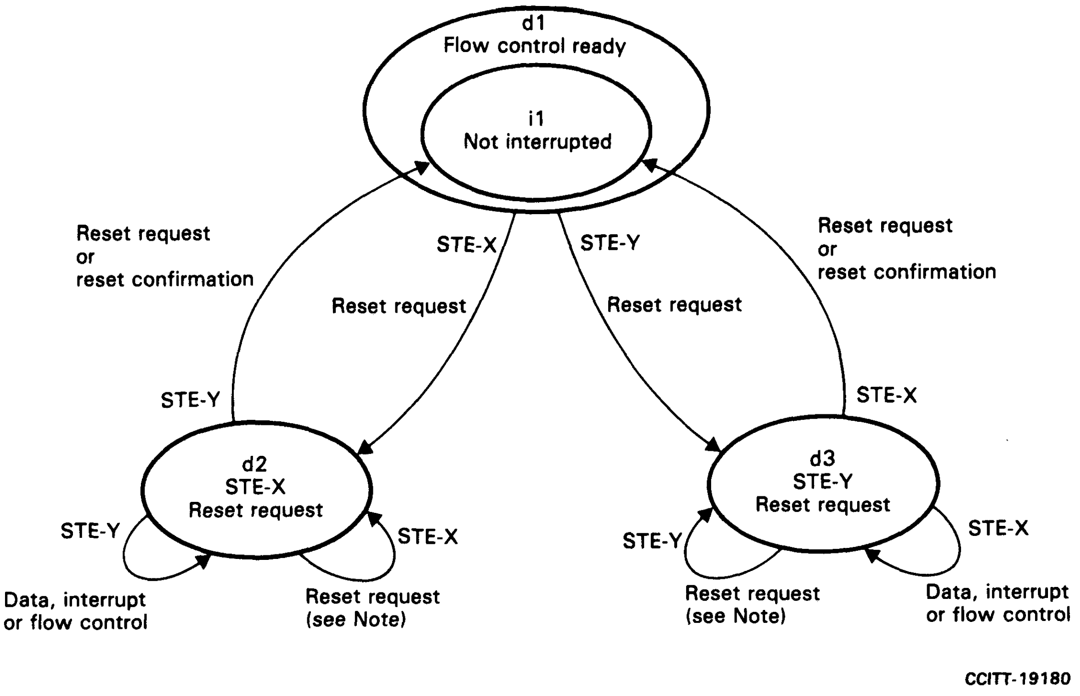
Note 1 – STE-X/Y shall issue a *clear request* packet and proceed to states p6/p7.

Note 2 – This transition takes place after time-out T33 expires the first time.

Note 3 – This transition takes place without the transmission of the packet after time-out T33 expires the second time.

FIGURE B-2/X.75

State diagrams for the transfer of call establishment and call clearing packets within the packet layer ready (r1) state on a logical channel



Note – This transition may take place after time-out T32 expires the first time.

FIGURE B-3/X.75

Diagram of states for the transfer of reset packets within the data transfer (p4) state on a logical channel

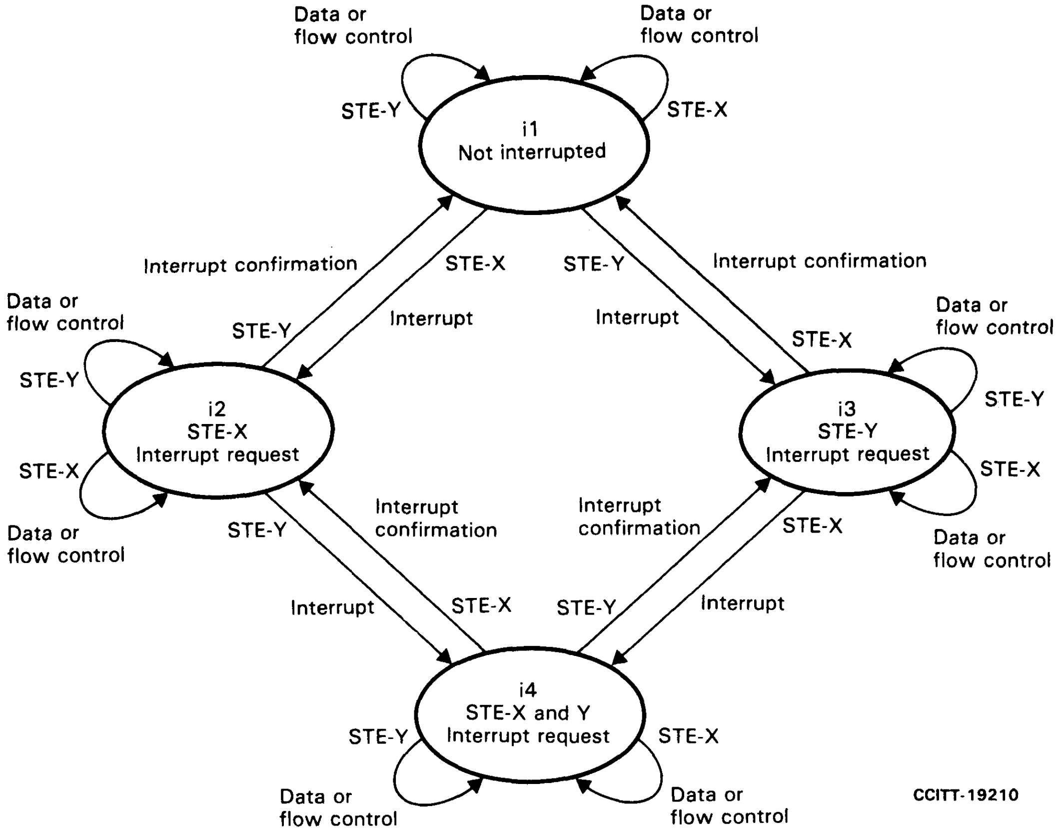


FIGURE B-4/X.75

Diagram of states for the transfer of data, flow control and interrupt packets within the flow control ready (d1) state on a logical channel

ANNEX C

(to Recommendation X.75)

Actions taken by the STE on receipt of packets in a given state of the packet layer X/Y interface

Note – Actions are specified for STE-Y only. STE-X should follow the same procedure.

TABLE C-1/X.75

Action taken by STE-Y on receipt of packets

State of the interface as perceived by STE-Y Packet received by STE-Y	Any state
Any packet with unassigned logical channel (see Note)	DISCARD
Any packet with less than 2 octets	
Any packet with an incorrect general format identifier	
Any packet with correct general format identifier and assigned logical channel (see Note)	(See Table C-2/X.75)

DISCARD STE-Y discards the received packet and takes no subsequent action.

Note – Assigned logical channel includes the case where bits 1 to 4 of octet 1 and bits 1 to 8 of octet 2 are all 0s.

TABLE C-2/X.75

Action taken by STE-Y on receipt of packets in a given state: restart

State of the interface as perceived by STE-Y Packet received by STE-Y	Packet layer ready r1	STE-X restart request r2	STE-Y restart request r3
Restart request	NORMAL (r2)	DISCARD (r2)	NORMAL (r1)
Restart confirmation	ERROR (r3) #17	ERROR (r3) (see Note 1) #18	NORMAL (r1)
Restart request or confirmation with bit 1 to 4 of octet 1 or bit 1 to 8 of octet 2 ≠ 0	(See Table C-3/X.75)	ERROR (r3) (see Note 1) #41	DISCARD (r3)
Data, interrupt, flow control, reset, call set-up or clear when both logical channel number and logical channel group number are not all 0s		ERROR (r3) (see Note 1) #18	
Packet having a packet type identifier which is shorter than 1 octet or is incompatible with the ones defined in § 4 of the text when both logical channel number and logical channel group number are not all 0s		ERROR (r3) (see Note 1) #38 or #33	
Data, interrupt, flow control, reset, call set up, clear, packet having a packet type identifier which is shorter than 1 octet or is incompatible with the ones defined in § 4 of the text when both logical channel number and logical channel group number are all 0s	DISCARD (r1)	DISCARD (r2)	DISCARD (r3)

NORMAL	The action taken by STE-Y follows the normal procedures as defined in § 3 of the text (see Note 2).
DISCARD	STE-Y discards the received packet and takes no subsequent action.
ERROR	STE-Y discards the received packet and indicates restarting with "network congestion" cause and decimal diagnostic value #n.

Note 1 – If STE-Y issues a *restart request* packet as a result of an error condition in state r2, it should follow the actions described in Annex D.

Note 2 – In the two following error situations the STE will invoke the ERROR (r3) procedure:

a) a *restart request* packet or *restart confirmation* packet received in state r3 exceeds the maximum permitted length, is too short or (where detection of non-octet alignment is made at packet layer) is not octet aligned; diagnostic values #39, #38 and #82 respectively are used.

b) a *restart request* packet received in state r1 exceeds the maximum permitted length, is too short or (where detection of non-octet alignment is made at packet layer) is not octet aligned; diagnostic values #39, #38 and #82 respectively, are used.

TABLE C-3/X.75

Action taken by STE-Y on receipt of packets specifying an assigned logical channel in a given state: call establishment and clearing

State of the interface as perceived by STE-Y / Packet received by STE-Y	Packet layer ready r1					
	Ready p1	STE-X call request p2	STE-Y call request p3	Data transfer p4	STE-X clear request p6	STE-Y clear request p7
Call request	NORMAL (p2)	ERROR (p7) # 21	ERROR (p7) # 116	ERROR (p7) # 23	ERROR (p7) (see Note 1) # 25	ERROR (p7) # 26
Call connected	ERROR (p7) # 20	ERROR (p7) # 21	NORMAL (p4) (see Note 2)	ERROR (p7) # 23	ERROR (p7) (see Note 1) # 25	DISCARD (p7)
Clear request	NORMAL (p6)	NORMAL (p6)	NORMAL (p6)	NORMAL (p6)	DISCARD (p6)	NORMAL (p1)
Clear confirmation	DISCARD (p1)	ERROR (p7) # 21	ERROR (p7) # 22	ERROR (p7) # 23	ERROR (p7) (see Note 1) # 25	NORMAL (p1)
Data, interrupt, flow control or reset	ERROR (p7) # 20	ERROR (p7) # 21	ERROR (p7) # 22	(See Table C-4/X.75)	ERROR (p7) (see Note 1) # 25	DISCARD (p7)
Restart request or confirmation with bit 1 to 4 of octet 1 or bit 1 to 8 of octet 2 ≠ 0	ERROR (p7) # 41	ERROR (p7) # 41	ERROR (p7) # 41		ERROR (p7) (see Note 1) # 41	
Packet having a packet type identifier which is shorter than 1 octet or is incompatible with the ones defined in § 4 of the text	ERROR (p7) # 38 or # 33	ERROR (p7) # 38 or # 33	ERROR (p7) # 38 or # 33		ERROR (p7) (see Note 1) # 38 or # 33	

NORMAL	The action taken by STE-Y follows the normal procedures as defined in § 3 of the text. However, if an error condition specified in Annex F occurs, STE-Y discards the received packet and indicates clearing with the cause and diagnostic codes specified in Annex F.
DISCARD	STE-Y discards the received packet and takes no subsequent action.
ERROR	STE-Y discards the received packet and indicates clearing with "network congestion" cause and decimal diagnostic value # n.

Note 1 – If STE-Y issues a *clear request* packet as a result of an error condition in state p6, it should follow the actions described in Annex D.

Note 2 – The ERROR (p7) procedure is invoked if STE-Y receives a *call connected* packet in response to a *call request* packet from STE-Y requesting the *fast select* facility with restriction on response.

TABLE C-4/X.75

Action taken by STE-Y on receipt of packets specifying an assigned logical channel in a given state: reset

State of the interface as perceived by STE-Y Packet received by STE-Y	Data transfer p4		
	Flow control ready d1	STE-X reset request d2	STE-Y reset request d3
Reset request	NORMAL (d2)	DISCARD (d2)	NORMAL (d1)
Reset confirmation	ERROR (d3) # 27	ERROR (d3) # 28	NORMAL (d1)
Data, interrupt or flow control	(See Table C-5/X.75)	ERROR (d3) # 28	DISCARD (d3)
Restart request or confirmation with bit 1 to 4 of octet 1 or bit 1 to 8 of octet 2 ≠ 0	ERROR (d3) # 41	ERROR (d3) (see Note 1) # 41	DISCARD (d3)
Packet having a packet type identifier which is shorter than 1 octet or is incompatible with the ones defined in § 4 of the text	ERROR (d3) # 38 or # 33	ERROR (d3) (see Note 1) # 38 or # 33	
Invalid packet type on a permanent virtual circuit	ERROR (d3) # 35	ERROR (d3) (see Note 1) # 35	

NORMAL	The action taken by STE-Y follows the normal procedures as defined in § 3 of the text (see Note 2).
DISCARD	STE-Y discards the received packet and takes no subsequent action.
ERROR	STE-Y discards the received packet and indicates resetting with "network congestion" cause and decimal diagnostic value # n.

Note 1 – If STE-Y issues a *reset request* packet as a result of an error condition in state d2, it should follow the actions described in Annex D.

Note 2 – In the following error situations the STE will invoke the ERROR (d3) procedure: the received packet exceeds the maximum permitted length, is too short or (where detection of non-octet alignment is made at packet layer) is not octet aligned; diagnostic values # 39, # 38 and # 82 respectively are used.

TABLE C-5/X.75

Action taken by receipt of packets specifying an assigned logical channel in a given state: data, interrupt or flow control

State of the interface as perceived by STE-Y / Packet received by STE-Y	Flow control ready d1			
	Not interrupted i1	STE-X interrupt request i2	STE-Y interrupt request i3	STE-X and Y interrupt request i4
Interrupt	NORMAL (i2)	DISCARD (i2) or ERROR (d3) (see Note 1) # 44	NORMAL (i4)	DISCARD (i4) or ERROR (d3) (see Note 1) # 44
Interrupt confirmation	DISCARD (i1)	DISCARD (i2)	NORMAL (i1)	NORMAL (i2)
Data with out of sequence P(S) or P(S) outside of window	ERROR (d3) # 1	ERROR (d3) # 1	ERROR (d3) # 1	ERROR (d3) # 1
Data with M bit violation	ERROR (d3) # 103	ERROR (d3) # 103	ERROR (d3) # 103	ERROR (d3) # 103
Data with inconsistent Q bit setting	NORMAL (i1) or ERROR (d3) # 83 (see Note 3)	NORMAL (i2) or ERROR (d3) # 83 (see Note 3)	NORMAL (i3) or ERROR (d3) # 83 (see Note 3)	NORMAL (i4) or ERROR (d3) # 83 (see Note 3)
Data or flow control with invalid P(R)	ERROR (d3) # 2	ERROR (d3) # 2	ERROR (d3) # 2	ERROR (d3) # 2
A first data packet after entering state d1 with P(S) ≠ 0	ERROR (d3) # 1	ERROR (d3) # 1	ERROR (d3) # 1	ERROR (d3) # 1
When modulo 128 numbering is used, a flow control or data packet with octet 4 shorter than 1 octet	ERROR (d3) # 38	ERROR (d3) # 38	ERROR (d3) # 38	ERROR (d3) # 38
Valid data or flow control	NORMAL (i1)	NORMAL (i2)	NORMAL (i3)	NORMAL (i4)

NORMAL	The action taken by STE-Y follows the normal procedures as defined in § 3 of the text (see Note 2).
DISCARD	STE-Y discards the received packet and takes no subsequent action.
ERROR	STE-Y discards the received packet and indicates reset with "network congestion" cause and decimal diagnostic value # n.

Note 1 – According to § 3.3.5 an STE receiving a further *interrupt* packet in the time between receiving one *interrupt* packet and transferring the interrupt confirmation may either discard this *interrupt* packet or reset the virtual call or the permanent virtual circuit.

Note 2 – In the following error situations the STE will invoke the ERROR (d3) procedure: the received packet exceeds the maximum permitted length, is too short or (where detection of non-octet alignment is made at packet layer) is not octet aligned; diagnostic values # 39, # 38 and # 82 respectively are used.

Note 3 – According to § 3.3.4 if an STE detects that the value of the Q bit has changed within a packet sequence it may reset the virtual call or permanent virtual circuit.

ANNEX D

(to Recommendation X.75)

Actions taken by the STE on time-outs in the packet layer

Under certain circumstances, the STE Y/X is required to respond to a packet from the STE X/Y within a stated maximum time. If any of these maximum times are exceeded, a time-out in the STE X/Y will initiate the actions summarized in Tables D-1/X.75 and D-2/X.75. Therefore, this must be taken into account in the STE design.

TABLE D-1/X.75

STE X/Y time-outs (first time)

Time-out number	Time-out value	State of the logical channel	Started when	Normally terminated when	Actions to be taken the first time the time-out expires	
					Toward STE Y/X	Toward network
T30	180 sec	r2/r3	STE X/Y issues a *restart request* packet	STE X/Y leaves the r2/r3 state (i.e., a *restart confirmation* or *restart request* packet is received)	STE X/Y signals a *restart request* packet (*network congestion,* # 52) again, and restarts time-out T30	For permanent virtual circuits, the STE signals a *reset request* packet (*network congestion,* # 52)
T31	200 sec	p2/p3	STE X/Y issues a *call request* packet	STE X/Y leaves the p2/p3 state (e.g., a *call connected, clear request* or *call request* packet is received)	STE X/Y enters the p6/p7 state signalling a *clear request* packet (*network congestion,* # 49)	STE X/Y signals a *clear request* packet (*network congestion,* # 49)
T32	180 sec	d2/d3	STE X/Y issues a *reset request* packet	STE X/Y leaves the d2/d3 state (e.g., a *reset confirmation* or *reset request* packet is received)	STE X/Y signals a *reset request* packet (*network congestion,* # 51) again and restarts time-out T32	STE X/Y signals *reset request* packet (*network congestion,* # 51)
T33	180 sec	p6/p7	STE X/Y issues a *clear request* packet	STE X/Y leaves the p6/p7 state (e.g., a *clear confirmation* or *clear request* packet is received)	STE X/Y signals a *clear request* packet (*network congestion,* # 50) again, and restarts time-out T33	

TABLE D-2/X.75

STE X/Y time-outs (second time)

Time-out number	Actions to be taken the second time the time-out expires	
	Toward STE Y/X	Toward network
T30	STE X/Y enters the r1 state *Note* – Further actions may be initiated at higher level	For permanent virtual circuits, STE X/Y signals a *reset request* packet (*network congestion*, # 52)
T31	(Not possible; T31 is not restarted after it has expired)	
T32	For virtual calls, STE X/Y enters the p6/p7 state signalling a *clear request* packet (*network congestion*, # 51) For permanent virtual circuits, STE X/Y enters the d1 state	For virtual calls, STE X/Y signals a *clear request* packet (*network congestion*, # 51) For permanent virtual circuits, STE X/Y signals a *reset request* packet (*network congestion*, # 51)
T33	STE X/Y enters the p1 state.	

ANNEX E

(to Recommendation X.75)

Coding of network generated diagnostic fields in X.75 clear, reset and restart packets

TABLE E-1/X.75

(See Notes 1, 2, 3 and 9)

Diagnostics	Bits								Decimal
	8	7	6	5	4	3	2	1	
No additional information	0	0	0	0	0	0	0	0	0
Invalid P(S)	0	0	0	0	0	0	0	1	1
Invalid P(R)	0	0	0	0	0	0	1	0	2
	0	0	0	0	1	1	1	1	15
Packet type invalid	0	0	0	1	0	0	0	0	16
For state r1	0	0	0	1	0	0	0	1	17
For state r2	0	0	0	1	0	0	1	0	18
For state r3	0	0	0	1	0	0	1	1	19
For state p1	0	0	0	1	0	1	0	0	20
For state p2	0	0	0	1	0	1	0	1	21
For state p3	0	0	0	1	0	1	1	0	22
For state p4	0	0	0	1	0	1	1	1	23
For state p5	0	0	0	1	1	0	0	0	24
For state p6	0	0	0	1	1	0	0	1	25
For state p7	0	0	0	1	1	0	1	0	26
For state d1	0	0	0	1	1	0	1	1	27
For state d2	0	0	0	1	1	1	0	0	28
For state d3	0	0	0	1	1	1	0	1	29
	0	0	0	1	1	1	1	1	31
Packet not allowed	0	0	1	0	0	0	0	0	32
Unidentifiable packet	0	0	1	0	0	0	0	1	33
Call on one way logical channel (Note 4)	0	0	1	0	0	0	1	0	34
Invalid packet type on a permanent virtual circuit	0	0	1	0	0	0	1	1	35
Packet on unassigned logical channel	0	0	1	0	0	1	0	0	36
Reject not subscribed to (Note 4)	0	0	1	0	0	1	0	1	37
Packet too short	0	0	1	0	0	1	1	0	38
Packet too long	0	0	1	0	0	1	1	1	39
Invalid general format identifier	0	0	1	0	1	0	0	0	40
Restart with non-zero in bits 1-4, 9-16	0	0	1	0	1	0	0	1	41
Packet type not compatible with facility/utility (Note 5)	0	0	1	0	1	0	1	0	42
Unauthorized interrupt confirmation	0	0	1	0	1	0	1	1	43
Unauthorized interrupt	0	0	1	0	1	1	0	0	44
Unauthorized reject (Note 4)	0	0	1	0	1	1	0	1	45
	0	0	1	0	1	1	1	1	47

TABLE E-1/X.75 *(continued)*

Diagnostics	Bits								Decimal
	8	7	6	5	4	3	2	1	
Time expired	0	0	1	1	0	0	0	0	48
For incoming call/call request (Note 6)	0	0	1	1	0	0	0	1	49
For clear indication/request (Note 6)	0	0	1	1	0	0	1	0	50
For reset indication/request (Note 6)	0	0	1	1	0	0	1	1	51
For restart indication/request (Note 6)	0	0	1	1	0	1	0	0	52
	0	0	1	1	1	1	1	1	63
Call set-up or clearing problem	0	1	0	0	0	0	0	0	64
Facility/utility code not allowed (Note 5)	0	1	0	0	0	0	0	1	65
Facility/utility parameter not allowed (Note 5)	0	1	0	0	0	0	1	0	66
Invalid called address	0	1	0	0	0	0	1	1	67
Invalid calling address	0	1	0	0	0	1	0	0	68
Invalid facility length	0	1	0	0	0	1	0	1	69
Incoming call barred	0	1	0	0	0	1	1	0	70
No logical channel available	0	1	0	0	0	1	1	1	71
Call collision	0	1	0	0	1	0	0	0	72
Duplicate facility/utility requested (Note 5)	0	1	0	0	1	0	0	1	73
Non-zero address length	0	1	0	0	1	0	1	0	74
Non-zero facility length	0	1	0	0	1	0	1	1	75
Facility/utility not provided when expected (Note 5)	0	1	0	0	1	1	0	0	76
Invalid CCITT-specified DTE facility	0	1	0	0	1	1	0	1	77
	0	1	0	0	1	1	1	1	79
Miscellaneous	0	1	0	1	0	0	0	0	80
Improper cause code from DTE/STE (Note 7)	0	1	0	1	0	0	0	1	81
Octet non-aligned	0	1	0	1	0	0	1	0	82
Inconsistent Q bit setting	0	1	0	1	0	0	1	1	83
NUI problem	0	1	0	1	0	1	0	0	84
	0	1	0	1	1	1	1	1	95
Inter-network call set-up or clearing problem	0	1	1	0	0	0	0	0	96
Unknown calling DNIC	0	1	1	0	0	0	0	1	97
TNIC mismatch	0	1	1	0	0	0	1	0	98
Call identifier mismatch	0	1	1	0	0	0	1	1	99
Negotiation error in utility parameter value	0	1	1	0	0	1	0	0	100
Invalid utility length	0	1	1	0	0	1	0	1	101
Non-zero utility length	0	1	1	0	0	1	1	0	102
M bit violation	0	1	1	0	0	1	1	1	103
	0	1	1	0	1	1	1	1	111

TABLE E-1/X.75 *(continued)*

Diagnostics	Bits								Decimal
	8	7	6	5	4	3	2	1	
Inter-network problem	0	1	1	1	0	0	0	0	112
Remote network problem	0	1	1	1	0	0	0	1	113
Inter-network protocol problem	0	1	1	1	0	0	1	0	114
Inter-network link out of order	0	1	1	1	0	0	1	1	115
Inter-network link busy	0	1	1	1	0	1	0	0	116
Transit network facility problem	0	1	1	1	0	1	0	1	117
Remote network facility problem	0	1	1	1	0	1	1	0	118
Inter-network routing problem	0	1	1	1	0	1	1	1	119
Temporary routing problem	0	1	1	1	1	0	0	0	120
Unknown called DNIC	0	1	1	1	1	0	0	1	121
Maintenance action	0	1	1	1	1	0	1	0	122
	0	1	1	1	1	1	1	1	127
Reserved for network specific diagnostic information (Note 8)	1	0	0	0	0	0	0	0	128
	1	1	1	1	1	1	1	1	255

Note 1 – Not all diagnostic codes need apply to a specific network, but those used are coded as in the table.

Note 2 – A given diagnostic need not apply to all packet types (i.e. *reset request*, *clear request* and *restart request* packets).

Note 3 – The first diagnostic in each grouping is a generic diagnostic and can be used in place of the more specific diagnostics within the grouping. The decimal 0 diagnostic code can be used in situations where no additional information is available.

Note 4 – Only generated at a user interface (see Recommendation X.25).

Note 5 – When associated with the cause "Network congestion", this indicates a utility problem; when associated with any other valid cause (see Tables 13/X.75, 15/X.75 and 17/X.75) this indicates a facility problem at a user interface.

Note 6 – When associated with the cause "Network congestion", this indicates an X.75 packet timer problem; when associated with any other valid cause (see Tables 13/X.75, 15/X.75 and 17/X.75) this indicates a packet timer problem at a user interface.

Note 7 – When associated with the cause "Network congestion", this indicates an invalid cause detected on an X.75 link; when associated with any other valid cause (see Tables 13/X.75, 15/X.75 and 17/X.75) this indicates an invalid cause detected at a user interface.

Note 8 – When the associated cause is "Network congestion", diagnostic codes in this range may by bilateral agreement between Administrations, be transferred across an X.75 link. However, the receiving network will alter such values, as described in §§ 4.2.3.2, 4.4.3.2 or 4.5.1.2 as appropriate, before passing them to another network or across a user interface.

Note 9 – When the associated cause is "Network congestion", diagnostic codes in the range 1 to 111 will be altered by the receiving network, as described in §§ 4.2.3.2 , 4.4.3.2 or 4.5.1.2 as appropriate, before passing them to another network or across a user interface.

ANNEX F

(to Recommendation X.75)

Association of error conditions with cause and diagnostic codes

a) *Call request packet*

Error condition	Cause	Specific diagnostics (see Note 3 of Annex E)
1. Not octet aligned packet (where detection of non-octet alignment is made at packet level if implemented; see § 3)	Network congestion	# 82
2. Address contains a non-BCD digit	Network congestion	# 67, 68
3. Address less than four digits	Network congestion	# 67, 68
4. Bit 8 of the octet which indicates the facility length field not set to 0	Network congestion	# 69
5. No combination of utilities could equal utility length	Network congestion	# 101
6. Facility or utility length larger than remainder of packet	Network congestion	# 38
7. Utility value conflicts (e.g., a particular combination not supported)	Network congestion	# 66
8. Utility code not allowed	Network congestion	# 65
9. Utility value not allowed or invalid	Network congestion	# 66
10. Utility expected and not provided	Network congestion	# 76
11. Packet too short	Network congestion	# 38
12. Address length larger than remainder of packet	Network congestion	# 38
13. Call user data larger than 16, or 128 octets in case of fast select facility	Network congestion	# 39
14. Class coding of the utility corresponding to a length of parameter larger than remainder of packet	Network congestion	# 101
15. Utility code (except TNIC and RPOA) repeated	Network congestion	# 73
16. Duplicate TNIC	Network congestion	# 66
17. Unknown calling network identification	Network congestion	# 97
18. Bits 7 or 8 of the utility length field octet not set to 0	Network congestion	# 101
19. Unknown number	Not obtainable	# 67
20. Incoming call barred	Access barred	# 70
21. Closed user group protection	Access barred	# 65

Error condition	Cause	Specific diagnostics (see Note 3 of Annex E)
22. Reverse charging rejected	Reverse charging acceptance not subscribed	# 0
23. Fast select rejected	Fast select acceptance not subscribed	# 0
24. National address smaller than national address format permits	Not obtainable	# 67, 68
25. National address larger than national address format permits	Not obtainable	# 67, 68
26. Called DTE out of order	Out of order	# 0 # greater than 127
27. No logical channel available	Number busy	# 71
28. Call collision	Number busy	# 71, 72
29. The remote DTE/DCE interface does not support a function or a facility request	Incompatible destination	# 0
30. Procedure error at the remote DTE/DCE interface	Remote procedure error	(see Annex E)
31. Network congestion or fault condition within the network	Network congestion	# 0 # greater than 127 (see Note 8 to Annex E)
32. Planned maintenance activity within the network	Network congestion	# 122
33. Network fault condition detected other than at the local STE-X/Y interface	Network congestion	# 113
34. X.75 protocol error detected other than at the local STE-X/Y interface	Network congestion	# 114
35. No inter-Administration service agreement is recorded for calls from the calling network to the called network	Access barred	# 119
36. No inter-Administration service agreement is recorded for calls from the calling network to the called network using the routing indicated	Access barred	# 119
37. The inter-Administration service agreement does not permit calls using the requested facility(ies) from the calling network to the called network	Incompatible destination	# 118

Error condition	Cause	Specific diagnostics (see Note 3 of Annex E)
38. The routing process is unable to determine any suitable outgoing link for the called network	Not obtainable	# 121
39. The routing process is unable to determine a suitable outgoing link with a free logical channel	Network congestion	# 116
40. Call collision is detected on the selected outgoing link	Network congestion	# 116
41. The routing indicated in the received *call request* packet is too long for an overall routing conforming to X.110 to be provided [e.g., because alternative routing has already been used]	Network congestion	# 120
42. The routing indicated in the received *call request* packet cannot be extended to provide an overall routing in conformance with X.110 [e.g., because the prior use of alternative routing means that a circular routing would be formed]	Network congestion	# 120
43. Each of the suitable outgoing links determined by the routing process is subject to an unplanned outage	Network congestion	# 115
44. The routing process is unable to determine a suitable operational outgoing link supporting the requested facility(ies)	Network congestion	# 117
45. The routing process is unable to determine a suitable operational outgoing link supporting the parameter value of a requested facility	Network congestion	# 117
46. None of the suitable outgoing links determined by the routing process is operational, and at least one is subject to a planned outage for essential maintenance	Network congestion	# 122
47. Requested RPOA out of order	RPOA out of order	# 0
48. Requested RPOA invalid or not supported	RPOA out of order	# 119
49. NUI utility value invalid/unsupported or NUI utility required but absent	Access barred	# 84

Note – Error conditions 19 to 30 are examples for problems related to the destination network.

b) *Call connected packet*

Error condition	Cause	Specific diagnostics (see Note 3 of Annex E)
1. Not octet aligned packet (where detection of non-octet alignment is made at packet level if implemented; see § 3)	Network congestion	# 82
2. Address contains a non-BCD digit	Network congestion	# 67, 68
3. Address less than four digits	Network congestion	# 67, 68
4. Bit 8 of the octet which indicates the facility length field not set to 0	Network congestion	# 69
5. No combination of utilities could equal utility length	Network congestion	# 101
6. Facility or utility length larger than remainder of packet	Network congestion	# 38
7. Utility value conflicts (e.g., a particular combination not supported)	Network congestion	# 66
8. Utility code not allowed	Network congestion	# 65
9. Utility value not allowed or invalid	Network congestion	# 66
10. Utility expected and not provided	Network congestion	# 76
11. Packet too short	Network congestion	# 38
12. Address length larger than remainder of packet	Network congestion	# 38
13. Call user data larger than 128 octets in case of *fast select* facility	Network congestion	# 39
14. Call user data present (if *fast select* facility not requested)	Network congestion	# 39
15. Class coding of the utility corresponding to a length of parameter larger than remainder of packet	Network congestion	# 101
16. Utility code (except TNIC and RPOA) repeated	Network congestion	# 73
17. Unknown calling network identification	Network congestion	# 97
18. Bits 7 or 8 of the utility length field octet not set to 0	Network congestion	# 101
19. Duplicate TNIC	Network congestion	# 66
20. The *call request* packet indicated *fast select* with restriction on response	Network congestion	# 42
21. Call identifier mismatch	Network congestion	# 99
22. TNIC mismatch	Network congestion	# 98
23. Negotiation error in utility parameter value	Network congestion	# 100
24. NUI utility value invalid/unsupported or NUI utility required but absent	Access barred	# 84

c) *Clear request packet*

Error condition	Cause	Specific diagnostics (see Note 3 of Annex E)
1. Not octet aligned packet (where detection of non-octet alignment is made at packet level if implemented; see § 3)	Network congestion	# 82
2. Packet too short	Network congestion	# 38
3. Packet too long	Network congestion	# 39
4. Address length fields incorrectly set to non-zero	Network congestion	# 74
5. Utility length field incorrectly set to non-zero	Network congestion	# 102
6. Call user data larger than 128 in case of *fast select* facility (if *fast select* facility requested)	Network congestion	# 39
7. Call user data present (if *fast select* facility not requested)	Network congestion	# 39
8. Improper cause code from STE (if implemented; see § 4.2.3.1)	Network congestion	# 81

d) *Clear confirmation packet*

Error condition	Cause	Specific diagnostics (see Note 3 of Annex E)
1. Not octet aligned packet (where detection of non-octet alignment is made at packet level if implemented; see § 3)	Network congestion	# 82
2. Packet length larger than 3 octets	Network congestion	# 39

APPENDIX I

(to Recommendation X.75)

Examples of multilink resetting procedures

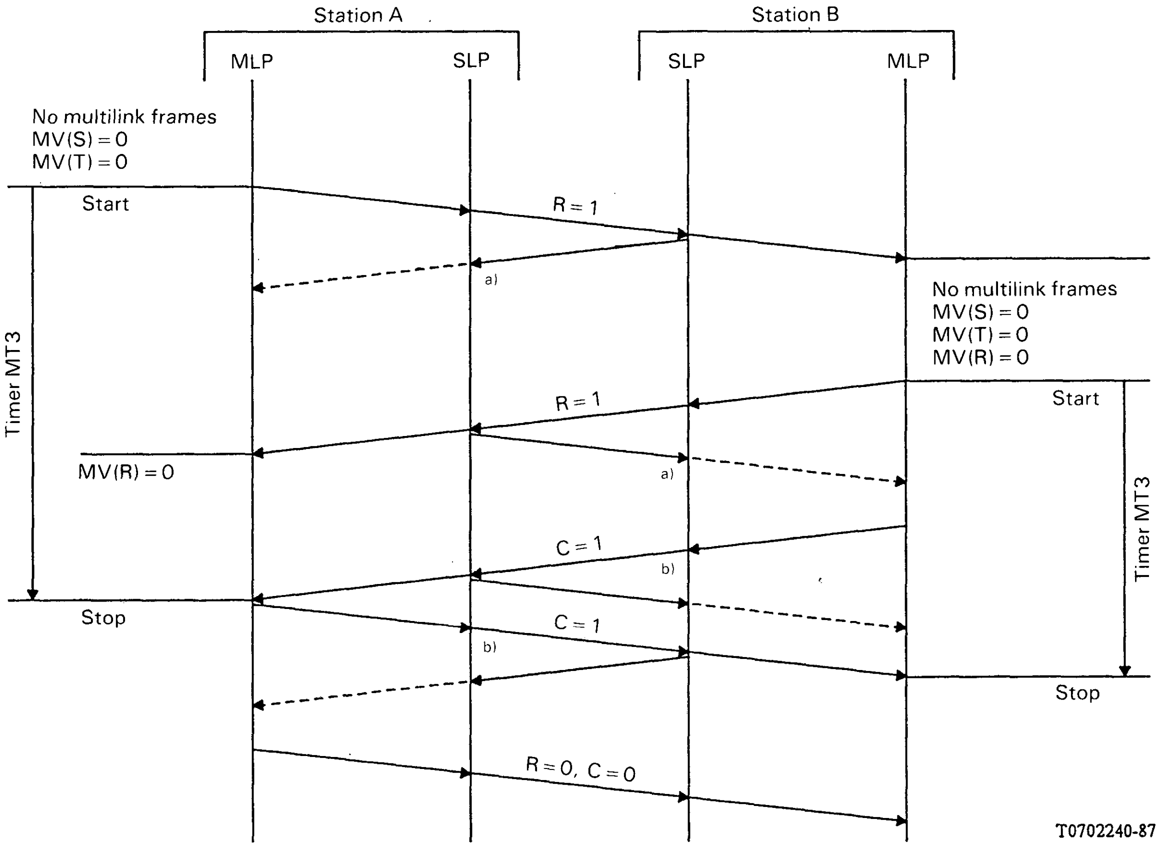
I.1 *Introduction*

The following examples illustrate application of the multilink resetting procedures is the case of:

a) MLP reset initiated by a single STE; and

b) MLP reset initiated by both STEs simultaneously.

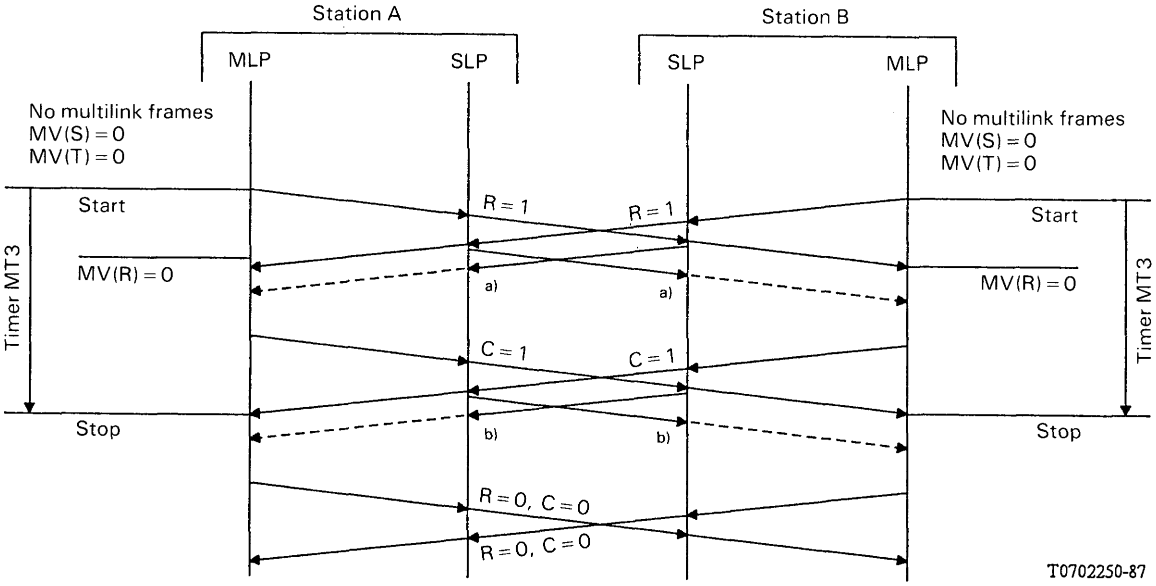
I.2 *MLP reset initiated by a single STE*



a) The SLP frame that acknowledges delivery of the multilink frame with R = 1.

b) The SLP frame that acknowledges delivery of the multilink frame with C = 1.

I.3 *MLP reset initiated by both STEs simultaneously*



[a] The SLP frame that acknowledges delivery of the multilink frame with R = 1.

[b] The SLP frame that acknowledges delivery of the multilink frame with C = 1.

Recommendation X.80

INTERWORKING OF INTEREXCHANGE SIGNALLING SYSTEMS FOR CIRCUIT SWITCHED DATA SERVICES

(Geneva, 1980; amended at Malaga-Torremolinos, 1984)

The CCITT,

considering

(a) that Recommendations X.60 and X.71 define two different signalling systems which are intended for use on international circuits between synchronous data networks;

(b) that Recommendation X.70 defines a signalling system which is intended for use on international circuits between anisochronous data networks;

(c) that Administrations and RPOAs have expressed interest in implementing Recommendations X.60, X.70 or X.71 as national signalling systems between national data switching exchanges;

(d) that Recommendations X.60, X.70 and X.71 have been defined to include the necessary signals to allow interworking between any combination of these signalling systems;

(e) that there is a need to define the specific interworking requirements between these signalling systems;

unanimously declares

that interworking between any combination of signalling systems conforming to Recommendations X.60, X.70 and X.71 should be as defined in this Recommendation.

1 General

1.1 *Principles*

This Recommendation provides a set of interworking specifications for CCITT circuit-switched data signalling systems. Interworking is defined as the controlled transfer of signalling information across an interface between different signalling systems where the significance of the transferred information is identical, or where significance is translated in a defined manner, and includes the performance of the appropriate interworking procedures in association with the transfer. These interworking procedures will be performed by an interworking function at a boundary between the two interworking signalling systems.

Interworking commences at call set-up when a link is established between two circuits using different signalling systems and continues throughout the call until release of the connection occurs. Interworking ceases with the release of the connection, whether the release is initiated by the reception of a clear condition from either of the signalling systems involved or by the interworking function itself in response to some abnormal condition.

1.2 *Presentation*

The specifications are basically represented by flow charts consistent with the CCITT Specifications and Descriptions Language (SDL), described in Recommendations Z.101 to Z.103; and are used to describe the logical requirements of the interworking function. In addition, two tables are included to show the signalling sequences required for a typical interworking situation. Narrative description has been reduced to a minimum.

SDL provides a method of presentation which is both comprehensive and independent of implementation, ensuring that all interworking conditions can be covered in a systematic manner. The logic of each signalling system is covered in the relevant signalling Recommendations X.60, X.70 or X.71.

2 Interworking procedures between Recommendations X.60 and X.71

The present § 2 details the specific requirements for interworking between an X.60 and an X.71 signalling system.

Table 1/X.80 illustrates the relationship between the signals on the X.60 side of the interworking function and the corresponding signals on the X.71 side. It illustrates the simple case of a basic call which originates in an "X.60 network" and terminates in an "X.71 network", and which does not invoke any additional facilities, and it assumes that the call is successful. The call clear-down is initiated by the customer in the X.60 network.

There are, however, several combinations of facilities which could be required on a particular call which complicate the interworking procedures, in particular the instant of connect through. In Table 1/X.80 the reception of the *Call Connected (CC)* signal from the Recommendation X.71 signalling system defines the conclusion of the call set-up sequence at the interworking point and hence the instant of connect through. If the call involves additional facilities the reception of the *Transit Through Connect (TTC)* signal from the Recommendation X.71 signalling system initiates the additional protocols necessary to setting up the call successfully. Table 2/X.80 illustrates an example involving these additional protocols for a call requiring both calling and called line identities and including a positive call progress indication.

Appendix I to this Recommendation illustrates further examples of interworking situations which can occur for the X.60 to X.71 case. The appendix illustrates examples of interworking situations where two "X.71 networks" transit an "X.60 network" or two "X.60 networks" transit an "X.71 network".

2.1 *Interworking from Recommendation X.60 to X.71*

Figure 1/X.80 shows the transit exchange interworking functions required to enable an X.60 to X.71 call to be connected.

In response to the selection information sent to the X.71 network, one of two signals may be received: *CC* or *TTC* as described above.

TABLE 1/X.80

X.60 to X.71 interworking situation—simple case, customer in X.60 network initiates clear

Originating network	Inter-network data circuit		X.60		Transit exchange interworking unit functions	X.71		Destination network
	→	←	→	←		→	←	
Seize free trunk	TF →	← TF						
Send trunk seized	TS →							
Send address message			AM →		Address message received			
					Seize free trunk			
					Send calling signal	CS →		
					Send selection signals	SS →		Selection signals received
							← C.CONF	Send call confirmation signal
					Call connected received		← CC	Call connected sent
								Connect through
Call accepted message received		← CA		← CAM 1	Call accepted message sent		← CA	Call accepted signal sent by customer
Connect through					Connect through	TS →		
Ready for data sent by customer	RD →					RD →		
Data phase	DATA →	← DATA				DATA →	← DATA	Data phase
Customer clears	CLEAR[a)] →				Clear recognized (optionally)	CLEAR →		Clear recognized
Disconnect call		← TF						
Clear request message sent			CM →		Clear message received	CLEAR →		
					Call disconnected			Call disconnected
					Maintain clear signal			
				← CM	Clear message sent			

X.60 signals + messages

TF	Trunk free
TS	Trunk seized
AM	Address message
CAM 1	Call accepted (CC) message
CM	Clear message

X.71 signals

CS	Calling signal
SS	Selection signals
C.CONF	Call confirmation
CC	Call connected
CLEAR	

Customer signals

RD	Ready for data
CA	Call accepted
DATA	
Clear request	

---

a) Customer clear signal same as trunk free.

TABLE 2/X.80

X.60 to X.71 interworking situation — complex case, call set-up only, no internetwork data circuits shown for X.60 network

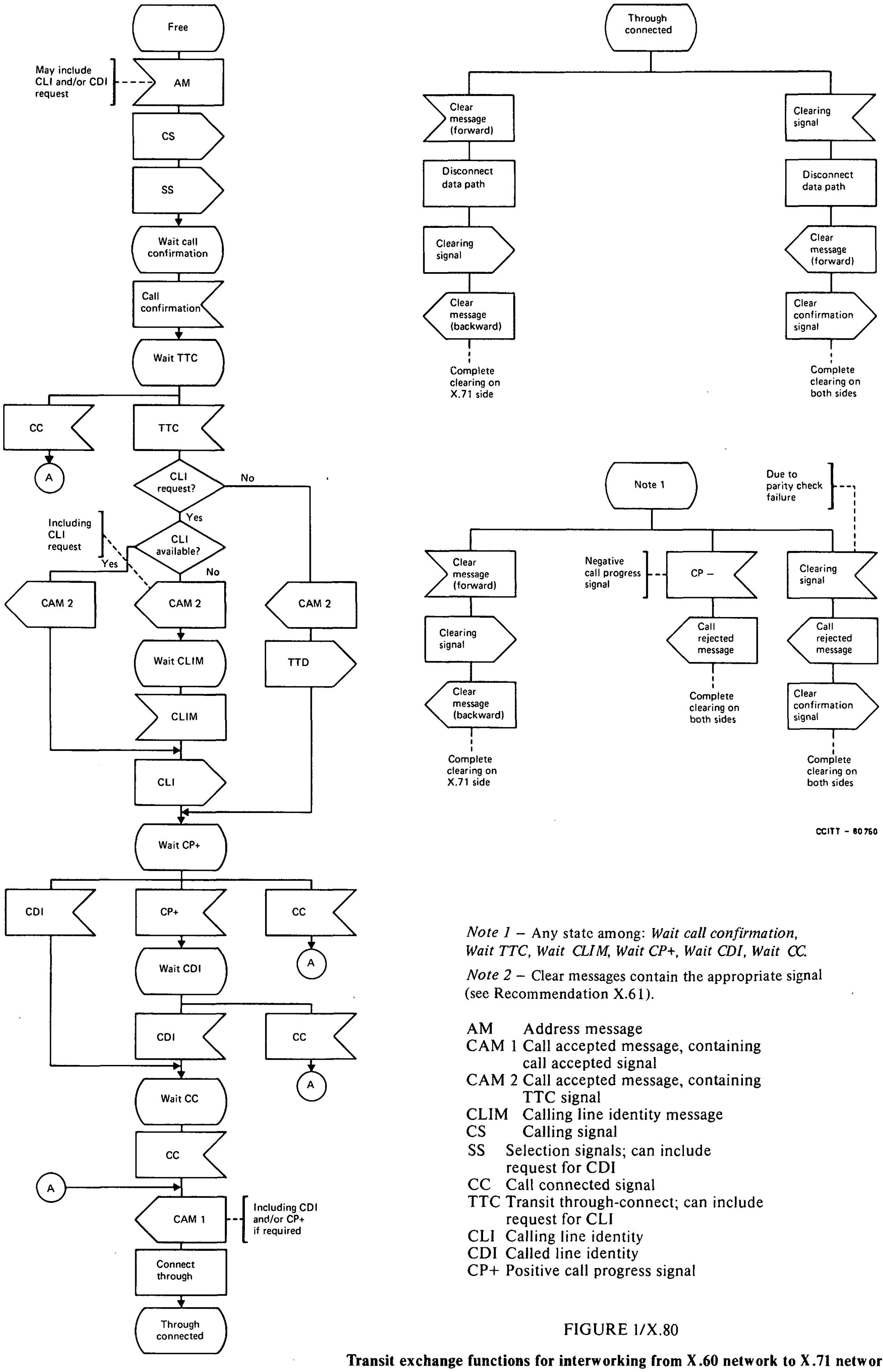
Originating network	X.60		Transit exchange interworking unit functions	X.71		Destination network
	→	←		→	←	
Seize free trunk Send trunk seized						
Send address message (includes a request for called line identity)	AM (CDIR)		Address message received			
			Seize free trunk			
			Send calling signal	CS		
			Send selection signals (including request for called line identity)	SS		Selection signals received
					C.CONF	Call confirmation signal sent
			TTC signal received		TTC	Transit through-connect (including request for calling line identity)
Call accepted message (TTC) received		CAM 2	Call accepted message (TTC) sent (including request for calling line identity)			
Calling line identity message sent	CLIM		Calling line identity message received			
			Calling line identity sent	CLI		Calling line identity received
						Called line identity sent
			Called line identity, call progress and call connected signals received		CDI/CP+	Call progress signal sent
					CC	Call connected signal sent
Call accepted message received		CAM 1	Call accepted message sent including called line identity and call progress signal			
Connect through			Connect through			Connect through

X.60 messages

AM	Address message
CAM 2	Call accepted (TTC)
CAM 1	Call accepted (CC)
CLIM	Calling line identity

X.71 signals

CS	Calling signal	TTC	Transit through-connected
SS	Selection	CLI	Calling line identification
C.CONF	Call confirmation	CDI	Called line identification
CC	Call connected	CP+	Positive call progress



Note 1 – Any state among: *Wait call confirmation, Wait TTC, Wait CLIM, Wait CP+, Wait CDI, Wait CC.*

Note 2 – Clear messages contain the appropriate signal (see Recommendation X.61).

AM	Address message
CAM 1	Call accepted message, containing call accepted signal
CAM 2	Call accepted message, containing TTC signal
CLIM	Calling line identity message
CS	Calling signal
SS	Selection signals; can include request for CDI
CC	Call connected signal
TTC	Transit through-connect; can include request for CLI
CLI	Calling line identity
CDI	Called line identity
CP+	Positive call progress signal

FIGURE 1/X.80

Transit exchange functions for interworking from X.60 network to X.71 network

The consequent *Call Accepted* Message(s) (CAM) which are transmitted on the X.60 side of the interworking unit function can contain:

a) The *call accepted* signal when a *CC* signal was received on the X.71 side. Note that this type of CAM (designated CAM 1) can also contain the *called line identity* and/or a *positive call progress* signal for calls which have initiated the additional protocols and are now ready to connect through. [See c) below.]

b) The *transit through connect* signal when a *TTC* signal was received on the X.71 side. The *TTC* signal may or may not *request the calling line identity.* The consequent *call accepted* message (designated CAM 2) can therefore contain:

   i) a request for the *calling line identity* if it was requested and it is not available;

   ii) no request if the calling line identity is already available as part of the *originating address* message;

   iii) no request if the *calling line identity* was not requested.

   In i) a *calling line identity* message is received from the X.60 side in response to the CAM 2. Then the *calling line identity* can be transmitted on the X.71 side.

   In ii) the *calling line identity* can be transmitted on the X.71 side.

   In iii) a Transit centres Through-Connected (TTD) signal is transmitted on the X.71 side.

c) A *positive call progress* signal and/or the *called line identity* when they were received on the X.71 side preceding the *call connected (CC)* signal. This information can be included in the CAM 1 sent on the X.60 side to complete the through-connection.

2.2 *Interworking from Recommendation X.71 to X.60*

Figure 2/X.80 shows the transit exchange interworking functions required to enable an X.71 to X.60 call to be connected.

The signals that can be transmitted on the X.71 side of the interworking unit function in response to a CAM 1 or CAM 2 message are as follows:

a) The *call connected (CC)* signal either directly or after the transmission of *called* and/or *calling line identification* and/or a *positive call progress* signal.

b) If the CAM 1 or CAM 2 contains a *request for calling line identity,* a *TTC* signal is transmitted with a request for *calling line identification* on the X.71 side. In response, the *calling line identity* is received from the X.71 side and a *calling line identity message* transmitted on the X.60 side.

   Note – If the *calling line identity* is sent as a result of a CAM 2 request, a subsequent CAM 1, which may contain the *called line identity,* must be sent on the X.60 side in order to complete the call set-up.

c) Where *calling line identity* is not required by a CAM 2, a *TTC* signal is sent on the X.71 side. A *TTD* signal will be received in response and this may be received before or after a CAM 1, which may contain the *called line identity,* has been received from the X.60 side in order to complete the call.

d) In b) and c) a *positive call progress* signal and/or the *called line identity* can be sent on the X.71 side before the *CC* signal.

e) Where a CAM 1 is received without a *request for calling line identity* but including the *called line identity* or a *positive call progress* signal, a *TTC* signal is sent on the X.71 side. When a *TTD* signal has been received in response, the *called line identity* and/or the *positive call progress* signal may be sent before the *CC* signal.

2.3 *Call clear-down*

A *call clear-down* signal can originate in either the X.60 or X.71 network. The interworking function must therefore be capable of detecting *clear* signals and messages, which can occur at any time during the call set-up or data phase of a call, and take the appropriate action as detailed below:

a) *A* clear request *signal received from the X.71 network*

   This will initiate disconnection of the call at the interworking function, the transmission of a *clear confirmation* signal to the X.71 network and a *clear* message to the X.60 network. At this point interworking ceases and each network clears down according to the normal X.60 or X.71 procedures.

b) *A* clear message *received from the X.60 network*

This will initiate disconnection of the call at the interworking function, the transmission of a *clear* message to the X.60 network and a *clear request* signal to the X.71 network. At this point interworking ceases and each network clears down according to the normal X.60 or X.71 procedures.

Note – The interworking function may optionally detect the internetwork data circuit *clear request* signal initiated by the user in the X.60 network. This will cause the disconnection of the call at the interworking function and initiate the same procedures as described above.

2.4 *Call failure conditions during call set-up*

Supervision of time-outs during call set-up takes place as stated in the signalling Recommendations X.71 and X.60 respectively.

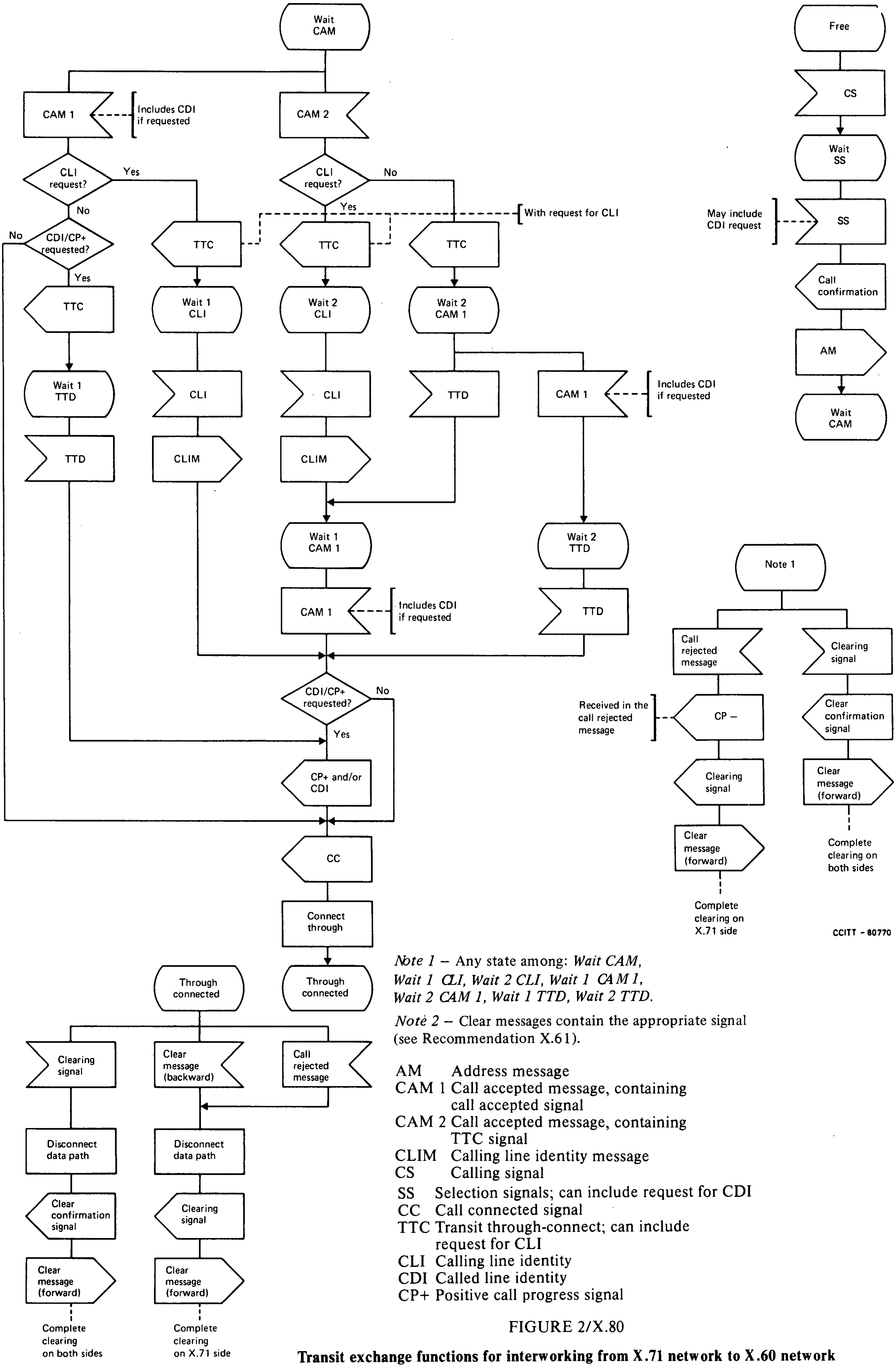
The expiry of such a supervisory-timer will lead to normal clearing as shown in Figures 1/X.80 and 2/X.80.

3 Interworking procedures between Recommendations X.70 and X.71

Since the proposals for Recommendations X.70 and X.71 are closely related, interworking between an X.70 and an X.71 network should be straightforward; however the interworking procedures required are for further study.

4 Interworking procedures between Recommendations X.60 and X.70

The required interworking procedures should be similar to the Recommendation X.60/71 case; however they are for further study.



Note 1 – Any state among: *Wait CAM, Wait 1 CLI, Wait 2 CLI, Wait 1 CAM 1, Wait 2 CAM 1, Wait 1 TTD, Wait 2 TTD.*

Note 2 – Clear messages contain the appropriate signal (see Recommendation X.61).

AM	Address message
CAM 1	Call accepted message, containing call accepted signal
CAM 2	Call accepted message, containing TTC signal
CLIM	Calling line identity message
CS	Calling signal
SS	Selection signals; can include request for CDI
CC	Call connected signal
TTC	Transit through-connect; can include request for CLI
CLI	Calling line identity
CDI	Called line identity
CP+	Positive call progress signal

FIGURE 2/X.80

Transit exchange functions for interworking from X.71 network to X.60 network

APPENDIX I

(to Recommendation X.80)

Interworking cases, Recommendations X.60/X.71

The following sequence charts illustrate examples of complex transit interworking situations.

Key to sequence charts

O – Originating network
T – Transit network
D – Destination network
X – Data path through-connect

Recommendation X.60

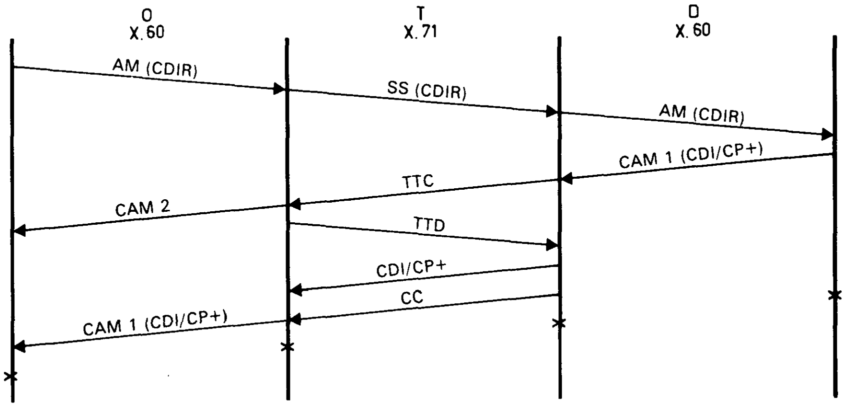
AM – Address message
AM (CLI) – Address message with calling line identity (CLI)
AM (CDIR) – Address message with request for called line identity (CDI)
AM (CLI + CDIR) – Address message with CLI and request for CDI
CAM 1 – Call accepted message contains call accepted signal; can contain CDI, CLI request and/or positive call progress signal
CAM 2 – Call accepted message; TTC signal can contain CLI request
CLIM – Calling line identity message

Recommendation X.71

CS – Calling signal
SS – Selection signals; can include request for called line identity
C.CONF – Call confirmation signal
CC – Call connected signal
TTC – Transit through-connect signal can include request for Calling Line Identity
TTD – Transit centres through-connected signal
CLI – Calling line identity
CDI – Called line identity
CP+ – Positive call progress signal

I.1 *Interworking, Recommendations X.60/X.71/X.60*

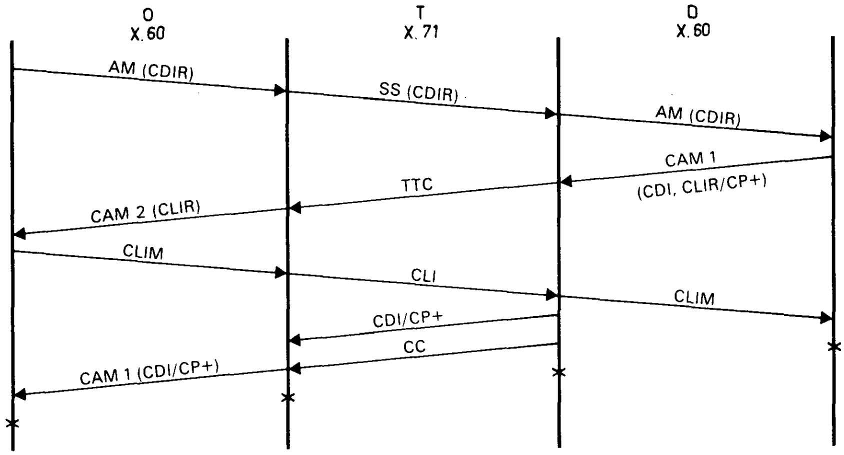
a) *Called line identity and/or positive call progress signal required, calling line identity not required*



CCITT-34820

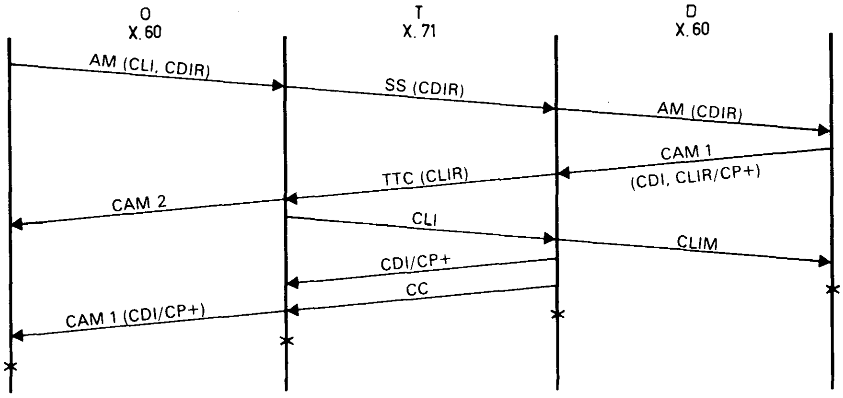
b) *Both called and calling line identity required and/or positive call progress signal*

First case



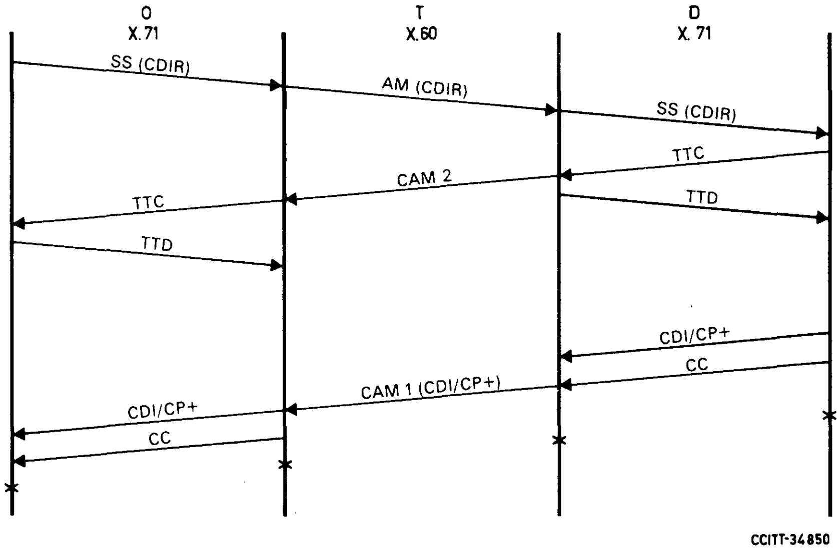
CCITT-34830

Second case

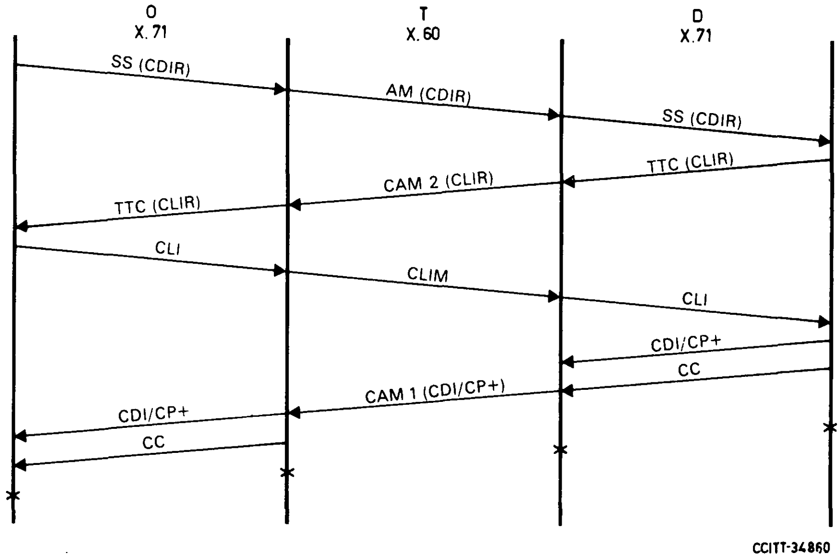


CCITT-34840

a) *Called line identity and/or positive call progress signal, calling identity not required*



b) *Calling line identity and called line identity and/or positive call progress signal required*



Recommendation X.81

INTERWORKING BETWEEN AN ISDN CIRCUIT-SWITCHED AND A CIRCUIT-SWITCHED PUBLIC DATA NETWORK (CSPDN)

(Melbourne, 1988)

The CCITT,

considering

(a) that I-Series Recommendations describe the integrated services digital network (ISDN);

(b) that interface characteristics, line multiplexing and inter-exchange signalling for use in CSPDNs are described by Recommendations such as X.26/X.27/X.50/X.51/Q.761 to Q.766/X.60/X.61/X.71/X.80;

(c) that Recommendation X.200 describes the reference model for Open System Interconnection;

(d) that Recommendation X.213 describes the Network Service Definition for Open Systems Interconnection for CCITT applications;

(e) that Recommendation X.300 defines the general principles for interworking between public networks, and between public networks and other networks for the provision of data transmission services;

(f) that Recommendation X.301 defines the general arrangements for call control within a subnetwork and between subnetworks for the provision of data transmission services;

(g) that Recommendation X.302 defines the general arrangements for internal network utilities within a subnetwork and between subnetworks for the provision of data transmission services;

(h) that Recommendation X.305 describes functionalities of subnetworks relating to the support of the OSI Network Service;

(i) that Recommendation X.10 describes categories of access to PSPDNs and ISDNs for the provision of data transmission services;

(j) that the I.230 Series Recommendations describe the bearer services supported by an ISDN;

(k) that Recommendation X.30 describes the support of X.21, X.21 *bis* and X.20 *bis* based DTEs on the ISDN;

(l) the need for arrangements when interworking between ISDN circuit switched and CSPDNs for the provision of data transmission services.

unanimously declares the view

that the scope of this Recommendation is intended to cover interworking between an ISDN using circuit-switched connection types and a CSPDN.

CONTENTS

0 Introduction

This Recommendation is one of a set of Recommendations produced to facilitate considerations of interworking between networks. It is based on Recommendation X.300, which defines the general principles for interworking between public networks, and between public networks and other networks for the provision of data transmission services. Recommendation X.300 indicates in particular how collections of physical equipment can be represented as "Subnetworks" for consideration in interworking situations.

This Recommendation describes the interworking arrangements between ISDNs (circuit switched bearer) and CSPDNs for the provision of data transmission services.

1 Scope and field of application

The purpose of this Recommendation is to describe the detailed arrangements for the interworking between CSPDNs and ISDNs (CS) for the provision of data transmission services. These arrangements are applicable only to the interworking involving transmission capabilities, and not to interworking involving communication capabilities as described in Recommendation X.300.

a) within the scope of interworking between an ISDN using circuit-switched connection types and a CSPDN the following cases of network interworking can be identified:

  i) where the terminals connected to the interworking networks use identical data transmission services and have identical high layer capabilities. The identity of data transmission services in both networks assumes that both terminals involved in a communication belong to the same user class of service;

  ii) where the terminals connected to the interworking networks use non-identical data transmission services but have identical high layer capabilities. In this case two terminals involved in a communication may belong to different user classes of service, e.g. user classes of service 4 and 30.

b) Not within the scope of this Recommendation is asynchronous mode of operation at the network-to-network interface (ISDN-to-CSPDN interface). In the case, that the ISDN supports the connection of asynchronous mode terminals via an appropriate terminal adaptor (TA) (see Recommendation X.30) and that interworking has to be provided to asynchronous terminals connected to a CSPDN, then the interworking arrangements as for synchronous mode operation will be used. Asynchronous to synchronous conversion may be provided within the TA for ISDN-connected terminals and at the terminal or at the CSPDN for CSPDN-connected terminals.

   Note – The typing of subnetworks in this Recommendation is based on the support for the OSI connection-mode Network Service and is therefore only valid in this context.

2 References

[1] Recommendation X.1
[2] Recommendation X.2
[3] Recommendation X.10
[4] Recommendation X.20
[5] Recommendation X.21
[6] Recommendation X.21 *bis*
[7] Recommendation X.25
[8] Recommendation X.27
[9] Recommendation X.30
[10] Recommendation X.50
[11] Recommendation X.51
[12] Recommendation X.60
[13] Recommendation X.61
[14] Recommendation X.71
[15] Recommendation X.300
[16] Recommendation X.321
[17] I.230 and I.250 Series Recommendations
[18] Recommendation G.703
[19] Recommendation G.708
[20] Recommendation G.811
[21] Recommendation Q.761-Q.766

3 Definitions

This Recommendation makes use of the folowing terms defined in the Recommendation as indicated:

Terms	Defined in Recommendation
Bearer service	I.230 Series
Data transmission service	X.10; X.300
OSI network layer service	X.213
User class of service	X.1
Supplementary services	I.250 Series; X.2

4 Abbreviations

DTE	Data terminal equipment
DCE	Data circuit-terminating equipment
SS No.7	Common channel signalling system number 7
CSPDN	Circuit switched public data network
ISDN	Integrated services digital network
IWF	Interworking functions
LAPB	Link access protocol, balanced
OSI-NLS	Open system interconnection – Network layer service
HDLC	High level data link control
TA	Terminal adaptor
TE	Terminal equipment
UC	User class of service

Abbreviations of SS No. 7 messages and X.71 signals

SS No. 7 messages:

ACM	Address complete
ANS	Answer
IAM	Initial address
INF	Information
INR	Information message
REL	Release
RLC	Release complete
RLSD	Released

X.71 signals:

CC	Call connected
CCF	Call confirmation
CDI	Called line identification
CLI	Calling line identification
CLEAR	Clearing signal
CLEAR C.	Clear confirmatiom
SEL	Selection signals
TTC	Transit through connect

Additional information contained in SS No. 7 messages and X.71 signals

CDI	Called line identification
CDIR	Called line identification request
CLI	Calling line identification
CLIR	Calling line identification request

5 General aspects

This Recommendation, in describing interworking arrangements between two subnetworks for the provision of data transmission services, adheres to the general principles of Recommendation X.300. The environments of these two subnetworks are described in the following sections.

5.1 *CSPDN*

The CSPDN provides circuit switched data transmission services as defined in Recommendations X.1 and X.2 for the provision of data transmission services, the CSPDN may be accessed by DTEs by the categories of access B as defined in Recommendation X.10. Other possibilities of access, which are not relevant to this Recommendation are described in Recommendation X.321.

5.2 *ISDN*

The ISDN provides circuit switched data transmission services/bearer services/supplementary services as defined in Recommendation X.1, the I.230 Series and I.250 Series. For the provision of data transmission services the ISDN may be accessed by TE2s by the categories of access S as defined in Recommendation X.10 and by TEs as defined in the I.230 Series. (circuit-mode 64 kbit/s unrestricted, unstructured). Other possibilities of access which are not relevant to this Recommendation are described in Recommendation X.321.

5.3 *Call control between the CSPDN and ISDN*

The general arrangements for call control between the CSPDN and ISDN circuit switched are as defined in Recommendation X.301. Network utililities used between the CSPDN and ISDN circuit switched are as defined in Recommendation X.302 (not visible for users).

6 Specification of interworking functions

The interworking functions specified hereafter have been grouped in accordance with their assignment to OSI-model-layers 1 to 3.

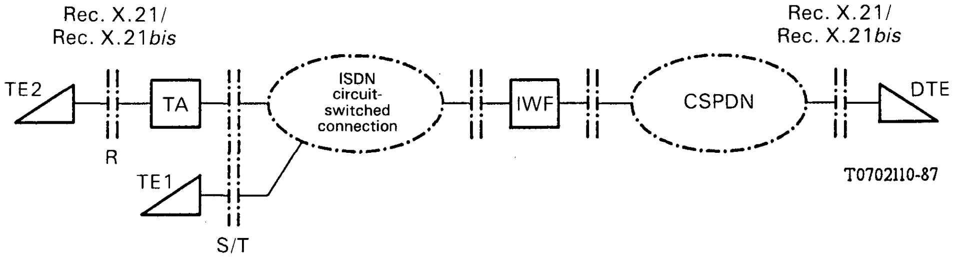
6.1 *Interworking functions for identical data transmission services*

Reference configuration of the interworking between a circuit switched ISDN and a CSPDN using identical bearer services.

As can be seen from Figure 1/X.81, the following terminal to terminal interworking cases may arise for which network interworking capabilities are necessary for user classes 3-7/categories S and B of Recommendation X.10.

ISDN	CSPDN
(Categories S1-S5)	(Categories B1-B5)
TE2 + TA X.21	DTE X.21
TE2 + TA X.21	DTE X.21 *bis*
TE2 + TA X.21 *bis*	DTE X.21
TE2 + TA X.21 *bis*	DTE X.21 *bis*
TE1 (see Note in Figure 1/X.81)	

TE2 and DTE involved in an end-to-end communication must be of the same user class. In this case a CSPDN Gateway must also support the same user class of service.



TE1 ISDN Terminal Equipment Type 1 (Note)
TE2 ISDN Terminal Equipment Type 2
TE2 and DTE Belonging to the same user class (in user classes of service 3-7)

Note – In cases where a TE1-type terminal is used, the bit stream at the S/T-interface has to comply to the X.30 frame structure.

FIGURE 1/X.81

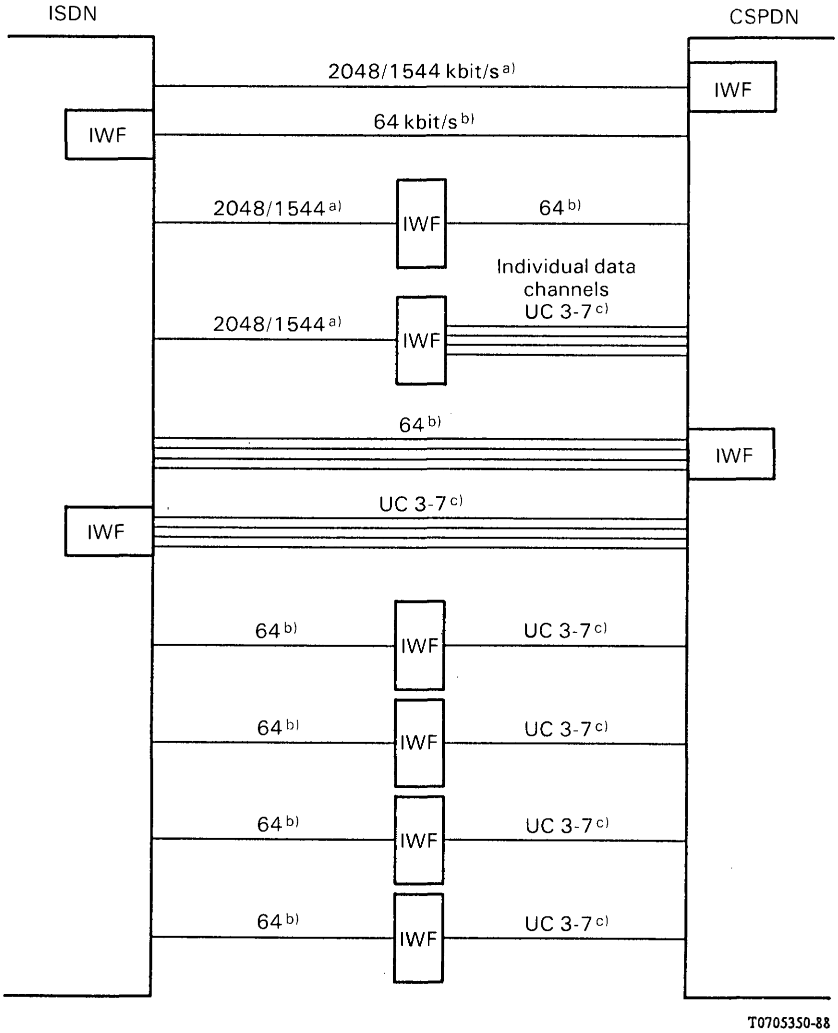
Reference configuration

6.1.1 *Physical link characteristics and interworking functions assigned to layer 1*

6.1.1.1 *Location of the interworking functions*

The interworking functions may be provided by the ISDN, or by the CSPDN or separately between the interworking networks.

The interworking links may be provided as multiplexed lines offering a multiplicity of channels at the conversion point or as individual channels. Consequently, the conversion facility has to provide layer 1 multiplexing/demultiplexing functions in the case where multiplexed lines are connected to it. Multiplexing may take place only on the ISDN side or only on the CSPDN side or on both sides of the location, where the interworking functions are provided. The question as to whether the interconnection lines will be multiplexed or not will depend on the location of the IWF, i.e. on whether the conversion facilities are installed at the data switching exchange or at the ISDN exchange. In either case the location of the IWF should be considered from an economic viewpoint regardless of whether the conversion facilities are designed as separate hardware and software-based modules or as integrated parts of either an ISDN exchange or a data switching exchange (see also Figure 2/X.81).



IWF Interworking Functions
UC User class

a) See § 6.1.1.1 a).
b) See § 6.1.1.1 b).
c) See § 6.1.1.1 c).

FIGURE 2/X.81

Alternative IWF location and interfaces

Depending on the location of the IWF (see Figure 2/X.81), the electrical characteristics on one or both sides of the IWF shall comply with the current Recommendations on interface characteristics as follows:

a) Recommendation G.703/G.708 when offering the 2048/1544 kbit/s interface for 32/24 × 64 kbit/s channels on the ISDN side;

b) Recommendation G.703/X.27 when offering the 64 kbit/s interface for data channels multiplexed in accordance with the X.50 or the X.51 or the X.51 multiplex scheme on the CSPDN side;

c) any interface recommendation applicable to individual data channels offered on the CSPDN side at speeds specified for user classes of service 3-7 and their associated 6 + 2 and 8 + 2 envelope bit rates.

6.1.1.2 *Timing requirements*

Since interworking takes place between two networks, the ISDN and the CSPDN, phase and/or clock adjustment must take place. The timing requirements of the IWF are the same, as specified in Recommendation G.811 for the interconnection of digital link.

6.1.1.3 *Bit rate adaption*

Since switching in the ISDN is provided only for 64 kbit/s channels and no standards are currently available for switching of subchannels, the bit rates of user classes of service 3-7 have to be adapted to 64 kbit/s.

The bit rate adaption mechanism shall be in accordance with Recommendation X.30, §§ 2.1.1 and 2.2.1. On the CSPDN side of the IWF there will be the need to allocate the information which is contained within 40 bit frames received from the ISDN on a 64 kbit/s bearer into either 6 + 2 or 8 + 2 envelopes.

The same applies to an incoming envelope stream received from a data network. The bit positions within the envelopes have to be allocated to the relevant positions in an outgoing 40 bit frame to be sent to the ISDN.

Concerning the layer 1 relay functions of an IWF for rate adaption, the following two configurations, presented by Figure 3/X.81, have been identified.

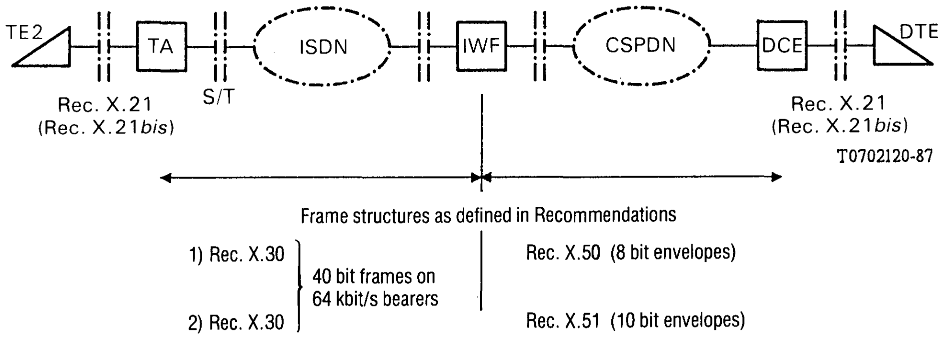


FIGURE 3/X.81

Rate adaption configurations

The bit rate adaption for both directions is restricted to the transfer of information and status. Framing bits, housekeeping bits and any other bits will not be transferred by the IWF. Mapping of housekeeping functions is a subject of further study.

(1) Interworking with an X.50-structured CSPDN

(8 bit envelope structure)

For interworking with a CSPDN utilizing the 8 bit envelope structure as identified in Recommendation X.50, § 1.1 a). This implies the suppression of each fourth status bit on transition from CSPDN to ISDN and replication of each third status bit on transition from ISDN to CSPDN is necessary.

(2) Interworking with an X.51-structured CSPDN

(10 bit envelope structure)

For interworking with a CSPDN utilising the 10 bit envelope structure as identified in Recommendation X.51, the data and status bits are mutually assembled for retransmission in the coordination defined in X.51 respectively X.30. An example is shown in Figure 4/X.81.

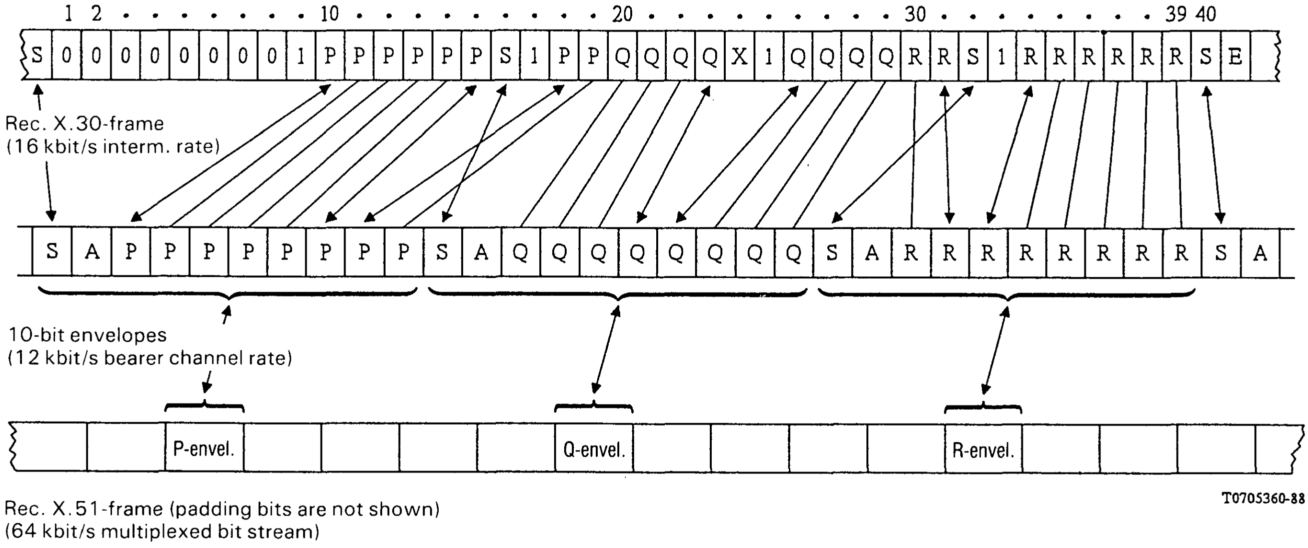


FIGURE 4/X.81

Exemple of Recs X.30/X.51 rate adoption
(shown is the 9.6 kbit/s case)

6.1.2 *Support of the OSI-network layer service (OSI-NLS)*

The support of the OSI-NLS provided by the interworking networks is shown in Figure 5/X.81 which is in accordance with Recommendation X.321.

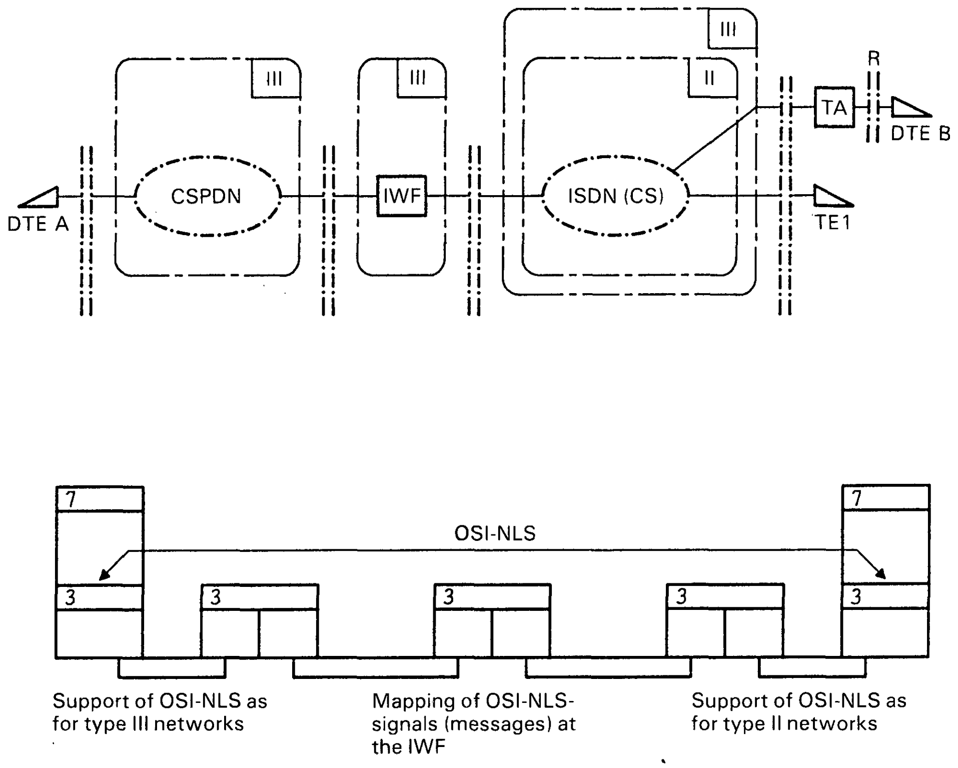
Mapping of OSI-NLS – signals by the IWF for the call establishment phase is for further study.

Note – Since the CSPDN is not capable of giving full support to the OSI-NLS during the call establishement phase, the IWF have to react on requests, which may arise at the ISDN-side and which cannot be handled successfully within the CSPDN.

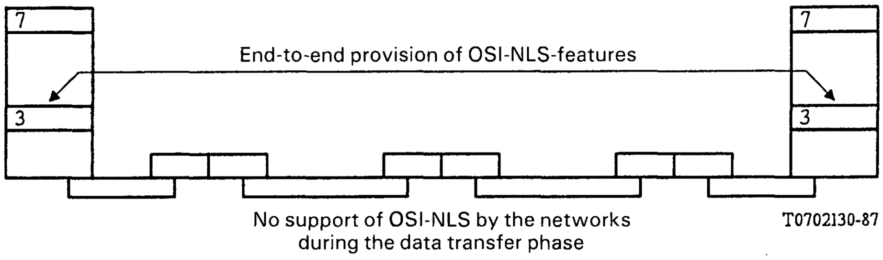
6.1.3 *Signalling conversion (protocol mapping)*

While on the ISDN side of the IWF common channel signalling with the SS No. 7 ISDN USER PART can always be assumed, signalling on the data network side can be either channel associated utilizing the X.71 signalling scheme or SS No. 7 based on X.60 and X. 61 or Q.761 to Q.766.

The logical representation of the mapping functions in the case of signalling conversion from SS No. 7 to X.71 is shown in Figure 6/X.81 and § 6.1.3.1 to 6.1.3.3. A typical configuration of an end-to-end data connection including several signalling conversion points is shown in Figure 7/X.81.



Call establishment phase

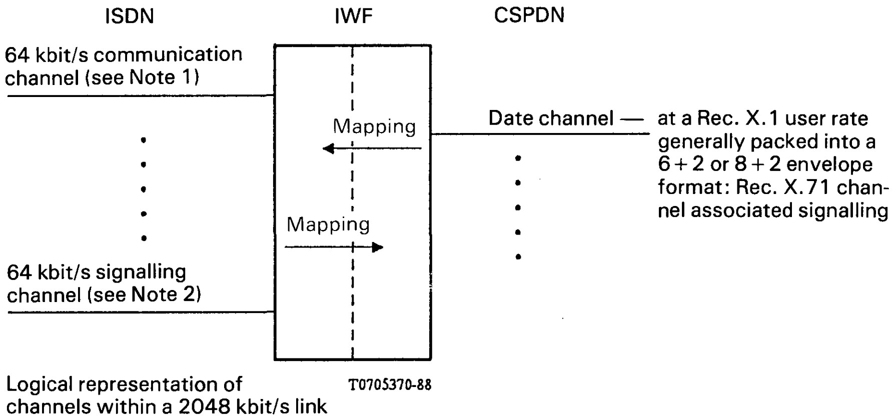


Data transfer phase

Note – Categorization of type II and III networks see Rec. X.300.

FIGURE 5/X.81

Support of the OSI-Network Layer Service



Note 1 – 64 kbit/s communication channel bearing a data channel at a Rec. X.1 user rate (non-transparent ISDN connection).

Note 2 – Common channel signalling SS No. 7, ISDN USER PART.

FIGURE 6/X.81

Signalling conversion

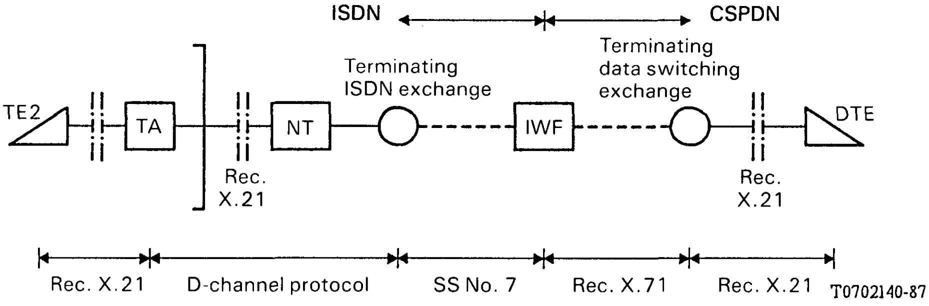
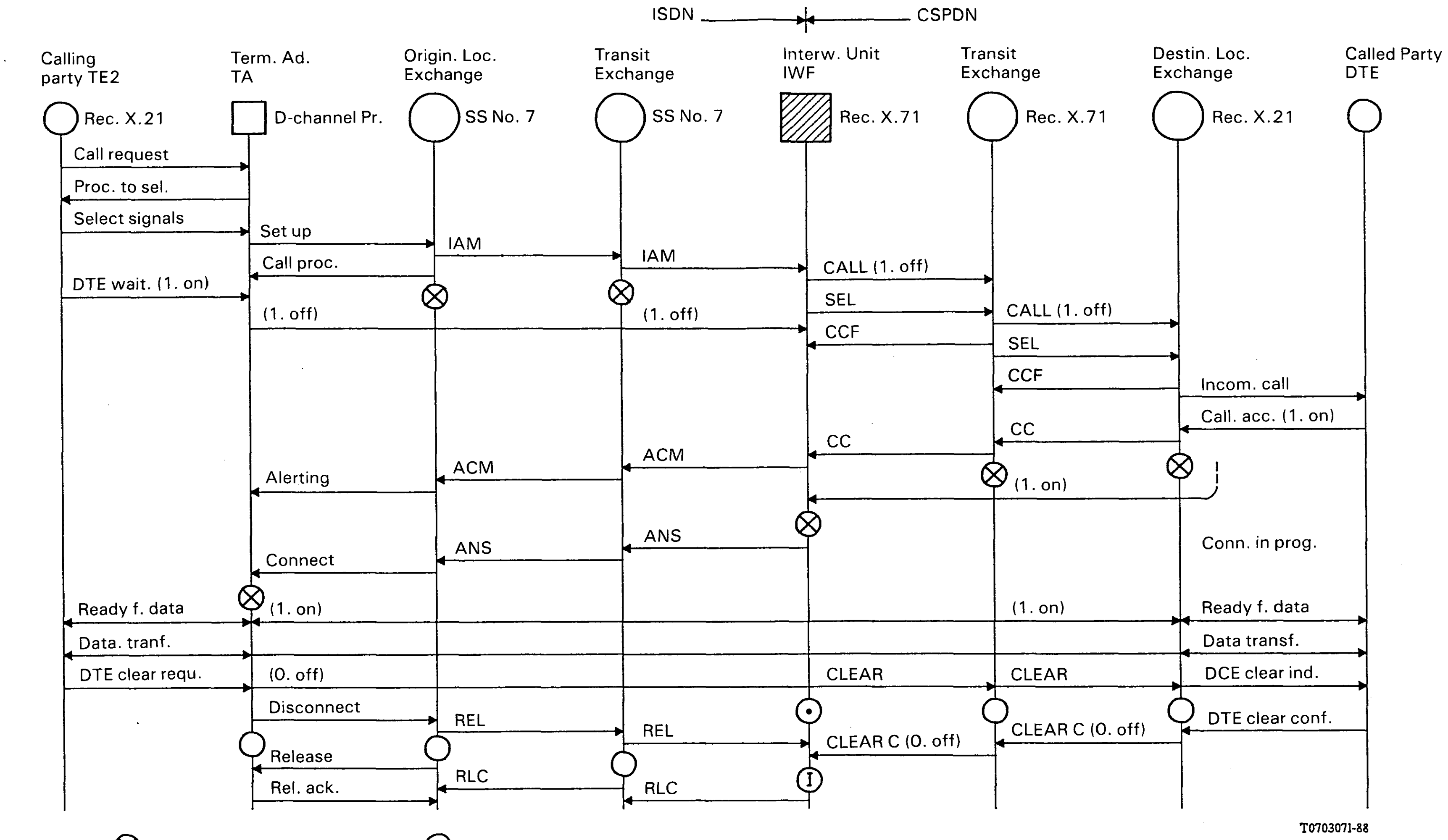


FIGURE 7/X.81

Signalling conversion points

6.1.3.1 *Signalling conversion for a basic call from ISDN to CSPDN*

Figure 8/X.81 illustrates the signalling conversion procedure in the IWF referring to the simple case of a basic call which originates in an ISDN and terminates in an X.71 network, and which does not invoke any additional facilities. The call is assumed to be successful and the clear down is initiated by the user of the ISDN. Besides of the signalling conversion functions in the IWF Figure 8/X.81 also shows the relevant signalling events of the D-channel protocol and of Recommendation X.21.



⊗ Switch path completed

○ Switch path removed

⊙ Outgoing side removal

Ⓘ Incoming side removed

Note – Abbreviations of SS No. 7 messages and Rec. X.71 signals see the end of the Recommendation.

FIGURE 8/X.81

Signalling conversion of a basic call from ISDN to CSPDN

The call set-up sequence begins in the IWF at the moment when the SS No.7 initial address message (IAM)[1] is received from the adjacent ISDN exchange. At this point of time the 64 kbit/s communication channel is already through-connected within the ISDN. In the following the X.71 call signal and selection signals (SEL) are sent to the adjacent CSPDN exchange, which confirms the call with the call confirmation signal (CCF). The call is forwarded link by link to the destination exchange of the CSPDN which calls according to Recommendation X.21 the DTE. After the DTE has accepted the call the destination exchange of the CSPDN sends back the X.71 call connected signal (CC). The CC signal is transmitted link by link towards the IWF. Parallel with the link by link transmission of the CC signal the data channel is switched through within the CSPDN. The reception of the CC signal and of the terminating throughconnection signal (1, ON) which is sent across the throughconnected data channel by the called DTE defines the instant of connect through in the IWF.

The IWF sends instead of the CC signal the SS No. 7 address complete message (ACM). After the data channel through-connection within the IWF the SS No. 7 answer message (ANS) is sent to the adjacent ISDN exchange. Both messages are transmitted link by link towards the originating exchange of the ISDN. The originating exchange of the ISDN sends after reception of ANS the CONNECT message (according to Recommendation Q.931) to the terminal adaptor (TA) of the calling TE2, and as a consequence throughconnection of the data channel in the TA can be performed according to Recommendation X.30. Now ready for data alignment followed by data transfer can take place between TE2 and DTE according to Recommendation X.21.

Clearing is initiated for example at the TE2 by DTE clear request (0, OFF), which is transmitted transparently via the data channel to the DTE. Accompanying this inslot signal the TA sends the DISCONNECT message (according to Recommendation Q.931) to the originating ISDN exchange. From there the No. 7 RELEASE message is transmitted link by link towards the IWF and as a consequence the 64 kbit/s communication channel is cleared down. The clearing down of the data channel within the CSPDN is initiated by the reception of the clearing signal (0, OFF).

6.1.3.2 *Signalling conversion for a basic call from CSPDN to ISDN*

Figure 9/X.81 illustrates the signalling conversion procedure in the IWF referring to the simple case of a basic call which originates in a CSPDN and terminates in an ISDN, and which does not invoke any additional facilities. The call is assumed to be successful and the clear down is initiated by the customer of the CSPDN. After throughconnection of the 64 kbit/s communication channel in the ISDN exchanges the SS No. 7 ANS message is received in the IWF. As a consequence the IWF sends the X.71 CC signal to the adjacent CSPDN exchange. The CC signal is transmitted link by link towards the originating exchange of the CSPDN. By this way the data channel is switched through within the CSPDN.

After the data channel finally is switched through in the originating exchange of the CSPDN and in the terminal adaptor ready for data alignment between DTE and TE2 can be performed.

6.1.3.3 *Signalling conversion for a complex call set-up between ISDN and CSPDN*

If the call involves additional facilities in comparison with the basic call additional procedures are necessary in Recommendation Q.761 to 765 and in Recommendation X.71.

Figures 10/X.81 and 11/X.81 illustrate examples involving these additional procedures for call set-up requiring both calling and called line identification.

6.1.3.3.1 *Call set-up from ISDN to CSPDN* (Figure 10/X.81)

In this case the called line identification (CDI) is requested by means of an additional information (CDIR) within the X.71 selection signals (SEL). The request of calling line identification (CLIR) is contained in the X.71 Transit through connect signal (TTC) and in the SS No. 7 information message (INR).

The called line identification (CDI) is transmitted via the CSPDN as a separate X.71 signal and via the ISDN as an additional information of the address complete message (ACM).

The calling line identification (CLI) is transmitted via the ISDN by means of the information message (INF) and via the CSPDN as a separate X.71 signal.

If the calling line identification (CLI) is already contained in the SS No. 7 IAM independently if it is requested or not, the SS No. 7 messages INR and INF may be omitted.

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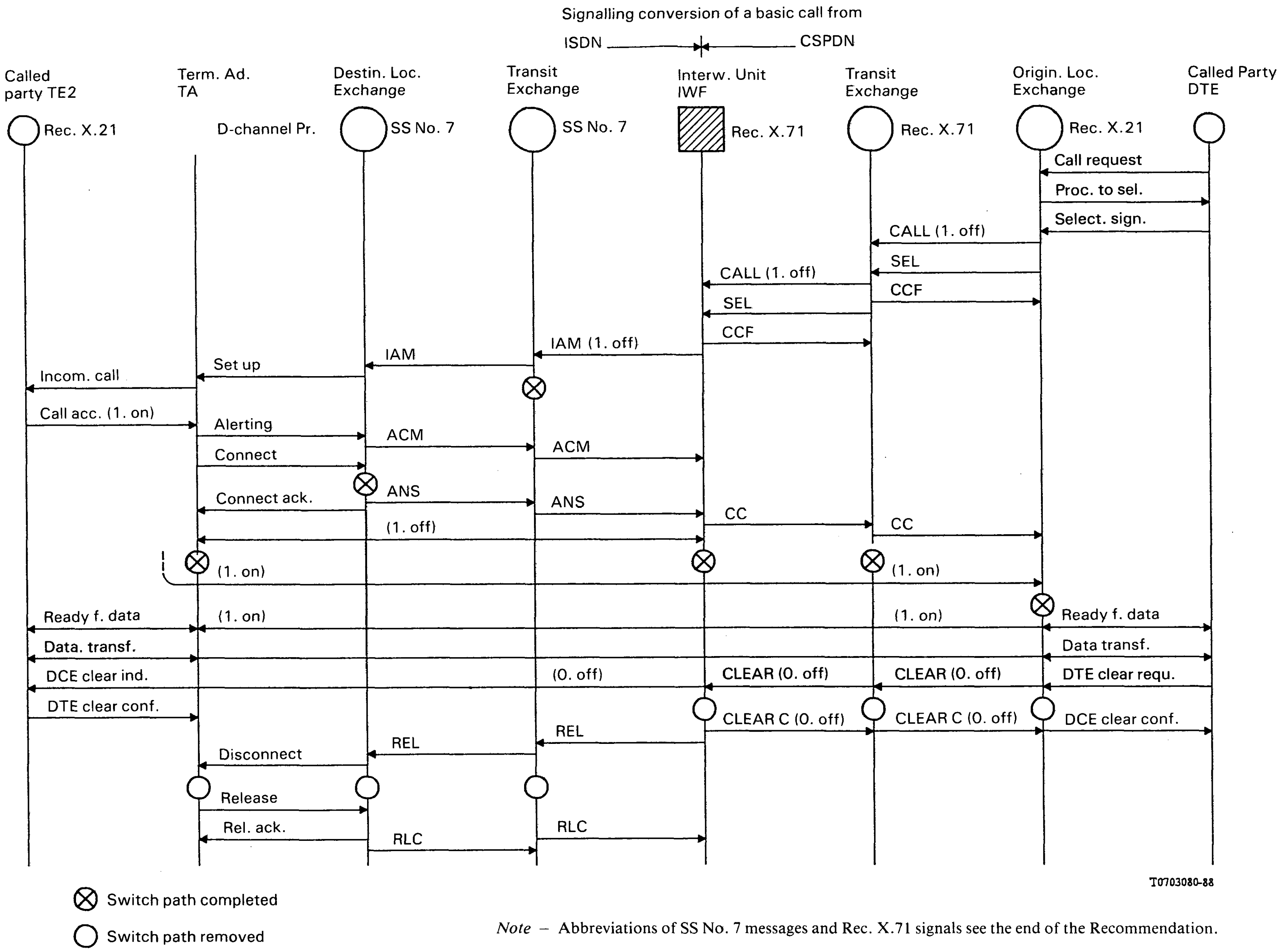
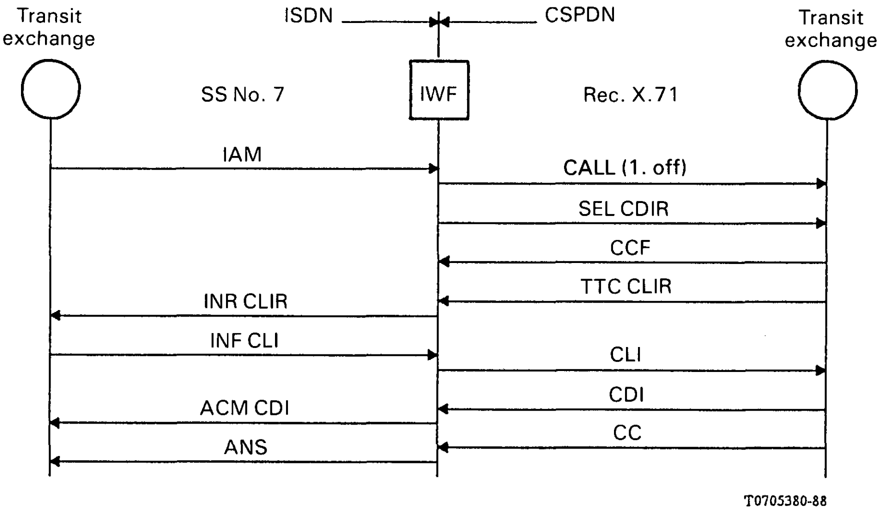
[1] Containing all information for call set-up.



Note – Abbreviations of SS No. 7 messages and Rec. X.71 signals see the end of the Recommendation.

FIGURE 9/X.81

Signalling conversion of a basic call from CSPDN to ISDN



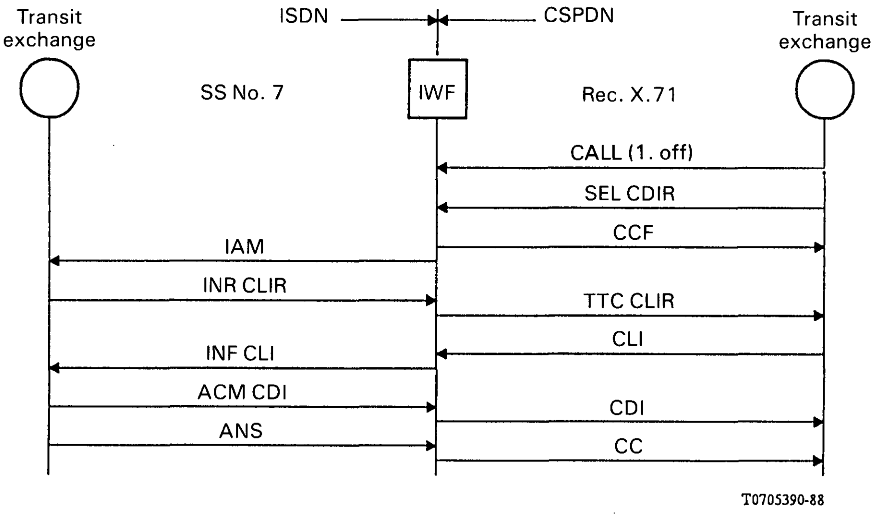
Note – Abbreviations of SS No. 7 messages and Rec. X.71 sign. see the end of the Recommendation.

FIGURE 10/X.81

Call set up with calling/called line identification from ISDN to CSPDN

6.1.3.3.2 *Call set-up from CSPDN to ISDN* (Figure 11/X.81)

Calling and called line identification and their requests are carried within the same SS No. 7 message and X.71 signals as described under § 6.1.3.3.1.



Note – Abbreviations of SS No. 7 messages and Rec. X.71 sign. see the end of the Recommendation.

FIGURE 11/X.81

Call set up with calling/called line identification from CSPDN to ISDN

6.1.3.4 *Ready for data alignment*

For user classes 3-7 a ready for data alignment procedure is executed after end-to-end connection has been established. The purpose of the ready for data alignment procedure is to indicate to the communicating terminals the exact point in time of the entry into the data transfer phase. The ready for data alignment signal is defined by the reception of a 1/ON-signal at the user/network interfaces at both ends. The 1/ON-signal is transmitted:

- in the ISDN by setting the data bits of the X.30-frames to one and the associated status bits to ON,
- in the CSPDN by setting the data bits of the envelopes to one and the associated status bits to ON.

For the ready for data alignment signal the IWF are transparent.

6.1.4 *Protocol mapping for supplementary services*

The IWF shall map the protocols which are necessary to support supplementary services. Bearing in mind, that for each network, the ISDN as well as for the CSPDN an individual set of supplementary services is defined, three different situations of interworking will arise:

a) A specific supplementary service is supported by both networks equivalently. In this case a one to one mapping by the IWF is possible.

b) A specific supplementary service is supported by the ISDN but not supported equivalently by the CSPDN. In this case;
   - either the request for this service coming from the ISDN has to be refused by the IWF, or
   - the request for this service coming from the ISDN may be mapped to the CSPDN but with reduced functionality.

c) A specific supplementary service is supported by the CSPDN but not supported equivalently by the ISDN. In this case;
   - either the request for this service coming from the CSPDN has to be refused by the IWF, or
   - the request for this service coming from the CSPDN may be mapped to the ISDN but with reduced functionality.

A table containing the supplementary services supported by CSPDN as specified in Recommendation X.2 is given in Table 1/X.81. The supplementary services for data transmission supported by ISDN circuit switched are specified in the I.250 Series Recommendations.

6.1.5 *Mapping of service signals and causes*

The IWF shall map signals and causes used in each of the interworking networks. Bearing in mind, that the list of causes used in the ISDN and list of service signals used in the CSPDN are not fully identical, a complete one to one mapping of all signals is not possible.

A table containing the service signals of the CSPDN and the causes of the ISDN and their mapping is for further study.

6.2 *Interworking functions for non-identical data transmission services*

Reference configuration of the interworking between a circuit-switched ISDN and a CSPDN using different data transmission services:

In Figure 12/X.81 end-to-end communication between terminals to different user classes of service is assumed. As an example, TE1 may belong to user class of service 30 at a data signalling rate of 64 kbit/s and the DTE may belong to user class of service 4 at a data signalling rate of 2400 bit/s category B2. The interworking functions in Figure 12/X.81 are partly different from those in Figure 1/X.81.

6.2.1 *Physical link characteristics and interworking functions assigned to layer 1*

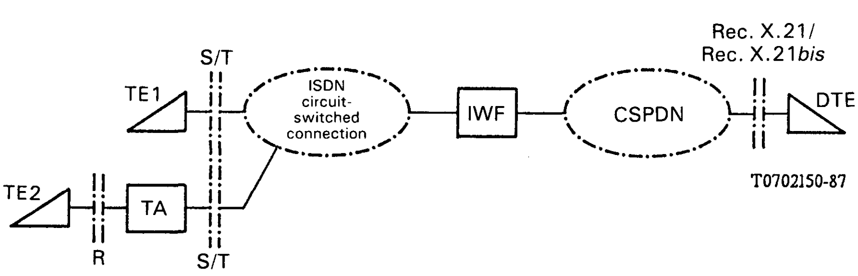
6.2.1.1 *Location of the interworking functions*

Location and features of the interworking functions are the same as described in § 6.1.1.1 for the cases where the data transmission services are identical.

TABLE 1/X.81

Supplementary services supported by CSPDN

1	Optional user facilities assigned for an agreed contractual period
1.1	Direct call
1.2	Closed user group
1.3	Closed user group with outgoing access
1.4	Closed user group with incoming access
1.5	Incoming calls barred within a closed user group
1.6	Outgoing calls barred within a closed user group
1.7	Calling line identification
1.8	Called line identification
1.9	Bilateral closed user group
1.10	Bilateral closed user group with outgoing access
1.11	Incoming calls barred
1.12	Reverse charging acceptance
1.13	Connect when free
1.14	Waiting allowed
1.15	Redirection of calls
1.16	On-line facility parameter registration/cancellation
1.17	DTE inactive registration/cancellation
1.18	Data and time indication
1.19	Hunt group
2	Optional user facilities requested by the DTE on a per-call basis
2.1	Direct call
2.2	Abbreviated address calling
2.3	Multi-address calling
2.4	Reverse charging
2.5	RPOA selection
2.6	Charging information
2.7	Called line identification



Note – In cases where a TE1-type terminal is used, the bit stream at the S/T-interface complies with the Rec. X.30 frame structure.

FIGURE 12/X.81

6.2.1.2 *Timing requirements*

Phase and clock adjustment regulations are not applicable in the case of non-identical data transmission rates.

6.2.1.3 *Bit rate conversion*

For bit rate conversion between the two non-identical data transmission services, a flow control is needed, since the effective mean data transfer rate must be decreased to the rate of the slower terminal.

The flow control method required must be derived from the protocols applied in both terminals.

6.2.1.3.1 *Protocol compatible terminals*

Assuming a HDLC-based terminal to terminal protocol rate conversion can be provided by the IWF by flag insertion/extraction and use of intermediate buffer capacity:

In an existing connection, both the ISDN-section and the CSPDN-section are bit-transparent, but differing data signalling rates are used. It is assumed that both terminals are protocol-compatible above the physical layer of the reference model. Complying examples may be teletex-terminals, but also multi-mode terminals, as long as they are in the same mode, e.g. in the teletex mode. A further assumption is that the terminal-to-terminal protocol includes a flow control facility, based on HDLC. The data transfer is then decreased to the capability of the slower terminal by injection of interframe time fill flags into the ISDN section. The only functions related to rate conversion to be effected by the IWF are to extract interframe time fill from the ISDN/CSPDN directed data stream, to inject the necessary interframe time fill into the CSPDN/ISDN directed data stream, and intermediate buffering.

The IWF-functions needed during the data transfer phase can be restricted then to:

1) provision of the physical layer adaptation functions by use of suitable interface modules to access both networks, and
2) support of the rate conversion provided by the terminals' flow control facility by provision of an intermediate buffer capacity and flag extraction/insertion.

The buffer capacity must be related to the maximum frame length and window size. In this respect, all link channel states and also exception conditions have to be considered, including their description in X.25.

Figures 13/X.81 and 14/X.81 below are examples for frame sequences on both interfaces of the IWF.

6.2.1.3.2 *Terminals with different protocols*

This case requires further study.

6.2.2 *Support of the OSI-network layer service (OSI-NLS)*

Is as described in § 6.1.2.

6.2.3 *Signalling conversion (protocol mapping)*

Is for further study, considering also § 6.1.3.

6.2.4 *Protocol mapping for supplementary services*

Is as described in § 6.1.4.

6.2.5 *Mapping of service signals and causes*

Is as described in § 6.1.5.

7 **Operation and maintenance**

Is for further study.

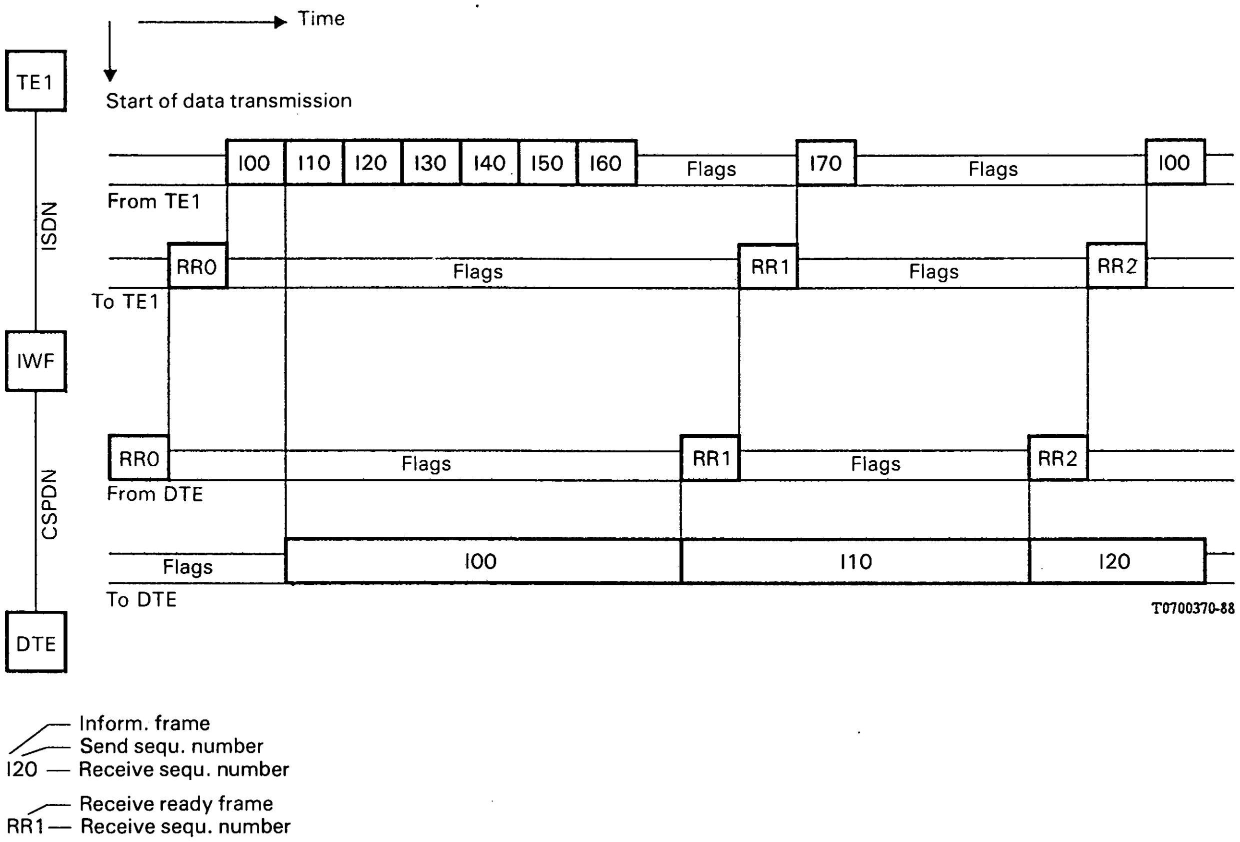


FIGURE 13/X.81

Transmission ISDN to CSPDN example

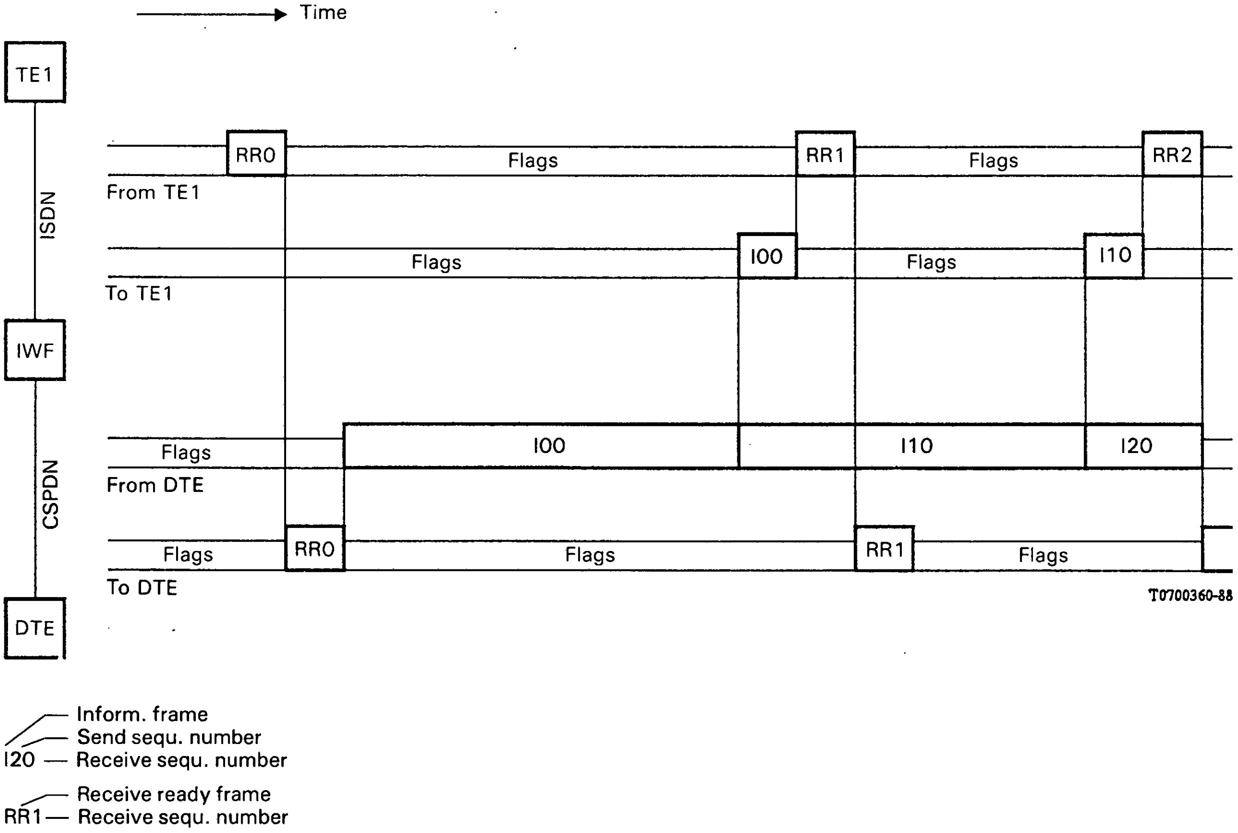


FIGURE 14/X.81

Transmission CSPDN to ISDN example

Recommendation X.82

DETAILED ARRANGEMENTS FOR INTERWORKING BETWEEN CSPDNs AND PSPDNs BASED ON RECOMMENDATION T.70

(Melbourne, 1988)

The CCITT,

considering

(a) that administrations are currently operating CSPDNs and PSPDNs;

(b) that it is essential to make it possible to allow interworking between DTEs connected to the different types of PDNs;

(c) that CCITT services, e.g. telematic services, can be carried by PSPDN or CSPDN or both, as defined in existing Recommendations T.70 and X.300;

(d) that Recommendation T.70 defines the network-independent basic transport service for telematic services;

(e) that Recommendation X.300 defines general principles for interworking between public networks, and between public networks and other networks for the provision of data transmission services;

(f) that Recommendation X.322 defines general arrangements for interworking between PSPDNs and CSPDNs for the provision of data transmission services;

(g) that Recommendation X.75 defines procedures for PSPDN/PSPDN interworking and that Recommendation X.71 defines procedures for CSPDN/CSPDN interworking;

(h) that Recommendation X.25 defines the user interface to PSPDNs, and that Recommendations X.21/X.21 *bis* define the user interface to CSPDNs;

unanimously declares the view

that detailed arrangements for interworking between CSPDNs and PSPDNs on the basis of Recommendation T.70 for telematic services shall be in accordance with the procedures specified in this Recommendation.

CONTENTS

0 Introduction

This Recommendation is one of a set of Recommendations produced to facilitate considerations of interworking between networks. It is based on Recommendation X.300, which defines the general principles for interworking between public networks, and between public networks and other networks for the provision of data transmission services. Recommendation X.300 indicates in particular how collections of physical equipment can be represented as "Subnetworks" for consideration in interworking situations.

This Recommendation describes the interworking arrangements between circuit switched public data networks (CSPDN) and packet switched public data networks (PSPDN) based on CCITT Recommendation T.70.

1 Scope and field of application

The purpose of this Recommendation is to describe the detailed arrangements for the interworking between CSPDNs and PSPDNs based on Recommendation T.70. These arrangements are applicable only to the interworking involving telematic services, and not to the interworking involving communication capabilities described in Recommendation X.300.

The mapping of protocol data units taken from different protocols is limited to the capabilities of each of the protocols. The functions that are required to provide the full OSI connection-mode network service (CONS) and their relation to this Recommendation can be found in the Appendix I.

2 References

Recommendation T.70	Network-independent basic transport service for telematic services,
Recommendation X.1	International user classes of service in public data networks and integrated services digital networks (ISDNs),
Recommendation X.2	International data transmission services and optional user facilities in public data networks and ISDNs,
Recommendation X.21	Interface between data terminal equipment (DTE) and data circuit-terminating equipment (DCE) for synchronous operation on public data networks,
Recommendation X.21 *bis*	Use on public data networks of data terminal equipment (DTE) which is designed for interfacing to synchronous V-series modems,
Recommendation 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,
Recommendation X.71	Decentralized terminal and transit control signalling system on international circuits between synchronous data networks,
Recommendation X.75	Packet switched signalling system between public networks providing data transmission services,
Recommendation X.121	International numbering plan for public data networks,
Recommendation X.300	General principles for interworking between public networks, and between public networks and other networks for the provision of data transmission services.

3 Definitions

No specific definitions need to be taken into account.

4 Abbreviations

CLI	Calling Line Identification
CONS	Connection-mode Network Service
COT	Class of Traffic
CP	Call Progress
CSPDN	Circuit Switched Public Data Network
CUG	Closed User Group
DM	Disconnected Mode
DNIC	Data Network Identification Code
DTE	Data Terminal Equipment
EOS	End of Selection
IWF	Interworking Function
NS	Network Service
OSI	Open Systems Interconnection
PDN	Public Data Networks
PSPDN	Packet Switched Public Data Network
QOS	Quality of Service
SABM	Set Asynchronous Balanced Mode
TID	Terminal Identifier
TTC	Transit Through Connect
TTD	Transit Centres Through Connected
UA	Unnumbered Acknowledgement
UC	User Class

5 General aspects

This Recommendation, in describing interworking arrangements between two subnetworks, adheres to the general principles of Recommendation X.300. The environments of these two subnetworks are described in the following sections.

5.1 *Circuit switched public data network*

The CSPDN provides switched data transmission services as defined in Recommendation X.1 and X.2 for the provision of data transmission services. The transmission capability of the CSPDN may also be used for the provision of telematic services defined in the T-series Recommendations.

Note – For additional application rules for telematic services see § 3.3 of CCITT Recommendation T.70.

5.2 *Packet switched public data network*

The PSPDN provides packet switched data transmission services as defined in Recommendation X.1 and X.2 for the provision of data transmission services. The transmission capability of the PSPDN may also be used for the provision of telematic services defined in the T-series Recommendations.

Note – For additional application rules for telematic services see § 3.1 of CCITT Recommendation T.70.

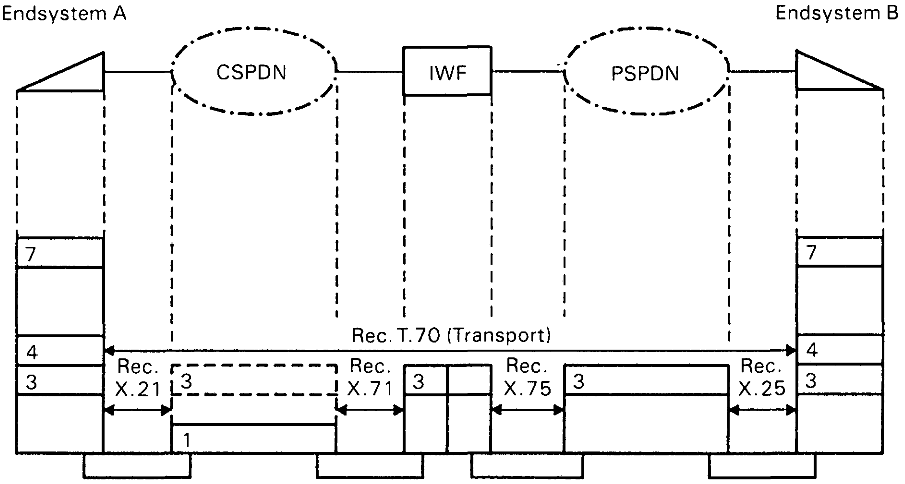


FIGURE 1/X.82

Interworking between CSPDN and PSPDN based on Recommendation T.70

6 Specification of interworking functions

This section describes the detailed mapping for the interworking between CSPDNs and PSPDNs based on Recommendation T.70.

6.1 *Connection establishment phase*

6.1.1 *Connection establishment initiated on the CSPDN side*

Figures 2/X.82 to 6/X.82 show the signalling when a terminal connected to the CSPDN initiates a call towards a terminal in the PSPDN. The order of events in time might be different for unrelated signals received from the circuit and packet switched side, depending on the transmission delays and response times in both subnetworks.

While Figure 2/X.82 represents the successful connection establishment several possible unsuccessful call attempts are shown in figures 3/X.82 to 6/X.82.

6.1.1.1 *Selection signals*

Any non allocated selection character shall result in the call being cleared with a call progress signal.

1) First class-of-traffic character (1st COT)

   Alternative routing allowed/not allowed is not transferred to the PSPDN.

   The implementation of transit/terminal indicator Bit 1 is as follows:

   If Bit 1 = 0 the DNIC is included in the selection signals;

   If Bit 1 = 1 the DNIC is not included in the selection signals.

2) First user class character (1st UC)

   1st UC is only used to indicate that 2nd UC follows.

3) Second user class character (2nd UC)

   When the 2nd UC indicate TELETEX by “1001” in the bits b1 to b4, this shall be mapped to “00000010” in the first octet of the call user data field of the call request packet of Recommendation X.75. In case there is no 2 UC or in cases of coding other than “1001” the call attempt may either be rejected or continued.

   Note – This reflects the status of Recommendation T.70. However, the transparency of the call user data field needs further consideration. Other possible mapping requirements e.g. to the class of traffic facility of Recommendation X.75 are for further study.

4) Second class-of-traffic character (2nd COT)

   Bit b1 is used to indicate national/international traffic, and is not to be transferred in the Recommendation X.75 signalling. The bits b2 and b3 indicate whether the called line identification of the called terminal is requested (b2) and/or the closed user group characters of the calling terminal are following.

5) Third class-of-traffic character (3rd COT)

   When receiving a 3rd COT the call attempt may either be rejected or continued. The possible future use of the 3rd COT is for further study.

6) Closed user group characters(CUG)

   The closed user group characters are transferred as CUG utility in Recommendation X.75. If the CUG sequence contains less than 4 characters, excluding DNIC, zeros are inserted in the X.75 CUG utility. If a CUG DNIC is not included in the X.71 signalling, a dummy DNIC of "0000" is inserted in the CUG utility.

7) Network or service identification signal

   The IWF T70 shall return a DNIC.

8) Called DTE address

   The selection signals received from the CSPDN are transferred into the X.75 address field. If the called DTE address does not contain a DNIC, a DNIC shall be added by the IWF T.70.

6.1.1.2 *Transit through connect signal (TTC)*

The Calling line identification shall always be required (b2 = 1).

6.1.1.3 *Calling line identification signal (CLI)*

The calling line identification signals are used in the calling DTE address field in the X.75 Call Request Packet. If the calling line ID does not contain a CNIC, a DNIC may be added by the IWF T.70.

6.1.1.4 *Call progress signal (CP)*

The call progress signal "terminal called" is sent to inform the calling terminal that the call is being established.

The IWF may repeat this CP in order to avoid that the call attempt is cleared by the calling DTE before the call attempt in the PSPDN is either connected or timed out and cleared.

6.1.1.5 *Call request packet*

1) Called and calling DTE address

   The called DTE address is obtained from the DNIC and the called terminal number in the X.71 selection signals. The end of selection character is not transferred.

   The calling DTE address is obtained from the X.71 calling line identification signals. If no DNIC is included, the DNIC of the calling network will be inserted.

2) Network utilities

   The throughput class indication shall be signalled and the value indicated shall be mapped to the bit rate on the circuit switched side. A lower throughput class in the call connected packet may be accepted.

   The default value of 2 for window size, and 128 for packet size, shall apply for all calls and needs therefore not be signalled. The use of other values is subject to bilateral agreement.

3) User facilities

   No user facilities shall be signalled.

4) Call user data

   See § 6.1.1.1, item 3).

6.1.1.6 *SABM command*

The calling terminal uses address (B) in commands and (A) in responses, the IWF T.70 uses address (A) in commands and (B) in responses according to Recommendation T.70.

After receiving the originating through connection signal the IWF T.70 shall wait for the link set up by the DTE connected to the CSPDN. After a time-out condition the IWF T.70 may attempt to set up the link level itself.

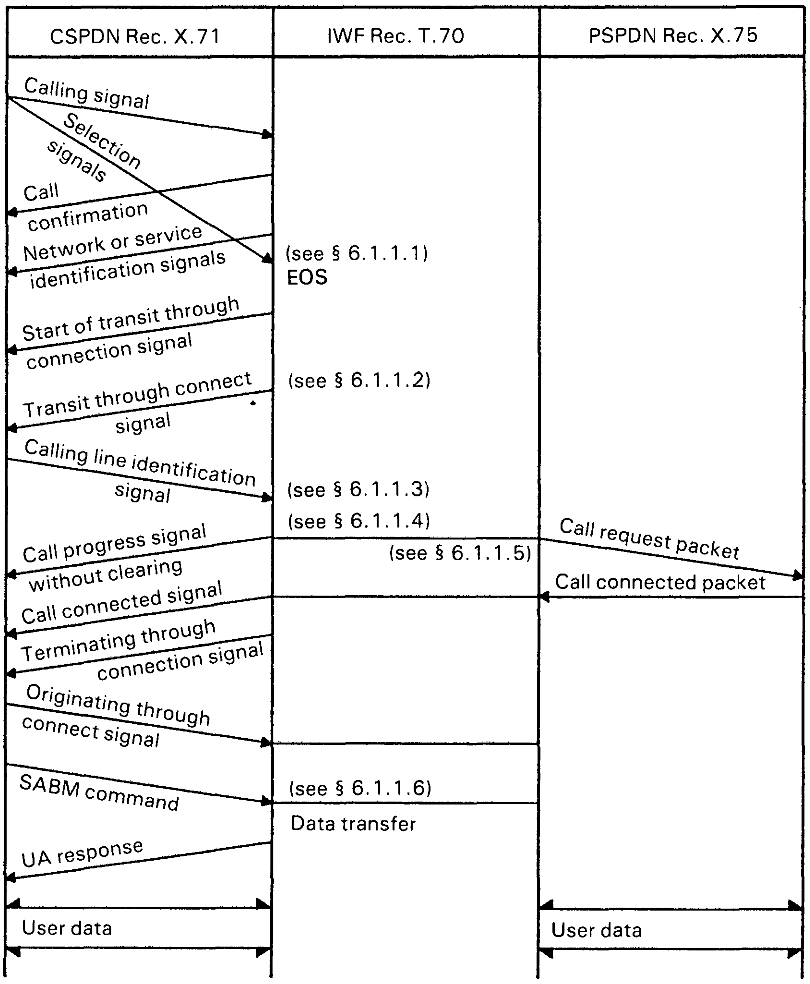


FIGURE 2/X.82

Successful connection establishment CSPDN ⟶ PSPDN

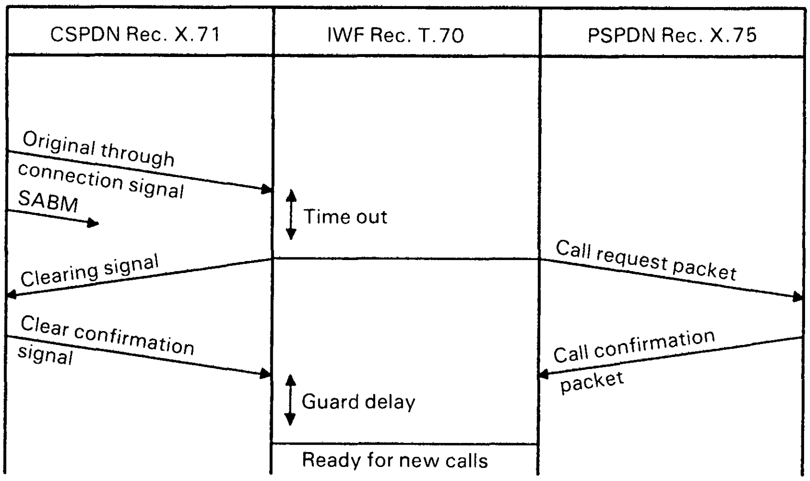


FIGURE 3/X.82

Unsuccessful connection establishment CSPDN ⟶ PSPDN
(IWF Rec. T.70 does not receive an SABM)

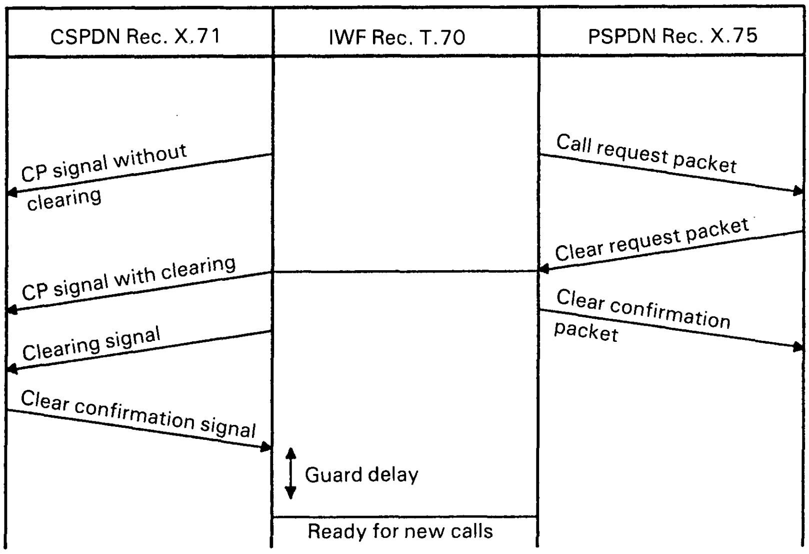


FIGURE 4/X.82

Unsuccessful connection establishment CSPDN ⟶ PSPDN
(IWF Rec. T.70 receives a clear request packet)

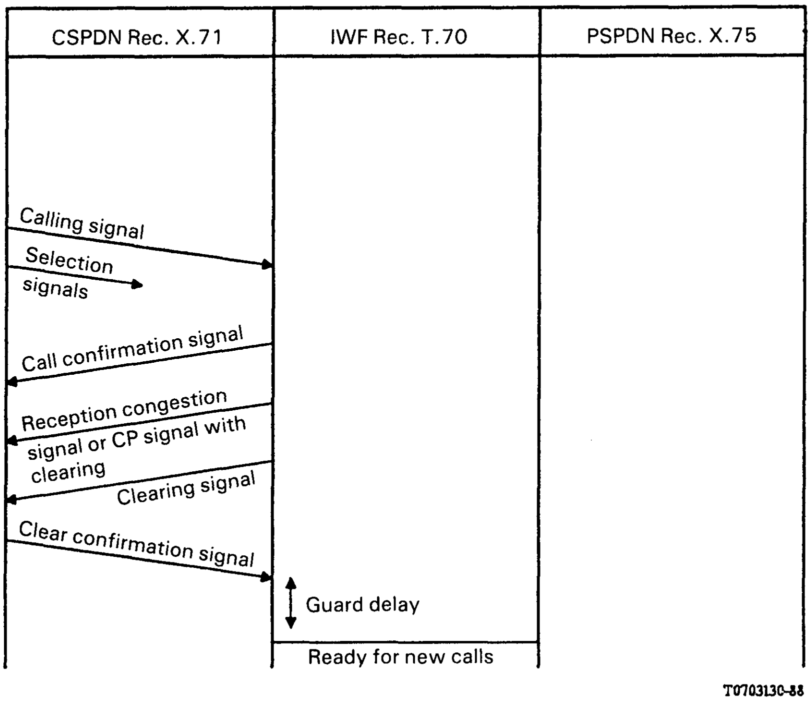


FIGURE 5/X.82

Unsuccessful connection establishment CSPDN ⟶ PSPDN
(IWF Rec. T.70 busy or no logical channels available)

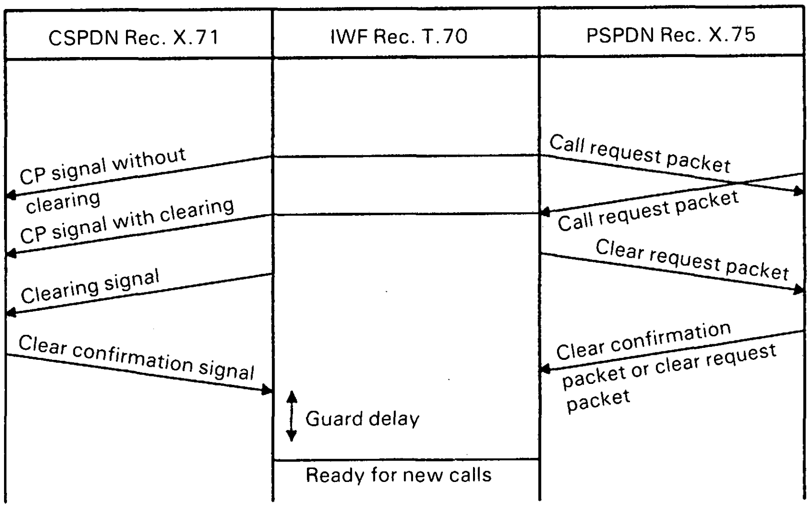


FIGURE 6/X.82

Unsuccessful connection establishment CSPDN ⟶ PSPDN
(Collision of call request packets on the PSPDN-side)

6.1.2 *Connection establishment initiated on the PSPDN side*

Figures 7/X.82 to 11/X.82 show the signalling when a terminal connected to the PSPDN initiates a call towards a terminal in the CSPDN. The order of events in time might be different for unrelated signals received from the circuit and packet switched side, depending on the transmission delays and response times in both subnetworks.

While Figure 7/X.82 represents the successful connection establishment several possible unsuccessful call attempts are shown in Figures 8/X.82 to 11/X.82.

6.1.2.1 *Call request packet*

The called DTE address always includes a DNIC which may be transferred to the CSPDN.

The calling DTE address is stored and will be transferred later in the calling line identification signal, if requested by the called DTE.

1) Network utilities

   The transit network identification code shall not be transferred in the X.71 signalling, since DNICs cannot be signalled in X.71 in the forward direction.

   The call identifier shall not be signalled in X.71.

   If the received throughput class indication is greater than the data signalling on the circuit switched side, the value of the data signalling rate shall be returned. In all other cases the received value is returned.

   The default values of 2 for window size, and 128 for packet size, shall apply for all calls. The use of other values is subject to bilateral agreement.

   According to Recommendation T.70 terminals shall not use the fast select facility. the receipt of fast select indication shall result in sending a clear request packet.

   The closed user group and closed user group with outgoing access are mapped to the corresponding signals in Recommendation X.71.

   The transit delay indication cannot be signalled in Recommendation X.71.

2) User facilities

For further study.

3) Call user data

The telematic protocol identifier as defined in Recommendation T.70 is mapped to the 2nd UC in Recommendation X.71 (all remaining information will get lost). When receiving other codes in the first octet of the call user data field of the call request packet of Recommendation X.75 the call attempt may either be rejected or continued.

Note – This reflects the status of Recommendation T.70. However, the transparency of the call user data field needs further consideration. Other possible mapping requirements e.g. to the class of traffic facility of Recommendation X.75 are for further study.

6.1.2.2 *Selection signals*

1) First class-of-traffic character (1st COT)

"Alternative routing allowed" is signalled. The X.75 called DTE address may be passed unchanged to the CSPDN or the DNIC may be stripped.

Note – The X.75 addresses of the called and calling DTE always include a DNIC.

Bit 1 is set accordingly. Bit 1 = 0: DNIC included; Bit 1 = 1: DNIC not included. "UC follows" is indicated.

2) First user class character (1st UC)

1st UC is only used to indicate that the second class-of-traffic (2nd COT) character and second user class (2nd UC) character follows.

3) Second user class character (2nd UC)

When the first octet of the call user data field of the call request packet of Recommendation X.75 indicates Teletex by "00000010", this shall be mapped to "1001" in the bits b1 to b4 of the 2nd UC in Recommendation X.71. In cases of coding other than "00000010" the call attempt may either be rejected or continued.

Note – This reflects the status of Recommendation T.70. However, the transparency of the call user data field needs further consideration. Other possible mapping requirements e.g. to the class of traffic facility of Recommendation X.75 are for further study.

4) Second class-of-traffic (2nd COT)

Bits 1, 2 and 4 of the 2nd COT are always set to "0". Bit 3 is set according to the X.75 utilities received.

5) Third class-of-traffic character (3rd COT)

The third class-of-traffic character is not transmitted by the IWF T.70.

6) Closed user group (CUG)

See § 6.1.1.1, item 6).

7) Transit through connect signal (TTC)

The calling line identification is only signalled if it has been requested in the TTC signal. The transit centres through-connected (TTD) signal is sent in other cases.

6.1.2.3 *SABM command*

The calling DTE uses address (B) in commands and (A) in responses, the called DTE uses address (A) in commands and (B) in responses, according to CCITT Recommendation T.70.

6.1.2.4 *Call connected packet*

If any network or service identification signals are received that indicate transit networks, these are transferred in the Transit network identification network utility. The Throughput class is set according to the procedure described under "Network utilities" (see § 6.1.2.1, item 1)).

The time relation between "UA Response Frame" received and "Call Connected Packet" sent is for further study.

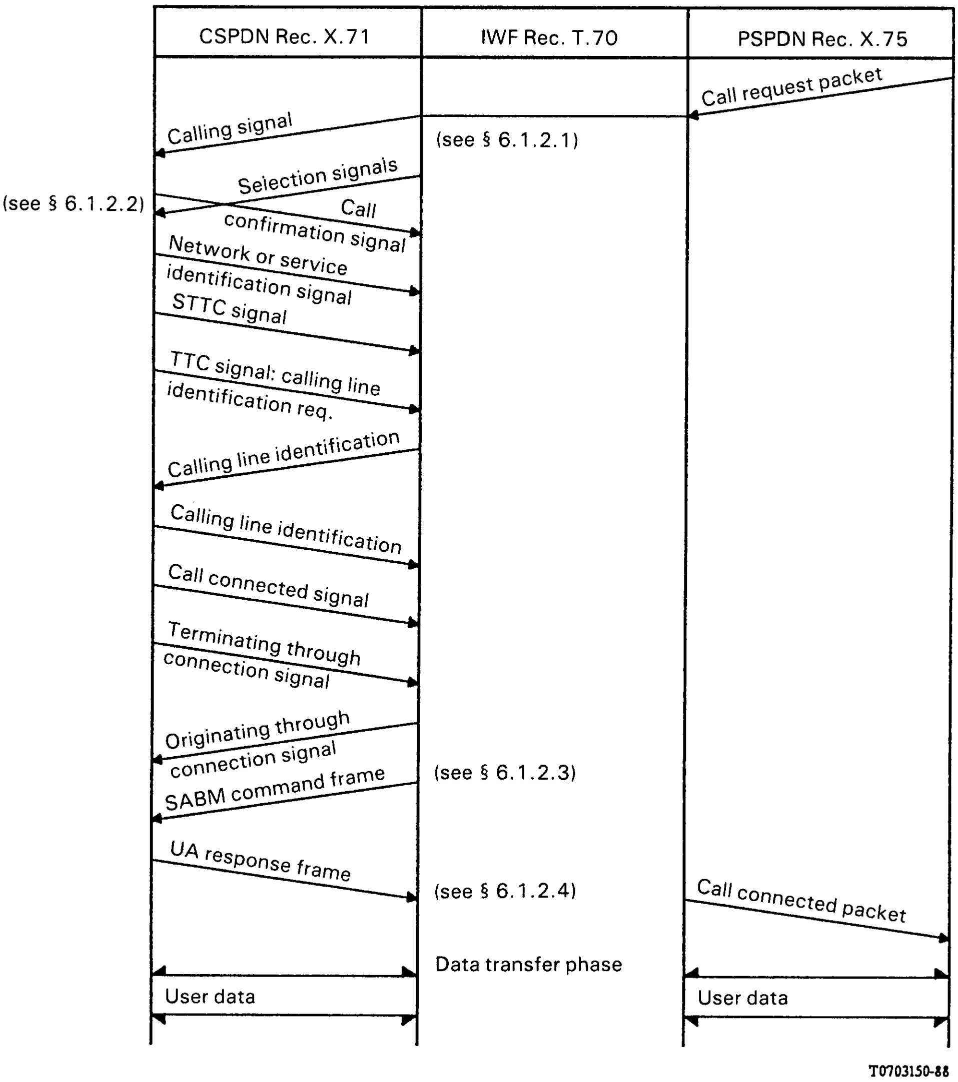


FIGURE 7/X.82

Successful connection establishment PSPDN ⟶ CSPDN

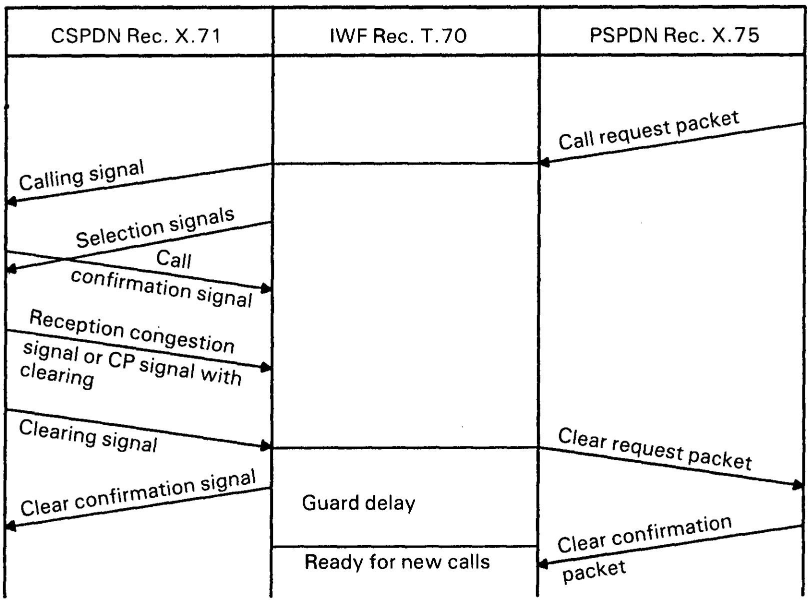


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FIGURE 8/X.82

Unsuccessful connection establishment PSPDN —► CSPDN (CSPDN or endsystem not available)

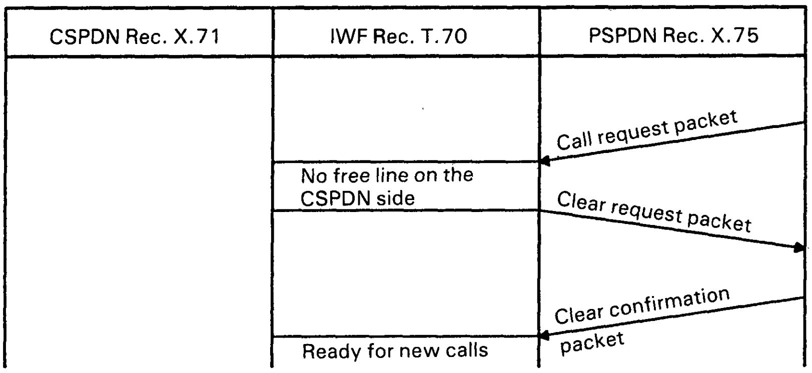

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FIGURE 9/X.82

Unsuccessful connection establishment PSPDN —► CSPDN (no free line on the CSPDN-side available)

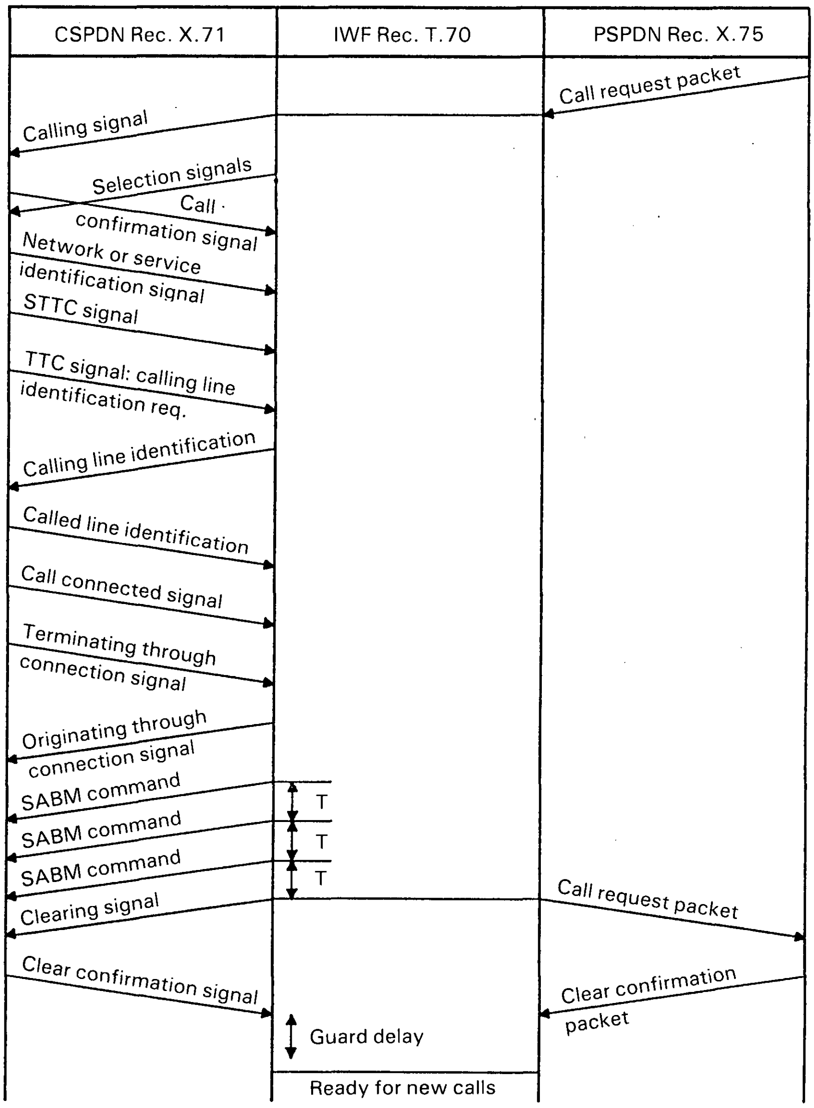


FIGURE 10/X.82

Unsuccessful connection establishment PSPDN ⟶ CSPDN (SABM command not answered)

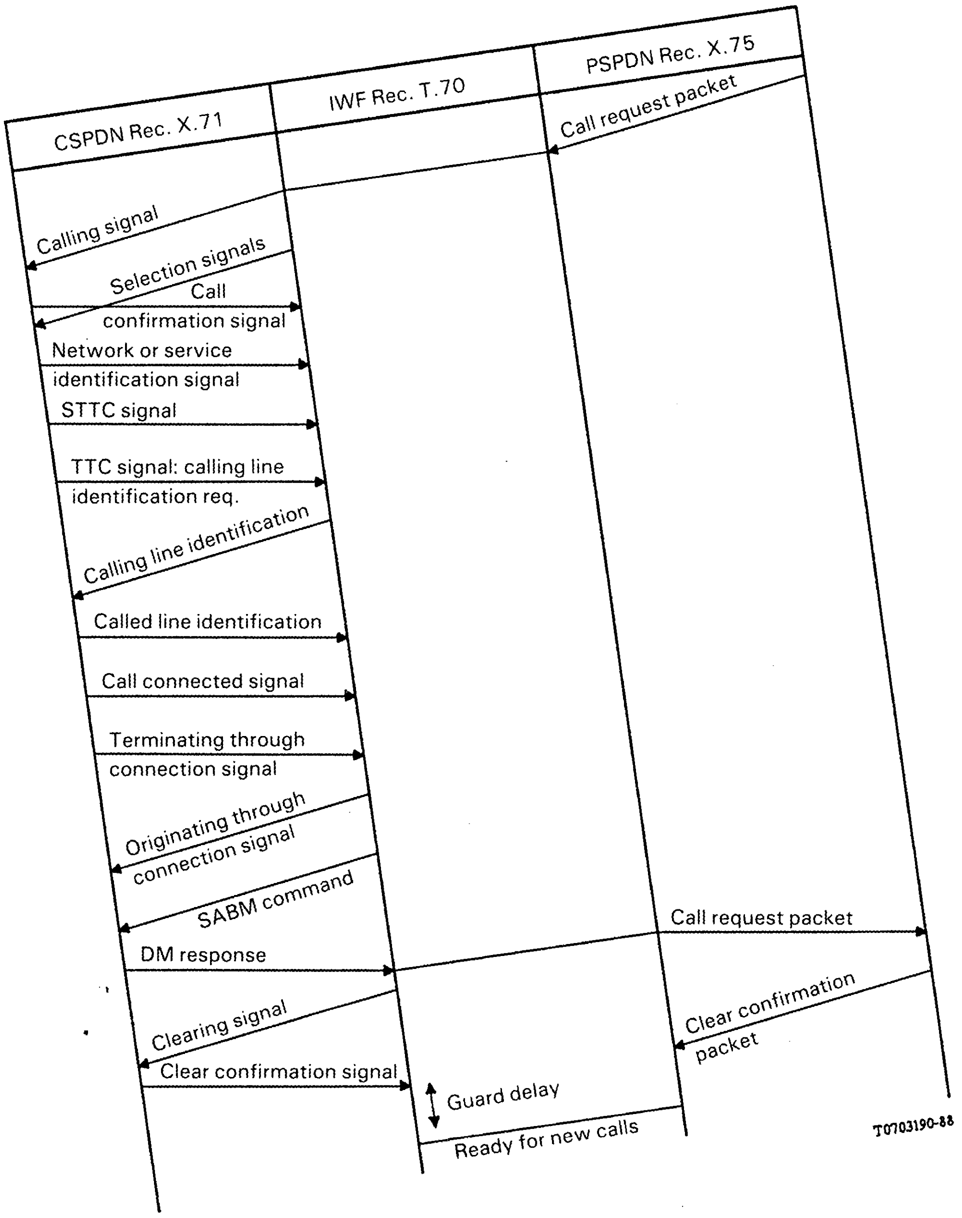
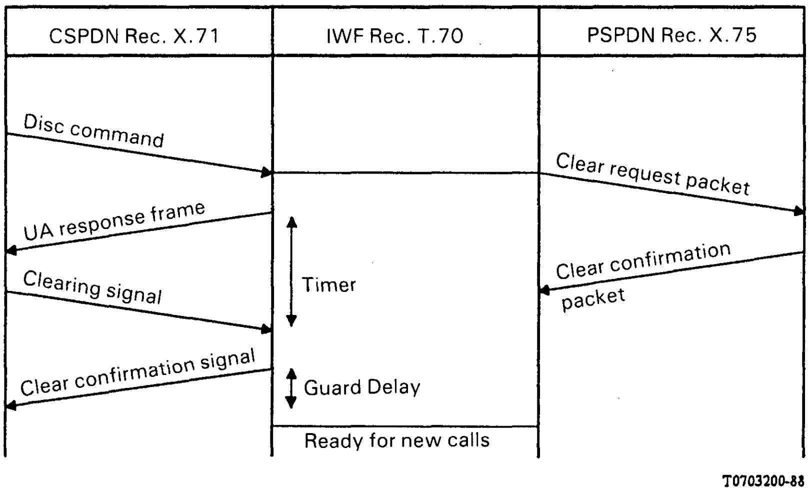


FIGURE 11/X.82

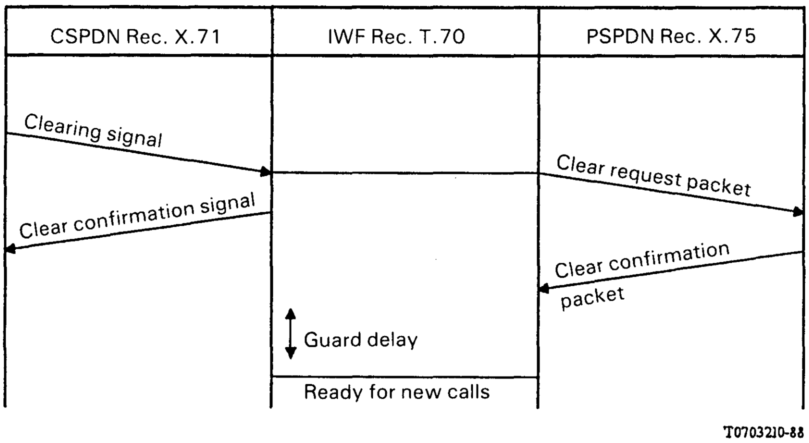
Unsuccessful connection establishment PSPDN → CSPDN (SABM command answered with DM Response)

6.2 *Connection release phase*

6.2.1 *Connection release initiated on the CSPDN-side* (Figure 12/X.82)



a) Possibility 1

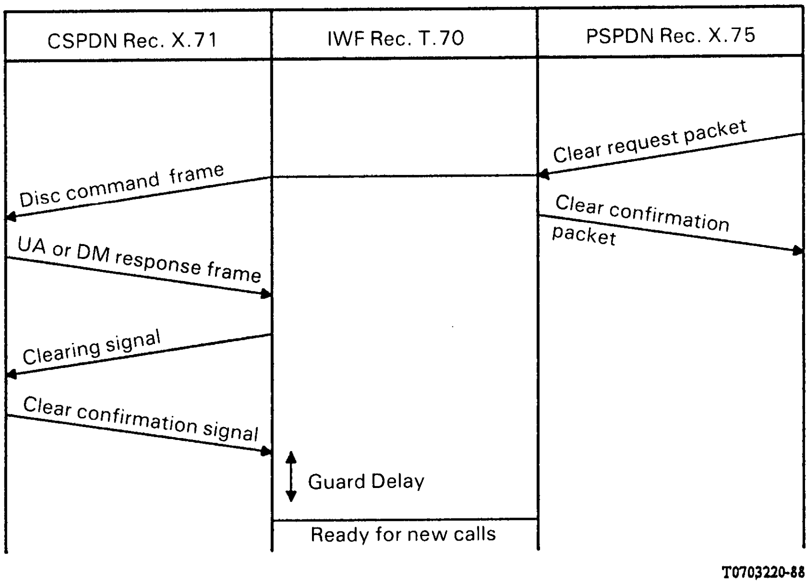


b) Possibility 2

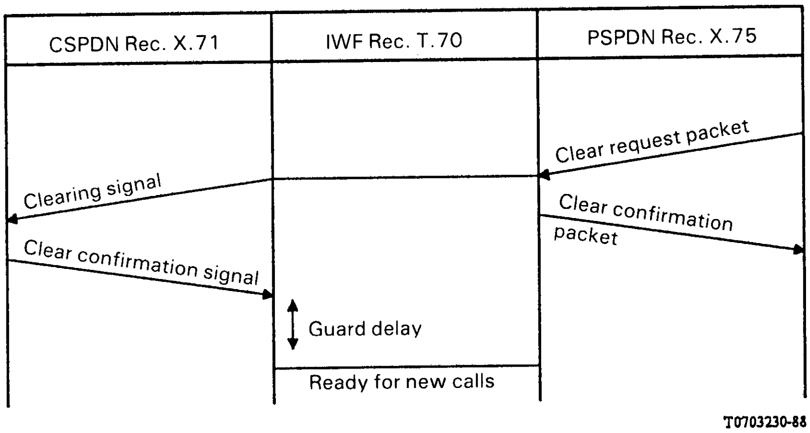
FIGURE 12/X.82

Connection Release initiated on the CSPDN-side

6.2.2 *Connection release initiated on the PSPDN-side* (Figure 13/X.82)



a) Possibility 1

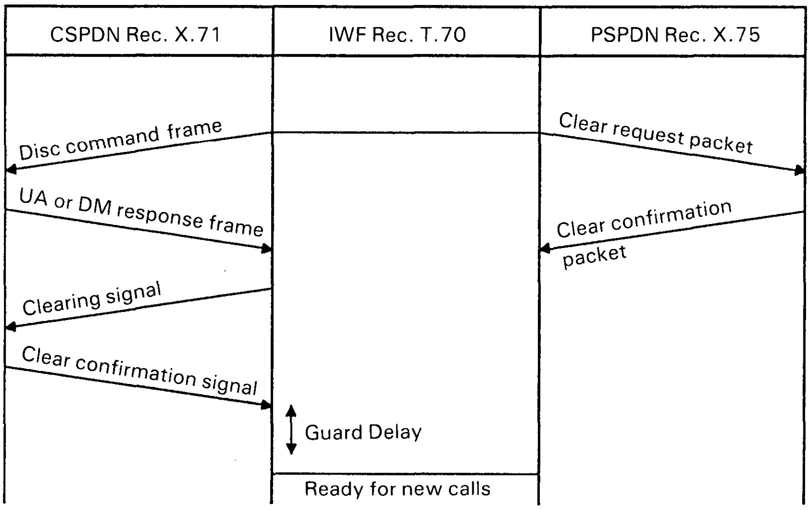


b) Possibility 2

FIGURE 13/X.82

Connection Release initiated on the PSPDN-side

6.2.3 *Connection release initiated by the IWF T70* (Figure 14/X.82)



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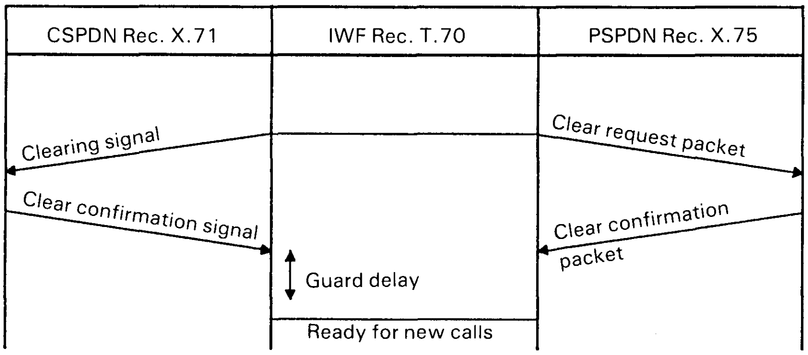
a) Possibility 1



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b) Possibility 2

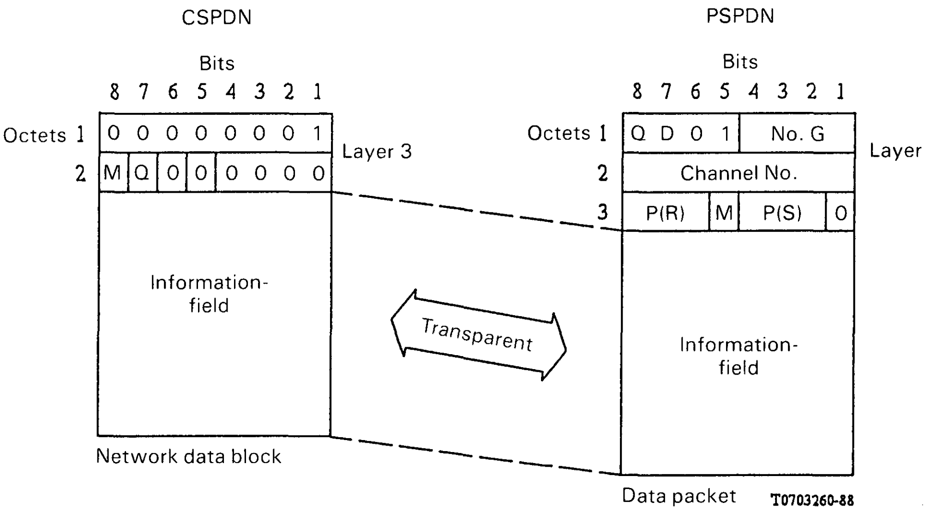
FIGURE 14/X.82

Connection Release initiated by the IWF T.70

6.3 *Data transfer phase*

6.3.1 *Handling of user data*

The user data shall be transferred transparently from the user data field in the data packet on the packet switched side to the user data field in the network data block on the circuit switched side, and vice versa (see Figure 15/X.82).



M-Bit	More data bit
Q-Bit	Qualifier bit
D-Bit	Delivery confirmation bit
G No.	Logical channel group number
Channel No.	Logical channel number
P(R)	Packet receive sequence number
P(S)	Packet send sequence number

FIGURE 15/X.82

Data transfer phase

6.3.2 *Handling of Qualifier bit (Q-bit)*

Independent from the value of the Q-Bit in an incoming data packet the IWF T.70 may

a) use Q-bit = 0 in the corresponding outgoing network data block, and vice versa;

b) transfer it transparently. This means that the value of the Q-bit in an incoming data packet shall be transferred in the corresponding outgoing network data block, and vice versa.

6.3.3 *Handling of Delivery confirmation bit (D-bit)*

The D-bit in incoming Data Packets is ignored. The IWF T70 shall set the D-bit = 0 in outgoing Data Packets.

6.3.4 *Handling of More data bit (M-bit)*

Network Data Blocks with more than 128 octets (up to 2048) are segmented into a Data Packet sequence with a number of octets in each Data Packet according to the selected value for the packet size. If the received M-bit of the last Data Packet in the sequence is set to 0. In all other cases the M-bit = 1.

Received Data Packets may be directly transferred in Network Data Blocks of the same size with the M-bit unchanged and transferred as one Network Data Block.

6.3.5 *Reset*

When a Reset request is received from the packet switched side this shall result in a disconnection of the link on the circuit switched side. After successful disconnection and setting up of the link again the reset is confirmed. The procedure is shown in Figure 16/X.82.

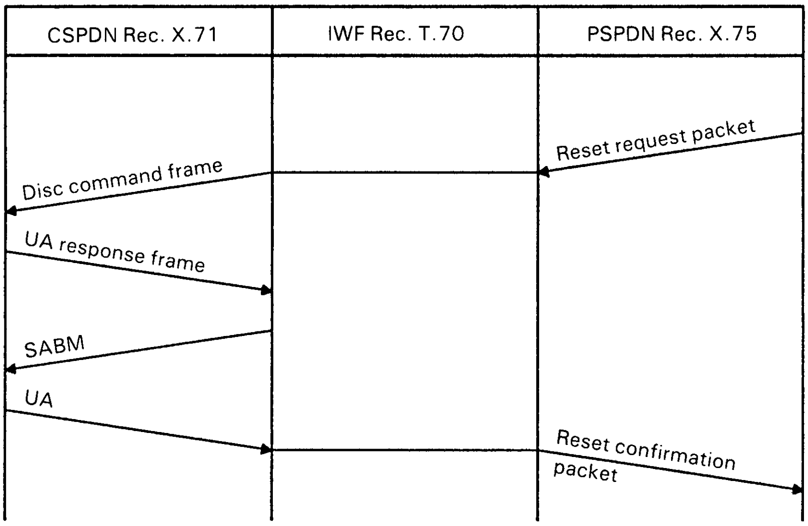


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FIGURE 16/X.82

Reset

7 Restart request

7.1 *Restart request initiated by the PSPDN*

An incoming Restart request packet from the packet switched side results in a clearing of all related circuits on the circuit switched side. This is described in Figure 17/X.82.

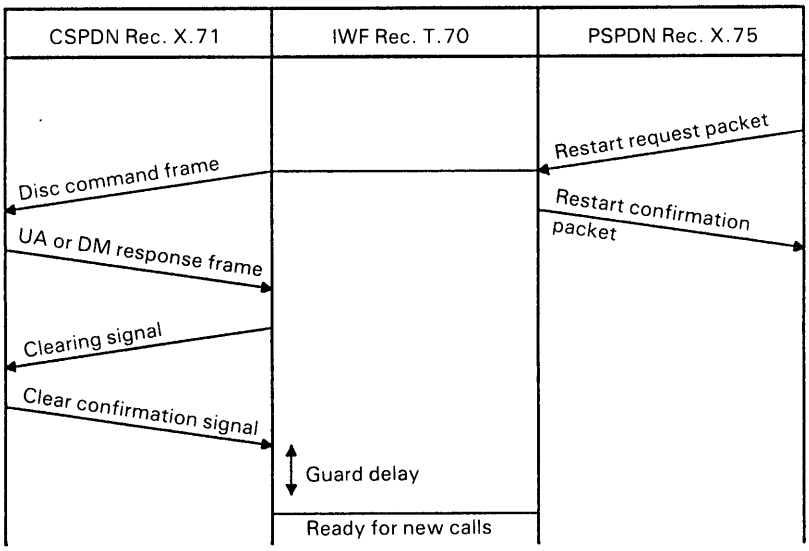

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FIGURE 17/X.82

Restart Request initiated by the PSPDN

7.2 *Restart request iniciated by the IWF T.70*

The IWF T.70 may request a restart by clearing all circuit on the circuit switched side and sending Restart request on the packet switched side. This is shown in Figure 18/X.82.

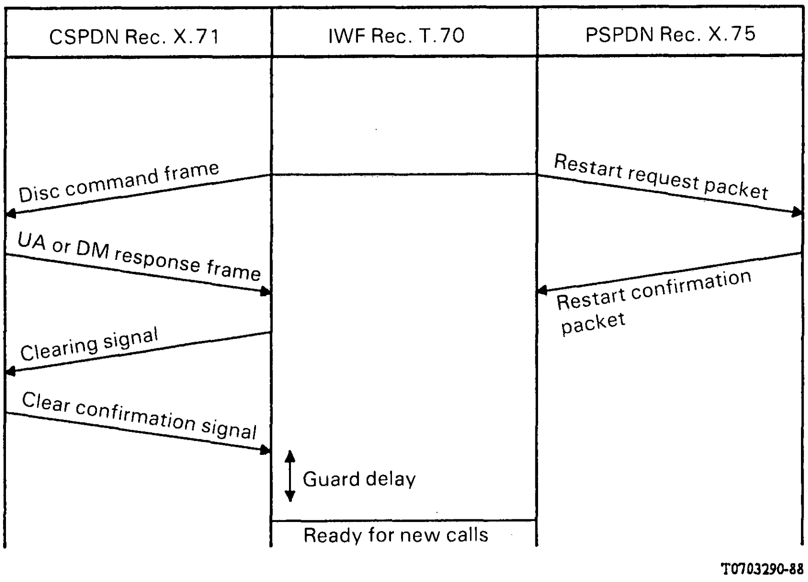


FIGURE 18/X.82

Restart Request initiated by the IWF T.70

APPENDIX I

(to Recommendation X.82)

Items in relationship to the OSI connection-mode network service

1) Connection establishment phase	
– Network address extension	not supported
– Receipt confirmation selection	not supported (see Note 1)
– Expedited data	not supported (see Note 1)
– QOS	not supported (see Note 1)
– NS user data	not supported (see Note 2)
2) Data transfer phase	
– D-bit	not supported (see Note 1)
3) Connection release phase	
– Network address extension	not supported,
– NS user data	not supported (see Note 2).

Note 1 – NS provider options, when provided within a subnetwork, would lead to additional actions and events (i.e. receipt confirmation, EXPEDITED DATA transfer).

Note 2 – The objective is to make this parameter a mandatory parameter to be supported by all subnetworks in the future. However, a number of existing subnetworks cannot support it now. During the interim period, while these subnetworks exist and are not modified to provide this parameter, it is considered as a provider-option. No negotiation mechanism is needed in the connection-mode network service. Limiting, in some subnetworks, the length of NS-user-data to be provided to a value lower than 128 octets (e.g. 16 to 32 octets) for an interim period would imply fewer changes to existing interfaces and signalling systems and would simplify the introduction of such a service in existing subnetworks.

SECTION 4

NETWORK ASPECTS

Recommendation X.92

HYPOTHETICAL REFERENCE CONNECTIONS FOR PUBLIC SYNCHRONOUS DATA NETWORKS

(Geneva, 1976; amended at Malaga-Torremolinos, 1984)

The CCITT,

bearing in mind

(a) the international user classes of service indicated in Recommendation X.1;

(b) the overall user-to-user performance objectives;

(c) the need to standardize the procedures for use over public synchronous data networks;

(d) in the case of packet switching, the need to standardize several procedural levels,

unanimously recommends

the use of the five hypothetical reference connections contained in this Recommendation.

1 The five hypothetical reference connections set down in the present Recommendation (see Figure 1/X.92) are intended for assessing the overall customer-to-customer performance objectives, for determining some data characteristics requirements of the various items in the connections and for setting limits to the impairments these items may introduce.

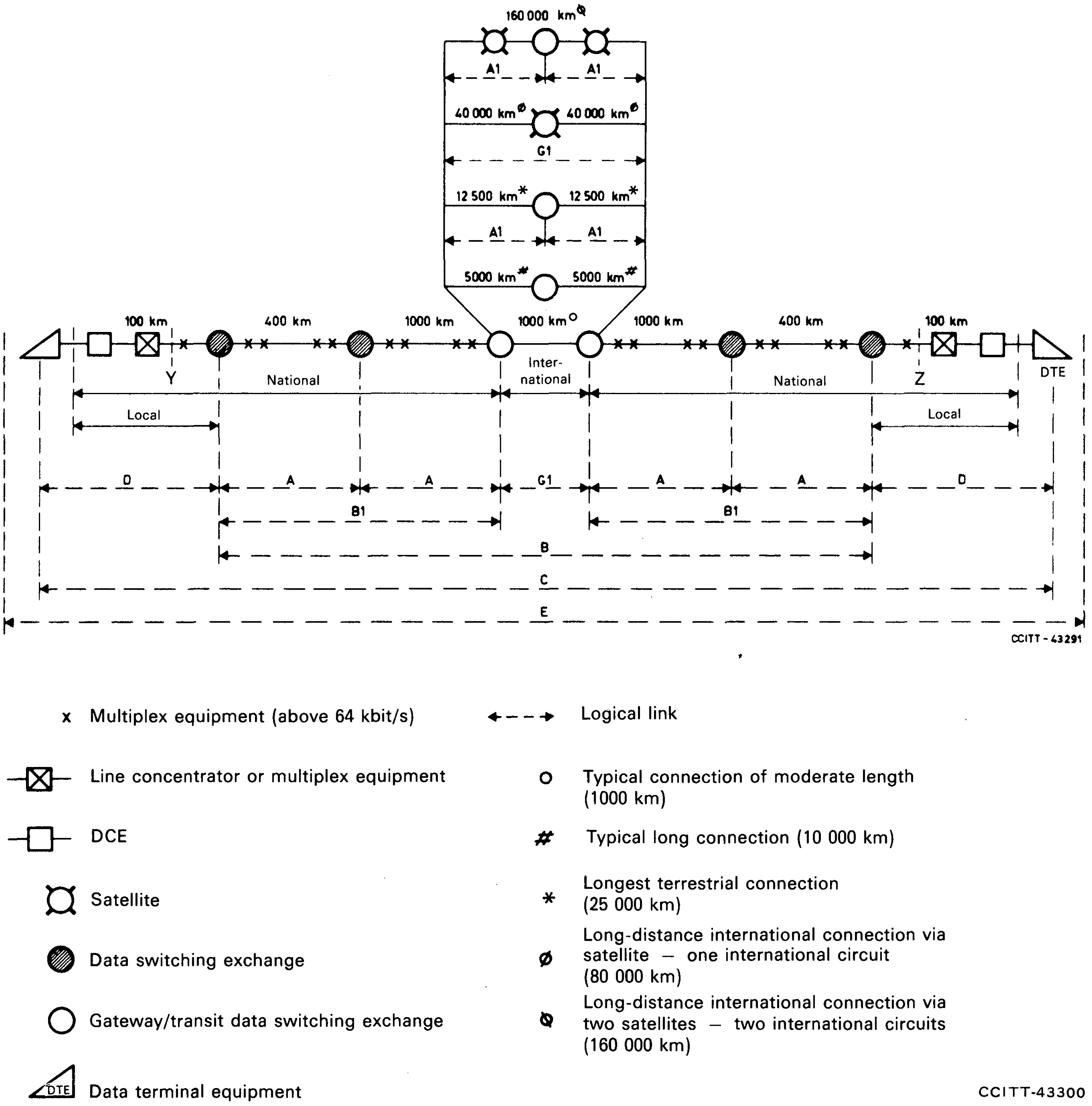
These hypothetical reference connections should be used for circuit-switched services, packet-switched services and leased line services in public synchronous data networks.

Other hypothetical reference connections may be set up in the future after experience of the design of synchronous public data networks has been gained.

2 The hypothetical reference connections of Figure 1/X.92 are intended for the user data signalling rates as recommended in Recommendation X.1.

Between points Y and Z, transmission takes place over 64 kbit/s digital paths. Such paths may include digital sections using modems over analogue facilities.

It should be assumed that the signalling for the circuit-switched data call control follows the same route as the data connection.



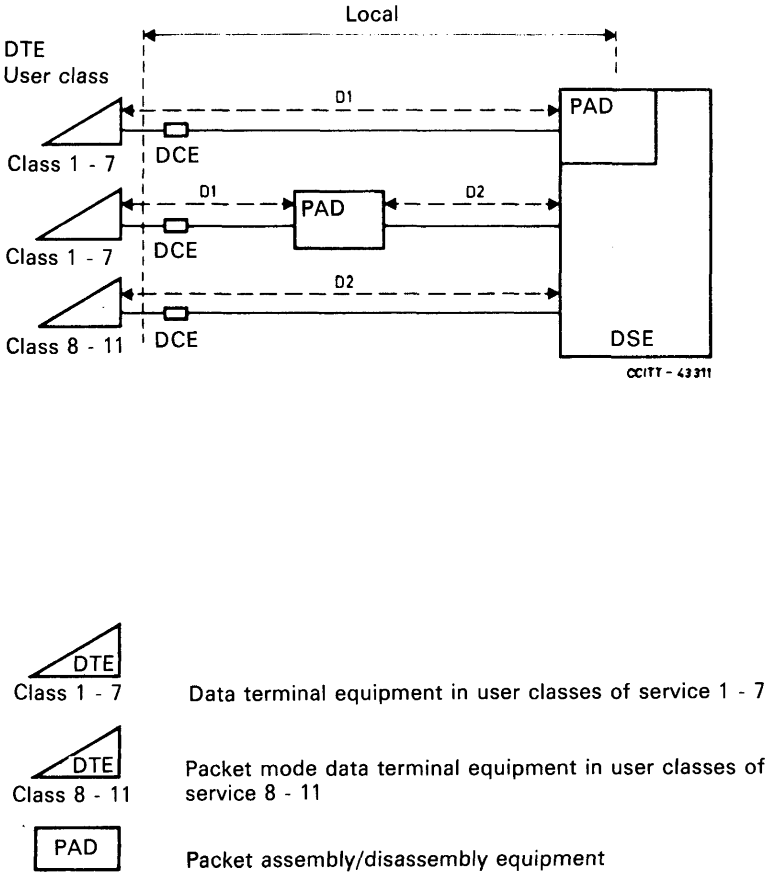
Link A = Data link between two adjacent data switching exchanges in a national network
Link A1 = Data link between two adjacent gateway data switching exchanges in an international connection
Link B = Data link between a source DSE and a destination DSE
Link B1 = Data link between a local DSE and a gateway DSE
Link G1 = Data link between a source gateway DSE and a destination gateway DSE in an international connection
Link C = Data link between source DTE and destination DTE
Link D = Data link between source DTE and the source local DSE or the data link between destination DTE and destination local DSE
Link E = Data link between communicating processes

FIGURE 1/X.92

Hypothetical reference connections for public synchronous data networks

3 The logical links to be considered in the case of packet switching are indicated in Figure 1/X.92 by the dotted lines.

To allow for the incorporation of packet assembly/disassembly facilities, the variants to logical Link D of Figure 1/X.92, which are shown in Figure 2/X.92, are recognized.



Data terminal equipment in user classes of service 1 - 7

Packet mode data terminal equipment in user classes of service 8 - 11

Packet assembly/disassembly equipment

Link D1 = Data link between a data terminal equipment in user class of service 1- 7 and a packet assembly/disassembly equipment

Link D2 = Data link between a data terminal equipment in user class of service 8- 11 or a packet assembly/disassembly equipment and a local data switching exchange

Note 1 – A user may see two different types of logical interfaces with the network (Links D1 and D2).

Note 2 – Link D2 could provide an interface for a single access terminal as well as for a multiple access terminal.

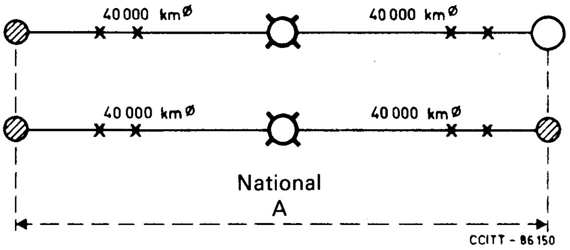
FIGURE 2/X.92

Variants of logical link D

4 It would be permissible to include a satellite in the transmission path of the national or local link. To allow for this, the variants of logical Links A and D of Figure 1/X.92, which are shown in Figures 3/X.92 and 4/X.92 respectively, are recognized.

In any connection, the maximum number of links via satellite should not exceed three (see Note).

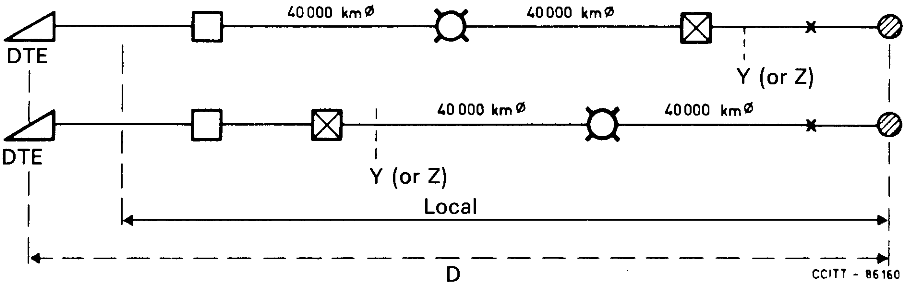
Note – The maximum number of links via satellite allowable in a connection requires further study, in the light of signalling time-outs and quality of service considerations.



Note – For legends, see Figure 1/X.92.

FIGURE 3/X.92

Variants of logical link A



Note – For legends, see Figure 1/X.92.

FIGURE 4/X.92

Variants of logical link D

Recommendation X.96

CALL PROGRESS SIGNALS IN PUBLIC DATA NETWORKS

(Geneva, 1976; amended at Geneva, 1980 and Malaga-Torremolinos, 1984, and Melbourne, 1988)

The CCITT,

considering

that the establishment of public data networks for data transmission in various countries and the subsequent international interconnection of these networks creates the possibility that, in certain circumstances, there is a need to inform the caller about the progress of the call,

unanimously declares

that call progress signals should be returned to the caller to indicate the circumstances which have prevented the connection being established to a called number;

that call progress signals should be returned to the caller to indicate in some circumstances the progress made towards establishing the call;

that in addition, for packet-switched services, call progress signals should also be transmitted:

- if a problem is detected at a DTE/DCE interface which may have an impact on data integrity,
- for the virtual call (VC) service, to the calling and called DTEs when a call is reset or cleared after having been established,
- for the permanent virtual circuit (PVC) service between two DTEs to both DTEs when the permanent virtual circuit is reset.

The call progress signals and their related circumstances giving rise to them are defined in Table 1/X.96.

Call progress signal format and coding shall be in accordance with relevant interface specifications in the Series X Recommendations.

In a circuit-switched service, call progress signals may only be transmitted during the call set-up phase. In a packet-switched service they may also be transmitted during the data transfer phase and the call clearing phase of a virtual call.

The call progress signals are categorized according to their significance to the network or DTE and the type of action expected of the DTE receiving the signal – refer to Table 2/X.96.

The sequence of call progress signals in Table 1/X.96 implies, for categories C and D, the order of call set-up processing by the network. In general the DTE can assume, on receiving a call progress signal, that no condition higher up the table is present. Network congestion, network out-of-order, long-term network congestion and no connection are exceptions to this general rule. The actual coding of call progress signals does not necessarily reflect this sequence.

Except as noted in Note 4 to Table 1/X.96, all call progress signals will be extended to the DTE unmodified. Users and DTE manufacturers are warned to make due allowance for possible later extensions to this table by providing appropriate fall-back routines for unexpected signals.

TABLE 1/X.96

Call progress signal	Definition	Category	Applicable to			See Note
			Circuit switching	Packet switching		
				VC	PVC	
Terminal called	The incoming call was signalled to the DTE and call acceptance is awaited.	A	(M)	–	–	1
Redirected call	The call has been redirected to another number assigned by the called subscriber.	A	(M)	–	–	
Connect when free	The called number is busy and the call has been placed in a queue. The call will be connected when the called number becomes free if the caller waits.	A	(M)	–	–	
Registration/cancellation confirmed	The facility registration or cancellation requested by the calling DTE has been confirmed by the network.	B	(M)	(M)	(M)	11
Redirection facility active	The redirection facility is active.	B	(M)	–	–	2
Redirection facility not active	The redirection facility is not active.	B	(M)	–	–	2
No connection	Cause unspecified.	C1	M	–	–	
Selection signal transmission error	A transmission error has been detected in the selection signals by the first Data Switching Exchange (DSE).	C2	M	–	–	
Local procedure error	A procedure error caused by the DTE is detected by the DCE at the local DTE/DCE interface. Possible reasons are indicated in relevant Series X interface Recommendations (e.g.: incorrect format, expiration of a time-out).	D1	M	M	M	3
Network congestion	A condition exists in the network such as: 1) temporary network congestion, 2) temporary fault condition within the network, including procedure error within a network or an international link.	C2	M	M	M	
Network out of order	Temporary inability to handle data traffic.	C2	–	–	M	

TABLE 1/X.96 *(cont.)*

Call progress signal	Definition	Category	Applicable to			See Note
			Circuit switching	Packet Switching		
				VC	PVC	
Invalid facility request	A facility requested by the calling DTE (circuit switching or packet switching services) or the called DTE (packet-switching service only) is detected as invalid by the DCE at the local DTE/DCE interface. Possible reasons include: – request for a facility which has not been subscribed to by the DTE, – request for a facility which is not available in the local network, – request for a facility which has not been recognized as valid by the local DCE.	D1	M	M	–	
RPOA out of order	The RPOA nominated by the calling DTE is unable to forward the call.	D2	(M)	(M)	–	4
Changed number	The called DTE has been assigned a new number.	D1	M	–	–	
Not obtainable	The called DTE address is out of the numbering plan or not assigned to any DTE.	D1	M	M	–	
Access barred	The calling DTE is not permitted the connection to the called DTE. Possible reasons include: – unauthorized access between the calling DTE and the called DTE; – incompatible closed user group.	D1	M	M	–	
Reverse charging acceptance not subscribed	The called DTE has not subscribed to the reverse charging acceptance facility.	D1	FS	(M)	–	
Incompatible user class of service	The called DTE belongs to a user class of service which is incompatible with that of the calling DTE.	D1	M	–	–	5

TABLE 1/X.96 *(cont.)*

Call progress signal	Definition	Category	Applicable to			See Note
			Circuit switching	Packet switching		
				VC	PVC	
Fast select acceptance not subscribed	The called DTE has not subscribed to the fast select acceptance facility.	D1	–	(M)	–	
Incompatible destination	The remote DTE/DCE interface or the transit network does not support a function or facility requested.	D1	–	M	M	
Ship absent	The called ship is absent.	D1	–	M	–	13
Out of order	The remote number is out of order. Possible reasons include: – DTE is uncontrolled not ready; – DCE power off; – network fault in the local loop; – in packet switched services only: X.25 level 1 not functioning, X.25 level 2 not in operation.	D1 or D2	See Note 6	M See Note 7	M	8
Network fault in the local loop	The local loop associated with the called DCE is faulty.	D2	M Unless out of order is provided	See Note 7		9
DCE power off	Called DCE has no mains power or is switched off.	D1	M	See Note 7		9
Uncontrolled not ready	Called DTE is uncontrolled not ready.	D1	M	See Note 7		9
Controlled not ready	Called DTE is signalling controlled not ready.	D1	M	FS	FS	1
Number busy	The called DTE is detected by the DCE as engaged on other call(s), and therefore as not being able to accept the incoming call.	C1	M	M	–	
Call the information service	The called number is temporarily unobtainable. Call the network information service for details.	D1	M	–	–	

TABLE 1/X.96 *(end)*

Call progress signal	Definition	Category	Applicable to			See Note
			Circuit switching	Packet switching		
				VC	PVC	
Remote procedure error	A procedure error caused by the DTE or an invalid facility request by the remote DTE is detected by the DCE at the remote DTE/DCE interface. Possible reasons are indicated in relevant Series X interface Recommendations.	D1	–	M	M	
Long term network congestion	A major shortage of network resource exists.	D2	M	–	–	10
Network operational	Network is ready to resume normal operation after a temporary failure or congestion.	C2	–	–	M	
Remote DTE operational	Remote DTE/DCE interface is ready to resume normal operation after a temporary failure or out of order condition (e.g. restart at the DTE/DCE interface). Loss of data may have occurred.	C1	–	–	M	
DTE originated	The remote DTE has initiated a clear, reset or restart procedure.	B or D1	–	M	M	12
PAD clearing	The call has been cleared by the local PAD as an answer to an invitation from the remote DTE (X.28 only).	B	–	M (X.28 only)	–	
Private/public network reached	See Annex F of Recommendation X.21.	A	–	–	–	14
DTE interactive	The called DTE has registered for being inactive until the date and time indicated.	D1	–	–	–	

– Not applicable.

M Mandatory in all networks.

(M) Mandatory where the relevant optional user facility is provided.

FS Further study.

Note 1 – The international implications of *controlled not ready* and *manual answering* are for further study.

Note 2 – Sent as confirmation/answer for the *redirection activation/deactivation* facility.

Note 3 – For circuit switching, applicable only to the calling DTE.

Note 4 – The *RPOA out-of-order* call progress signal will not be returned to a DTE which does not subscribe to the *RPOA selection* facility.

Note 5 – Some networks may use the *not obtainable* call progress signal to signal this condition.

Note 6 – Used as an alternative signal in networks where one or more of the conditions *uncontrolled not ready, DCE power off* and *network fault in the local loop* cannot be uniquely identified.

Note 7 – Although the basic *out-of-order* call progress signal is transmitted for these conditions, the diagnostic field in the *clearing* or *resetting* packet may give more precision.

Note 8 – The fact that a DTE is also out of order when the link access procedure level is not operating correctly is a subject for further study.

Note 9 – Should be provided where the network can identify the condition.

Note 10 – Activated by the operational staff of the network.

Note 11 – Applicable only to the DTE/DCE interface (*restart* packets in case of packet switching service).

Note 12 – Possible reasons for this include *reverse charging not accepted. Reset* and *restart* are not applicable to the circuit-switching service.

Note 13 – Used only in conjunction with mobile maritime service.

Note 14 – Refer to Notes 3 and 4 of Annex F to Recommendation X.21.

TABLE 2/X.96

Category	Significance
A	Call not cleared. Calling DTE is expected to wait.
B	Call cleared because the procedure is complete.
C1 and C2	*Call cleared* The call has failed due to conditions of a temporary nature. The DTE may try again after a suitable delay as the next attempt may be successful. However, after a number of unsuccessful call attempts with the same response, the action taken by the DTE should be as defined in category D1 or D2. Some Administrations may specify by regulation the interval between and maximum number of call attempts permitted by a DTE in these circumstances. *Reset* (for packet-switched services only) The DTE may continue to transmit data recognizing that data loss may have occurred.
D1 and D2	*Call cleared* The calling DTE should take other action to clarify when the call attempt might be successful. *Reset* (for permanent virtual circuit only) The DTE should cease data transmission and take other action as appropriate.
C1 and D1	Due to subscriber condition.
C2 and D2	Due to network condition.

INTERNATIONAL ROUTING PRINCIPLES AND ROUTING PLAN FOR PUBLIC DATA NETWORKS

(Geneva, 1980; amended at Malaga-Torremolinos, 1984,
and Melbourne, 1988)

The CCITT,

considering

(a) that Recommendation X.1 defines the international user classes of service for public data networks;

(b) that Recommendation X.92 defines the hypothetical reference connection for public synchronous data networks;

(c) that Recommendation X.121 defines the international numbering plan for public data networks;

(d) that Recommendation X.122 defines numbering plan interworking between a PSPDN and an ISDN or PSTN in the short-term;

(e) that Recommendations X.130 and X.131 define the quality of service parameters for circuit-switched public data networks (CSPDN);

(f) that Recommendations X.135 and X.136 define the quality of service parameters for packet-switched public data networks (PSPDN);

(g) that Recommendation X.75 defines terminal and transit call control procedures and data transfer systems on international circuits between packet-switched public data networks;

(h) that Recommendations X.70, X.71, X.60 and X.61 define the signalling systems for circuit-switched public data networks;

(i) that Recommendation X.353 defines the routing principles interconnecting PDNs with mobile-satellite systems,

unanimously declares

that the following routing principles should be applied for the establishment of circuit-switched calls or packet-switched virtual calls when interconnecting public data networks.

Note – There is a requirement for a Recommendation that defines the dimensioning of routes to enable a specified grade of service to be met. (This requires further study).

1 Introduction

1.1 This Recommendation should be applied to public data networks and referred to when Administrations are planning an interconnection between public data networks. The Recommendation contains guidelines which Administrations should follow and also gives examples of specific routings. Its aim is to encourage the spread of public data networks internationally and foster a commonality of understanding leading to an orderly development of the international network, making effective and economical use of network resources. It is hoped that its use will enable the evolution of inter-networking to take place between public data networks, ISDNs, international telephone networks and other public networks. It is recognized that the plan will need to be reviewed periodically to ensure that it is in step with actual practice taking place within the international public data networks. In order to aim at a better understanding of the international routing plan for public data networks, an international public data network model is illustrated in Figure 1/X.110. This model consists of a set of national public networks and shows the interconnection of national public data networks and international data switching exchanges (IDSEs). Public data networks have evolved in different ways in many countries. The model shown illustrates six types of networks that have evolved as follows:

a) some countries may have more than one PDN and also more than one IDSE. See country A in Figure 1/X.110;

b) some countries may have one IDSE which stands alone from that country's PDN. See country B in Figure 1/X.110;

c) some countries may have one PDN and gain international access through one IDSE. See country C in Figure 1/X.110;

d) some countries may not have a PDN but utilize an IDSE for international connections. See country D in Figure 1/X.110;

e) some countries may have more than one PDN each with its own IDSE. See country E in Figure 1/X.110;

f) some countries may have more than one PDN each sharing one or more IDSE. See country F in Figure 1/X.110.

1.2 Circuits between IDSEs in the same country are not classed as international links.

1.3 A list of the terms and definitions used in this Recommendation are recorded in Annex A.

2 Description of an international route

2.1 The basic function of routing a call (or selecting a route for a call) consists of selecting the network equipment (i.e., outgoing link) that will be used for transferring data for that call.

2.2 The route used for an international call always consists of three parts:

- an originating national network part, from the calling DTE to the originating IDSE (i.e., through the originating PDN);
- an international network part, from the originating IDSE to the destination IDSE (i.e., through the international public data network IPDN);
- a destination national network part, from the destination IDSE to the called DTE (i.e., through the destination PDN).

Note – For maritime satellite data transmission systems a maritime satellite data switching exchange (MSDSE) would function as the originating and destination IDSE.

2.3 The planning of the international network part is subject to CCITT study.

2.4 The planning of the originating and destination national network parts is a national matter; however, the quality of service (e.g., transfer delay) provided on international connections should be considered in these national networks.

3 General routing principles

3.1 The planning of international data traffic routes is the responsibility of the Administrations concerned and is subject to bilateral agreements.

3.2 The traffic route within the international network part should be so planned as to encompass no more than four international data links in tandem.

3.3 When planning traffic routes, quality of service (QOS) requirements should be taken into account. One such QOS requirement is the overall transfer delay of the connection. In considering the overall transfer delay, the number of satellite links is of major significance. However, it is noted that transfer delay in PSPDNs exclusive of satellite links, may also be significant and is for further study.

3.4 In accordance with Recommendation X.92, no more than three satellite links should normally be included in an overall PDN route. The international network part should normally not include more than two satellite links. (Refer to Annex B.)

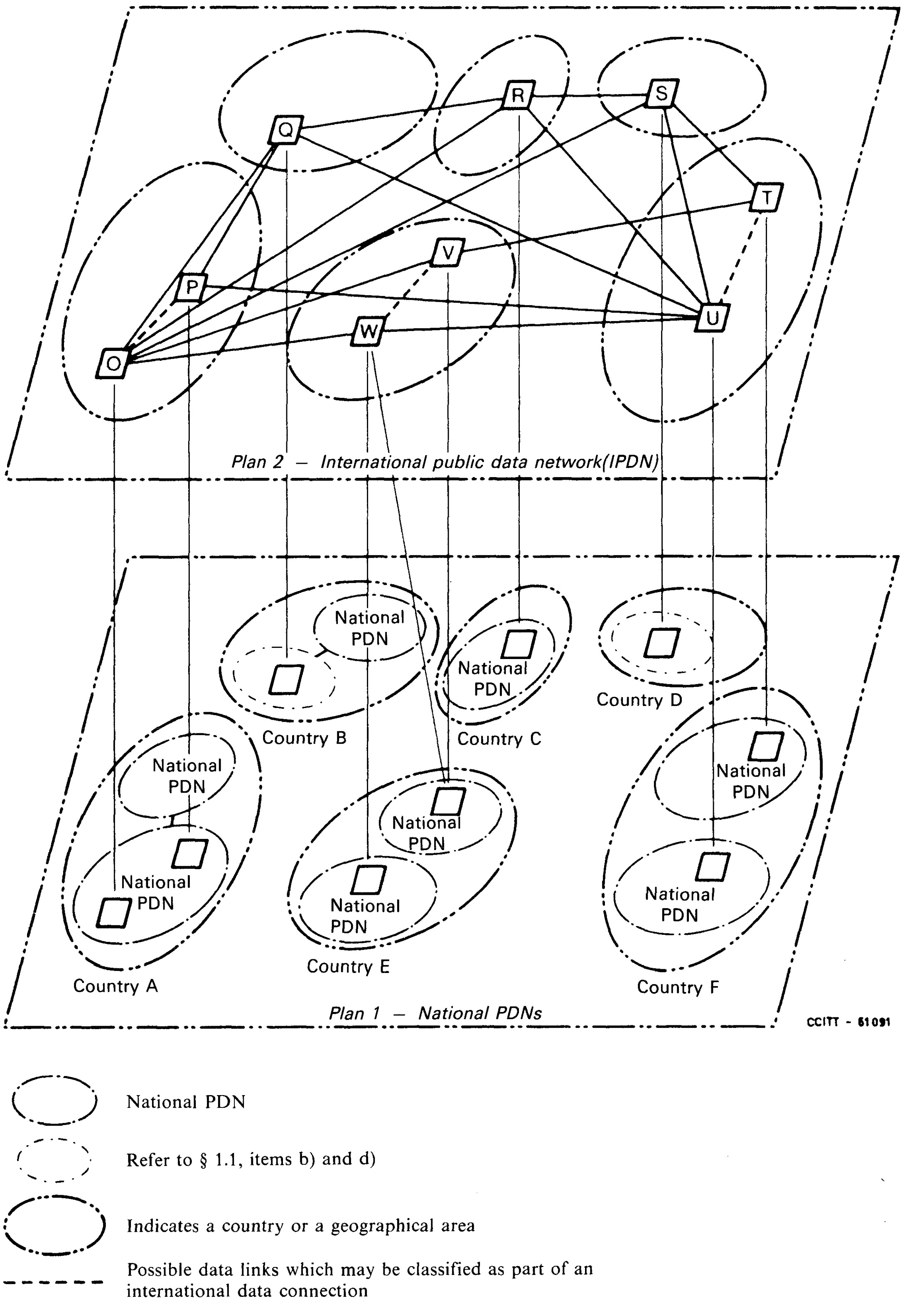


FIGURE 1/X.110

International public data network model

3.5 Traffic routes will normally comprise direct call routes and alternative call routes.

3.6 Traffic routes should be planned so as to avoid the possibility of circular call routings.

3.7 When planning traffic routes, use should be made of time-zone differences.

3.8 The routing of a call is a matter under the responsibility of Administrations and should, where possible, follow one of the traffic routes agreed to in § 3.1.

3.9 All Administrations concerned with the routing of a given call should be able to obtain the necessary information for that call (e.g., the DNIC of each network concerned).

Note – The application of this principle to circuit-switched networks is for further study.

3.10 The international network part for a call should be selected on a link-by-link basis by the IDSEs concerned.

3.11 The international call route for a connection is selected by the IDSEs concerned. Under normal conditions, when a call route for a specific call has been established, that call route should be used for the entire duration of that call.

3.12 Calls should be routed using the minimum number of international data links taking into account the economics and practicalities of the situation.

3.13 If a trunk cannot satisfy the throughput requirements made by the originating subscriber it will be necessary to select one of the possible alternative call routes.

4 Specific routing possibilities through the IPDN

4.1 *Routing possibilities required for maintaining the quality of service*

Specific routing possibilities may be considered for maintaining a good quality of service, for example:

- the selection of a reliable route for a call, in order to avoid clearing of the call by the network (or reset of the call in the case of a virtual call), due to internal network problems;
- the availability of more than one call route between originating and destination networks, in order to avoid a call request being barred if one call route is temporarily not available.

4.2 *Service characteristics associated with a route*

During a call establishment, a public data network may have to consider some aspects of the network service characteristics to make routing decisions.

Whenever several traffic routes can be used between two users, in addition to the availability of those traffic routes at a given time, it is important that the service characteristics associated with any one of those traffic routes be considered (e.g., throughput available, acceptance of some facilities, etc.).

4.3 *Specific conditions associated with a route*

During call establishment, a public data network may have to consider specific conditions such as reverse charge request, access protection (closed user group, incoming calls barred), etc. Under such circumstances, as far as possible the Administrations should endeavour to provide call routings subject to:

a) the availability of the facilities required;

b) bilateral agreement.

Failing that, the call should be barred.

5 Routing procedures applicable to international interworking between PDNs of the same type and also between PSPDNs and ISDNs and/or PSTNs in the short term

5.1 International Data Switching Exchanges (IDSEs) will recognize the calling and the called data network identification codes (DNICs) or data country codes (DCCs) to determine the destination of a call and the call route. (See Notes 1, 2, 3 and 4).

Note 1 – The application of this principle to circuit-switched networks is for further study.

Note 2 – For PSPDN to PSPDN interworking, possible digit analysis of the first one or more digit(s) beyond the 4 digit DNIC field is to be determined on a bilateral basis if necessary.

Note 3 – For routing of calls to ISDN, PSTN and Mobile Satellite Systems from PSPDN, a digit analysis capability of at least 1 digit beyond the 4 digit DNIC field is required.

Note 4 – RPOA selection shall have no influence on the determination of the call route between the IDSEs.

5.2 The presence of an escape code, 0 or 9, as defined in Table 2/X.121 will have special significance for PSPDN to ISDN and PSTN routing as follows:

i) a value of 0 for the escape code will require that the IDSE route the call either to a digital interface to an ISDN or to a transit IDSE. (See Note.)

ii) a value of 9 for the escape code will require that the IDSE route the call either to an analogue interface to a PSTN, an ISDN or to a transit IDSE. (See Note.)

Note – To select a route, an IDSE may choose to examine up to 5 digits (escape code and first four digits of E.164 number).

5.3 The selection of links (e.g., satellite and/or submarine cable) for a given call route should be determined by the Administrations concerned on a per call basis.

5.4 The same call route will be maintained for the duration of a call.

5.5 Barring procedures for particular call routes will be provided by each Administration and will be the subject of bilateral agreement.

5.6 Transit networks will check routing information of each call to prevent circular routings.

6 Identification of IDSEs and ISDNs involved in an international call in the short term

Any Administration involved in providing transit IDSE(s) or ISDN(s) for an international call should be identified at the time of the call establishment by means of a DNIC or 4 digit ISDN Identification Code allocated to that administration (Notes 1, 2).

Note 1 – Exceptionally, a DNIC or ISDN Identification Code may need to be allocated to an Administration that would offer transit only and no direct subscriber access, for the purpose of identifying the transit ISDE(s) or ISDN(s).

Note 2 – The Administrations of the originating and destination networks are already identified within the calling and called DTE terminal addresses, and therefore do not require any additional identification at the time of the call establishment.

There may be more than one ISDE provided by the same Administration. Several independently operated networks may be provided by the same Administration. Independently operated networks may need to be identified even when the same Administration is concerned. Two or more IDSEs provided within the same independently operated network should be identified by the same DNIC (Note 3).

Note 3 – The provision of one DNIC or ISDN Identification Code for a transit, independently operated network is considered to be sufficient for covering the international accounting requirements, and for avoiding unexpected loops of calls between independently operated networks. The identifications needed for tracing the exact path of a call for maintenance are for further study.

7 Multiple IDSEs provided by one Administration

7.1 *In the originating or destination country*

The use by some Administrations of multiple originating and/or destination IDSEs could, in some cases, result in the routing of a call over a circuit between two IDSEs in the originating or destination country. Such circuits may be regarded as national links in applying this Recommendation.

7.2 *In a transit country*

Some Administrations may find it desirable to route transit traffic between two IDSEs in their own country. Such circuits need not be counted as one of the four international links allowed in this Recommendation, but from a transmission point of view must be counted as an additional international circuit.

8 International routing plan

8.1 Administrations may plan any traffic route providing it conforms to the principles in this Recommendation.

8.2 Since traffic routes can comprise direct and alternative routes, individual call routes should use the minimum number of IDSEs.

8.3 Many combinations of call routes are possible, some examples of which are contained in Appendix I.

8.4 Call rerouting can be planned if the required network management signals are available. An example of call rerouting is contained in Appendix I.

9 Network information required to enable optimum routings to be planned

Administrations should compile information concerning the quality of service parameters and network status of their networks for dissemination, on request, to those other interested Administrations who may wish to utilize it. These exchanges of information will enable Administrations to make optimum routing decisions when planning networks. Annex C contains a typical list of information of the type that should be available.

ANNEX A

(to Recommendation X.110)

Terms and Definitions related to routing in the PDN

This Annex contains terms and definitions that will be utilized in the PDN routing plan. These terms and definitions are based, as far as possible, on the currently available documentation both with CCITT and IEC (International Electrotechnical Vocabulary, Chapter 701).

To aid understanding, Figure A-1/X.110 records the relationship between the terms traffic route, alternative traffic route, call route, originating IDSE (IDSE-O), destination IDSE (IDSE-D), transit IDSEs (IDSE-X and IDSE-Y).

A.1 traffic route

A predetermined sequence of *trunk circuits* that is used to carry traffic between two points.

A.2 alternative traffic route

Between two given points more than one *traffic route* may exist. The availability of the option of using one of several routes is referred to as alternative traffic route.

A.3 **call route**

The sequence of circuits that is used to provide a *connection* between two points.

A.4 **call routing**

The action taken by an exchange of selecting a given *call route* from a number of *traffic routes.*

A.5 **call rerouting**

The action of changing a proposed *call route* during the attempted establishment of a *connection.*

A.6 **originating PDN**

A set of equipment and/or circuits which enable connection of a calling DTE to the originating IDSE.

A.7 **destination PDN**

A set of equipment and/or circuits which enable connection of a destination IDSE to the called DTE.

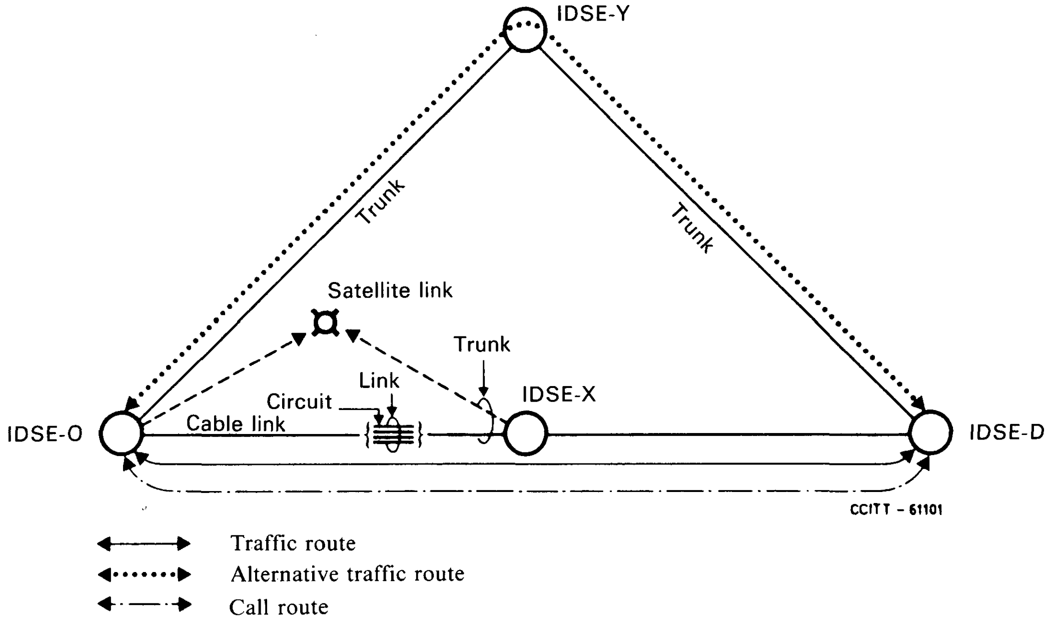


FIGURE A-1/X.110

The network part of the international public data network (IPDN)

ANNEX B

(to Recommendation X.110)

The use of satellite links in overall CSPDN routes

B.1 *General considerations*

1) When a satellite circuit is selected as the transmission path in an international connection, it should be noted that satellite circuits have some specific characteristics which need to be taken into account when used in PDNs.
2) It should be allowed to include a satellite link in the transmission path of a national network part of an international connection, recognizing that, in some cases, subscriber access may only be available via national or regional satellite systems.
3) It should be noted that in the international maritime satellite system for data communication services, only satellite paths are available in each ocean region.

In considering the above, the maximum number of satellite links allowed in an international connection, including both international and national network parts, should be three.

B.2 *Principles in each PDN*

B.2.1 *Originating national PDN*

It would be preferable to select routes which have a higher quality and minimum transfer delay for the national network part in the international connection. This would afford the maximum flexibility in the selection of the international links.

B.2.2 *Originating/transit IDSEs*

No more than two satellite links should be used in the international network part of the connection.

For calls to and from the maritime satellite data transmission system no more than one satellite link should be used in the international network part of the connection.

B.2.3 *Destination IDSE*

It is for further study to establish if the number of satellite links which have been applied to each call should be conveyed to the national PDNs in its call establishment stage.

If three satellite links have already been used in the connection so far, the use of a further satellite link in the destination Administration to complete the call should only be allowed with the consent of the Administrations concerned.

B.2.4 *Destination national PDN*

It would be preferable not to select any satellite links except in the case where no other possible route is available for this call.

ANNEX C

(to Recommendation X.110)

Routing information

The following information is typical of that which may be exchanged by Administrations during traffic routing negotiations:

1) name of country and DNIC to which their IDSEs are connected, indicating 1, 2, 3 or 4 link connections;
2) number of circuits and speed of transmission on each link using satellite or cable;
3) mode of working;
4) busy hour for each trunk and IDSE;
5) alternative traffic routes;
6) quality of service requirements;
7) facilities provided;
8) internetworking provided.

APPENDIX I

(to Recommendation X.110)

International routing plan – examples of routes

I.1 Administrations will wish to provide their routes in an economical way. When high volumes of traffic are forecast, a direct route with no intermediate IDSE will be planned and routes with low volumes will be switched at one or more transit IDSEs. Alternative routes will be provided over which the traffic will be carried when the direct route is unavailable. The routing algorithm will normally be: high usage route (direct), alternative route 1, alternative route 2. Administrations can make use of their agreed routes by offering them to third party Administrations for their own routes. Care should be taken to ensure that no routes planned in this way would involve inclusion of any more than 4 international links.

I.2 Figures I-1/X.110 to I-3/X.110 depict some typical routes that Administrations are likely to plan.

I.2.1 *Direct route* (high usage route)

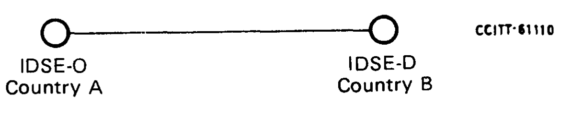


FIGURE I-1/X.110

Direct route

I.2.2 *Routes via intermediate countries* (low volumes of traffic)

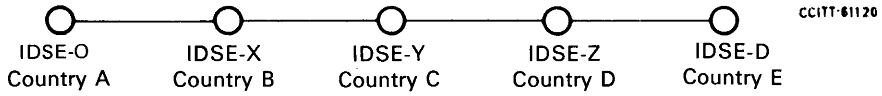
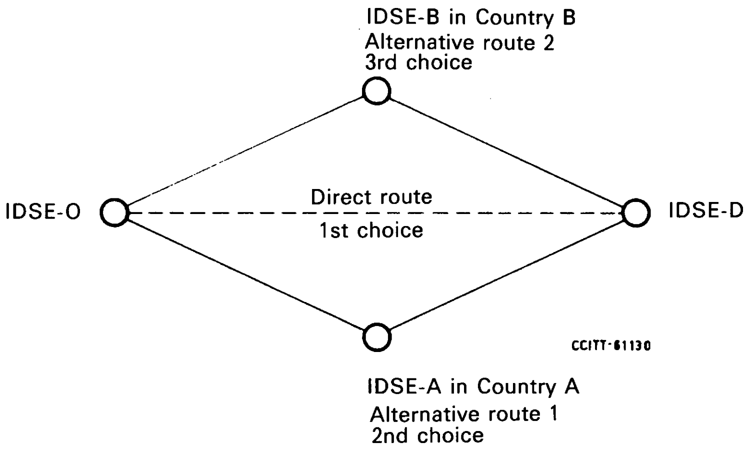


FIGURE I-2/X.110

Limiting condition route via 3 intermediate IDSEs

I.2.3 *Alternative route*



Note 1 – Likely routing selection process:

First choice – Direct route
Second choice – Alternative 1 via IDSE-A
Third choice – Alternative 2 via IDSE-B.

Note 2 – A similar routing algorithm may exist at intermediate IDSEs and care should be taken to ensure that the call is not routed using more than four links.

FIGURE I-3/X.110

Alternative route

I.2.4 *Routing plan in cases where direct routes are available*

Within the economic and political constraints of a country, the alternative routes should be selected with the following sequences for some particular connections.

The first alternative route selection would be made in the originating IDSE to one of the transit IDSEs which have direct routes to the destination IDSE (Figure I-4/X.110). If this is not the case, selection will be made to the transit IDSE without a direct route to the destination.

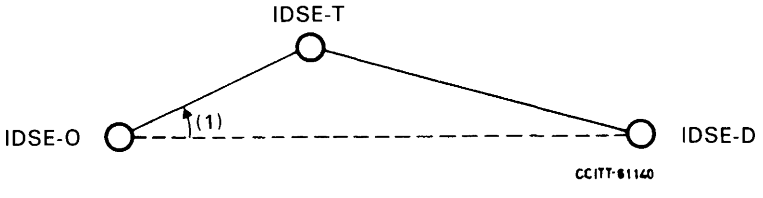


FIGURE I-4/X.110

The second alternative routing will be made in the first transit IDSE IDSE-T_1 to the second transit IDSE IDSE-T_2 with a direct route to the destination of this connection (Figure I-5/X.110).

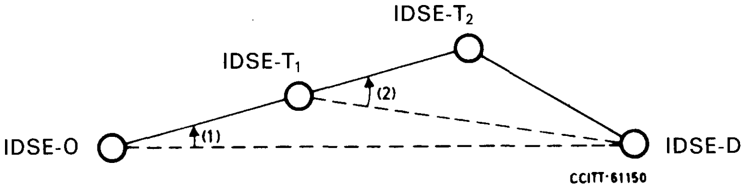


FIGURE I-5/X.110

The third alternative routing should be made in the same way, indicated in Figure I-6/X.110.

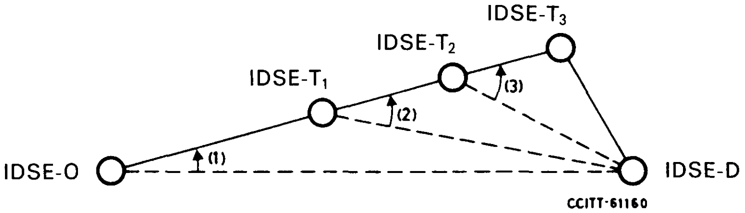


FIGURE I-6/X.110

I.2.5 *Routing plan in cases where no direct route is available*

In the case of traffic congestion between IDSE-O and IDSE-T_1, it is preferable to take another transit IDSE which has a direct route to the destination IDSE-D, if possible (Figure I-7/X.110).

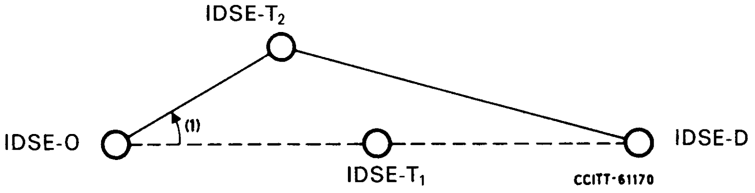


FIGURE I-7/X.110

If the originating IDSE-O must select a route to the transit IDSE-T_2 which has no direct route available to the destination IDSE-D, the subsequent transit IDSE may be the IDSE-T_1 (Figure I-8/X.110) or IDSE-T_3 (Figure I-9/X.110) if no direct route is available between IDSE-T_2 and IDSE-D.

The routing plan for the connection from IDSE-T_1 to IDSE-D would be the same as the plan indicated in § I.2.4 above.

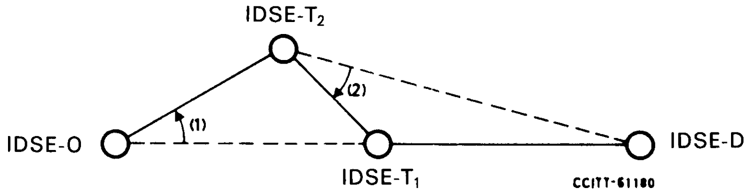


FIGURE I-8/X.110

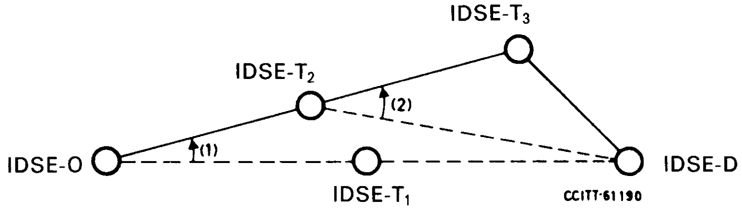
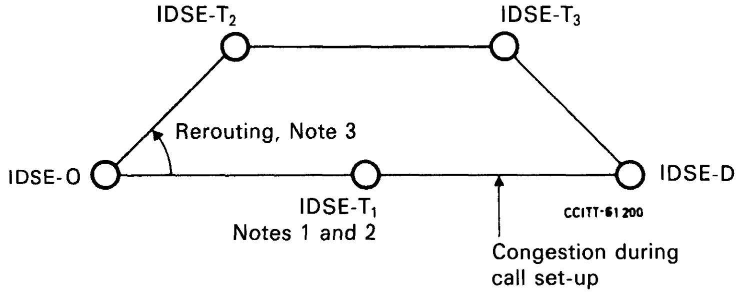


FIGURE I-9/X.110

I.3 *Rerouting*

The concept of rerouting considers calls that fail at an intermediate IDSE during call set-up. The details of call rerouting are for further study, however, Figure I-10/X.110 records the concept.



Note 1 – Call attempt reaches IDSE-T_1.

Note 2 – No route is provided between IDSE-T_1 and IDSE-T_3.

Note 3 – Call rerouting is attempted via IDSE-T_2, IDSE-T_3 and IDSE-D.

FIGURE I-10/X.110

Recommendation X.121

INTERNATIONAL NUMBERING PLAN FOR PUBLIC DATA NETWORKS[1]

(provisional, Geneva, 1978; amended, Geneva, 1980, Malaga-Torremolinos, 1984 and Melbourne, 1988)

The CCITT,

considering

(a) that the purpose of an International Numbering Plan for Public Data Networks is to facilitate the introduction of public data networks and provide for their interworking on a worldwide basis;

(b) that there could be a number of public data networks in a country;

(c) that the International Numbering Plan should permit the identification of a country as well as a specific public data network in that country;

(d) that the International Numbering Plan should provide means for interworking with other numbering plans;

(e) that Recommendation E.164 describes the Numbering Plan for the ISDN Era;

(f) that Recommendation F.69 describes the Plan of Telex Destination Codes;

(g) that the International Numbering Plan for Data Networks should provide for substantial spare capacity to accommodate future requirements,

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[1] In this Recommendation, the word "country" means country or geographical area.

unanimously declares

that the International Numbering Plan for Public Data Networks should be as defined in this Recommendation.

1 Design considerations

The design considerations that form the basis of this Numbering Plan are as follows:

1.1 The international data number is to determine only the specific DTE/DCE interface and, in particular, to identify a country, and a network, if several data networks exist in the same country.

1.2 Where a number of public data networks are to be established in a country, it should not be mandatory to integrate the numbering plans of the various networks.

1.3 The number of digits comprising the code used to identify a country and a specific public data network in that country should be the same for all countries.

1.4 A national data number assigned to a DTE/DCE interface should be unique within a particular national network. This national data number should form part of the international data number which should also be unique on a worldwide basis.

1.5 The number of digits to be used in an international data number should be governed by national and international requirements but a reasonable limit on the overall number of digits should be imposed.

1.6 The Numbering Plan should make provision for the interworking of data terminals on public data networks with data terminals on public telephone and telex networks and on Integrated Services Digital Networks (ISDNs).

Note – The term "telex" employed in this Recommendation includes TWX networks.

1.7 The Numbering Plan should not preclude the possibility of a single national network providing an integrated telecommunications system for services of all kinds.

1.8 Where multiple RPOA facilities exist providing service to or within the same country, provision for the selection of a specific RPOA facility should be allowed for in the *facility request* part of the *selection* signals.

Note – The term RPOA in this Recommendation refers to Recognized Private Operating Agency.

2 Characteristics and application of the Numbering Plan

2.1 *Number system*

2.1.1 The 10-digit numeric character set 0-9 should be used for numbers (or addresses) assigned to DTE/DCE interfaces on public data networks. This principle should apply to both national and international data numbers.

2.1.2 Use of the above number system will make it possible for data terminals on public data networks to interwork with data terminals on public telephone and telex networks and on Integrated Services Digital Networks (ISDNs).

2.2 *Data network identification codes and data country codes*

2.2.1 A Data Network Identification Code (DNIC) could be assigned as follows:

2.2.1.1 To each public data network (PDN) within a country;

2.2.1.2 To non-zoned service, such as the Public Mobile Satellite System (see § 2.2.10);

2.2.1.3 To a public switched telephone network (PSTN) or to an ISDN for the purpose of making calls from DTEs connected to a PDN to DTEs connected to that PSTN or ISDN;

2.2.1.4 To a group of PDNs within a country, when permitted by national regulations;

2.2.1.5 To a group of private data networks connected to PDNs within a country, where permitted by national regulations.

Note – For administrative purposes, including charging, a group of networks which have been assigned a single DNIC, will, in the international context, be considered as a single entity.

2.2.2 In the system of data network identification codes, the first digit of such codes should be in accordance with Table 1/X.121.

TABLE 1/X.121

First digit of data network identification code

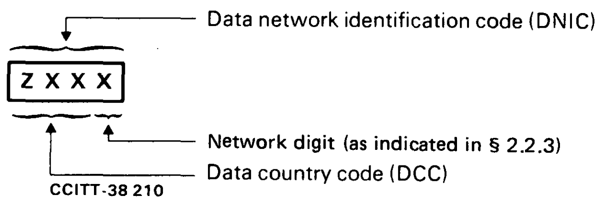
1	Reserved (see also § 2.2.10)
2 3 4 5 6 7	For data network identification codes (DNICs)

Note 1 – The allocation of codes for non-zoned services, other than the mobile satellite systems, is for further study.

Note 2 – Digits 8, 9 and 0 are used as escape codes, not being part of the DNIC. They are defined in § 2.6.

2.2.3 All data network identification codes (DNICs) should consist of four digits. The first three digits should always identify a country and could be regarded as a Data Country Code (DCC). The fourth, or network digit, should identify a specific data network in the country.

2.2.4 Each country should be assigned at least one 3-digit data country code (DCC). The data country code (DCC) in conjunction with the fourth digit can identify up to 10 public data networks. The format for data network identification codes (DNIC) should be as indicated in Figure 1/X.121.



X denotes any digit from 0 through 9

Z denotes any digit from 2 through 7 as indicated in § 2.2.2

FIGURE 1/X.121

Format for data network identification codes (DNIC)

2.2.5 The system of data network identification codes (DNIC) indicated in §§ 2.2.2 and 2.2.4 above will provide for 600 data country codes (DCC) and a theoretical maximum of 6000 DNIC.

2.2.6 In the case where a country requires more than 10 DNICs, additional data country codes (DCCs) could be assigned to the country (see § 2.2.8).

2.2.7 A list of data country codes (DCC) to be used in the development of data network identification codes (DNIC) is given in Annex D to this Recommendation. This list was prepared in accordance with the requirement that the first digit of a DNIC, which is also the first digit of the embedded data country code (DCC), should be restricted to the digits 2-7 inclusive (see § 2.2.2 above). As first digits of data country codes (DCC), the digits 2-7 are arranged to represent world zones.

2.2.8 The assignment of data country codes (DCC) is to be administered by the CCITT. The assignment of network digits will be made nationally and the CCITT Secretariat notified.

The Member countries of the International Telecommunication Union not mentioned in this list who wish to take part in the international data service or those Members who require an additional data country code(s) (DCC) should ask the Director of the CCITT for the assignment of an available 3-digit data country code(s) (DCC). In their request, they may indicate the available 3-digit code(s) preferred.

Assignments by the Director of the CCITT of data country codes (DCC) as well as assignments by countries of the network digits will be published in the Operational Bulletin of the International Telecommunication Union.

2.2.9 Examples indicating how data network identification codes (DNICs) could be developed, are given in Annex A to this Recommendation.

2.2.10 *International data number for stations in the Public Mobile Satellite Systems*

The DNICs allocated to Public Mobile Satellite Systems are 111S where the digit S indicates the ocean area. The digit S has the values as shown in Annex C.

The mobile station is identified by a unique mobile earth station number (INMARSAT mobile number) common for telephony, telex, data transmission and other services as defined in Recommendation E.215/F.125. The first digit of the mobile earth station number (INMARSAT mobile number) is the digit "T" defined in Recommendation E.215/F.126 and is used for discrimination between different Public Mobile Satellite Systems (such as the INMARSAT Standard A, B and C and aeronautical systems).

The complete international data number for mobile earth stations is composed as follows:

111S + mobile earth station number + X

where X is an optional digit which, if present, designates a particular DTE associated with the mobile earth station.

Note 1 – In the INMARSAT Mobile Satellite Systems, the use of the S digit for indicating the ocean area in which the mobile earth station is located at the time of the call is considered a temporary arrangement. It is recognized that such an arrangement should be avoided in the future, if possible, since it requires the calling user to know the exact area of a destination mobile earth station at the time of the call, and such an area may change from time to time for the mobile earth station.

Note 2 – Digit "X" requires further studies regarding aeronautical and land mobile earth stations.

2.3 *International data number*

2.3.1 A data terminal on a public data network when called from another country should be addressed by the international data number assigned to its DTE/DCE interface. The international data number should consist of the data network identification code (DNIC) of the called public data network, followed by the network terminal number (NTN) of the called DTE/DCE interface, or, for example, where an integrated numbering scheme exists within a country, the data country code (DCC) followed by the National Number (NN) of the called DTE/DCE interface, i.e.:

International data number = DNIC + NTN, or, DCC + NN

2.3.2 The Network Terminal Number (NTN) should consist of the full address that is used when calling the data terminal from within its serving public data network. The national number (NN) should consist of the full address used when calling the data terminal from another terminal within the national integrated numbering scheme. These numbers should consist of all the digits necessary to uniquely identify the corresponding DTE/DCE interface within the serving network and should not include any prefix (or access code) that might be employed for such calling.

Note 1 – Network terminal numbers (NTN) or national numbers (NN) may be assigned by a PDN to DTEs connected to other public networks, when interworking capabilities are provided with that PDN.

Note 2 – An example of the development of NTNs where a DNIC is assigned to a group of public or private data networks connected to PDNs within a country, is shown in Annex B.

2.4 *Number of digits*

2.4.1 International data numbers could be of different lengths but should consist of at least 5 digits but not more than 14 digits.

With the data network identification code (DNIC) fixed at 4 digits and the data country code (DCC) fixed at 3 digits, it would, therefore, be possible to have a network terminal number (NTN) of 10 digits maximum, or, a national number (NN) of 11 digits maximum.

Note 1 – The limit of 14 digits specified above applies exclusively to the international data number information. Adequate register capacity should be made available at data switching exchanges to accommodate the above digits as well as any additional digits that might be introduced for signalling, or other purposes.

Note 2 – After time "T" (see Recommendation E.165) the maximum number of digits of the international ISDN number will be 15. The need of extending the maximum capacity of the X.121 data number is for further study.

2.5 *Prefixes*

2.5.1 A prefix is an indicator consisting of one or more digits, allowing the selection of different types of address formats. Prefixes are not part of the international X.121 format and are not signalled over internetwork or international boundaries.

2.5.2 To distinguish between different address formats within a public data network (e.g. national data number and international data number formats), a prefix would generally be required. Any such prefix does not form a part of the data number. Pending further study, the use and composition of such a prefix is a national matter. However, the possible need to accommodate such a prefix with regard to digit register capacity should be noted. It is also a national matter to decide on evaluation of prefixes, escape code, parts of the international data number of incoming path of entry for routing or other purposes.

Note – In the case of Recommendation X.25 access, a prefix indicating international data number format can only be one digit.

2.6 *Escape codes*

An escape code is an indicator consisting of one digit. It indicates that the following digits are a number from a different numbering plan.

An escape code when required has to be carried forward through the originating network and can be carried across internetwork and international boundaries.

Digits used for escape codes are the digits 8, 9 and 0. The allocation and their purpose are shown in Table 2/X.121. The escape codes are not part of the international data number but are part of the "international X.121 format" (see Figure 21/X.121).

TABLE 2/X.121

Allocation of escape codes

8	Indicates that the digits which follow are from the F.69 Numbering Plan
9	Indicates that the digits which follow are from the E.164 Numbering Plan (Notes 2, 3 and 4)
0	Indicates that the digits which follow are from the E.164 Numbering Plan (Notes 1, 3 and 4)

Note 1 – In this case, 0 is to indicate that a digital interface between the PDN and the destination network (ISDN or integrated ISDN/PSTN) is requested.

Note 2 – In this case, 9 is to indicate that an analogue interface on the destination network (PSTN or integrated ISDN/PSTN) is requested.

Note 3 – In the case of calls from a PSPDN to an integrated ISDN/PSTN which does not require a distinction between digital and analogue interfaces, only a single escape code (e.g. 9 or 0) may be required. However all PSPDNs interworking with ISDNs, PSTNs and integrated ISDN/PSTNs should also support both 9 and 0 escape codes when acting as an originating, transit or destination network.

Note 4 – In the context of this Recommendation, the E.163 numbering plan is assumed to be a sub-set of the E.164 numbering plan.

Note 5 – Escape codes may be replaced by signalling means after time "T" (for the definition of time "T", see Recommendation E.165).

2.7 *Number analysis – international calls between public data networks*

2.7.1 In the case of international calls between public data networks, provision should be made in originating countries to interpret the first three digits of the international data number. These digits constitute the data country code (DCC) component of the data network identification code (DNIC) and identify the terminal country. This information is required in the originating country for routing purposes.

2.7.2 In originating countries, it might also be necessary to interpret the fourth, or network digit of a DNIC and, if the originating network requires it, the first digit after the DNIC. Such interpretation would provide the identity of a specific network in a country where several public data networks are in service. This information might be required for billing purposes or for the selection of specific routes to called networks. An example of the requirement for interpretation of the fifth digit is the use of this digit in the Mobile Satellite Systems for selection of a particular mobile system (digit "T", see § 2.2.10 above).

Note 1 – With regard to number analysis and routing in the case of interworking with PSTN and ISDN, see Recommendations X.110 and X.122.

Note 2 – With regard to RPOA selection, see § 1.8 above.

2.7.3 Countries receiving international calls for public data networks should receive the complete international data number. However, where a country of destination indicates that it does not wish to receive the data country code (DCC) component of the DNIC, arrangements should be made to suppress the DCC.

2.7.4 For destination countries with more than ten public data networks, interpretation of the first three digits of the DNIC [i.e., the data country code (DCC)] would identify the group of networks within which the called network is included. Interpretation of the fourth, or network, digit of the DNIC would identify the called network in that group. Interpretation of the first three digits would also make it possible to verify that an incoming call has in fact reached the correct country.

2.7.5 In the case of destination countries where there are fewer than ten public data networks, the first three digits of the DNIC could provide the verification indicated in § 2.7.4 above. Interpretation of the fourth, or network, digit of the DNIC would identify the specific network being called.

2.7.6 In transit countries the complete international data number must be received. Interpretation of the first three digits would identify the called country. Interpretation of the fourth or network digit would identify a specific data network in the called country. Interpretation of the fourth digit might be required for billing purposes or for route selection beyond the transit country. It might also be necessary in the transit network to analyse the fifth digit to allow selection of a particular public mobile system (e.g. digit "T", see § 2.2.10 above).

2.7.7 Where a data call is to be routed beyond a transit country through a second transit country, the complete international data number should always be sent to the second transit country. Where the data call is to be routed by a transit country to the country of destination, the arrangements indicated in § 2.7.3 above should apply.

2.8 *Numbering plan interworking*

Details on numbering plan interworking are outlined in Recommendation X.122 (see also Recommendations E.165, E.166, X.301 and I.330).

Transit cases are considered in these Recommendations. For routing aspects see also Recommendation X.110.

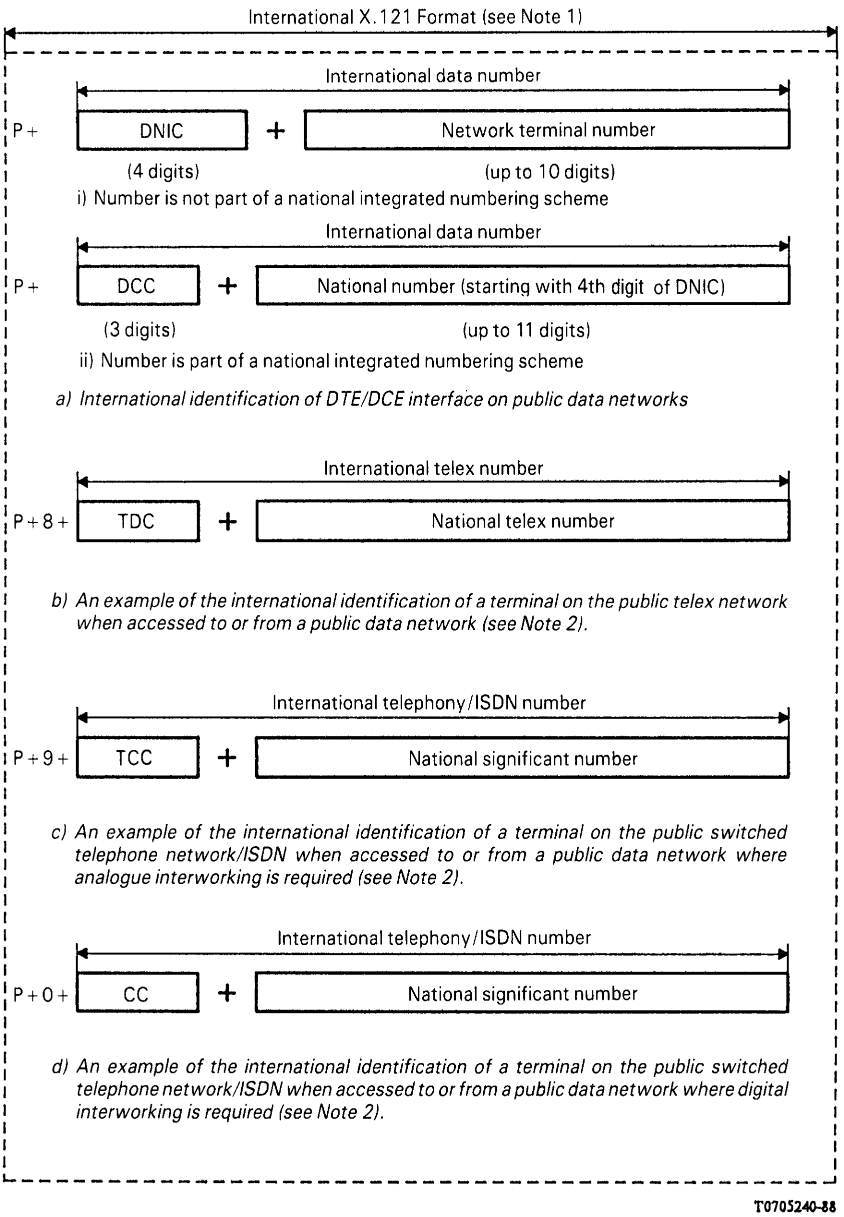
2.9 *Directories and letterheads*

2.9.1 Directories for public data networks should include information on the procedures to be followed for making international data calls. A diagram, such as that of Figure 2/X.121, could assist the customer in these procedures.

2.9.2 With regard to the prefix shown in Figure 2/X.121, it should be noted that the same prefix (designated P) could be used for all four types of calls. The choice of prefix is, however, a national matter.

2.9.3 With regard to RPOA selection (see § 1.8 above), it should be noted that an RPOA facility request designator would be used either in international data calls or within certain countries. Provision of this facility as well as the designation of the RPOA facility selection designator is a national matter in the originating country.

2.9.4 With regard to the publication of international data numbers on letterheads or other written material, it is recommended that the network terminal number (NTN) or national number (NN) should be easily distinguished within the international number, i.e. that there be a space between the 4-digit DNIC and the network terminal number (NTN) or, between the 3-digit data country code (DCC) and the national number (NN), where the fourth digit of the DNIC is included in the national number (NN).



a) International identification of DTE/DCE interface on public data networks

b) An example of the international identification of a terminal on the public telex network when accessed to or from a public data network (see Note 2).

c) An example of the international identification of a terminal on the public switched telephone network/ISDN when accessed to or from a public data network where analogue interworking is required (see Note 2).

d) An example of the international identification of a terminal on the public switched telephone network/ISDN when accessed to or from a public data network where digital interworking is required (see Note 2).

P	Prefix
DNIC	Data network identification code
DCC	Data country code
TCC	Telephone country code
TDC	Telex destination code
CC	Country code as defined by Recommendation E. 163
NDC	National destination code

Note 1 – The term "International X.121 Format" refers to the formats included within the dotted lines and excludes prefixes.

Note 2 – This illustrates the case where the data terminal on the public telephone or telex networks or on the ISDN is identified by the telephony/ISDN or telex number. Other cases are possible. The various interworking scenarios are described in separate Recommendations. It should also be noted that in the case of calls from a PSPDN to an integrated ISDN/PSTN which does not require a distinction between digital and analogue interfaces, only a single escape code (e.g. 9 or 0) may be required. However all PSPDNs interworking with ISDNs, PSTNs and integrated ISDN/PSTNs should also support both 9 and 0 escape codes when acting as an originating, transit or destination network.

FIGURE 2/X.121

International X.121 Format

ANNEX A

(to Recommendation X.121)

Development of data network identification codes (DNICs)

Example 1

In this example, it is assumed for illustrative purposes only, that the Netherlands has established its first public data network. To develop the data network identification code (DNIC) for this network, it would be necessary for the Netherlands to assign to it a network digit to follow the listed data country code (DCC) 204 (see Annex D). Assuming that the Netherlands selected the digit 0 as the network digit, the data network identification code (DNIC) for this initial network would be 2040.

Example 2

In this example, it is assumed for illustrative purposes only, that five public data networks have been established in Canada. To develop the data network identification codes for these networks, it would be necessary for Canada to assign to each of these networks a network digit to follow the listed data country code (DCC) 302 (See Annex D). Assuming that Canada assigned the network digits 0-4 to the five networks, the resulting data network identification codes (DNIC) would be 3020, 3021, 3022, 3023 and 3024.

Example 3

In this example, it is assumed for illustrative purposes only, that eight public data networks have been established in the United States of America. It is also assumed that network digits 0-7 would be assigned by the United States of America to follow the listed data country code (DCC) 310 (see Annex D). The data network identification codes (DNIC) thus formed for these eight networks would be 3100, 3101, 3102, 3103, 3104, 3105, 3106 and 3107.

If, some time later, four additional public data networks were to be introduced in the United States of America, two of the four new networks could be assigned network digits 8 and 9 in association with data country code (DCC) 310, to produce the data network identification codes (DNIC) 3108 and 3109.

For the remaining two public data networks, the United States of America would have to ask the CCITT for an additional data country code (DCC). A request for a code next in sequence, i.e. 311, could be made if this code appeared to be spare. If code 311 could be made available it would be assigned to the United States of America. If it was not available, a spare code in the "300" series of data country codes (DCC) would be assigned. Assuming data country code (DCC) 311 was available and issued to the United States of America, the two remaining public data networks could be assigned network digits 0 and 1 in association with data country code (DCC) 311, to produce the data network identification codes (DNIC) 3110 and 3111.

The data network identification codes (DNIC) for the 12 public data networks would then be 3100, 3101, 3102, 3103, 3104, 3105, 3106, 3107, 3108, 3109, 3110 and 3111.

Example 4

In this example, it is assumed for illustrative purposes only, that a public data network is to be established in each of two Caribbean islands that are part of the group of islands known as the French Antilles. The islands concerned are Guadeloupe and Martinique.

To develop the data network identification codes (DNIC) for these public data networks, it is assumed that the French Administration would assign network digit 0 to the network in Guadeloupe and network digit 1 to the network in Martinique and associate these network digits with the listed data country code (DCC) 340 for the French Antilles (see Annex D). The data network identification codes (DNIC) thus formed would be 3400 for Guadeloupe and 3401 for Martinique.

This example indicates that the system of data network identification codes (DNIC) is appropriate for application to groups of islands or regions of a country since one data country code (DCC) could provide for up to ten public data networks dispersed over several islands or regions. At the same time such island or regional networks would be distinguishable from each other.

ANNEX B

(to Recommendation X.121)

Development of network terminal numbers NTNs where a data network identification code (DNIC) is assigned to a group of public data networks or to a group of private data networks connected to public data networks within a country

The following is a guideline for allocating within a country data numbers for DTE/DCE interfaces on private data networks which are in turn connected to public data networks where permitted by national regulations.

Note – In the context of this annex, a private data network identification code (PNIC) may also be used to identify a specific public data network in a group of public data networks that share a common DNIC.

B.1 A private data network identification code (PNIC) is assigned to each private data network contained within a group of private data networks identified by a specific DNIC. The private data network identification code (PNIC)-digits are the first digits of the NTN.

B.2 All private data network identification codes (PNICs) consist of up to six digits. The format for the private data network identification codes (PNICs) is as follows:

ZXXXXX Private data network identification code (PNIC)

Z denotes any digit from 2 through 9 as indicated in § B.3 (See Note in Table B-1/X.121.)

X denotes any digit 0 through 9.

B.3 In the system of private data network identification codes (PNICs), the first digit of such codes is in accordance with Table B-1/X.121.

TABLE B-1/X.121

First digit of private data network identification code

0 1	See Note
2 3 4 5 6 7 8 9	For private data network identification codes (PNICs)

Note – The use of 0 or 1 depends on the national use of 0 or 1.

B.4 If a country has more private data networks than can be grouped under one DNIC or, if the public data networks within a country are not all interconnected, another DNIC may be allocated for each new group of private data networks.

B.5 If a private data network requires more numbers for DTE/DCE interfaces than can be grouped under one PNIC, multiple PNICs may be allocated to a single private data network.

B.6 The assignment of private data network identification codes (PNICs) is administered nationally.

ANNEX C

(to Recommendation X.121)

List of DNICs for non-zoned systems public mobile satelite sistems

Code	*Area*	*System*
1110	Spare	
1111	Atlantic Ocean	INMARSAT Mobile satellite data transmission system
1112	Pacific Ocean	INMARSAT Mobile satellite data transmission system
1113	Indian Ocean	INMARSAT Mobile satellite data transmission system
1114	Spare	
1115	Spare	
1116	Spare	
1117	Spare	
1118	Spare	
1119	Spare	

ANNEX D

(to Recommendation X.121)

List of data country or geographical area codes

Note – The countries or geographical areas shown in this Annex include those that already have code assignments in the case of other public telecommunication networks.

Zone 2

Code	*Country or Geographical Area*
202	Greece
204	Netherlands (Kingdom of the)
206	Belgium
208	France
212	Monaco
214	Spain
216	Hungarian People's Republic
218	German Democratic Republic
220	Yugoslavia (Socialist Federal Republic of)
222	Italy
226	Romania (Socialist Republic of)
228	Switzerland (Confederation of)
230	Czechoslovak Socialist Republic
232	Austria
234	United Kingdom of Great Britain and Northern Ireland
235	United Kingdom of Great Britain and Northern Ireland
236	United Kingdom
237	United Kingdom
238	Denmark
240	Sweden

Zone 2 *(cont.)*

Code	*Country or Geographical Area*
242	Norway
244	Finland
250	Union of Soviet Socialist Republics
260	Poland (People's Republic of)
262	Germany (Federal Republic of)
266	Gibraltar
268	Portugal
270	Luxembourg
272	Ireland
274	Iceland
276	Albania (Socialist People's Republic of)
278	Malta (Republic of)
280	Cyprus (Republic of)
284	Bulgaria (People's Republic of)
286	Turkey
288	Faroe Islands
290	Greenland
292	San Marino (Republic of)

Zone 2, Spare Codes: 62

Zone 3

Code	*Country or Geographical Area*
302	Canada
308	St. Pierre and Miquelon
310	United States of America
311	United States of America
312	United States of America
313	United States of America
314	United States of America
315	United States of America
316	United States of America
330	Puerto Rico
332	Virgin Islands (USA)
334	Mexico
338	Jamaica
340	French Antilles
342	Barbados
344	Antigua and Barbuda
346	Cayman Islands
348	British Virgin Islands
350	Bermuda
352	Grenada
354	Montserrat
356	St. Kitts
358	St. Lucia
360	St. Vincent and the Grenadines
362	Netherlands Antilles
364	Bahamas (Commonwealth of the)
366	Dominica
368	Cuba
370	Dominican Republic
372	Haiti (Republic of)
374	Trinidad and Tobago
376	Turks and Calcos Islands

Zone 3, Spare Codes: 68

Zone 4

Code	*Country or Geographical Area*
404	India (Republic of)
410	Pakistan (Islamic Republic of)
412	Afghanistan (Democratic Republic of)
413	Sri Lanka (Democratic Socialist Republic of)
414	Burma (Socialist Republic of the Union of)
415	Lebanon
416	Jordan (Hashemite Kingdom of)
417	Syrian Arab Republic
418	Iraq (Republic of)
419	Kuwait (State of)
420	Saudi Arabia (Kingdom of)
421	Yemen Arab Republic
422	Oman (Sultanate of)
423	Yemen (People's Democratic Republic of)
424	United Arab Emirates
425	Israel (State of)
426	Bahrain (State of)
427	Qatar (State of)
428	Mongolian People's Republic
429	Nepal
430	United Arab Emirates (Abu Dhabi)
431	United Arab Emirates (Dubai)
432	Iran (Islamic Republic of)
440	Japan
441	Japan
450	Korea (Republic of)
452	Viet Nam (Socialist Republic of)
454	Hong Kong
455	Macao
456	Democratic Kampuchea
457	Lao People's Democratic Republic
460	China (People's Republic of)
467	Democratic People's Republic of Korea
470	Bangladesh (People's Republic of)
472	Maldives (Republic of)

Zone 4, Spare Codes: 65

Zone 5

Code	*Country or Geographical Area*
502	Malaysia
505	Australia
510	Indonesia (Republic of)
515	Philippines (Republic of the)
520	Thailand
525	Singapore (Republic of)
528	Brunei Darussalam
530	New Zealand
535	Guam
536	Nauru (Republic of)
537	Papua New Guinea
539	Tonga (Kingdom of)
540	Solomon Islands
541	Vanuatu (Republic of)
542	Fiji

Zone 5 *(cont.)*

Code	*Country or Geographical Area*
543	Wallis and Futuna Islands
544	American Samoa
545	Kiribati (Republic of)
546	New Caledonia and Dependencies
547	French Polynesia
548	Cook Islands
549	Western Samoa

Zone 5, Spare Codes: 78

Zone 6

Code	*Country or Geographical Area*
602	Egypt (Arab Republic of)
603	Algeria (People's Democratic Republic of)
604	Morocco (Kingdom of)
605	Tunisia
606	Libya (Socialist People's Libyan Arab Jamahiriya)
607	Gambia (Republic of the)
608	Senegal (Republic of)
609	Mauritania (Islamic Republic of)
610	Mali (Republic of)
611	Guinea (Republic of)
612	Côte d'Ivoire (Republic of)
613	Burkina Faso
614	Niger (Republic of the)
615	Togolese Republic
616	Benin (People's Republic of)
617	Mauritius
618	Liberia (Republic of)
619	Sierra Leone
620	Ghana
621	Nigeria (Federal Republic of)
622	Chad (Republic of the)
623	Central African Republic
624	Cameroon (Republic of)
625	Cape Verde (Republic of)
626	Sao Tome and Principe (Democratic Republic of)
627	Equatorial Guinea (Republic of)
628	Gabonese Republic
629	Congo (People's Republic of the)
630	Zaire (Republic of)
631	Angola (People's Republic of)
632	Guinea-Bissau (Republic of)
633	Seychelles
634	Sudan (Republic of the)
635	Rwandese (Republic of)
636	Ethiopia
637	Somali Democratic Republic
638	Djibouti (Republic of)
639	Kenya (Republic of)
640	Tanzania (United Republic of)
641	Uganda (Republic of)
642	Burundi (Republic of)
643	Mozambique (People's Republic of)
645	Zambia (Republic of)
646	Madagascar (Democratic Republic of)
647	Reunion (French Department of)

Zone 6 *(cont.)*

Code	*Country or Geographical Area*
648	Zimbabwe (Republic of)
649	Namibia
650	Malawi
651	Lesotho (Kingdom of)
652	Botswana (Republic of)
653	Swaziland (Kingdom of)
654	Comoros (Islamic Federal Republic of the)
655	South Africa (Republic of)

Zone 6, Spare Codes: 47

Zone 7

Code	*Country or Geographical Area*
702	Belize
704	Guatemala (Republic of)
706	El Salvador (Republic of)
708	Honduras (Republic of)
710	Nicaragua
712	Costa Rica
714	Panama (Republic of)
716	Peru
722	Argentine Republic
724	Brazil (Federative Republic of)
730	Chile
732	Colombia (Republic of)
734	Venezuela (Republic of)
736	Bolivia (Republic of)
738	Guyana
740	Ecuador
742	Guiana (French Department of)
744	Paraguay (Republic of)
746	Suriname (Republic of)
748	Uruguay (Eastern Republic of)

Zone 7, Spare Codes: 80

ANNEX E

(to Recommendation X.121)

Definitions regarding the international numbering plan for public data networks

E.1 data country code

In the context of the international numbering plan for public data networks, a component of the international X.121 format consisting of three digits allocated by CCITT and published in Recommendation X.121.

E.2 data network identification code (DNIC)

In the context of the international numbering plan for public data networks, a component of the international X.121 format consisting of four digits. The first three digits are regarded as the data country code (DCC), the fourth digit identifies a network in that country according to Recommendation X.121.

Note 1 – The digit allocated by countries to establish, together with the data country code the data network identification code, should be notified to the CCITT Secretariat.

Note 2 – CCITT publishes a list of data network identification codes.

E.3 escape code

In the context of the international numbering plan for public data networks, an indicator consisting of one digit which indicates that the following digits are a number from a different numbering plan.

Note – An escape code is part of the international X.121 format.

E.4 international data number

In the context of the international numbering plan for public data networks, the address information comprising the data country code (DCC) and the national number, or the data network identification code (DNIC) and the (national) network terminal number, according to Recommendation X.121.

E.5 international data number format

In the context of the international numbering plan for public data networks, a numbering plan format comprising of the digits of the international data number, according to Recommendation X.121.

E.6 international X.121 format

In the context of the international numbering plan for public data networks, a format consisting of digits which are to be transferred across international boundaries, according to Recommendation X.121.

Note 1 – See also "international data number format".

Note 2 – Escape codes, if required, are part of the international X.121 format and are allowed by digits of another international numbering plan.

Note 3 – Prefixes do not belong to the international X.121 format.

E.7 numbering plan

In the context of the international numbering plan for public data networks, the specification given in Recommendation X.121.

Note – Other international numbering plans are contained in Recommendations E.163, E.164 and F.69.

E.8 numbering plan interworking

In the context of the international numbering plan for public data networks, the methods to establish interworking between networks applying different international numbering plans.

Note – Examples of numbering plan interworking are given in Recommendations X.122, E.166 and I.332.

E.9 prefix

In the context of the international numbering plan for public data networks, an indicator consisting of one or more digits, allowing the selection of different numbering formats. Prefixes are not part of the international X.121 format.

Note – Prefixes are a national matter.

NUMBERING PLAN INTERWORKING BETWEEN A PACKET SWITCHED PUBLIC DATA NETWORK (PSPDN) AND AN INTEGRATED SERVICES DIGITAL NETWORK (ISDN) OR PUBLIC SWITCHED TELEPHONE NETWORK (PSTN) IN THE SHORT-TERM

(Melbourne, 1988)

The CCITT,

considering

(a) that Recommendation X.121 defines the international numbering plan for Public Data Networks (PDNs);

(b) that Recommendation E.164/I.331 defines the numbering plan for the ISDN era;

(c) that Recommendation X.300 defines the general principles and arrangements for interworking between Public Networks, and between Public Networks and other networks for the provision or data transmission services;

(d) that Recommendation I.330 defines ISDN numbering and addressing principles;

(e) that Recommendation I.332 defines the numbering principles for interworking between ISDNs and dedicated networks with different numbering plans;

(f) that Recommendation E.166 defines numbering and addressing interworking in the ISDN era;

(g) that Recommendation E.165 defines the timetable for coordinated implementation of the full capability of Recommendation E.164;

(h) the Recommendation X.110 defines the international routing principles and routing plan for Public Data Networks (PDNs);

(i) that Recommendation X.25 defines the interface between DTE and DCE for terminals operating in the packet mode on PDNs and connected to PDNs by dedicated circuit;

(j) that Recommendation X.32 defines the interface between DTE and DCE for terminals operating in the packet mode and accessing a PSPDN through a PSTN or a Circuit Switched Public Data Network (CSPDN);

(k) that Recommendation X.31 defines the support of packet mode terminal equipment by an ISDN;

(l) that Recommendation X.75 defines the packet switched signalling system between public networks providing data transmission services;

(m) that the need that DTEs can communicate through different networks with different numbering plans for the purpose of interworking CCITT defined services;

(unanimously) declares the view

that numbering plan interworking between PSPDNs and ISDNs or PSTNs in the short term be in accordance with the procedures specified in this Recommendation.

CONTENTS

1 Introduction

Numbering plan interworking is a fundamental requirement for successful completion of call connections between networks with different numbering plans, e.g., from a PDN to an ISDN.

This Recommendation defines the general procedures applicable to PSPDNs up to the time "T" as specified in Recommendation E.165. In particular, this Recommendation describes the interworking scenarios for PSPDNs with ISDNs or PSTNs for a short-term. Recommendation E.166 deals with interworking scenarios for ISDNs with other networks.

2 Scope

2.1 The scope of this Recommendation is to define the procedures applicable to PSPDNs for the purpose of numbering plan interworking with networks which use the E.164 (E.163) numbering plan. In this Recommendation it is assumed that E.163 is a subset of E.164. In the short-term the length of the E.164 numbers are restricted to 12 digits as per Recommendation E.165.

2.2 This Recommendation applies to numbering plan interworking across international boundaries. Its applicability to calls within a single country is a national matter.

2.3 The procedures specified in this Recommendation may not apply after time "T". Time "T" is defined in Recommendation E.165.

2.4 Interworking involving more than two networks is included in the scope of this Recommendation.

2.5 Procedures at the man machine interface and the DCE/DTE interface are outside the scope of this Recommendation.

3 Procedures

3.1 *Use of standardized escape codes from the X.121 numbering plan*

3.1.1 *General*

The terms "prefix" and "escape code" are defined in §§ 2.5 and 2.6 of Recommendation X.121. The values of the escape codes for use by PDNs should be in accordance with Table 2/X.121 of Recommendation X.121.

Two escape codes from X.121 to E.164 numbering plan have been allocated for the following reasons:

1) to provide routing criteria from PDN to PSTN and ISDN, and
2) to indicate the type of interface that has been requested.

Specific use of escape codes is described in the following sections.

The use of prefix digits is outside the scope of this Recommendation.

3.1.2 *Escape code 0*

Escape code 0 is used to escape from the X.121 numbering plan to the E.164 numbering plan. It indicates that a digital interface between the PSPDN and the destination network (ISDN or integrated ISDN/PSTN) is requested (see also § 3.1.4).

An Optional Network Specific Digit(s) (ONSD) that represents the same functionality as the internationally agreed value 0 may be used to avoid any conflict with existing use of 0 within a network. When an ONSD is used, the translation of that digit into the internationally agreed value 0 at the network boundary is the responsibility of the originating (or transit) network.

Other number formats may be used when interworking is not across an international boundary.

3.1.3 *Escape code 9*

Escape code 9 is used to escape from the X.121 numbering plan to the E.163 or E.164 numbering plan. It indicates that an analogue interface on the destination network (PSTN or integrated ISDN/PSTN) is requested (modem required) (see also § 3.1.4).

An Optional Network Specific Digit(s) (ONSD) that represents the same functionality as the internationally agreed value 9 may be used to avoid any conflict with existing use of 9 within a network. When an ONSD is used, the translation of that digit into the internationally agreed value 9, at the network boundary is the responsibility of the originating (or transit) network.

Other number formats may be used when interworking is not across an international boundary.

3.1.4 It is recognized that special situations can exist where calls from a PSPDN to an integrated ISDN/PSTN which does not require a distinction between digital and analogue interfaces, then only a single escape code (i.e. 9 or 0) is required. However, all PSPDNs interworking with ISDNs, PSTNs and ISDN/PSTNs should support both 9 and 0 escape codes when acting as an originating, transit or destination network.

3.1.5 Alternatively, in some countries (or RPOAs) a DNIC may be allocated to an ISDN, as it is now applicable for PSTN. The use of a DNIC for that purpose is the decision of the country (or RPOAs). In this case, the PDN using a DNIC to identify terminals on an interconnected ISDN should be capable of generating escape codes 9 and/or 0 for escaping to an ISDN/PSTN or PSTN that does not use the DNIC solution. The translation of the X.121 number to an E.164 number at the destination is permitted on a national basis. As far as the international subscriber is concerned, the called terminal has an X.121 number and the conversion, if required, is done in the destination country.

3.2 *General description on the use of escape codes for PSPDNs to ISDN/PSTN or PSTN interworking*

3.2.1 The DTE enters the number of the ISDN or PSTN terminal to be called, preceded by an appropriate escape code N (see §§ 3.1.2 and 3.1.3). The full called number, N + E.164, is sent from the DTE to the PDN for routing to the correct network interface.

3.2.2 In some networks, an Optional Network Specific Digit(s) (ONSD) may be used, which represents the same functionality as the internationally agreed value as described in §§ 3.1.2 and 3.1.3. In this case an ONSD can be used in place of N. The ONSD must be translated into N before it crosses an international boundary.

3.3 *Digit analysis*

3.3.1 For routing implications, refer to Recommendation X.110.

3.3.2 The number analysis capability of a PSPDN for interworking with ISDN, PSTN should be five digits. The need to analyze more than five digits is for further study.

4 **Short-term interworking scenarios**

4.1 *General*

The short term in this Recommendation means the time period up to time "T" specified in Recommendation E.165.

The following figures illustrate examples of short-term numbering plan interworking scenarios. The scenarios are not exhaustive.

Note to figures

The presence and exact format of the called and calling addresses at the DTE/DCE interface are network dependent, including possible use of prefixes.

The diagrams show only number flows carried in the call request packet and do not indicate any numbering arrangement necessary for call set up within the ISDN or PSTN. For details of call set up arrangements refer to Recommendations X.31, X.32 and the X.300-series.

In this Recommendation when an X.31 terminal is being called using an E.164 number, it is recognized that digital interworking is required (E.164 digital) and when an X.32 terminal is being called using an E.164 number, analogue interworking is required (E.164 analogue).

Key to the figures

Arrows associate the number flows to that portion of the call.

The positioning of the interworking function (IWF) symbol, on the diagram, does not imply a fixed position within a network nor make any statement on the functionality of the IWF. The IWF is a logical representation of the gateway process which handles required protocol translations beween two dissimilar networks. The specific location or implementation of an IWF should be referred to the appropriate Recommendation.

In the following scenarios, the number referred to as "X.121" is "International Data Number" i.e., DNIC + NTN or DCC + NN as specified in § 2.3.1 of Recommendation X.121 and Figure 2/X.121.

Abbreviations used are explained in Annex A.

4.2 *Short-term numbering plan interworking between two dissimilar networks*

Table 1/X.122 introduces the possible numbering plan interworking scenarios. Reference to a specific Recommendation is for the case where its procedures are directly applicable. The other numbers in the table indicate the corresponding scenarios in this Recommendation.

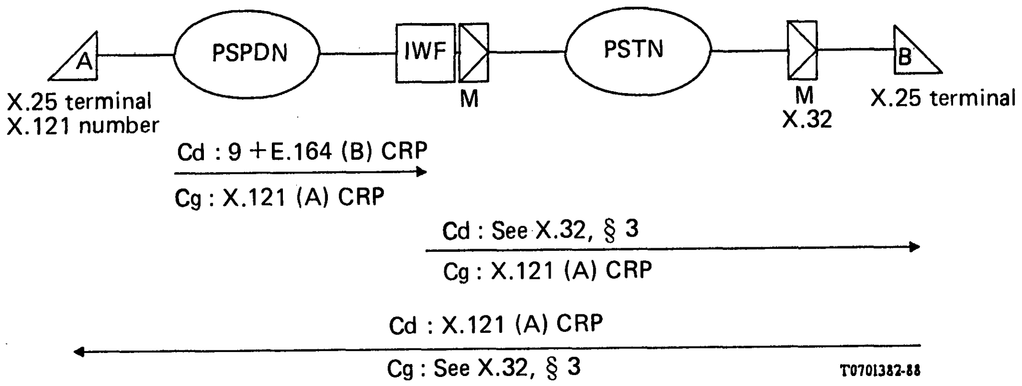
TABLEAU 1/X.122

Numbering plan interworking (short-term) between two networks

From/To	PSPDN	X.121			
	CSPDN	X.121	X.121		
	ISDN	§ 4.2.2	(Note)	E.164	
	PSTN	§ 4.2.1	(Note)	E.164	E.164
NETWORK / NETWORK		PSPDN	CSPDN	ISDN	PSTN
		To/From			

Note – Outside the scope of this Recommendation.

4.2.1 *PSPDN to/from PSTN (X.32)* (Figure 1/X.122)



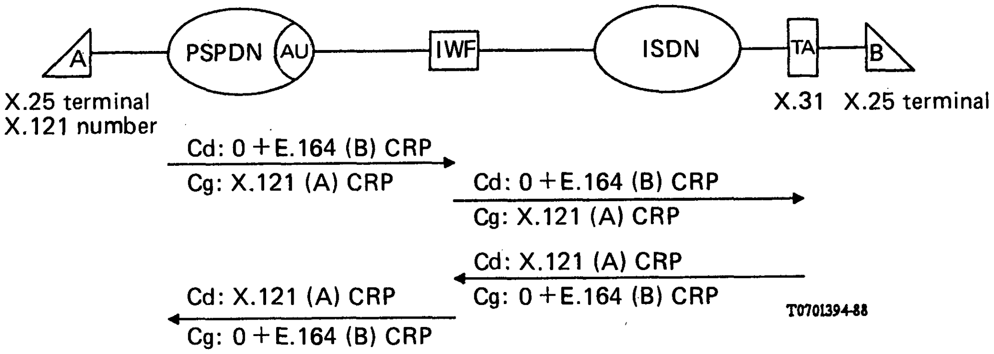
Note 1 – The use of a DNIC to identify a PSTN may be used as an alternative to the above scenario (see § 3.1.5)

Note 2 – This numbering interworking principle should also be applied to the character type DTE

FIGURE 1/X.122

4.2.2 *PSPDN to/from ISDN (X.31)*

4.2.2.1 Case A scenarios using permanent, or switched access from the Access Unit (AU) to the terminal connected to the ISDN when the ISDN terminal is not allocated an X.121 number (see Figure 2/X.122).

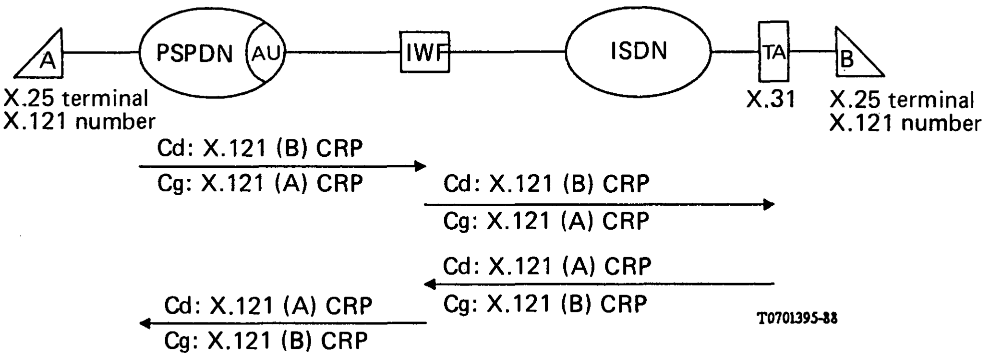


Note 1 – The use of a DNIC to identify an ISDN may be used as an alternative to the above scenario (see § 3.1.5).

Note 2 – The diagram shows only number flows carried in the call request packet and does not indicate any numbering arrangement necessary for call set up within the ISDN. For details of call set up arrangements refer to Recommendations X.31, X.32 and the X.300-series.

FIGURE 2/X.122

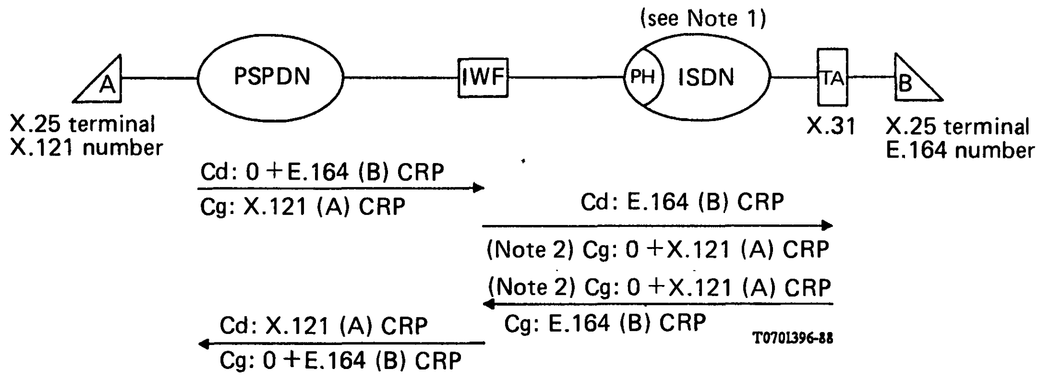
4.2.2.2 Case A scenarios using permanent or switched access from the access unit (AU) to the terminal in the ISDN when the ISDN terminal is allocated an X.121 number (see Figure 3/X.122).



Note – The diagram shows only number flows carried in the call request packet and does not indicate any numbering arrangement necessary for call set up within the ISDN. For details of call set up arrangements refer to Recommendations X.31, X.32 and the X.300-series.

FIGURE 3/X.122

4.2.2.3 Case B scenarios using permanent or switched access from the packet handler (PH) to the terminal in the ISDN (see Figure 4/X.122).



Note 1 – In some implementations the PH functions logically belonging to the ISDN may reside in a node of PSPDN (see Recommendation X.31 § 5.2)

Note 2 – The use of the Escape Code 0 in this instance is explained in E.166.

Note 3 – The use of a DNIC to identify an ISDN may be used as an alternative to the above scenario (see § 3.1.5).

Note 4 – The diagram shows only number flows carried in the call request packet and does not indicate any numbering arrangement necessary for call set up within the ISDN. For details of call set up arrangements refer to Recommendations X.31, X.32 and the X.300-series.

FIGURE 4/X.122

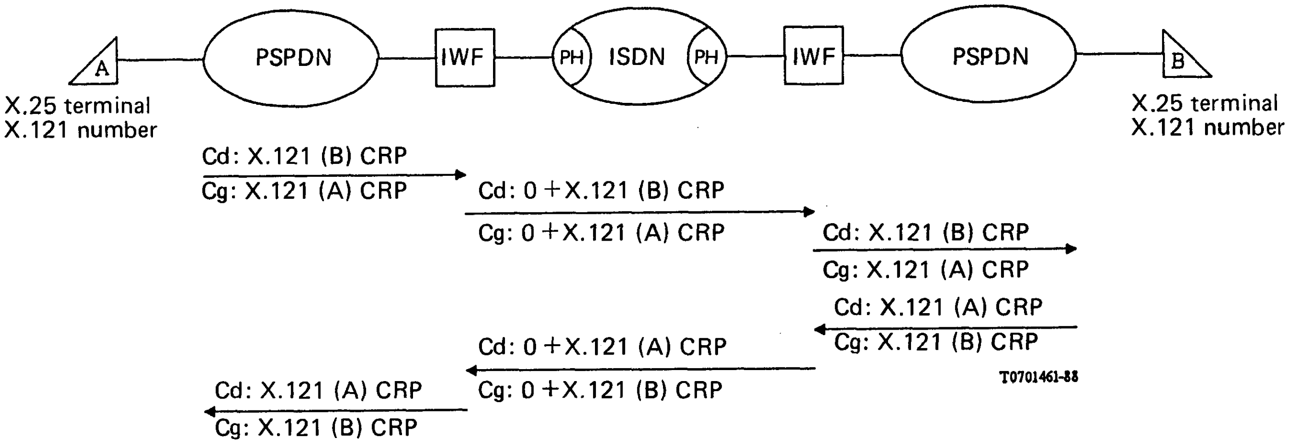
4.3 *Numbering plan interworking via an intermediate network – short term*

TABLE 2/X.122

Numbering plan interworking via an intermediate network – short-term

Scenarios	Interconnected networks
§ 4.3.1	PSPDN – ISDN – PSPDN
§ 4.3.2	ISDN – PSPDN – ISDN
§ 4.3.3	ISDN – PSPDN – ISDN
§ 4.3.4	ISDN – PSPDN – PSTN
§ 4.3.5	ISDN – PSPDN – ISDN
§ 4.3.6	ISDN – PSPDN – ISDN/PSTN

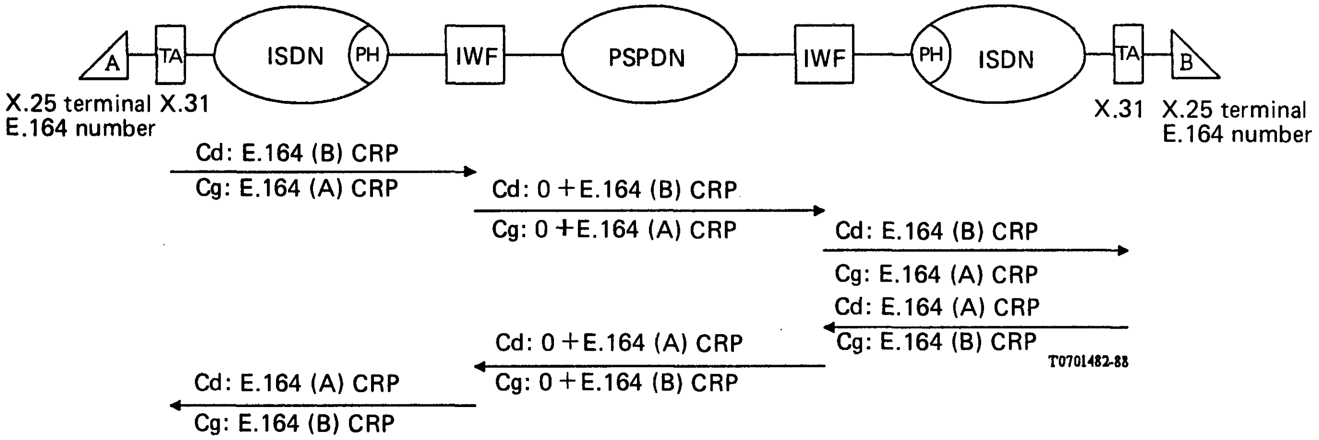
4.3.1 *PSPDN – ISDN – PSPDN* (Figure 5/X.122)



Note – The diagram shows only number flows carried in the call request packet and does not indicate any numbering arrangement necessary for call set up within the ISDN. For details of call set up arrangements refer to Recommendations X.31, X.32 and the X.300 - series.

FIGURE 5/X.122

4.3.2 *ISDN – PSPDN – ISDN for X.31 Case B to X.31 Case B* (Figure 6/X.122)

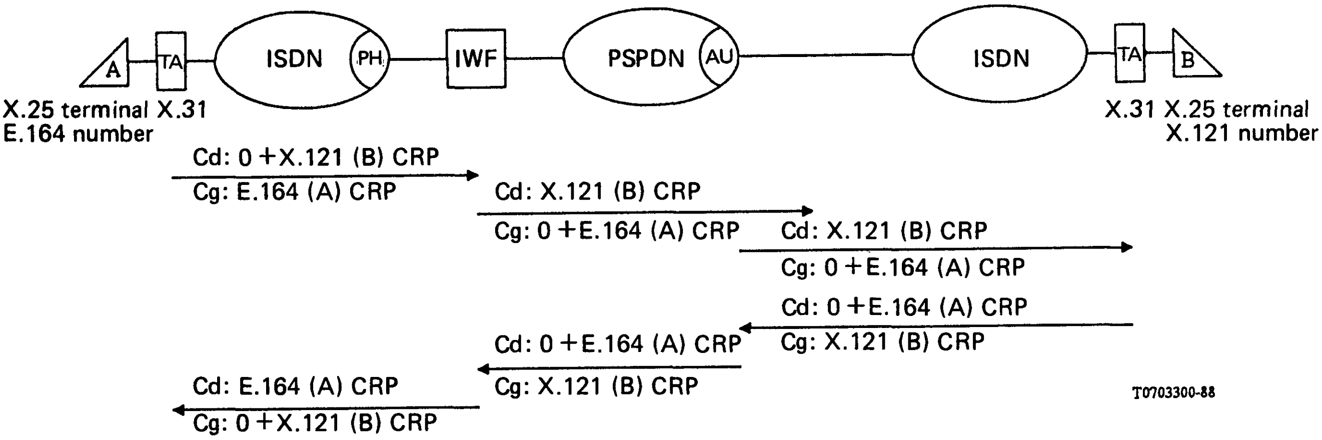


Note 1 – The use of a DNIC to identify an ISDN or PSTN may be used as an alternative to the above scenario (see § 3.1.5).

Note 2 – The diagram shows only number flows carried in the call request packet and does not indicate any numbering arrangement necessary for call set up within the ISDN. For details of call set up arrangements refer to Recommendations X.31, X.32 and the X.300-series.

FIGURE 6/X.122

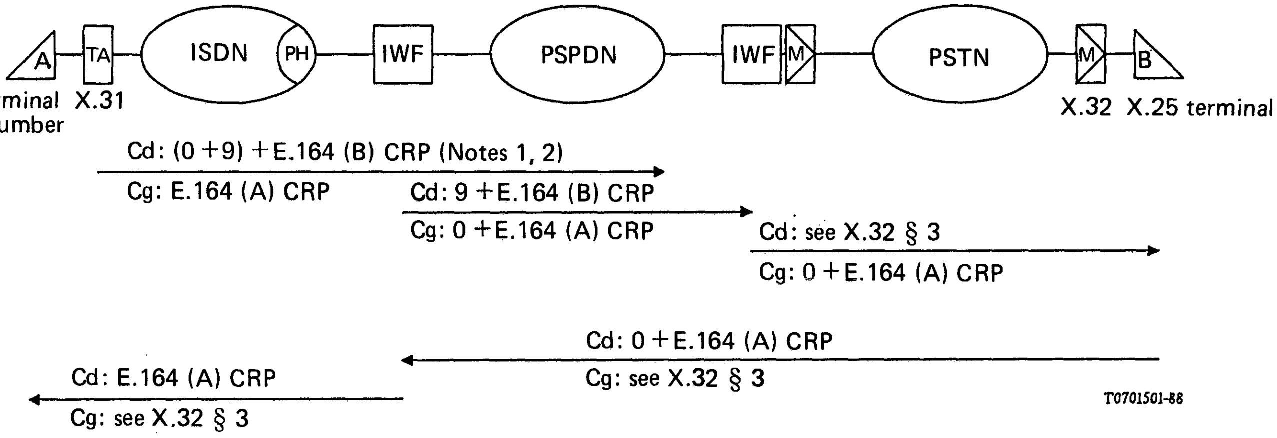
4.3.3 *ISDN – PSPDN – ISDN for X.31 Case B to X.31 Case A when the Case A terminal is allocated an X.121 number* (Figure 7/X.122)



Note – The diagram shows only number flows carried in the call request packet and does not indicate any numbering arrangement necessary for call set up within the ISDN. For details of call set up arrangements refer to Recommendations X.31, X.32 and the X.300-series.

FIGURE 7/X.122

4.3.4 *ISDN – PSPDN – PSTN* (Figure 8/X.122)



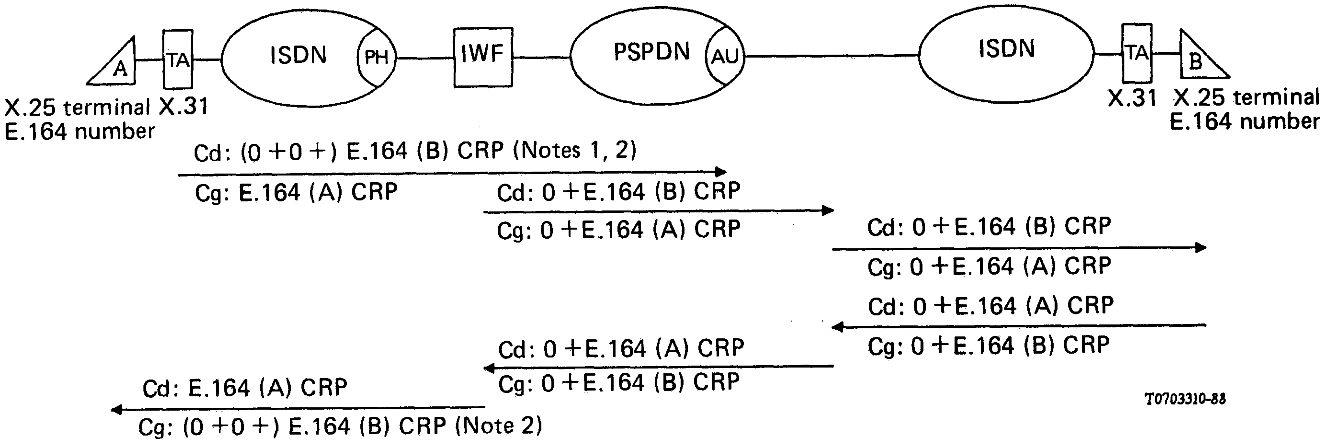
Note 1 – The use or non use of two contiguous escape codes 9 and 0 preceeding E.164 (B) is a matter of the NETWORK to which TERMINAL A is connected. The implementation of this scenario may depend on the national use of Prefixes.

Note 2 – The first escape code "0" is used to escape to the international X.121 format (see Figure 2/X.121)

Note 3 – The diagram shows only number flows carried in the call request packet and does not indicate any numbering arrangement necessary for call set up within the ISDN and PSTN. For details of call set up arrangements refer to Recommendations X.31, X.32 and the X.300 - series.

FIGURE 8/X.122

4.3.5 *ISDN – PSPDN – ISDN for X.31 Case B to X.31 Case A when the Case A terminal is not allocated an X.121 number* (Figure 9/X.122)



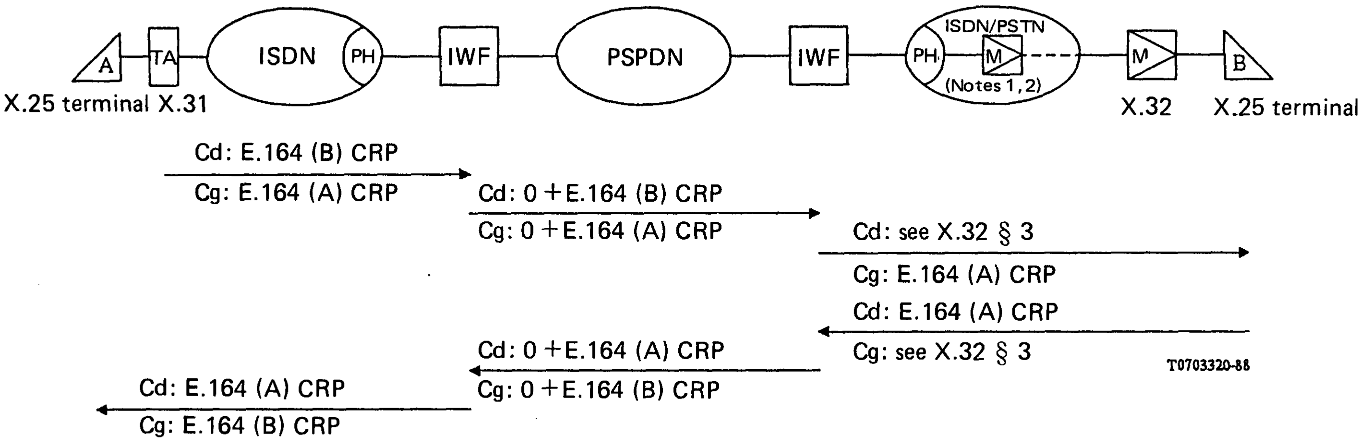
Note 1 – The use or non use of two contiguous escape code 0's preceeding E.164 (B) is a matter of the NETWORK to which TERMINAL A is connected. The implementation of this scenario may depend on the national use of Prefixes.

Note 2 – The first escape code "0" is used to escape to the international X.121 format (see Figure 2/X.121).

Note 3 – The diagram shows only number flows carried in the call request packet and does not indicate any numbering arrangement necessary for call set up within the ISDN and PSTN. For details of call set up arrangements refer to Recommendations X.31, X.32 and the X.300 - series.

FIGURE 9/X.122

4.3.6 *ISDN – PSPDN – ISDN/PSTN* (Figure 10/X.122)



Note 1 – The location of the modem is implementation dependent.

Note 2 – X.32 procedures between the packet handler in the destination network and terminal B are not fully defined.

Note 3 – The scenario may apply in the case of an integrated ISDN/PSTN, where X.32 functionality does not need to be indicated by an escape code within the destination ISDN/PSTN network.

FIGURE 10/X.122

ANNEX A

(to Recommendation X.122)

Abbreviations

A, B,	Designation of the terminals used in the scenarios
AU	access unit
Cd	called number
Cg	calling number
CRP	call request packet or equivalent
CSPDN	circuit switched public data network
DCE	data circuit terminating equipent
DNIC	data network identification code
DTE	data terminal equipment
ISDN	integrated service digital network
IWF	interworking function
M	modem
ONSD	optional network specific digit
PDN	public data network
PH	packet handler
PSPDN	packet switched public data network
PSTN	public switched telephone network
T	time "T" specified in Recommendation E.165

Recommendation X.130

CALL PROCESSING DELAYS IN PUBLIC DATA NETWORKS WHEN PROVIDING INTERNATIONAL SYNCHRONOUS CIRCUIT-SWITCHED DATA SERVICES

(Geneva, 1980; amended at Malaga-Torremolinos, 1984)

The CCITT,

considering

(a) that Recommendation X.1 specifies the user classes of service applicable to networks offering public data services;

(b) that Recommendation X.2 specifies the international user services and facilities to be offered by public data networks;

(c) that Recommendations X.21 and X.21 *bis* define the DTE/DCE interface for circuit switched services;

(d) that Recommendation X.60 specifies the common channel signalling for synchronous data networks;

(e) that Recommendation X.71 specifies the channel associated signalling for synchronous data networks;

(f) that Recommendation X.92 specifies the hypothetical reference connections for public data networks;

(g) that Recommendation X.110 specifies the routing plan to be applied in the international portions of public data networks;

(h) that Recommendation X.213 specifies the OSI Network Layer service;

(i) that Recommendation X.140 specifies the user-oriented quality of service parameters applicable to data services,

unanimously declares

that when public data networks provide international synchronous circuit-switched data services according to Recommendations X.21 and X.21 *bis*, the values of call processing delays specified in this Recommendation shall be taken as provisional worst-case values that should not be exceeded under the conditions specified therein.

Introductory note – Design objectives that take into account both user needs and network costs are for further study.

1 Introduction

1.1 Quality of service in circuit-switched public data networks has been considered in five basic areas as follows:

i) call processing delays (Recommendation X.130);

ii) failures due to congestion (blocking) (Recommendation X.131);

iii) failures due to malfunction;

iv) loss of service; and

v) transmission performance (including throughput).

This Recommendation specifies the objectives for i) above. Each of the other areas of circuit-switching quality of service identified above will be the subject of a separate Recommendation in the X-series.

1.2 In telecommunication networks it is necessary, for economic reasons, to limit the resources provided for carrying the offered traffic. This limitation may affect the quality of service to the user of circuit-switched services in two different ways: by call processing delays and by blocking. Both of these aspects, that are consequences of the finite traffic handling capacity of the network, constitute the grade of service. Grade of service together with malfunction, loss of service and transmission performance constitute the quality of service.

1.3 In this Recommendation the values for the network delay are quoted for two types of connection according to Recommendation X.92 as follows:

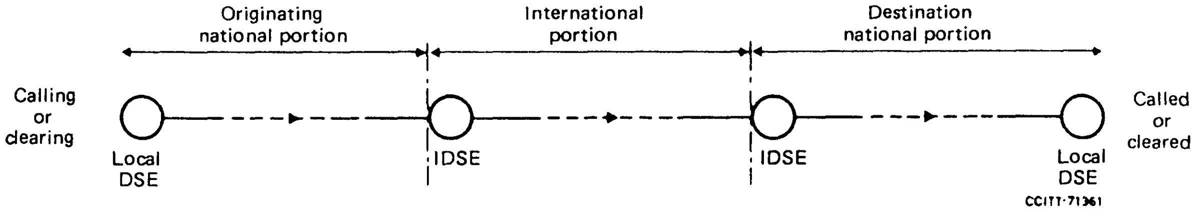
Type 1: Typical terrestrial international connection of moderate length with no satellite circuits either in the national or international portions (International portion: 1000 km).

Type 2: Long distance international connection with a satellite circuit in one national portion and two satellite circuits in the international portion (International portion: 160 000 km).

Where appropriate, values are also specified separately for the following network portions:

- originating national network,
- international portion,
- destination national network.

The boundaries for these portions are shown in Figure 1/X.130.



Note 1 – A DSE may also function as an IDSE.

Note 2 – Arrows indicate direction of call set-up or clear-down.

FIGURE 1/X.130

National/international boundaries for call set-up and clear-down functions

For the present, the values apply also to other normal routing options within the international portion.

Following the allocation of a delay allowance to the international portion of an international transit connection, it will be necessary to further apportion the allowance to individual transit networks and/or their component parts within the international portion. The means by which useful and realistic constraints can be applied, consistent with maintaining the maximum possible freedom for each involved Administration in the design and implementation of its own network, is for further study.

1.4 The values for call processing delays established in this Recommendation are to be considered as design objectives in network planning together with the forecast traffic for the planned period. The actual delay performance that will be obtained will depend on the accuracy of the traffic estimations. Normally, the actual delay performance will not coincide with the one used as a basis for planning. Furthermore, if the network is planned for the traffic forecast at the end of the period considered, the actual delay performance of the network may be better than the design value, worsening gradually to the end of the planning period as traffic increases.

The non-coincidence of busy hours in originating and destination national networks as well as in the international network will improve the overall delay performance with respect to the sum of the nominal delays of the constituent parts of the connection.

1.5 Delays are specified under conditions of normal busy hour load and are expressed where appropriate in terms of mean and 95% probability values. The term "mean" is taken to be the expected value of delay in the statistical sense. The "95% probability" value is taken as the limit within which 95% of the delays fall. Delays at higher loadings are for further study.

1.6 Call processing delays are defined for a basic call which does not include any optional user facilities, e.g. those defined in Recommendation X.21.

1.7 Where appropriate, separate limits are quoted for common channel signalling and channel associated signalling between DSEs.

For common channel signalling, the values given in this Recommendation are also applicable to lower signalling rates (less than 4800 bit/s), when the associated mode of operation is used.

1.8 The quality of service implications of regional or national satellite systems using demand assignment for resource allocation require further study.

2 Call connection delay

See Annex A for an explanation of the delay elements t1 to t6.

2.1 *Total call connection delay (TCCD)*

The **total call connection delay (TCCD)** is the time interval between the transmission of the *call request* signal and receipt of the *ready for data* signal by the calling DTE. A full explanation of the elements of TCCD is contained in Annex A. Objectives for the network-dependent components of TCCD are provided below.

2.2 *Call request delay (t1)*

Call request delay is considered to be a national matter and consequently the specification of its value is not appropriate to this Recommendation.

2.3 *Overall network post selection delay*

Overall network post selection delay is the sum of t3 and t5. It should not exceed the values given in Tables 1/X.130 and 2/X.130.

If on any call the overall network post selection delay exceeds X seconds, the call will be considered for quality of service purposes to have failed. The precise value of X is for further study but it should be at least 30 seconds.

2.4 *Network portion post selection delays (t3 + t5)*

The contribution from each network portion to the overall network post selection delay should not exceed the values given in Tables 3/X.130 and 4/X.130.

2.5 *Ready for data delay (t6)*

The need for specification of this parameter is for further study.

3 Call clearing delays

3.1 *Clear request delay (CLRD)*

Clear request delay (CLRD) is the delay between transmission of a *clear request* signal and receipt of the *DCE ready* signal by the clearing DTE. Clear request delay is considered to be a national matter and consequently the specification of its value is not appropriate to this Recommendation.

TABLE 1/X.130

Overall network post selection delay for common-channel signalling

User rate (bit/s)	Statistic	Delay (ms)	
		Connection type	
		1	2
600	Mean	1800	3500
	95%	2700	4400
2 400	Mean	1500	3200
	95%	2200	3900
4 800	Mean	1300	3000
	95%	1900	3600
9 600	Mean	1300	3000
	95%	1900	3600
48 000	Mean	1300	3000
	95%	1900	3600

Note – See introductory note to this Recommendation.

TABLE 2/X.130

Overall network post selection delay for channel-associated signalling

User rate (bit/s)	Statistic	Delay (ms)	
		Connection type	
		1	2
600	Mean	2200	4000
	95%	3300	5100
2 400	Mean	1800	3600
	95%	2700	4500
4 800	Mean	1700	3500
	95%	2500	4400
9 600	Mean	1600	3400
	95%	2400	4200
48 000	Mean	1500	3300
	95%	2200	4100

Note – See introductory note to this Recommendation.

TABLE 3/X.130

Contributions to network post selection delays for common-channel signalling

User rate (bit/s)	Statistic	Originating national portion (ms)		Destination national portion (ms)		International portion (ms)	
		Number of satellites		Number of satellites		Connection type	
		0	1	0	1	1	2
600	Mean	700	1200	800	1300	300	1500
	95%	1100	1600	1200	1800	500	1700
2 400	Mean	600	1100	700	1200	200	1400
	95%	900	1500	1100	1600	300	1600
4 800	Mean	500	1000	600	1100	200	1400
	95%	800	1300	900	1500	300	1600
9 600	Mean	500	1000	600	1100	200	1400
	95%	800	1300	900	1500	300	1600
48 000	Mean	500	1000	600	1100	200	1400
	95%	800	1300	900	1500	300	1600

Note – See introductory note to this Recommendation.

TABLE 4/X.130

Contributions to network post selection delays for channel-associated signalling

User rate (bit/s)	Statistic	Originating national portion (ms)		Destination national portion (ms)		International portion (ms)	
		Number of satellites		Number of satellites		Connection type	
		0	1	0	1	1	2
600	Mean	800	1300	1000	1500	400	1700
	95%	1200	1800	1500	2100	600	2000
2 400	Mean	700	1200	800	1300	300	1600
	95%	1100	1600	1200	1800	500	1900
4 800	Mean	600	1100	800	1300	300	1600
	95%	900	1500	1200	1800	500	1900
9 600	Mean	600	1100	700	1200	300	1600
	95%	900	1500	1100	1600	500	1900
48 000	Mean	600	1100	700	1200	200	1500
	95%	900	1500	1100	1600	400	1700

Note – See introductory note to this Recommendation.

3.2 *Network clear indication delay (NCID)*

Network clear indication delay (NCID) is the delay between transmission of a *clear request* signal by the clearing DTE and the receipt of the DCE *clear indication* signal by the cleared DTE. It should not exceed the values given in Tables 5/X.130 and 6/X.130.

If on any call the overall network clear indication delay exceeds Y seconds the call will be considered for quality of service purposes to have failed. The precise value of Y is for further study but it should be at least 30 seconds.

TABLE 5/X.130

Overall network clear indication delay for common-channel signalling

User rate (bit/s)	Statistic	Delay (ms)	
		Connection type	
		1	2
600	Mean	900	1900
	95%	1300	2400
2 400	Mean	700	1700
	95%	1100	2100
4 800	Mean	600	1600
	95%	900	1900
9 600	Mean	600	1600
	95%	900	1900
48 000	Mean	600	1600
	95%	900	1900

Note – See introductory note to this Recommendation.

TABLE 6/X.130

Overall network clear indication delay for channel-associated signalling

User rate (bit/s)	Statistic	Delay (ms)	
		Connection type	
		1	2
600	Mean	1100	2100
	95%	1600	2700
2 400	Mean	900	1900
	95%	1300	2400
4 800	Mean	800	1800
	95%	1200	2300
9 600	Mean	800	1800
	95%	1200	2300
48 000	Mean	800	1800
	95%	1200	2300

Note – See introductory note to this Recommendation.

3.3 *Network portion clear indication delay (NPCID)*

The contribution from each network portion to the overall network clear indication delay should not exceed the values given in Tables 7/X.130 and 8/X.130.

TABLE 7/X.130

Contributions to network clear indication delay for common-channel signalling

User rate (bit/s)	Statistic	Originating national portion (ms)		Destination national portion (ms)		International portion (ms)	
		Number of satellites		Number of satellites		Connection type	
		0	1	0	1	1	2
600	Mean	300	600	400	700	200	900
	95%	500	800	600	900	300	1100
2 400	Mean	200	500	300	600	200	900
	95%	300	600	500	800	300	1100
4 800	Mean	200	500	300	600	100	800
	95%	300	600	500	800	200	900
9 600	Mean	200	500	300	600	100	800
	95%	300	600	500	800	200	900
48 000	Mean	200	500	300	600	100	800
	95%	300	600	500	800	200	900

Note – See introductory note to this Recommendation.

TABLE 8/X.130

Contributions to network clear indication delay for channel-associated signalling

User rate (bit/s)	Statistic	Originating national portion (ms)		Destination national portion (ms)		International portion (ms)	
		Number of satellites		Number of satellites		Connection type	
		0	1	0	1	1	2
600	Mean	400	700	500	800	200	900
	95%	600	900	800	1100	300	1100
2 400	Mean	300	600	400	700	200	900
	95%	500	800	600	900	300	1100
4 800	Mean	300	600	300	600	200	900
	95%	500	800	500	800	300	1100
9 600	Mean	300	600	300	600	200	900
	95%	500	800	500	800	300	1100
48 000	Mean	300	600	300	600	200	900
	95%	500	800	500	800	300	1100

Note – See introductory note to this Recommendation.

3.4 *Clear confirmation delay (CLCD)*

Clear confirmation delay (CLCD) is the delay between transmission of a *DTE clear confirmation* signal and receipt of a *DCE ready* signal by the cleared DTE. Clear confirmation delay is considered to be a national matter and consequently the specification of its value is not appropriate to this Recommendation.

ANNEX A

(to Recommendation X.130)

A.1 *Total call connection delay (TCCD) elements*

The total call connection delay is the sum of the following elements (see Figure A-1/X.130):

t1: delay between transmission of the *call request* signal and receipt of the *proceed to select* signal by the calling DTE.

t2: time between receipt of the *proceed to select* signal and transmission of the *end of selection* signal by the calling DTE.

t3: delay between transmission of the *end of selection* signal by the calling DTE and receipt of the *incoming call* signal by the called DTE.

t4: delay between receipt of the *incoming call* signal and transmission of the *call accepted* signal by the called DTE.

and if t5 > t6,

t5: delay between transmission of the *call accepted* signal by the called DTE and receipt of the *ready for data* signal by the calling DTE.

or if t5 < t6,

t6: delay between transmission of the *call accepted* signal and receipt of the *ready for data* signal by the called DTE.

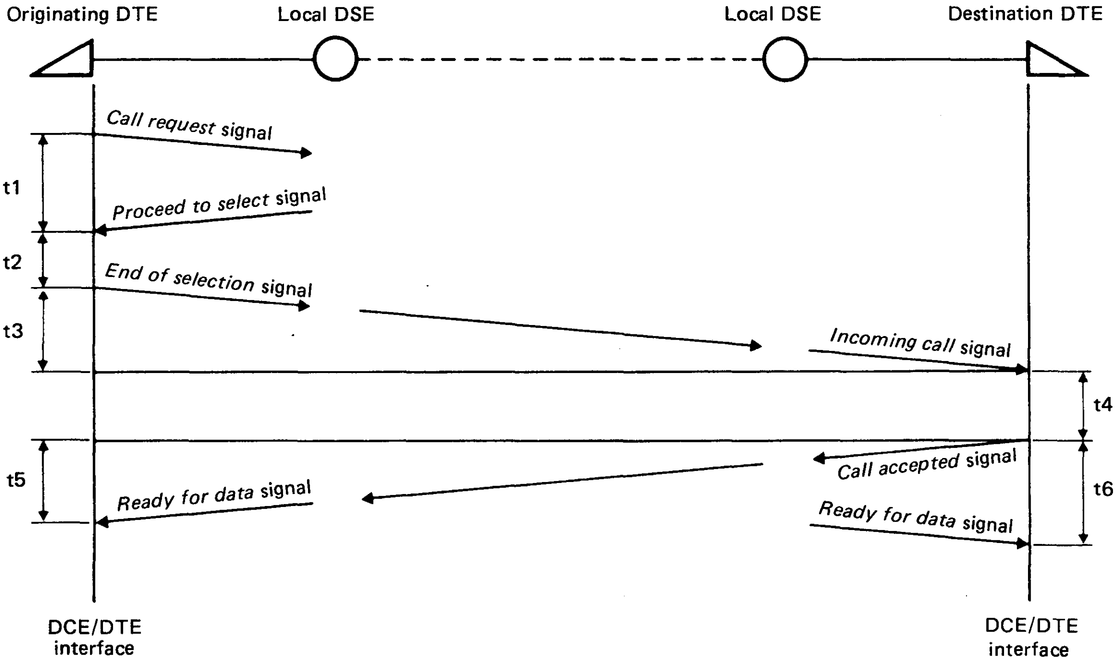


FIGURE A-1/X.130

Total call connection delay (TCCD)

A.1.1 *User-dependent call connection delay (UCCD)*

Considering the above elements of TCCD, t2 is dependent on the source DTE. Similarly t4 is dependent on the destination DTE. It is therefore inappropriate to specify values for these times in this Recommendation, but the following observations are relevant:

A.1.1.1 *Selection time (t2)*

Selection times for automatic calls from the DTE are given in Table A-1/X.130.

TABLE A-1/X.130

Selection time

User rate (bit/s)	Selection time (t2) (ms)
600	260
2 400	70
4 800	40
9 600	20
48 000	5

A.1.1.2 *Call acceptance delay (CAD) (t4)*

If the CAD exceeds 500 ms with automatic answer or 60 seconds with manual answer the DCE will initiate clearing.

A.1.2 *Network-dependent call connection delay (NCCD)*

Considering the elements of TCCD in § A.1, it has been shown in § A.1.1 that t2 and t4 are user-dependent and are not regarded as network performance parameters.

The network (dependent) call connection delay is therefore the sum of the remaining elements. Hence:

$$\text{NCCD} = t1 + t3 + t5$$

Recommendation X.131

CALL BLOCKING IN PUBLIC DATA NETWORKS WHEN PROVIDING INTERNATIONAL SYNCHRONOUS CIRCUIT-SWITCHED DATA SERVICES

(Former X.132, Geneva, 1980; amended at Malaga-Torremolinos, 1984)

The CCITT,

considering

(a) that Recommendation X.1 specifies the user classes of service applicable to networks offering public data services;

(b) that Recommendation X.2 specifies the international user services and facilities to be offered by public data networks;

(c) that Recommendations X.21 and X.21 *bis* define the DTE/DCE interface for circuit switched services;

(d) that Recommendation X.60 specifies the common channel signalling for synchronous data networks;

(e) that Recommendation X.71 specifies the channel associated signalling for synchronous data networks;

(f) that Recommendation X.92 specifies the hypothetical reference connections for public data networks;

(g) that Recommendation X.110 specifies the routing plan to be applied in the international portions of public data networks;

(h) that Recommendation X.213 specifies the OSI Network Layer service;

(i) that Recommendation X.140 specifies the user-oriented quality of service parameters applicable to data services,

unanimously declares

that when public data networks provide international synchronous circuit-switched data services, according to Recommendations X.21 and X.21 *bis*, the values of call blocking probability specified in this Recommendation shall be taken as provisional, worst-case values that should not be exceeded under the conditions specified therein.

Introductory note – Design objectives that take into account both user needs and network costs are for further study.

1 Introduction

1.1 Quality of service in circuit-switched public data networks has been considered in five basic areas as follows:

i) call processing delays (Recommendation X.130);

ii) failures due to congestion (blocking) (Recommendation X.131);

iii) failures due to malfunction;

iv) loss of service; and

v) transmission performance (including throughput).

This Recommendation specifies the objectives for ii) above. Each of the other areas of circuit-switching quality of service identified above will be the subject of a separate Recommendation in the X-series.

1.2 In telecommunication networks it is necessary, for economic reasons, to limit the resources provided for carrying the offered traffic. This limitation may affect the quality of service to the user of circuit-switched services in two different ways: by call processing delays and by blocking. Both of these aspects, that are consequences of the finite traffic handling capacity of the network, constitute the grade of service. Grade of service together with malfunction, loss of service and transmission performance constitute the quality of service.

1.3 In this Recommendation the values for the network blocking are quoted for two types of connection according to Recommendation X.92 as follows:

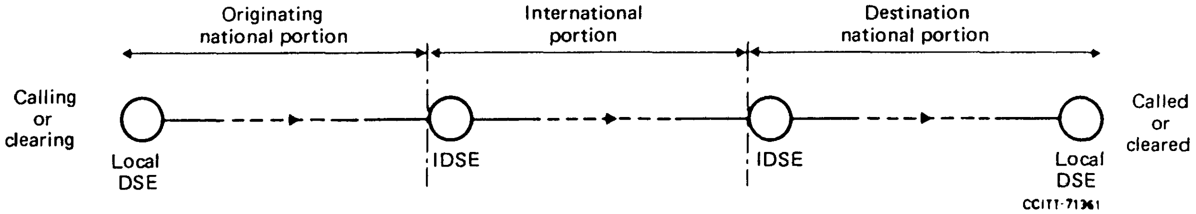
Type 1: Typical terrestrial interconnection of moderate length with no satellite circuits either in the national or international portions. (International portion: 1000 km.)

Type 2: Long distance international connection with a satellite circuit in one national portion and two satellite circuits in the international portion. (International portion: 160 000 km.)

Where appropriate, values are also specified separately for the following network portions:

- originating national network,
- international portion,
- destination national network.

The boundaries for these portions are shown in Figure 1/X.131.



Note 1 – A DSE may also function as an IDSE.

Note 2 – Arrows indicate direction of call set-up or clear-down.

FIGURE 1/X.131

National/international boundaries for call set-up and clear-down functions

For the present, the values apply also to other normal routing options within the international portion.

Following the allocation of a blocking allowance to the international portion of an international transit connection, it will be necessary to further apportion the allowance to individual transit networks and/or their component parts within the international portion. The means by which useful and realistic constraints can be applied, consistent with maintaining the maximum possible freedom for each involved Administration in the design and implementation of its own network, is for further study.

1.4 The values for blocking probability established in this Recommendation are to be considered as design objectives in network planning together with the forecast traffic for the planned period. The actual blocking performance that will be obtained will depend on the accuracy of the traffic estimations. Normally the actual blocking performance will not coincide with the one used as a basis for planning. Furthermore, if the network is planned for the traffic forecast at the end of the period considered, the actual blocking performance of the network may be better than the design value, worsening gradually to the end of the planning period as traffic increases.

The non-coincidence of busy hours in originating and destination national networks as well as in the international network will improve the overall blocking performance with respect to the sum of the nominal blocking probabilities of the constituent parts of the connection.

1.5 The blocking probabilities are specified under conditions of normal busy hour load. Blocking probabilities for higher loadings are for further study.

1.6 Blocking probabilities are defined for a basic call which does not include any optional user facilities, e.g. those defined in Recommendation X.21.

1.7 Recommendation X.21 permits the following blocking situations:

i) non-reception of *proceed to select*;

ii) non-connection of call.

Item i) is considered to be a national matter and consequently the specification of its value is not appropriate to this Recommendation. Objectives for item ii) are contained in § 2 of this Recommendation.

1.8 The quality of service implications of regional or national satellite systems using demand assignment for resource allocation require further study.

2 Probability of non-connection due to congestion (blocking probability)

Probability of non-connection due to congestion is the probability that a calling DTE does not receive the *ready for data* signal but does receive a *network congestion* signal within 20 seconds after transmission of the *end of selection* signal (or within 60 seconds when manual answering is permitted at the called DTE).

2.1 *Overall probability of non-connection due to congestion*

The overall probability of non-connection due to congestion for an end-to-end connection seen from the customer point of view should not exceed the following values:

Connection type 1: 13%

Connection type 2: 15%

See Introductory note to this Recommendation.

2.2 *Network portion probability of non-connection due to congestion*

The contribution of each network portion to overall probability of non-connection due to congestion should not exceed the values shown in Table 1/X.131.

TABLE 1/X.131

Contributions to network probability of non-connection due to congestion

Originating national portion	Destination national portion	International portion	
		Connection type	
		1	2
5%	5%	3%	5%

Note – See introductory note to this Recommendation.

Recommendation X.134

PORTION BOUNDARIES AND PACKET LAYER REFERENCE EVENTS: BASIS FOR DEFINING PACKET-SWITCHED PERFORMANCE PARAMETERS

(Melbourne, 1988)

The CCITT,

considering

(a) that Recommendation X.1 specifies the international user classes of service in public data networks;

(b) that Recommendation X.2 specifies the international data transmission services and optional user facilities in public data networks;

(c) that Recommendation X.25 specifies the DTE/DCE interface for packet mode terminals connected to public data networks by dedicated circuit;

(d) that Recommendation X.75 specifies the packet-switched signalling system between public networks providing data transmission services;

(e) that Recommendation X.323 specifies general arrangements for interworking between packet-switched public data networks;

(f) that Recommendation X.96 specifies call progress signals in public data networks;

(g) that Recommendation X.110 specifies the international routing principles and routing plan for public data networks;

(h) that Recommendation X.213 defines the OSI Network Layer service;

(i) that Recommendation X.140 defines general quality of service parameters for communication via public data networks;

(j) that Recommendation X.135 specifies speed of service performance values for public data networks when providing international packet-switched service;

(k) that Recommendation X.136 specifies accuracy and dependability (including blocking) performance values for public data networks when providing international packet-switched service;

(l) that Recommendation X.137 specifies availability performance values for public data networks when providing international packet-switched service,

unanimously declares

(1) that the portion boundaries defined in this Recommendation shall be used in apportioning the performance of an international packet-switched data communication service provided in accordance with Recommendations X.25 and X.75;

(2) that the packet layer reference events specified in this Recommendation shall be used in the definition of packet-switched performance parameters for data communication services provided in accordance with Recommendations X.25 et X.75.

1 Introduction

1.1 This Recommendation is the first in a series of four CCITT Recommendations (X.134-X.137) that define performance parameters and values for international packet-switched data communication services. Figure 1/X.134 illustrates the scope of these four Recommendations and the relationships among them.

1.2 This Recommendation divides a virtual connection into basic sections whose boundaries are associated with X.25 and X.75 interfaces. The performance of collections of these basic sections can be measured using the packet-switched performance parameters defined in Recommendations X.135-137. In order to apportion the performance of an international virtual connection, Recommendation X.134 defines two particular collections of basic sections for which performance values will be specified: national portions and international portions. As defined, every international virtual connection contains two national portions and one international portion. The performance of these three portions can be combined in the calculation of the end-to-end virtual connection performance. These Recommendations do not specify performance values for other collections of basic sections; however, the ability to decompose a virtual connection into its basic sections will be useful in planning the performance of national and international portions.

1.3 The performance parameters in Recommendations X.135-X.137 are defined in terms of packet layer reference events which can be observed at the boundaries between basic sections and thus can be observed at the portion boundaries. This Recommendation defines the performance significant packet layer reference events.

1.4 For comparability and completeness, packet-switched network performance is considered in the context of the 3 × 3 performance matrix defined in Recommendation X.140. Three protocol-independent data communication functions are defined in the matrix: access, user information transfer, and disengagement. These general functions correspond to call set-up, data (and interrupt) transfer, and call clearing in packet-switched virtual call services conforming to the X.25 and X.75 Recommendations. Each function is considered with respect to three general performance concerns (or "performance criteria"): speed, accuracy, and dependability. These express, respectively, the delay or rate, degree of correctness, and degree of certainty with which the function is performed.

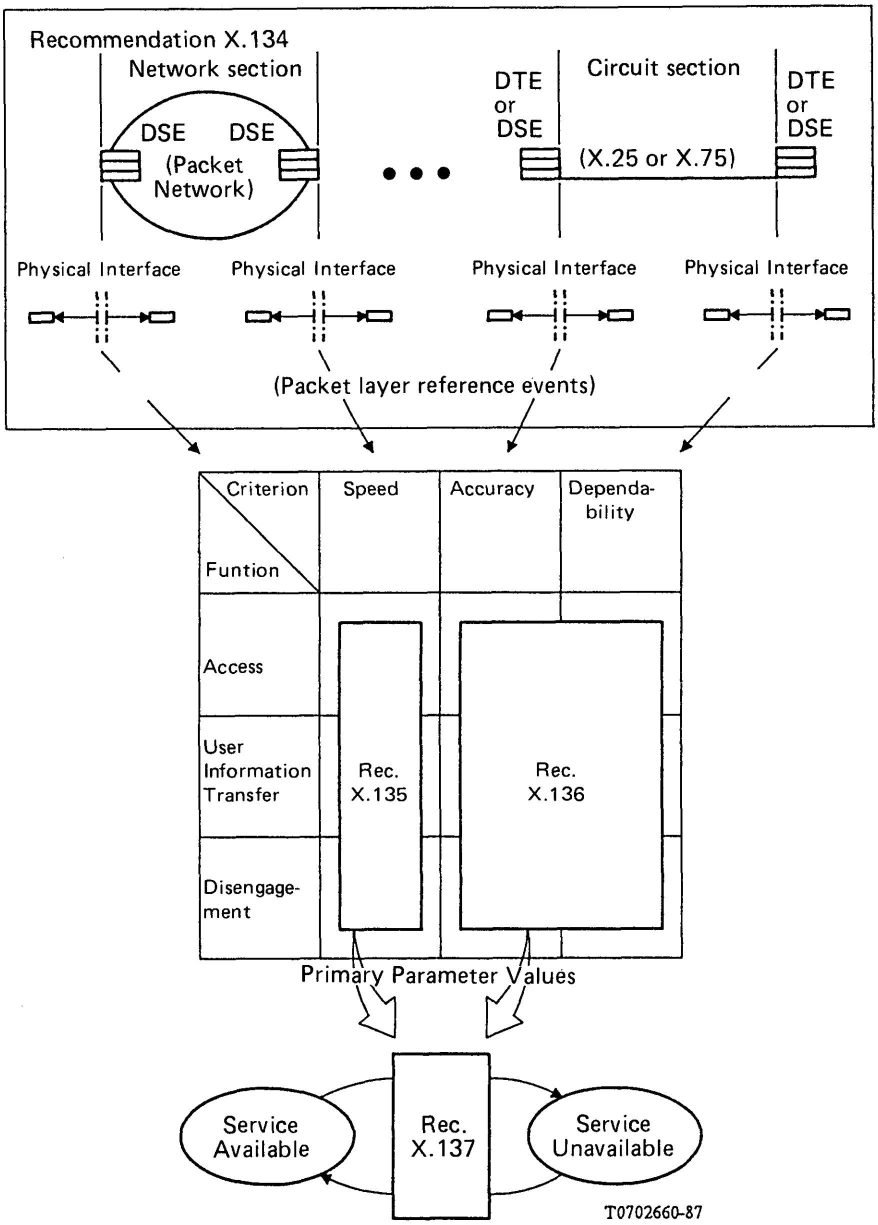


FIGURE 1/X.134

Packet-switched service performance description framework

1.5 Recommendation X.135 defines protocol-specific speed of service parameters and values associated with each of the three data communication functions. Recommendation X.136 defines protocol-specific accuracy and dependability parameters and values associated with each function. The Recommendation X.135 and Recommendation X.136 parameters are called "primary parameters" to emphasize their direct derivation from packet layer reference events.

1.6 An associated two-state model provides a basis for describing overall service availability. A specified availability function compares the values for a subset of the primary parameters with corresponding outage thresholds to classify the service as "available" (no service outage) or "unavailable" (service outage) during scheduled service time. Recommendation X.137 specifies the availability function and defines the availability parameters and values that characterize the resulting binary random process.

1.7 In order to relate the network performance values given in Recommendations X.135 to X.137 to the service receivable at points within the scope of the DTEs, further elements must be included.

1.7.1 In particular, specification of service performance at the layer 3/4 boundary (OSI Network Service) must include those processes within the DTEs concerned with the transfer of packets from the physical circuit of the DTE/DCE interface to the layer 3/4 boundary at each end of the virtual connection, however they may be implemented. This processing may include elements associated with OSI layers 1, 2 and 3 and may involve transmission across private wide-area and/or local-area networks.

1.7.2 Specification of service performance for the user or application, if required, must similarly include in addition those processes within the DTEs concerned with the transfer of information from the layer 3/4 boundary to the layer 7 upper boundary beyond each of the virtual connection, however they may be implemented. This processing may include elements associated with OSI layers 4, 5, 6 and 7.

1.7.3 Additional protocol- or service-specific reference events would need to be defined to cover these aspects, but are outside the scope of this Recommendation. However, the parameter definitions of Recommendations X.135 to X.137 can readily be adapted to correspond to any extension of scope.

2 Virtual connection sections and portions

In the context of Recommendations X.134-X.137, the following definitions apply:

An **access circuit section** is the physical circuit or set of circuits connecting a DTE to the local DSE. It does not include any parts of the DTE or DSE. These recommendations assume that X.25 procedures are used on an access circuit section.

An **internetwork circuit section** is the physical circuit or set of circuits connecting a DTE in one network with a DSE in a different network. It does not include any parts of either DSE. These recommendations assume that X.75 procedures are used on an internetwork circuit section.

A **circuit section** is either an access circuit section or an internetwork circuit section.

A **network section** consists of the network components that provide a virtual connection between two circuit sections. The network provider is responsible for the performance of the network section.

An **access network section** is a network section connected to (at least) one access circuit section.

A **transit network section** is a network section between two internetwork circuit sections.

A **basic section of a virtual connection** is either an access network section, a transit network section, an access circuit section, or an internetwork circuit section.

A **section boundary (or boundary)** separates a network section from the adjacent circuit section or it separates an access circuit section from the adjacent DTE.

A **national portion of an international virtual connection** is a collection of adjacent alternating network sections and circuit sections entirely within the borders of one nation. The national portion connects a DTE to an internetwork circuit section that crosses the national border. The national portion includes the access circuit section and excludes the internetwork circuit section that crosses the national border. A national portion always includes one access circuit section and one access network section, and it may include one or more pairs of internetwork circuit sections and transit network sections.

There are two national portions of any international virtual circuit.

An **international portion of an international virtual connection** is the set of basic sections between the two national portions. An international portion may be a single internetwork circuit section crossing a national border or it may be two (or more) internetwork circuit sections together with one (or more) transit network sections.

There is one international portion of any international virtual circuit and that international portion will cross one or more national borders.

For purposes of allocating the performance of an international virtual connection, this Recommendation defines a **portion boundary** as a section boundary delimiting a national or international portion.

Figure 2/X.134 illustrates the definitions and delimitation of the virtual connection sections and portions. A typical international virtual connection is shown including the two access circuit sections and the two DTEs.

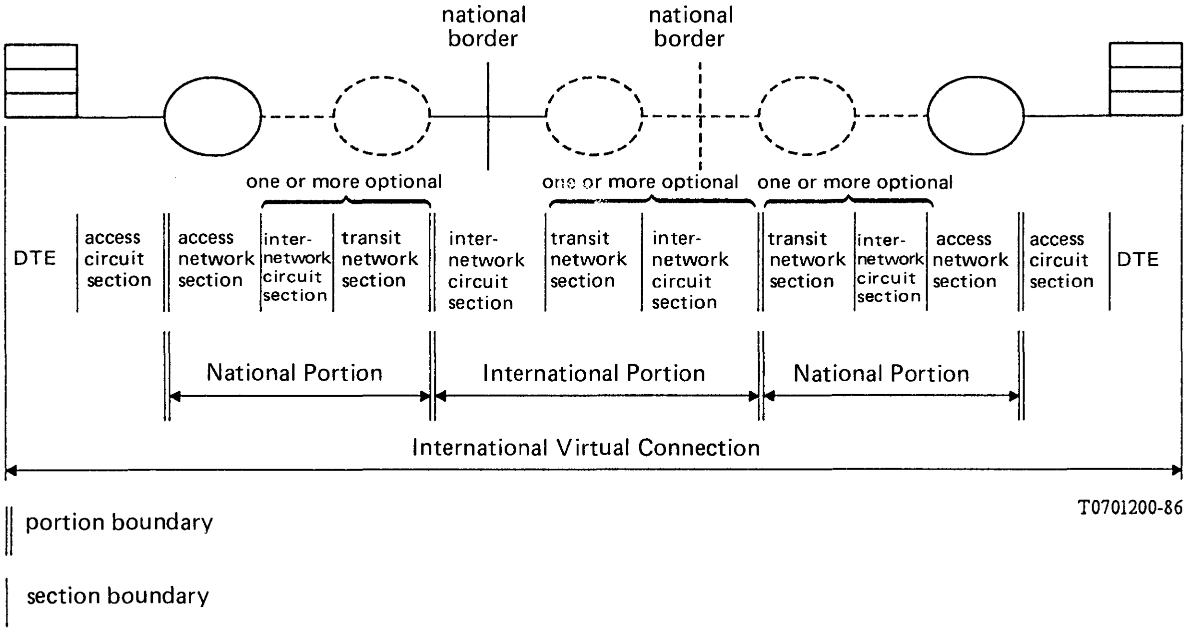


FIGURE 2/X.134

Apportionment of an international virtual connection

3 Packet layer reference events

3.1 *Definitions*

In the context of Recommendations X.134-X.137:

A **packet layer reference event** 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 Recommendations X.25 and X.75.

Two classes of packet layer reference events are defined.

A **packet entry event** is 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).

A **packet exit event** is 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 frame retransmissions occur, the packet exit event occurs with the first transmission and the packet entry event occurs with the next transmission.

Figure 3/X.134 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, 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 was changed by the event.

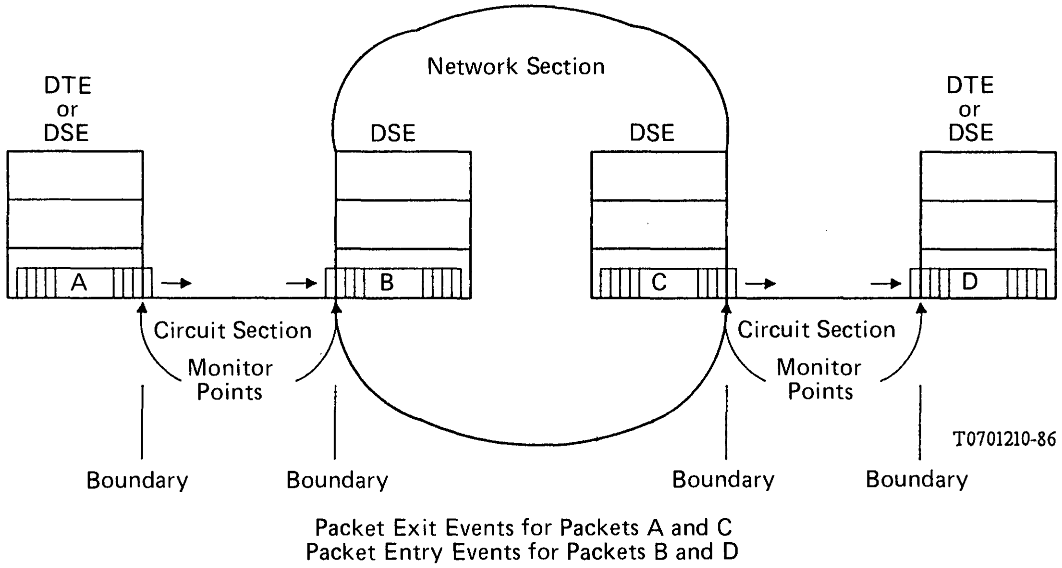


FIGURE 3/X.134

Example Packet Layer-Reference Events

3.2 *Performance-significant reference events*

The performance-significant reference events are the packet layer reference events useful in defining performance parameters. Table 1/X.134 lists performance-significant X.25 packet layer reference events associated with the boundaries of access circuit sections. Table 2/X.134 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 Recommendations X.135-X.137.

The entries in Tables 1-2/X.134 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 Recommendations X.25 and X.75 are addressed in the tables.

The states identified in the tables differ from those defined in 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.

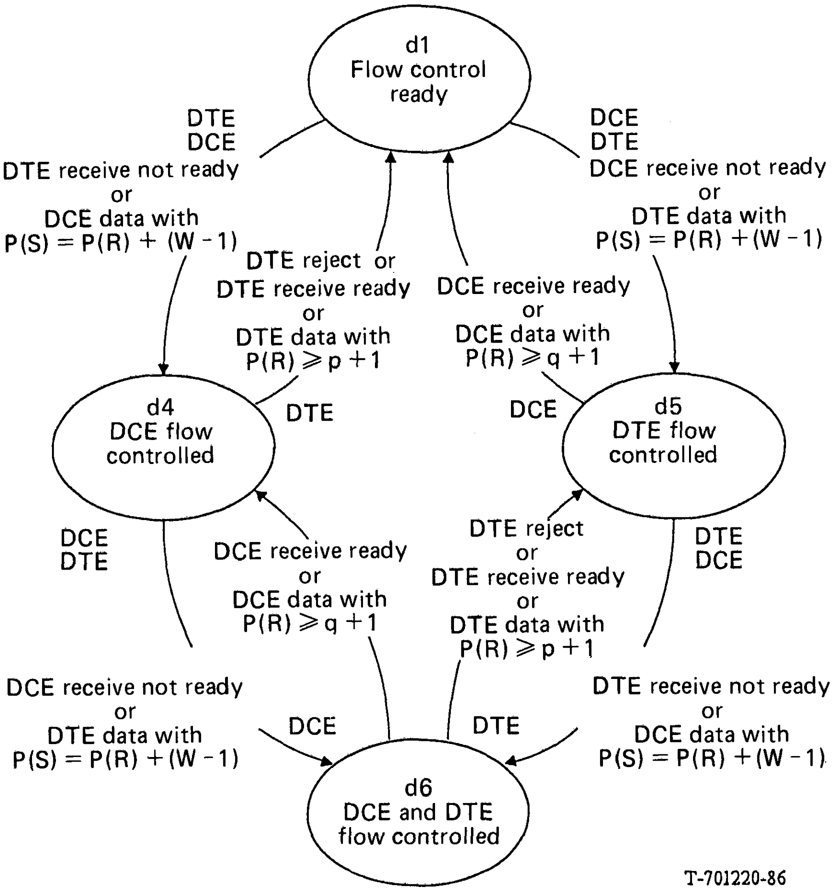
Three ancillary X.25 states and three ancillary X.75 states are defined in this Recommendation to permit more accurate description of flow control effects. The new X.25 states are "DCE flow controlled," "DTE flow controlled," and "DTE and DCE flow controlled." The new X.75 states are "STE X flow controlled," "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/X.134. A state diagram for the ancillary X.75 flow control states is shown in Figure 5/X.134. In each case, the new states are numbered d4-d6.

Three ancillary state variables are defined:

- *lwt* – lower edge of the window on the transmit side. This variable contains the latest P(R) received either in a data packet, an RR, or an RNR. The value may be implicitly represented using the upper window edge (and the window size).
- *npr* – next data packet to be received. This variable contains the P(S) of the next data packet to be received.
- *ric* – received interrupt count. Because only one unacknowledged interrupt packet can exist in a particular direction, the interface must record 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.

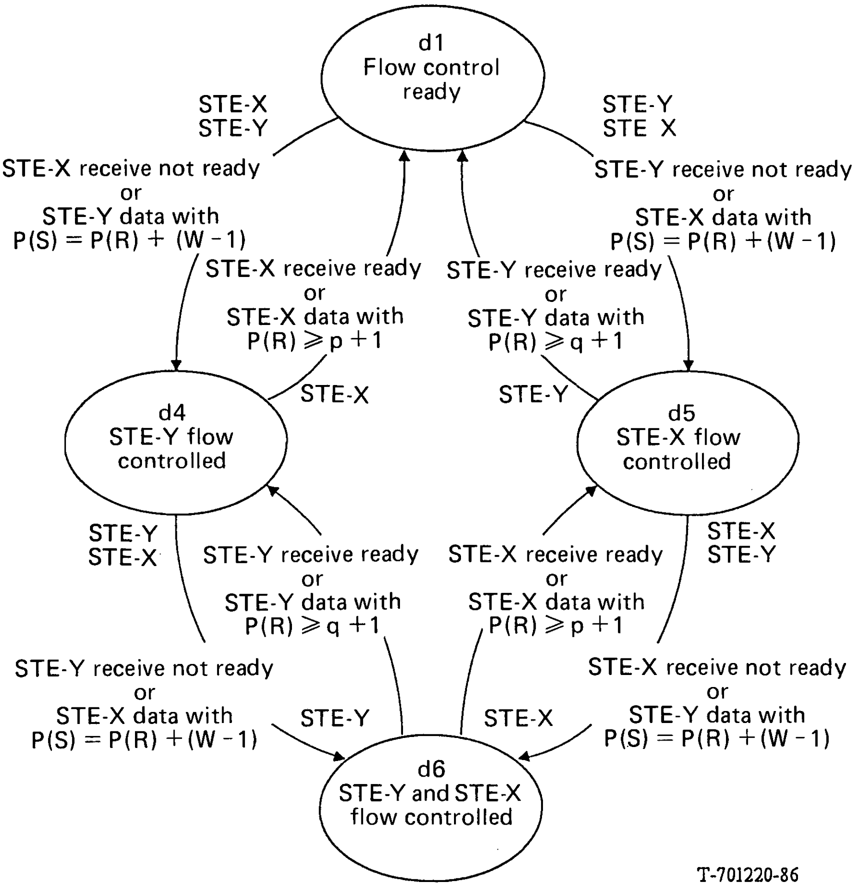
When the tables list more than one aspect of the state that might be changed as a result of a particular packet's entry or exit, each of those changes represents a distinct packet layer reference event that can be used in defining different performance parameters. For example, in Table 1/X.134, event 9a would be used where the correct receipt of the data is relevant, and 9b would be used when the receipt of the acknowledgement is relevant. Event 26b would be used in association with permanent virtual circuits and 26a with other logical channels.



Note – Variables p and q represent the send sequence numbers of the last DTE data and DCE data paquets transferred across the DTE/DCE interface, respectively.

FIGURE 4/X.134

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 paquets transferred across the STE-X / STE-Y interface, respectively.

FIGURE 5/X.134

Diagram of STE-X/STE-Y flow control states

TABLE 1/X.134

X.25 packet layer reference events

Number	Packet type	Resulting state
1	Incoming Call	p3 (DCE Waiting)
2	Cal 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
9a	DCE Data	npr becomes P(S) + 1
9b	DCE Data	lwt becomes P(R)
9c	DCE Data	d1 (Flow Control Ready)
10a	DTE Data	npr becomes P(S) + 1
10b	DTE Data	lwt becomes P(R)
10c	DTE Data	dl (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
15a	DCE RR	lwt becomes P(R)
15b	DCE RR	d1
16a	DTE RR	lwt becomes P(R)
16b	DTE RR	dl
17a	DCE RNR	lwt becomes P(R)
17b	DCE RNR	d5 (DTE Flow Controlled)
17c	DCE RNR	d6 (DTE + DCE Flow Controlled)
18a	DTE RNR	lwt becomes P(R)
18b	DTE RNR	d4 (DCE Flow Controlled)
18c	DTE RNR	d6
19	DTE REJ	npr becomes P(R) (Note 1)
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	r2 (DTE Restart Request)
26a	DCE Restart Confirmation	pl
26b	DCE Restart Confirmation	dl
27a	DTE Restart Confirmation	pl
27b	DTE Restart Confirmation	dl
(Note 2)		

Note 1 – This is npr from the perspective of the DTE.

Note 2 – 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.134

X.75 packet layer reference events

Number	Packet type	Resulting state	
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)
5a	Data	npr	becomes P(S)+1
5b	Data	lwt	becomes P(R)
5c	Data	d1	(Flow Control Ready)
6a	Interrupt	i2 or i3	(STE Interrupt Request)
6b	Interrupt	i4	(STE-X and Y Interrupt Request)
7a	Interrupt Confirmation	i1	(No Interrupt Request)
7b	Interrupt Confirmation	i2 or i3	
8a	RR	lwt	becomes P(R)
8b	RR	d1	
9a	RNR	lwt	becomes P(R)
9b	RNR	d4 or d5	(STE Flow Controlled)
9c	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)
13a	Restart Confirmation	p1	
13b	Restart Confirmation	d1	

Recommendation X.135

SPEED OF SERVICE (DELAY AND THROUGHPUT) PERFORMANCE VALUES FOR PUBLIC DATA NETWORKS WHEN PROVIDING INTERNATIONAL PACKET-SWITCHED SERVICES

(Malaga-Torremolinos, 1984; amended at Melbourne, 1988)

The CCITT,

considering

(a) that Recommendation X.1 specifies the international user classes of service in public data networks;

(b) that Recommendation X.2 specifies the international data transmission services and optional user facilities in public data networks;

(c) that Recommendation X.25 specifies the DTE/DCE interface for packet mode terminals connected to public data networks by dedicated circuit;

(d) that Recommendation X.75 specifies the packet switched signalling system between public networks providing data transmission services;

(e) that Recommendation X.323 specifies general arrangements for interworking between packet-switched public data networks;

(f) that Recommendation X.96 specifies call progress signals in public data networks;

(g) that Recommendation X.110 specifies the international routing principles and routing plan for public data networks;

(h) that Recommendation X.213 defines the OSI Network Layer service;

(i) that Recommendation X.140 defines general quality of service parameters for communication via public data networks;

(j) that Recommendation X.134 specifies portion boundaries and packet layer reference events for defining packet-switched performance parameters;

(k) that Recommendation X.136 specifies accuracy and dependability (including blocking) performance values for public data networks when providing international packet-switched service;

(l) that Recommendation X.137 specifies availability performance values for public data networks when providing international packet-switched service,

unanimously declares

(1) that the speed of service parameters defined in this Recommendation shall be used in the planning and operation of international packet-switched data communication services provided in accordance with Recommendations X.25 and X.75;

(2) that in such services, the performance values specified in this Recommendation shall be taken as worst-case limits under the conditions specified herein.

1 Introduction

1.1 This Recommendation is the second in a series of four CCITT Recommendations (X.134-X.137) that define performance parameters and values for international packet-switched data communication services. Figure 1/X.135 illustrates the scope of these four Recommendations and the relationships among them.

1.2 Recommendation X.134 divides a virtual connection into basic sections whose boundaries are associated with X.25 and X.75 interfaces; defines particular collections of basic sections, called virtual connection portions, for which performance values will be specified; and defines a set of packet layer reference events (PEs) which provide a basis for performance parameter definition. The basic sections consist of network sections and circuit sections. They are delimited, in each case, by physical data terminal equipment (DTE) or data switching equipment (DSE) interfaces. Virtual connection portions are identified either as national portions or international portions. Each PE is defined to occur when a packet crossing a section boundary changes the state of the packet layer interface.

1.3 For comparability and completeness, packet-switched network performance is considered in the context of the 3 × 3 performance matrix defined in Recommendation X.140. Three protocol-independent data communication functions are identified in the matrix: access, user information transfer, and disengagement. These general functions correspond to call set-up, data (and interrupt) transfer, and call clearing in packet-switched virtual call services conforming to the X.25 and X.75 Recommendations. Each function is considered with respect to three general performance concerns (or "performance criteria"): speed, accuracy, and dependability. These express, respectively, the delay or rate, degree of correctness, and degree of certainty with which the function is performed.

1.4 This Recommendation defines protocol-specific speed of service parameters and values associated with each of the three data communication functions. Recommendation X.136 defines protocol-specific accuracy and dependability parameters and values associated with each function. The Recommendation X.135 and Recommendation X.136 parameters are called "primary parameters" to emphasize their direct derivation from packet layer reference events.

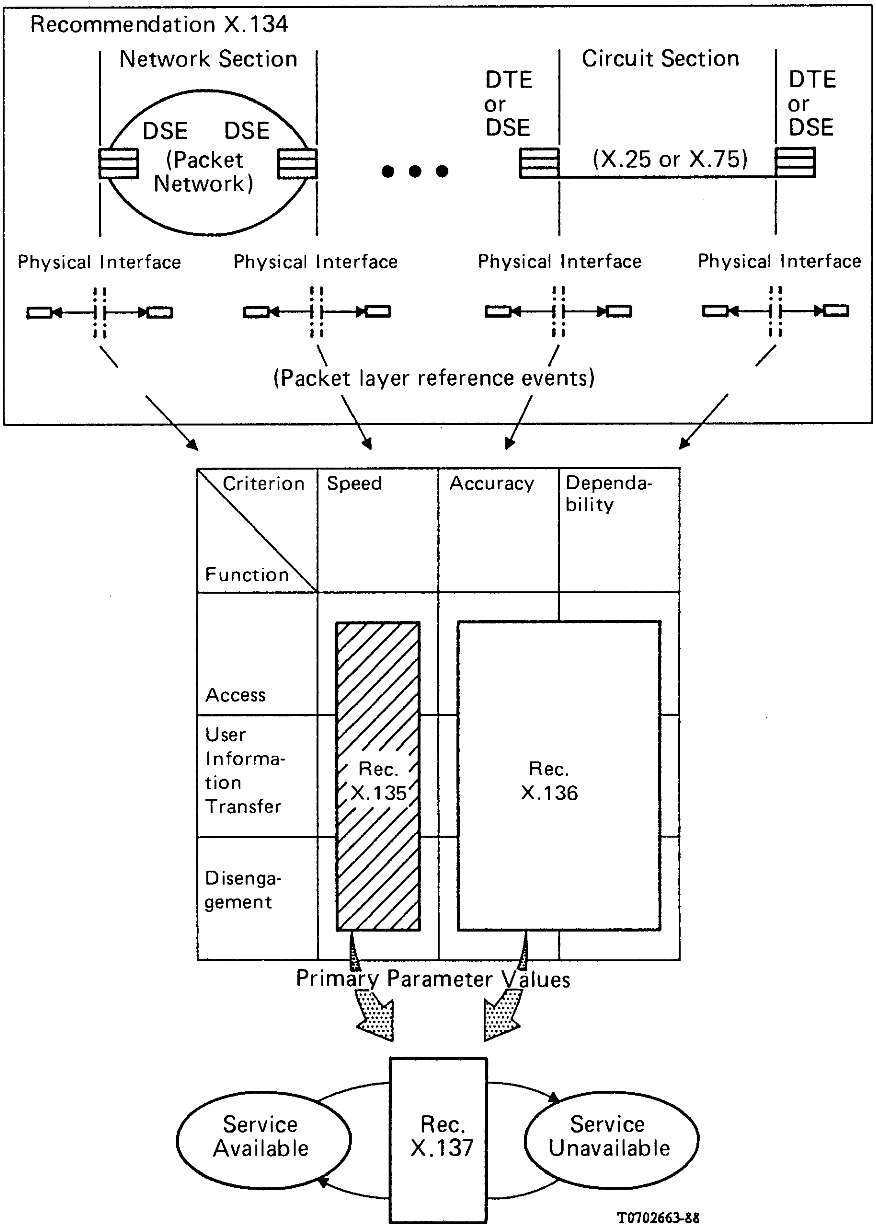


FIGURE 1/X.135

Packet-switched service performance description framework

1.5 An associated two-state model provides a basis for describing overall service availability. A specified availability function compares the values for a subset of the primary parameters with corresponding outage thresholds to classify the service as "available" (no service outage) or "unavailable" (service outage) during scheduled service time. Recommendation X.137 specifies the availability function and defines the availability parameters and values that characterize the resulting binary random process.

1.6 Four speed of service parameters are defined in this Recommendation: one access parameter (call set-up delay), two user information transfer parameters (data packet transfer delay and throughput capacity), and one disengagement parameter (clear indication delay). Each parameter can be applied to any basic section or portion of a virtual connection. This generality makes the parameters useful in performance allocation and concatenation.

1.7 This Recommendation specifies delay and throughput values for national portions and international portions of two types (Table 1/X.135). Performance values for data terminal equipment are not specified, but the parameters defined in this Recommendation may be employed in such specification to assist users in establishing quantitative relationships between network performance and quality of service (see Recommendation X.140).

TABLE 1/X.135

Virtual connection portion types for which perfomance values are specified [a)]

Portion type	Typical characteristics
National A	Terrestrial connection via an access network section
National B	Connection via an access network section with one satellite circuit; or via an access network section and one or more transit network sections
International A	Connection via a direct terrestrial internetwork circuit section
International B	Connection via two satellite circuits and one transit network section; or via one satellite circuit and two or more transit network sections

[a)] The values specified for Type B portions also apply to virtual connection portions not explicitly identified as Type A or Type B.

1.8 Worst-case mean and 95% probability values for call set-up delay, data packet transfer delay, throughput capacity, and clear indication delay are specified for each virtual connection portion type identified in Table 1/X.135. The term "worst case" means that these values should be met during the normal busy hour in the worst-performing virtual connection portion used in providing international packet-switched service. The performance of a virtual connection portion will normally be much better than the worst-case values specified in this Recommendation.[1)] Design objectives that take into account more demanding user applications and network performance and connectivity enhancements are for further study.

Numerical methods for combining individual portion performance values to estimate end-to-end performance are also provided in this Recommendation. DTE to DTE values for two particular hypothetical reference connections are derived using these methods in Annex C.

2 Call set-up delay

Call set-up delay 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 and then call set-up delay between a pair of section boundaries (B_i, B_j) is 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. Values are specified for call set-up delay observed between section boundaries.

[1)] Supplement 1 presents delay and throughput values measured on particular connections at particular times and is for illustrative purposes only.

2.1 *Definition of call set-up delay at a single section boundary*

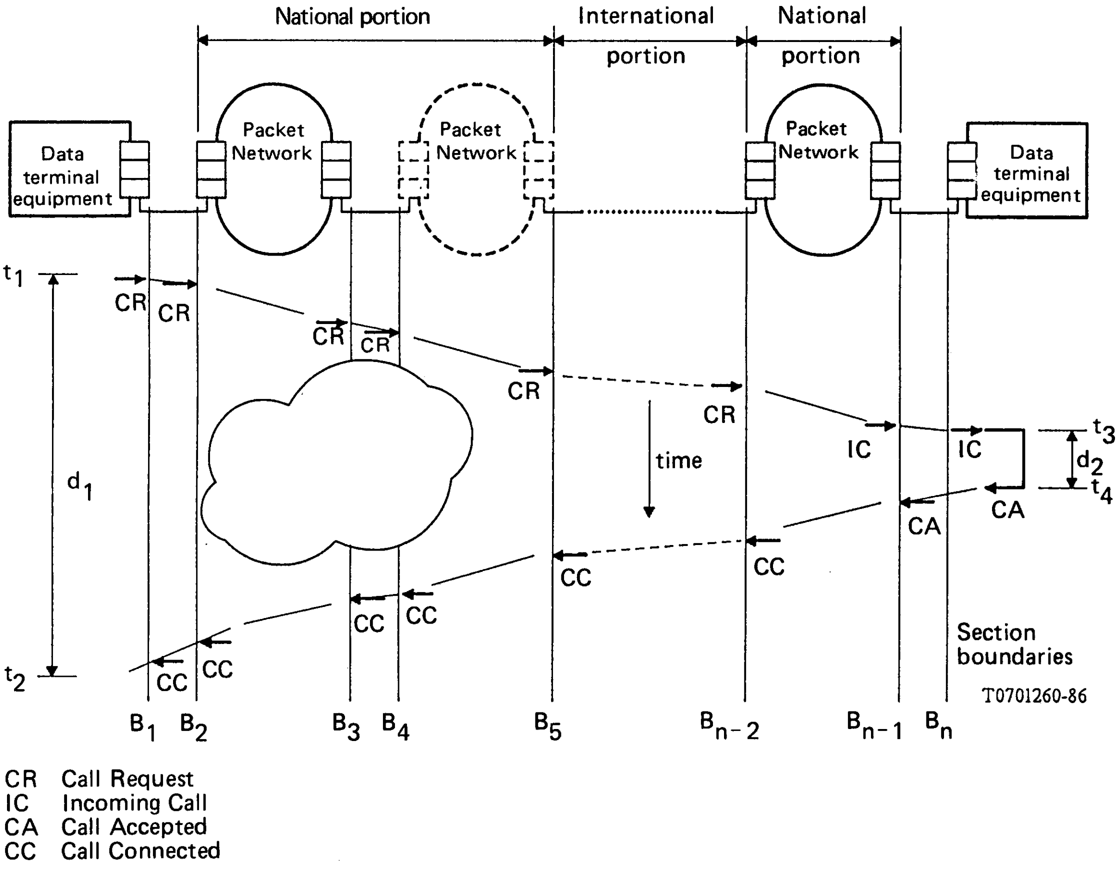
Call set-up delay at a section boundary, B_i, is defined using two Recommendation X.134 packet layer reference events (PEs). It is the period of time that starts when either a call request or an incoming call packet creates a PE at B_i, and ends when the corresponding call connected or call accepted 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 2/X.135. 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 X.134 PEs used in measuring call set-up delay at each section boundary are identified in Table 2/X.135.



Note – (t_1, t_2) and (t_3, t_4) may be observed on the calling side and called side of any virtual connection portion.

FIGURE 2/X.135

Call set-up delay events

TABLE 2/X.135

Packet layer reference events (PEs) used in measuring call set-up delay [a]

X.134 packet layer reference event / Circuit section	Starting PE	Ending PE
Calling DTE access circuit section	2 (X.25)	3 (X.25)
Called DTE access circuit section	1 (X.25)	4 (X.25)
Internetwork circuit section	1 (X.75)	2 (X.75)

[a] The PE numbers in this table refer to Tables 1/X.134 and 2/X.134 in Recommendation X.134.

2.2 *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 measured 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, e.g., B_1 and B_n in Figure 2/X.135. This end-to-end delay excludes the called user response time. The **national portion call set-up delay** is the call set-up delay between the boundaries delimiting a national portion, e.g., B_1 and B_5 in Figure 2/X.135. The **international portion call set-up delay** is the call set-up delay between the boundaries delimiting an international portion, e.g., B_5 and B_{n-2} in Figure 2/X.135.

2.3 *Values*

Table 3/X.135 defines worst-case call set-up delay values for each of the four virtual connection portion types identified in Table 1/X.135. DTE to DTE call set-up delay values for two hypothetical reference connections are calculated in Annex C. All values are based on (and only apply under) the following assumptions[2]:

1) Normal busy hour load conditions for the observed virtual connection. The definition of "normal busy hour load" as a traffic description is for further study.
2) A basic call, in which none of the optional user facilities defined in Recommendation X.25 are used and no call user data is sent.
3) Data link layer windows of entities outside the portion being specified are open (not flow controlled).

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[2] Values for other conditions are for further study. In the case of extremely long access lines and/or excessive delays in the access circuit section transmission equipment, these values may be exceeded.

The defined values consist of mean and 95% probability values. The mean is the expected value of the call set-up delay distribution. The 95% probability value is the value below which 95% of the call set-up delay values lie. Call set-up attempts that are unsuccessful under the conditions of Recommendation X.136 are excluded and are addressed separately in that Recommendation.

TABLE 3/X.135

Worst-case call set-up delay values for virtual connection portions

Statistic	Virtual connection portion type			
	National		International	
	A	B	A	B
mean (ms)	1000 + X	1600 + X	250	1600
95% (ms)	1500 + X	2100 + X	250	1800

In Table 3/X.135, the value X depends on the signalling rate of the access circuit section that is included in the national portion. Table 4/X.135 presents the X values for user classes of service 8-11 in Recommendation X.1[3]. The X values for other signalling rates may be computed using the formula

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

where R is the signalling rate in kilobits per second.[4].

TABLE 4/X.135

X-values for Table 3/X.135

X.1 user class of service	R (kbit/s)	X (Milliseconds)
8	2.4	167
9	4.8	84
10	9.6	42
11	48.0	9

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[3] These X values are not intended to represent the delay performance of the access circuit section, since these values do not include propagation delays, multiplexing delays, or the effects of retransmission.

[4] The formula assumes that the transfer of each call set-up packet across an access circuit section involves the transmission of 25 octets: 5 octets of frame level overhead, a 5-octet packet header, and 15 octets of DTE address information

The call set-up delay values defined in Table 3/X.135 are intended to be used as worst-case limits in planning international packet-switched services. The actual delay performance achieved on 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 internetwork circuit sections. Variation away from the worst-case value for each factor can improve the performance.

The overall call set-up delay value for a set of concatenated virtual connection portions can be calculated directly by adding the individual portion means defined in Table 3/X.135. A method of calculating an overall 95% probability call set-up delay value for a set of concatenated virtual connection portions from the individual 95% probability values is described in Annex C.

3 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.

3.1 **data packet transfer delay** *definition*

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 X.134 PEs used in measuring data packet transfer delay at each section boundary are identified in Table 5/X.135.

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.

TABLE 5/X.135

Packet layer reference events (PEs) used in measuring data packet transfer delay

X.134 packet layer reference event / Circuit section	Starting/Ending PE
Source access circuit section	10a (X.25)
Destination access circuit section	9a (X.25)
Internetwork circuit section	5a (X.75)

The **end-to-end data packet transfer delay** is the one-way delay between DTE boundaries, e.g., B_1 and B_n in Figure 2/X.135. The **national portion data packet transfer delay** is the delay between the boundaries delimiting a national portion, e.g., B_1 and B_5 in Figure 2/X.135. The **international portion data packet transfer delay** is the delay between the boundaries delimiting an international portion, e.g., B_5 and B_{n-2} in Figure 2/X.135.

3.2 *Values*

Table 6/X.135 defines worst case data packet transfer delay values for each of the four virtual connection portion types identified in Table 1/X.135. DTE to DTE data packet transfer delay values for two hypothetical reference connections are calculated in Annex C. All values are based on (and only apply under) the following assumptions[5]:

1) Normal busy hour load conditions for the observed virtual connection. The definition of "normal busy hour load" as a traffic description is for further study.
2) A user data field length of 128 octets.
3) Data link and packet layer windows on the receiving DTE side of the portion being specified are open.

The defined values consist of mean and 95% probability values. The mean is the expected value of the data packet transfer delay distribution, excluding values that exceed a specified maximum data packet transfer delay. The 95% probability value is the value below which 95% of the data packet transfer delay values lie. Data packet transfer attempts that are unsuccessful under the conditions of Recommendation X.136 are excluded and are addressed separately in that Recommendation.

TABLE 6/X.135

Worst-case data packet transfer delay values for virtual connection portions

Statistic	Virtual connection portion type			
	National		International	
	A	B	A	B
mean (ms)	350 + Y	650 + Y	215	950
95% (ms)	525 + Y	825 + Y	215	1125

In Table 6/X.135, the value Y depends on the signalling rate of the access circuit section that is included in the national portion. Table 7/X.135 presents the Y values for user classes of service 8-11 in Recommendation X.1[6]. The Y values for other signalling rates may be computed using the formula

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

where R is the signalling rate in kilobits per second[7].

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[5] Values for other conditions are for further study. In the case of extremely long access lines and/or excessive delays in the access circuit section transmission equipment, these values may be exceeded.

[6] These Y values are not intended to represent the delay performance of the access circuit section, since these values do not include propagation delays, multiplexing delays, or the effects of retransmission.

[7] The formula assumes that the transfer of a data packet across an access circuit section involves the transmission of 136 octets: 5 octets of frame level overhead, a 3-octet packet header, and 128 octets of user data.

TABLE 7/X.135

Y-values for Table 6/X.135

X.1 user class of service	*R* (kbit/s)	*Y* (Milliseconds)
8	2.4	453
9	4.8	227
10	9.6	113
11	48.0	23

The data packet transfer delay values defined in Table 6/X.135 are intended to be used as worst-case limits in planning international packet-switched services. The actual delay performance achieved on 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 internetwork circuit sections. Variation away from the worst-case value for each factor can improve the performance.

The overall mean data packet transfer delay value for a set of concatenated virtual connection portions can be calculated directly by adding the individual portion means defined in Table 6/X.135. A method of calculating an overall 95% probability data packet transfer delay value for a set of concatenated virtual connection portions from the individual 95% probability values is described in Annex C.

4 Throughput parameters

This section defines three throughput parameters: throughput, steady-state throughput, and throughput capacity. Values are specified for throughput capacity.

4.1 **throughput** *definition*

Throughput for a virtual connection section is the number of user data bits successfully transferred in one direction across that section per unit time[8]. Successful transfer means that no user data bits are lost, added, or inverted in transfer.

Assume:

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

The X.134 PEs used in measuring throughput are the same as those used in measuring data packet transfer delay, as identified in Table 5/X.135.

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[8] User data bits are the bits of the user data field in data packets of the X.25 or X.75 packet level (protocols and data above the packet level). Framing, routing, bit stuffing, error control, and other protocol fields introduced by all protocols at or below the packet level are excluded.

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:

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

Recommendation X.136 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.

4.2 **steady-state throughput** *definition*

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 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[9].

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, ... 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 **throughput capacity** *definition*

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. 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 (see Annex B).

The end-to-end throughput capacity is the throughput capacity between DTE boundaries, e.g., B_1 and B_n in Figure 2/X.135. The national portion throughput capacity is the throughput capacity between the boundaries delimiting a national portion, e.g., B_1 and B_5 in Figure 2/X.135. The international portion throughput capacity is the throughput capacity between the boundaries delimiting an international portion, e.g., B_5 and B_{n-2} in Figure 2/X.135.

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[9] Ancillary information on steady-state throughput measurement is provided in Annex B.

4.4 *Values*

Table 8/X.135 defines worst-case throughput capacity values for each of the four virtual connection portion types identified in Table 1/X.135. DTE to DTE throughput capacity values for two hypothetical reference connections are calculated in Annex C. All values are based on (and only apply under) the following assumptions[10]:

1) Normal busy hour load conditions for the observed virtual connection. The definition of "normal busy hour load" as a traffic description is for further study. No other traffic on the access circuit sections.
2) 9600 bit/s signalling rates on the access circuit sections. Applicability of the specified throughput capacity values to lower access circuit section signalling rates is for further study.
3) A user data field length of 128 octets. Requested throughput class corresponding to 9600 bit/s. (Note that the throughput class finally applying to the call may be lower than the requested throughput class.)
4) Packet layer window sizes of 2 and data link layer window sizes of 7 on the access circuit sections.
5) D bit not used ($D = 0$).
6) Values apply to either direction of transfer.
7) No unavailability (as defined in Recommendation X.137) during the observation period.
8) No resets or premature disconnects (as defined in Recommendation X.136) during the observation period.
9) Throughput capacity sample sizes of 200 packets (in the case of the first measurement technique specified in § 4.2) or 2 minutes (in the case of the alternative measurement technique specified in § 4.2).

TABLE 8/X.135

Worst-case throughout capacity values for virtual connection portions

Statistic	Virtual connection portion type			
	National		International	
	A	B	A	B
mean (bit/s)	3000	2400·	2000	1800
95% (bit/s)	2400	2000	1800	1500

The defined values consist of mean and 95% probability values. The mean is the expected value of the throughput capacity distribution. The 95% probability value is the value above which 95% of the throughput capacity measurements lie.

The throughput capacity values defined in Table 8/X.135 are intended to be used as worse-case limits in planning international packet-switched services. The actual throughput capacity 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 internetwork circuit sections. Variation away from the worse-case value for each factor can improve the performance. The throughput capacity values defined here will not necessarily be achieved concurrently with the delay values defined in Table 6/X.135.

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[10] Values for other conditions are for further study.

An upper bound for the throughput capacity of a set of concatenated virtual connection portions can be derived from the individual portion throughput capacities as follows. If a portion between boundaries B_i and B_j has throughput capacity T_1 under load L_1, and a portion between boundaries B_k and B_m has throughput capacity T_2 under load L_2, and those portions are concatenated so that $B_j = B_k$ with L_1 and L_2 unchanged, then the resulting portion has throughput capacity.

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

Further information on estimating the throughput capacity of a set of concatenated virtual connection portions is provided in Annex C.

5 Clear indication delay

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

5.1 clear indication delay *definition*

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 X.134 PEs used in measuring clear indication delay at each section boundary are identified in Table 9/X.135.

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 9/X.135

Packet layer reference events (PEs) used in measuring clear indication delay

X.134 packet layer reference event / Circuit section	Starting/Ending PE
Clearing DTE access circuit section	6 (X.25)
Cleared DTE access circuit section	5 (X.25)
Internetwork circuit section	3 (X.75)

The **end-to-end clear indication delay** is the one-way delay between DTE boundaries, e.g., B_1 and B_n in Figure 2/X.135. The **national portion clear indication delay** is the delay between the boundaries delimiting a national portion, e.g., B_1 and B_5 in Figure 2/X.135. The **international portion clear indication delay** is the delay between the boundaries delimiting an international portion, e.g., B_5 and B_{n-2} in Figure 2./X.135.

5.2 *Values*

Table 10/X.135 defines worst case clear indication delay values for each of the four virtual connection portion types identified in Table 1/X.135. DTE to DTE clear indication delay values for two hypothetical reference connections are calculated in Annex C. All values are based on (and only apply under) the following assumptions[11]:

1) Normal busy hour load conditions for the observed virtual connection. The definition of "normal busy hour load" as a traffic description is for further study.
2) Data link layer windows on the cleared DTE side of the portion being specified are open.
3) The extended format of the clear request packet is not used.

TABLE 10/X.135

Worst-case clear indication delay values for virtual connection portions

Statistic	Virtual connection portion type			
	National		International	
	A	B	A	B
mean (ms)	500 + Z	800 + Z	110	800
95% (ms)	750 + Z	1050 + Z	110	900

The defined values consist of mean and 95% probability values. The mean is the expected value of the clear indication delay distribution, excluding values that exceed a specified maximum clear indication delay. The 95% probability value is the value below which 95% of the clear indication delay values lie. Unsuccessful call clear attempts are excluded and are addressed separately in Recommendation X.136.

In Table 10/X.135, the value Z depends on the signalling rate of the access circuit section that is included in the national portion. Table 11/X.135 presents the Z values for user classes of service 8-11 in Recommendation X.1[12].

The Z values for other signalling rates may be computed using the formula

$$Z = 80/R \text{ ms}$$

where R is the signalling rate in kilobits per second[13].

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[11] Values for other conditions are for further study. In the case of extremely long access lines and/or excessive delays in the access circuit section transmission equipment, these values may be exceeded.

[12] These Z values are not intended to represent the delay performance of the access circuit section, since these values do not include propagation delays, multiplexing delays, or the effects of retransmission.

[13] The formula assumes that the transfer of each call clearing packet across an access circuit section involves the transmission of 10 octets: 5 octets of frame level overhead and 5 octets of packet header information.

TABLE 11/X.135

Z-values for Table 10/X.135

X.1 user class of service	*R* (kbit/s)	*Z* (Milliseconds)
8	2.4	34
9	4.8	17
10	9.6	9
11	48.0	2

The clear indication delay values defined in Table 10/X.135 are intended to be used as worst-case values in planning international packet-switched services. The actual delay performance achieved on 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 internetwork circuit sections. Variation away from the worst-case value for each factor can improve the performance.

The overall mean clear indication delay value for a set of concatenated virtual connection portions can be calculated directly by adding the individual portion means defined in Table 10/X.135. A method of calculating an overall 95% probability clear indication delay value for a set of concatenated virtual connection portions from the individual 95% probability values is described in Annex C.

ANNEX A

(to Recommendation X.135)

Factors to be specified in reporting throughput performance

Many factors affect the throughout capacity that can be obtained on a virtual connection section.

A.1 *Signalling rates*

The choice of signalling rates on circuit sections bounds throughput. In general, faster signalling rates improve throughput.

A.2 *Interface windows*

The choice of window size has an effect on throughput. In general, larger window sizes improve throughput. For maximum throughput, each user-controllable window size should be optimized with respect to delays and retransmission rates.

A.3 *Packet length*

The choice of packet length has an effect on throughput. In general, the use of larger packets improves throughput. For maximum throughput, packet sizes should be optimized with respect to the known error properties of the access links.

A.4 *Additional virtual connections*

Throughput of a tested virtual connection is dependent on the number of additional virtual connections and the loading in each direction on each connection. Throughput per virtual connection decreases as the number of additional virtual connections or the loading on the individual connections increases. When stating the throughput capacity of a virtual connection portion, the number of additional active virtual connections on the access circuit sections should be specified. Also, the total throughput in each direction on those virtual connections should be reported. For example:

> "The throughput capacity of a virtual connection on this international portion is at least 1.2 kbit/s. There can be at most 4 additional virtual connections transmitting in the same direction between the same two portion boundaries at the same throughput."

A.5 *Time-of-day*

When measuring throughput it is assumed that the loads on many connection components cannot be user controlled or observed. However, it is assumed that those loads are correlated with time-of-day, day-of-week, and holidays. Thus users can improve their throughput by transmitting at particular times.

A.6 *Direction*

If the direction of the measurement affects the throughput capacity, the direction should be specified when stating throughput capacity. Otherwise the capacities in the two directions will be assumed to be equal.

A.7 *Throughput class*

Network internal windows and acknowledgement schemes may or may not be a function of a virtual connection's requested or default throughput class. For maximum throughput and when measuring throughput capacity, the throughput class for the virtual connection should be set to the maximum allowed by the section being measured. Because the optimum throughput class is always the maximum allowable, a statement of throughput capacity need not explicitly specify the throughput class.

A.8 *D bit usage*

If the *D* bit is set to 1 during a throughput measurement, that fact should be reported. Otherwise, the *D* bit setting need not be reported.

A.9 *Delay*

Throughput and data packet transfer delay are related. If the throughput is specified under a delay constraint, then the delay should be reported.

A.10 *Reporting throughput capacity*

Throughput capacity reports should specify the values of the controllable factors that were in effect during the throughput capacity measurement. All factors listed in this Annex should be reported unless otherwise specified. A typical report might specify conditions as follows:

> "For this connection the network throughput capacity is at least 4.1 kbit/s. The capacity was measured using two 9.6 kbit/s access circuit sections, data link layer window sizes of 7, packet layer window sizes of 2, and 128 octet user data fields. No additional virtual connections were present on either of the access circuit sections. The capacity was measured during the busiest hour of the weekday. The average data packet transfer delay during the measurement period was 500 milliseconds. The precision of the throughput measurement is plus or minus 0.1 kbit/s."

With such statements, the throughput capacity is more easily verified and more easily matched to the throughput needs of potential users.

ANNEX B

(to Recommendation X.135)

Ancillary information on throughput measurement and the application of throughput capacity values

The following points should be noted with regard to throughput measurement:

- A measurement of steady-state throughput requires a measurement size of $k = 200$ packets. An alternative is to specify a value for the measurement time period $(t_2 - t_1)$ of 2 minutes.
- When measuring steady-state throughput, data packets A_1 through A_k need not constitute a single complete packet sequence.
- One way of verifying successful transfer of the test sequence in a steady-state throughput measurement is to transfer another complete packet sequence.
- Throughput-related measurements should not be conducted with user data sequences with high density of binary "ones" to avoid biasing the results by the effects of bit stuffing.

The following describes one way of applying the throughput capacity parameter. The discussion uses throughput capacity to design an international circuit section.

Assuming:

m = the mean throughput per call (for the duration of the call)
n = the total number of calls present at any time
p = the number of those calls requiring the throughput capacity at any instant in time
b = the bit rate of the international internetwork circuit section and
T = the throughput capacity objective per call

Then the bit rate b should be:

$$b \geqslant (m * n) + p(T - m)$$

The actual m, n, and p values may be network dependent and reflect basically the population of the access line speeds and their traffic characteristics. It is therefore recommended that the value of b is chosen considerably higher than the value of $(m * n)$. The number of logical channels assigned to international internetwork links should depend on the relationship of the values b and m.

ANNEX C

(to Recommendation X.135)

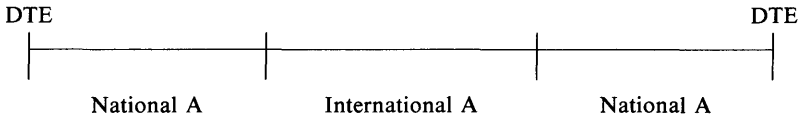
Representative end-to-end speed of service performance

This Annex provides two examples to illustrate how end-to-end (DTE to DTE) speed of service performance can be estimated from the individual virtual connection portion performance values specified in X.135. Two example concatenations of Type A and Type B virtual connection portions are defined. The end-to-end call set-up delay, data packet transfer delay, throughput capacity, and clear indication delay are calculated for each example. 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 individual network portions.

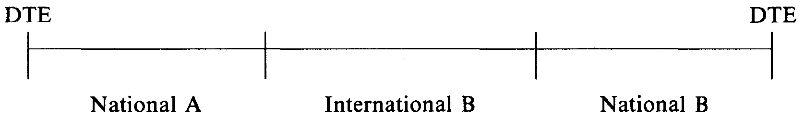
C.1 *Definition of the example end-to-end connections*

For ease of reference the two example end-to-end (i.e., DTE to DTE) connections presented in this Annex will be referred to as "Type 1" and "Type 2" configurations. These hypothetical, but representative, configurations use the portion boundaries and packet layer reference events described in X.134. Figure 2/X.135 shows the relevant network boundaries and Table 1/X.135 defines the virtual connection portion types.

The Type 1 configuration is defined to be:



The Type 2 configuration is defined to be:



C.2 *End-to-end speed of service performance for the Type 1 and Type 2 configuration examples*

End-to-end speed of service performance values have been calculated for the example Type 1 and Type 2 connection configurations and are reported below in Tables C-1/X.135 and C-2/X.135. These calculations have been made by applying the methods derived in § C.3 (below) to the individual network portions that, for convenience in defining these examples, are characterized by the worst-case speed of service performance values specified in X.135.

The end-to-end performance for the mean call set-up delay, data packet transfer delay, and clear indication delay are computed by simply summing the mean delays associated with the appropriate individual network portions.

Example – For the Type 1 configuration the end-to-end mean call set-up delay in milliseconds is computed by referring to Table 3/X.135 and adding the mean values for the National A and International A portion types:

$$(1000 + X) + (250) + (1000 + X) = 2250 + 2 \ast X$$

The end-to-end performance for the 95th percentile call set-up delay, data packet transfer delay, and clear indication delay can be determined by assuming (see § C.3) that the variance of the end-to-end delay is the sum of the variances of the individual network portion delays.

Example – For the Type 1 configuration, referring to Table 3/X.135 and § C.3, the 95th percentile value for the end-to-end call set-up delay in milliseconds is:

$$(2250 + 2 \ast X) + [((1500 + X) - (1000 + X))^2 + ((250) - (250))^2 + ((1500 + \mathrm{X}) - (1000 + X))^2]^{0.5}$$
$$= 2957 + 2 \ast X$$

The end-to-end performance for the mean and 95th percentile for throughput capacity are determined by assuming that:

1) the end-to-end throughput at any particular time is the minimum taken over all the individual network portions; and
2) the throughput of an individual network portion is an independent and normally distributed random variable. § C.3 derives formulas that combine the overlapping individual probability distributions to give the end-to-end throughput capacity distribution.

Example – Numerical computations of the end-to-end mean and 95th percentile throughput capacities for the Type 1 and Type 2 configurations are provided as examples in § C.3.2.

TABLE C-1/X.135

End-to-end speed of service performance for the type 1 configuration example

Statistic	Type 1 configuration	
	Mean	95%ile
Call set-up delay (ms)	$2250 + 2 * X$	$2957 + 2 * X$
Data packet transfer delay (ms)	$915 + 2 * Y$	$1162 + 2 * Y$
Throughput capacity (bit/s)	1999	1800
Clear indication delay (ms)	$1110 + 2 * Z$	$1464 + 2 * Z$

TABLE C-2/X.135

End-to-end speed of service performance for the Type 2 configuration example

Statistic	Type 2 configuration	
	Mean	95%ile
Call set-up delay (ms)	$4200 + 2 * X$	$4935 + 2 * X$
Data packet transfer delay (ms)	$1950 + 2 * Y$	$2284 + 2 * Y$
Throughput capacity (bit/s)	1797	1500
Clear indication delay (ms)	$2100 + 2 * Z$	$2467 + 2 * Z$

The parameters X, Y and Z depend on the signalling rate of the access circuit section that is included in the national portion. Definitions, relevant assumptions, and values for X, Y, and Z can be found in the appropriate sections of X.135. As noted in § 4.4 of X.135, a 9.6 kbit/s signalling rate for the access circuit sections is assumed for the worst-case throughput capacity performance values.

C.3 *Methods for calculating mean and 95% points of delays and throughputs of packet-switched services with two or more concatenated portions*

This section describes the methods used in calculating end-to-end speed of service performance from individual network portion performance values.

C.3.1 *Delays*

It is assumed that a packet-switched service has n portions with delays $d_1, d_2, \ldots, d_n$ varying randomly with means $m_1, m_2, \ldots, m_n$ and 95% points $z_1, z_2, \ldots, z_n$. Then the total delay $D = d_1 + d_2 + \ldots + d_n$ has a distribution with mean

$$M = m_1 + m_2 + \ldots + m_n$$

(with no further assumption). In order to obtain the 95% point of D it is assumed that the delays d_i are statistically independent and that $z_i = m_i + k\sigma_i$ with the same k for all portions, where σ_i is the standard deviation of d_i. The like equality is also assumed for D, i.e., $Z = M + k\sigma_D$, where Z is the 95% point of D. These equalities are true for normal distributions with $k = 1.645$. Then the variance of D is the sum of the variances of the d_i. It follows that the 95% point of D is given by

$$Z = M + [(z_1 - m_1)^2 + (z_2 - m_2)^2 + \ldots + (z_n - m_n)^2]^{1/2}$$

The assumption of normality seems reasonable, but other assumptions are possible and could give substantially different answers.

C.3.2 *Throughputs*

It is assumed that a packet-switched service has n portions with throughputs $T_1, T_2, \ldots, T_n$ varying randomly and independently with means $M_1, M_2, \ldots, M_n$ and 5% points (points exceeded by 95% of the values) $Z_1, Z_2, \ldots, Z_n$. The net throughput of the service is assumed to be $V = \min (T_1, T_2, \ldots, T_n)$. The cumulative distribution function (cdf) of T_i is the probability that T_i is less than or equal to any value, say t, and is denoted by $F_i(t)$:

$$F_i(t) = \text{Prob}\,[T_i \leqslant t],\ i = 1, 2, \ldots, n$$

The probability density function (pdf) of T_i is the derivative of $F_i(t)$ and is denoted by $f_i(t) = dF_i/dt$.

In order to calculate the mean, say M_{V_n}, and the 5% point, $V_{.05,\,n}$, of the net throughput V, it is in general not sufficient to consider just the portion M_i's and Z_i's; it is necessary to combine the entire distributions $F_i(t)$ (or $f_i(t)$) to obtain the pdf of V, to be denoted by $g_n(v)$. However, in the important special case that the portion with the usually smallest throughput (the "slowest portion") has a distribution that is not overlapped at all by the distributions of the larger throughputs, then the net throughput distribution is identical with that of the slowest portion, having the same mean and 5% point in particular. If the overlap of any other distribution with the slowest portion's distribution is negligible, then the same conclusion can be drawn. Later examples will suggest how much overlap can be considered negligible.

The case of general distributions is now resumed, that with $n = 2$ at first. Integration in the two dimensions of (T_1, T_2) shows that the pdf of V is given by

$$g_2(v) = f_1(v)\,[1 - F_2(v)] + f_2(v)\,[1 - F_1(v)] \qquad \text{(C-1)}$$

The mean net throughput of the service is then

$$M_{V2} = \int_0^\infty v g_2(v)\, dv \tag{C-2}$$

and the 5% point is the value $V_{.05,\,2}$ such that

$$\int_0^{V_{.05,\,2}} g_2(v)\, dv = 0.05 \tag{C-3}$$

If $f_1(t) = f_2(t)$, then

$$g_2(v) = 2 f_1(v)\,[1 - F_1(v)] \tag{C-4}$$

It is now assumed that the portion throughput distributions are normal and that they are sufficiently concentrated that the tail of the fitted normal distribution to the left to zero is negligible (as is true for all the numerical values in X.135). The assumption is expressed in terms of the standard normal pdf $\varphi(u)$ and cdf $\Phi(x)$:

$$\varphi(u) = \frac{1}{\sqrt{2\pi}}\, e^{-u^2/2},\ \Phi(x) = \int_{-\infty}^{x} \varphi(u)\, du \tag{C-5}$$

Then

$$f_i(t) = \frac{1}{\sigma_i}\, \varphi\left(\frac{t - M_i}{\sigma_i}\right),\ F_i(t) = \int_{-\infty}^{t} f_i(y)\, dy \tag{C-6}$$

where the standard deviation $\sigma_i = (M_i - Z_i)/1.64485$. In the case $f_1(t) = f_2(t)$, then

$$g_2(v) = \frac{2}{\sigma_1}\, \varphi\left(\frac{v - M_1}{\sigma_1}\right) \left[1 - \Phi\left(\frac{v - M_1}{\sigma_1}\right)\right] \tag{C-7}$$

The case $n = 3$ is now considered. The pdf $g_3(v)$ of $V_3 = \min\,(T_1,\ T_2,\ T_3)$ can be obtained by iteration on the distribution of $V_2 = \min\,(T_1,\ T_2)$ since $V_3 = \min\,(V_2,\ T_3)$. Hence

$$g_3(v) = g_2(v)\,[1 - F_3(v)] + f_3(v)\,[1 - G_2(v)] \tag{C-8}$$

where $g_2(v)$ is given by (C-1) and $G_2(v)$ is its indefinite integral,

$$G_2(v) = \int_0^{v} g_2(x)\, dx \tag{C-9}$$

If all three pdf's $f_i(t)$ are identical, the $g_3(v)$ simplifies to

$$g_3(v) = 3 f_1(v)\,[1 - F_1(v)]^2 \tag{C-10}$$

Normal as well as identical distributions are now assumed. Then, from (C-5), (C-6), and (C-10),

$$\begin{aligned} M_{V3} &= \int_0^\infty v g_3(v)\, dv \\ &= M_1 + 3\sigma_1 \int_{-\infty}^{\infty} u\, \varphi(u)\, [1 - \Phi(u)]^2\, du \\ &= M_1 - 3\sigma_1 \int_0^{\infty} u\, \varphi(u)\, [2\, \Phi(u) - 1]\, du \\ &= M_1 - \sigma_1 K_3, \end{aligned} \qquad \text{(C-11)}$$

where $K_3 = 0.8463$ by Teichroew (1956). Likewise

$$V_{.05,\,3} = M_1 + \sigma_1\, U_{.05,\,3} \qquad \text{(C-12)}$$

where

$$3 \int_{-\infty}^{U_{.05,\,3}} \varphi(u)\, [1 - \Phi(u)]^2\, du = 0.05 \qquad \text{(C-13)}$$

By integration

$$\Phi(-\,U_{,05,\,3}) = 1 - 0.095^{1/3} = 0.016952 \qquad \text{(C-14)}$$

Hence from any cumulative normal distribution table, $U_{.05,\,3} = 2.121$.

Example 1 – Calculate the mean and 95th percentile net throughputs assuming there are three identical and normal portion distributions with $M_1 = M_2 = M_3 = 2000$ bit/s and $Z_1 = Z_2 = Z_3 = 1800$ bit/s. Then $\sigma_1 = \sigma_2 = \sigma_3 = 2000/1.645 = 121.6$ bit/s. From (C-11):

$$M_{V3} = 2000 - 121.6 \times 0.8463 = 1897 \text{ bit/s}$$

From (C-12) and (C-14)

$$V_{.05,\,3} = 2000 - 121.6 \times 2.121 = 1742 \text{ bit/s}$$

Example 2 – Consider the Type 1 configuration. From Table 8/X.135, $M_1 = M_2 = 3000$ bit/s, $M_3 = 2000$ bit/s, $Z_1 = Z_2 = 2400$ bit/s, $Z_3 = 1800$ bit/s. With normal distributions there is slight but probably negligible overlap of the larger throughputs with the smallest throughput; the probability of either national throughput being less than or equal to the *upper* 5% point of the international throughput, 2200 bit/s, is 0.014. Hence, at least approximately, $M_{V3} = M_3 = 2000$ bit/s, $V_{.05,\,3} = Z_3 = 1800$ bit/s.

This can be checked by numerical integration. Since this will come up in other applications, general formulas are given here. When $f_1(v) = f_2(v)$, $G_2(v)$ in (C-9) becomes

$$G_2(v) = 2F_1(v) - [F_1(v)]^2$$

When the distributions are also normal, it follows from (C-8) and (C-5) that

$$g_3(v) = \left[1 - \Phi\left(\frac{v - m_1}{\sigma_1}\right)\right] \left\{\frac{2}{\sigma_1}\, \varphi\left(\frac{v - m_1}{\sigma_1}\right) \left[1 - \Phi\left(\frac{v - m_3}{\sigma_3}\right)\right] + \right.$$
$$\left. + \frac{1}{\sigma_3}\, \varphi\left(\frac{v - m_3}{\sigma_3}\right) \left[1 - \Phi\left(\frac{v - m_1}{\sigma_1}\right)\right]\right\} \qquad \text{(C-15)}$$

Hence the mean throughput for a three-portion network with two portions identical is, with the change of variable $u = (v - m_1)/\sigma_1$,

$$M_{V3} = \int_{-\infty}^{\infty} (m_1 + \sigma_1 u)[1 - \Phi(u)] \left\{ Z\varphi(u)\left[1 - \Phi\left(\frac{m_1 - m_3 + \sigma_1 u}{\sigma_3}\right)\right] + \right.$$

$$\left. + \frac{\sigma_1}{\sigma_3}\varphi\left(\frac{m_1 - m_3 + \sigma_1 u}{\sigma_3}\right)[1 - \Phi(u)]\right\} du \tag{C-16}$$

This can be integrated numerically using a pocket calculator and the National Bureau of Standards *Tables of Normal Probability Functions.* Since these tables give the integral of $\varphi(u)$ from $-x$ to x, say $S(x)$, rather than $\Phi(x)$, the following substitution is made in (C-16) (in three places):

$$1 - \Phi(u) = \begin{cases} [1 - S(u)]/2 & \text{if } u \geqslant 0 \\ [1 + S(|u|)]/2 & \text{if } u < 0 \end{cases} \tag{C-17}$$

In the above Example 2, (C-16) becomes

$$\frac{2M_{V3}}{\sigma_1} = \int_{-\infty}^{\infty} (8.225 + u)\,[1 \pm S(|u|)]\,\{\varphi(u)[1 \pm S(|8.225 + 3u|)]$$

$$+ 1.5\,\varphi\,(8.225 + 3u)[1 \pm S(|u|)]\}\,du$$

Numerical integration with $\Delta u = 0.1$ and the Trapezoidal Rule yields $M_{V3} = 1999.09$ bit/s. With Simpson's Rule $M_{V3} = 1999.11$ bit/s. Hence the slight overlap of the distributions of the two larger throughputs with the smaller throughput distribution reduces the mean net throughput by less than 1 bit/s. The effect on the lower 5% point will be much less, so $V_{.05,\,3} = 1800$ bit/s. However, comparison with Example 1 shows that *complete* overlap of three portion distributions does reduce the throughput substantially below that of an individual portion.

Example 3 – Consider the Type 2 configuration. From Table 8/X.135, $M_1 = 3000$, $M_2 = 2400$, $M_3 = 1800$, $Z_1 = 2400$, $Z_2 = 2000$, $Z_3 = 1500$ (all bit/s). Three non-identical portions result in an integral substantially messier than (16). It could be programmed on a computer, but that is unnecessary because a tight bound can be obtained by replacing the fastest portion by one identical with the next faster portion and using (16). Doing so with $\Delta u = 0.1$ and the Trapezoidal Rule gives $M_{V3} = 1794.4$ bit/s; the more accurate Simpson's Rule gives $M_{V3} = 1794.7$ bit/s. Since M_{V3} must be less than or equal to $M_3 = 1800$ bit/s, the mean throughput with the original three non-identical portions is bounded by 1795 and 1800 bit/s. It is estimated as 1797 bit/s with an error probably no more than 1 bit/s. The effect on the lower 5% point will be even less; numerical integration with $\Delta u = 0.1$ gives $V_{.05,\,3} = 1499.2$ bit/s when the fastest portion is replaced by one identical with the next faster portion, so it is estimated that the original network has $V_{.05,\,3} = 1500$ bit/s to the nearest unit.

These examples suggest the following when the smallest throughput distribution is not greatly overlapped by others, and this applies no matter how many portions there are:

General Rule – If the mean throughput of the slowest portion is less than the mean of the next slowest portion by at least twice the difference between the mean and 95%ile of the slowest portion or of the next slowest portion, whichever difference is larger, then the mean and 95%ile of the throughput of the network are the same as those of the slowest portion (with negligible error). (This rule can probably be relaxed by replacing "twice" by "1.5 times" or deleting "twice" without incurring too much error in practice.)

The case of general n is considered similarly. With different distributions $f_i(t)$ the pdf $g_n(v)$ of $V_n = \min(T_1, T_2, \ldots, T_n)$ is obtainable by iteration from $g_{n-1}(v)$:

$$g_n(v) = g_{n-1}(v)\,[1 - F_n(v)] + f_n(v)\,[1 - G_{n-1}(v)]$$

If all $f_i(t)$ are identical, then

$$g_n(v) = nf_i(v)\,[1 - F_i(v)]^{n-1}$$

If, in addition, normal distributions are assumed for the $f_i(t)$, then the mean net throughput is

$$\begin{aligned} M_{Vn} &= M_1 + n\,\sigma_1 \int_{-\infty}^{\infty} u\,\varphi(u)\,[1 - \Phi(u)]^{n-1}\,du, \\ &= M_1 - n\,\sigma_1 \int_{0}^{\infty} u\,\varphi(u)\,\{\Phi^{n-1}(u) - [1 - \Phi(u)]^{n-1}\}\,du \\ &= M_1 - K_n\sigma_1 \qquad \text{(C-18)} \end{aligned}$$

and the 5% point of the net throughput is

$$V_{.05,\,n} = M_1 - \sigma_1\,U_{.05,\,n} \qquad \text{(C-19)}$$

where

$$\Phi(-V_{.05,\,n}) = 1 - 0.95^{1/n} \qquad \text{(C-20)}$$

The values K_n and $U_{.05,\,n}$ can be tabulated as a function of n:

n	1	2	3	4	5
K_n	0	0.5642	0.8463	1.0294	1.1630
$U_{.05,\,n}$	1.645	1.955	2.121	2.234	2.319

C.4 *Notes on key assumptions, results, and implications*

For further study.

Reference

[1] TEICHROEW, D., Tables of expected values of order statistics and products of order statistics for samples of size twenty and less from the normal distribution, *Annals of Mathematical Statistics*, **27**, pp. 410-426, 1956.

ACCURACY AND DEPENDABILITY PERFORMANCE VALUES FOR PUBLIC DATA NETWORKS WHEN PROVIDING INTERNATIONAL PACKET-SWITCHED SERVICES

(Malaga-Torremolinos, 1984; amended at Melbourne, 1988)

The CCITT,

considering

(a) that Recommendation X.1 specifies the international user classes of service in public data networks;

(b) that Recommendation X.2 specifies the international data transmission services and optional user facilities in public data networks;

(c) that Recommendation X.25 specifies the DTE/DCE interface for packet mode terminals connected to public data networks by dedicated circuit;

(d) that Recommendation X.75 specifies the packet-switched signalling system between public data networks providing data transmission services;

(e) that Recommendation X.323 specifies general arrangements for interworking between packet-switched public data networks;

(f) that Recommendation X.96 specifies call progress signals in public data networks;

(g) that Recommendation X.110 specifies the international routing principles and routing plan for public data networks;

(h) that Recommendation X.213 defines the OSI Network Layer service;

(i) that Recommendation X.140 defines general quality of service parameters for communication via public data networks;

(j) that Recommendation X.134 specifies portion boundaries and packet layer reference events for defining packet-switched performance parameters;

(k) that Recommendation X.135 specifies speed of service performance values for public data networks when providing international packet-switched service;

(l) that Recommendation X.137 specifies availability performance values for public data networks when providing international packet-switched service,

unanimously declares

(1) that the accuracy and dependability parameters defined in this Recommendation shall be used in the planning and operation of international packet-switched data communication services provided in accordance with Recommendations X.25 and X.75;

(2) that in such services, the performance values specified in this Recommendation shall be taken as worst-case limits under the conditions specified herein.

1 Introduction

1.1 This Recommendation is the third in a series of four CCITT Recommendations (X.134-X.137) that define performance parameters and values for international packet-switched data communication services. Figure 1/X.136 illustrates the scope of these four Recommendations and the relationships among them.

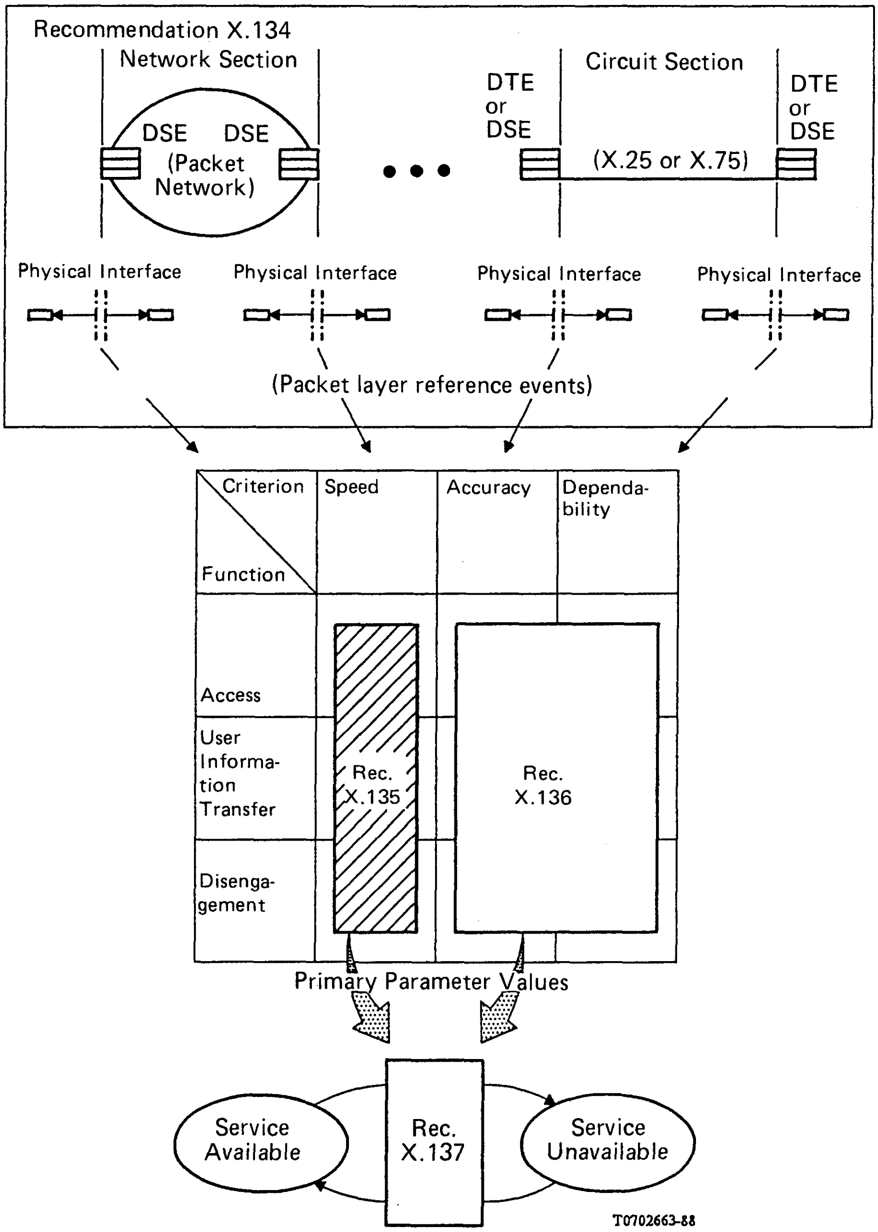


FIGURE 1/X.136

Packet-switched service performance description framework

1.2 Recommendation X.134 divides a virtual connection into basic sections whose boundaries are associated with X.25 and X.75 interfaces; defines particular collections of basic sections, called virtual connection portions, for which performance values will be specified; and defines a set of packet layer reference events (PEs) which provide a basis for performance parameter definition. The basic sections consist of network sections and circuit sections. They are delimited, in each case, by physical data terminal equipment (DTE) or data switching equipment (DSE) interfaces. Virtual connection portions are identified either as national portions or international portions. Each PE is defined to occur when a packet crossing a section boundary changes the state of the packet layer interface.

1.3 For comparability and completeness, packet-switched network performance is considered in the context of the 3 × 3 performance matrix defined in Recommendation X.140. Three protocol-independent data communication functions are identified in the matrix: access, user information transfer, and disengagement. These general functions correspond to call set-up, data (and interrupt), transfer, and call clearing in packet-switched virtual call services conforming to the X.25 and X.75 Recommendations. Each function is considered with respect to three general performance concerns (or "performance criteria"): speed, accuracy, and dependability. These express, respectively, the delay or rate, degree of correctness, and degree of certainty with which the function is performed.

1.4 Recommendation X.135 defines protocol-specific speed of service parameters and values associated with each of the three data communication functions. This Recommendation defines protocol-specific accuracy and dependability parameters and values associated with each function. The Recommendation X.135 and X.136 parameters are called "primary parameters" to emphasize their direct derivation from packet layer reference events.

1.5 An associated two-state model provides a basis for describing overall service availability. A specified availability function compares the values for a subset of the primary parameters with corresponding outage thresholds to classify the service as "available" (no service outage) or "unavailable" (service outage) during scheduled service time. Recommendation X.137 specifies the availability function and defines the availability parameters and values that characterize the resulting binary random process.

1.6 Eight accuracy and dependability parameters are defined in this Recommendation: two access parameters (call set-up error probability and call set-up failure probability), five user information transfer parameters (residual error rate, reset stimulus probability, reset probability, premature disconnect stimulus probability, and premature disconnect probability), and one disengagement parameter (call clear failure probability). Each parameter can be applied to any basic section or portion of a virtual connection. This generally makes the parameters useful in performance allocation and concatenation.

1.7 This Recommendation specifies accuracy and dependability values for national and international portions of two types (Table 1/X.136). Performance values for data terminal equipment are not specified, but the parameters defined in this Recommendation may be employed in such specification to assist users in establishing quantitative relationships between network performance and quality of service (see Recommendation X.140).

TABLE 1/X.136

Virtual connection portion types for which performance values are specified [a)]

Portion type	Typical characteristics
National A	Terrestrial connection via an access network section
National B	Connection via an access network section with one satellite circuit; or via an access network section and one or more transit network sections
International A	Connection via a direct terrestrial internetwork section
International B	Connection via two satellite circuits and one transit network section; or via one satellite circuit and two or more transit network sections

[a)] The values specified for Type B portions also apply to virtual connection portions not explicitly identified as Type A or Type B.

1.8 Worst-case values for each of the eight accuracy and dependability parameters are specified below for each virtual connection portion type identified in Table 1/X.136. The term "worst case" means that these values should be met during the normal busy hour in the worst-performing virtual connection portion used in providing international packet-switched services. The performance of a virtual connection portion may be better than the worst-case values specified in this Recommendation. Design objectives that take into account more demanding user applications and network performance and connectivity enhancements are for further study.

Numerical methods for combining individual portion performance values to estimate end-to-end performance are also provided in this Recommendation. DTE to DTE values for two particular hypothetical reference connections are derived using these methods in Annex B.

2 Access parameters

This section specifies worst-case values for two access parameters: call set-up error probability and call set-up failure probability.

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 2/X.136 identifies the sequence of four particular events that occur at these boundaries during a successful call set up[1]. 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 timeout period[2]. 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.

2.1 *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 Recommendations X.25 and X.75.

2.1.1 **call set-up error probability** *definition*

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 2/X.136, 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 timeout 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.

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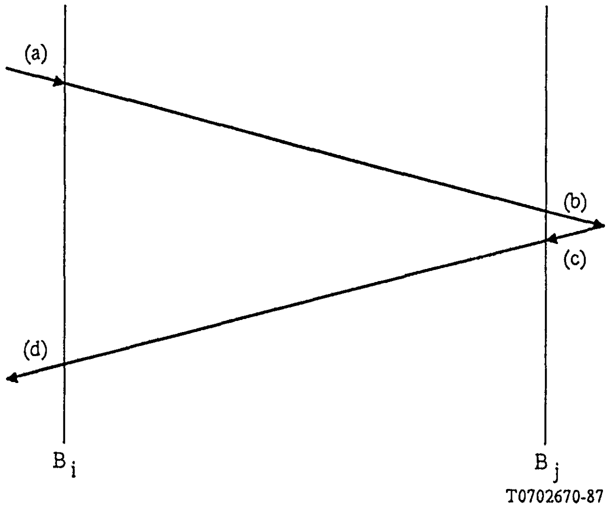
[1] The PE numbers in Figure 2a/X.136 refer to Tables 1 and 2 in Recommendation X.134.

[2] This period corresponds to timer T21 in Recommendation X.25.

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

a) Packet layer reference events (PEs)



b) Event sequence

FIGURE 2/X.136

Packet layer reference events occurring during successful call set-up

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.

Call set-up error probability does not apply to the fast select mode of data transfer. The optional user call redirection facilities in X.25 (including hunt group, call redirection, call forwarding subscription, call forwarding selection, call redirection or forwarding notification, and called line address modified notification) are assumed not to be used in the calculation of this parameter.

The specific X.134 PEs used in measuring call set-up error probability at each section boundary are those identified in Figure 2/X.136.

2.1.2 *Values*

The contribution from each network portion to the overall call set-up error probability under the conditions described in this Recommendation shall not exceed the values specified in Table 2/X.136.

TABLE 2/X.136

Worst-case call set-up error probability values for virtual connection portions

Statistic	Virtual connection portion type			
	National		International	
	A	B	A	B
Probability	10^{-5}	2×10^{-5}	[a]	2×10^{-5}

[a] The Type A international virtual connection portion consists only of a physical circuit. Its contribution to call set-up error probability is expected to be negligible.

Note – All specified values are provisional.

2.2 *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 Recommendations X.25 and X.75.

2.2.1 **call set-up failure probability** *definition*

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 2/X.136, 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 timeout period[3]:

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 X.134 PEs used in measuring call set-up failure probability at each section boundary are those identified in Figure 2/X.136.

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[3] Recommendation X.96 places limits on the frequency at which a DTE can repeat call attempts to a given destination.

2.2.2 *Values*

The contribution from each network portion to the overall call set-up failure probability under the conditions described in this Recommendation shall not exceed the values specified in Table 3/X.136.

TABLE 3/X.136

Worst-case call set-up failure probability values for virtual connection portions

Statistic	Virtual connection portion type			
	National		International	
	A	B	A	B
Probability	5×10^{-3}	10^{-2}	[a]	10^{-2}

[a] The Type A international virtual connection portion consists only of a physical circuit. Its contribution to call set-up failure probability is expected to be negligible.

Note – All specified values are provisional.

2.2.3 *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 timeout.
3) All logical channels at the called DTE are in use.

3 User information transfer parameters

This section specifies worst-case values for five user information transfer parameters; residual error rate, 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.

3.1 *Residual error rate*

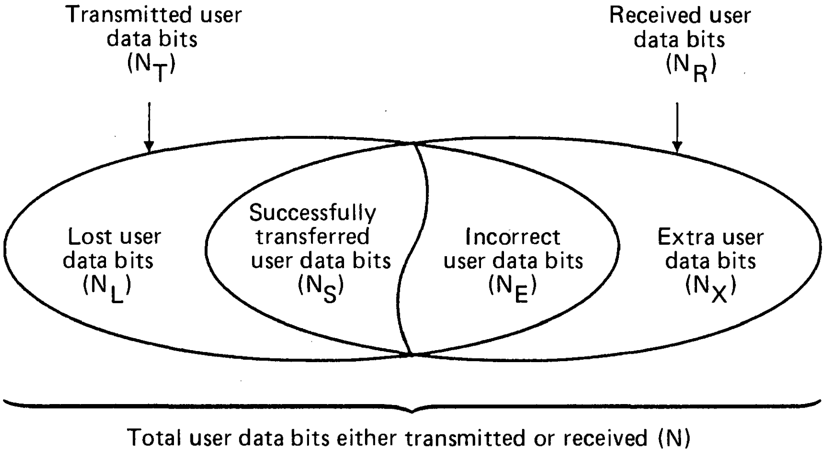
Residual error rate applies to both virtual call and permanent virtual circuit services. This parameter is used to measure the accuracy of the general function of user information transfer in public packet-switched services conforming to Recommendations X.25 and X.75.

3.1.1 **residual error rate** *definition*

Residual error rate is the ratio of total incorrect, lost, and extra (e.g. duplicate) user data bits to total user data bits transferred across either section boundary in a population of interest.

User data bits are the bits of the user data field in data packets of the X.25 or X.75 packet layer (protocols and 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.

Relationships among the quantities identified above are defined in Figure 3/X.136. Incorrect user data bits are user data bits that are inverted in transfer between the section boundaries, i.e., bits whose binary value observed at the section boundary on the destination side of a virtual connection portion 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 portion at one section boundary, but are not transferred out of the virtual connection portion 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 rate. Extra user data bits are user data bits that are transferred out of a virtual connection portion at one section boundary, but were not previously transferred into the virtual connection portion at the other. Extra user data bits include duplicated user data bits and misdelivered user data bits.



$$RER = \frac{N_E + N_L + N_X}{N}$$

$$N = N_L + N_S + N_E + N_X$$

FIGURE 3/X.136

Components of residual error rate

The specific X.134 PEs used in measuring residual error rate at each section boundary are identified in Table 4/X.136. Only user data bits in data packets that create the specified PEs are counted in calculating residual error rate estimates.

TABLE 4/X.136

Packet layer reference events (PEs) used in measuring residual error rate

Circuit section	Starting/Ending PE
Source access circuit section	10a (X.25)
Destination access circuit section	9a (X.25)
Internetwork circuit section	5a (X.75)

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. A simple, approximate method of calculating residual error rate values is presented in Annex A. Other methods of equivalent or superior accuracy are acceptable.

3.1.2 *Values*

The contribution from each network portion to the overall residual error rate of a virtual connection provided under the conditions described in this Recommendation shall not exceed the values specified in Table 5/X.136. This specified values are based on an assumed data packet length of 128 octets.

TABLE 5/X.136

Worst-case residual error rate values for virtual connection portions

Statistic	Virtual connection portion type			
	National		International	
	A	B	A	B
Probability	10^{-10}	2×10^{-10}	[a]	2×10^{-10}

[a] The Type A international virtual connection portion consists only of a physical circuit. Its contribution to residual error rate is expected to be negligible.

Note – All specified values are provisional.

3.1.3 *Components of residual error rate*

In some applications, it may be important to specify probability limits for the individual failure outcomes illustrated in Figure 3/X.136 in addition to the overall residual error rate. The general user information error, user information loss, and extra user information delivery probabilities defined in Recommendation X.140 may be specialized to the corresponding user data bit-oriented measures as follows.

- 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.
- 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.
- 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; i.e., each failure outcome is expressed in proportion to the total number of opportunities for that outcome to occur. The mathematical relationship between residual error rate (*RER*) and the three user data bits transfer failure probabilities defined above is as follows.

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

3.2 *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 Recommendations X.25 and X.75.

3.2.1 **reset stimulus probability** *definition*

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[4]. 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 expected number of reset stimuli generated within that section and transferred across the boundary per virtual connection second.

3.2.2 **reset probability** *definition*

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 Recommendation X.134 packet layer reference events listed in Table 6/X.136.

TABLE 6/X.136

Packet layer reference events (PEs) used in measuring reset probability

Boundaries of section	Pair of PEs
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)]

a) Pairs of PEs resulting from reset events

Boundaries of section	Pair of PEs
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)]

b) Additional PE pairs resulting from reset events on PVCs

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[4] 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.

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 substracting 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 X.134 PEs used in measuring reset probability at each section boundary are identified in Table 6/X.136.

3.2.3 *Values*

The contribution from each network portion to overall reset stimulus probability and reset probability under the conditions described in this Recommendation shall not exceed the values specified in Table 7/X.136.

TABLE 7/X.136

Worst-case reset stimulus probability and reset probability values for virtual connection portions

Statistic	Virtual connection portion type			
	National		International	
	A	B	A	B
Reset stimulus probability (reset stimuli per VC second)	10^{-6}	10^{-6}	[a]	10^{-6}
Reset probability (resets per VC second)	10^{-5}	2×10^{-5}	N/A	2×10^{-5}

[a] The Type A international virtual connection portion consists only of a physical circuit. Its contribution to reset stimulus probability is expected to be negligible.

Note – All specified values are provisional.

The reset stimulus and reset probabilities for a set of concatenated virtual connection portions may be estimated from the individual portion probabilities as follows. Assume between boundaries (B_i, B_j) the reset probability is R_1 and the reset stimulus probabilities are S_{1i}, S_{1j}. Assume between boundaries (B_j, B_k) the reset probability is R_2 and the reset stimulus probabilities are S_{2j}, S_{2k}. Then on a VC passing through B_j the reset probability between B_i and B_k is approximately ($R_1 + R_2 + S_{1j} + S_{2j}$). See Figure 4/X.136. The reset stimulus probability at B_i is S_{1i} and the reset stimulus probability at B_k is S_{2k}.

3.3 *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 Recommendations X.25 and X.75.

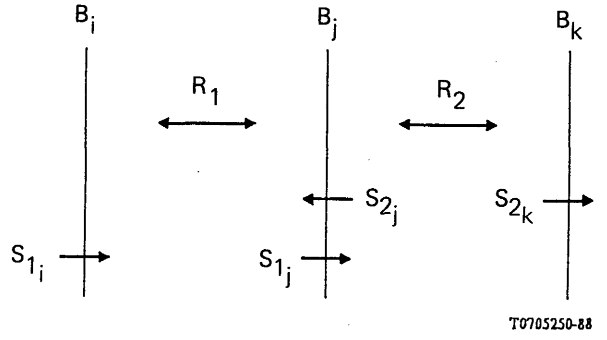


FIGURE 4/X.136

Reset stimulus and reset probabilities for concatenated virtual connection portions

3.3.1 **premature disconnect stimulus probability** *definition*

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[5]. An example of of a premature disconnect stimulus is the transmission of an incorrect packet type into a virtual connection section. A premature disconnect stimulus applies only to virtual call services.

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.

3.3.2 **premature disconnect probability** *definition*

A premature disconnect event is defined to have been generated within a section when, in the absence of an external premature disconnect stimulus, two packet exit the section – one at each boundary – creating any one of the pairs of packet layer reference events listed in Table 8/X.136. A premature disconnect event applies only to virtual call services.

TABLE 8/X.136

Packet layer reference events (PEs) used in measuring premature disconnect probability

(Pairs of PEs resulting from premature disconnect events)

Boundaries of section	Pair of PEs
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 [3(X.75) 12(X.75)]

---

[5] 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.

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 substracting from the result any premature disconnect stimuli that enter the section during that period.

Note – Premature disconnect events may be associated with a loss of packets.

The specific X.134 PEs used in measuring premature disconnect probability at each section boundary are identified in Table 8/X.136.

3.3.3 *Values*

The contribution from each network portion to overall premature disconnect stimulus probability and premature disconnect probability under the conditions described in this Recommendation shall not exceed the values specified in Table 9/X.136.

TABLE 9/X.136

Worst-case premature disconnect stimulus probability and premature disconnect probability values for virtual connection portions

Statistic	Virtual connection portion type			
	National		International	
	A	B	A	B
Premature disconnect stimulus probability (premature disconnect stimuli per VC second)	10^{-7}	10^{-7}	10^{-7}	10^{-7}
Premature disconnect probability (premature disconnects per VC second)	5×10^{-6}	10^{-5}	N/A	10^{-5}

Note – All specified values are provisional.

The premature disconnect stimulus and premature disconnect probabilities for a set of concatenated virtual connection portions may be estimated from the individual portion probabilities in a manner analogous to that described in § 3.2.3.

4 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.

4.1 **call clear failure probability** *definition*

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 clean 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 10/X.136.

TABLE 10/X.136

Packet layer reference events (PEs) used in measuring call clear failure probability

Circuit section	X.134 Packet layer reference event	
	Starting PE	Ending PE
Clearing DTE access circuit section	6(X.25)	–
Cleared DTE access circuit section	–	5(X.25) (does not occur)
Internetwork circuit section	3(X.75)	3(X.75) (does not occur)

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

4.2 *Values*

The contribution from each virtual connection portion to the overall call clear failure probability under the conditions described in this Recommendation shall not exceed the values specified in Table 11/X.136.

TABLE 11/X.136

Worst-case call clear failure probability values for virtual connection portions

Statistic	Virtual connection portion type			
	National		International	
	A	B	A	B
Probability	10^{-5}	2×10^{-5}	[a]	2×10^{-5}

[a] The Type A international virtual connection portion consists only of a physical circuit. Its contribution to call clear failure probability is expected to be negligible.

Note – All specified values are provisional.

4.3 *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 Recommendation. Recovery mechanisms for such occurrences are defined in both the Recommendations X.25 and X.75 protocols. Clear confirmation at X.25 interfaces is a national matter.

ANNEX A

(to Recommendation X.136)

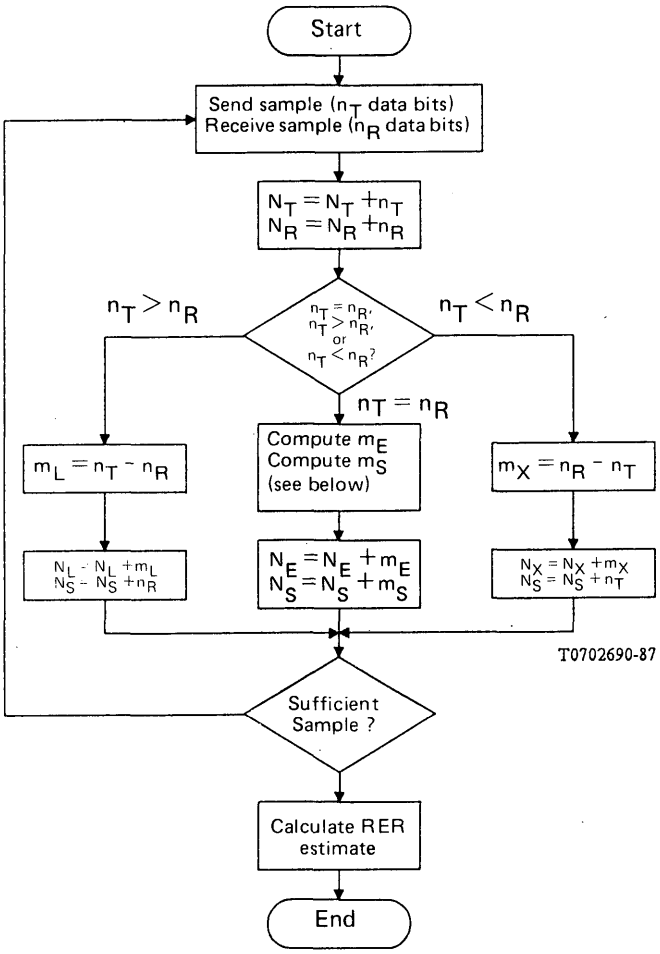
Ancilliary information on accurancy and dependability measurement

The following points should be noted with regard to accuracy and dependability measurement:

- The ratios used to calculate the probabilities are understood to be estimates of the true probabilities.
- The periods of observation for accuracy – and dependability – related probabilities, as well as the concept of busy hour itself, for packet services, are for further study.

Figure A1/X.136 illustrates a simple approximate method of calculating residual error rate. A sample consisting of n_T user data bits is transmitted typically in many successive packets. (A 128-octet packet length is assumed.) A corresponding sample consisting of n_R user data bits is received. If $n_T = n_R$, the transmitted and received user data bits are compared bit for bit, and the number of incorrect data bits in the sample is estimated by m_E, the number of corresponding transmitted and received bits that do not match. If $n_T > n_R$, the number of lost data bits in the sample is estimated by $m_L = (n_T - n_R)$. If $n_T < n_R$, the number of extra data bits in the sample is estimated by $m_x = (n_R - n_T)$. If a reset request or clear request is issued during the transfer of a measurement sample, that sample is excluded in calculating the RER estimate.

The outcome totals in each sample are accumulated over a number of samples sufficient to calculate the residual error rate with the desired precision. Guidelines for relating overall sample size with desired precision are for further study. It should be noted that the approximate method of residual error rate estimation presented here will not produce unbiased estimates if more than one category of bit transfer failure occurs in the same sample. Other, more exact methods of estimating residual error rate may also be employed.



$$m_E = \sum_{i=1}^{n_T} b_i^T \oplus b_i^R \qquad i = \text{bit position}$$

$$m_S = n_T - m_E$$

FIGURE A-1/X.136

An approximate method of calculating residual error rate

ANNEX B

(to Recommendation X.136)

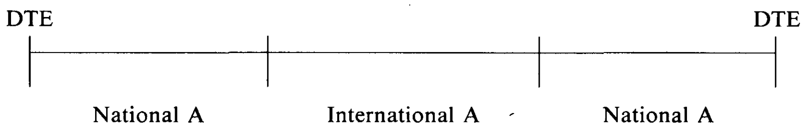
Representative end-to-end accuracy and dependability performance

This annex provides two examples to illustrate how end-to-end (DTE to DTE) accuracy and dependability performance can be estimated from the individual virtual connection portion performance values specified in Recommendation X.136. Two example concatenations of Type A and Type B virtual connection portions are defined. The end-to-end call set-up error probability, call set-up failure probability, residual error rate, reset stimulus probability, reset probability, premature disconnect stimulus probability, premature disconnect probability, and call clear failure probability are calculated for each example. 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.

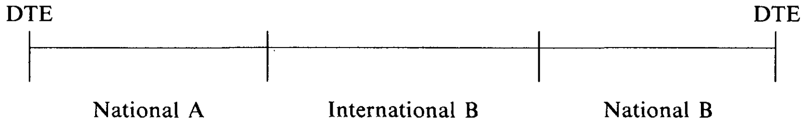
B.1 *Definition of the example end-to-end connections*

For ease of reference, the two example end-to-end (i.e., DTE to DTE) connections presented in this annex will be referred to as “Type 1” and “Type 2” configurations. These hypothetical, but representative, configurations use the portion boundaries and packet layer reference events described in Recommendation X.134. Table 1/X.136 defines the virtual connection portion Types.

The Type 1 configuration is defined to be:



The Type 2 configuration is defined to be:



B.2 *End-to-end accuracy and dependability performance for the Type 1 and Type 2 configuration examples*

End-to-end accuracy and dependability performance values have been calculated for the example Type 1 and Type 2 connection configurations and are reported below in Tables B-1/X.136 and B-2/X.136. These calculations have been made by applying the methods described below to the individual network portions that, for convenience in defining these examples, are characterized by the worst-case accuracy and dependability performance values specified in Recommendation X.136.

Assuming that the performance associated with the individual network portions are statistically independent, then a very close approximation to the end-to-end performance can be obtained for the call set-up error probability, call set-up failure probability, residual error rate probability, and call clear failure probability by simply summing the respective probabilities for the concatenated individual connection portions. Note that this procedure assumes that the approximation error caused by dropping the higher order terms in combining these individual portion probabilities is usually not significant and therefore can be ignored for most cases of practical interest.

Example: To compute the end-to-end probability of call set-up error for the Type 1 configuration, refer to Table 2/X.136 to obtain the individual portion probabilities (National A: probability = 10^{-5}; International A: probability = 0). The end-to-end probability of call set-up error is then $10^{-5} + 0 + 10^{-5} = 2 * 10^{-5}$.

The approximate end-to-end performance at each boundary for the reset probability, reset stimulus probability, premature disconnect stimulus probability, and premature disconnect probability can be calculated using methods in §§ 3.2.3 and 3.3.3 of Recommendation X.136.

Example: To compute the end-to-end performance for the reset probability for the Type 2 configuration, refer to Table 7/X.136 to obtain the individual portion probabilities. The end-to-end probability of reset at the boundaries can be calculated as $10^{-5} + 0 + 10^{-5} + 0 + 10^{-6} + 0 + 10^{-6} = 2.2 * 10^{-5}$.

Example: To compute the end-to-end performance for the reset stimulus probability for the Type 1 configuration, refer to Table 7/X.136 to obtain the individual portion probabilities. The end-to-end probability of reset stimulus at the boundaries can be determined by inspection as 10^{-6}.

TABLE B-1/X.136

End-to-end accuracy and dependability performance for the Type 1 configuration example

Type 1 configuration	
Statistic	End-to-end value
Call set-up error probability	$2 * 10^{-5}$
Call set-up failure probability	$1 * 10^{-2}$
Residual error rate	$2 * 10^{-10}$
Reset stimulus probability	$1 * 10^{-6}$
Reset probability	$2.2 * 10^{-5}$
Premature disconnect stimulus probability	$1 * 10^{-7}$
Premature disconnect probability	$1.04 * 10^{-5}$
Call clear failure probability	$2 * 10^{-5}$

TABLE B-2/X.136

End-to-end accuracy and dependability performance for the Type 2 configuration example

Type 2 configuration	
Statistic	End-to-end value
Call set-up error probability	$5 * 10^{-5}$
Call set-up failure probability	$2.5 * 10^{-2}$
Residual error rate	$5 * 10^{-10}$
Reset stimulus probability	$1 * 10^{-6}$
Reset probability	$5.4 * 10^{-5}$
Premature disconnect stimulus probability	$1 * 10^{-7}$
Premature disconnect probability	$2.54 * 10^{-5}$
Call clear failure probability	$5 * 10^{-5}$

B.3 *Notes on key assumptions, results and implications*

For further study.

Recommandation X.137

AVAILABILITY PERFORMANCE VALUES FOR PUBLIC DATA NETWORKS WHEN PROVIDING INTERNATIONAL PACKET-SWITCHED SERVICES

(Melbourne, 1988)

The CCITT,

considering

(a) that Recommendation X.1 specifies the international user classes of service in public data networks;

(b) that Recommendation X.2 specifies the international data transmission services and optional user facilities in public data networks;

(c) that Recommendation X.25 specifies the DTE/DCE interface for packet mode terminals connected to public data networks by dedicated circuit;

(d) that Recommendation X.75 specifies the packet-switched signalling system between public data networks providing data transmission services;

(e) that Recommendation X.323 specifies general arrangements for interworking between packet-switched public data networks;

(f) that Recommendation X.96 specifies the international routing principles and routing plan for public data networks;

(g) that Recommendation X.110 specifies the international routing principles and routing plan for public data networks;

(h) that Recommendation X.213 defines the OSI Network Layer service;

(i) that Recommendation X.140 defines general quality of service parameters for communication via public data networks;

(j) that Recommendation X.134 specifies portion boundaries and packet layer reference events for defining packet-switched service performance parameters;

(k) that Recommendation X.135 specifies speed of service performance values for public data networks when providing international packet-switched service;

(l) that Recommendation X.136 specifies accuracy and dependability (including blocking) performance values for public data networks when providing international packet-switched service.

unanimously declares

(1) that the availability parameters defined in this Recommendation shall be used in the planning and operation of international packet-switched data communication service provided in accordance with Recommendations X.25 and X.75,

(2) that in such service, the performance values specified in this Recommendation shall be taken as worst-case limits under the conditions specified herein.

1 Introduction

1.1 This Recommendation is the fourth in a series of four CCITT Recommendations (X.134-X.137) that define performance parameters and values for international packet-switched data communication services. Figure 1/X.137 illustrates the scope of these four Recommendations and the relationships among them.

1.2 Recommendation X.134 devides a virtual connection into basic sections whose boundaries are associated with X.25 and X.75 interfaces; defines particular collections of basic sections, called virtual connection portions, for which performance values will be specified; and defines a set of packet layer reference events (PEs) which provide a basis for performance parameter definition. The basic sections consist of network sections and circuit sections. They are delimited, in each case, by physical data terminal equipment (DTE) or data switching equipment (DSE) interfaces. Two types of virtual connection portions are identified: national portions and international portions. Each PE is defined to occur when a packet crossing a section boundary changes the state of the packet layer interface.

1.3 For comparability and completeness, packet-switched network performance is considered in the context of the 3 × 3 performance matrix defined in Recommendation X.140. Three protocol-independent data communication functions are identified in the matrix: access, user information transfer, and disengagement. These general functions correspond to call set-up, data (and interrupt) transfer, and call clearing in packet-switched virtual call services conforming to the X.25 and X.75 Recommendations. Each function is considered with respect to three general performance concerns (or “performance criteria”): speed, accuracy, and dependability. These express, respectively, the delay or rate, degree or correctness, and degree of certainty with which the function is performed.

1.4 Recommendation X.135 defines protocol-specific speed of service parameters and values associated with each of the three data communication functions. Recommendation X.136 defines protocol-specific accuracy and dependability parameters and values associated with each function. The Recommendation X.135 and X.136 parameters are called “primary parameters” to emphasize their direct derivation from packet layer reference events.

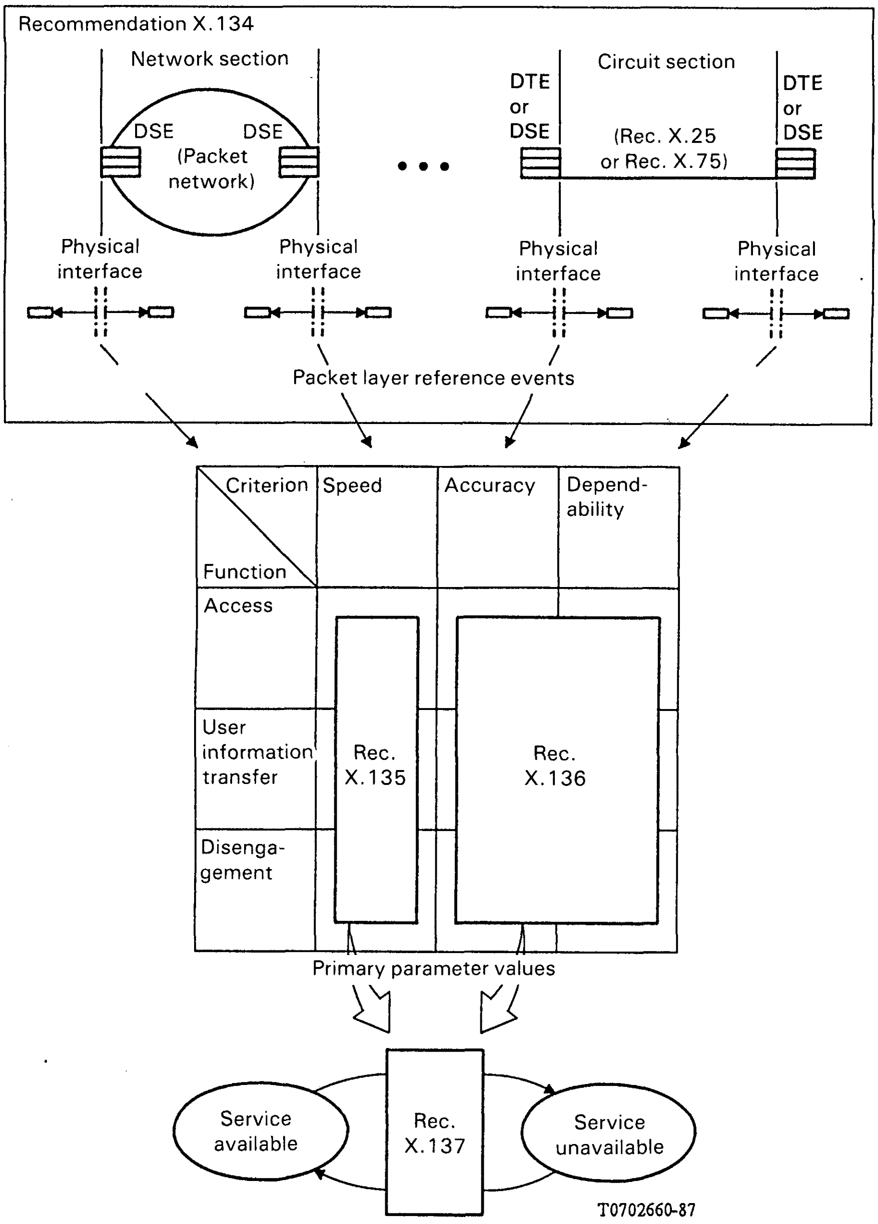


FIGURE 1/X.137

Packet-switched service performance description framework

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

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

1.7 This Recommendation specifies availability values for national and international portions of two types (Table 1/X.137). Performance values for data terminal equipment are not specified, but the parameters defined in this Recommendation may be employed in such specification to assist users in establishing quantitative relationships between network performance and quality of service (see Recommendation X.140).

TABLE 1/X.137

Virtual connection portion types for which performance values are specified [a]

Portion type	Typical characteristics
National A	Terrestrial connection via an access network section
National B	Connection via an access network section with one satellite circuit; or via an access network section and one or more transit network sections
International A	Connection via a direct terrestrial internetwork section
International B	Connection via two satellite circuits and one transit network section; or via one satellite circuit and two or more transit network sections

[a] The values specified for type B portions also apply to virtual connection portions not explicitly identified as type A or type B.

1.8 Worst-case values for each of the two availability parameters are specified below for each virtual connection portion type identified in Table 1/X.137. The term "worst case" means that these values should be met during the normal busy hour in the worst-performing virtual connection portion used in providing international packet-switched service. The performance of a virtual connection portion may be better than the worst-case values specified in this Recommendation. Design objectives that take into account more demanding user applications and network performance and connectivity enhancements are for further study.

Numerical methods for combining individual portion performance values to estimate end-to-end performance are also provided in this Recommendation. DTE to DTE values for two particular hypothetical reference connections are derived using these methods in Annex B.

2 Availability function

Eight performance parameters from Recommendations X.135 and X.136 are used in computing the availability of a virtual connection: throughput capacity (X.135), call set-up failure probability (X.136), call set-up error probability (X.136), residual error rate (X.136), reset probability (X.136), reset stimulus probability (X.136), premature disconnect probability (X.136), and premature disconnect stimulus probability (X.136). 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 2/X.137.

TABLE 2/X.137

Outage criteria for the availability decision parameters

Availability decision parameters	Outage criteria
Call set-up failure probability (cfp) Call set-up error probability (cep)	(cfp + cep) > 0.9
Throughput capacity (tc)	tc < 80 bit/s
Residual error rate (rer)	rer > 10^{-3}
Reset probability (rp) Reset stimulus probability (rsp_1, rsp_2)	(rsp_1 + rp + rsp_2) > 0.015
Premature disconnect probability (pdp) Premature disconnect stimulus probability ($pdsp_1$, $pdsp_2$)	($pdsp_1$ + pdp + $pdsp_2$) > 0.01

Note – These outage criteria are provisional.

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; and
2) all continuous periods when the link is not in the information transfer phase are less than 1 second in length; and
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 the following reasons:

1) a nonfunctional physical circuit; or
2) a data link layer controller either unable or unwilling to establish the information transfer phase; or
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; and
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, i.e. failures of data link controllers or physical circuits outside the section in question.)

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 2/X.137.

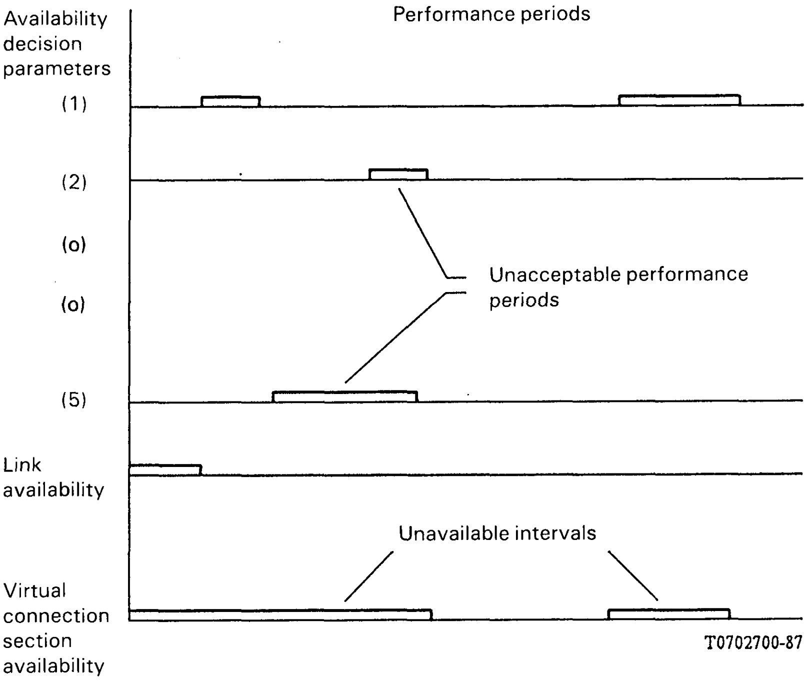


FIGURE 2/X.137

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. A minimal availability test meeting these restrictions is defined in Annex B.

3 Availability parameters

This section specifies worst-case values for two availability parameters: service availability and mean time between service outages.

3.1 **Service availability** *definition*

Service availability applies to both virtual call and permanent virtual circuit services. The service availability for a virtual connection portion 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 would be 24 hours per day, 7 days per week.[1] A procedure for estimating the availability of a section is described in Annex A.

3.2 *Definition of* **mean time between service outages**

Mean time between service outages (MTBSO) 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 schedules service time are concatenated. Annex A describes a procedure for estimating the mean time between the service outages of a section.

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[1] Other scheduled service times may be specified in some networks.

Mean time between service outages as defined in this Recommendation is closely related to mean time between failures.

3.3 *Values*

The conribution from each network portion to overall service availability and mean time between service outages under the conditions described in this Recommendation shall not be worse than the values specified in Table 3/X.137. The mean time between service outage values for national portions exclude up to 5% of virtual connections to account for geographical and climatic extremes.

TABLE 3/X.137

Worst-case service availability and mean time between service outage values for virtual connection portions

Statistic	Virtual connection portion type			
	National		International	
	A	B	A	B
Service availability (percent)	99.5	99	99.5	99
Mean time between service outages (hours)	1200	800	1600	800
Mean time to service restoral MTTSR [a] (hours)	(6)	(8)	(8)	(8)

[a] The parenthetical values given in Table 3/X.137 represent the mean time to service restoral that would result if the service availability and the mean time between service outage values are achieved as stated in the table. Any improvements in MTBSO should be used to improve service availability and should not be used to degrade MTTSR.

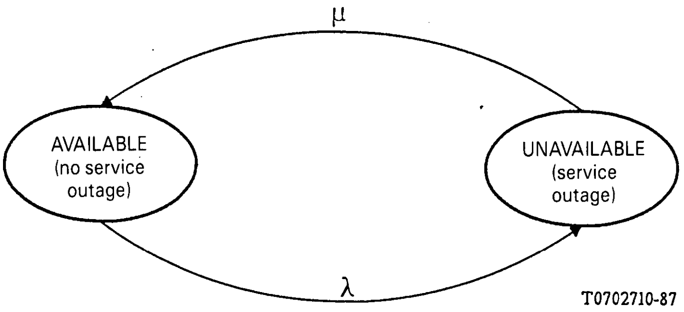
Note – All specified values are provisional.

3.4 *Related parameters*

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

- **mean time to service restoral (MTTSR)** is the average duration of unavailable service time intervals.
- **failure rate** (λ) is the average number of transitions from the available state to the unavailable state per unit available time.
- **restoral rate** (μ) is the average number of transitions from the unavailable state to the available state per unit unavailable time.
- **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 3/X.137.



a) *State diagram*

$$MTBSO = \frac{1}{\lambda}$$

$$MTTSR = \frac{1}{\mu}$$

$$A = 100\left(\frac{MTBSO}{MTBSO + MTTSR}\right) = 100\left(\frac{\mu}{\lambda + \mu}\right)$$

$$U = 100 - A = 100\left(\frac{MTTSR}{MTBSO + MTTSR}\right) = 100\left(\frac{\lambda}{\lambda + \mu}\right)$$

b) *Parameter relationships*

FIGURE 3/X.137

Basic availability model and parameters

ANNEXE A

(to Recommendation X.137)

Sampling estimation of availability parameters

A.1 *A minimal test for 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 unavailable. This test and its decision criteria are defined to be the minimum criteria necessary to sample the availability 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.

Phase I: Attempt N[2] 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).
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 rate 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 of 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.

A.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 durations 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.

A.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".[3]

---

[2] The number N of consecutive call set-ups depends on the values of the outage decision parameter, to provisionally 4, and so for further study.

[3] Each counter is initially set to zero.

For each pre-defined interval:

- 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.
- 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.
- 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 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.[4] 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 is missed, and the mean time between service outages is 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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[4] 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.

ANNEX B

(to Recommendation X.137)

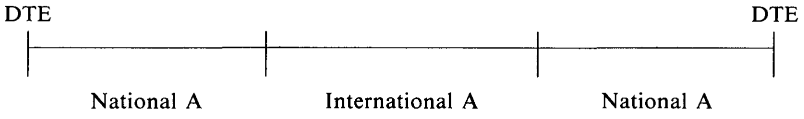
Representative end-to-end availability performance

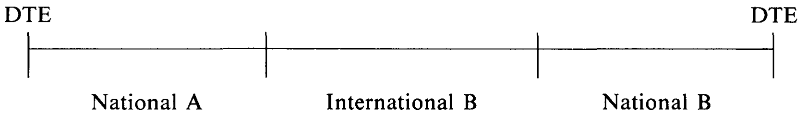
This annex provides two examples to illustrate how end-to-end (DTE to DTE) availability performance can be estimated from the individual virtual connection portion performance values specified in Recommendation X.137. Two example concatenations of Type A and Type B virtual connection portions are defined. The end-to-end service availability and mean time between service outages are calculated for each example. Although alternative network models and statistical assumptions are possible, the methods presented in this annex provide one practical way of estimating the end-to-end performance from the performance of the individual network portions.

B.1 *Definition of the example end-to-end connections*

For ease of reference the two example end-to-end (i.e. DTE to DTE) connections presented in this annex will be referred to as "Type 1" and "Type 2" configurations. These hypothetical, but representative, configurations use the portion boundaries and packet layer reference events described in Recommendation X.134. Table 1/X.137 defines the virtual connectin portion types.

The Type 1 configuration is defined to be:





B.2 *End-to-end availability performance for the Type 1 and Type 2 configuration examples*

End-to-end availability performance values have been calculated for the example Type 1 and Type 2 connection configurations and are reported below in Tables B-1/X.137 and B-2/X.137. These calculations have been made by applying the methods described below to the individual network portions that, for convenience in defining these examples, are characterized by the worst-case accuracy and dependability performance values specified in Recommendation X.137.

Assuming that the service availability performance values associated with the individual network portions are statistically independent, then the end-to-end performance values can be calculated by multiplying the percent of time each of the network portions is available.

Example: To compute the end-to-end service availability for the Type 1 configuration, refer to Table 3/X.137 to obtain the individual portion availabilities (National A: percent = 99.5; International A: percent = 99.5). The end-to-end availability in percent is then:

$$99.5 \cdot 99.5 \cdot 99.5 = 98.5$$

The end-to-end performance for the mean time between service outages can be estimated by assuming that the times between service outages in each individual network portion are independent and exponentially distributed. It follows from these assumptions that the end-to-end mean time between service outages performance 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 mean time between service outages for each of the N network portions, T_i ($i = 1, 2, \dots, N$), is expressed in hours.

Example: For the Type 1 configuration the National A portion mean time between service outages is 1200 hours and the International A portion is 1600 hours (refer to Table 3/X.137). The end-to-end performance objective is then:

$$[1200^{-1} + 1600^{-1} + 1200^{-1}]^{-1} = 436 \text{ heures}$$

TABLE B-1/X.137

End-to-end availability and mean time between service outage performance for the type 1 configuration example

Type 1 configuration	
Statistic	End-to-end value
Service availability (percent)	98.5
Mean time between service outages (hours)	436

TABLE B-2/X.137

End-to-end availability and mean time between service outage performance for type 2 configuration example

Type 2 configuration	
Statistic	End-to-end value
Service availability (percent)	97.5
Mean time between service outages (hours)	300

GENERAL QUALITY OF SERVICE PARAMETERS FOR COMMUNICATION VIA PUBLIC DATA NETWORKS

(Malaga-Torremolinos, 1984; amended at Melbourne, 1988)

The CCITT,

considering

(a) that users of data transmission services need general parameters which express their quality of service requirements without reference to any particular service or the means of its provision;

(b) that providers of data transmission services need similar general parameters for representing offered services, and for relating user quality of service requirements to network performance capabilities;

(c) that Recommendations X.130 and X.131 define protocol-specific performance parameters and objectives for circuit-switched public data networks;

(d) that Recommendation X.134 specifies portion boundaries and packet-layer reference events for defining packet-switched performance parameters;

(e) that Recommendations X.135, X.136 and X.137 define protocol-specific performance parameters and values for packet-switched public data networks;

(f) that Recommendation X.200 defines the Reference Model of Open Systems Interconnection (OSI) for CCITT applications;

(g) that Recommendation X.213 defines the OSI Network Service;

(h) that Recommendation X.300 defines general principles and arrangements for interworking among public data networks, and between public data networks and other networks,

unanimously recommends

that the general parameters defined below be used in specifying the end-to-end quality of public data network services as seen from the user's point of view.

1 Scope and application

1.1 This Recommendation defines a set of general quality of service (QOS) parameters for public data networks (PDNs). The parameters have two essential characteristics:

1) they focus on performance *effects* which are observable at the network interfaces, rather than their causes within the network; and
2) their definitions are based on protocol-independent events (e.g. access request) rather than protocol-specific interface events (e.g. issuance of an X.21 *call request* signal).

These characteristics make the parameters independent of application, network, and service. With proper specialization, they may be used to specify or measure the quality of any data communication service, irrespective of network internal design or network access protocol. Examples of data communication services to which the parameters may be applied are circuit-switched services, packet-switched services, and leased circuit services. The parameters are applicable to both connection-oriented and connectionless services.

1.2 The general quality of service parameters defined in this Recommendation are specifically designed to be used in relating the performance capabilities of particular network services with user requirements (see Figure 1/X.140). The network specific performance parameters defined in other X-series Recommendations are focused on specific service interface protocols (e.g. X.21, X.25) and specific network configurations (e.g. X.92, X.110). They are essential for network design and operation and component performance specification, but are not necessarily understandable or relevant to users. Similarly, performance requirements of users are often focused on particular applications (e.g. electronic funds transfer, text editing) and may not be directly useful to network providers. An example is the data processing parameter "response time". The general parameters provide a "common language" for relating the two. They enable users to specify communication requirements without presupposing any particular service, network, or protocol, and enable providers to describe service performance in terms that are relevant to users, but not specialized to any particular application.

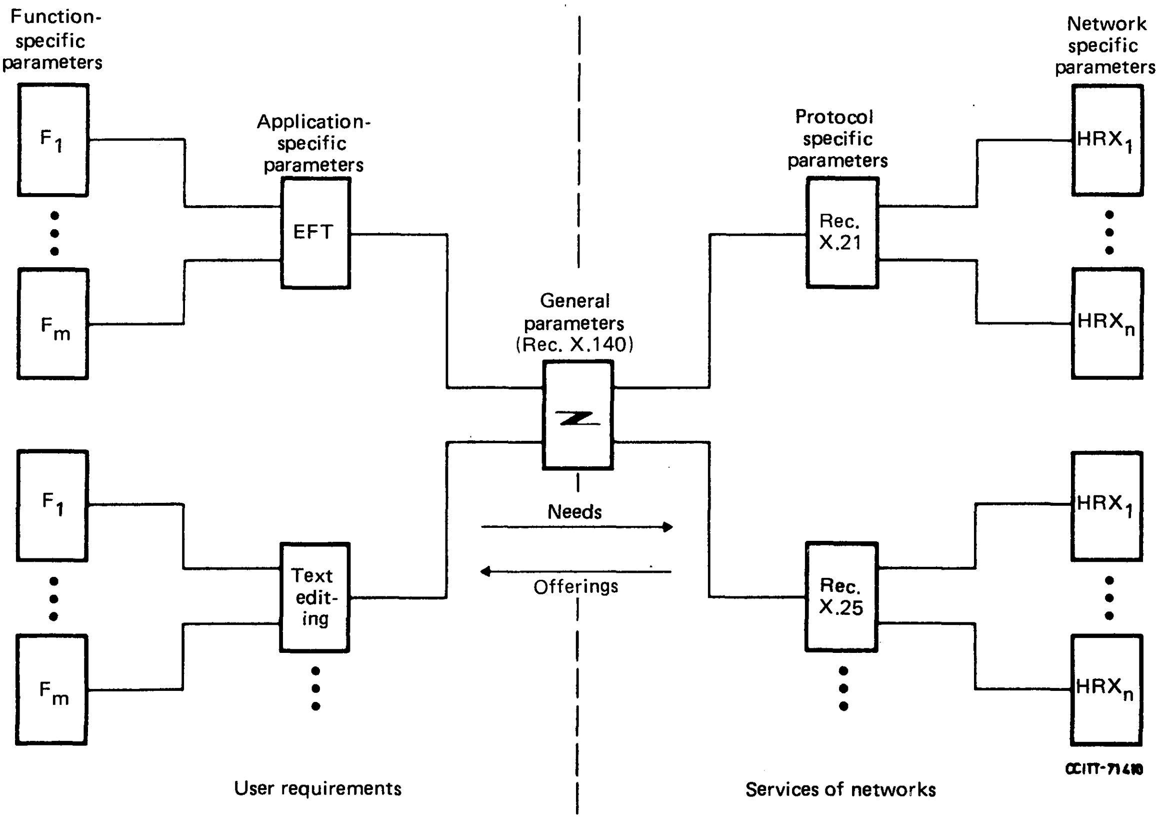
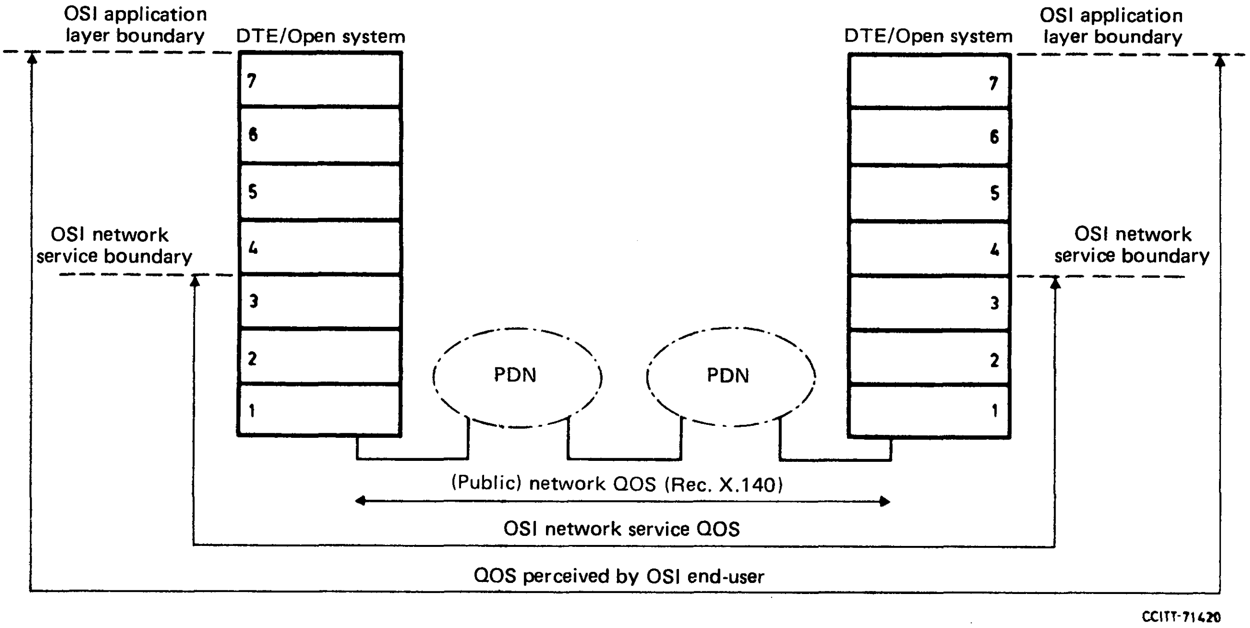


FIGURE 1/X.140

"Common language" function of the general QOS parameters

1.3 The general parameters are principally intended to describe communication performance at interfaces between public data networks and customer DTEs. The detailed characteristics of such network-user interfaces depend on the type of network service and the user application. The quality of X.21-based circuit-switching networks is described in terms of signals occurring at the physical DTE/DCE interfaces (e.g. *call request*, *incoming call*). The quality of X.25-based packet-switching networks is described in terms of corresponding events (or state transitions) occurring at the packet layer of X.25. Specific relationships between the X.140 parameters and the circuit-switching and packet-switching network performance parameters defined in the X.130-series Recommendations are described in Annexes A and B, respectively.

1.4 Many applications of public data networks will conform to the Reference Model of open systems interconnection for CCITT applications (CCITT Recommendation X.200). In that model, QOS parameters are defined as abstract boundaries between layers. Public data networks provide support for the OSI network service (Recommendation X.213). The QOS parameters defined in the OSI network service reflect those aspects of public data network quality of service that are observable and significant to OSI network service users. The general relationship between PDN quality of service and OSI network service quality is illustrated in Figure 2/X.140. Specific relationships between the X.140 QOS parameters and the network layer QOS parameters are defined in Annex C. Relationships among the general parameters, the X.130-series parameters, and the OSI network service performance QOS parameters are summarized in Figure 3/X.140.



Note 1 – QOS parameters and values for specific PDN's (CSPDN, PSPDN) are specified by separate Recommendations.

Note 2 – The signalling of QOS information in various interworking situations is not the subject of this Recommendation.

Note 3 – Interfaces are defined in § 1.3.

Note 4 – Application to private networks may be possible at the discretion of individual Administrations.

FIGURE 2/X.140

QOS relationships in an OSI context

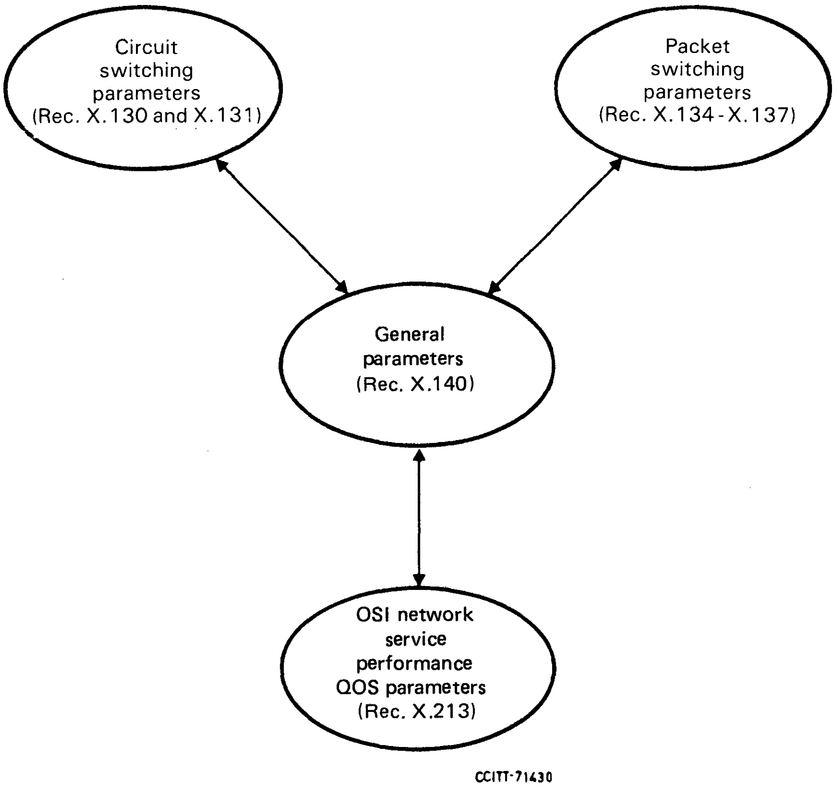


FIGURE 3/X.140

Relationships among the general parameters, the X.130-series parameters, and the OSI network service performance QOS parameters

1.5 Because the X.140 parameters are based on protocol-independent events, they may also be applied at higher layers in the OSI model. Application of the X.140 parameters at the OSI end user interfaces is illustrated in Figure 2/X.140. Details of parameter specialization, relationships to application-specific parameters, and the mapping of end user QOS values into corresponding lower layer values are for further study.

1.6 A requirement also exists to describe QOS at higher layers (above the network layer) in non-OSI applications. An example is the X.28/X.29 PAD facility. Use of the X.140 parameters to express QOS characteristics in such applications, and possible relationships to PDN QOS parameters, are also for further study.

1.7 Some public networks will have the capability to signal QOS requests and conditions, or to permit users to "negotiate" certain QOS characteristics of the network. This Recommendation does not describe any public data network capabilities of this kind, nor does it specify how they might be used. The provision and use of such capabilities will be the subject of other Recommendations (for example, Recommendations describing how public data networks may support the OSI network service). In the case of interworking between networks, such capabilities are described in Recommendation X.300.

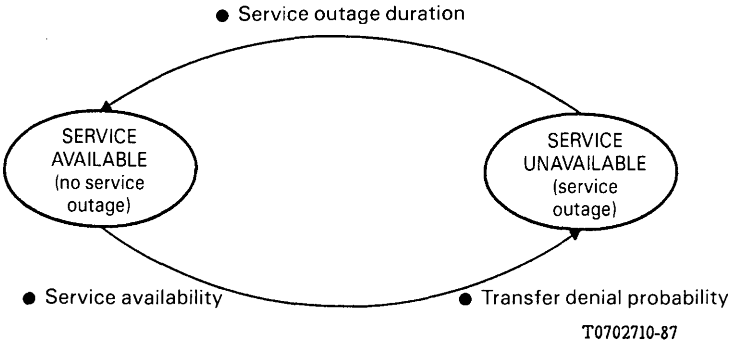
1.8 This Recommendation does not specify values for the general QOS parameters. Values may be specified either by the service user, in characterizing a particular data communication requirement, or by the service provider, in characterizing a particular service offering. Values may be measured by either the users or the provider.

1.9 To ensure comparability, stated values for the general parameters should be accompanied by supplementary information which clearly identifies their scope of application and statistical meaning. User delays may be "factored out" of stated delay and transfer rate values using the method defined below. The same method can be used to factor out provider delays in cases where an assessment of user performance is desired.

1.10 Figure 4/X.140 identifies the general QOS parameters defined in this Recommendation. The parameters are of two types: primary parameters and availability parameters. The primary parameters describe performance during periods of normal service operation, in the absence of service outages. The availability parameters describe the frequency and duration of service outages.

Function Criterion	Speed	Accuracy	Dependability
Access	● Access delay	● Incorrect access probability	● Access denial probability
User information transfer	● User information transfer delay ● User information transfer rate	● User information error probability ● Extra user information delivery probability ● User information misdelivery probability	● User information loss probability
Disengagement	● Disengagement delay	● Disengagement denial probability	

a) *Primary parameters*



b) *Availability parameters*

FIGURE 4/X.140

Summary of user-oriented QOS parameters

1.11 Three protocol-independent data communication functions are considered in defining the primary parameters: access, user information transfer, and disengagement. These general functions correspond to connection set-up, data transfer, and connection clearing in connection-oriented services. They are also applicable to connectionless services. Each function is considered with respect to three general performance concerns (or "performance criteria"): speed, accuracy, and dependability. These express, respectively, the delay or rate, degree of correctness, and degree of certainty with which the function is performed.

1.12 An associated two-state model provides a basis for describing overall service availability. A specified availability function compares the values for a subset of the primary parameters with corresponding outage thresholds to classify the service as "available" (no service outage) or "unavailable" (service outage) during scheduled service time. The availability parameters characterize the resulting binary random process.

1.13 The remainder of this Recommendation is comprised of three sections. Section 2 defines the set of user-oriented QOS parameters. Section 3 describes a method of separating delays into user and network components and determining "responsibility" for timeout performance failures. Section 4 specifies supplementary information which should be provided in conjunction with any statement of parameter values.

2 Parameter definitions

This section provides definitions for the fourteen user-oriented QOS parameters.

2.1 access parameters

Performance of the access function is described by three parameters: access delay, incorrect access probability, and access denial probability.

2.1.1 *Access delay*

Access delay is the value of elapsed time between an access request and successful access.

An access request is any interface signal that notifies the network of a user's desire to initiate a data communication session.

Elapsed time values are calculated only on access attempts that result in successful access. The successful access outcome is indicated in one of two ways:

1) by network issuance of a *ready for data* or equivalent signal to the calling user before access timeout, in networks that provide such a signal; or
2) by the fact that at least one bit of user information is input to the system before access timeout, in networks that do not provide a *ready for data* or equivalent signal. In connection-oriented services, there is the additional requirement that the intended called user must have been contacted and committed to the data communication session during the access attempt. This requirement distinguishes successful access outcomes from incorrect access outcomes, as discussed in § 2.1.2 below.

Access delay is divided into user-dependent and network-dependent components. Values for the network-dependent components are specified in network-specific Recommendations (e.g. Recommendation X.135).

2.1.2 *Incorrect access probability*

Incorrect access probability is the ratio of total access attempts that result in incorrect access to total access attempts in a specified sample.

Incorrect access is essentially the case of a "wrong number". It occurs when the network establishes a physical or virtual circuit connection to a user other than the one intended by the call originator, and then does not correct the error before the start of user information transfer. Incorrect access can only occur in connection-oriented services, since the network does not establish a connection between users in connectionless services. Incorrect access is distinguished from successful access (in connection-oriented services) by the fact that the intended called user is not contacted and committed to the data communication session during the access attempt. Values for network-specific parameters corresponding to incorrect access probability are contained in network-specific Recommendations (e.g. X.136).

2.1.3 *Access denial probability*

Access denial probability is the ratio of total access attempts that result in access denial to total access attempts in a specified sample.[1]

Access denial (also termed network blocking) can occur in two ways:

1) the network issues a blocking signal to the originating user during the access period (preventing the start of user information transfer); or
2) the network delays excessively in responding to user actions during the access period, with the result that user information transfer is not initiated before access timeout. Access denial is distinguished from service outage by the fact that some active response (i.e. interface signal) is issued by the network during the access attempt.

An access attempt can also fail as a result of user blocking. Such failures are excluded from network performance measurement. User blocking is defined as any case where an access attempt fails as a result of incorrect performance or non-performance on the part of a user. Examples of user blocking include the following:

a) either the originating or the called user issues a termination (or blocking) signal to the network during the access period (preventing the start of the user information transfer); or
b) the originating or the called user delays excessively in responding to network actions during the access period, with the result that user information transfer is not initiated before access timeout. An example of the latter is the case where the called user does not answer an incoming call.

Access timeout occurs (i.e. an access attempt is considered to have failed for performance assessment purposes) whenever the duration of an individual access attempt exceeds a specified value. A procedure for distinguishing access denial from user blocking is described in § 3. Values for network-specific parameters corresponding to access denial probability are contained in network-specific Recommendations (e.g. X.136).

Note – Delay to access denial is not included as a parameter because its effect on the users is considered to be insignificant.

2.2 **User information transfer parameters**

Performance of the user information transfer function is described by six parameters: user information transfer delay, user information transfer rate, user information error probability, extra user information delivery probability, user information misdelivery probability, and user information loss probability.

2.2.1 *User information transfer delay*

User information transfer delay is the value of elapsed time between the start of transfer and successful transfer of a specified user information unit (e.g. block).

The start of user information unit transfer occurs, for any given user information unit, when two conditions have been met:

1) all bits in the unit are physically present within the network facility; and
2) the network has been authorized to transmit them. Authorization may either be an explicit user action (e.g. typing carriage return at a buffered CRT terminal) or an implicit part of inputting the user information itself (e.g. typing a single character at an unbuffered asynchronous terminal).

The successful transfer outcome is declared (on end of transfer) when an information unit is transferred from the source user to the intended destination user within the specified transfer timeout period, and the delivered unit has exactly the form and content intended by the source. The form or content of an information unit successfully delivered to a destination user may differ from that input by the source if desired conversions are performed within the network.

---

[1] This ratio and all other probability ratios defined in this Recommendation are actually *estimates* of the true probability values.

The end of user information unit transfer records the output of user information units to the destination user in essentially the same way as the start of transfer records their input at the source. It is defined to occur when:

a) all bits in the unit are physically present within the destination user facility; and

b) the destination user has been notified that the information is available for use. The notification may be explicit or implicit.

The user information unit used in defining user information transfer delay is a contiguous group of user information bits delimited at the source user-network interface for transfer to a destination user as a unit. The specific number of bits in such a unit may be defined by the provider in specifying an offered service, or by the user in specifying a service requirement. User information transfer delay is divided into user-dependent and network-dependent components. Values for the network-dependent components are specified in network-specific Recommendations (e.g. X.135).

2.2.2 *User information transfer rate*

User information transfer rate is the total number of successfully-transferred user information units in an individual transfer sample divided by the input/output time for that sample.

The input/output time for a transfer sample is the larger of the input time or the output time for that sample (Figure 5/X.140). The sample input time begins when the transfer sample (defined above) begins, and ends when either:

1) all digits in the sample have been input to the network, and the network has been authorized to transmit them; or

2) sample input/output timeout occurs.

The sample output time begins when the first user information digit in the sample is delivered by the network to the destination user. It ends when either:

- the last digit of user information in the sample is delivered to the destination user; or
- sample input/output timeout occurs.

As noted earlier, either the input or the output of a transfer sample may be delayed excessively by a user (rejected sample). Such failures are excluded from network performance measurement. As in the case of user information transfer denial probability, rejected samples are distinguished from valid transfer samples using the procedure described in § 3.

Note – A "maximum user information transfer rate" which excludes the effect of user input/output delays can be calculated using the procedure described in § 3. Values for network-specific parameters corresponding to user information transfer rate are contained in network-specific Recommendations (e.g., X.135).

2.2.3 *User information error probability*

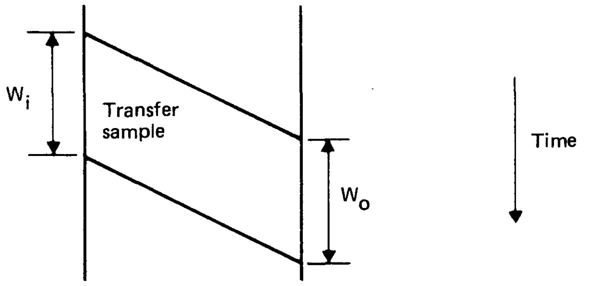
User information error probability is the ratio of total incorrect user information units to total successfully transferred user information units *plus* incorrect user information units in a specified sample.

A transferred user information unit is defined to be an incorrect user information unit when the value of one or more digits in the unit is in error, or when some, but not all, digits in the unit are lost digits or extra digits (i.e. digits that were not present in the original signal).

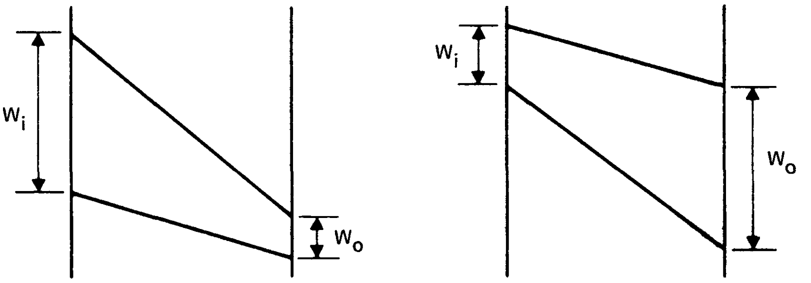
Bit error ratio is a limiting case of user information error probability in which the user information unit length, on which the error performance is based, is a single binary digit.

The proportion of errored seconds is a particular case of user information error probability in which the user information unit length is defined as one second. The number of digits contained in each user information unit in this case is numerically equal to the digit rate per second. This parameter is usually expressed in the form of the percentage of its complement, i.e. as a percentage of error-free seconds (% EFS). A similar parameter, the percentage of error-free deciseconds (% EFdS), can be defined based on a user information unit length of 100 ms.

Values for network-specific parameters corresponding to user information error probability are contained in network-specific Recommendations (e.g. X.136).



Case 1 – No rate conversion: $W_i = W_o$



Case 2 – Rate increase: $W_i > W_o$ *Case 3 – Rate reduction:* $W_i < W_o$

$$\text{User information transfer rate} = \frac{B1_s}{\text{Max } (W_i \text{ or } W_o)}$$

$B1_s$ = Total successful transfer outcomes in the transfer sample

FIGURE 5/X.140

User information transfer rate

2.2.4 *Extra user information delivery probability*

Extra user information delivery probability is the ratio of total (unrequested) extra information units to total information units received by a destination user in a specified sample.

An information unit received by a particular destination user is declared to be an extra information unit when none of the bits in the unit were input to the system by the source user for delivery to that destination. Unless misdelivered user information units are explicitly identified in a measurement process, they will be counted as extra information units. Values for network-specific parameters corresponding to extra user information delivery probability are contained in network-specific Recommendations (e.g. X.136).

2.2.5 *User information misdelivery probability*

User information misdelivery probability is the ratio of total misdelivered user information units to total user information units transferred between a specified source and destination user in a specified sample.

A misdelivered user information unit is a user information unit transferred from a source user to a particular destination user that was actually intended for delivery to a different destination user. It is considered inconsequential whether the information is correct or incorrect in content. Values for network-specific parameters corresponding to user information misdelivery probability may be contained in network-specific Recommendations (under study).

2.2.6 *User information loss probability*

User information loss probability is the ratio of total lost user information units to total transmitted user information units in a specified sample.

A transmitted user information unit is declared to be a lost user information unit when none of the bits in the unit are delivered to the intended destination user within the specified timeout period, and the network is responsible.

User information may also remain undelivered as a result of user information refusal – i.e. non-delivery attributable to excessive delay on the part of a user. An example is a destination user's exercise of flow control. Such outcomes are excluded from network performance measurement.

Transfer timeout occurs (i.e., a transfer attempt is considered to have failed for performance assessment purposes) whenever the duration of an individual transfer period exceeds a specified value. A procedure for distinguishing user information loss from user information refusal is described in § 3. Values for network-specific parameters corresponding to user information loss probability are contained in network-specific Recommendations (e.g. X.136).

2.3 **disengagement parameters**

Performance of the disengagement function is described by two parameters: disengagement delay and disengagement denial probability.

2.3.1 *Disengagement delay*

Disengagement delay is the value of elapsed time between the start of a disengagement attempt for a particular user and successful disengagement of that user.

The disengagement request notifies the system of a user's desire to terminate an established data communications session. It is complementary to the access request in most networks.

Elapsed time values are calculated only on disengagement attempts that result in successful disengagement. The successful disengagement outcome is indicated in one of two ways:

1) by network issuance of a *clear confirmation* or equivalent signal to the requesting user before disengagement timeout, in networks that provide such a signal; or
2) by the fact that the user is able to initiate a new access before disengagement timeout, in networks that do not provide a *clear confirmation* or equivalent signal.

Disengagement delays may be defined independently for each participating user when significantly different values are expected. Disengagement delays are divided into user-dependent and network-dependent components. Values for the network-dependent components are specified in network-specific Recommendations (e.g. X.135).

2.3.2 *Disengagement denial probability*

Disengagement denial probability is the ratio of total disengagement attempts that result in disengagement denial to total disengagement attempts in a specified sample.

The disengagement denial outcome is indicated in one of two ways:

1) by the absence of a *clear confirmation* or equivalent signal within the disengagement timeout period (in networks that provide such a signal); or
2) by the inability of the user to initiate a new access within the specified disengagement timeout period (in networks that do not provide a *clear confirmation* or equivalent signal).

In some networks, a disengagement attempt can also fail as a result of user disengagement blocking. User disengagement blocking is defined as any case where a disengagement attempt fails as a result of incorrect performance or non performance on the part of a user. Examples of user disengagement blocking include the following:

1) a user issues a disengagement blocking signal to the network during the disengagement period (preventing the termination of a connection-oriented data communication session); or
2) a user delays excessively in responding to network actions during the disengagement period, with the result that disengagement is not completed before disengagement timeout. Such failures are excluded from network performance measurement.

Disengagement timeout occurs (i.e. a disengagement attempt is considered to have failed for performance assessment purposes) whenever the duration of an individual disengagement attempt exceeds a specified value. A procedure for distinguishing disengagement denial from user disengagement blocking is described in § 3. Values for network-specific parameters corresponding to disengagement denial probability are contained in network-specific Recommendations (e.g. X.136).

2.4 **Availability parameters**

Three parameters are defined to describe overall service availability: service availability, user information transfer denial probability, and service outage duration.

2.4.1 *Service availability*

Service availability is the ratio of the aggregate time during which satisfactory or tolerable service is, or could be, provided to the total observation period.

In practice, the observation period may consist of several non-contiguous smaller time intervals. The time during which satisfactory or tolerable service is available includes all time that is not within the service outage duration as defined above. The criteria by which the service is judged to be unacceptable are for further study. Such study will embrace consideration of the parameters, including call-related events, that are relevant, and consideration of the observation period(s) and performance thresholds for unacceptability. Values for the network-specific parameter(s) corresponding to service availability are contained in network-specific Recommendations (e.g. X.137).

2.4.2 *User information transfer denial probability*

User information transfer denial probability is the ratio of total transfer denials to total transfer samples during a specified observation period.

A transfer sample is a discrete observation of network performance in transferring user information between a specified source and destination user. A transfer sample begins on input of a selected user information digit at the source user interface, and continues until the outcomes of a given number of transfer attempts have been determined.

A transfer denial is a transfer sample in which the observed performance is worse than a specified minimum acceptable level. Transfer denials are identified by comparing the measured values for four supported quality of service parameters with specified transfer denial thresholds. The four supported parameters are user information error probability, user information loss probability, extra user information delivery probability, and user information transfer rate. Transfer denial includes cases where the network unilaterally terminates user data transmission (e.g. reset or clearing due to network congestion).

A transfer sample may also indicate performance worse than the minimum acceptable level if:

1) the source or destination user intentionally disengages during the sample transfer period; or
2) a user delays excessively in inputting or accepting the sample data (e.g., through exercise of flow control). Such failures (called rejected samples) are excluded from network performance measurement.

A transfer sample input/output timeout occurs (i.e., a transfer sample is considered to have failed for performance assessment purposes) whenever the duration of an individual sample input or output period exceeds a specified value. A procedure for distinguishing a transfer denial from a rejected sample is described in § 3. Values for network-specific parameters corresponding to user information transfer denial probability are contained in network-specific Recommendations (e.g. X.137).

2.4.3 *Service outage duration*

Service outage duration is the duration of any continuous period of time for which satisfactory or tolerable service is not available. It is recognized that the determination of an outage condition requires a finite observation period.

A service outage includes any period during which the user is unable or would be unable to elicit any response from the network; i.e. the network is "dead". It also includes any period during which the service provided by the network is unacceptable because of, for example, poor error performance or throughput. The criteria by which the service is judged to be unacceptable are for further study. Such study will embrace consideration of the parameters, including call-related events, that are relevant, and consideration of the observation period(s) and performance thresholds for unacceptability. Values for the network-specific parameter(s) related to service outage duration are contained in network-specific Recommendations (e.g. X.137).

3 Distinguishing network and user components of performance

This paragraph describes a method of separating delays into network and user components and determining "responsibility" for timeout performance failures. This is accomplished by dividing selected performance periods into alternating "responsibility intervals" of two types:

1) intervals in which the network is responsible for creating the next event in a sequence of interface events leading to the accomplishment of a specified data communication function (e.g., access);
2) intervals in which a user is responsible for creating the next event in such a sequence.

A simple illustration of this concept is provided in Figure 6/X.140. The four interface events in a typical connection establishment sequence divide the connection establishment period into three responsibility intervals: two network-dependent intervals surrounding one user-dependent interval. User responsibility intervals must normally be "factored out" in specifying network performance objectives, since their durations are not under network control.

Figure 7/X.140 illustrates the responsibility transfer concept in more detail. Two general types of responsibility transfer events are identified and are defined below. Both are defined with respect to particular data communication functions and associated performance periods, which are defined in § 3.3.

3.1 *Network-user responsibility transfer*

A network-user responsibility transfer occurs upon issuance of any interface signal that:

1) initiates user activity needed to accomplish a specified function;
2) solicits a subsequent user response indicating that the required activity has been completed; and
3) suspends network activity on the function pending the expected response. Examples are network issuance of an *incoming call* signal (in Recommendation X.21) or packet (in Recommendation X.25) to a called user.

3.2 *User-network responsibility transfer*

A user-network responsibility transfer occurs on issuance of any interface signal that:

1) initiates network activity needed to accomplish a specified function;
2) solicits a subsequent network response indicating that the required activity has been completed; and
3) suspends user activity on the function pending the expected response. Examples are user issuance of *call request* and *call accepted* signals and packets in Recommendations X.21 and X.25, respectively.

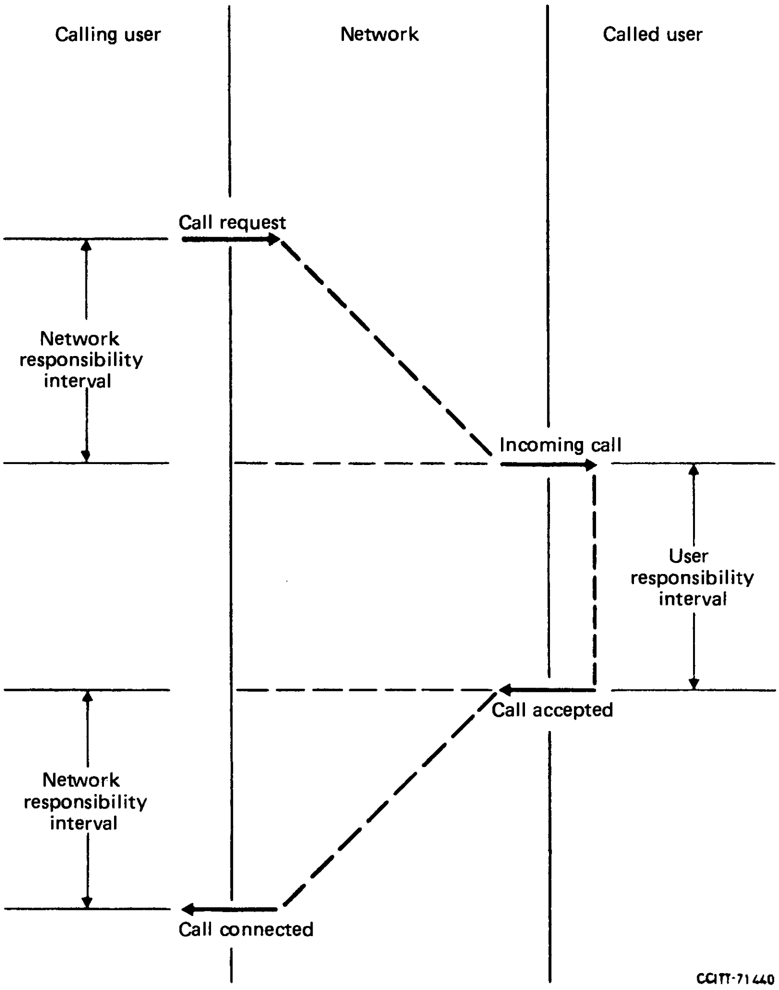
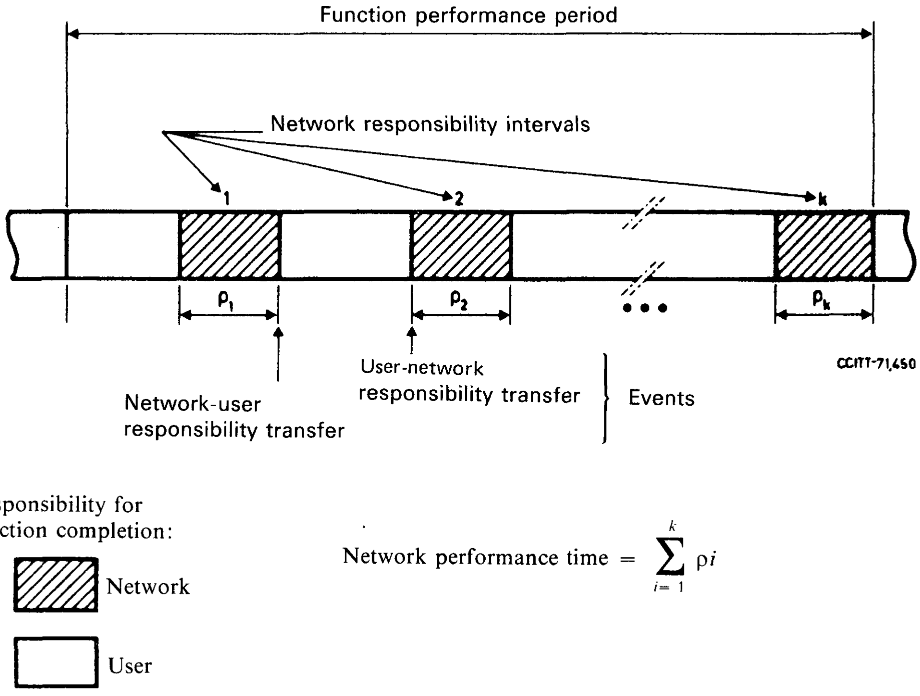


FIGURE 6/X.140

Illustration of network-user responsibility intervals



$$\text{Network performance time} = \sum_{i=1}^{k} \rho i$$

FIGURE 7/X.140

Use of responsibility transfer events in calculating network performance times

3.3 *Use of the responsibility transfer events*

The responsibility transfer events may be used in defining user and network responsibility intervals within four specific performance periods:

1) the period between the beginning and the end of an access attempt;
2) the period between the beginning and the end of a block transfer attempt;
3) the period between the beginning and the end of a disengagement attempt (for a specified user);
4) the period delimiting the larger of the input time or the output time for an individual transfer sample (as discussed in § 2.2.2).

Defining user and network responsibility intervals within the access, block transfer, and disengagement performance periods enables the specification of separate network and user values for access delay, user information transfer delay, and disengagement delay.

Separation of the above performance periods into user and network components also provides a method of establishing "responsibility" for timeout performance failures i.e. whether the user or the network should be charged with the failure when a performance trial is not completed within the established timeout period (and no blocking signals are issued). This decision is made by comparing the user performance time for the trial that failed with a specified maximum user performance time. If the observed user performance time exceeds the specified maximum, the failure is attributed to the user; otherwise, the failure is attributed to the network. This procedure is used in distinguishing access denial from user blocking (§ 2.1.3); user information loss from user information refusal (§ 2.2.6); and disengagement denial from user disengagement blocking (§ 2.3.2). It is also used in distinguishing the transfer denial and rejected sample outcomes in defining user information transfer denial probability (§ 2.4.2) and user information transfer rate (§ 2.2.2).

4 Supplementary information

This section specifies supplementary information which should be provided in conjunction with any statement of values for the general QOS parameters. The specified information is of two types:

1) information which identifies the intended scope of application of the parameter values;
2) information which identifies the particular statistical meaning each value expresses.

Significant differences between user requirement specifications, provider service specifications and measurement reports are noted.

4.1 *Scope of application*

The intended scope of application of stated QOS values should be defined by specifying the following interface and usage characteristics:

1) user-network interfaces to which the values apply;
2) interface event sequences (e.g., call request, incoming call, call accepted, call connected, etc.) by which the specified data communication service is provided in a typical instance;
3) service refusal actions allowed by the user/network interface protocol (e.g. network clearing in response to a user call request);
4) population of users (or communication instances, such as calls) to which the values apply;
5) operating conditions (or range of conditions) under which the values may be expected to hold.

Particular characteristics may be specified generally or more precisely, depending on the type of specification. User requirement specifications define a service need (and any constraints imposed by the user application) without reference to a particular service offering. User-network interfaces and interactions should be defined generally in such specifications, with particular mechanical, electrical, or procedural characteristics identified only where necessary. The population of users and any user-controlled operating conditions (e.g. service time interval, offered traffic) should be defined explicitly. User delays, user information unit lengths, user input/output rates and the selection of user facilities such as abbreviated address calling should also be explicitly defined.

Provider service specifications describe the quality of a particular offered service, often without references to any particular user application. The user-network interfaces and interactions are normally defined explicitly in such specifications (e.g. by reference to an interface specification such as Recommendations X.21 or X.25). The user (or call) population and the operating conditions may be defined more generally, since they refer to potential rather than actual usage.

Measurement reports summarize the actual performance of a network service. Both interface and usage conditions should be specified in detail in such reports to ensure comparability. Details which should be specified include:

1) the particular user-network interfaces instrumented;
2) the exact user-network interaction sequence observed during the measurements, and the performance significance assigned to each interface event;
3) the sampling plan used to select the measurement points, times, and conditions; and
4) the values or ranges of relevant operating conditions. Confidence limits and levels associated with each measured value should also be stated.

4.2 *Statistical meaning*

In order to clearly define the statistical meaning of a stated QOS parameter value, it is necessary to:

1) identify the particular distribution characteristic the parameter represents; and
2) specify values for any variables which may influence the parameter definition.

Any generally accepted statistical measure may be used in stating values for the general QOS parameters. "Mean" and "95 percentile" values are specified for the protocol-specific performance parameters in the X.130-series Recommendations.

The principal variables which may influence the definitions of the general QOS parameters are listed below:

1) *performance timeouts*
   - access
   - disengagement
   - transfer
   - sample input/output
2) *maximum user performance times*
   - access
   - disengagement
   - transfer
   - sample input/output
3) *transfer denial criteria*
   - transfer sample size
   - user information error probability threshold
   - user information loss probability threshold
   - extra user information delivery probability threshold
   - user information transfer rate threshold
4) *service outage criteria* (for further study)
   - observation period(s)
   - defining events
   - supported parameters
   - unacceptable performance thresholds.

The performance timeouts establish upper bounds on the associated delay distributions. The maximum user performance times provide a basis for identifying and eliminating user-caused failures. The transfer denial and service outage criteria distinguish "unacceptable" performance periods from periods of "satisfactory or tolerable" service.

Specifications should also indicate whether stated values are "target" or "minimum acceptable" values.

ANNEX A

(to Recommendation X.140)

Relationships between the general QOS parameters and the circuit-switched service performance parameters

This annex describes relationships between the general QOS parameters defined in Recommendation X.140 and the X.21-based, circuit-switched service performance parameters for which limits are specified in Recommendations X.130 and X.131. It illustrates one application of the general parameters and provides a framework for relating user QOS needs with the performance capabilities of circuit switching PDNs. Such relationships may be defined either to allocate a user requirement among network elements, in cases where the network performance values are selectable, or to derive resultant QOS values from network performance values, in cases where the latter are fixed.

In the example presented here, it is assumed that quality is to be specified at a pair of DTE/DCE physical interfaces conforming to Recommendation X.21. The call set-up and clearing sequences presented are derived from Annex B of that Recommendation.

Table A-1/X.140 lists the general parameters and the circuit-switched parameters in the rows and columns of a matrix and indicates qualitative relationships between them. Specific network performance parameters are listed for call processing delays (Recommendation X.130) and call blocking (Recommendation X.131).

Within the delay and blocking categories, a mark at a particular row/column intersection indicates that the corresponding parameters are interdependent and should be considered jointly in service performance specification. Each general parameter is influenced by a corresponding circuit-switched parameter, and may influence its value if the latter is selectable. Detailed relationships between the general parameters and the corresponding X.130 delay and X.131 blocking parameters are described below.

Figure A-1/X.140 illustrates the relationship between the access delay and the X.130 parameter network post selection delay. Access delay here describes the total time between the user's issuance of an X.21 *call request* and the network's subsequent issuance of *ready for data.* The X.130 parameter network post selection delay describes two specific network-dependent components of access delay.

Figure A-2/X.140 illustrates the relationship between disengagement delay and the X.130 parameter network clear indication delay. Two independent disengagement delays are identified:

1) Originator disengagement delay – the total time between *DTE clear request* and *DCE ready* at the clearing DTE interface.

2) Non-originator disengagement delay – the total time between *DTE clear request* at the clearing DTE interface and *DCE ready* at the cleared DTE interface.

Network clear indication delay contributes directly to non-originator disengagement delay, but does not include the delays associated with issuance of the *DTE clear confirmation* and *DCE ready* signals at the cleared DTE interface.

Access denial probability corresponds to the probability of blocking in the X.21 application. It includes cases where no *network congestion* signal is issued.

The X.140 access and disengagement parameters may be used to describe the quality of X.21 leased circuit services by simply specifying their values as zero.

TABLE A-1/X.140

Qualitative relationships between the general parameters and the circuit-switched parameters

Circuit-switched parameters (X.21 protocol) General parameters	Delay (X.130)		Blocking (X.131)
	Network post selection delay	Network clear indication delay	Probability non-connection due to congestion (blocking probability)
Access delay	■		
Incorrect access probability			
Access denial probability			■
User information transfer delay			
User information transfer rate			
User information error probability			
Extra user information delivery probability			
User information misdelivery probability			
User information loss probability			
Disengagement delay		■	
Disengagement denial probability			
Service availability			
User information transfer denial probability			
Service outage duration			

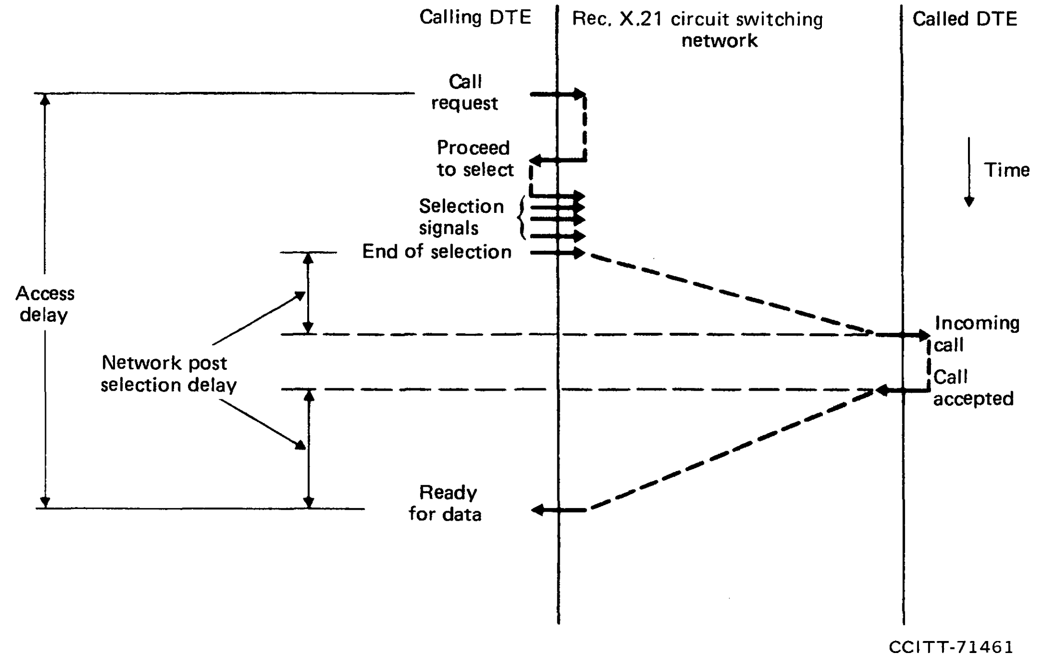
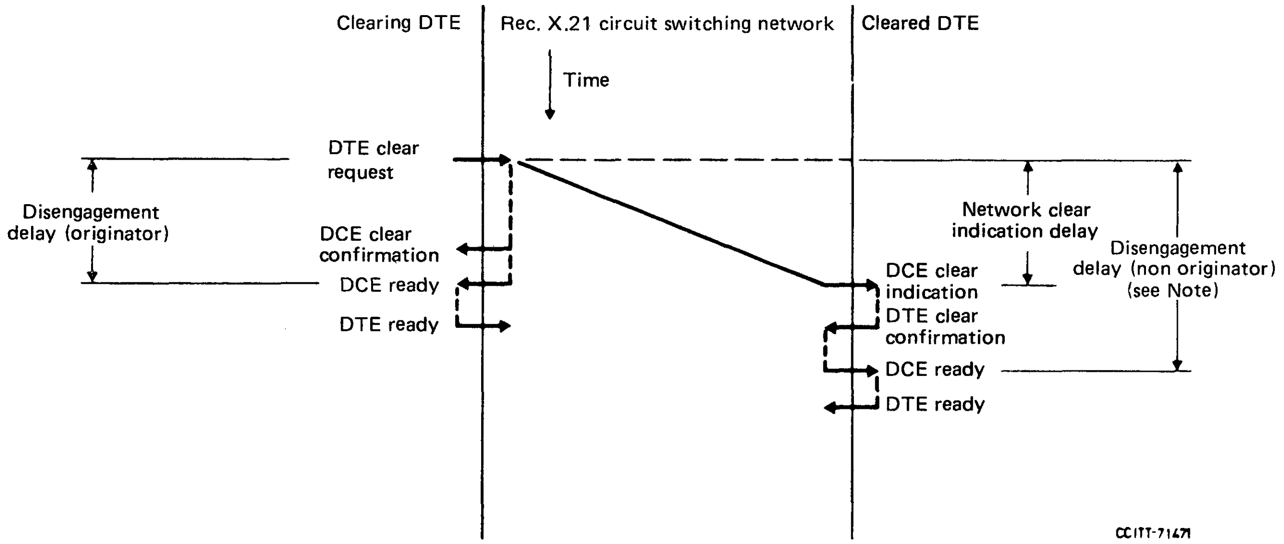


FIGURE A-1/X.140

Relationship between access delay and the Recommendation X.130 network post selection delay



Note – Specification of separate disengagement delays for each participating user is optional.

FIGURE A-2/X.140

Relationship between disengagement delay and the Recommendation X.130 network clear indication delay

ANNEX B

(to Recommendation X.140)

Relationships between the general QOS parameters and the packet-switched service performance parameters

This annex describes relationships between the general QOS parameters defined in Recommendation X.140 and the X.25-based, packet-switched service performance parameters for which limits are specified in X.130-series Recommendations. It illustrates a second application of the general parameters and provides a framework for relating user QOS needs with the performance capabilities of packet-switched PDNs. Such relationships may be defined either to allocate a user requirement among network elements, in cases where the network performance values are selectable, or to derive resultant QOS values from network performance values, in cases where the latter values are fixed.

In the example presented here, it is assumed that quality is to be specified in terms of packet-layer reference events observed at the physical boundaries separating communicating DTEs from their adjacent access circuit sections. It is assumed that Recommendation X.25 procedures are used on the access circuit sections. The section boundaries and the specific packet reference events are defined in Recommendation X.134. The packet-switched service performance parameters are defined in Recommendations X.135-X.137. The call set-up and clearing sequences presented here are derived from state diagrams presented in Annex B of Recommendation X.25.

Table B-1/X.140 lists the general parameters and the packet-switched service parameters in the rows and columns of a matrix and indicates qualitative relationships between them. Each set of parameters is grouped in four categories: access parameters, user information transfer parameters, disengagement parameters, and availability parameters.

A mark at a particular row/column intersection in the matix indicates that the corresponding parameters are interdependent and should be considered jointly in service performance specification. Each general parameter is influenced by one or more packet-switched service parameters, and may influence their values if the latter are selectable. Detailed relationships between the general parameters and corresponding X.135-X.137 parameters are described below.

The relationship between access delay and call set-up delay is illustrated in Figure B-1/X.140. As described in Recommendation X.135, call set-up delay can be defined either at a single virtual connection section boundary or between two section boundaries. When defined at the calling DTE boundary B_1, call set-up delay differs from access delay in only one respect: it includes the modulation time (X) of the call request packet on the calling DTE access circuit section, while access delay does not. When defined between the calling and called DTE boundaries B_1 and B_n, call set-up delay differs from access delay in one additional respect: it excludes the called DTE response time (i.e., the call set-up delay at boundary B_n).

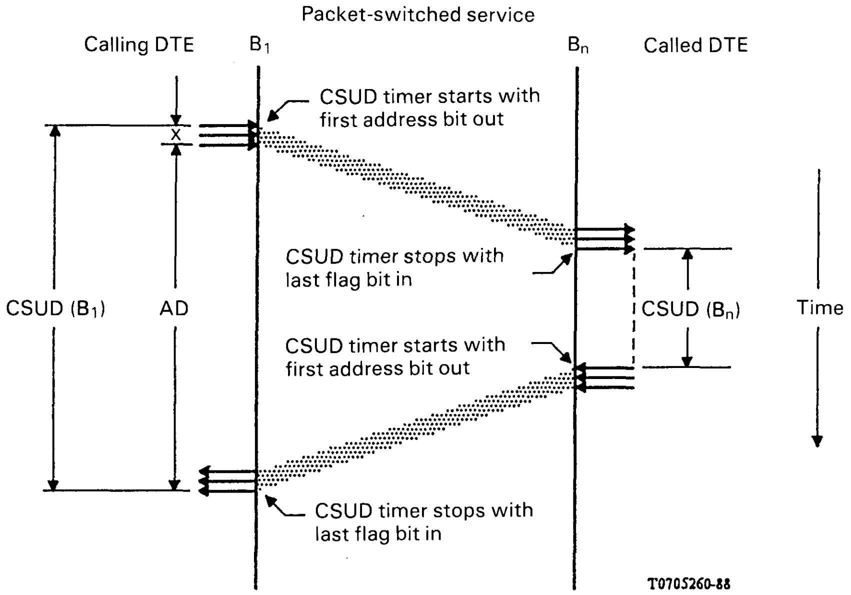
The general parameters user information transfer delay and disengagement delay correspond closely with the packet-switched service parameters data packet transfer delay and call clearing delay, respectively, when each parameter is defined at the X.25 DTE boundaries. Data packet transfer delay includes the modulation time (Y) of the data packet on the originating DTE access circuit section, while user information transfer delay does not. Similarly, call clearing delay includes the modulation time (Z) of the clear request packet on the clearing DTE access circuit section, while disengagement delay does not.

The general parameters incorrect access probability, access denial probability, and disengagement denial probability are essentially identical to the packet-switched service parameters call set-up error probability, call set-up failure probability, and call clear failure probability, respectively, as defined at the X.25 DTE boundaries. The packet-switched service parameter throughput capacity expresses the maximum continuously achievable (steady-state) value of the general parameter user information transfer rate; the former parameter also differs from the latter in that its definition allows it to be measured at a single boundary.

TABLE B-1/X.140

Qualitative relationships between the general parameters and the packet-switched service parameters

Packet-switched service parameters (X.25 protocol) / General parameters	Call set-up delay	Call set-up error probability	Call set-up failure probability	Data packet transfer delay	Through-put capacity	Residual error rate	Reset stimulus probability	Reset probability	Premature disconnect stimulus probability	Premature disconnect probability	Clear indication delay	Call clear failure probability	Service availability	Mean time between service outages
Access delay	■													
Incorrect access probability		■												
Access denial probability			■											
User information transfer delay				■										
User information transfer rate					■									
User information error probability						■								
Extra user information delivery probability						■								
User information misdelivery probability						■								
User information loss probability						■	■	■	■	■				
Disengagement delay											■			
Disengagement denial probability												■		
Service availability													■	
User information transfer denial probability													■	
Service outage duration														■



$$\mathrm{CSUD} = \mathrm{CSUD}(B_1) - \mathrm{CSUD}(B_n)$$
$$\mathrm{AD} = \mathrm{CSUD} + \mathrm{CSUD}(B_n) - X$$

AD Access delay
CSUD Call set-up delay

FIGURE B-1/X.140

Relationship between access delay and call set-up delay

The packet-switched service parameter residual error rate combines the three general parameters user information error probability, extra user information delivery probability, and user information loss probability in a single composite accuracy measure. The mathematical relationship between residual error rate and the three general parameters is specified in Recommendation X.136. There is no packet-switched service parameter that corresponds directly with the general parameter user information misdelivery probability; however, misdelivered data is counted as extra data under the Recommendation X.136 definitions, and is thus reflected indirectly in the definition of residual error rate. The reset and premature disconnect parameters defined in Recommendation X.135 are protocol dependent and thus have no direct counterparts among the general parameters specified in Recommendation X.140. Their values will normally influence the X.140 parameter user information loss probability.

Both Recommendation X.137 and Recommendation X.140 define a measure of service availability. The former measure specializes the latter by identifying the particular decision parameters and thresholds that are to be used in defining outages in a packet-switched service. Recommendation X.140 defines a closely-related general parameter, user information transfer denial probability, that provides a sampled measure of unavailability. It is based on a specific definition of outage that differs from that presented in Recommendation X.137 in one respect: the former definition includes only user information transfer parameters among the supported (decision) parameters used in identifying outages, while the latter definition includes call set-up parameters as well. The X.140 parameter service outage duration and the X.137 parameter mean time between service outages provide complementary information on the frequency of transitions between the available and unavailable states.

ANNEX C

(to Recommendation X.140)

Relationships between the general QOS parameters and the OSI network layer service performance parameters

This annex describes relationships between the general QOS parameters defined in Recommendation X.140 and the OSI network service (NS) performance[2] parameters defined in Recommendation X.213. It illustrates application of the general parameters to a specific (abstract) OSI service interface – the interface between the transport and network layers.

Table C-1/X.140 lists the general parameters and the NS performance parameters in the rows and columns of a matrix and indicates qualitative relationships between them. Each set of parameters is grouped in four categories: access parameters, user information transfer parameters, disengagement parameters, and availability parameters.[3]

Recommendation X.213 defines exact counterparts to five X.140 parameters: access delay, user information transfer delay, user information transfer rate, disengagement delay, and disengagement denial probability.

The X.140 parameters subdivide the X.213 parameters into more detailed components in two cases. The X.213 parameter NC establishment failure probability subsumes two X.140 parameters: access denial probability and incorrect access probability. Values for the X.140 parameters could be added to calculate their X.213 counterpart. The X.213 parameter residual error rate subsumes four X.140 parameters: user information loss probability, user information misdelivery probability, extra user information delivery probability, and user information error probability. The relationships among these probabilities are illustrated in Figure C-1/X.140. Each parameter is normalized so that its possible values range between 0 and 1.

The X.213 parameter NC Resilience is protocol dependent and this has no direct counterpart among the general parameters specified in Recommendation X.140. Its value will normally influence the X.140 parameter user information loss probability.

The X.140 parameter user information transfer denial probability corresponds closely with the X.213 parameter transfer failure probability; the two differ only in the detailed definition of the supported (decision) parameters used in defining transfer denial (or failure).

Two X.140 parameters have no X.213 counterparts: service outage duration and service availability.

---

[2] Recommendation X.213 distinguishes QOS parameters which describe performance from those which describe other service characteristics [network connection (NC) protection priority, and maximum acceptable cost]. Only the former parameters are addressed here.

[3] Transfer failure probability is included among the user information (data) transfer parameters in Recommendation X.213; that Recommendation does not identify availability as a separate parameter category.

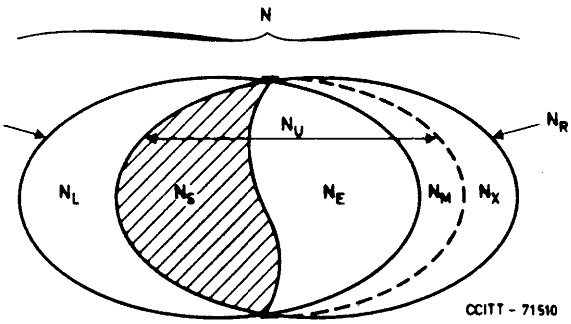
TABLE C-1/X.140

Qualitative relationships between the general parameters and the OSI network layer service performance QOS parameters

OSI network service parameters (X.213) / General parameters	NC establishment delay	NC establishment failure probability	Transit delay	Throughput	Residual error rate	NC resilience	NC release delay	NC release failure probability	Transfer failure probability
Access delay	●								
Incorrect access probability		■							
Access denial probability		■							
User information transfer delay			●						
User information transfer rate				●					
User information error probability					■				
Extra user information delivery probability					■				
User information misdelivery probability					■				
User information loss probability					■	■			
Disengagement delay							●		
Disengagement denial probability								●	
Service availability									
User information transfer denial probability									■
Service outage duration									

● General parameter is identical to the corresponding X.213 parameter when specialized to the OSI network service interface.

■ Corresponding parameters are interdependent but not identical.



Rec. X.213

$$\text{RER} = \frac{N_L + N_E + N_X}{N}$$

(N_M not distinguished from N_X)

N_T	Number transmitted
N_R	Number received
N_L	Number lost
N_S	Number successfully transferred
N_E	Number received with errors
N_M	Number misdelivered
N_X	Number of extras

Rec. X.140

$$P(L) = N_L / N_T$$
$$P(X) = N_X / N_R$$
$$P(E) = N_E / N_U$$
$$P(M) = N_M / N_U$$

P(L)	User information loss probability
P(X)	Extra user information delivery probability
P(E)	User information error probability
P(M)	User information misdelivery probability
RER	Residual error rate

FIGURE C-1/X.140

Relationships among the Recommendations X.140 and X.213 transfer failure probabilities

Recommendation X.141

GENERAL PRINCIPLES FOR THE DETECTION AND CORRECTION OF ERRORS IN PUBLIC DATA NETWORKS

(Malaga-Torremolinos, 1984)

The CCITT,

considering

(a) that errors have to be detected and corrected with a very high degree of reliability;

(b) that some error correction procedures may be more advantageous than others depending on transmission delays in the network and on the distribution (with time) of errors;

(c) that the distribution (with time) of errors at the ends of a path may depend on forward error correction procedures implemented in one or more of the path segments;

(d) that the applicability of some error correction procedures may be affected by the number of satellite systems in the connection, which may be in national or international links or in the Maritime Mobile Service;

(e) that different error correction procedures may be concatenated in some connections, *unanimously recommends*

that the general principles identified in this Recommendation be taken into account in the design and application of procedures for the detection and correction of errors in public data networks.

1 General

1.1 The purpose of this Recommendation is to describe general principles applicable to the detection and correction or recovery of link transmission errors in public data networks.

1.2 Two fundamental objectives of error control procedures are:

- to ensure an incidence of undetected errors that is within acceptably low probability limits;
- to ensure that detected errors are corrected or recovered using an error control procedure consistent with data throughput and sequencing requirements which apply when the error rate of the Physical Layer is within the fully acceptable and the tolerance limits of specified performance.

1.3 In the context of the Reference Model of Open Systems Interconnection, it is noted in Recommendation X.200 that each (N) peer protocol should include sufficient control information to enable the (N) entities to detect or recover from error conditions within its purview. Reporting detected but unrecovered errors is a service that must be provided by each layer.

Specifically, it is an objective of the Data Link Layer to detect and possibly correct errors which may occur in the Physical Layer.

1.4 For any particular error detection arrangement the probability of undetected errors will generally tend to increase:

- with increasing error rate,
- for any given error rate, as the error distribution becomes less random and as the length of error bursts increases,
- with increasing frame length,
- possibly due to scrambling arrangements which may have factors in common with the generating polynomial used for error detection.

1.5 Data throughput in the presence of errors depends on the design of the error control procedure, which in turn depends on the following conditions:

- error rate,
- error distribution,
- scrambling and/or multiplexing arrangements insofar as they affect the error distribution or error rate,
- transmission path (propagation) time delay,
- data signalling rate,
- frame length,
- window size,
- buffer memory resources at the sending and receiving end of the link.

2 Types of error occurrences

Error occurrences are typically of three types distinguished by characteristic error distributions with time:

- random errors,
- burst errors,
- errors due to uncontrolled slip.

It is likely that one type of error occurrence will be predominant in any particular link, depending on the type of transmission systems employed (i.e. cable, microwave radio relay, or satellite, with or without forward error correction).

In the design of error control procedures for a link, it is important to identify any tendency for the predominance of a particular type of error occurrence.

3 Error control procedures

3.1 *Types of procedure*

Two types of error control commonly employed in public data networks (PDNs) are:

- forward error correction, a coding method employed with the objective of detecting and correcting errors in received data instead of requesting retransmission,
- ARQ procedures wherein transmitted information is formatted in frames with error detection encoding, and error recovery is achieved by automatic repetition upon request from the data receiver of a frame or of all information already transmitted starting with the requested frame. Timeout recovery serves as a backup for the ARQ procedure.

3.2 *Forward error correction*

Forward error correction (FEC) does not require the provision of a backward mechanism in order to operate. FEC is usually applied at the Physical Layer of the reference model, typically within transmission systems whose error performance might not otherwise meet required limits.

The capability of FEC techniques commonly used in PDNs to control errors tends to be restricted to the correction of a limited number of errors (typically 2 or 3 errors) within each coded information block or constraint of block length. For this reason these FEC procedures are most effective in of situations where error occurrences are predominantly random.

Depending on multiplexing arrangements and sometimes on other arrangements in the Physical Layer such as scrambling and encryption, residual uncorrected errors after FEC may tend to be grouped in clusters or error bursts. When the number of errors within a coded information frame or constraint length of code exceeds the correction capability of the FEC algorithm, the total number of errors in the cluster or burst may be increased rather than reduced by the FEC facility.

Usually, it will not be feasible to notify the Data Link Layer of detected but uncorrected errors via FEC facilities of the Physical Layer which may perform their error control function at a multichannel, multiplexed signal level of the transmission system.

In the adaption of data signals at recommended bit rates below 64 kbit/s for transmission at 64 kbit/s, sufficient redundancy will be introduced in some cases for forward error correction to be undertaken on a majority voting basis without special forward error correction encoding. With this arrangement, a large number of different error patterns can be detected and corrected.

Alternatively, or in addition, the frame checking sequence of the ARQ error control procedure may also be used to distinguish between correctly and incorrectly received information in redundant signal streams.

3.3 *ARQ procedures*

3.3.1 *General*

ARQ procedures require the provision of forward and backward channels, usually with simultaneous transmission capability.

ARQ control procedures of error detection and error recovery are included in the functions of the Data Link Layer and may also be implemented in the functions of higher layers of the reference model.

3.3.2 *Error detection*

3.3.2.1 *Frame checking sequence*

The 16-bit frame checking sequence (FCS) described below is used for error detection in the packet transfer procedures of Recommendations X.25 and X.75, in the Signalling System No. 7 signalling link procedure of Recommendation Q.703 and in the Link Access Procedure on the D-channel of an ISDN as described in Recommendation Q.921 (I.441).

The same generator polynomial is also used in the encoding and checking process of Recommendation V.41.

The 16 FCS bits are generated at the transmitter. They are the 1s complement of the sum (modulo 2) of:

1) the remainder of x^h $(x^{15} + x^{14} + x^{13} + \ldots + x^2 + x + 1)$ divided (modulo 2) by the generator polynomial $x^{16} + x^{12} + x^5 + 1$, where h is the number of bits in the frame existing between, but not including, the final bit of the opening flag and the first bit of the FCS, excluding bits inserted for transparency, and

2) the remainder after multiplication by x^{16} and then division (modulo 2) by the generator polynomial $x^{16} + x^{12} + x^5 + 1$ of the content of the frame existing between, but not including the final bit of the opening flag and the first bit of the FCS, excluding bits inserted for transparency.

As a typical implementation, at the transmitter, the initial remainder of the division is preset to all 1s and is then modified by division by the generator polynomial (as described above) on the address, control and information fields; the 1s complement of the resulting remainder is transmitted as the 16-bit FCS sequence.

At the receiver the correspondence between the check bits and the remaining part of the frame is checked. If a complete correspondence is not found the appropriate error recovery procedure is initiated.

As a typical implementation at the receivers, the initial remainder is preset to all 1s, and the serial incoming protected bits including the check bits (after the bits inserted for transparency are removed) when multiplied by x^{16} and then divided by the generator polynomial will result in a remainder of 0001110100001111 (x^{15} through x^0, respectively) in the absence of transmission errors.

Explanatory notes concerning the FCS error detection procedure described above are given in Appendix I.

The procedure will detect:

a) all odd numbers of errors within a frame,

b) any error burst not exceeding 16 bits in length,

c) all two-bit errors when the code length is less than 32768 bits,

d) a large percentage of other error patterns (with even numbers of errors).

3.3.2.2 *Use of scramblers*

The following system design consideration should be taken into account concerning the use of self-synchronizing scramblers:

Where self-synchronizing scramblers (i.e. scramblers which effectively divide the message polynomial by the scrambler polynomial at the transmitter and multiply the received polynomial by the scrambler polynomial at the receiver) are used, the scrambler polynomial and the generating polynomial for error detection must have no common factors in order to ensure satisfactory performance of the error-detecting system. Where this condition cannot be maintained, the scrambling process must precede the error detection encoding process and the descrambler process must follow the error detection decoding process. Where additive (i.e. non-self-synchronizing) scramblers are used or where the scrambling takes place at a multi-channel multiplexed signal level, this design precaution need not be observed.

3.3.2.3 *Frame integrity*

The integrity of the frame format must be maintained in order to assure proper functioning of the error detection procedure described in § 3.3.2.1.

The frame structure for all transmissions is distinguished by opening and closing flags, each consisting of one 0 followed by six contiguous 1s and one 0. A single flag may be used as both the closing flag for one frame and the opening flag for the next frame. To ensure that the unique flag sequence is not simulated, the entire frame content between two flag sequences is examined at the transmitter and a 0 bit is inserted after all sequences of 5 contiguous 1 bits (including the last 5 bits of the FCS). At the receiver, the frame content is reexamined and any 0 bit which directly follows 5 contiguous 1 bits is discarded.

At the receiver, a frame validity check is carried out to detect any invalid frames not properly bounded by two flags or having fewer than the specified minimum number of bits. Invalid frames are treated in the same way as frames with detected errors.

3.3.3 *Error recovery procedures*

In accordance with ARQ concepts, error recovery is vested in the traffic control procedure wherein all information frames are numbered sequentially in order of transmission, from 0 through modulus minus 1 (where *modulus* is the modulus of the sequence numbers). Typically, the modulus equals 8 or 128 and the sequence numbers cycle through the entire range.

Valid frame without errors received in proper sequence are acknowledged in responses from the receiver to the transmitter, while invalid frames and frames with errors are discarded by the receiver and completely ignored. Frame recovery action is initiated by the receiver when a valid frame without errors does not have the expected sequence number. Consequently, when one or more frames are discarded for lack of validity or for errors, the number of the next correctly received frame will be out of sequence, causing the receiver to initiate the prescribed frame recovery procedure.

If, due to a transmission error, the receiver does not receive (or receives and discards) a single information frame or the last in a sequence of information frames, then the out-of-sequence condition which would otherwise serve to initiate error recovery procedures at the receiver will not be detected. In this case, frame recovery will be initiated at the transmitter via a time-out procedure as follows:

For traffic control purposes, the receiver must send acknowledgment response to the transmitter confirming the receipt of valid, error free frames. After a specified time-out period with outstanding transmitted frames and no acknowledgement or frame recovery responses from the receiver, appropriate recovery action is initiated at the transmitter to determine the point at which retransmission must begin.

Alternative types of error recovery procedure are available as follows:

- reject procedure,
- selective reject procedure,
- selective reject-reject procedure.

Each of these procedures requires that storage be provided at the transmitter for all information frames already sent but not yet acknowledged to be correctly received.

The data throughput efficiency obtainable as a function of error rate and distribution may depend significantly on the type of error recovery procedure, particularly in transmission links with associated long time delays (e.g. links via satellite). The complexity of error recovery implementation, including frame storage requirements at the receiver, is another consideration which plays an important part in selecting the most advantageous error recovery procedure to suit a particular situation.

3.3.3.1 *Reject (REJ) procedure*

The *reject* (REJ) error recovery procedure is used by the receiver to request retransmission of information frames commencing with a specified sequence number and to simultaneously acknowledge satisfactory reception of all preceding information frames.

The rejected frame and all subsequent information frames already in transit at the time that the REJ response reaches the transmitter will be retransmitted.

After sending the REJ response, the receiver discards all incoming information frames until the lost frame is recovered. This procedure minimizes frame storage requirements at the receiver, but under marginal conditions of error performance it may result in poor throughput efficiency depending on the round-trip transmission delay between the transmitter and receiver.

With the REJ error recovery procedure, the window size should allow a maximum number k of outstanding frames, where k is the smallest integer not less than r, calculated as follows:

$$r = \frac{T \cdot D}{L}$$

where

T is the transmission rate (bit/s)

D is the round-trip delay (seconds)

L is the information frame length (bits).

3.3.3.2 *Selective reject procedure*

The *selective reject* (SREJ) response is used by the receiver to request retransmission of a single information frame identified by its sequence number and to simultaneously acknowledge satisfactory reception of all preceding information frames.

In accordance with the foregoing definition of the SREJ request, only one SREJ condition can be outstanding at any one time. Consequently, the ability of the SREJ procedure to make efficient use of the Physical Layer falls off very quickly as the rate of frame error occurrence exceeds one per round-trip propagation delay.

This difficulty may be alleviated by an alternative SREJ procedure which suppresses the acknowledgement function of the SREJ request and consequently allows a station to send a following SREJ request for retransmission of another fault information frame before the information frame in response to the first SREJ request has been correctly received. This alternative procedure can be particularly advantageous in the case of high speed transmission via satellite.

Pursuant to the alternative procedure, the selective reject frame SREJ is used to request retransmission of a single information frame numbered N(R) and the information frames numbered up to N(R) – 1 are not considered as accepted.

Subsequent information frames already in transit when the SREJ response reaches the transmitter will not be repeated (if received correctly). Hence, there is a minimum reduction in throughput efficiency as a function of increasing error rate on transmission paths with long time delays.

This advantage of the SREJ procedure, is realized at the cost of providing considerable frame storage capability and some processing for frame resequencing at the receiver.

With the SREJ recovery procedure, the window size should allow a maximum number k of outstanding frames where k is an integer not less than r as follows:

$$r = 2\,\frac{T \cdot D}{L}$$

3.3.3.3 *Selective reject-reject procedure*

A proposed selective reject-reject procedure is described below:

If the receiver detects the loss of a single information frame then, after satisfactory receipt of the next information frame, it sends a SREJ response to recover the lost frame. All information frames received satisfactorily in sequence after the lost frame are stored at the receiver pending recovery of the lost frame.

If the receiver detects the sequential loss of two information frames, it sends a REJ response and discards all subsequently received information frames until the lost frame is recovered.

If the loss of another information frame is detected prior to recovery from the SREJ exception condition, the receiver will store all information frames received subsequent to the first lost frame and prior to the second lost frame and will discard all information frames thereafter until the first lost frame is recovered. After recovery of the first lost frame, the receiver will send a REJ response for recovery of the second lost frame and the subsequently received but discarded frames.

Operated over transmission paths with long time delays, the SREJ-REJ error recovery procedure results in values of throughput efficiency as a function of error rate which are somewhat inferior to those obtained with the SREJ procedure and significantly better than those obtained with the REJ procedure.

For any prevailing error rate, the average occupancy of buffer memory for frame storage at the receiver with the SREJ-REJ procedure is significantly less than corresponding average buffer occupancy with the SREJ procedure.

With the SREJ-REJ error recovery procedure, the window size should allow a maximum number of outstanding frames, k, where k is the smallest integer not less than r calculated as follows:

$$r = 2\,\frac{T \cdot D}{L}$$

4 Concatenation of error control procedure

4.1 *Concatenation of FEC and ARQ procedures*

In links via satellite, the reduction in data throughput as a function of increasing error rate may be minimized by the concatenated use of both forward error correction and ARQ procedures.

Since forward error correction reduces the effective data throughput of the transmission medium under acceptable operating conditions when the error rate is low and is most effective for only a small percentage of the time when the error performance is marginal, the use of a more efficient ARQ procedure (e.g. SREJ instead of REJ) may be considered as an alternative to the concatenation of FEC and ARQ procedures.

4.2 *Concatenation of FEC procedures*

The utilization of two stages of forward error correction (FEC) coding may yield a very significant improvement in the performance of a satellite link. Figure 1/X.141 illustrates the general configuration of such a two-stage concatenated coding system. The diagram shows two pairs of interleavers which are discussed below and may be omitted in some cases.

The purpose of the interleaver pairs is to break up bursts of errors and to scatter these errors in a manner so as to minimize the probability that a long uncorrectable error burst will be presented to a decoder. The channel symbol interleaver pair may be deleted if the channel errors are known to be statistically independent. When an inner decoder error occurs, the decoder will output a burst of errors. Therefore, it is necessary to include an intercode interleaved pair which is designed to transform the occasional inner decoder error bursts so that they can be efficiently corrected by the outer decoder. Thus, a well designed concatenated coding system includes codes and interleavers designed to complement each other. The inner code should correct nearly all of the channel errors and the outer code should correct the residual errors caused by inner decoder failures and errors.

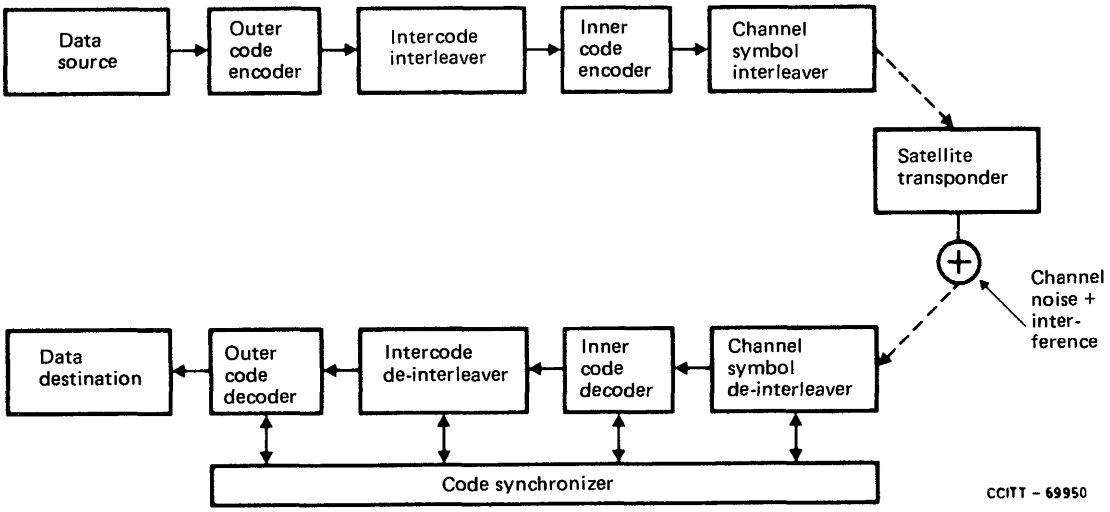


FIGURE 1/X.141

Two-stage concatenated FEC configuration

A final consideration in the selection of a concatenated code is that of decoding delay and synchronization. Care must be taken in selecting the codes and interleavers to insure that:

1) the delay introduced by the FEC system is small with respect to the satellite propagation time (approximately 250 ms), and
2) the data loss incurred due to a loss of synchronization is minimized.

APPENDIX I

(to Recommendation X.141)

Explanatory notes concerning the frame checking sequence

The following abbreviations are used in the explanatory notes given below:

$G(x)$ is the polynomial representing the k-bit sequence between the opening flag and the start of the FCS

$P(x)$ is the generator polynomial $(x^{16} + x^{12} + x^5 + 1)$

$L(x)$ is the polynomial representing 16 contiguous ones, $(x^{16} + x^{15} + x^{14} + \dots + x + 1)$

$R(x) = \overline{\text{FCS}}$ is the remainder obtained from the modulo 2 division:

$$\frac{x^{16}\, G(x) + x^k\, L(x)}{P(x)} = Q\,(x) + \frac{R(x)}{P(x)}$$

In the frame checking sequence (FCS), the multiplication of $G(x)$ by x^{16} corresponds to shifting the message $G(x)$ 16 places, thus providing the space of 16 bits for the FCS.

The addition of $x^k\, L(x)$ to $x^{16}\, G(x)$ is equivalent to inverting the first 16 bits of $x^{16}\, G(x)$ and corresponds to initializing the initial remainder to a value of all ones. This addition is provided to protect against the obliteration of leading flags, which may be non-detectable if the initial remainder is zero. The complementing of $R(x)$, by the transmitter, at the completion of the division ensures that the received, error-free message will result in a unique, non-zero remainder at the receiver. The non-zero remainder provides protection against potential non-detectability of the obliteration of trailing flags.

At the transmitter, the FCS is added to $x^{16}\, G(x)$ and results in a total message $M(x)$ of length:

$$n = k + 16, \text{ where } M(x) = x^{16}\, G(x) + FCS$$

At the receiver, the incoming $M(x)$ is multiplied by x^{16}, added to $x^n\, L(x)$ and divided by $P(x)$ as shown below:

$$\frac{x^{16}\, M(x) + x^n\, L(x)}{P(x)} = \frac{x^{16}\, [x^{16}\, G(x) + FCS + x^k\, L(x)]}{P(x)}$$

The following expressions are derived from those shown above, noting that the addition of $L(x)$ without carry to a polynomial $R(x)$ of the same length is equivalent to a bit by bit inversion of $R(x)$ and substituting $FCS = \overline{R(x)} = R(x) + L(x)$, then rearranging the terms of the numerator:

$$\frac{x^{16}\, [x^{16}\, G(x) + R(x) + L(x) + x^k\, L(x)]}{P(x)} =$$

$$\frac{x^{16}\, [x^{16}\, G(x) + x^k\, L(x) + R(x)] + x^{16}\, L(x)}{P(x)} = Qr(x) + \frac{Rr(x)}{P(x)}$$

If the transmission is error-free, the term $[x^{16}\, G(x) + x^k\, L(x) + R(x)]$ will be divisible by $P(x)$ and the remainder after division will be:

$$\frac{x^{16}\, L(x)}{P(x)},$$

or 0001110100001111 (x^{15} through x^0 respectively).

If the transmission is error-free and the FCS is inverted before division at the receiver, the remainder will be zero because inverting the FCS is equivalent to adding another $x^{16}\, L(x)$ to the numerator and

$$\frac{x^{16}\, L(x) + x^{16}\, L(x)}{P(x)} = 0.$$

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SECTION 5

MAINTENANCE

Recommendation X.150

PRINCIPLES OF MAINTENANCE TESTING FOR PUBLIC DATA NETWORKS USING DATA TERMINAL EQUIPMENT (DTE) AND DATA CIRCUIT-TERMINATING EQUIPMENT (DCE) TEST LOOPS

(Geneva, 1980; amended at Malaga-Torremolinos, 1984)

1 Introduction

The CCITT,

considering

(a) the increasing use being made of data transmission systems;

(b) the volume of the information circulating on data transmission networks;

(c) the savings to be made by reducing interruption time on such data circuits;

(d) the importance of being able to determine responsibilities (of necessity involving several parties) in maintenance questions for networks; and

(e) the advantages of standardization in this field,

unanimously declares

that the locating of faults can be facilitated in many cases by data circuit loop testing procedures in DTEs and DCEs.

2 Scope

This Recommendation specifies the principles of maintenance testing for public data networks using the DTE/DCE test loops. The definition of the loops, the principles for the control of the loops and the principles for the indications to be presented when the loops are activated are described herein for general application with all DTE/DCE interfaces. Specific details concerning the implementation of these principles are contained in the individual DTE/DCE interface Recommendations. The maintenance philosophy, definitions and general principles are applicable to interface Recommendations, such as X.20, X.20 *bis*, X.21, X.21 *bis*, and X.22.

2.1 *Maintenance philosophy*

The provision of DTE and DCE test loops in public data networks is based upon the following maintenance philosophy:

a) test loops may be used by an Administration's test centre(s) to test the operation of either leased lines or circuit-switched network subscriber lines, including either all or part of the DCE, without necessarily requiring the dispatch of network maintenance personnel to the subscriber's premises;

b) where allowed by national testing principles, DCE test loops may also be used by a DTE to test the operation of network connection or leased lines. Where provided, the intent is for the DTE to make initial tests to isolate a trouble condition either in the DTEs or in the data circuit.

2.2 *Loop testing principles*

The provision of loop testing capabilities should be based on the following principles:

a) the loops should be transparent, e.g. they should be bit sequence independent;

b) loop testing is a disruptive type of testing, e.g. when a loop has been established transmission of data is not possible;

c) test loops may be established from any state; however, when loop testing is commenced from the *data transfer* state, the same test procedures may be used for both leased lines and circuit-switched connections;

d) when allowed, the preferred DTE test sequence is: loop 1 – loop 2 – loop 3, sequentially from both ends of the circuit.

3 Definition of the loops

Nine loops are defined as shown in Figure 1/X.150. For clarity these nine loops have been grouped as follows:

a) DTE test loop – type 1 loop	(§ 3.1)
Loop 1	(§ 3.1.1)
b) Local test loops – type 3 loops	(§ 3.2)
Loop 3d	(§ 3.2.1)
Loop 3c	(§ 3.2.2)
Loop 3b	(§ 3.2.3)
Loop 3a	(§ 3.2.4)
c) Subscriber-line test loops – type 4 loops	(§ 3.3)
Loop 4a	(§ 3.3.1)
Loop 4b	(§ 3.3.2)
d) Network test loops – type 2 loops	(§ 3.4)
Loop 2b	(§ 3.4.1)
Loop 2a	(§ 3.4.2)

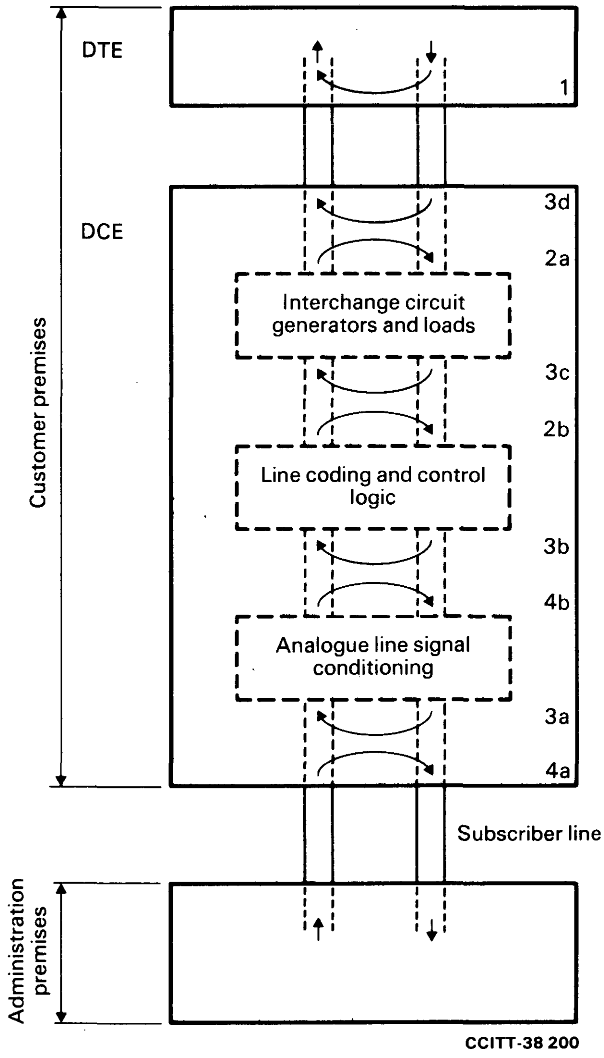
3.1 *DTE test loop – type 1 loop*

3.1.1 *Loop 1*

This loop is used as a basic test of the operation of the DTE, by looping back the transmitted signals inside the DTE for checking. The loop should be set up inside the DTE as close as possible to the DTE/DCE interface.

Loop 1 may be established from either the *data transfer* or *ready* state.

In some networks, for short routine tests during the *data transfer* state, the DTE should either maintain the same status on the interchange circuits as before the test or where possible send the *DTE controlled not ready* signal.



Note – The back-to-back loopbacks (e.g. 3d/2a. 3c/2b. 3b/4b and 3a/4a) that are provided should be configured in such a manner that there is no active equipment between the loopbacks. For example: an Administration may operate the back-to-back loopbacks simultaneously in the same relay or switch.

FIGURE 1/X.150

If the loop is established from the *data transfer* state, the DCE may continue to deliver data to the DTE during the test as though the DTE were in normal operation. It will be the responsibility of the DTEs to recover from any errors that might occur while the test loop is activated.

If the loop is established from the *ready* state, the DTE should continue to monitor so that an incoming call may be given priority over a routine test. Alternatively, in cases where the DTE cannot accept incoming calls, DTEs shall signal one of the *not ready* states.

The DCE continues to present signal element timing and, if implemented, byte timing. The DTE need not make use of the timing information during a loop test.

3.2 *Local test loops – type 3 loops*

Local test loops (type 3 loops) are used to test the operation of the DTE, the interconnecting cable and either all or parts of the local DCE, as discussed below.

Where allowed by national testing principles, loop 3 may be established from any state.

For testing on leased circuits and for short duration testing on circuit-switched connections, the DCE should either continue to present toward the line the conditions that existed before the test (e.g. either *data transfer* or *ready* state) or send the *controlled not ready* state to the remote DTE. Where this is not practical (e.g. in some cases for loop 3a) or desirable (e.g. for long duration testing in circuit-switched applications), the DCE should terminate an existing call and, if possible, signal toward the subscriber-line one of the *not ready* states.

The DCE continues to present signal element timing, and, if implemented, byte timing. The DTE must make use of the timing information.

3.2.1 *Loop 3d*

This loop is used to test the operation of the DTE, including the inter-connecting cable, by returning transmitted signals to the DTE for checking. The loop is set up inside the local DCE and does not include interchange circuit generators and loads. The DCE may make either type 2 or type 4 loop tests during the loop 3d test condition.

Note – While test loop 3d is operated, the effective length of the interface cable is doubled. Therefore, to insure proper operation of loop 3d, the maximum DTE/DCE interface cable length should be one-half the length normally appropriate for the data signalling rate in use.

3.2.2 *Loop 3c*

This loop is used to test the operation of the DTE, including the inter-connecting cable and DCE interchange circuit generators and loads. In this case, the Note in § 3.2.1 concerning restrictions of interconnecting cable length does not apply.

3.2.3 *Loop 3b*

This loop is used as a test of the operation of the DTE and the line coding and control logic and circuitry of the DCE. It includes all the circuitry of the DCE, with the exclusion of the line signal conditioning circuitry (e.g. impedance matching transformers, amplifiers, equalizers, etc.). The delay between transmitted and received test data is a few octets.

Note – In some DCEs, the setting of loop 3b will result in momentary loss of envelope alignment causing spurious signals to appear on receiving interchange circuits for a period of time. This may impact upon the DTE test procedure. Refer to the DTE/DCE Recommendations for information regarding the signal element timing. In some networks, the setting of loop 3b will cause clearing of existing connections.

3.2.4 *Loop 3a*

This loop is used to test the operation of the DTE and the DCE. The loop should include the maximum amount of circuitry used in DCE working including, in particular, the line signal conditioning circuitry. It is recognized that, in some cases, the inclusion of devices (e.g. attenuators, equalizers or test loop translators) may be necessary in the loopback path. The subscriber line is suitably terminated during a loop 3a test condition. The delay between transmitted and received test data is a few octets.

Note – In some DCEs, the setting of loop 3a will result in momentary loss of envelope alignment causing spurious signals to appear on the receiving interchange circuits for a period of time. This may impact upon the DTE test procedure. Refer to the DTE/DCE interface Recommendations for information regarding the signal element timing. In some networks, the setting of loop 3a will cause clearing of existing connections.

3.3 *Subscriber line test loops – type 4 loops*

Subscriber line test loops (type 4 loops) are provided for the maintenance of lines by the Administrations. The DCE signals to the local DTE a *not ready* or *test mode* state, as appropriate for the specific DTE/DCE interface. The DCE continues to present signal element timing and, if implemented, byte timing.

Note – In the case of loops 4 and 2 (see below) the DCE may signal the local DTE in such a manner that the DTE can distinguish a test mode from a network failure.

3.3.1 *Loop 4a*

This loop is only provided in the case of 4-wire subscriber lines. Loop 4a is for the maintenance of lines by Administrations. When receiving and transmitting pairs are connected together, the circuit under test may not be measured as a data circuit. Loop 4a may be established inside the DCE or in a separate device.

3.3.2 *Loop 4b*

This loop is used by Administrations to test the operation of the subscriber line including the line signal conditioning circuitry in the DCE. When the receiving and transmitting circuits are connected at this point, loop 4b provides a connection that can be considered as a data circuit.

Note – The implementation may be such that some impairment of the performance is expected since the DCE, in this case, does not perform a complete signal regeneration/conversion.

3.4 *Network test loops – type 2 loops*

Network test loops (type 2 loops) are used by the Administration's test centre(s) to test the operation of the leased line or subscriber line and either all or part of the DCE as discussed below.

Where allowed by national testing principles, loop 2 may also be used by a DTE, as follows:

- in case of switched circuit networks when the DTEs are in the *data transfer* state, to test the operation of the network connection including the remote DCE;
- in case of leased lines in the *ready* state, to test the operation of the line including the remote DCE.

The DCE signals the local DTE a *not ready* or *test mode* state as appropriate for the specific DTE/DCE interface (see Note to § 3.3). The DCE continues to present signal element timing and, if implemented, byte timing.

3.4.1 *Loop 2b*

This loop is used by either the Administration's test centre(s) and/or the remote DTE to test the operation of the subscriber line and all the circuitry of the DCE, with the exception of interchange circuit generators and loads.

3.4.2 *Loop 2a*

This loop is used by either the Administration's test centre(s) or the remote DTE to test the operation of the subscriber line and the entire DCE.

Note – While in the loop 2a condition, the DCE may present an *open circuit* condition to the DTE on certain interchange circuits. It is assumed that the DCE detects electrical signal faults condition as *not ready* as appropriate for the specific electrical characteristics.

4 **Minimum implementation of test loops**

4.1 *DCE test loops*

Sufficient test loops should be provided in the DCE to allow the customer and/or the Administration's maintenance personnel to positively distinguish between DTE and DCE/line faults.

The DCE will implement at least one of the four local test loops (type 3). The DCE will also implement at least one of the two network test loops (type 2). The implementation of the loops within the DCE is a national matter. Implementation of test loops beyond the minimum set specified above may be provided by some Administrations.

4.2 *DTE test loops*

It is suggested that all new DTEs provide loop 1.

5 Loop control

5.1 *General*

Where available, the means for remotely controlling a loop in one country from a location in another country are described in appropriate DTE/DCE interface Recommendations.

In leased circuit services, subscriber-line and network test loops should not be activated before the customer has been informed. However, some Administrations may activate these loops when abnormal conditions are detected in the network without first informing the customer.

In circuit-switched services, subscriber-line and network test loops should not be activated when the DTE is engaged in a call. In case of a collision between call request and the activation of these loops, the loop activation command will have priority and the call request will be cancelled. These loops may be activated without the prior knowledge and agreement of the customer for periods which normally do not exceed one second.

5.2 *Control of the local test loops*

To facilitate the testing of the DTE by the customer, manual activation (by means of a switch of the DCE) will be provided for at least one of the four local test loops (type 3). The precise implementation is a national matter. However, customer-controlled automatic activation of the local test loops via the DTE/DCE interface should be considered.

Where available, the means for DTE control of these loops via signals in the DTE/DCE interface are described in the appropriate DTE/DCE interface Recommendations.

Note – With the introduction of the electrical interchange circuits as defined in Recommendations X.26/X.27, some Administrations may locate the DCE in a location that is remote (up to 1000 metres) from the DTE. Therefore, manual loop activation by the customer may be either difficult or impossible. Thus, some form of an automatic activation of these loops should be considered. Also the limitations in the Note of § 3.2.1, loop 3d, should be considered.

5.3 *Control of the network test loops*

5.3.1 *General*

Each network test loop implemented in the DCE will be activated either by a manual switch on the DCE or remotely from the Administration's test centre(s) or both. The means for loop activation, the method for achieving remote control and the method for notifying the network of manual activation are national matters. Random signals may be delivered to the DTE prior to closing of loops.

Where loop 2a or 2b is provided for customer use, the procedure for its use is subject to the relevant interface Recommendations.

5.3.2 *Leased circuits*

5.3.2.1 *Point-to-point leased lines*

In case of point-to-point line circuits, the Administrations will provide one or more of the following:

a) customer control of the network test loop in the local DCE via a manually operated switch on the DCE;

b) customer control of the network test loop via the remote DTE/DCE interface.

   Note – The provision for the remote control of a loop in one country from a location in another country is subject to bilateral agreement between the affected Administrations.

c) remote control from the Administration's data test centre.

When available, the method for activation of the network test loop in a DCE by a command signal from a remote DTE/DCE interface will be as described in the appropriate DTE/DCE interface Recommendation.

5.3.2.2 *Centralized multipoint leased circuits*

The principle of maintenance testing for centralized multipoint circuits are for further study. In this regard, the specific details concerning the implementation of these principles are contained in the individual DTE/DCE interface Recommendations.

5.3.3 *Switched networks*

In a similar manner to the technique decribed in § 5.3.2.1 above, Administrations may provide a means for remotely controlling the network test loop from their test centre(s). This is recognized to be a national matter; however, if provided, the procedure to be used is described in the appropriate interface and signalling Recommendations. The remote control of network test loops in a DCE via the remote DTE/DCE interface is subject to bilateral agreement between the affected Administrations.

Note – Where allowed by national testing principles, the DTE may establish a network test loop in the remote DCE when in the *data transfer* state by means of a signal across the DTE/DCE interface. In this case, it should be possible for the DTEs to re-enter the data transfer state after deactivation of the network test loop. It will be the responsibility of the DTEs to recover from any errors that might occur while the test loop is activated.

5.4 *Control of subscriber-line test loops*

The provision of and use of subscriber-line test loops is a national matter. They are designed for the maintenance of lines by Administrations in case of 4-wire subscriber lines.

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SECTION 6

ADMINISTRATIVE ARRANGEMENTS

Recommendation X.180

ADMINISTRATIVE ARRANGEMENTS FOR INTERNATIONAL CLOSED USER GROUPS (CUGs)

(Geneva, 1980; amended at Melbourne, 1988)

The CCITT,

bearing in mind

(a) the introduction of international CUGs in public networks that provide packet-switched and circuit-switched data transmission services;

(b) the need to standardize a scheme for international CUG numbers;

(c) the need to standardize administrative procedures for allocation of international CUG numbers and the establishment of international CUGs,

unanimously recommends the following

1 One subscriber, hereafter called the "responsible subscriber", shall clearly be allocated the responsibility for all organizational matters relating to an international CUG. The responsible subscriber shall be nominated by the subscribers intending to form an international CUG (see Note 1).

2 The Administration of the country housing this "responsible subscriber" (hereafter called the "coordinating Administration") shall act as the controlling and coordinating Administration for that CUG and shall carry out the discussions with the responsible subscriber about changes to the CUG. The coordinating Administration shall also be responsible for allocating the international CUG number (ICN) and for issuing the necessary information to other Administrations involved in the CUG.

3 If the public network is a data network, the DNIC or DCC used for the construction of the ICN would be one proper to the coordinating Administration. If the public network is ISDN, the INIC (see Note 5) would be used for the construction of the ICN. If the responsible subscriber changes his country of residence, the ICN shall be changed in accordance with the DNIC, DCC or INIC of the new coordinating Administration.

4 The ICN allocated by the coordinating Administration shall be retained for the period of existence of the international CUG even if the location of members of the CUG changes, as long as the responsible subscriber is located within the area of the coordinating Administration.

5 The ICN will be represented by two decimal numbers A/B, where A is the DNIC, DCC (plus one digit) or INIC in accordance with § 3 above, and B is a 1-5 digit number (see Note 2).

6 In order to allow efficient conversion of CUG information, where required, for international CUG calls, restrictions on allocation of the ICNs used by each coordinating Administration have to be applied. Provisionally the following guidelines should apply:

i) each coordinating Administration should allocate the ICNs to international CUGs in sequence within a certain range of the available total range of ICNs;

ii) information should be sent regularly to the Administrations concerned about the size and allocation of the range used for ICNs in i) above;

iii) the size of the range used should not be bigger than necessary for the operation of the network.

7 The following procedures should apply for the interchange of information between Administrations and subscribers of an international CUG. Where a subscriber belongs to more than one international CUG, the procedures in this Recommendation must be applied separately for each international CUG.

7.1 A subscriber applying for membership of an international CUG shall apply to his own Administration using standard application procedures. He should supply full details of the responsible subscriber (see Note 3).

7.2 The Administration receiving the request should pass details in a standard format to the coordinating Administration as indicated in Annex A.

7.3 The coordinating Administration will then verify with the responsible subscriber whether the application can be permitted and, if acceptable, will inform the applying subscriber's Administration of the ICN allocated for that particular CUG.

7.4 The Administration of the applying subscriber shall inform the coordinating Administration when the applying subscriber is connected.

7.5 The existing member Administrations of an international CUG shall be informed by the coordinating Administration when a subscriber of a new Administration has become a member of that CUG.

7.6 Changes in membership of an international CUG or cessation of a CUG shall similarly be arranged between the responsible subscriber and the coordinating Administration following individual applications from the members of the CUG concerned.

7.7 The coordinating Administration shall, on request from the responsible subscriber or one of the member Administrations of an international CUG, supply information (print out) of all subscribers in that particular CUG. The approval of the responsible subscriber is needed in the latter case (see Note 4).

8 The fact that the applying subscriber may be a member of other CUGs or may wish to have outgoing or incoming access in addition to the CUG facility is of no relevance to the administrative procedures contained in this Recommendation.

Note 1 – It is assumed that in a CUG of an international organization the headquarters branch probably will be the responsible subscriber.

Note 2 – According to Recommendation X.300, the value B should not be greater than $2^{16} - 1 = 65\,535$.

Note 3 – To simplify the administrative arrangements, the procedures in §§ 6.1 and 6.2 should be followed also in the case where the applying subscriber is a branch of an international organization and another branch of that organization is the responsible subscriber.

Note 4 – Legal, or other, considerations may preclude some coordinating Administrations from supplying such information at the request of other member Administrations of an international CUG.

Note 5 – For ISDN identification, a four-digit field ISDN Network Identification Code (INIC) is used. INIC consists of 0 + E.164 country code + national network digit(s). In order to identify additional ISDNs, some countries may also use 9 + E.164 country code + national network digit(s). Alternative ways of ISDN network identification are for further study.

ANNEX A

(to Recommendation X.180)

The format and information to be passed to the coordinating Administration following an application from a subscriber to join or cease membership of an international closed user group is as follows:

1 Application to join/cease membership of an international closed user group (CUG) on a public network has been received from:

Firm
Address
Country
Public Network No.
Date of application

2 The responsible subscriber for this international CUG is:

Firm
Address
Public Network No.

3 The responsible subscriber's international CUG number (ICN)

4 The applying subscriber requires the following facility (if any):

- incoming calls barred within the CUG,
- outgoing calls barred within the CUG.

Recommendation X.181

ADMINISTRATIVE ARRANGEMENTS FOR THE PROVISION OF INTERNATIONAL PERMANENT VIRTUAL CIRCUITS (PVCs)

(Malaga-Torremolinos, 1984; amended at Melbourne, 1988)

The CCITT,

considering

(a) that the permanent virtual circuit service is listed in Recommendation X.2;

(b) that the permanent virtual circuit service may not be available for all international connections and may not be available for the maritime mobile services (as noted in Recommendation X.2);

(c) that the international application of the permanent virtual circuit service depends on bilateral agreement (as noted in Recommendation X.2);

(d) that Recommendation X.25 defines the DTE/DCE interface characteristics and Recommendation X.75 defines the procedures for international circuits between two STEs;

(e) that there is a need to standardize administrative procedures for the exchange of information characterizing the international permanent virtual circuit among Administrations concerned, in order to establish the international permanent virtual circuit, *unanimously recommends the following*

1 One of the two subscribers to be connected by the international PVC (hereafter called the "responsible subscriber") shall clearly be allocated the responsibility for all organizational matters relating to an international PVC. The responsible subscriber shall be nominated by the subscribers to be connected by the international PVC (see Note 1).

2 The Administration of country housing this "responsible subscriber" (hereafter called the "source Administration") shall act as the controlling and coordinating Administration for that international PVC and shall carry out the discussions with the responsible subscriber about any changes to the international PVC.

3 The source Administration will normally gather information required for charging and accounting for the international PVC (see Note 2).

4 The exchange of information for the provision of international PVCs will be as follows:

4.1 Source Administration and destination Administration (the Administration housing the non-responsible subscriber) will need to exchange information for the provision of the international PVC. This information should include:

a) international numbers of both subscriber DTEs,
b) throughput class (both directions),
c) name of the international PVC (see Note 3),
d) proposal on routing,
e) quality of service parameters (see Note 4),
f) service inauguration date and lifetime of the international PVC.

4.2 If transit networks must be involved, the route should be agreed on among Administrations involved. For the efficiency of negotiation among Administrations, the source Administration may initiate the negotiation with its adjacent Administration after which negotiation between subsequent Administrations follows, until the destination Administration has been reached. The information described in § 4.1 should be conveyed to each transit Administration, and the route should be determined satisfying the requested throughput classes and quality of service (see Note 4).

4.3 Between adjacent Administrations, information to be exchanged should include the following items in addition to the information described in § 4.1:

a) the identity of STE X/Y interface,
b) logical channel group number and logical channel number,
c) window size (both directions),
d) packet size (both directions).

5 The following procedures should apply for the interchange of information between Administrations and subscribers of an international PVC. Where a subscriber has subscribed to more than one international PVC, the procedures in this Recommendation must be applied separately for each international PVC.

5.1 A subscriber applying for the provision of an international PVC shall apply to his own Administration using standard application procedures. He should supply full details of the responsible subscriber.

5.2 The Administration should, in case the request did not come from the responsible subscriber, pass details in a standard format to the source Administration as indicated in Annex A.

5.3 The source Administration will then verify with the responsible subscriber whether the application can be permitted.

5.4 The source Administration will then negotiate with the destination Administration and possibly transit Administrations whether the application can be permitted, taking into account § 4.

5.5 If the conditions in §§ 5.3 and 5.4 are accepted, the source Administration will agree with the subscriber on the logical channel group number and logical channel number to which the international PVC will be connected. The destination Administration will do so with the other subscriber.

5.6 The source Administration will inform both subscribers involved when the international PVC is established.

5.7 If, during the lifetime of the international PVC, rerouting of the international PVC is necessary, the responsible subscriber will be informed by the source Administration of any changes of either charging or quality of service of the international PVC (see Note 5).

Note 1 – It is assumed that the responsible subscriber is one of the two subscribers to be connected by the international PVC.

Note 2 – Other arrangements are possible on a bilateral basis.

Note 3 – A name of an international PVC should be unique among the Administrations involved, (e.g., the specification of the international number, the logical channel group number and the logical channel number to which the international PVC is connected for both subscribers).

Note 4 – The parameters concerning quality of service are for further study.

Note 5 – The impact of a possible inclusion of an automatic (re)establishment procedure is for further study.

ANNEX A

(to Recommendation X.181)

The format and information to be passed to the source Administration following an application from a subscriber for the provision of an international PVC is as follows:

1 Application for the establishment of an international permanent virtual circuit (PVC) via several public networks has been received from:

Firm
Address
Country
International number
Date of application

2 The applicant as indicated in 1) above requests to be connected via an international PVC with:

Firm
Address
Country
International number

3 The responsible subscriber's international data number:

4 The following is requested:

- throughput class,
- service inauguration date,
- requested lifetime of the international PVC.

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PART II

SUPPLEMENT TO RECOMMENDATION X.135

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Supplement No. 1

SOME TEST RESULTS FROM SPECIFIC NATIONAL AND INTERNATIONAL PORTIONS

(referenced in Recommendation X.135)

This supplement presents test results and is intended for illustrative purposes only. It contains results for national A and national B portions. The results were measured in the DATEX-P network, which is operated by the Deutsche Bundespost in the Federal Republic of Germany.

Since these figures apply to one network under a specific network traffic load at a specific time, they cannot be taken in any way to be presentative of the current or likely performance of either other networks of this same network at a different point of time. They are included for the sole purpose of summarizing one experiment in which the network performance was better than that defined in Recommendation X.135.

The above implies that many factors, including a particular set of equipment types, a specific configuration, distribution of network traffic loading, network topology, and network-specific dimensioning rules, impact the values obtained.

1 National A portion delay and throughput values

Table 1 presents call set-up delay, data packet transfer delay, throughput capacity, and clear indication delay values measured in a DATEX-P configuration selected to represent this National A portion of an international virtual connection. The measurements were taken during the busy hour on a representative set of connections. These results demonstrate that the delay and throughput performance provided in the National A portion can be much better than is indicated by the worst-case values specified in Recommendation X.135.

2 National B portion delay and throughput values

Table 2 presents call set-up delay, data packet transfer delay, throughput capacity, and clear indication delay values measured in a DATEX-P configuration selected to represent the National B portion of an international virtual connection. The measured configuration included a 128 kbit/s satellite circuit. The measurements were taken during the busy hour. These results demonstrate that the delay and throughput performance provided in the National B portion can be much better than is indicated by the worst-case values specified in Recommendation X.135.

TABLE 1

Measured national A portion delay and throughput capacity values

Statistic	Measured national A value			
	Minimum	Mean	95th Percentile	Maximum
Call set-up delay (ms)	388	450	517	588
Data packet transfer delay (ms)	147	169	193	203
Throughput capacity (bit/s)	–	6287	–	–
Clear indication delay (ms)	85	107	142	180

Note 1 – The measurements summarized in this table were conducted in January 1987. All reported values are based on measurements of at least 5 different 3-hop paths within the DATEX-P network. Each reported delay value is an average of at least 100 indvidual measurements, including at least 20 measurements on each path. The reported throughput capacity value is an average of 40 individual measurements, each involving the transfer of at least 450 packets.

Note 2 – The data packet transfer delay and throughput capacity values were measured using data packets having a 128-octet user data field. In the throughput capacity measurements, the signalling rate on the access circuit sections was 9600 bit/s; the packet layer window size on the access circuit sections was 2; and the network internal packet layer window size was 4. (The network internal window is a network specific throughput class implementation in which higher negotiated throughput classes result in larger network internal window.)

Note 3 – The clear indication delay values were estimated by measuring the time between transmission of a clear indication packet and receipt of the corresponding clear confirmation packet at the clearing DSE, and dividing the result by 2. Clear confirmation has end-to-end significance in the DATEX-P network.

Note 4 – The reported delay values do not include delays in the access circuit sections or the DTEs.

TABLE 2

Measured national B portion delay and throughput capacity values

Statistic		Measured nation B values			
		Minimum	Mean	95th percentile	Maximum
Call set-up delay (ms)		1040	1089	1126	1197
Data packet transfer delay (ms)		471	495	531	537
Throughput capacity (bit/s)					
Network internal window size	4	–	4127	–	–
	7	–	5350	–	–
	15	–	8595	–	–
Clear indication delay (ms)		406	432	455	468

Note 1 – The measurements summarized in this table were conducted in January 1987. All reported values are based on measurements of at least 5 different 3-hop paths (including 1 satellite-hop) within the DATEX-P network. Each reported delay value is an average of at least 100 individual measurements, including at least 20 measurements on each path.

Each reported throughput capacity value is an average of at least 40 individual measurements, each involving the transfer of at least 450 packets.

Note 2 – The data packet transfer delay values were measured using data packets having a 128-octet user data field.

In each measurement, the signalling rate on the access circuit sections was 9600 bit/s and the packet layer window size on the access circuit section was 2.

Note 3 – The clear indication delay values were estimated by measuring the time between transmission of a clear indication packet and receipt of the corresponding clear confirmation packet at the clearing DSE, and dividing the result by 2. Clear confirmation has end-to-end significance in the DATEX-P network.

Note 4 – The reported delay values do not include delays in the access circuit sections or the DTEs.

Note 5 – The measured values demonstrate that the packet layer network internal window size can strongly influence the throughput capacity of virtual connection portions that contain a satellite circuit.

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