INTERNATIONAL TELECOMMUNICATION UNION

ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

E.301

(03/93)

TELEPHONE NETWORK AND ISDN

OPERATION, NUMBERING, ROUTING AND MOBILE SERVICE

IMPACT OF NON-VOICE APPLICATIONS ON THE TELEPHONE NETWORK

FOREWORD

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

The World Telecommunication Standardization Conference (WTSC), which meets every four years, established the topics for study by the ITU-T Study Groups which, in their turn, produce Recommendations on these topics.

ITU-T Recommendation E.301 was revised by the ITU-T Study Group II (1988-1993) and was approved by the WTSC (Helsinki, March 1-12, 1993).

NOTES

1 As a consequence of a reform process within the International Telecommunication Union (ITU), the CCITT ceased to exist as of 28 February 1993. In its place, the ITU Telecommunication Standardization Sector (ITU-T) was created as of 1 March 1993. Similarly, in this reform process, the CCIR and the IFRB have been replaced by the Radiocommunication Sector.

In order not to delay publication of this Recommendation, no change has been made in the text to references containing the acronyms "CCITT, CCIR or IFRB" or their associated entities such as Plenary Assembly, Secretariat, etc. Future editions of this Recommendation will contain the proper terminology related to the new ITU structure.

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

CONTENTS

Page

1 Introduction 1

2 Scope 1

3 Related Recommendations 2

4 Category 1 applications 2

- 4.1 Voice 2
- 4.2 Non-voice 2
- 4.3 Mixed 3

5 Signalling and transmission considerations for Category 1 applications 3

- 5.1 Signal interference 3
- 5.2 Transmission 3
- 5.3 Potential solutions 3

6 DCME for Category 1 applications 4

- 6.1 Speech quality considerations 4
- 6.2 Impact of DLC on blocking performance 5
- 6.3 Other considerations 5
- 6.4 Tandem DCME connections 6

7 Category 2 applications 6

8 Signalling, dialling and routing considerations for Category 2 applications 7

- 8.1 Signalling considerations 7
- 8.2 Dialling and numbering considerations 7
- 8.3 Routing considerations 8

9 Network architecture considerations 8

- 9.1 Overlay networks 8
- 9.2 Integrated networks 8

10 Recommendation history 8

Annex A – Teletraffic characteristics of non-voice traffic 8

- A.1 Mean call duration 8
- A.2 24-hour traffic profile 8

Annex B 11

Annex C 12

Recommendation E.301

IMPACT OF NON-VOICE APPLICATIONS ON THE TELEPHONE NETWORK

(Modified at Helsinki, 1993)

1 Introduction

recognizing

that true Integrated Service Digital Networks (ISDNs) as described in the I-Series Recommendations, will evolve from Public Switched Telephone Networks (PSTNs);

that this evolutionary process is already well under way in that digital capabilities are being introduced in PSTNs all over the world, in place of analogue facilities;

that although the realization of true ISDNs worldwide is unlikely to occur for many years, the introduction of digital capabilities nevertheless provides the opportunity for Administrations to improve the quality of existing services and simultaneously also offer new "ISDN-like" services; and

that these same digital capabilities may also introduce network elements that may adversely impact service quality in ways not yet fully understood;

this Recommendation provides an analysis of some of the problems which may be encountered in the existing telephone network during the PSTN to ISDN transition period. Further, this Recommendation also described network control and routing procedures that can be used to provide high levels of service quality for all services until a comprehensive ISDN can be achieved.

2 Scope

From a network capabilities point of view, it is convenient to separate the various types of traffic that PSTNs may be expected to carry into the following two broad categories.

Category 1 applications are those traffic streams that can be carried over analogue networks but will henceforth be carried on digital facilities in order to realize the quality improvements inherent in digital transmission. Examples of such applications include data and facsimile with further details contained in section 4.

Category 2 applications are primarily those traffic streams that inherently require digital facilities and in addition may require further network actions to accommodate requirements peculiar to the particular traffic stream. Examples of such applications include switched digital services with fuller details given in clause 7. However, there may be cases where

3 Related Recommendations

The following Recommendations cover related topics in the evolution of PSTNs towards ISDN:

- Rec. E.164 *Numbering plan for the ISDN era*;
- Rec. E.171 *International telephone routing plan*;
- Rec. E.172 *ISDN routing plan*;
- Rec. G.721 *32 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM)*;
- Rec. G.726 *Adaptive pulse code modulation for 40, 32, 24, 16 kbit/s(ADPCM)*;
- Rec. G.727 *5-, 4-, 3-, 2- bits sample, embedded ADPCM*;
- Rec. G.728 *16 kbit/s low-delay CELP (LD-CELP) speech coding algorithm*;
- Rec. G.763 *Digital circuit multiplication equipment using 32 kbit/s ADPCM and digital speech interpolation*;
- Rec. G.766 *Facsimile demodulation and remodulation in DCME*;
- Rec. P.84 *Subjective listening test method for evaluating digital circuit multiplication and packetized voice systems*.

4 Category 1 applications

In this clause we consider three types of Category 1 applications classified as the following.

4.1 Voice

Voice traffic is mainly composed of speech conversation. Because of the rapid deployment of Digital Circuit Multiplication Equipment (DCME) which utilizes speech interpolation and compression to efficiently utilize digital facilities, increasing volumes of non-voice traffic can cause problems that can affect the performance of voice traffic.

4.2 Non-voice

Voice-band traffic which is not speech. Example applications for which the present telephone network is capable of providing bearer services include:

- data (analogue coded);
- facsimile;
- scrambled speech;

b) non-voice traffic often has a 24-hour traffic profile different from voice traffic; this is especially true for international routes, where time zone differences result in the peaks of voice and non-voice traffic occurring at different times (see Annex A for some typical traffic profiles);

c) mean call holding times are often significantly shorter than voice traffic.

4.3 Mixed

Traffic which has both voice and non-voice transmission. An example of one such service would be videotelephony.

5 Signalling and transmission considerations for Category 1 applications

The increasing presence of non-voice traffic in Category 1 applications may result in signalling and transmission problems.

5.1 Signal interference

Non-voice service signals can interfere with telephone circuit signalling systems and vice versa.

Data or facsimile signals can interfere with signalling systems which use in-band line signalling such as Signalling Systems No. 4, No. 5 and R1. Thus such non-voice calls should use the standardized systems set out in the Series-V and -T Recommendations since these are designed to prevent interference with the standard signalling systems, either by avoiding the particular signalling frequencies or by operating the guard circuit of the signalling receiver.

Despite the safeguards mentioned above, it may sometimes happen that the signalling receiver is momentarily operated by the carried service signal. In this case the splitting device in the signalling receiver will operate and cause a short discontinuity in the received service signal.

5.2 Transmission

5.2.1 Interference to transmission systems

If the proportion of non-voice calls is large, it can increase the overall power loading in a transmission assembly (group or supergroup). This can cause distortion in the group of signals and/or the operation of power limiters which can adversely affect other calls or services in the same transmission assembly.

5.2.2 Interference by transmission systems

It may be the case that ordinary speech channels do not provide an adequate transmission path for some types of

5.3.1 It should be established for each bilateral relationship what commercial and regulatory arrangements exist which recognize the need to provide for non-voice services within prescribed Quality of Service parameters.

5.3.2 If it is decided by the Administration concerned that certain services must be supported, then two approaches can be taken:

a) only transmission systems allowing suitable performance for non-voice services are used;

b) separate routings are established for the whole or part of the networks, where unsuitable transmission would otherwise occur.

5.3.3 In case b) above, it is necessary to know when subscribers are initiating non-voice calls. There are two methods for achieving this:

i) the subscriber line is known to be one originating only non-voice calls, e.g. it is a facsimile terminal;

ii) the subscriber sends some form of service indication to the network, identifying a non-voice call request (e.g. Recommendation E.131).

If these indications are directly available at the exchange where the separate routing is selected, then path selection need only combine this indication with the dialled digits. In other cases it is necessary for a suitable signalling system to be employed to carry this indication forward to the special selection point. This may be done using signalling systems including special call categories. In particular, a call category "data call" is provided in Signalling Systems R2, No. 6 and No. 7, also No. 5 by bilateral agreement. The separate routing may be continued throughout the network using either "path of entry" indications at the exchanges concerned or the special call category signals within the signalling system (see Recommendation E.172).

6 DCME for Category 1 applications

As mentioned in 4.1, in order to economize on the provision of international voice channels, international digital transmission systems are increasingly being deployed with speech interpolation systems such as DCME. Information on speech interpolation systems can be found in Supplement 2 of Fascicle VI.1. Circuit gains are realized by speech compression and by exploiting the silent period normally existing during speech conversations. DCME equipment uses Low Rate Encoding (LRE) for non-voice signals and a combination of LRE and Variable Bit Rate Encoding (VBR) for speech signals (see Recommendation P.84, *Blue Book* for a definition of LRE and VBR). In the absence of VBD traffic, circuit gains of 4:1 are achievable. Annex B shows a functional view of a typical DCME configuration.

6.1 Speech quality considerations

Continuous non-voice service signals will cause the continuous operation of the speech detectors and give rise to permanent association of the telephone circuit to the transmission channel. This in turn increases the probability of

6.1.2 The DLC capability should be provided at both ends of the DCME system. When a DLC signal is sent to the switch, the selection of trunk circuits associated with the specified DCME should be skipped immediately.

6.1.3 Wherever possible, Administrations should avoid the connection of one particular circuit sub-group to more than one DCME. This is necessary to avoid controlling too much traffic. If one circuit sub-group is assigned to more than one DCME, then even if only one DCME is in an overload condition, all traffic in that circuit sub-group is controlled and the blocking may therefore increase.

Also, if the selection scheme of trunk circuits with DCME is simply the ascending or descending order, traffic will be concentrated into one particular DCME. This may be a cause of speech clipping and degradation in time of overload as the ABS decreases, especially at the non-voice peak hours. Thus, as a general principle, Administrations should balance the load on all DCMEs connected to a single circuit group. The circuit selection scheme described in Figure C.1, can be considered to decrease the probability of overloading a particular DCME unit. Thus, the selection order for the circuit sub-groups is 1A-1B-2A-2B-3A-3B-4A-4B. This order may be implemented as a hunt group or, if the technology permits, on a call-by-call basis. Balanced loading is even more important if the DLC capability cannot be provided since this may be the only method available to maintain service quality.

6.2 Impact of DLC on blocking performance

In periods of high non-voice activity, there is the possibility of high blocking as a result of DLC. Annex D shows the blocking, as a function of the per cent of VBD traffic, on a group of 120 circuits, with the total load fixed at 103 Erlangs. The triplet (T, w, s) in Figure D.1 represents a combination of the maximum allowable number of VBD calls, the ABS threshold for DLC activation and the speech activity factor respectively. Two values, 37 and 57 for T are considered. The range of consideration for w is 3.3 to 3.7 and for s is 0.35 to 0.40.

6.2.1 The Administrations concerned should observe the operating condition of each DCME system and measure the ratio of non-voice traffic on each route on a regular basis. Network planning practices should consider the quantity of non-voice traffic present within the international network. The proper observation and maintenance of DCME operation can help optimize the Quality of Service provided within the international network.

6.2.2 For traffic streams containing a high degree of VBD traffic, Administrations should consider operating the DCME at lower compression ratios than with traffic streams with primarily voice traffic.

6.3 Other considerations

6.3.1 Given the high growth rate of facsimile traffic, expected compression ratios may not be achievable. This difficulty may be alleviated by the use of facsimile demodulation technology. This technology permits, for a given number of derived circuits, the accommodation of a higher percentage of facsimile traffic than would be possible otherwise. With appropriate dimensioning of the DCME, the result will be more consistent quality of speech.

6.3.4 It is recommended that Administrations consider the volume of voice and non-voice traffic when establishing the compression ratio. As a result, there is a possibility that the peak time of required telephone circuits and bearer circuits may appear at different hours. Therefore the number of required telephone circuits with speech interpolation systems and bearer circuits need to be dimensioned considering the 24-hour traffic profiles of both voice and non-voice.

6.3.5 Administrations should note that for DCME circuit groups (or circuit sub-groups) with DLC, high blocking may not necessarily imply lack of derived circuits. The problem could be insufficient bearer capacity. Administrations should ensure that remedial measures are directed at the appropriate network elements. Administrations are further cautioned that traditional relationships between blocking call attempts and observed traffic may not apply.

6.4 Tandem DCME connections

Tandem DCME connections occur when two links in an international connection are both equipped with DCME. DCME is used extensively in the international networks, especially on long haul routes where the circuit efficiency makes its use economically attractive. It is likely that penetration of DCME will continue to increase, and hence the likelihood of switched transit calls involving tandem connection of DCMEs will also increase. This could result in tandem encoding/decoding of voice and non-voice traffic. Although the impact on quantizing distortion units (QDU's) of such connections requires further study, present estimates are that if Recommendation G.721 coding is used, the international portion of tandem connections could result in an accumulated QDU allocation of 8, which is considered unacceptable. Non-voice applications such as facsimile are known to be adversely affected by such tandem encodings. Whether voice is similarly affected from a customer perception point of view is a matter for further study. If a tandem connection is anticipated, the Administrations concerned should take the following network control procedures to maintain an acceptable Quality of Service for both voice and non-voice traffic:

– select a circuit sub-group derived by DCME equipped with facsimile demodulation capability. Although this option does not alleviate the accumulation of QDU's, because of reduced load on the bearer, some improvement in speech quality and blocking call attempts is anticipated;

– select a circuit sub-group which is normally derived by DCME but ADPCM/DSI functions inactivated;

– select a non-DCME derived circuit sub-group.

Annex E provides further clarification of these options.

Tandem DCME connections could also result in tandem signal detection of voice and non-voice traffic. This is expected to have an adverse impact on voice traffic because signal levels vary during a conversation.

In addition, if the tandem DCME units are engineered with different signal activity detection thresholds and/or if each portion of the link is individually optimized for signal detection performance, then further degradation of speech quality in the form of clipping and signal loss can result.

Thus from a Quality of Service viewpoint, Administrations establishing routing arrangements that result in tandem DCME connections should carefully negotiate signal detection performance issues.

It should be noted that in some cases no formal service definition exists in CCITT Recommendations, and will not be developed. This is because these services will vary from network to network since different PSTNs will have different capabilities. The introduction of such services internationally will be by bilateral agreement between Administrations.

To provide these and other Category 2 services, the PSTN, in addition to having digital transmission facilities, may need to perform additional functions so that these services are feasible. Examples of such functions are:

i) ensuring that only compatible digital circuits must be selected, e.g. all circuits use transparent 64 kbit/s transmission;

ii) disable or bypass all digital speech processing systems (e.g. DCME, DSI, echo control equipment, other compression systems) in the data transmission phase;

iii) disable or bypass any μ-law to A-law converters in the data transmission phase;

iv) disable or bypass all echo suppressors or cancellers in the data transmission phase;

v) prevent the use of digital transmission attenuation pads;

vi) permit the use either in-band or out-of-band network and access signalling.

As mentioned in clause 2, these functions may be used to handle some Category 1 applications also. For example, in a switched transit arrangement, the originating Administration may provide for transit calls to bypass any DCME circuit groups to the transit Administration in order to avoid the problems with tandem DCME connections discussed in 6.4.

Details for these arrangements are for further study. In order that these arrangements may be provided from the originating network to the destination network, the signalling system applied should have the capability to convey such non-voice service requests; for example, in the case of the Telephone User Part (TUP) of Signalling System No. 7, at least such an additional function must be implemented among Administrations concerned in order to convey the customer request for "unrestricted bearer capability" to the transit and destination networks. It should be also noted that terminal compatibility cannot be negotiated between the originating terminal and destination terminal within the capability of TUP. In this case, therefore, the subscriber can only communicate with the destination number which, he knows in advance, is accommodating a compatible non-voice terminal.

8 Signalling, dialling and routing considerations for Category 2 applications

This clause identifies some signalling and routing techniques which can be utilized for providing the types of services identified in clause 7.

8.1 Signalling considerations

8.3 Routing considerations

From a routing point of view, it is necessary for a switching node to know what kind of service is requested in order to select appropriate network facilities. In an ISDN this is achieved by examining various parameters and indicators in the signalling system (see Recommendation E.172). In the absence of these indicators, other techniques can be used. For example, customers subscribing to the services described could be provided with a dedicated digital access link to a network node which supports the service. Appropriate route selection could then be made based on analysis of the path of entry and dialled numbers.

9 Network architecture considerations

The following subclauses apply to those situations where the switching nodes in the PSTN are shared by both Category 1 and 2 services.

To provide the necessary functionalities, Administrations may use two different types of network architecture.

9.1 Overlay networks

In this arrangement, network transmission facilities meeting the above requirements may be dedicated for the exclusive use of these services. The advantage of this approach is that the required facilities are already provisioned and the switching node need only match up the requested capability with already existing facilities. The disadvantage is that these facilities may not be suitable for Category 1 applications. For example, a circuit with echo cancellers or suppressors permanently disabled may not be suitable for voice. Thus, there is a loss of efficiency in this inability to share capacity.

9.2 Integrated networks

In this arrangement, all facilities are capable of handling all types of services. We now gain the efficiency of pooled resources. However, the switching node and the appropriate network elements must now set up the required bearer capabilities on a call-by-call basis. The processing overhead thus increases. The current trend is for Administrations to begin with overlay networks and migrate towards shared networks as switching technology advances in functionality.

10 Recommendation history

- First published 1988 (*Blue Book*).
- Revised 1992.

Annex A

Teletraffic characteristics of non-voice traffic

(This annex forms an integral part of this Recommendation)

Voice communication is only possible when calling and called parties are present at both ends and therefore, generally align with the schedule of human activities. Thus, peak hours of voice and non-voice traffic may differ. In Figure A.2, countries A and B have similar peak hours for both traffic streams while country C has two peaks, one (earlier) for voice and the other for non-voice. This can contribute to flattening the traffic profile thus making more efficient use of the circuit group. It should also be noted that non-voice traffic may sharpen, the peak of the profile in case of short overlapping of business hours between two countries. This may affect the dimensioning of the network and require additional circuits to cover only a short period of time.

It is therefore important that countries measure and understand the traffic on their routes so that efficient dimensioning of the network can be undertaken.

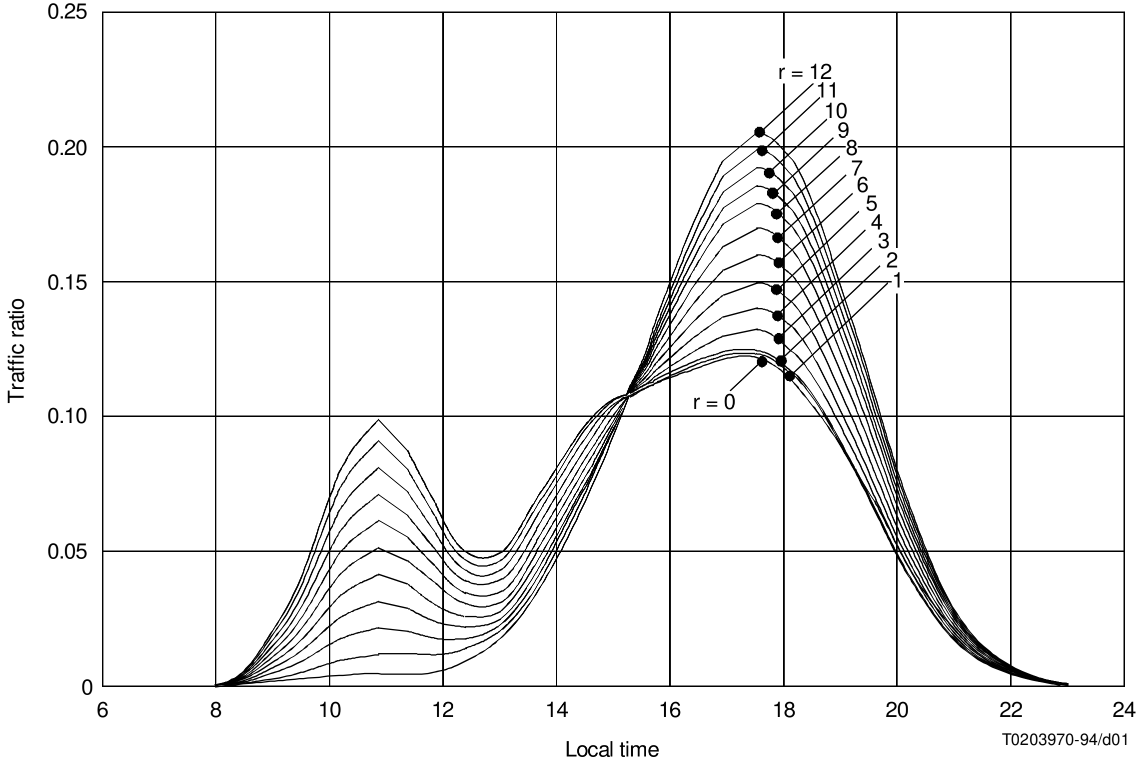
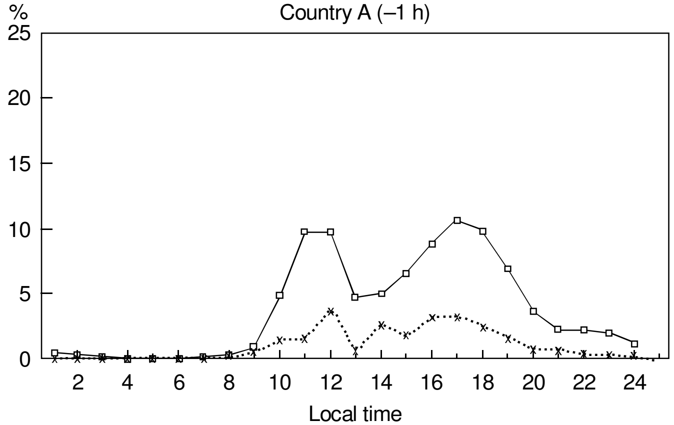


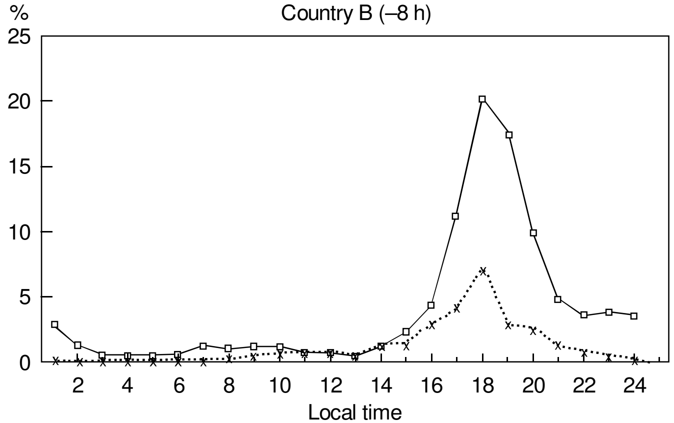
FIGURE A.1/E.301

24-hour traffic profile for telex and record-type telecommunication services (calculated)

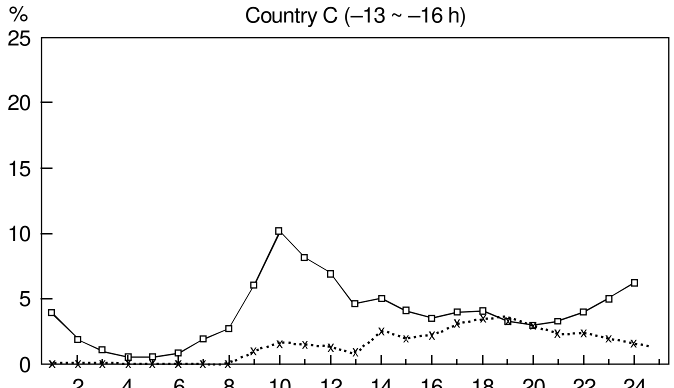
Country A (–1 h)



Country B (–8 h)



Country C (–13 ~ –16 h)



Annex B

(This annex forms an integral part of this Recommendation)

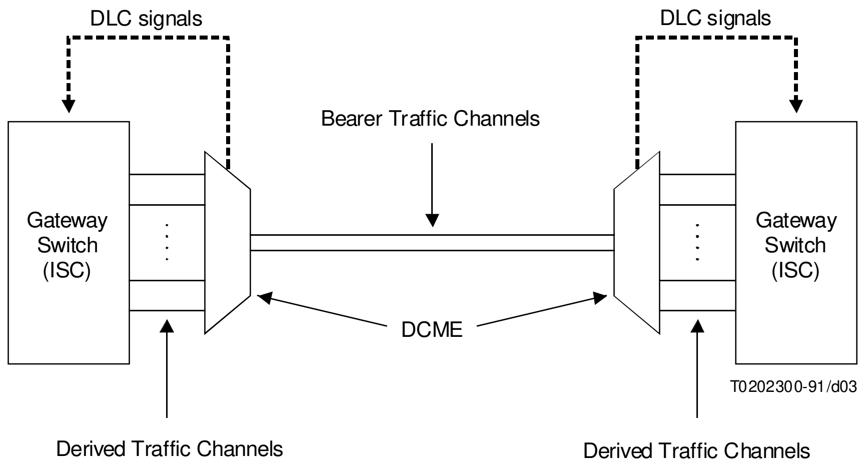


FIGURE B.1/E.301

A functional view of a typical DCME configuration

Annex C

(This annex forms an integral part of this Recommendation)

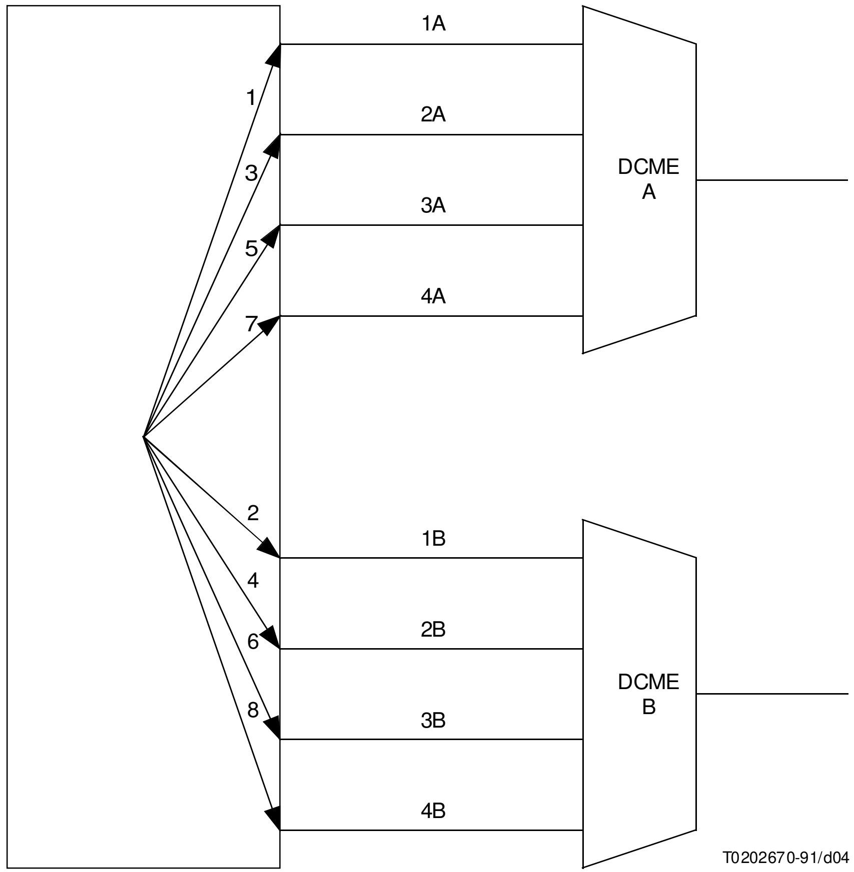


FIGURE C.1/E.301

Circuit selection scheme

Annex D

(This annex forms an integral part of this Recommendation)

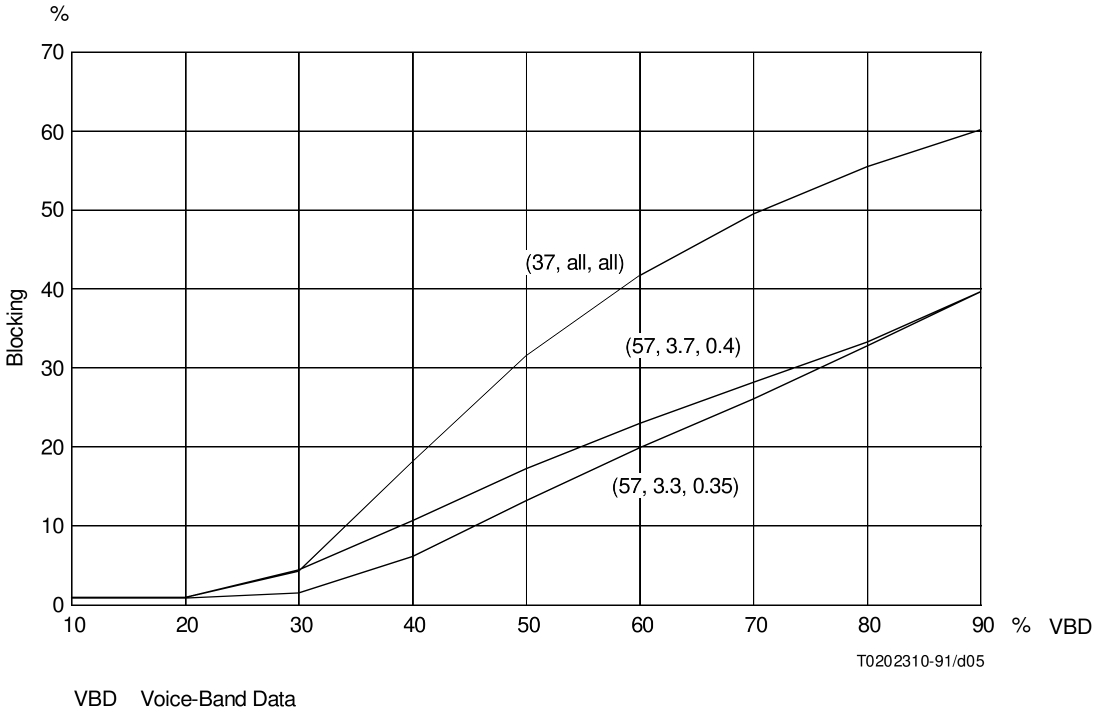


FIGURE D.1/E.301

Effect on VBD component on blocking

Trunks: 120

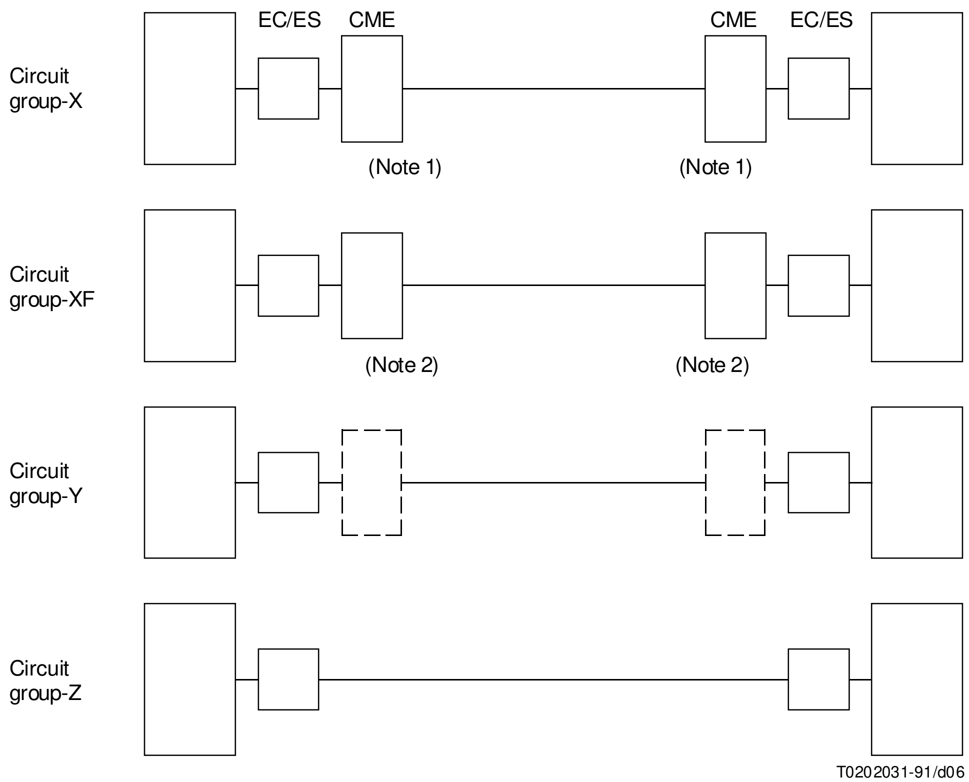
Total Load: 103 Erlangs

Voice Holding Time: 300 s

VBD Holding Time: 100 s

Annex E

(This annex forms an integral part of this Recommendation)



NOTES

1 DCME equipped with facsimile demodulation.

2 Changeable to 64 kbit/s clear circuit by inactivating ADPCM/DSI functions on call-by-call basis.

FIGURE E.1/E.301

Circuit groups to be provided and its combination

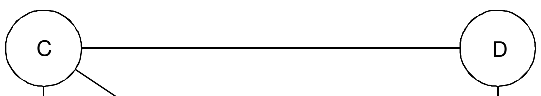


TABLE E.1/E.301

Examples of possible circuit groups

Anticipated connection	Selected circuit group at originating ISC A	Possible circuit group at transit ISC C	Possible circuit group at transit ISC D
A-B	X, XF, Y or Z		
A-C-B	X	XF, Y or Z	
	XF	X, XF, Y or Z	
	Y		
	Z		
A-C-D-B	X	XF, Y or Z	XF, Y or Z
	XF	X	XF, Y or Z
		XF	X, XF, Y or Z
		Y	X, XF, Y or Z
		Z	X, XF, Y or Z
	Y	X	XF, Y or Z
		XF	X, XF, Y or Z
		Y	X, XF, Y or Z
		Z	X, XF, Y or Z
	Z	X	XF, Y or Z
		XF	X, XF, Y or Z
		Y	X, XF, Y or Z
		Z	X, XF, Y or Z
NOTE – Selection order of circuit groups is subject to the agreement among Administrations concerned.			