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**TELEPHONE NETWORK AND ISDN**

**QUALITY OF SERVICE, NETWORK MANAGEMENT AND TRAFFIC ENGINEERING**

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# DIMENSIONING OF DIGITAL CIRCUIT MULTIPLICATION EQUIPMENT (DCME) SYSTEMS

## ITU-T Recommendation E.528

(Previously "CCITT Recommendation")

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## FOREWORD

The ITU-T (Telecommunication Standardization Sector) is a permanent organ of the International Telecommunication Union (ITU). 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, establishes the topics for study by the ITU-T Study Groups which, in their turn, produce Recommendations on these topics.

The approval of Recommendations by the Members of the ITU-T is covered by the procedure laid down in WTSC Resolution No. 1 (Helsinki, March 1-12, 1993).

ITU-T Recommendation E.528 was prepared by ITU-T Study Group 2 (1993-1996) and was approved under the WTSC Resolution No. 1 procedure on the 19th of February 1996.

NOTE

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

# CONTENTS

## SUMMARY

Digital Circuit Multiplication Equipment (DCME) allows a high degree of utilization of digital circuits in Public Switched Telephone Networks (PSTN) and Integrated Services Digital Networks (ISDN) providing integration of traffic containing voice, Voice Band Data (VBD) and the facsimile portion of Voice Band Data (VBD) traffic.

Dimensioning method for DCME system, described in this Recommendation, deals with a group of trunks (circuit group, as defined in Recommendation E.600) between two switches, where DCMEs are used to achieve a statistical multiplexing gain.

This Recommendation also describes the GOS parameters that play a part in the DCME gain calculation.

This Recommendation deals with a point-to-point DCME single clique operation configuration. Dimensioning of complex multiclique/multidestination types of DCME systems is for further study.

# DIMENSIONING OF DIGITAL CIRCUIT MULTIPLICATION EQUIPMENT (DCME) SYSTEMS

*(Geneva, 1996)*

## 1 Scope

Digital Circuit Multiplication Equipment (DCME) allows a high degree of utilization of digital circuits in Public Switched Telephone Networks (PSTN) and Integrated Services Digital Networks (ISDN) providing integration of traffic containing voice, Voice Band Data (VBD) and the facsimile portion of Voice Band Data (VBD) traffic.

Dimensioning method for DCME system, described in this Recommendation, deals with a group of trunks (circuit group, as defined in Recommendation E.600) between two switches, where DCMEs are used to achieve a statistical multiplexing gain. The purpose of the dimensioning method is to determine the size of the group of trunks, the number and properties of the DCME output (bearer) channels and the number of DCMEs for specific traffic mixes – voice, VBD and facsimile to provide a satisfactory Grade of Service (GOS) cost-effectively.

Further, this Recommendation also describes the GOS parameters that play a part in the DCME gain calculation. The DCME gain is described in this Recommendation as a ratio of a number of trunk channels to a number of available bearer channels and is used to determine availability and specific properties of the DCME bearer channels.

This Recommendation deals with a point-to-point DCME single clique operation configuration. Dimensioning of complex multiclique/multidestination types of DCME systems is for further study.

## 2 References

The following Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision: all users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published.

- CCITT Recommendation G.766 (1992), *Facsimile demodulation/remodulation for digital circuit multiplication equipment*.
- ITU-T Recommendation G.763 (1994), *Digital circuit multiplication equipment using ADPCM (Recommendation G.726) and digital speech interpolation*.
- CCITT Recommendation G.726 (1990), *40/32/24/16 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM)*.
- ITU-T Recommendation E.301 (1993), *Impact of non-voice applications on the telephone network*.
- CCITT Recommendation E.520 (1988), *Number of circuits to be provided in automatic and/or semi-automatic operation, without overflow facilities*.
- CCITT Recommendation E.721 (1991), *Network grade of service parameters and target values for circuit-switched services in the evolving ISDN*.
- ITU-T Recommendation P.84 (1993), *Subjective listening test method for evaluating digital circuit multiplication and packeted voice systems*.
- ITU-T Recommendation Q.50 (1993), *Signalling between Circuit Multiplication Equipment (CME) and International Switching Centres (ISC)*.

## 3 Definitions

The following terms and definitions applicable to DCME are described in clause 2/G.763. Definitions relating to DCME:

These definitions are used throughout the texts of this Recommendation.

**3.1** Average Encoding Rate – Average number of Bits per Sample (bit/sample) (ABS)

**3.2** Bearer Channel (BC)

**3.3** Trunk Channel (TC)

**3.4** Clique

**3.5** DCME gain

**3.6** Digital Speech Interpolation (DSI) gain

**3.7** Facsimile demodulation/remodulation

**3.8** Freezeout

**3.9** Freezeout fraction

**3.10** Low Rate Encoding (LRE) gain

**3.11** Speech Activity Factor (AF)

**3.12** Variable Bit Rate (VBR)

The following definitions are specific to DCME dimensioning as described in this Recommendation.

**3.13** **clipping**: Clipping is an alteration of the front-end of a speech burst due to lack of sufficient time for processing. Occurs in most cases as a result of speech detectors processing functions employed by DSI systems when detectors try to process a front of a speech burst but lack the sufficient time to operate. Competitive clipping is the impairment caused by the overload control strategy which allows freezeout to occur when bearer channels are temporarily unavailable.

**3.14** **DCME gain for voice $G_V$**: $G_V$ is the product of Low Rate Encoding (LRE) and Digital Speech Interpolation (DSI) gain factors.

**3.15** **DCME link**: A number of bearer channels between two DCMEs transmitting concentrated information in each direction.

**3.16** **nibble**: Nibble is the term which refers to a 4-bit/sample-coded channel and corresponds to a data rate of 32 kbit/s. It is always associated with the capacity of a bearer channel. The term full bearer channel refers to a channel consisting of 61 nibbles, with the exception of the Time Slot zero and a single nibble used for Assignment Message Channel (AMC).

**3.17** **time slot (TS)**: Time Slot is the term which refers to an 8-bit/sample-coded channel and corresponds to data rate of 64 kbit/s. It is also used to refer to a trunk channel.

**3.18** **point-to-point mode**: A DCME operational mode in which the traffic is exchanged between two corresponding DCMEs and the trunk channel traffic is interpolated over available bearer channels for only one destination in each direction.

## 4 Abbreviations

For the purposes of this Recommendation, the following abbreviations are used:

| | |
|---|---|
| ABS | Average Bits per Sample |
| ADPCM | Adaptive Differential Pulse Code Modulation |
| AF | Activity Factor |

| | |
|---|---|
| CME | Circuit Multiplication Equipment |
| DCME | Digital Circuit Multiplication Equipment |
| DLC | Dynamic Load Control |
| DSI | Digital Speech Interpolation |
| FEC | Forward Error Correction |
| G3 | Group 3 FAX |
| GOS | Grade of Service |
| $G_v$ | Gain for Voice |
| ISDN | Integrated Services Digital Network |
| LRE | Low Rate Encoding |
| PSTN | Public Switched Telephone Network |
| QOS | Quality of Service |
| VBD | Voice Band Data |

# 5 Dimensioning elements

## 5.1 Determining Quality of Service (QOS) and Grade of Service (GOS) parameters

The traffic offered to networks containing DCME is determined in accordance with methods specified in the E.500-Series of Recommendations. Determination of a DCME gain, as a prerequisite for DCME dimensioning, for the overall composite traffic load requires distinction (in accordance with Recommendation E.301) among and between voice, VBD (non-facsimile and facsimile). Determination of GOS and QOS should provide an initial input for dimensioning of DCME systems; in the ISDN case Recommendation E.721 should provide the necessary guidance in respect to GOS parameters. The following parameters are considered for dimensioning of DCME systems:

– QOS:
  - level of the front-end clipping;
  - a tolerated number of bit per sample used for speech encoding.

– GOS:
  - call blocking probability.

## 5.2 General principles of dimensioning method

The following principles are observed in the DCME dimensioning method described in this Recommendation:

- A single link of point-to-point DCME is dimensioned by calculating DCME gain for voice, the number of necessary nibbles (size of bearer channel) and DCME gain for assumed trunk capacity and traffic mix.
- Then, the necessary number of DCME links for the given trunk group is determined, to meet the required call blocking probability objectives.

## 5.3 Parameters relevant to dimensioning calculations

### 5.3.1 General parameters

- A number of trunk channels versus a number of bearer channels.
- Trunk channel occupancy and speech activity. It is usual to assume a 35% to 40% of speech activity depending on the type of a language. Outside of the route busy hour occupancy of the trunk channels is lower than in the busy hour. This dictates a reduction of the Speech Activity Factor to about 27% outside the busy hour to allow the maximum of 40% during the busy hour.

- VBD traffic. A factor to be considered is a percentage of the VBD traffic that may vary depending upon the route and time of day.
- Ratio of half-to-full duplex VBD.
- Signalling between DCMEs may hold channels active for significant periods not allowing any interpolation during the signalling processing periods.
- 64 kbit/s clear channel traffic. A significant factor is a required number of 64 kbit/s clear channels, each absorbing two 32 kbit/s bearer channels.
- Minimum acceptable speech quality. It is determined by the encoding rate of the LRE process and by the amount of speech lost while newly active trunk channels are being connected to bearer channels. If a large number of these channels are in competition, the beginning of a speech burst is more likely to be clipped; if relatively few trunk channels are active, a part of a speech burst could be frozen.

#### 5.3.2 ADPCM mode for Low Rate Encoding

The ADPCM method is assumed to be used for coding speech with bit rates less than 64 kbit/s (e.g. the 32 kbit/s). This technique is commonly used in DCME to increase the circuit capacity. The ADPCM increases the channel occupancy by means of encoding at a lower number of bit per sample than the PCM. An ordinary PCM encodes a voice channel at 64 kbit/s using a rate of 8-bit/sample encoding. The ADPCM coding takes into account the nature of a signal, and is capable of reducing the encoding rate:

- 5-bit/sample for VBD signals at 40 kbit/s data rate;
- 4-bit/sample for a regular voice signal at 32 kbit/s data rate;
- 3- or 2-bit/sample for an overload-voice signal, corresponding to 24 kbit/s and 16 kbit/s data rates.

Compressing the signal by the ADPCM in this manner could double the channel capacity.

#### 5.3.3 Digital Speech Interpolation

The DSI method is assumed to be used for speech processing. This technique minimizes inactive periods during a conversation, creating extra channel occupancy. The gain based on the DSI processing is defined as the ratio of a number of voice channels over a number of bearer nibbles (4-bit segments) required to accommodate the voice channels with a given (user determined) level of speech quality. Typically a gain of up to 3:1 (minimum of 2:1) could be produced by the DSI processing for the speech activity of 30% to 40%.

Interpolated voice channels are assigned to available bearer channels. If a bearer capacity is not available when a voice channel becomes active, front-end clipping of the voice spurt occurs. Overload bearer channels are created to reduce the clipping probability during the peak traffic conditions. This in turn requires reduction of the ADPCM encoding rate for voice channels from normal 4-bit/sample to a lower value; typically, greater or equal to 3-bit/sample of 3/4-bit overload condition and greater or equal to 2-bit/sample for 2/3-bit overload condition.

### 5.4 DCME gain determination (in accordance with Recommendation G.766)

#### 5.4.1 DCME gain for voice $G_V$

Equations for $G_V$ calculation provided in Annex B are quoted from Recommendation G.763.

The formulae used for the $G_V$ calculation are empirical; they are the result of more formal mathematical analysis based on the statistical model of DCME that takes into account the Poisson arrival attributed to speech spurts. Values of the coefficients a and b, called curve fitting coefficients, reflect the appropriate AFs for a number of trunks less than 80. A user guide exists on DCME link dimensioning, in the form of a computer program that simulates different bearer pool sizes and different traffic mixes and provides generic calculation of a DCME gain for a variety of cases. Appendix II contains information on how to obtain this guide.

## 6 Method for dimensioning of DCME links and gain computation

Dimensioning the DCME link requires computing the minimum number of bearer channels (BCs) which would accommodate the trunk traffic, while providing the required transmission quality. An appropriate method is provided here to compute the impact of specifically chosen volumes of voice/VBD/data/facsimile calls reflected by the overall DCME gain calculation. Annex B provides computation formulae. Examples of the results are in Annex A. The gain calculation task is broken into the following steps:

a) Determine the preassigned traffic: preassigned 64 kbit/s channels, preassigned 40 kbit/s channels and preassigned 32 kbit/s channels. If preassigned 40 kbit/s channels are selected, bit-banks will be created (one bit-bank equals 4-bit) and the unused bits are assigned to the bit-banks. These bits, then could be used to compensate for the bits deficit often created by dynamically loaded data channels.

b) Determine the percentage of input trunk channels to be used as data channels.

c) Determine the percentage of data trunk channels to be used as facsimile channels.

d) Determine the percentage of these facsimile channels to be assigned as G3 FAX calls (9.6 kbit/s standard facsimile calls).

e) Select the voice Activity Factor (AF) and the encoding rates – the number of bits per sample per channel.

f) From step b), a number of input trunk channels for voice and data channels can be determined.

g) From steps c) and f), a number of facsimile channels in the trunk can be determined.

h) From steps g) and d), a number of standard facsimile channels in the trunk can be determined.

i) From steps e) and f), and applying the formulae as shown in Table A.1, the voice gain can be computed and the number of nibbles in the bearer channel can also be computed.

j) Nibbles computation for data calls, VBD, non-standard facsimile, and standard facsimile calls.

k) Following the nibbles determination in the bearer channel for all fractions of the required traffic, the DCME gain could be calculated as: DCME gain = 2 × number of input trunk channels/(total number of nibbles required in bearer channel + overhead nibbles).

Specific basic principles of DCME operation are summarized in this clause in order to introduce the assumptions used for the gain determination and associated bearer channel capacity computation:

- A data channel of 40 kbit/s requires 5-bit/sample encoding. This is accommodated by allocating a 4-bit bearer nibble and taking the fifth bit from a special 4-bit bank channel. Four data channels can use the same bit bank.
- Each clear channel call (64 kbit/s) requires two 32 kbit/s transmission channels.
- Preassigned channels include data rates of 32 kbit/s, 40 kbit/s and 64 kbit/s. The bit-banks created by the preassigned 40 kbit/s channels can be shared with the interpolated data channels.
- A large amount of data calls all engaged in facsimile transmission and most of the facsimile is transmitted in a half-duplex mode, with only a minimum amount of information sent in the receive direction (acknowledgement signal). The DCME specifications reflect the fact that the activity of the facsimile channel is 100% in the transmit direction and a very small percentage in the receive direction (silence elimination).

## 7 Determination of the number of DCME links

### 7.1 Determination of blocking probabilities for point-to-point DCME links

In the case where the activation of DLC is assumed in a single point-to-point DCME configuration, increase of call blocking probabilities as described in Recommendation E.301 should be taken into account. To estimate the blocking probabilities of this case, the events such as a number of admissible VBD calls and Average Encoding Rate (ABS), a number of bit per sample that trigger the activation of DLC, should also be taken into account. Appendix I gives an example of an analytical method for computing blocking probabilities in cases where DLC is used. Details are for further study.

### 7.2 Multiple system configuration

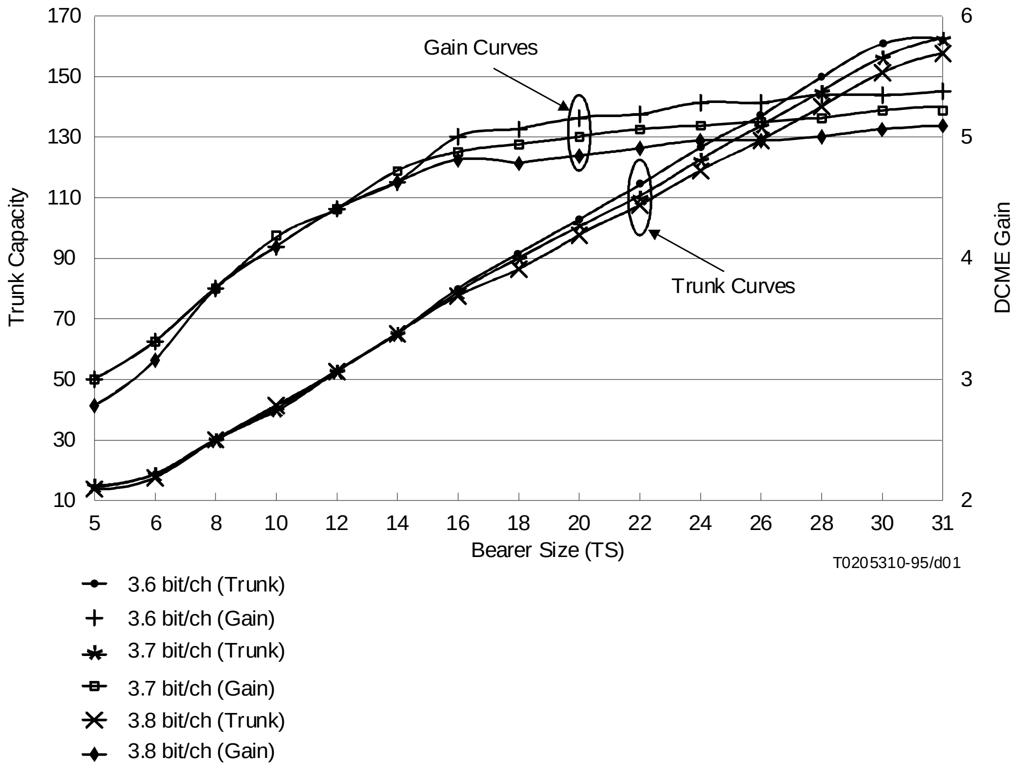
To meet objectives of low call blocking probabilities, multiple DCME links should be deployed in a trunk group. A simple approximation method may be used to determine a number of DCME links in such a way that each link achieves the blocking probability objective independently. In this case, an assumption is made that the offered traffic is randomly split into N homogeneous traffic streams, each of the links is dedicated to one of the N streams and no overflow from one link to another is permitted. Dimensioning methods for the configuration of heterogeneous systems or with a specific trunk selection rule needs further study.

# Annex A

(This annex forms an integral part of this Recommendation)

### A.1 Effects of specifically chosen parameters on DCME dimensioning

Figures A.1 and A.2 show examples of $G_V$ gain curves and corresponding capacities of the bearer channels under specific GOS conditions. In both cases the traffic does not contain VBD channels and in particular the VBD carrying the facsimile channels.



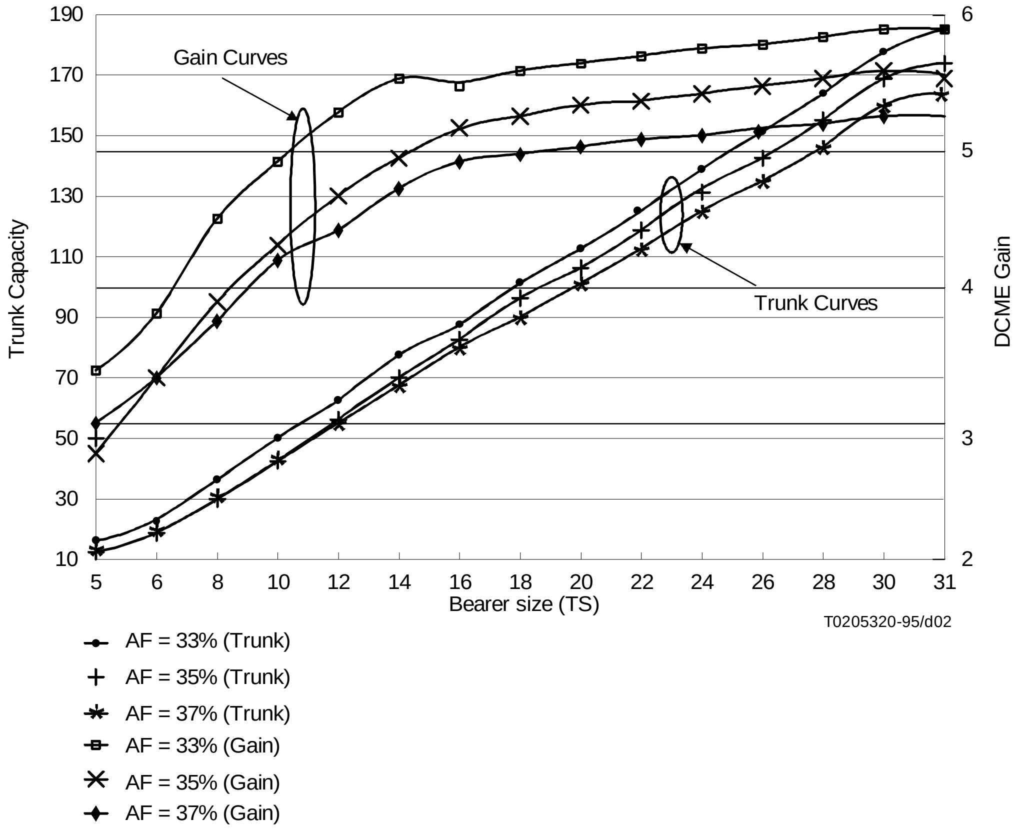

Conditions: No VBD, no facsimile, no demodulation/remodulation, no FEC.
TS = Time Slot (8-bit) = 2 nibbles (4-bit). AF = 37%.

FIGURE A.1/E.528

**Impact of encoding rates – bit/sample per channel – on DCME gain**

Figure A.1 illustrates the impact of the encoding rates per channel (bit/sample per channel) on the overall DCME gain with fixed voice Activity Factor, in this case, AF = 37%. In this example, the maximum gain occurs at the minimum encoding rate of 3.6-bit/sample per channel. The split in the gain value between different sample rates becomes noticeable at a specific bearer channel size, in this example, at a size of 14 Time Slots (TS). It is also shown that as encoding rates increase, the DCME gain decreases.

Figure A.2 shows the impact of a variable voice Activity Factor (AF) on the DCME gain with a constant encoding rate, in this example, 3.7-bit/sample.



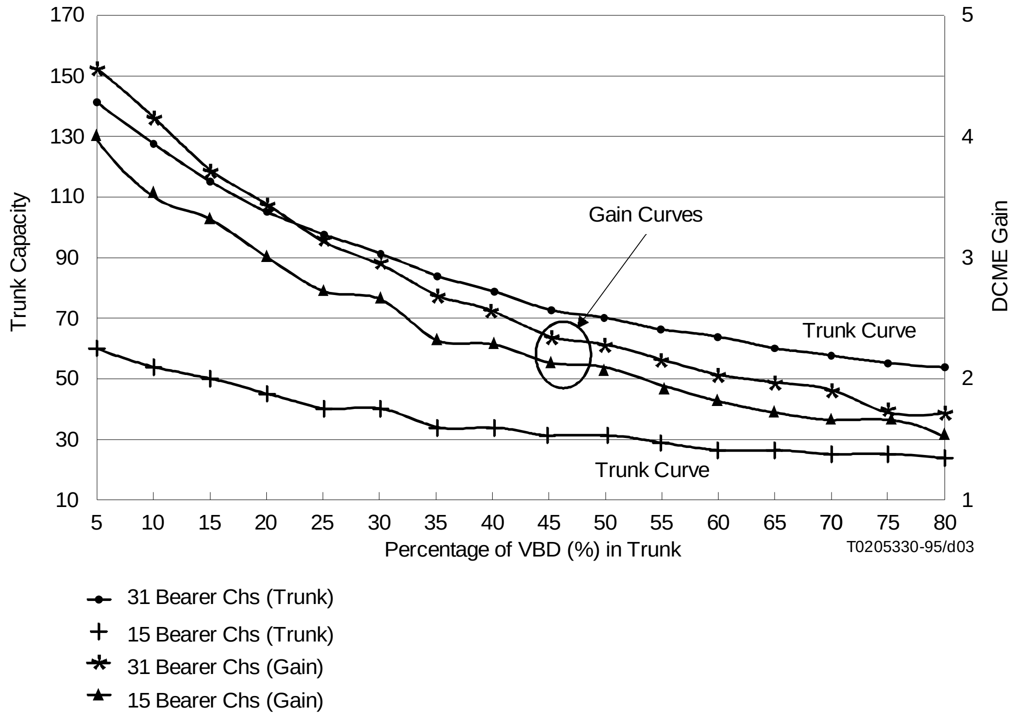

Conditions: No VBD, no facsimile, no demodulation/remodulation, no FEC.
TS = Time Slot (8-bit) = 2 nibbles (4-bit). Encoding rate? bit/sample per channel = 3.7.

FIGURE A.2/E.528

**Impact of voice Activity Factor on DCME gain**

The AF variations could correspond to a potentially higher DCME gain than in the previous example, Figure A.1. The gain for the AF 33% – the lowest in this example – ranges between 3.2 and 5.9 as compared with the Lowest Rate Encoding – 3.6-bit/sample per channel (Figure A.1) – that corresponds to the gain in the range of 3-5.5.

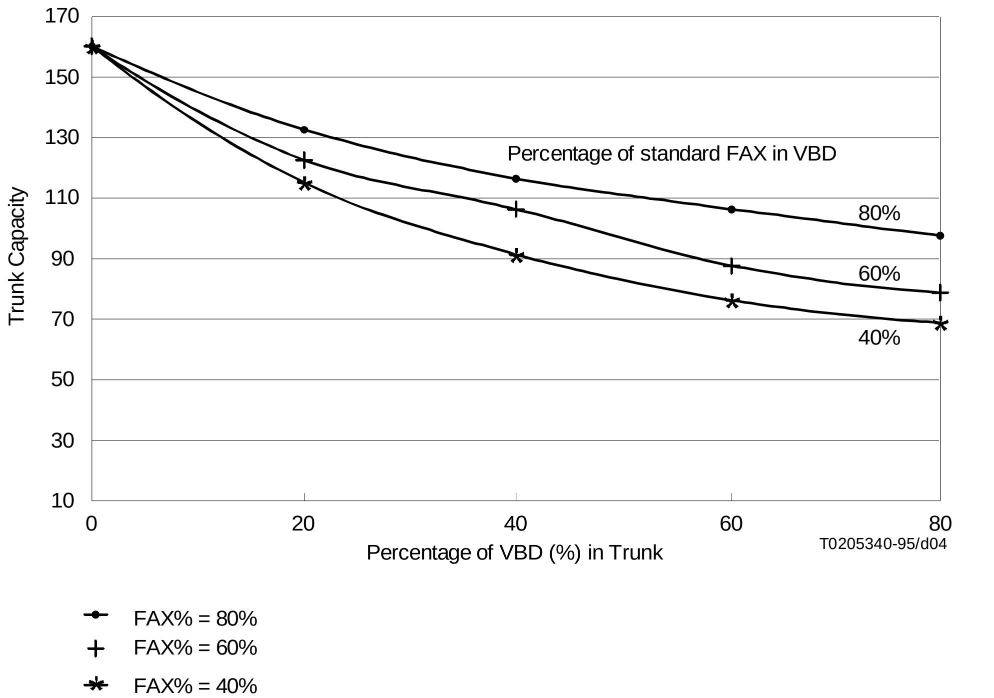
Figures A.3 and A.4 both show the impact of various percentages of the VBD traffic loading at the DCME input (trunks) on the DCME gain.




Conditions: VBD only, no facsimile remodulation/demodulation, FEC is disabled.

FIGURE A.3/E.528

**Impact of various VBD traffic loads (percentages of traffic) on DCME gain**




Conditions: Voice Activity Factor (AF) = 37%, encoding rate bit/sample per channel = 3.7, FEC = on, bearer channel = CEPT 31 channels.

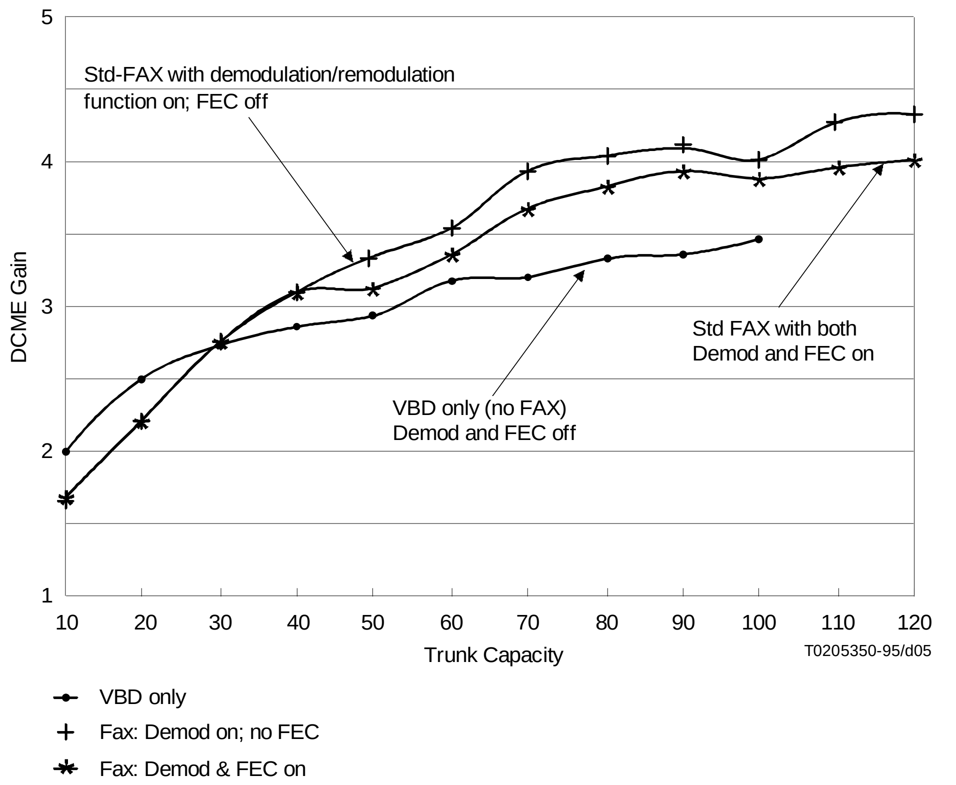
FIGURE A.4/E.528

**Variations of standard facsimile loading to VBD channels**

It is assumed that in the example of Figure A.3, the traffic contains no facsimile calls in the VBD traffic and that the facsimile-associated functions – demodulation and Forward Error Correction (FEC) – are disabled.

The VBD traffic, in the example of Figure A.4, contains various percentages of facsimile calls (80%, 60% and 40%) with the facsimile-associated functions – demodulation/remodulation and Forward Error Correction (FEC) – enabled. The AF and encoding rates are chosen to be constant and equal correspondingly to 37% and 3.7-bit/sample. In this example, DCME connects to the full bearer channel (CEPT).

Figure A.5 below shows the impact of the facsimile calls and the impact of facsimile functions – demodulation/remodulation and FEC – on the DCME gain. The standard facsimile is the G3-type-FAX transmitted at 9.6 kbit/s.




Conditions: 20% of trunks channels is VBD, 90% of VBD trunk channels is FAX channels, 80% of FAX channels is standard, FAX AF = 37%. Encoding rate: bit/sample per channel = 3.7.

FIGURE A.5/E.528

**Impact of demodulation/remodulation and FEC applied to standard facsimile and VBD traffic**

The figure clearly shows that the facsimile calls with the demodulation/remodulation function yield better gain than the ADPCM-path facsimile calls. The graphs also indicate the FEC impact on the gain. When the FEC is enabled, a DCME gain for the standard facsimile calls slightly decreases because the FEC uses the bearer capacity.

Figure A.5 also illustrates an impact on the DCME gain produced by the changing percentage of standard facsimile calls (at 9.6 kbit/s) for a given 20% share of data trunk channels. The gain improves as the percentage of standard facsimile calls increases. This improvement would occur if more facsimile calls were routed through 9.6 kbit/s facsimile paths than through ADPCM paths.

## A.2 An example of DCME dimensioning calculation

An example of DCME link dimensioning is included in this Recommendation. The formulae used for the calculations are specified in Table A.1 and conform to Recommendation G.763. Two cases are considered in this example. The first case considers only the voice DCME traffic. In the second case, examples include the traffic mixes of preassigned traffic, dynamic loaded voice, voiceband, non-standard and standard facsimile with demodulation/remodulation and FEC functions.

In both cases, the calculations were performed for an AF of 37% and 3.7-bits/sample but with different traffic mixes. Tables A.1 and A.2 provide, in sequential form, the stages of DCME total gain calculation for the specified traffic mixes.

TABLE A.1/E.528

**DCME gain calculation for trunk sizes of 161 to 105 with various traffic mixes**

| Description | Case No. 1 | Case No. 2 |
|---|---|---|
| Input: trunk assumptions | | |
| Total number of trunks: N | 161 | 105 |
| Number of preassigned 64 kbit/s channels: $P_{64}$ | 0 | 1 |
| Number of preassigned 40 kbit/s channels: $P_{40}$ | 0 | 5 |
| Number of preassigned 32 kbit/s channels: $P_{30}$ | 0 | 1 |
| Percentage of trunks for VBD channels: $p_{VBD}$ | 0 | 20% |
| Percentage of VBD trunks for FAX channels: $p_{fax}$ | 0 | 90% |
| Percentage of FAX-channel trunks for standard FAX: $p_{stdfax}$ | 0 | 80% |
| Voice Activity Factor: AF | 37% | 37% |
| Sampling rates of a channel (bit/sample) | 3.7 | 3.7 |
| Trunk channel allocation calculation | | |
| Number of trunks for data channels: $N_{data}$<br>$N_{data} = p_d \times N$ | 0 | 21 |
| Number of trunks for voice channel: $N_{voice}$<br>$N_{voice} = N - N_{data} = (1 - p_d) \times N$ | 161 | 84 |
| Number of VBD channels for FAX channels: $N_{fax}$<br>$N_{fax} = N_{data} \times p_{fax}$ | 0 | 19 |
| Number of FAX channels for standard FAX: $N_{stdfax}$<br>$N_{stdfax} = N_{fax} \times p_{stdfax}$ | 0 | 15 |
| Number of FAX channels for non-standard (ADPCM) FAX: $N_{adpcmfax}$<br>$N_{adpcmfax} = N_{fax} - N_{stdfax}$ | 0 | 4 |
| Number of trunks for VBD channels: $N_{vbd}$<br>$N_{vbd} = N_{data} - N_{fax}$ | 0 | 2 |
| Overhead bearer channel (nibbles) consideration, bearer channel assumptions | | |
| FAX demodulation on/off:<br>if on, number of spare FAX banks, $N_{faxbk} = 2 + 1$,<br>includes the Fax Control Channel (FCC);<br>if off, number of spare FAX banks, $N_{faxbk} = 0$, all in nibbles | 0 | 3 |
| FEC on/off<br>if FEC on, FEC = k/n = 21/32 (Rec. G.766, pp. 5-10, defines rates for FEC);<br>if FEC off, FEC = 1 | 1 | 21/32 |
| Bit-bank for preassigned 40 kbit/s channels: $B_{bankp40}$ | 0 | 2 |
| Assignment Channel (AMC) = 1 nibble | 1 | 1 |
| Total overhead channel (nibbles) required in bearer channel (OVCH):<br>$OVCH = AMC + FCC + N_{faxbank} + B_{bankp40}$ | 1 | 6 |

TABLE A.2/E.528

**DCME gain calculation to trunk sizes of 161 to 105 with various traffic mixes**

| Description | Case No. 1 | Case No. 2 |
|---|---|---|
| FAX channel consideration, FAX parameters assumptions | | |
| Percentage of 9.6 kbit/s occupied by transmit side fax channel for silence elimination advantage: (TxSE) | 100% | 100% |
| Percentage of standard FAX transmit rate: ($Tx_{fax}$) | 100% | 100% |
| ADPCM gain = $G_d$ (64 000/40 000 = 8/5) = 1.6 | 1.6 | 1.6 |
| Average duration of FAX's low-speed signal in one direction: (referred to V21-type modem) $V21_{time} = 4.81$ | 4.81 | 4.81 |
| Typical page transmission time: $Page_{time} = 30$ s | 30 | 30 |
| Bearer channel rate: $V21_{rate} = 3$ kbit/s | 3 K | 3 K |
| Bearer channel rate of 9.6 kbit/s for standard FAX $B_{cRate} = f_{dlc}/2$ (ms) = 21/2 (ms) | 10.5 K | 10.5 K |
| Mean bit rate for a 9.6 single call: $M_{rate96}$ $M_{rate96} = (V21_{Rate} \times V21_{time} + B_{cRate} \times Page_{Time})/(V21_{time} + Page_{Time})$ | 9.46 K | 9.46 K |
| Gain for 9.6 standard fax call: $S_{faxg96}$ $S_{faxg96} = (1/FEC) \times (64\,000/M_{rate96})$ if FEC on, FEC = 21/32; if FEC off, FEC = 1 | 6.77 | 10.31 |
| Silence elimination advantage: $AD_{si} = 1/TX_{fax}$ | 1 | 1 |
| Bearer channel (nibbles) allocation calculation | | |
| Gain for voice channel: $G_v$ | 5.3 | 5.08 |
| Bearer nibbles for voice channel: $NB_v$ $NB_v = (N_{voice}G_v) \times 2$ | 61 | 34 |
| Bearer nibbles for VBD channels: $NB_{vbd}$ $NB_{vbd} = 2 \times N_{vbd}/ADPCM_{gain}$ | 0 | 2 |
| Bearer nibbles for non-standard FAX channel: $NB_{adpcmfax}$ $NB_{adpcmfax} = 2 \times N_{adpcmfax}/ADPCM_{gain} \times AD_{si}$ | 0 | 5 |
| Bearer nibbles for standard FAX channel: $NB_{stdfax}$ $NB_{stdfax} = 2 \times N_{stdfax}/S_{fax96} \times AD_{si}$ | 0 | 7 |
| Total required bearer channel (nibbles) and DCME gain calculation | | |
| Total required bearer channel (nibbles): ($NB_{total}$) $*$ $NB_{total} = NB_v + NB_{vbd} + NB_{adpcmfax} + NB_{stdfax} + OVHD + P_{64} \times 2 + P_{40} + P_{32}$ | 62 | 52 |
| DCME gain: $G_{DCME} = N/NB_{total}$ | 5.19 | 4.04 |

## Annex B

(This annex forms an integral part of this Recommendation)

TABLE B.1/E.528

**DCME gain for voice $G_v$**

| Sample rate (bits/sample) | Number of trunks (N) | Formula | Activity Factor (AF) | | |
|---|---|---|---|---|---|
| | | | 33% | 35% | 37% |
| 3.6 | $N < 80$ | $G_v = (a + b \times \ln(N)) \times 2$ | $a = 0.23$<br>$b = 0.61$ | $a = 0.04$<br>$b = 0.60$ | $a = 0.30$<br>$b = 0.51$ |
| | $N > 80$ | $G_v = \dfrac{1.1388 \times N}{N \times AF + \sqrt{(N \times AF)}} \times 2$ | $AF = 0.33$ | $AF = 0.35$ | $AF = 0.37$ |
| 3.7 | $N < 80$ | $G_v = (a + b \times \ln(N)) \times 2$ | $a = 0.23$<br>$b = 0.61$ | $a = 0.04$<br>$b = 0.60$ | $a = 0.27$<br>$b = 0.52$ |
| | $N > 80$ | $G_v = \dfrac{1.1081 \times N}{N \times AF + \sqrt{(N \times AF)}} \times 2$ | $AF = 0.33$ | $AF = 0.35$ | $AF = 0.37$ |
| 3.8 | $N < 80$ | $G_v = (a + b \times \ln(N)) \times 2$ | $a = 0.24$<br>$b = 0.59$ | $a = 0.01$<br>$b = 0.61$ | $a = 0.28$<br>$b = 0.51$ |
| | $N > 80$ | $G_v = \dfrac{1.0789 \times N}{N \times AF + \sqrt{(N \times AF)}} \times 2$ | $AF = 0.33$ | $AF = 0.35$ | $AF = 0.37$ |

## Appendice I

(This appendix does not form an integral part of this Recommendation)

### I.1 Dynamic Load Control (DLC) functionality

The key DCME action that effects the congestion in the trunk channels is Dynamic Load Control (DLC) which is, in general, sent by DCME when Average Bits per Sample (ABS) reaches a specific threshold predetermined by a trade-off between the speech quality and bearer capacity and stops the International Switching Centre (ISC) from sending new calls. DLC may also be activated if the percentage of a Voice Band Data (VBD) call exceeds a specified limit.

Point-to-point DLC functionality is illustrated in Figure I.1.

The key elements for dimensioning of the point-to-point links that contain DCME are the maximum achievable $G_v$ and the maximum allowable percentage of VBD calls.

### I.2 Maximum achievable $G_v$

The DCME gain associated with voice channels, $G_v$, is achieved by a combination of two techniques: Digital Speech Interpolation (DSI) and Low Rate Encoding (LRE) based on ADPCM. A combination of these two methods more than quadruples the voice channel capacity. With increased occupancy of the bearer channels, DCME dynamically adjusts down the sampling rate of voice calls by the mechanisms of Variable Rate Encoding (VRE), lowering the ABS of these calls. With lowering ABS, $G_v$ increases and more of the bearer channels capacity becomes available for the voice calls. At the point, chosen by the user, when decreasing ABS level may cause an unacceptable speech quality and $G_v$ is at the maximum, DCME invokes DLC that stops ISC from sending new voice calls.

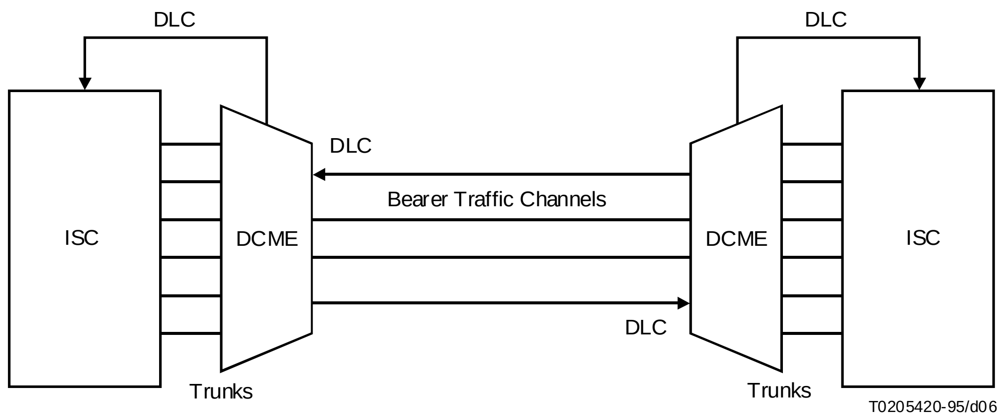



FIGURE I.1/E.528

**DLC functionality for point-to-point configuration**

## I.3 Maximum allowable percentage of VBD calls

The gain associated with VBD is achieved by utilization of LRE. ADPCM coding could double the data bearer occupancy (dbo). If the number of VBD calls is increasing, DCME will react to the increasing load and generate a DLC that will stop ISC from sending the VBD calls to this DCME. At this point, the percentage of VBD calls has reached the limit and is considered the maximum allowable by traffic requirements of a network.

## I.4 Determination of the blocking probabilities for dimensioning of point-to-point links that include DCME

### I.4.1 DLC activation and deactivation

Dimensioning of the DCME single links depends on the call blocking objectives of each traffic stream (voice and VBD) and the overall traffic throughput.

The call blocking probability of a DCME link with given trunk and bearer sizes for a given traffic load assumptions is calculated by analyzing the probability distribution of system states as given in Reference [2].

In this calculation the main parameter is f(j,k,AF,m) – a function that defines the ABS value for voice encoding, and the result is determination of parameters that activate and deactivate DLC. The relevant parameters are:

k is a number of voice calls in progress through the system;

j is a number of VBD calls in progress through the system;

m is a number of preassigned channels without the provision of DSI (clear channels);

AF is a voice Activity Factor.

Other relevant parameters that are used for these calculations are:

T a maximum allowable number of VBD calls connected through the system corresponds to data bearer occupancy (dbo);

w a minimum acceptable value of the ABS;

i type of call (i = 1 assumed for voice calls and i = 2 assumed for VBD calls).

The DLC signal triggering, then becomes a function of the values for T and w as shown below.

$$g_i(j,k) \begin{cases} 0 \text{ if } f(j,k,AF,m) < w \\ 0 \text{ if } k > T \\ 1 \text{ for all the other cases} \end{cases}$$

When voice calls (k) and VBD calls (j) are in progress and a call of i = 1 or i = 2 arrives, it is accepted with the probability of $g_i(j,k)$. The $g_i(j,k)$, in terms of the traffic stream directed to DCME, represents a function that describes the acceptance or denial of new arriving to DCME voice and VBD calls.

### I.4.2 Bearer channel loading conditions

The DLC signal, to be sent to the ISC, is determined by the bearer channels loading conditions. Examples of the load conditions for voice and VBD are given in clause 9/G.763 and B.1/G.763.

If voice calls are not present in the mix (k = 0), then all calls are coded at the maximum of up to 4-bit/sample, ABS value is not changing and therefore would not trigger DLC. DLC in this case will be triggered by the data load threshold, set by the user, e.g. 80% for the High Load (HL) threshold and 60% for the Low Load (LL) threshold.

If the voice channels are present (k > 0), the calculation results will show what value of ABS is corresponding to the given bearer capacity. DLC in this case is triggered by the thresholds based on ABS changing values (typically 3.6-bit/sample for HL threshold and up to 4-bit/sample for LL threshold).

The two load conditions, HL and LL, that trigger DLC activation and deactivation are established by clause 9/G.763.

- Denial of new calls to DCME is represented by the value of $g_i(j,k) = 0$ at the HL condition where the measured average level of ABS is less than the HL threshold (e.g. 3.6-bit/sample), or where the measured average dbo is greater than the high data load threshold (e.g. 80%). At this point DLC is activated.

- Acceptance of new calls to DCME is represented by the value of $g_i(j,k) = 1$ at the LL condition where the measured average value of ABS is greater than the LL threshold (e.g. 3.9-bit/sample) and the measured average dbo is less than the low data load threshold (e.g. 60%). At this point, DLC is deactivated.

This appendix does not address the presence of 64 kbit/sec. unrestricted channels. The effect of these channels on triggering DLC is for further study.

## References


[1] Recommendation G.763.

[2] CHADRAMOHAN (J.): A traffic engineering model for trunk groups with digital multiplication systems, *Teletraffic and Datatraffic in a period of Change. ITC-13A*, Jensen and V. B. Iversen (editors), Elsevier Science Publications. (B. V.) North Holland IAC, pages 243-246, 1991.

[3] CCITT *Blue Book* Volume V: Telephone Transmission Quality – P-Series of Recommendations.

# Appendix II

(to Recommendation E.528)

Soft version of the user guide is available on diskettes, from INTELSAT.

This appendix is not considered as an integral part of Recommendation E.528. Please address your request to: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