ANNEX B

(to Recommendation G.763)

## B.1 *An example of a DLC double averaging technique*

The average number of encoding bits per sample is obtained using a double average process.

a) The first stage averaging is computed at discrete time instances once every n DCME frames, where n is operator selectable (n = 4, 16, 32, 64 or 128). The result of the computation is the ensemble average <Se> taken over the ensemble of BCs which are carrying voice traffic and will result in one of the following possible outcomes:

—

—

with M =

voice band data, bit banks and 64 kbit/s on-demand traffic counted in the measurement frame.

N =

frame.

Two ensemble averages should be determined:

—

bits/sample <Se> based on ~~<Sea> —~~ actual counts of M and N.

—

<Se> based on the ~~<Sep> —~~ actual count of N and a reduced count of M—2.

b) The second stage averaging should be a moving discrete time averaging of <Sea> and <Sep>:

—

<Sea>.

—

<Sep>.

The value of Sta may be used as a measure of the average number of encoding bits/sample when determining the dynamic load control condition for voice and voice band data channels.

The value of Stp may be used as a measure of the average number of encoding bits/sample when determining the dynamic load control condition for on-demand 64 kbit/s channels.

## B.2 *Transmit activity detector threshold and operate time characteristic*

A typical response to a sinusoidal stimulus signal in the band 300 to 3400 Hz will be as given below:

*Average signal power* (see Note)
*Operate time*

< -40 dBm0

OFF

³ -40 dBm0, £ -30 dBm0

Figure B-1/G.763

> -30 dBm0

2 ms < t < 4 ms

The operate time requirements will be satisfied while permitting tolerances on the average signal power of any stimulus signal in the frequency band at boundary conditions as follows:

-40 dBm0 + 1.5 dB

-30 dBm0 + 1.0 dB

A typical rate of change of the transmit activity detector adaptive threshold will be between 2.5 dB/s and 20.0 dB/s.

*Note* — The activity detector should not indicate activity for idle channel noise less than -40 dBm0.
FIG. B-1/G.763

## B.3 *Data/speech discriminator*

Functionally, the data/speech (D/S) discriminator determines whether the activity on each transmit IT is speech or data and provides a speech/data indication to the hangover control and signal classification process.

The implementation of the D/S discriminator may be performed by a combination of spectral analysis and 2100 Hz tone detection.

The following requirements should be met with the modem types and bit rates given in Table 7/G.763.

### B.3.1 *Output conditions*

The D/S discriminator analyzes the activity on each transmit IT and provides the following output conditions:

| *Activity* | *Output condition* |
|---|---|
| Speech | “Voice” |
| Tones except 2100 Hz | “Voice” |
| Data signal (see Note) | “Data” |
| 2100 Hz | “Data” |

*Note* — V.23 modem signals may be classified as either voice or data dependent upon the design of the data/speech discriminator.

The D/S discriminator provides a continuous output condition indicating the presence of either speech or data on the ITs. The current output condition should be maintained upon termination of activity on the IT or until the output condition of a subsequent activity is determined. Upon system start-up or map change, the D/S discriminator should be reset to “Voice”.

### B.3.2 *Accuracy*

The missed detection probability of data as speech or speech as data should be less than 0.5%.

### B.3.3 *Response time*

The D/S discriminator should update its output condition within 200 ms after any of the following changes in the signal characteristics on the IT:

— Inactive-to-speech

— Inactive-to-data

— Speech-to-data

— Data-to-speech

### B.3.4 *2100 Hz tone detection*

The 2100 Hz tone detector should meet the following requirements:

— Frequency range of tone:

2100 + 21 Hz

— Minimum amplitude of tone:

-25 dBm0

— Response time:

< 100 ms (for further study)

## B.4 *2400 Hz tone detector*

The 2400 Hz tone detector should meet the following requirements:

— Frequency of tone:

2400 Hz + 15 Hz

— Minimum amplitude of tone:

-25 dBm0

— Response time:

< 50 ms

— Missed detection probability:

< 0.5%.

## B.5 *Speech detector/echo control device interactions*

Consideration must be given to minimizing excessive channel loading which may exist as the result of network interactions between the DCME speech detector and an echo control device (see Figure B-2/G.763).

If the DCME utilizes an adaptive threshold speech detector, interaction between the speech detector threshold adjustment and the echo control operation may generate excessive activity in the channel. The echo control device modulates the terrestrial circuit noise accumulated between the telephone and the send-input port of the echo control device. The adaptive threshold speech detector may falsely classify this change in terrestrial circuit noise as speech and increase the load on the DCME. This will increase the occurrence of overload channels and/or freeze-out, thereby degrading the performance in the baseband channel. This interaction occurs as follows:

a) Receive speech arrives at the receive input (Rin) port of the echo control unit.

b) The echo suppression switch or canceller centre clipper activates, stopping the echo or residual echo and attenuating the near-end-generated analogue terrestrial noise (N1) present at the send input (Sin) port.

c) If very little noise is generated between the echo control send output (Sout) port and the DCME speech detector input, the speech detector threshold will adapt to its minimum level (typically -50 dBm0).

d) When the receive speech stops, after a suitable echo control unit handover time the echo suppression switch or canceller centre clipper will close and the near-end-generated terrestrial noise, as seen by the DCME speech detector will reappear as a step change in noise level.

e) This step change in noise level may exceed the speech detector threshold, causing the DCME to transmit a noise spurt as if it were speech. The noise spurt duration will be a function of the adaptation speed of the speech detector and the near-end-generated terrestrial noise level.

This sequence will be repeated for every speech spurt and will produce a very annoying speech-correlated noise spurt heard by the far-end talkers every time they stop speaking.

This interaction is not limited to single echo control device network configurations. A typical network configuration with multiple echo control devices interacting with a DCME speech detector is shown in Figure B-3/G.763. In this configuration, the DCME speech detector may respond to unit step increases in noise power which result from echo suppressor switch or echo canceller centre clipper activations in the send paths of echo control devices 1 and 3. The DCME speech detector will first experience a unit step increase in noise power from echo control device 3 switch activation, followed by a second step increase from echo control device 1 switch activation. The extent to which the DCME speech detector incorrectly responds to these step increases in noise power will be a function of the noise power levels N1, N2, N3, and N4, and the specific DCME speech detector threshold adaptation algorithm. For example, the dual step increases in noise presented to the DCME speech detector which result from switch or centre clipper activation at locations 1 and 3 will be masked if the power level of N4 is excessively high. Likewise, high noise power levels at N2 or N3 may mask step increases in noise power caused by echo control unit 1.

There are several methods for dealing with the interactions between the echo control devices and the DCME speech detector. In one approach, the echo control device could be modified to monitor the terrestrial-generated noise at the send-input port. When the send transmission path is broken, noise at the proper level is injected into the send-output toward the

DCME, keeping the noise seen by the speech detector at a constant level and avoiding speech detector activation. This approach is unacceptable due to the number of different echo control devices in use and the uniqueness of the application. In a second approach, the speech detector adaptive threshold would be frozen in the presence of speech on the corresponding receive channel. A third approach would be to specify an adaptive speech detector with a fast adaptation feature which would track step changes in noise level and minimize the noise spurts.

The transmit activity detector threshold should not adapt to Gaussian noise level variations which are due to the action of echo suppressors or echo cancellers. This may be accomplished by any means which is functionally equivalent to providing the transmit activity detector with a threshold inhibit signal from a receive activity detector when activity is present on the receive channel (see § 12.4).

Fig. B-2/G.763 = 7,5 cm

Fig. B-3/G.763 = 6 cm

## B.6 *Timing synchronization*

The following figures provide a number of examples of Doppler and plesiochronous slip buffer placements for a variety of network synchronization schemes. In the figures it is assumed that all buffers will derive their write clocks from the input bit stream.

### B.6.1 *Point-to-point operation*

#### B.6.1.1 *Terrestrial operation within a national network*

Figures B-4/G.763 and B-5/G.763 show methods of DCME terminal synchronization for operation within a national network.

Fig. B-4 y B-5/G.763 = 12 cm

### B.6.1.2 *Terrestrial operation between national networks*

Figures B-6/G.763, B-7/G.763 and B-8/G.763 show methods of terminal synchronization for operation between national networks via terrestrial networks. Plesiochronous buffers are required for networks as shown in Figures B-6/G.763 and B-7/G.763. Figure B-8/G.763 utilizes loop timing and therefore does not require plesiochronous buffering.
Fig. B-6/G.763 = 6,5 cm

Fig. B-7 y B-8/G.763 = 13 cm

### B.6.1.3 *Satellite operation between national networks based upon continuous digital carrier type services*

Figures B-9/G.763, B-10/G.763, B-11/G.763 and B-12/G.763 show methods of terminal synchronization for operation between national networks over a satellite link based upon asynchronous continuous digital carrier type services. Figure B-9/G.763 introduces controlled slips between the DCMEs which are limited to 1 in 70 days if G.811 clocks are available in both networks. The configuration shown in Figures B-10/G.763, B-11/G.763 and B-12/G.763 permit slip free operation between the DCMEs.

Fig. B-9, B-10 y Fig. B-11/G.763 = 6,5 cm

B-12 /G.763 = 6,5 cm

#### B.6.1.4 *Satellite operation between national networks based upon TDMA type services*

Figures B-13/G.763 and B-14/G.763 show a method of DCME terminal synchronization for operation between national networks over a satellite link based on TDMA-type services. An appropriate interface is provided in the TDMA terminal to permit interfacing the DCME with and without multi-clique over a primary multiplex port. The configuration shown in Figure B-13/G.763 permits slip free operation between the DCMEs.

### B.6.2 *Multi-clique operation*

#### B.6.2.1 *Terrestrial operation within a national network*

Figure B-15/G.763 shows a method of DCME terminal synchronization for operation within a national network. The cross connect function provides a means of assembling the received multi-clique pools on to a single primary multiplex.

#### B.6.2.2 *Terrestrial operation between national networks*

Figure B-16/G.763 shows a method of DCME terminal synchronization for operation between national networks via terrestrial facilities. Plesiochronous buffers are required to resolve timing differences between the various plesiochronous networks. Due to the multiple source nature of the multi-clique configuration, the plesiochronous buffers must be placed before the cross connect function.

#### B.6.2.3 *Satellite operation between national networks based upon continuous carrier type services*

Figure B-17/G.763 shows a method of DCME terminal synchronization for operation between national networks based on continuous digital satellite carriers. Plesiochronous/doppler buffers are required to resolve timing differences between the various plesiochronous networks and to remove satellite induced doppler shifts on the received data streams. Due to the multiple source nature of the multi-clique configuration, the plesiochronous/doppler buffers must be placed before the cross connect function.

Fig. B-13/G.763 = 22 cm

Fig. B-14G.763 = 22 cm

Fig. B-15/G.763 = 22 cm

Fig. B-16/G.763 = 22 cm

Fig. B-17/G.763 = 22 cm

### B.6.3 *Multi-destination operation*

#### B.6.3.1 *Terrestrial operation within a national network*

Figure B-18/G.763 shows a method of DCME terminal synchronization for operation within a national network. The received data streams are assumed to originate from mutually synchronized sources.

#### B.6.3.2 *Terrestrial operation between national networks*

Figure B-19/G.763 shows a method of DCME terminal synchronization for operation between national networks via terrestrial facilities. Plesiochronous buffers are required to resolve timing differences between the various plesiochronous networks. Due to the multiple source nature of the multi- destination configuration, the plesiochronous buffers must be placed before the DCME receive function.

#### B.6.3.3 *Satellite operation between national networks based upon continuous carrier type services*

Figure B-20/G.763 shows a method of DCME terminal synchronization for operation between national networks based on continuous digital satellite carriers. Plesiochronous/doppler buffers are required to resolve timing differences between the plesiochronous networks and to remove satellite induced doppler shifts on the received data streams. Due to the multiple source nature of the receive signals in the multi-destination configuration, the plesiochronous/doppler buffers must be placed before the DCME receiver.

#### B.6.3.4 *Satellite operation between national networks based upon TDMA-type services*

Figures B-21/G.763 and B-22/G.763 show a method of DCME terminal synchronization for operation between national networks over a satellite link based on TDMA-type services. An appropriate interface is provided in the TDMA terminal to permit interfacing the DCME over a primary multiplex port. The configuration shown in Figure B-21/G.763 permits slip free operation between the DCMEs.

### B.7 *Performance*

#### B.7.1 *Speech performance* (provisional)

Recommendation P.84 describes a subjective test method for comparing the performance of DCME systems against suitable reference conditions for carefully defined input signals. Recommendation P.84 consists of listening tests and is the recommended source of information about subjective testing of DCME. These tests are a first step and do not preclude the need for conversational tests.

It is recommended that a fixed delay be inserted in the transmit speech path to reduce the probability of front end clipping. This delay compensates for activity detection time and DCME assignment message connection delay. The delay should be such as to assure that the main speech spurt clipping is less than 5 ms.

### B.7.2 *Voice band data performance*

Extensive testing has demonstrated satisfactory voice band data performance for the 40 kbit/s algorithm specified in Recommendation G.726 for voice band data rates up to 9600 bit/s.

Voice band data at rates up to 12 000 bit/s can be accommodated by 40 kbit/s ADPCM. The performance of V.33 modems operating at 14 400 bit/s over 40 kbit/s ADPCM is for further study. Selection of a 64 kbit/s unrestricted channel through a DCME is also possible and may be used for V.33 modems operating at 14 400 bits.

Fig. B-18/G.763 = 22 cm

Fig. B-19/G.763 = 22 cm

Fig. B-20/G.763 = 22 cm

Fig. B-21/G.763 = 22 cm

Fig. B-22/G.763 = 22 cm

**Supplement No. 1**

DCME TUTORIAL

(to Recommendation G.763)

## 1 Use of digital circuit multiplication system (DCMS)

DCMS provide the means to reduce the cost of transmission (e.g. long distance transmission) by making use of the combination of digital speech interpolation (DSI) and low rate encoding (LRE) techniques.

DSI is used to concentrate a number of input channels (generally referred to as trunk channels) onto a smaller number of output channels (generally referred to as bearer channels). It does this by connecting a trunk channel to a bearer channel only for the period that the trunk channel is active, i.e. is carrying a burst of speech or voice-band data. Since in average conversations one direction of transmission is active only for 30% to 40% of the time, if the number of trunks is large the statistics of the speech and silence distributions will permit a significantly smaller number of bearer channels (bearer channel pool) to be used. Control information must also be passed between the terminals to make sure that bearer and trunk channel assignments at each end remain synchronized.

LRE uses digital filtering techniques to construct an estimate of the waveform at both the encoder and the decoder. Since the actual information rate of speech is much lower than the channel Nyquist rate the link used between the LRE encoder and the decoder can operate at a rate which is dependent mainly on the quality of the models and the permissible amount of transmission degradation. The CCITT has standardized in Recommendations G.726 and G.727 a type of LRE known as ADPCM, the performance of which has been extensively characterized. DCME uses the ADPCM defined in Recommendation G.726.

Facsimile compression uses recognition and decoding of some or all of the voice-band signals sent by the modem to enable the sub-multiplexing of the digital information from a number of trunk channels onto a reduced number of bearer channels with the object of enhancing both the quality and the efficiency of transmission as compared to rate reduction of the signals using ADPCM. This is under study.

The simplest way to use DCMS is in the single destination mode as shown in Figure 1/G.763. This mode of operation is most economic for the largest routes. For smaller routes there are two options:

— operation in multi-clique mode,

— operation in multi-destination mode.

Operation in multi-clique mode, see Figure 2/G.763, divides the bearer channels into a number of blocks or cliques, each associated with a different route. There is normally a fixed boundary between cliques, and trunk/bearer channel assignments are generally carried in a control channel within the clique to which they refer. This limits the dynamic processing of received channels to those which are contained in the wanted clique; selection of the wanted clique channels can be done using a simple static digital switch without reference to the assignment information. With a 2048 kbit/s bearer system in multi-clique DCMS the statistics of the DSI are unpromising with more than three routes. Recommendation G.763 provides for two cliques.

Operation in multi-destination mode, see Figure 3/G.763, permits any bearer channel to be associated with any trunk channel of any of a number of different routes. There is no segregation of routes on the bearer, and therefore at the receive terminal it is impossible to select the wanted channels without reference to the assignment information. Multi-destination mode is economic for very small routes via satellite, but practical difficulties limit the number of routes which it is desirable to have on a single DCMS.

## 2 Location

Location of DCME depends on its use. Equipment used in single destination mode or in multi-clique mode can in general be located at:

- ISC,
- earth station,
- cable head,

without significant restrictions. Equipment used in the multi-clique mode will typically be located at the ISC so that the advantages of DCMG can be extended over the national section. Equipment used in the multi-destination mode will typically be located at the earth station or cable head. The reason for this is that whereas in multi-clique mode the number of receive bearer channels at the DCME terminal is approximately equal to the number of transmit bearer channels, in multi-destination mode the number of receive bearer channels at the DCME terminal is the number of transmit bearer channels multiplied by the number of destinations. It therefore may be uneconomic to provide sufficient transmission capacity between earth station and ISC to permit location of multi-destination DCME at an ISC.

## 3 Transmission requirements

DCMS are usually required to carry any traffic which can be carried on ordinary General Switched Telephone Network (GSTN) connections. That includes voice-band data using V-Series Recommendation GSTN modems, facsimile calls following Recommendations T.4 and T.30 and using V.29 modems. In addition, in the ISDN 64 kbit/s unrestricted on-demand digital data and alternate speech/64 kbit/s unrestricted bearer services must be carried.

DCMS are primarily designed to maximize the efficiency of speech transmission. Use with voice-band data, especially at high rates, presents problems. These problems are mainly due to the difficulty for 32 kbit/s ADPCM of encoding voice-band data waveforms.

## 4 DCME gain (DCMG)

The gain of DCME is the input trunk channel transmission multiplication ratio, which is achieved through application of DCME, including LRE and DSI (for a specified speech quality at a certain level of bearer channel activity). The maximum available gain depends on:

- number of trunk channels;
- number of bearer channels;
- trunk channel occupancy;
- speech activity;
- voice-band data traffic;
- ratio of half duplex to full duplex voice-band data;

— type of signalling;

— 64 kbit/s traffic;

— minimum acceptable speech quality;

— dynamic load control threshold.

Of these the factor which has the greatest significance is the percentage of 64 kbit/s digital data traffic. This is because a trunk channel carrying 64 kbit/s traffic requires two 32 kbit/s bearer channels to be removed from the pool of channels available to the DSI process.

The peak percentage of voice-band data may vary between 5 and 30 per cent, depending on route. This is discussed in greater detail in Supplement No. 2.

The type of signalling system used on the route can significantly affect the gain. Continuously compelled signalling systems hold channels active for undesirably long periods. In the case of CCITT R2 digital signalling via a DCMS used on a satellite, the channel might be active for 5 to 14 seconds.

The measured speech activity depends on the characteristics of the activity detector. It is usual to assume 35 to 40 per cent. Channels with high ambient background noise can increase this activity factor. Outside of the route busy hour the occupancy of the trunk channels by traffic will be lower than in the route busy hour. The effect of this is to reduce the ensemble activity measured by the activity detector to about 27 per cent outside the route busy hour, whereas it will be close to the speech activity factor, i.e. about 40 per cent during the route busy hour.

The speech quality is governed by two main factors; the LRE encoding rate, and the amount of speech lost while a newly active trunk channel is awaiting connection to a bearer channel. If there are a great many newly active trunk channels in competition the beginning of a burst of speech is more likely to be clipped or frozen out than if relatively few trunk channels are active.

The speech quality of a DCME in a network with external echo control devices may be affected by clipping introduced by echo control devices and by a possible noise contrast effect. In particular when echo suppressors or echo cancellers are used on circuits where the near end generated noise is high with respect to the noise generated in the remainder of the link, suppression of the far end noise may be objectionable due to noise contrast. Possible means of eliminating this problem are use of echo control devices which insert idle line noise at the appropriate level during suppression periods, or insertion of idle line noise at the DCME during the relevant period when the echo control device is integrated in the DCME. Another approach is discussed in Annex B, § B.5 to Recommendation G.763.

When commissioning a new DCMS, observations should be made of the type and characteristics of the traffic which will use it. It is unwise to rely solely on customer complaints to indicate when a system is poorly dimensioned. This is because interactions between the DCMS and echo control (note) may obscure the true problem. Furthermore the consequence of trying to concentrate too many trunk channels onto too few bearers may be simply to increase the calling rate and to reduce the call holding time. This may result in greatly reduced quality, especially where continuously compelled signalling systems are used, and levels of trunk channel activity occur far above what was envisaged in the original system dimensioning.

*Note* — This highest speech quality is obtained when echo cancellers conforming to Recommendation G.165 (Red Book) are used for echo control. However echo suppressors conforming to Recommendations G.164 (Red Book) and G.161 (Yellow Book) may be used.

Two possible criteria for acceptable speech performance are an average of 3.7 bits per sample and less than 2.0% probability of clipping exceeding 50 ms, or alternatively that less than 0.5% of speech should be lost due to clipping.

Using the above criteria, approximations have been derived that relate the percentage of voice-band data and the number of trunk channels to the gain of a DCME. Approximations intended for use in initial system dimensioning are given in Supplement No. 2 to Recommendation G.763.

If a more accurate representation is required, then it will be necessary to do the first order Markov chain analyses referred to in the literature on DSI [1], [2], [3].

## 5 ISDN bearer services

DCMS are generally required to carry the full range of ISDN bearer services which can be provided on a 64 kbit/s channel as specified in Recommendation I.230 (Blue Book). These are:

— Circuit mode 64 kbit/s unrestricted, 8 kHz structured bearer service category.
This may be used among other things for speech, multiple sub-rate information streams multiplexed by the user, or for transparent access to an X.25 public network.

—

information transfer.

This is broadly similar to the preceding category, but with different access protocols.

— Circuit mode 64 kbit/s, 8 kHz structured bearer service category, usable for 3.1 kHz audio information transfer.
This bearer service provides the transfer of 3.1 kHz bandwidth audio information, such as for example voice-band data via modems, Group I, II and III facsimile information, and speech.

—

service category.
This service is similar to both the unrestricted and speech 64 kbit/s circuit-mode bearer services, but provides for the alternate transfer of either voice or unrestricted digital information at 64 kbit/s within the same call.

## 6 Restoration of services

For most applications the loss of traffic under failure conditions would be such that it would be insufficient to install a single pair of terminals on a route without a means of rapid changeover to spare equipment in the event of failure. This means that DCME is often used in a cluster of N active DCMEs for one standby. Automatic changeover permits the standby to be loaded with the configuration and synchronization information of the failed terminal. Other automatic fallback modes may be considered.

Failure of the transmission system between DCME terminals can be handled by normal transmission system restoration procedures. Failure of the transmission systems entering the DCME terminals from the exchanges may result in a wide range of different alarm conditions being experienced particularly where a multi-destination DCME terminal serves more than one exchange and more than one route. It is desirable to limit the generation of alarm conditions to the channels which have actually failed.

## 7 Control of transmission overload

The reduction in the number of bearer channels available to the interpolation process, due to the high activities of voice band and 64 kbit/s data services or statistical variations in the ensemble input speech activity can occur when the number of instantaneously active trunk channels exceeds the number of available bearer channels. Either event requires action to be taken to safeguard speech quality. There are four possible solutions:

- The system can be dimensioned so that with the maximum anticipated short-term trunk channel activities there is negligible probability of violating the speech quality criteria. This employs the DCMS very inefficiently outside the busy hour.
- A multi-destination system can be made to carry routes with widely different busy hours, so that though the trunk channels might have relatively low non-busy hour occupancy, the bearer channels would always be well loaded.

— Signals can be sent from the DCME to the exchange to busy out part of the route when the quality criteria are violated. This is known as dynamic load control (DLC), and can be an effective control method, but it cannot be retrospective and it is slow to take effect. Furthermore care must be taken to ensure that when circuits are returned to service the increase in bearer channel activity is not sufficient to result in the immediate reapplication of DLC.

— The signal to quantization performance can be traded against the clipping of speech bursts. By using variable rate ADPCM algorithms it is possible to quantize to three or optionally two rather than four bits on individual speech channels on a pseudo-cyclic basis for a given number of samples. In this way the system can be given a gradual degradation characteristic, rather than suddenly overloading.

In a DCME conforming to Recommendation G.763 all of these techniques may be used.

## 8 Transmission link performance monitoring

Experience with DCMEs has shown the value of using cyclic redundancy check information in the detection and tracing of certain faults. In order to provide a comprehensive set of long-term and short-term indicators the DCME should provide the following means of monitoring the performance of any digital path(s) terminated upon it:

— cyclic redundancy check (CRC);

— frame alignment signal (FAS);

— other primary rate alarms;

— far end block error information of distant CRC (FEBE);

— DCME control channel FAS;

— violations of the Golay FEC of the control channel(s).

**References**

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[2] BRADY (P.T.): A model for generating on-off speech patterns in 2-way conversation, *Bell System Technical Journal*, page 2445 *et seq*, September 1969.

[3] Special issue on bit rate reduction and speech interpolation, Guest Editors M.R. Aaron and N.S. Tayant, *IEEE Trans. on communications*, Vol. COM-30, No. 4, April 1982.

**Supplement No. 2**

DCME DIMENSIONING METHODS FOR DIFFERENT ROUTE CHARACTERISTICS

(to Recommendation G.763)

# 1 Introduction

This supplement draws attention to the implications of the measurements of channel occupancy and voice-band data levels which have been done on particular routes for which the number of voice-band data calls is either large in absolute terms, or large compared to the total number of calls.

# 2 Route profiles

Figure 1 shows the kind of profile which has been obtained from measurements on an FDM route between the United Kingdom and a country for which the proportion of voice-band data calls was suspected to be high. It can be seen from this that there are two peaks which are of interest in DCME dimensioning — one (the voice peak) where voice is the dominant feature with a relatively small amount of voice-band data, and another (the data peak) where voice-band data dominates voice.

*Note* — The data profile is not symmetric in each direction of transmission.

Voice-band data requires more bearer capacity than voice in a DCME system incorporating digital speech interpolation (DSI) and low rate encoding (LRE) and therefore it is not immediately obvious which of these peaks is the limiting factor when calculating the achievable gain of a DCME on a particular route. Each route has to be examined carefully to determine the achievable gain. The limiting value of the gain does not necessarily occur at either of the peaks and in practice a scan across several days' profiles is necessary to determine the achievable gain.
FIG. 1

Figure 2 shows a typical profile obtained from the TDMA route for the same country. Due to different traffic origins and loading distributions the voice and data peaks are coincident, and the transmit and receive profiles are more nearly symmetrical in this case.
FIGURE 2

## 3 DCME operation

Figure 3 shows a DCME consisting of a DSI stage and an LRE stage. Voice and voice-band data have to be treated separately in each of these stages when trying to access the achievable gain of a particular DCME faced with a particular route profile.

### 3.1 *DSI gain for voice*

This is dependent upon the number of input trunks carrying voice and it is *not a linear relationship*.

Fig. 3 = 6 cm

## 3.2 *DSI gain for data*

Facsimile is the dominant data service and can be considered as half duplex, i.e. on a particular call if data is flowing in one direction of transmission at a particular time, then the opposite direction is silent. If the total amount of facsimile traffic in one direction of transmission is balanced by an equivalent amount in the opposite direction of transmission then a technique known as silence elimination can be employed to free the opposite channel when data is flowing in one direction. This leads to a theoretical DSI gain of 2. However, if the total facsimile traffic on a route is not balanced in each direction of transmission, making silence elimination difficult to implement (or if silence elimination has not been built into a particular DCME) then the DSI gain for voice-band data is 1.

## 3.3 *LRE gain for voice*

Studies have indicated that the minimum acceptable average number of bits per sample is of the order of 3.6, which will be the threshold for operation of dynamic load control. Therefore the LRE gain for voice is unlikely to exceed 8/3.6.

## 3.4 *LRE gain for data*

The LRE gain for data depends on how many bits/sample a particular system allocates to a data call.

In this supplement all calculations assume the use of the 40 kbit/s encoding rate for voice-band data, in conformity with Recommendation G.763, therefore the LRE gain for data = 8/5.

Examples for facsimile compression are not presented.

# 4 Calculation of DCME gain

Table 1 gives some approximate non-analytical formulas for calculation of the voice part of the DCME gain. It should be noted that these approximations are strictly valid only for DCMEs conforming to Recommendation G.763 and having ideal speech detection (i.e. the activity indicated by the speech detector is the same as the actual speech activity).

µTABLE 1

**Formulas for voice interpolation gain (Gv)**

No. of

No. of

Formula

Activity factor (AF)

bits/sample

trunks (N)

33%

35%

37%

a = 0.23

a = 0.04

a = 0.30

N < 80

Gv = a + b ´ 1n(N)

3.6

b = 0.61

b = 0.60

b = 0.51



N > = 80

Gv =

### 4.1 *Limitations*

Ideally the calculation of the DCME gain would be done by a comprehensive computer modelling of the system in the way which has already been demonstrated with great success by Swedish Telecom Radio. Given an intimate knowledge of the route, in terms of its hourly, daily and seasonal variations in voice and voice-band data traffic flow, signalling systems, call holding times and effective/ineffective ratios over a period of time it may be possible to model the route with a high degree of accuracy, at least retrospectively. However the major limitation is the quality of the information fed into the model. To overcome this limitation the digital channel occupancy analyser (DCOA) has been developed. If the DCOA is used on a group of circuits which previous sampling or other information has shown to be typical then very useful dimensioning information results. The limitation then is the total permissible measuring time. In most cases, for operational reasons, greater than two weeks is unlikely to be feasible. This represents a severe limitation on the attempt to create an accurate model, such that for dimensioning (as opposed to the verification of the operation of the equipment) Monte Carlo type simulations do not appear to be necessary.

### 4.2 *Example gain calculations using simplified techniques*

The following examples illustrate the concepts outlined in § 2, and demonstrate the use of a simplified technique for DCME dimensioning using DCOA route profiles.

#### 4.2.1 *DCME dimensioning using the profile of a route without silence elimination*

*Assumptions:*

Number of trunk channels at service date = 240.

Figure 4 is the applicable DCOA route profile.

FIG. 4

*Remark:*

From experience or from rough calculations it can be seen that for the given number of trunk channels and quantity of voice-band data traffic at least three DCMEs each using 30 bearer channels are likely to be required, but let us assume that four DCMEs are to be used on the route in order to calculate the gain for the voice traffic (this gain is dependent upon how many DCMEs the voice traffic is spread over). This is to ensure that the DCMEs are not overloaded and may also allow for growth on the route. In practice an iterative procedure would have to be used to determine the optimum number of DCMEs for each route.

From Figure 4 there are two peaks to be considered. One is dominated by the amount of data (data peak) and the other is dominated by the amount of voice (voice peak):

*Data peak*

59% data:

number of data trunks

= 240 ´ 0.59

= 142 trunks,

= ~~0~~36

DSI gain

= ~~00~~1 (no silence elimination advantage to be gained because almost all

~~= 001~~ the data is in one direction of transmission)

LRE gain =

17% voice:

number of voice trunks

= 240 ´ 0.17

= ~~0~~41 trunks total

number of voice trunks

per DCME = ~~0~~10

DSI gain (for 10 trunks)

= 1.25 (from tables)

LRE gain

=

Hence the 64 kbit/s bearer channel requirement is:

= 23 (data) + 4 (voice)

= *27 bearer channels.*

The total bearer requirement is therefore:

27 ´ 4

= *108 bearer channels.*

*Voice peak*

13% data:

number of data trunks

= 240 ´ 0.13

= ~~0~~32 trunks total,

= 8

DSI gain

= ~~00~~1 (no silence elimination advantage to be gained

because almost all t

~~= 001 tthe~~ data is in one direction of transmission),

LRE gain

=

83% voice:

number of voice trunks = 240 ´ 0.83

= 200 trunks total

number of voice trunks
per DCME = θ50

DSI gain (for 50 trunks)= 1.92 (from tables)

LRE gain
=

Hence the 64 kbit/s bearer channel requirement per DCME is:

= 5 (data) + 12 (voice)

= *17 bearer channels.*

The total bearer requirement is therefore:

17 ´ 4

= *68 bearer channels.*

*Inference:*

It seems therefore that in this case the DCME dimensioning is determined by the number of trunk channels required to cope with the speech peak, and by the number of bearer channels required to handle the data peak. Since the number of channels shown as active by the DCOA is an average over the measurement interval, it is reasonable to assume that all 240 trunk channels, rather than only 132 were active for some brief duration. Assuming that only the wanted bearer channels are used, and neglecting the assignment channel, the achievable gain will be:

.

### 4.2.2 *DCME dimensioning using the profile of a route with silence elimination*

*Assumptions:*

Number of trunk channels at service date = 347.

Figure 5 is the applicable DCOA route profile.

Fig. 5 = 13,5 cm

*Remark:*

On this route it appears that use of silence elimination will give some benefits. Other DCOA measurements have indicated that there is approximately twice as much voice-band data activity in the transmit direction as in the receive direction. Therefore the achievable DSI gain on voice-band data due to silence elimination is of the order of 1.5. This assumes that there are as many transmit as receive bearer channels on each DCME terminal. Rough calculations and experience indicate that because of the relatively low voice-band data percentage of this example three DCMEs will probably be sufficient.

From Figure 5 there is only one peak to be considered:

15% data:

number of data trunks = 347 ´ 0.15

= θ52 trunks

= θ18

DSI gain
= 1.5 (due to silence elimination)

LRE gain
=

72% voice:

number of voice trunks = 347 ´ 0.72

= 250 trunks total

number of voice trunks
per DCME = θ83

DSI gain (for 83 trunks)= 2.08 (from tables).

Hence the 64 kbit/s bearer channel requirement per DCME is:

= 8 (data) + 19 (voice)

= *27 bearer channels.*

The total bearer requirement is therefore:

27 ´ 3

= *81 bearer channels.*

*Inference:*

In this case, assuming that only the wanted bearer channels are used, the DCME can achieve a gain of:

.

However, as was shown by the previous example, it would be very unwise to assume that a DCME gain as high as four will be achievable for all types of DCME, without careful consideration of the route conditions. A corollary to this is that when a DCME has been installed on a route its performance must be continually monitored to ensure that changes in the traffic distribution on the route do not impact seriously upon the transmission quality.

### 4.3 *Two pitfalls for the unwary*

Figure 6 shows a plausible example of a DCOA record, covering a typical two hour period. On the basis of the trunk occupancy percentage for the route it might be thought that the maximum bearer occupancy would be coincident with the peak in voice traffic, however this is not so. The actual maximum occurs immediately before, as Figure 7 shows, during period 2. The reason for this is that the voice-band data traffic peaks before the voice traffic. Administrations may wish to consider whether this is a likely state of affairs; whether for example the facsimile transmission of financial results at close of business on any particular day is likely to result in follow-up telephone conversations. The relevant information for each period is summarized in Table 2.

FIG. 6

FIG 7

µTABLE 2

**Comparison of trunk and bearer occupancies**

Period

1

2

3

4

% chs

% chs

% chs

% chs

Data occupancy

20 26

25 ~~0~~32.5

15 ~~0~~19.5

10 13

Speech occupancy

55 71.5

55 ~~0~~71.5

80 104~~.5~~

60 78

Total occupancy

75 97.5

80 104~~.5~~

95 123.5

70 91

Data bearers

13

~~0~~16.5

~~0~~10~~.5~~

~~0~~6.5

Speech bearers

15

15~~.5~~

~~0~~21~~.5~~

16

Total bearers

28

31.5

~~0~~31~~.5~~

22.5

§

Care must be taken when the short-term characteristics of a measured route are not known. This may be especially significant when the route is small, since the presentation of voice-band data traffic may not be very uniform. Over a five minute period 2:1 variations in the short-term voice-band data activity level are not unusual events. It might therefore be prudent to repeat any dimensioning exercises which use a DCOA profile, but doubling all the voice-band data occupancies, for comparison against the absolute maximum number of channels available when *all* voice activity is allocated 3 bits. If that comparison shows that clipping would be experienced under those conditions then a lower gain setting should be chosen, based on whichever is believed to be the limiting period.

## 5 Conclusion

An approach to dimensioning DCME systems has been demonstrated, which though not statistically rigorous, is nevertheless capable of giving reasonable estimates of system capabilities, given adequate input data. A number of potential dimensioning problems have been described, and the solutions outlined. These methods have been used successfully in the introduction of DCMEs on a number of routes.