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Appendix I

Rec. G.765

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GENERAL ASPECTS OF DIGITAL TRANSMISSION SYSTEMS

A GUIDE TO PCME

Appendix I to

ITU-T Recommendation G.765

(Previously "CCITT Recommendation")

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).

Appendix I to ITU-T Recommendation G.765 was prepared by ITU-T Study Group 15 (1993-1996) and was approved under the WTSC Resolution No. 1 procedure on the 13th of November 1995.

NOTE

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

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Appendix I to Recommendation G.765

A GUIDE TO PCME

(Geneva, 1995)

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

I.1 Terms and definitions

This Recommendation uses the following terms that are not currently defined in Recommendation G.701.

I.1.1 block

F: bloc
S: bloque

A block is a specific group of octets within a voice packet that is made of bits of the same significance.

I.1.2 block dropping

F: abandon de blocs
S: eliminación de bloques

Block dropping is a process by which one or more of the less significant blocks of a voice packet is dropped to alleviate congestion.

I.1.3 block dropping indicator (BDI)

F: indicateur d'abandon de blocs
S: indicator de eliminación de bloques

The Block Dropping Indicator (BDI) is the field of the voice packet header that indicates the number of blocks that have been dropped and the maximum number that can be dropped.

I.1.4 bursts

F: salves
S: ráfagas

Bursts are periods of high energy content signals present in the access channel of a wideband network.

I.1.5 coding type (CT) field

F: champ du type de codage
S: campo de tipo de codificación

The coding type field of a voice/voiceband data packet is a 5-bit sequence in the packet header that indicates the method of coding speech samples used at the originating endpoint before packetization.

I.1.6 core bits

F: bits essentiels
S: bits primarios

The core bits form the subset of the total codeword of an embedded ADPCM coder that must reach the decoder.

I.1.7 data link connection identifier (DLCI)

F: identificateur de connexion de liaison de données
S: identificador de conexión de enlace de datos

The DLCI is a 13-bit field that defines the destination address of a frame on a physical-link-by-physical-link basis.

I.1.8 delay equalization (EQ) bit

F: bit de compensation du délai
S: bit de igualación de retardo

The EQ bit is a parameter used to indicate to the terminating endpoint whether a packet should be built out or played out immediately to the channelized side.

I.1.9 digital circuit emulation (DICE) protocol

F: protocole d'émulation de circuits numériques
S: protocolo de emulación de circuitos digitales

The DICE protocol is a wideband protocol used to transport digital data that arrive on the channelized side through a specific format containing idle codes and repetition of user’s data.

I.1.10 digital speech interpolation

F: interpolation numérique de la parole
S: interpolación digital de la palabra

Digital speech interpolation is a process that takes advantage of inactive periods of a conversation to insert speech from other conversations and to remove silent periods. This is the same process used for a G.763 DCME.

I.1.11 embedded adaptive differential pulse code modulation (ADPCM)

F: modulation par impulsions et codage différentiel adaptatif imbriqué (MICDA)
S: modulación por impulsos codificados diferencial adaptable jerarquizada (MICDA)

Embedded ADPCM algorithms are ADPCM algorithms that quantize the difference between the input and the estimated signal into core bits and enhancement bits.

I.1.12 enhancement bits

F: bits d'amélioration
S: bits secundarios

The enhancement bits in embedded ADPCM are bits that are not used in the prediction process in both the encoder and the decoder. They enhance the coding performance when used in the decoding process by reducing the quantization noise in the reconstructed signal.

I.1.13 errored second (ES)

F: seconde erronée
S: segundo con errores

An errored second is a second during which one or more errors have occurred.

I.1.14 frame relay

F: relayage de trame
S: retransmisión de tramas

Frame relay is a method of transporting HDLC frames within a network whereby the network nodes provide error detection without retransmission. Retransmissions are done at the endpoints of the network only.

I.1.15 gap

F: pause
S: pausa

A gap is a period of signals of low-energy contents present at a speech detection device.

I.1.16 high level data link control (HDLC) protocol

F: protocole de commande de liaison de données à haut niveau (HDLC)
S: protocolo de control de enlace de datos de alto nivel (HDLC)

HDLC is a family of bit-oriented, link-layer protocols defined by the International Organization for Standardization (ISO).

I.1.17 keep alive alarm

F: alarme sur perte de réception de paquets
S: alarma de pérdida de paquetes

A keep alive alarm is an alarm generated by a terminating endpoint that experiences loss of keep alive packets.

I.1.18 more (M) bit

F: bit de continuation
S: bit más (M)

The M-bit is a bit used to indicate that more packets in sequence are to be expected by the terminating endpoint.

I.1.19 noise field

F: champ de bruit
S: campo de ruido

The noise field is the field of the packet header that indicates the level of background noise the terminating end may play out in the absence of packets.

I.1.20 originating endpoint

F: extrémité d'origine
S: punto extremo de origen

The originating endpoint of a wideband packet node is the point that receives channelized traffic, packetizes it, and sends it into the wideband packet network.

I.1.21 packet header

F: en-tête de paquet
S: encabezamiento de paquete

The G.764/G.765 packet header consists of octets 4 to 8 (inclusive) of the frame (HDLC flags are excluded from the octet count).

I.1.22 packet stream

F: train de paquets
S: tren de paquetes

A packet stream is a collection of logical links multiplexed together onto one physical channel between two endpoints of the wideband packet network.

I.1.23 packetization interval

F: intervalle de mise en paquets
S: intervalo de empaquetado

The packetization interval is the duration of the channelized traffic that has been packetized.

I.1.24 scheduled play-out time

F: horaire de restitution
S: horario de restitución

The scheduled play-out time is the time at which a received packet is to be played out.

I.1.25 terminating endpoint

F: extrémité de signalisation
S: punto extremo de terminación

The terminating endpoint of a wideband packet node is the part of the node that receives packetized traffic, depacketizes it, and then plays it back as channelized traffic.

I.1.26 time stamp (TS)

F: horodateur
S: indicación de tiempo

The time stamp is a field that records the cumulative variable queueing delay experienced by packet in transversing the network with a resolution of 1 ms.

I.1.27 unnumbered information (Ul) frame

F: trame d'information non numérotée
S: trama de información no numerada (UI)

A Ul frame is a frame used to transfer unacknowledged information between two link-layer entities. The format and encoding are the same as specified in Recommendation Q.92111.441. The CRC is derived over the entire frame.

I.1.28 unnumbered information with header check (UIH) frame

F: trame d'information non numérotée avec vérification d'en-tête
S: trama de información no numerada con verificación de encabezamiento (UIH)

The UIH frame is similar to the UI frame except that the CRC sequence is derived over selected bits of the frame and not the whole frame. In Recommendation G.764, the CRC calculation is over the frame and packet headers (the first 8 octets excluding flags).

I.1.29 virtual data link capability (VDLC) protocol

F: protocole de capacité de liaison virtuelle de données
S: protocolo de capacidad de enlace de datos virtual

The VDLC protocol is a wideband protocol that is used to transport digital data packets arriving from the channelized side in HDLC frames.

I.1.30 voice frame

F: trame de parole
S: trama de voz

The voice frame is a G.764 UIH frame that contains a voice packet in its information field.

I.1.31 voice information field

F: champ d'information de la parole
S: campo de información de voz

The voice information field is the field that contains blocks of voice traffic.

I.1.32 wideband packet network

F: réseau de paquets à large bande
S: red de paquetes de banda ancha

A wideband packet network is a packet network that offers transmission channels capable of supporting rates above 64 kbit/s and below the broadband rate of 150 Mbit/s.

I.2 PCME tutorial

A Packet Circuit Multiplication Equipment (PCME) uses several strands of technology. These strands can be divided into two categories. The first category consists of technologies common to other Digital Circuit Multiplication Equipment (DCME), such as digital speech interpolation, echo cancellation, speech compression, and facsimile demodulation. It should be noted, however, that for these common techniques, PCME may offer additional features, such as embedded coding of speech, which can be useful in congestion control.

The second category comprises technologies unique to PCME, such as packetization and frame relay.

A network operator can utilize the PCME to make several types of connections simultaneously (within the constraints of the capacity of the system). The operator defines how the PCME is configured for each type. A typical connection involves the compression and packetization of voiceband traffic (voice or data). Other connections may include 64 kbit/s uncompressed channels that can be cross-connected as well. It is also possible to cross-connect streams of packets at each point of the network; this allows a PCME to operate within frame relay networks as well.

The following is a brief description of these technology strands.

I.2.1 Digital speech interpolation

Conversational speech is not continuous and rarely uses both directions of transmission simultaneously. Digital Speech Interpolation (DSI) takes advantage of inactive periods in a conversation to insert speech from other conversations. Because speech activity occupies, on the average, about 40 percent of the total time in a conversation, reduction of the bit rate needed for voice can be done by removing all idle periods, that is, removing silent intervals from speech or by coding the silence at a lower bit rate than that used for speech.

Time Assignment Speech Interpolation (TASI) was originally used for undersea analogue cable systems then transposed to digital satellite transmission systems. Later on, digital speech interpolation was combined with variable rate ADPCM coding (see I.2.3) to increase the efficient use of the transmission path.

To recognize that speech is being transmitted, a highly sensitive speech detector is required. The quality of the speech detector at the originating end is one of the important factors that determines the overall quality of the speech. If the speech detector does not detect speech correctly, it could chop the word's beginning *(ftdnt-end clipping),* thereby causing severe degradation to the speech quality. In contrast, if the speech detector is too sensitive, then more silence intervals will be passed and the gain will be reduced. Usually, the speech detector extends the effective duration of the speech burst, an interval denoted as *hang-over* to avoid tail-end clipping of speech. This extension reduces the gain; therefore, it is usually recommended that the extension does not increase the actual speech activity by more than 5 percent.

Another problem, called *freeze-out,* occurs when a talker requires a channel while all channels are occupied so that the leading edge of the incoming speech is not transmitted. This is alleviated by adding a fixed delay in G.763 DCME, as well as G.764/G.765 PCME. A PCME, however, utilizes this time to packetize the incoming traffic.

The terminating endpoint usually plays a *background noise fill* instead of silence to minimize the discontinuities between the background noise for speech and silence. Recommendation G.764 specifies the level of the noise fill in the packet header. This noise fill may be used also to replace missing or delayed packets.

Careful selection of the noise power is necessary to avoid the problem of "noise pumping", an annoying contrast between the background noise during the silence period and the background noise during speech bursts [1].

Furthermore, the terminating endpoint should be able to distinguish gaps due to silence from gaps that result from missing/discarded packets. Recommendation G.764 specifies that an originating endpoint set the *More-Bit* or M-bit to 1 for all packets except for the last of a burst at which time it sets it to 0. This allows a transmitter to signal whether a packet is part of a burst or is the last packet in a burst.

With the M-bit mechanism, the terminating endpoint can distinguish the absence of packets at play-out time due to silence at the source from their absence due to packet loss in transit. When it finishes playing out a packet and the next packet is not available, the terminating endpoint checks the M-bit of the last packet. If this value is 0, then the absence of a packet indicates a real pause in the transmission at the originating end of the transmitting node. If the value of the M-bit of the last packet is 1, the absence indicates a late or last packet that had not arrived. This procedure will allow the terminating endpoint to use appropriate fill procedures for the speech packet.

I.2.2 Echo cancellation

Although echo results from any impedance mismatch in the circuit, its effects are more serious with long delays. The main source of echo in analogue circuits is the 2-wire to 4-wire conversion at the connection from the customer to the local central office. In local connections, a reduction of signal/noise ratio is usually sufficient to mask the echo. For a given delay, there is an optimal amount of loss that attenuates the echo without impairing the signal itself.

In the past, echo elimination in long delay circuits was done through echo suppressors by inserting a high loss in the echo return path. Echo suppressors, however, introduce front-end clipping and are ineffective during double talk. Echo cancellation avoids these drawbacks by removing the echo from the speech.

Echo cancellers are used on two-way voice connections. At each end of the circuit, the near-end canceller generates a replica of the echo from its receive side and subtracts it from the transmitted signal. For each sample period, the residual echo is used to update the coefficients of a linear filter whose impulse response mimics the echo path.

Note that in the Public Switched Telephone Network (PSTN), echo control is needed whenever digital speech interpolation is used because the speech detector may erroneously classify echoed signals as speech and reduce the compression gain. This is in addition to the need for echo cancellation to compensate for the delay due to packetization or to avoid “freeze-out”.

I.2.3 ADPCM coding of voiceband signals

The traditional method of coding voice signals in a digital network uses 64 kbit/s Pulse Code Modulation (PCM). Adaptive Differential Pulse Code Modulation (ADPCM) algorithms take advantage of the slow variation and predictability of speech to reduce the bit rate and increase transmission efficiency. The ITU-T has adopted algorithms of Recommendation G.726 for transmission at 40 kbit/s, 32 kbit/s, 24 kbit/s and 16 kbit/s. The embedded ADPCM algorithms of Recommendation G.727 can be used for the compression of voice and voiceband data signals in digital networks. They have an advantage over non-embedded algorithms in that, in a congested network, an intermediate node can reduce the bit allocation for particular incoming channels without having to exchange control messages between the various nodes in the path of the connection.

I.2.4 Facsimile demodulation/remodulation

Many factors affect the end-to-end performance of facsimile communications over compressed facilities. Thus, to improve the overall quality of facsimile communications over Intermediate Data Rate (IDR) satellite compressed facilities, the following approaches can be used:

– increase the reliability of the IDR link either by augmenting the transmission power or by using a more robust error correction coding or both;

– enhance the operation of the Circuit Multiplication Equipment (CME) for facsimile by introducing a service-specific Forward Error Correction (FEC) coding for facsimile such as in Recommendation G.766; and

– improve the performance of the national portions of the connection.

In 1991 and 1992, AT&T and COMSAT studied the impact of IDR satellite facility errors on Group III (G3) facsimile image quality in point-to-point connections. In these tests, a simulated IDR connected two pairs of widely used CME that incorporated both the ADPCM mode and the facsimile demodulation mode for the treatment of facsimile. The main conclusions were that, with no tail impairments in the national extensions, packetized facsimile traffic according to the demodulation/remodulation procedure of Recommendation G.765 provides significant performance improvement over PCM or ADPCM in the circuit-oriented mode (without service-specific FEC) on IDR links.

Test results for error-free pages show that the performance of packetized traffic is better than circuit-ohented PCM traffic and DCME with G.766 fax remodulation and without a G.766 FEC. Furthermore, the results for severely errored pages show that packetized remodulation is superior even up to BER of 10^{-5}**.** The reason that packetized demodulation can improve the end-to-end performance is the following: the combination of the impact of an error burst on a single packet and the distributed control structure of packet systems reduces the frequency of exposure to errors and explains the improvement for packetized transmission. A G.766 DCME with G.766 fax remodulation and G.766 FEC is better than a PCME because the FEC corrects the line errors.

One solution to improving facsimile performance is to use service-specific error-correction techniques, such as the DCME's BCH coder specified for facsimile demodulation/remodulation in Recommendation G.766. This solution improves G3 facsimile quality significantly.

Another alternative to improving performance is to add a FEC coder at the line level (for example, the IDR Reed-Solomon concatenated outer-codec). This approach can improve the performance of the 2.048 Mbit/s channel for all types of traffic. This includes, in addition to facsimile, other services, such as voice, digital data, high-speed voiceband data videotelephony, ISDN, and frame relay.

The most general approach, which also causes the smallest number of operational complexities, to boost the satellite transmission power, thereby improving the satellite performance objectives and the link quality for all types of traffic and services.

I.2.5 Packetization

In a circuit-switched connection, the switching and transmission paths are dedicated to the call even if there is no traffic. The main component of delay is the fixed delay due to transmission. In packet switching, information is exchanged in the form of blocks of limited size or "packets". At the source, long messages are divided into several packets which are then transmitted across the network and then reassembled at the destination to reconstitute the original message. The allocation of bandwidth for transmission is dynamic on a link-by-link basis. Packet switching, therefore, takes advantage of the bursty nature of speech and data traffic to multiplex the traffic from different users so that they can use the transmission bandwidth and switching resources dynamically. Packetization facilitates the integration of speech and data, and should lead to a more efficient use of the available resources. Data only networks must be engineered to accommodate bursts of traffic without causing congestion; this leads to an inefficient utilization of available bandwidth. Thus, by multiplexing data traffic with voiceband traffic, which is less bursty, a more efficient use of the bandwidth can be achieved. Furthermore, packet systems accommodate networking needs more readily because the necessary control information is available in the packet header. This greater flexibility in network operation allows easier integration of multimedia traffic (speech, video, data).

To facilitate the integration of different types of traffic, the Packet Voice Protocol of Recommendation G.764 builds on ISDN Recommendations to define procedures at the physical, link, and packet layers for the unified treatment of voice, voiceband data, and digital data. The link-level protocol is derived from Recommendation 1.441/Q.921 (that is, LAPD), and is similar to Recommendation Q.922 (LAPF), which is the protocol for data-only frame relay applications.

Recommendations G.764 and Q.922 have many similarities. Transmission bursts are enveloped in frames, each with a destination address. The address is called Data Link Connection Identifier (DLCI) and is stored in an address field of the frame header. Within the network, there is no flow control by windowing or by retransmission. The general class of frames that do not require acknowledgment consists of Unnumbered Information (UI) frames. Recommendation G.764

has extended frame relay to voiceband signal and has defined a new UIH frame for packet voice. The UIH voice frame has a Cyclic Redundancy Check over octets 2-9 of the frame. These octets include the address field (to ensure correct delivery), the control field (to guarantee the validity of the frame type), and the layer 3 header that contains the *time stamp* information.

The UIH allows the packetization of voice over transmission lines that are prone to errors. This makes the transmission more robust, which is important for speech coders that are not robust to line errors like PCM and ADPCM.

The information in G.764 packets allows the terminating endpoint to reconstitute continuous speech streams and to play them out at regular intervals despite varying packet arrival times. This involves:

1) preserving the relative time of information within one talk spurt through the use of packet sequence numbers; and

2) equalizing the delay of the various packets within a speech burst through the time stamp and build-out values.

I.2.5.1 Sequence numbers

Sequence numbers in the packet header allow the endpoints to determine whether a packet was lost. In addition, the endpoints use this information in conjunction with the time stamp to remove the variability in delay between talk spurts. The first packet of a voice spurt always has the sequence number of "0"; subsequent packets in the same burst have the numbers from 1 to 15, rolling back to 1.

I.2.5.2 Time stamp

The time stamp field in the packet header records the cumulative variable delays that the packet experiences traversing the network. Each node adds the amount of time it took to serve a packet to the time stamp before sending the packet. By the time it reaches the terminating edge, the value in the time stamp field will indicate the total queueing delay in the network. The packet is then delayed before play-out so that the end-to-end delay is equal to the value of the *build-out*. The value of the build-out delay is a trade-off between accepting excessively large delays and forcing a high number of lost packets.

I.2.5.3 Build-out procedures

In a packetized speech system, *build-out* is the maximum allowable variable delay on a given virtual circuit. The terminating endpoint uses this parameter to mask the timing jitter. This jitter results from the variability in the delay that the various packets of the same permanent virtual circuits encounter in their path. Thus, the use of build-out delay improves the subjective quality of the reconstituted speech.

The terminating endpoint uses this parameter to estimate how much the first packet of a speech spurt has to be delayed. Once it has estimated the play-out time, it will place the subsequent packets in sequence order and hold them in the play-out buffer for the following duration:

time before play-out = build-out delay – time stamp value

The terminating endpoint must, therefore, store the speech packets that arrive before their scheduled play-out time and then play them at regular intervals. Packets whose time stamp field exceeds the build-out delay are considered late and are dropped. The value of the build-out is the maximum allowable delay in the transmission path. Packets that experience delays exceeding that value are discarded. In the case of voice, noise packets fill the gap of the missing packets.

This scheme guarantees that the packets will remain in sequence and does not require clock synchronization between the endpoints. Furthermore, excessive packet dropping can be used to detect network congestion and then to invoke overload management strategies.

The build-out delay is added only once at the terminating endpoint; thus, if two G.764 networks are connected back-to-back, the delay is added at the last node only. This property assists the network planner/operator in maintaining the overall one-way delay within the limit of 400 ms of Recommendation G.114.

The selection of the value for build-out is a trade-off between accepting excessively large delays and dropping packets.

I.2.6 Frame relay and packet cross-connections

A PCME is capable of networking in the packet domain by cross-connecting packets. This feature is similar to the digital cross-connection capability except that it applies to the packet domain instead of the traditional circuit domain. This is the same feature that is used to frame relay digital data described in Recommendation Q.922.

However, there are some differences between cross-connections of G.764 frames and the frame relaying of Q.922 LAPF frames. These differences are at the physical, link, and packet layers as explained in I.2.6.1, I.2.6.2, and I.2.6.3.

I.2.6.1 At the physical layer

According to 3.1.1/G.764, bit inversion is needed to ensure adequate transport over primary applications that require code restrictions by maintaining the one's density requirements. This is not a requirement in Recommendation Q.922.

I.2.6.2 At the logical (link) layer

As explained earlier, Recommendation G.764 utilizes the Unnumbered Information with a Header check or UIH frame to allow the CRC check to be calculated over some part of the frame. Recommendation Q.922 specifies a check over the whole frame. If the CRC check is computed over the whole frame as in LAPF, any bit error will cause the dropping of the entire frame. In this case, the application of packetized speech becomes restricted to cases where the error rate is very low. This restriction would exclude the application of the protocol from many public and private networks worldwide, unless some degradation is accepted.

Recommendation G.764/G.765 specifies a 13-bit DLCI as a mean of addressing frames. The frame relay protocols use a 10-bit DLCI because three bits of the second octet have been used for forward and backward congestion indication and for marking frames. Therefore, when a PCME interfaces with a data-only frame relay network, some address translation must take place.

Each frame flowing in a PCME is identified by the couple {packet stream, DLCI}. A packet stream consists of packets flowing on several predefined 64 kbit/s time slots that are not necessarily contiguous but are on the same outgoing primary rate line. Before merging with a packet stream, the incoming full-rate (channelized) traffic may arrive on different access primary rate lines and on non-contiguous 64 kbit/s time slots. A DLCI identifies a given destination point for a given packet stream.

I.2.6.3 Layer 3

The issue of delay time variability is important for voice application. Recommendation Q.922 does not include this mechanism; therefore, the interface of a PCME and Q.922 devices should ensure that there is no large delay variability in the data-only frame relay network.

I.3 Performance

This subclause explains how PCME is used to achieve graceful degradation under overload.

I.3.1 Congestion control

In the following discussion, a *packet stream* consists of packets flowing on several predefined 64 kbit/s time slots that are not necessarily contiguous but are on the same outgoing primary rate line. Before packetization, the incoming full-rate (channelized) traffic may arrive on different access primary rate lines and on non-contiguous channels.

There are two types of congestion control mechanisms in a PCME:

a) block dropping for speech traffic; and

b) Dynamic Load Control (DLC).

The basic idea of block dropping is to use embedded algorithms. These are algorithms whose code words consist of a core block and several enhancement blocks. The core blocks are used both in the encoder and the decoder, while the enhancements blocks are used to reduce the quantization noise in the reconstructed signal. Core blocks must reach the decoder to avoid mistracking, but the enhancements blocks can be blocked from packets when overload conditions occur.

Bandwidth enforcement relies on dropping packets that have exceeded a user's allocated bandwidth. This is used in the data-only frame relay protocol of Recommendation Q.922 through the use of the DE bit.

Finally, the basis of DLC is to send signals to a switch with DLC capability, that is, to prevent the origination of new calls at the access side during congested periods. The signals may be as specified in Recommendation Q.50 or in one of several proprietary formats.

A tutorial on embedded ADPCM can be found in the Appendix to G.727. Block dropping is explained in more detail in the Appendix to G.764. Subjective investigations have demonstrated that with this scheme voice quality degrades gracefully under overload. In addition, the traffic smoothing effects and capacity advantage of this approach appear when the end-to-end performance over tandem links is compared with the traditional methods of tandem encoding.

I.3.2 Traffic smoothing effects of block dropping

The combination of packets from various voice sources with speech detection leads to high burstiness due to the correlations between successive interarrival times in the superposition stream. Without block dropping, these correlations tend to increase queueing delays and packet losses.

Simulation, analytical results, and live subjective testing have shown that block dropping during congestion speeds up the packet service rate and smoothes the burstiness of superposed packet voice traffic. The results have shown a dramatic improvement in capacity.

I.3.3 Performance of facsimile on IDR links

In this study, the criteria of Recommendations E.451, E.452, E.453 and E.456 are used to classify the pages. A page is called *severely errored* if it contains one or more of the following:

- twelve or more degradation events;
- three or more minor errors;
- one or more major errors.

A page *is errored* if it contains one or more degradation events, but is not severely errored.

A page *is error-free* if it contains no degradation events.

Table I.1 lists the number of facsimile test pages transmitted through DCME (circuit) or PCME (packet) that were analyzed for each of the test conditions. The goal was to send enough pages to have at least 100 error events strike the image transmissions for each test case.

Tables I.2 and I.3 list the percent of pages that were error-free and severely errored for each test condition.

Figures I.1 and I.2 plot the data in Tables I.2 and I.3 for error-free and severely errored pages, respectively.

Note that the percent error-free pages versus BER curves for all three methods of non-packetized transmission were very close.

TABLE I.1/G.765

Number of pages analyzed for final results

IDR BER	1×10^{-4}	3×10^{-5}	1×10^{-5}	3×10^{-6}	1×10^{-6}	3×10^{-7}	Total
PCM (circuit)	100	457	499	727	966	1381	4130
PCM (packet)	101	508	598	1099	1044	1200	4550
ADPCM (circuit)	74	196	401	415	344	638	2068
ADPCM (packet)	33	605	597	1276	1349	1198	5058
Remod w/o FEC (circuit)	75	400	278	499	458	774	2484
Remod (packet)	75	381	499	1280	1105	1298	4638

TABLE I.2/G.765

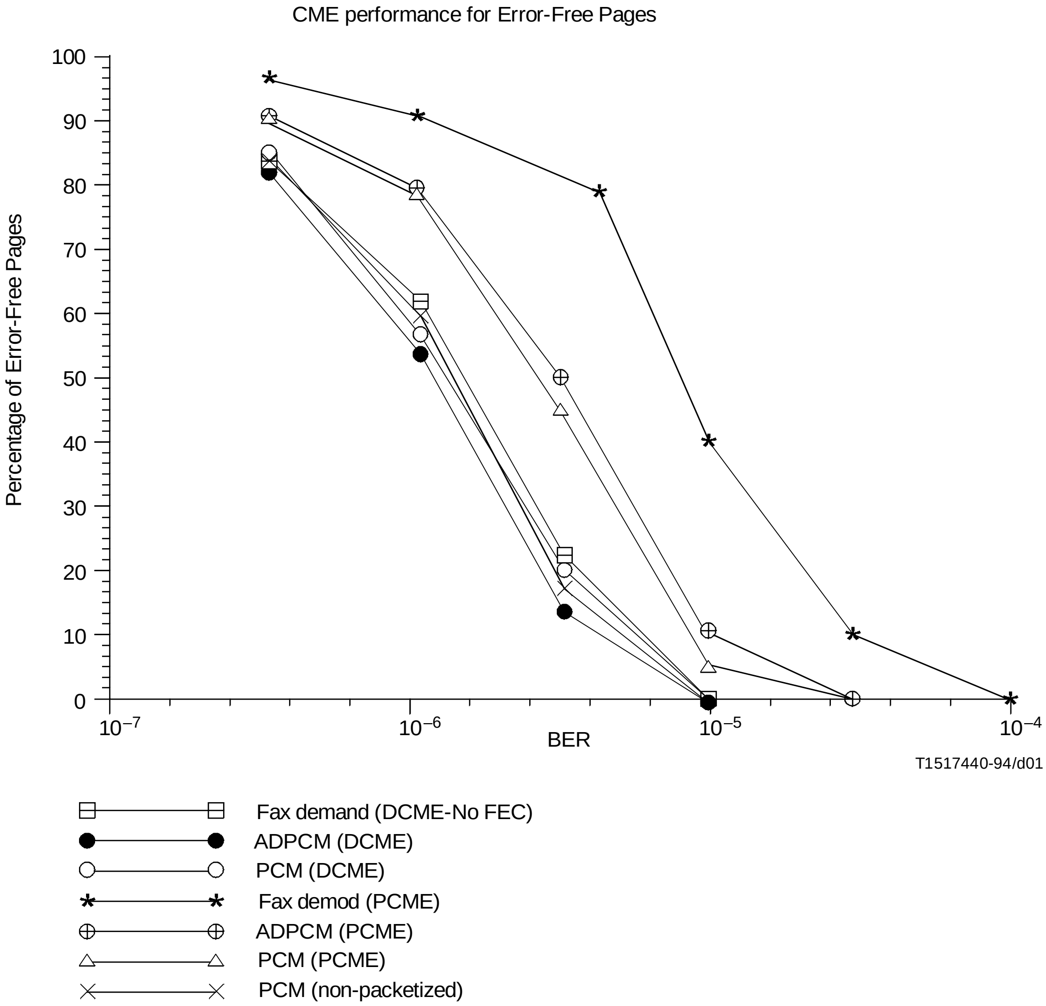
Percent error-free test pages for test data

IDR	PCM (%)		ADPCM (%)		Remodulation w/o FEC (%)	
BER	(circuit)	(packet)	(circuit)	(packet)	(circuit)	(packet)
3×10^{-7}	84.8	93.4	83.5	93.2	84.9	97.5
1×10^{-6}	59.2	76.8	52.9	77.5	62.0	90.9
3×10^{-6}	17.1	45.8	14.0	48.1	22.9	79.3
1×10^{-5}	0.2	6.0	0.0	11.1	0.4	43.3
3×10^{-5}	0.0	0.2	0.0	0.3	0.0	11.3
1×10^{-4}	0.0	0.0	0.0	0.0	0.0	1.3

TABLE I.3/G.765

Percent severely errored test pages for test data

IDR	PCM (%)		ADPCM (%)		Remodulation w/o FEC (%)	
BER	(circuit)	(packet)	(circuit)	(packet)	(circuit)	(packet)
3×10^{-7}	0.0	1.6	0.0	1.6	0.0	0.0
1×10^{-6}	0.0	6.4	0.0	6.4	0.0	0.0
3×10^{-6}	0.0	16.0	0.0	16.9	0.0	0.0
1×10^{-5}	0.4	54.7	3.3	49.9	0.4	0.0
3×10^{-5}	84.7	96.9	92.9	92.4	72.4	1.6
1×10^{-4}	100	100	100	100	100	32.0



NOTE – DCME with G.766 facsimile demodulation and remodulation and G.766 FEC were not included in this particular test since it offers virtually error-free service in this range of BER values.

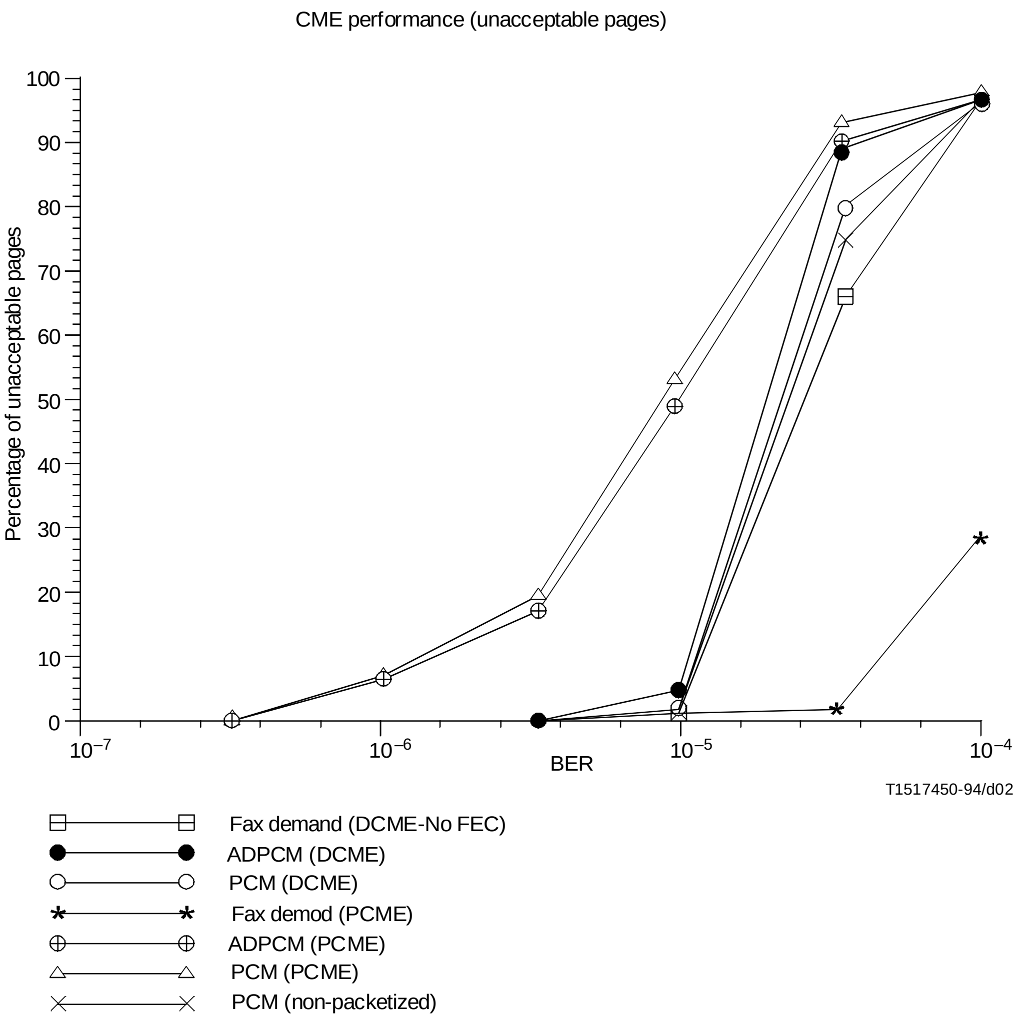
FIGURE I.1/G.765

Data plotted from Table I.2

Figure I.1 shows the measured results for percent error-free pages as a function of IDR BER for each of the test conditions. This figure shows that for every BER value tested, the packetized method of transmission resulted in a higher percentage of error-free pages than did the non-packetized methods of transmission (without the service-specific FEC). This includes the reference condition of PCM, DCME with ADPCM coding, and DCME with facsimile remodulation without FEC. This observation is valid for all cases of packetized traffic: PCM, ADPCM, and packetized remodulation traffic.

Figure I.2 shows that packetized remodulation performs better than circuit-oriented traffic and packetized PCM and ADPCM for the cases of severely errored images in the range $10^{-5} \leq \text{BER} \leq 10^{-4}$. Packetized PCM and ADPCM have higher levels of severely errored pages, especially when the BER is in the range $3 \times 10^{-7} < \text{BER} < 3 \times 10^{-5}$. This is due to error multiplication when the packets are lost.

The performance of the PCME is better as far as facsimile on IDR links is concerned due to the unique characteristics of the packet mode of transmission, as explained below.



NOTE – DCME with G.766 facsimile demodulation and remodulation and G.766 FEC were not included in this particular test since it offers virtually error-free service in this range of BER values.

FIGURE I.2/G.765

Data plotted from Table I.3

Measurements have determined that the typical burst length for a 2048 kbit/s IDR link ranges from about 20 to 40 bits with an average of 27 bits. Therefore, in packetized traffic, an IDR error burst typically affects only a single packet, that is, a single channel of transmission, whereas with circuit-oriented traffic several channels are disturbed.

With facsimile demodulation, when a packet is dropped, the terminating PCME can be designed to inject T.4 line-fill characters to prevent the receiving V.29 from hanging up and to minimize the effects of errors. If the dropped packet contained line-fill characters, no degradation occurs. When packet dropping introduces some degradation, the concentration of errors due to the burstiness of IDR errors is such that, in most cases, only one or two adjacent scan lines are affected.

An errored scan line affects the facsimile image quality in the same manner, no matter how many bit errors it contains. Thus, for the same number of total bits in error, the concentration of errors into one or two adjacent scan lines provides better performance than when the errors are scattered over many scan lines, each with fewer numbers of bits in error per scan line.

A simplified explanation is:

1) because demodulated and packetized facsimile uses less bearer bandwidth (relative to packetized ADPCM or PCM), its exposure to bursts of errors is reduced even further; and

2) the loss of packets containing demodulated facsimile has confined effects, whereas the loss of ADPCM packets causes the loss of the synchronization of the ADPCM decoder and/or loss of carrier for the modem. The reduced exposure to line errors in packet systems with facsimile demodulation and the less persisting effects of packet loss explain the dramatic improvement for packetized transmission depicted in Figures I.1 and I.2.

I.4 Planning and dimensioning

I.4.1 General

This subclause contains traffic load information to be used in the traffic engineering of PCME. Traffic curves and formulas are presented with the purpose of assisting on the optimization of bandwidth usage. Different traffic mixes, packet stream sizes, voice activity factors, and compression gain combinations have been used to generate average bits per sample data.

I.4.1.1 PCME gain operation

PCME achieves bandwidth gain by applying different signal processing algorithms on a per-channel basis. These algorithms are based on the detection and interpolation of channel activity, the classification of signal type and speed, the codification at different PCM/ADPCM rates, and the modulation of G3 facsimile data.

Because of the forward-acting nature of packet communications, where no handshake is required between nodes for the allocation of bandwidth, the gains obtained by applying different bandwidth-savings techniques are direction independent. In addition, the freeze-out effect is eliminated.

I.4.1.2 Digital Speech Interpolation (DSI) gain

The detection of energy and the interpolation of the information detected, along with other information submitted to similar processing, allow for gaps in transmission (of voice, voiceband data, facsimile) to be filled with traffic from other channels. The specific gains due to DSI are dependent on voice and voiceband equipment transmission patterns. In PCME, bandwidth allocation is direction independent; it is determined by the energy detector at each PCME endpoint.

I.4.1.3 Signal type and speed classification – Gain implications

The real-time classification of signal type (whether voice, G3 facsimile, or voiceband data) and, if voiceband data, the classification of signal speed, allows for the traffic to be processed optimally. The combination of these algorithms not only provides for bandwidth savings but also for the proper transport of traffic, therefore preserving information integrity. The gains through classification are obtained through the specific assignment of PCM/ADPCM coding rates and the demodulation of G3 facsimile.

I.4.1.4 PCM/ADPCM codification rate gain

Once the signal type and speed are classified, the signals are assigned a coding rate. The rate is programmable on a per-channel, per-signal-type, and speed basis. For voice, embedded ADPCM with 4 bits/sample and the capability of dynamically going down to 3 and 2 bits/sample under periods of congestion, is most commonly used. For voiceband data, coding at non-droppable bit rates of 8, 5, 4, 3, and 2 bits/sample are used to optimize bandwidth usage and signal quality. Codification gains depend on signal speeds and signal speed coding rates.

I.4.1.5 G3 facsimile modulation gain

For G3 facsimile traffic, the signal may be modulated to its original digital speed. This procedure allows for the growing amount of facsimile traffic to assume compression ratios similar to those of voice; without modulation, the amount of bandwidth taken by facsimile traffic would lower the amount of bandwidth available for voice, therefore reducing the quality of the voice.

I.4.2 PCME traffic engineering

The traffic engineering information described in this subclause applies to the five major PCME applications:

1) point-to-point;
2) thin route;
3) multidestination;
4) cascade;
5) transit traffic.

Up to eight compressed links may be simultaneously transmitted from a PCME terminal. The compression treatment onto each link is totally independent of the others.

The purpose of these graphs is to allow calculation of the gain for a PCME for a given set of fixed-traffic requirements and a given level of performance.

The voice gain in the following curves (see Figures I.3, I.4 and I.5) is defined as the number of access 64 kbit/s channels that would be required to transport the desired traffic without the PCME, divided by the number of time slots that are configured for the packet stream of the PCME that is transporting the traffic.

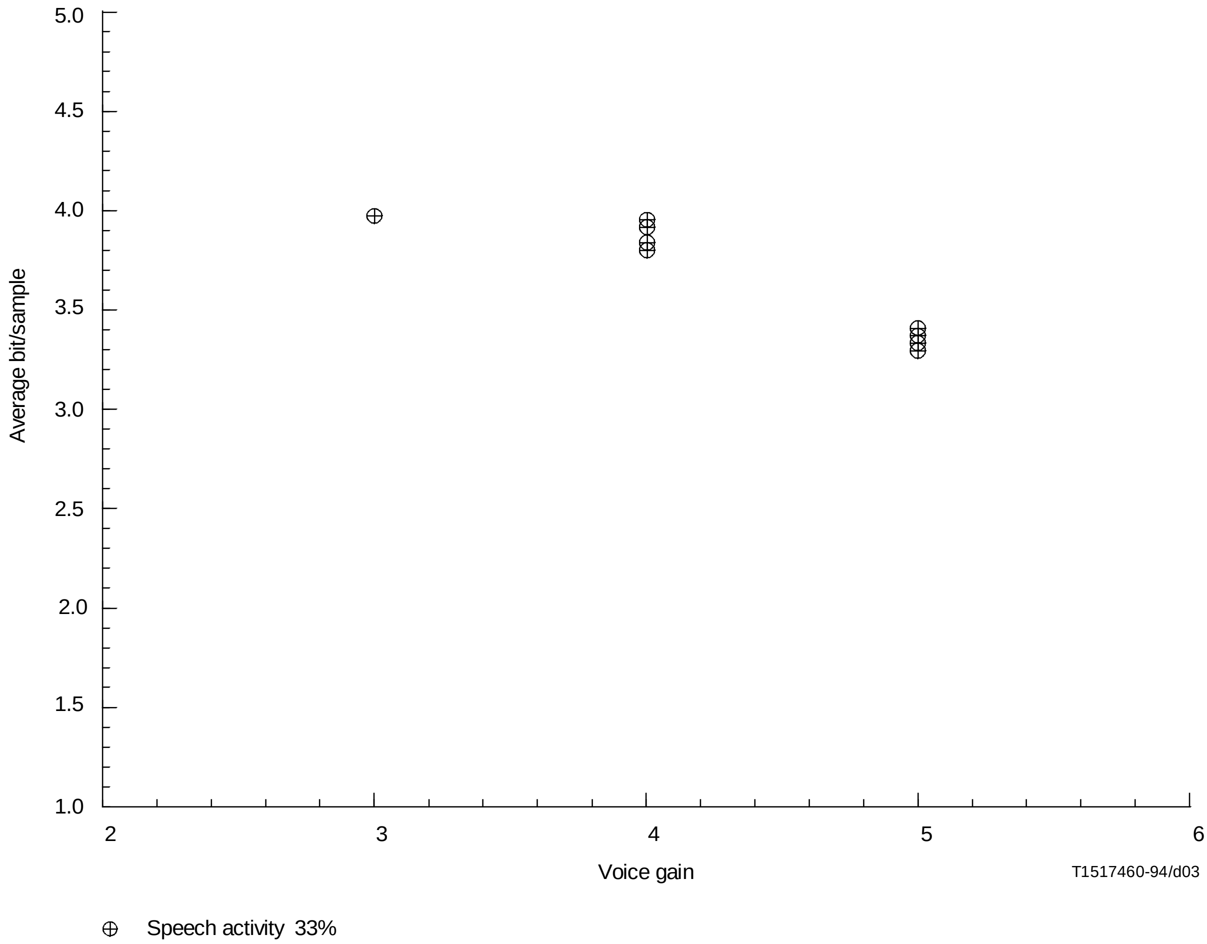


FIGURE I.3/G.765

Speech activity 33%

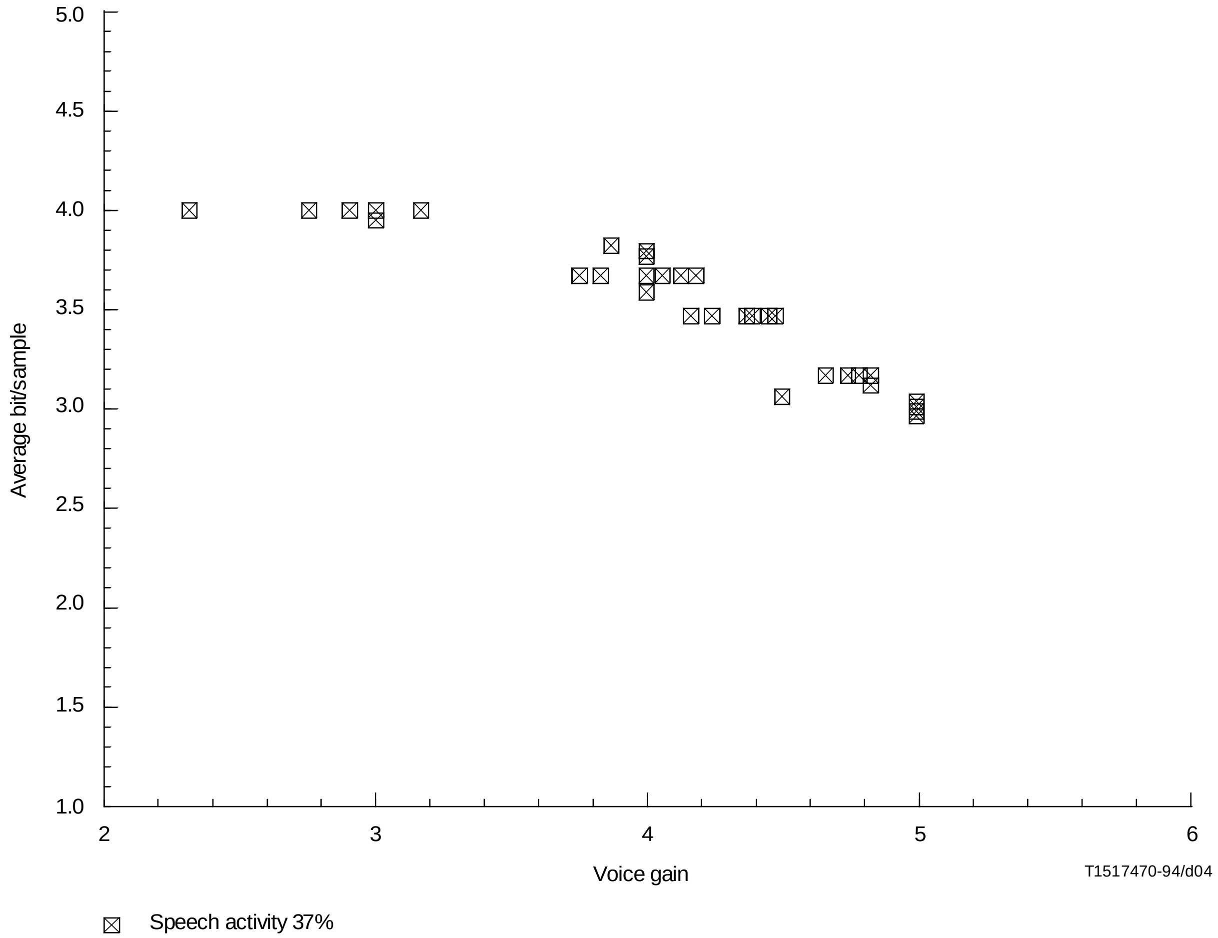


FIGURE I.4/G.765

Speech activity 37%

The definition of *performance level* is subjective as it relates to telephone users. For DCME, this subjective performance has been mapped into the quantitative parameters of average bit per sample and freeze-out (see Recommendation G.763). Because of the difference in the technology, this mapping is different for PCME. Also, because PCME rarely produces clipping or freeze-out, PCME can be configured with a significantly lower average bit/sample than conventional DCME (that is, the proprietary DCME or those that do not conform to the latest revision of Recommendation G.763) and yet have equal or better subjective performance.

In the following curves, the voice bandwidth on a packet stream is calculated by subtracting the total bandwidth used for any preassigned channels and expected voiceband data calls from the total bearer bandwidth of the packet stream.

The average bit/sample has been experimentally measured over a large number of conditions and different voice activity settings. In all cases, the setting was adjusted for no freeze-out.

NOTE – The simulator setting for the voice activity is generally less than the actual voice activity in the PCME; however, the exact difference is manufacturer-dependent and should be calibrated for each equipment.

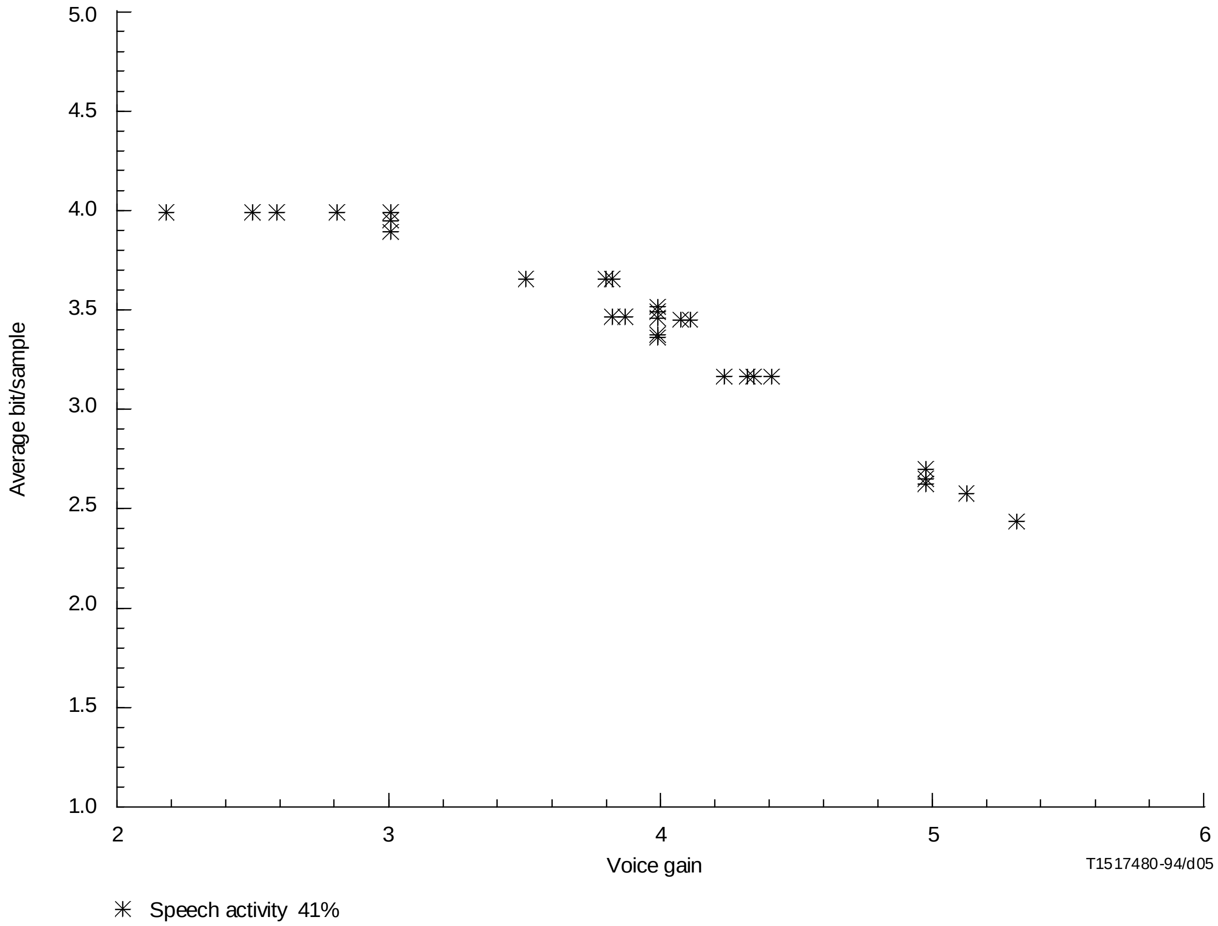


FIGURE I.5/G.765

Speech activity 41%

I.4.3 Bandwidth usage at various coding rates

In engineering network, it is important to estimate the maximum bandwidth usage at the various voiceband data coding rates. In making that estimation, it is essential to include the overhead due to the G.764 header (10 octets), HDLC bit stuffing, which on the average increases the packet length by a factor of 1.016, and during periods of continuous frame generation, the preceding and succeeding HDLC flag. Therefore, the required bandwidth to transport a user's information field of I octets is calculated as follows:

$$\text{Required bandwidth} = \frac{[(I + 10)(1.016) + 2]}{\text{packetization time } \tau} \text{ octets/per frame} \times 8 \text{ bits per object}$$

$$= \frac{8.128I + 97.28}{\tau} \text{ kbit/s}$$

The packetization time τ is 16 ms for voiceband data and 20 ms for DICE, VDLC and facsimile demodulated data. This expression shows that the effect of overhead increases when the information field decreases.

Therefore, the correspondence shown in Table I.4 can be made:

TABLE I.4/G.765

Correspondence of bandwidth variables

Traffic type	Transmission method	Information field I in octets	Packetization time τ in ms	Required bandwidth in kbit/s
64 kbit/s	Clear	128	16	71.1
9.6 kbit/s digital data	DICE	28	20	16.2
9.6 kbit/s voiceband data	40 kbit/s ADPCM	80	16	46.7
9.6 kbit/s facsimile	Facsimile demod/remod	24	20	14.6
7.2 kbit/s voiceband data	40 kbit/s ADPCM	80	16	46.7
7.2 kbit/s facsimile	Facsimile demod/remod	18	20	12.2
4.8 kbit/s voiceband data	32 kbit/s ADPCM	64	16	38.6
4.8 kbit/s facsimile	Facsimile demod/remod	12	20	9.74
2.4 kbit/s voiceband data	24 kbit/s ADPCM	48	16	30.46
2.4 kbit/s facsimile	Facsimile demod/remod	6	20	7.3
2.4 kbit/s voiceband data	24 kbit/s ADPCM	64	16	38

I.4.4 Average bit/s vs. traffic load

The graphs described in the previous paragraphs can be used to estimate the gain and the average bit/s for given traffic loads. The principle is as follows: for a given packet stream bandwidth and for a given activity factor and average bit per sample, we find the curve that corresponds to the packet stream width and the given activity factor. From these curves we determine the number of speech channels that correspond to the required average bit/sample. The required bandwidth for speech is then calculated.

NOTE 1 – All these curves where derived with zero packet dropping (that is, zero freeze-out).

For a given specific loading pattern, we estimate the total bandwidth requirements for the clear channels, voiceband channels, and facsimile channels. Under the assumption that all the facsimile channels are active at the same time, we then subtract all this bandwidth and estimate the bandwidth available for speech for the worst-case condition. Using the curves, we then estimate the worst gain under conditions of zero packet dropping (that is, zero freeze-out).

NOTE 2 – The gain calculated in this fashion is an extremely conservative number. Also, these rules do not apply other than in point-to-point situations.

For more complex networks or if more accurate estimates are needed, modeling tools should be used to account for the instantaneous variation in the coding rates for speech. One such tool is the Comnet II.5 simulation packet that is produced by CACI Products company. This product generates models of PCME networks and simulates their performance. The results of these simulations can be used to optimize network designs.

I.4.5 Build-out settings vs. traffic load

This paragraph contains guidelines for selecting the build-out value for voice traffic for various loading conditions and various packet stream sizes. The values are selected so that no packet dropping occurs (that is, 0 percent freeze-out).

To select the value of the build-out, we estimate the worst-case queueing delay for the network configuration at hand. In general, the selection of the value for build-out is a trade-off between accepting excessively large total delays and dropping packets. In circuit-emulation mode, the build-out delay is typically selected to be higher for digital data traffic than for voiceband traffic.

The value of the build-out depends on several factors, including the buffer length, the block dropping thresholds, the service time assigned for the voice queue and the digital data queue, the traffic mix on the voiceband data channel, the speech activity factors, and the packet stream width. In practice, 20 to 30 ms per hop were found for voiceband traffic, increasing to 50 to 70 ms when facsimile demodulation/remodulation is used. This additional delay takes into account the processing time for facsimile demodulation and for the timing constraints imposed by the T.30 protocol.

I.4.5.1 A simple formula

In the following we assume that the voiceband queue is used only (that is, there is no digital data traffic). This assumption is valid in international gateway applications. Let D_4 denote the service time for a frame containing a voice packet with 4 ADPCM bits per sample. The total number of octets with a full voice packet consists of 64 octets of speech, 10 octets of header and trailer information, 1 octet of bit-stuffing that takes place within the frame, and an octet of HDLC flag between the frames. Therefore:

$$D_4 = \frac{76 \text{ octets/pkt} \times 8 \text{ bits/octet}}{N \times 64 \text{ kbit/s}} \approx \frac{9.5}{N} \text{ ms/pkt}$$

where:

$N \times 64$ kbit/s is the bandwidth for the packet stream ($1 \leq N \leq 31$) and 64 kbit/s is the bandwidth of a time slot.

Let D_3 denote the service time for a frame containing a voice packet with 3 bits per sample, after dropping the block containing the least significant bits of each of the 128 samples in the packet due to congestion. The total size of the frame is now 60 octets. Therefore:

$$D_3 = \frac{60 \text{ octets/pkt} \times 8 \text{ bits/octet}}{N \times 64 \text{ kbit/s}} \approx \frac{7.5}{N} \text{ ms/pkt}$$

Finally, let D_2 denote the service time for a frame containing a voice packet with 2 bits per sample, after dropping the block containing the second least significant bits of each of the 128 samples in the packet due to congestion. The total size of the frame is now 44 octets. Therefore:

$$D_2 = \frac{44 \text{ octets/pkt} \times 8 \text{ bits/octet}}{N \times 64 \text{ kbit/s}} \approx \frac{5.5}{N} \text{ ms/pkt}$$

Let B_1 20 and $B_2 = 40$ denote the block-dropping thresholds (in packets), while the total buffer length is 140 octets. A reasonable estimate for the build-out delay is therefore:

$$\text{Build-out} = D_4 \times B_1 + D_3 \ (B_2 - B_1) + D_2 \times (140 - B_2)$$

This estimate is calculated for various types of coding algorithms under the condition that no packet dropping should occur. The estimate should be increased by about 35 ms when facsimile demodulation and remodulation is used.

I.4.5.1.1 Build-out estimates with (4,4) embedded ADPCM

In this case, there is no block dropping so that the coding is 32 kbit/s. ADPCM is used and the only response to congestion is to drop the packet. The following gives the build-out estimate for various packet stream sizes of N × 6.4:

Packet Stream Size N	6	8	16	24	30	31
Build-out (ms)	31.7	23.8	11.9	7.92	6.3	6.1

I.4.5.1.2 Build-out estimates with (4,3) embedded ADPCM

In this case, one block can be dropped for congestion control. The build-out value is chosen so that no packet is dropped due to excessive delays. The following gives the build-out estimate for various packet stream sizes of N × 64:

Packet Stream Size N	6	8	16	24	30	31
Build-out (ms)	56.7	42.5	21.25	14.17	11.3	11.0

I.4.5.1.3 Build-out estimates with (4,2) embedded ADPCM

In this case, two blocks can be dropped for congestion control. The build-out value is chosen so that no packet is dropped due to excessive delays. The following gives the build-out estimate for various packet stream sizes of N × 64:

Packet Stream Size N	6	8	16	24	30	31
Build-out (ms)	148.3	111.25	55.6	37.1	29.7	28.7

I.4.5.2 Experimental validation

The measurements were made on an actual system using the same parameters as in the calculation. The voice loading and the bandwidth used for various speech activity factors were varied. The minimum build-out value was selected to ensure that no packet dropping would occur. The measurements were for two voice-gain settings (4:1 and 5:1) at different speech activity factors and for various packet stream widths. See Table I.5 and also Figures I.6 and I.7.

TABLE I.5/G.765

Build-out with voice-gain settings (4:1, 5:1)

Gain	Activity factor	Build-out for various packet stream bandwidth kbit/s					
		6 × 64	8 × 64	16 × 64	24 × 64	30 × 64	31 × 64
4:1	33%	73	46	24	16	13	13
	37%	82	48	25	18	15	14
	41%	96	58	26	18	15	15
5:1	33%	136	102	33	20	20	17
	37%	146	109	43	22	20	19
	41%	152	114	53	28	28	25

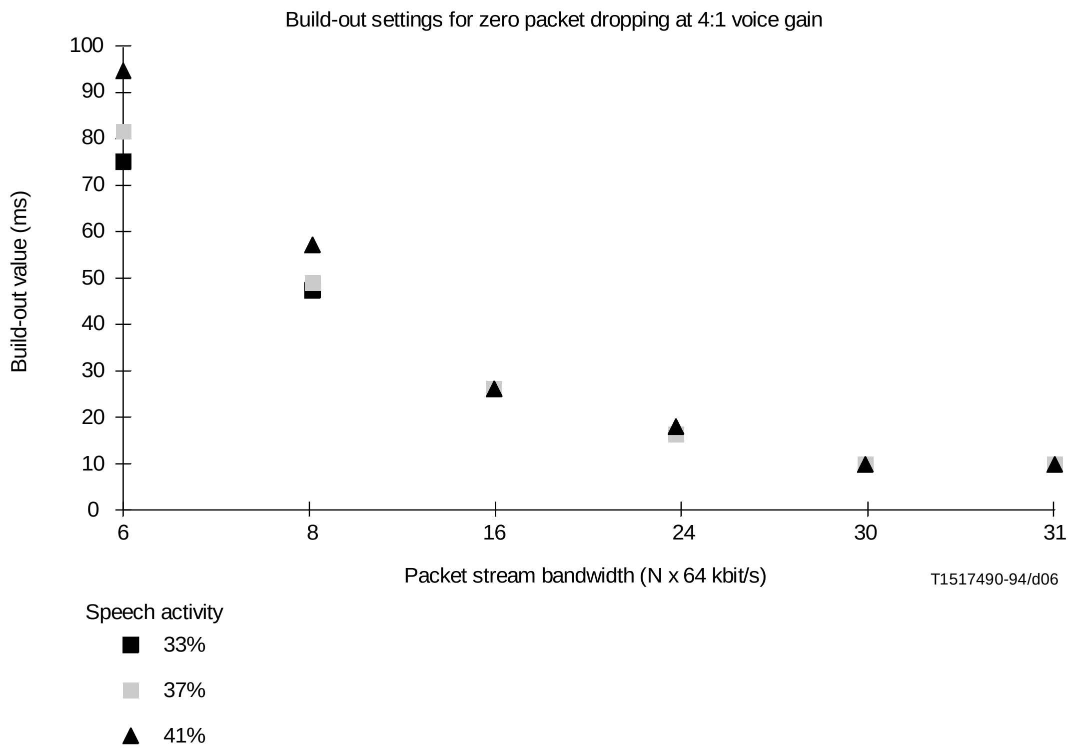


FIGURE I.6/G.765

Build-out settings (4:1 voice gain)

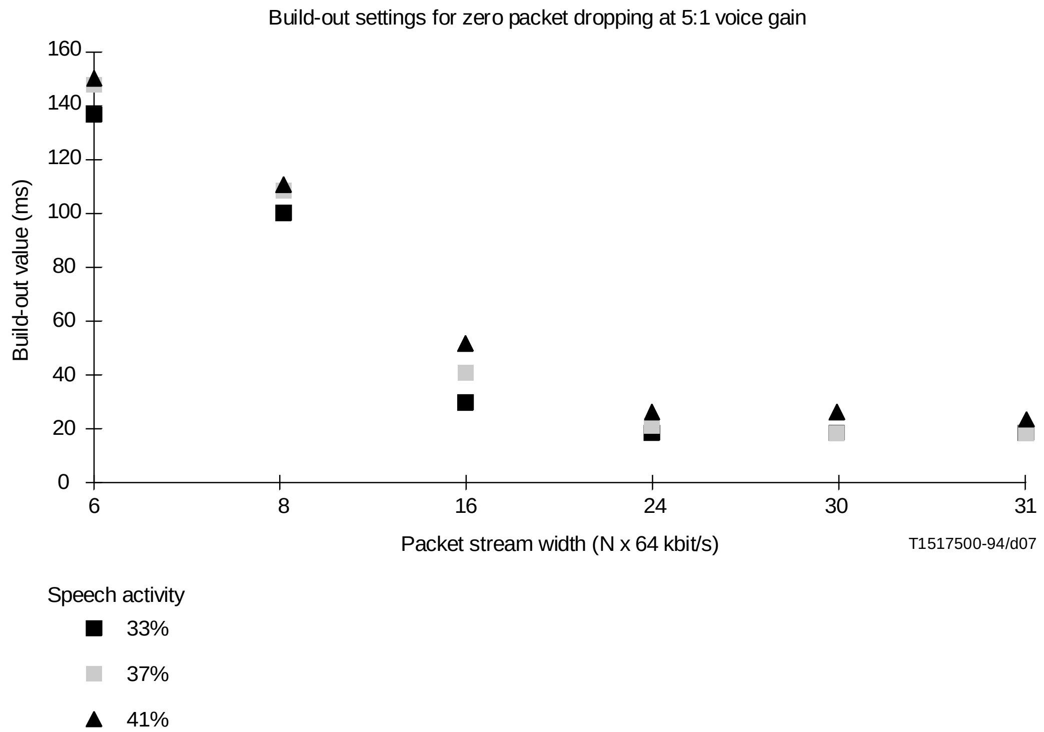


FIGURE I.7/G.765

Build-out settings (5:1 voice gain)

I.4.5.3 Discussion

The experimental results confirm the calculation as well as the field results. For 4:1 gain, the build-out value of 20 ms is sufficient for packet streams of 24 time slots or larger. To cover packet stream sizes of 16 or larger, the build-out value for 30 ms should be used. The value of 30 ms is also sufficient for voice gains of 5:1 for the packet stream sizes of at least 24 time slots.

For smaller packet stream sizes, the value of the build-out becomes larger for the same gain and if the same system parameters (block dropping thresholds, queue lengths, etc.) are used as for larger packet stream sizes.

I.5 Installation and acceptance

I.5.1 A possible configuration map for PCME

A PCME can be configured in several ways. The following is a proposed configuration map for a PCME compressing eight incoming 30-channel trunks onto two bearer packet streams.

Table I.6a) is for when time slot 16 does not carry user traffic (for example, it carries signalling).

TABLE I.6a)/G.765

From		To	
Trunk	Time slot	Packet stream	DLCI number
1	1 to 15 and 17 to 31	1	402 to 460
2	1 to 15 and 17 to 31	1	462 to 520
3	1 to 15 and 17 to 31	1	522 to 580
4	1 to 15 and 17 to 31	1	582 to 640
5	1 to 15 and 17 to 31	2	642 to 700
6	1 to 15 and 17 to 31	2	702 to 760
7	1 to 15 and 17 to 31	2	762 to 820
8	1 to 15 and 17 to 31	2	822 to 880

Table I.6b) is for when time slot 16 does carry user traffic.

TABLE I.6b)/G.765

From		To	
Trunk	Time slot	Packet stream	DLCI number
1	1 to 31	1	402 to 462
2	1 to 31	1	464 to 524
3	1 to 31	1	526 to 586
4	1 to 31	1	588 to 648
5	1 to 31	2	650 to 710
6	1 to 31	2	712 to 772
7	1 to 31	2	774 to 834
8	1 to 31	2	836 to 896

Other standard configurations are for further study.

I.6 Examples of applications

I.6.1 Point-to-point application

A PCME is ideally suited for facility-relief type of applications where the compression and dynamic bandwidth allocations are needed to:

a) reduce transmission costs;

b) offer interim relief;

c) satisfy temporary needs (for example, conventions or emergencies).

I.6.2 Point-to-multipoint application

I.6.2.1 Transient traffic applications satellite networks

Applications of a PCME in satellite networks rely on two advantages:

1) compression;

2) cross-connections of packets.

In the international satellite communications environment, links are typically established between the public switched telephone networks of two or more countries (although private network links are also available). There are two basic interconnections arrangements:

1) point-to-point links between two locations;

2) multi-destinational links serving several locations.

A PCME point-to-point system would be functionally equivalent to a point-to-point DCME link. Multi-destinational is particularly easy for packet systems since the address information is contained in the header of each packet. Because individual control channels are not needed for each destination, there is no limit to the number of destinations in a packet-based multi-destinational configuration.

A PCME can also be used to reduce the facilities needed between the satellite earth station and the associated switch (the so-called "back-haul" problem). The problem is totally solved by using the packet cross-connect capability of a network of PCMEs with one PCME placed next to the earth station.

I.6.3 Transient traffic applications

To take advantage of diurnal traffic non-coincidence, traffic between several areas with disparate time zones can be merged without requiring that the maximum transmission capacity be equal to the sum of the maximum transmission between each destination. In a network of PCMES, the traffic can transit without requiring successive transcoding. Using PCME for re-routing existing calls, PCME can offer a method to redirect existing calls from one transmission link onto another transmission link on the basis of the packet stream backup procedure described in clause 15/G.765.

The objective is to avoid the *camp-on* problem for calls with long holding times and still protect all existing traffic. This allows optimum usage of the transmission links, because a single, long-duration call can no longer block an entire link having a capacity of 120 calls for the duration of the camp-on.

As defined in Recommendation G.765, a *packet stream* is a collection of logical links multiplexed together onto one physical channel between two nodes of a wideband packet network. Recommendation G.765 defines automatic backup to mean backing up and restoring packet streams on different physical links, while keeping both ends synchronized. Backup can be automatically initiated after facility failures or due to scheduled changes to ensure optimal utilization of transmission resources. The effect of each of those backup events on the traffic is slightly different. In backups after failures, the traffic will inevitably be interrupted for a brief moment at least equal to the required, standard alarm-detection interval. Although this may go unnoticed in voice calls, the effects on voiceband data connections may be visible. In contrast, during call redirection for optimal utilization of transmission resources, it is possible to implement the backup without interruption of voice traffic. This is because one PCME will be using the backup physical link for transmission in one direction while the other PCME will still use the original physical link for the other direction.

I.6.3.1 **Call redirection** in **clusters of PCME**

In the case of a coupled cluster of PCME, the above method can be extended to re-route in-progress calls in a predefined order. Figure I.8 shows a cluster of four PCMEs, denoted as 1A through 4A, arranged in a cluster in one side connected to a matching cluster of PCME 1B through 4B. Each PCME cluster is formed via a simple daisy-chain interconnection of individual PCME. Each PCME receives traffic from several incoming primary-rate channels. For simplicity, it is assumed that the output of a PCME is a single packet stream that spans the entire capacity of a primary link (that is, a packet stream = a bearer). In general, the bandwidth of a packet stream may vary from a few time slots to the full primary rate, and a PCME may have several output physical links.

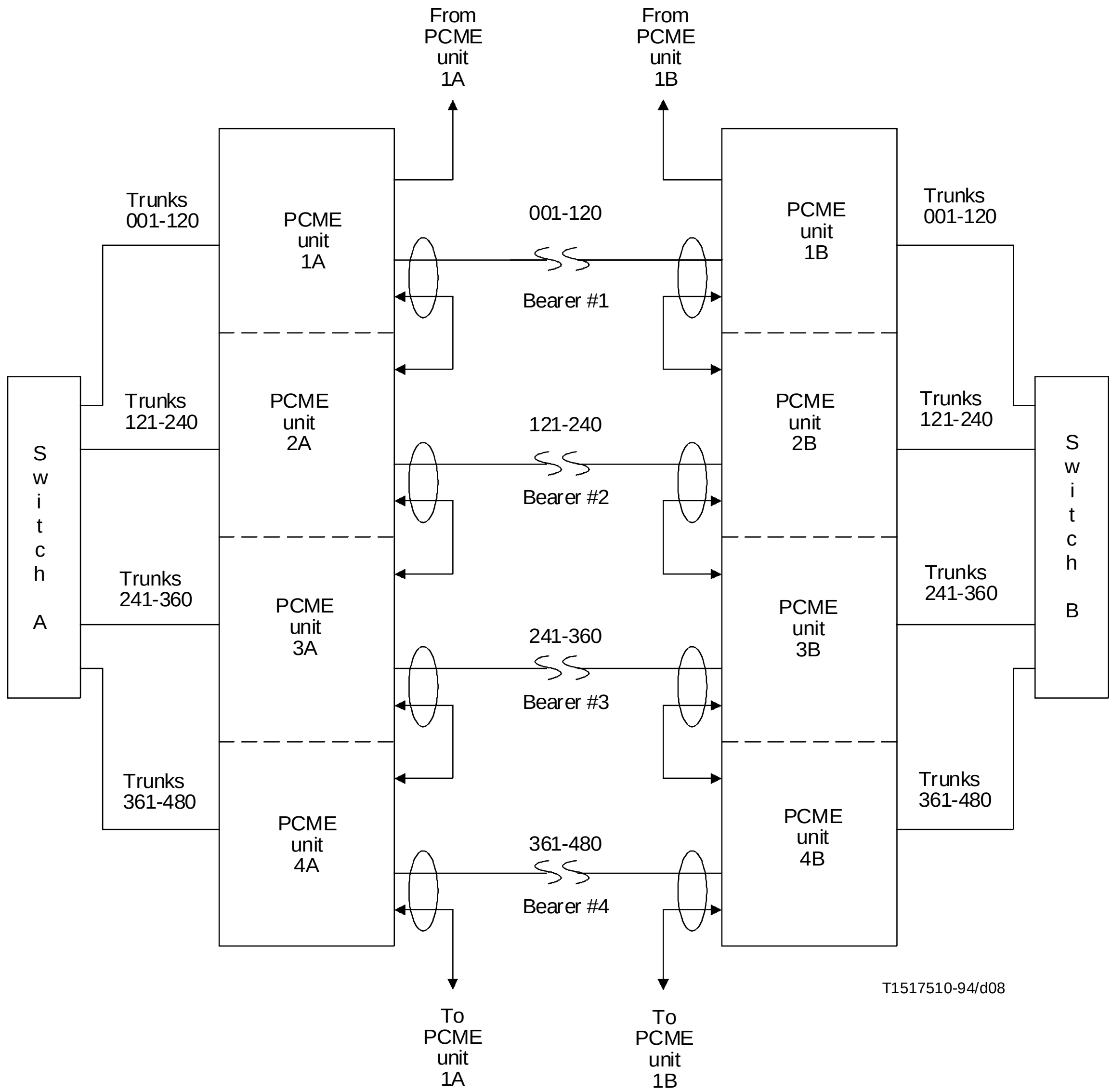


FIGURE I.8/G.765

Cluster of four PCMEs

Initially, the PCAU will packetize the incoming traffic and send it on the associated bearer. When the packet stream backup is started, the frames will be relayed onto the backup bearer that is physically connected to the next adjacent PCME. At this next adjacent PCME, one of two cases can take place:

– If the bearer associated with the adjacent PCME is not currently backed up, the relayed frames will flow onto that bearer along with that bearer's normal traffic.

– If the bearer is also backed up, the relayed frames, together with the next bearer's normal traffic, will be forwarded to the next backup bearer that is physically connected to the next adjacent PCME.

This frame-relaying will continue in a daisy-chain fashion. Frames will be relayed from one PCME to the next until they reach the PCME whose packet link is designated as the backup link for the original link. To ensure that the traffic ends up at its destination at the far end, the address space must be divided into disjointed sets. Thus, the incoming frames will be relayed from one PCME to the next PCME until they reach the PCME that corresponds to the address subspace to which they belong. This is the PCME that terminates the corresponding virtual circuit. At that PCME, the depacketization procedures described in Recommendation G.764/G.765 will take place, and the channelized information is played back at the outgoing access lines.

I.6.3.2 Example of a PCME cluster with hot switching capability

The following example may serve to clarify the above concepts. Initially, none of the packet streams are backed up, and the traffic for each of the trunk groups flows as shown in Figure I.8. At some point, the bandwidth on bearer #l of the first PCME or more. (As mentioned earlier, a PCME may have multiple bearers, but for simplicity, we shall ignore this possibility.) Switches A and B are told to stop establishing calls on trunks 001-120. (To facilitate our discussion, we denote the corresponding DLCIs to be also in the range 001-120, but of course this is not mandatory.)

Suppose that after ten minutes, all trunks are idle except trunk 024, which happens to be carrying a voiceband data connection that may persist indefinitely. This can be done by requesting that PCME #1 backup its packet stream so that all traffic from trunks 001 through 120 will now be routed onto bearer #2. All this is transparent to the end user, and because only trunk 024 is active, the net increase in loading on bearer #2 is negligible.

The same steps are taken if bearer #2 from PCME #2 is to be freed as well. Switches A and B stop establishing calls on trunks 121 through 240. Suppose that after 10 minutes, trunks 122 and 135 are still active (trunk 024 is also still active). Then, activation of the packet stream backup on bearer #2 will move the traffic from trunks 001 through 120 and 121 through 240 onto bearer #3. The effect is restricted to re-directing the traffic arriving on the channelized side on active trunks 024, 122, and 135.

To re-establish bearer #l between the two switches, a request to PCME #l to restore bearer #l will allow the traffic incoming on trunks 001 through 120 to once again be routed onto bearer #l instead of bearer #3. This will restore the voiceband data call of trunk 024 to its initial bearer. Again, this is done transparently to the end-user. Then, the switches are permitted to establish new calls on trunks 001 through 120. In the meantime, trunks 121 through 240 will continue to be routed on bearer #3.

With suitable engineering rules, a much larger number of active calls can be switched over to the backup facility, without any noticeable degradation of quality. This is especially true if the calls are carrying voice, because a PCME can drop 1 or 2 enhancement blocks to reduce the bandwidth needed for voice packets and avoid freeze-out. Assume that 30 calls in progress on bearer #l are re-routed over bearer #2, and that the bearer #2 was already carrying 120 calls using a 4:1 compression ratio. In this case, bearer #2 will be carrying 150 calls with a new compression ratio of 5:1. This new ratio fails well within the recommended operational guidelines for providing a toll quality service. In fact, field data have shown that with a packetized system, the average bit/sample can be driven down to 3.2 bit/s or even 2.8 bit/s without noticeable degradation in the voice quality.

This example shows a way for re-arranging transmission capacities without affecting any of the calls in progress, regardless of their source and prior re-routing. If only some traffic is to be protected, then the permanent virtual circuit restoration procedure described in clause 14/G.765 could be used.

I.6.4 Frame relay applications

A PCME allows the combination of voiceband, digital data, and image in a single private (or virtual private) network. Virtual private networks offer the advantages of private networks without their operational difficulties by using the resources of a network provider, including network maintenance and management. Finally, a hybrid virtual private network is a way for a network to provide virtual private network services with equipment at the customer premises.

A PCME with a Q.922 interface can support the connection between Wide Area Networks (WANS) and Local Area Networks (LANs). This is in addition to X.25 and Q.921 connectivity using VDLC. Thus, a PCME can interface with high performance LAN interconnect devices or routers that support the routing requirements for IEEE 802.3/Ethernet LANs across a diverse range of campus and WANS. These routers act as communication servers that support Q.922 functionality.

The connection between the router and the PCME can be through the router’s V.35 serial interface.

A PCME requires that the 1544 kbit/s primary-rate interface uses the Extended Superframe Format (ESF) of Recommendation G.703/G.704 and that the HDLC bit stream be inverted to operate on facilities that restrict the l’s density. A bit-inverted channel refers to taking a normal LAPPED frame after bit stuffing and inverting its contents before transmittal. This is done to preserve the l’s density requirement on all restricted primary-rate facilities. Normal HDLC bit stuffing guarantees that a zero will exist at least every 6 bits. By inverting this format a 1 is guaranteed in at least a 6-bit interval.

Note that some “non-fractional” DSU/CSUs cannot convert between Recommendation V.35 and a primary-rate without corrupting the HDLC stream. This type of modem will not work with any primary-rate equipment expecting a HDLC stream.

Subclause 3.2.7/G.764 specifies a maximum frame size of 490 octets between flags. Thus, if the PCME has no Q.922 interface, the router’s transmittal unit must be configured so that it does not allow frames with information fields exceeding 490 octets in size. If the PCME has a Q.922 interface, the PCME itself will carry the segmentation and the recombination of the frames at both endpoints.

The address mapping between Recommendations Q.922 and G.765 can be done either at the PCME or at the router before the frame encapsulation.

In an experiment, two SUN Sparc workstations exchanged a 5-Mbit (5137338 octets) file that was transmitted for various values of packet sizes. This was done to determine the effect on performance of segmenting the message before sending it into the PCME network into smaller packets. The sizes used were: 128, 450, 490, and 500 octets. The minimum size to avoid segmentation is 576. The results are presented in Table I.7.

TABLE I.7/G.765

5-Mbit file transfer from SUN Sparc to SUN 3 workstation

Transmission method	Frame size (octets)	Transmission time (ms)	Average rate (kbit/s)
Ethernet direct TCP/IP (10 Mbit/s)	1514	49 • 1	800 • 1
768 kbit/s (direct TCP/IP)	576 490	62 • 1 78 • 1	656 • 1 528 • 1
768 kbit/s (frame relay)	500-450 128	79 • 1 140 • 1	520 • 1 296 • 1

When the two workstations were connected directly through Ethernet, the file transfer took 49 s (at the rate of 800 kbit/s) with the maximum message size of 1514 octets. This is the maximum message size the workstation will transmit and is the maximum message size. When the two workstations were connected by a dedicated line through the routers, the transmission time increased to 62 s for a packet size of 576 octets, and to 78 for a size of 490 octets. Here the bandwidth available was a 768 kbit/s stream.

With frame relay through the PCME, the 5-Mbit file transfer lasted around 79 s for packet sizes from 450 octets to 500 octets. This shows that transition through the PCME has negligible effect on the total transmission time. When the packet size was reduced to 128 octets, the time jumped to 140 s (at the average rate of 296 kbit/s).

Transfers in the opposite direction, SUN 3 to SUN Sparc, took approximately 6 percent longer due to the differences in the performance of the workstations. For example, the dedicated line transfer took 84 s when the size was set at 490 octets. These results show that most of the delay is caused by segmentation of the message.

Parsing (and segmentation) at the workstation reduces the throughput by about 20 percent; frame relay adds an additional reduction of less than 1 percent. When the messages are divided into even smaller packets (for example, 128 octet packets), the transmission time increases dramatically.

A Channel Activity Simulator (CHAS 96-1) was used to provide the voice load, with an activity factor of 40 percent. Twelve voice channels were provisioned with digital speech interpolation. Voice coding was set to the embedded ADPCM (4,2) algorithm of Recommendation G.727. For normal load, 4 bits were used per voice sample. Under normal load, some channels may drop up to 3 bits per sample or 2 bits per sample. The load is assumed to consist of short bursts of highly concentrated traffic to allow the algorithm to shed the excess load in a manner that is imperceptible to the listener. Under an extremely heavy load, entire packets containing 16 ms bursts of speech will be dropped as well as packets containing data. Thus, the traffic conditions were such that 768 kbit/s of voice and 768 kbit/s of constant data were vying for the same 768 kbit/s packet stream. A number of buffering schemes and loads were used to determine the effect of integrating the LAN traffic with voice. With proper adjustment of the PCME's parameters, the throughput for LAN traffic was reduced from 768 kbit/s to 570 kbit/s. The voice quality was unaffected and voice samples retained their 4 bits. Since there was no flow control in the data generation, the reduction in data bandwidth was achieved by dropping frames. Increasing the voice load to 24 channels reduced the LAN traffic throughput to 414 kbit/s, with the voice maintaining an average of 3.3 bit/s.

The performance is even better in the more realistic case of non-constant LAN load. For example, with 48 active voice channels, the 5-Mbit file was transferred from the Sparc to the SUN 3 in 130 seconds (at the average rate of 320 kbit/s) while the average bit-per-voice sample was maintained at 3.9 bits per sample. In this case, the delay was caused by the flow control between the two terminals and not by frame dropping.

The real advantage of interconnecting the router and PCME in a LAN/WAN application would be in the application where integrated services are desired. It has been shown that voiceband traffic and LAN traffic may share the same transmission trunks, and a balance may be engineered between voice compression and LAN traffic throughput.

Figure I.8 illustrates a possible integrated network in which local nodes contain a router and channelized voice. Both types of traffic would be transmitted to a wideband packet network of PCME using hybrid facilities, that is, part of the primary-rate bandwidth would contain a frame relay packet stream while the rest of the bandwidth would contain voice circuits. When the voice circuits reach the first PCME, they would be converted to packet format and integrated directly with the LAN traffic. At the far end of the network, the voice circuits would be returned to their channelized state and be transmitted with the frame relay LAN traffic on hybrid facilities to the far-end local node.

I.6.5 Mobile telephony applications

I.6.5.1 Analogue cellular networks

In an analogue cellular network, the MTSO is connected to the Public Switched Telephone Network (PSTN) and to individual cell sites via wire lines (Figure I.9). The MTSO monitors the signal quality on the channels and switches between channels *(hand-off)* when the quality is below a specified level. It performs the re-routing operation needed when a hand-off takes place in addition to carrying out many other cellular call-processing operations. Furthermore, the MTSO generates billing data and custom-calling features such as call-forwarding.

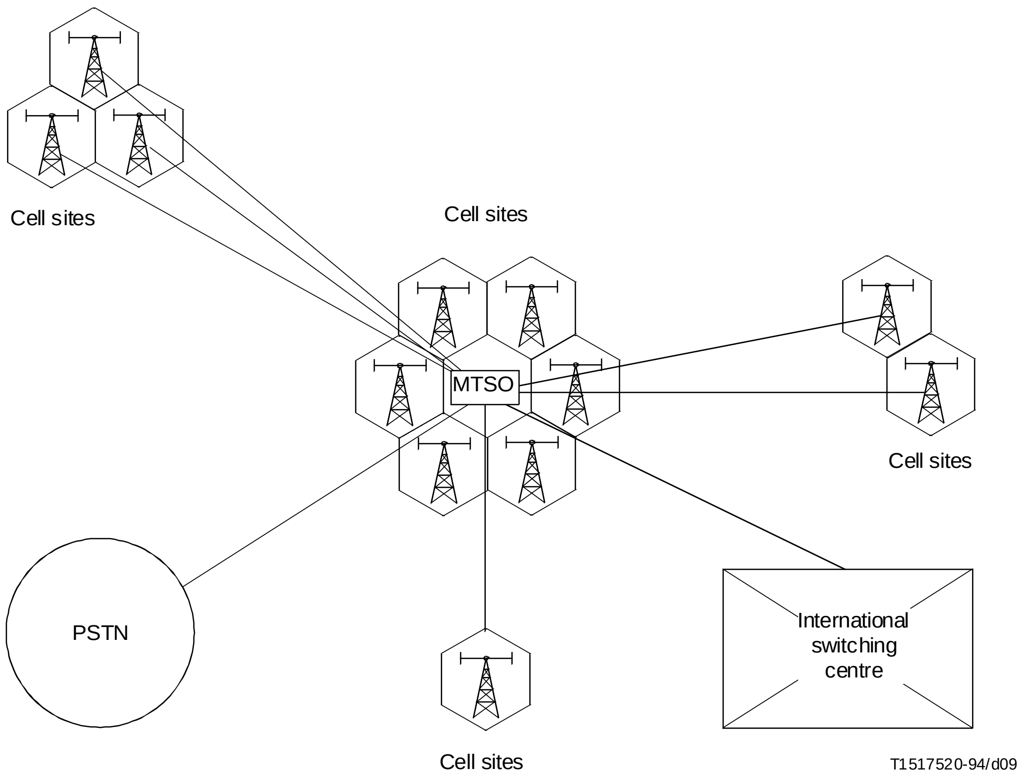


FIGURE I.9/G.765
Analogue cellular network

During the hand-off, there may be a brief loss of signal from about 200 ms to 1200 ms. When a mobile unit transmits facsimile, this loss will cause a streak of about 1/16th inch across the page for each hand-off. The frequency of hand-off is unpredictable, and the number of streaks per page depends on the machine. In extreme cases, the gap may result in a failure message or a disconnect.

For low-traffic densities, such as for isolated communities and rural areas, a decentralized architecture may be more economical. In this architecture, several cells are interconnected by a Switching Mobile Center (SMC) which supervises the operation of those few cells (Figure I.10). The SMCs act as the primary interface to the PSTN. They are interconnected to each other and to a Master Mobile Center (MMC) that serves as the primary man-machine interface and master data base.

In this distributed architecture, satellite links may provide the interconnections among the various SMCs and/or the MMC, thereby enabling a multicity, long-distance cellular network. This is particularly useful whenever there is a serious lack of microwave or other terrestrial systems. Compression equipment is used in cellular networks to increase the number of calls carried over the cellular network transmission media. Multiplexers are also used to combine the traffic from multiple data, voice, or video channels for transmission over a single primary-rate line. Finally, digital cross-connect systems are used at various levels of the transmission hierarchy to re-route calls without demultiplexing them. A PCME combines all these functions and adds the following functions:

– compression gain because of digital speech interpolation and ADPCM coding of speech;

– graceful congestion control that is adapted to the actual link condition;

– real-time signal classification of the incoming signal into voice, voiceband data at the various speeds, facsimile, etc., which allows the matching of the appropriate compression technique with the type of the signal;

– facility backup capability.

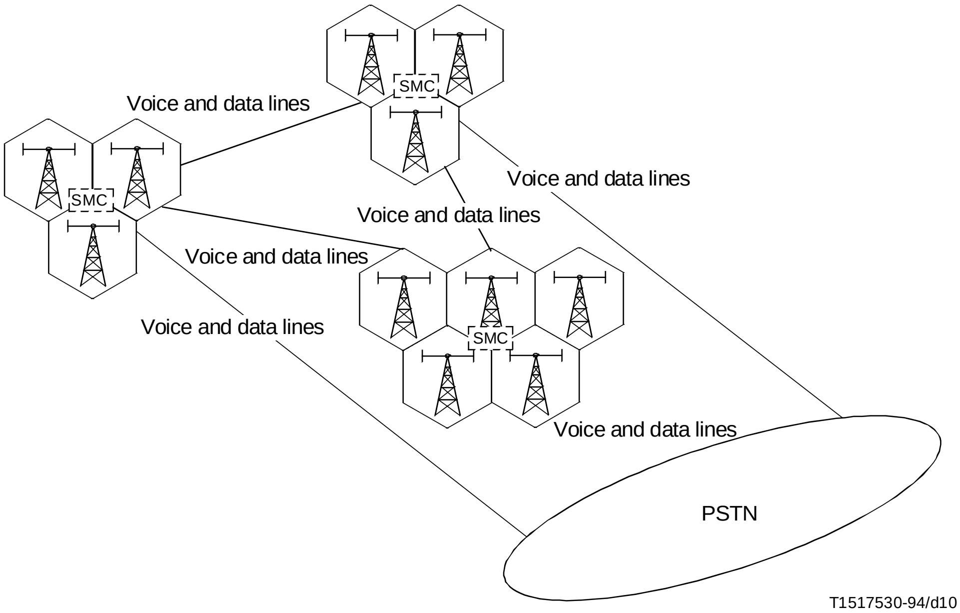


FIGURE I.10/G.765

Cellular network with SMC

PCME may also result in the following:

– the rate of premature disconnection of facsimile calls may be reduced because, when there is a gap during hand-off, the PCME inserts a filling pattern of zeros;

– during image transfer, faacsimile quality may be improved because demodulated and packetized far-simile is less susceptible to line errors, integrated and centralized management, and control of the various terminals.

The following network engineering issues must be addressed in a cellular network:

– silence removal;

– line Echo;

– tandeming;

– delay.

I.6.5.1.1 Silence removal

The principle of silence removal is to remove the silence segments from the speech. This is similar to digital speech interpolation in traditional compression applications. However, in mobile applications, the signal that arrives at the speech detector includes the reverberation due to the reflections of the sound waves on the hard surface inside the car as well as the vehicular noise. Therefore, the signal can be represented as:

$$S_{\text{speech detector}} = s(t) + r[s(t)] + n_{v(t)} + n(t)$$

where:

s(t) = speech signal;

r[s(t)] = reverberations of the speech;

$n_{v(t)}$ = vehicular noise;

n(t) = circuit noise.

The vehicle noise depends on several factors such as:

1) the car speed;

2) the status of the windows (open or closed);

3) the road conditions;

4) the status of the fan (on or off).

Furthermore, in hands-free applications, the background noise is not necessarily white, particularly if the vehicle windows are open. In these situations, it may be advantageous to increase the threshold level for the speech detector to avoid clipping by decreasing the sensitivity of the speech detector. This would allow the transmission of the background noise, if it reaches a defined level, at the expense of reducing the effective compression gain. The value of the threshold for the speech detector to pass the background noise must be defined subjectively during the equipment provisioning time, depending on the conditions from which calls are placed to the cell site.

I.6.5.1.2 Echo

This subclause discusses line echo only because acoustic echo is outside the scope of this discussion.

Echo canceling is needed for two reasons:

1) to avoid impairments to voice and voiceband data (including facsimile) traffic;

2) to achieve the maximum compression gain, since echo can fool the speech detector.

I.6.5.1.3 Tandeming

In current centralized architectures, all intelligence for the cellular network, including call set-up, resides in the MTSO (see Figure I.11) even if the traffic is not directed to the PSTN (that is, going to the same cell site or to a different cell site). Therefore, all traffic will be tandemed as follows:

a) *Voice/Voiceband data*

   – (Cellular to PSTN)
   Analog-PCM-ADPCM-PCM;
   This case does not include international traffic which will be similar to the next case.

   – (Cellular to Cellular)
   Analog-PCM-ADPCM-PCM-ADPCM-PCM-Analog.

b) *Facsimile:* Digitai-Analog-PCM-baseband-PCM-baseband-Analog-Digital.

The successive codings and decodings may affect performance. This tandeming effect does not apply to the data link between the MTSO and the cell site. In this case, the PCME may use either DICE or VDLC. It is also possible that the data link be provisioned for 64 kbit/s clear operation.

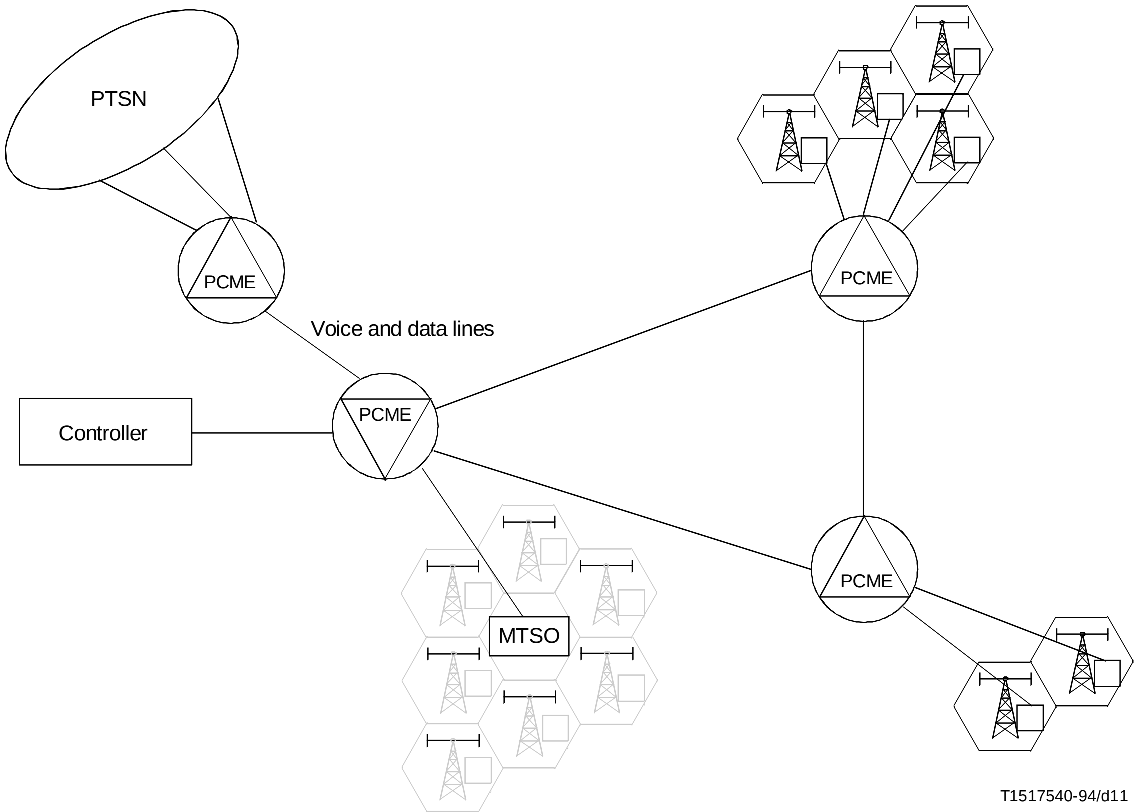


FIGURE I.11/G.765

Cellular network with MTSO

Because most of the traffic is to and from the national or regional PSTN, the likelihood of having more than one tandem encoding/decoding is small. It should be noted that for inter-cell and intra-cell traffic, the PCME capabilities for packet cross-connection and circuit cross-connection can be used to avoid tandem encoding. This may require some modifications to the cell-site equipment so that incoming calls can be routed on pre-defined time slots that correspond to their final destination. (See Figure I.12.)

I.6.5.1.4 Delay

The use of a PCME introduces the packetization delay and a build-out delay. This delay is added once to calls connected to the PSTN in each direction of transmission. If the call is to the same cell site or to another cell site, the total delay will be multiplied by two in each direction of transmission when a PCME is located between each cell site and the MTSO. The effect of delay on hand-off should be evaluated.

The value of the build-out delay is less if the traffic is mainly voice traffic and facsimile is not demodulated/remodulated. Also, the build-out for digital data traffic is independent from the build-out for voiceband traffic.

I.6.5.2 Digital cellular networks

The basis for this discussion will be the architecture of the GSM network. In this network, the base station subsystem consists of the Base Transceiver Subsystem (BTS) and Base Station Controller (BSC) that provides real-time control and management to a number of BTS's.

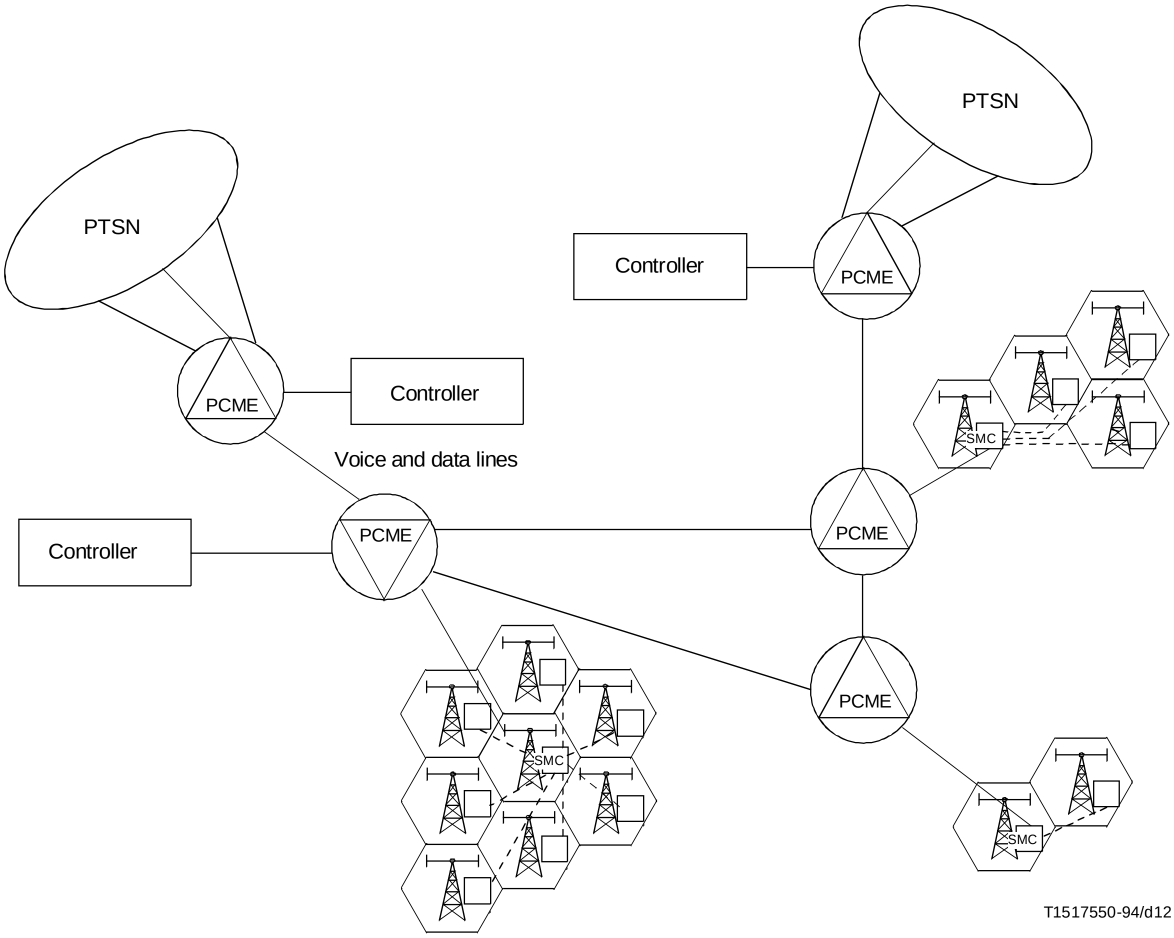


FIGURE I.12/G.765

Cellular network, inter-cell and intra-cell traffic

The BSC controls all the low-level control and cellular functionality of the Mobile Switching Center (MSC).

In some GSM implementations, 16 kbit/s transmission bandwidth are used between the BTSs and their corresponding BSC, so that 4 calls are carried in a 64 kbit/s time slot. However, in the BSC, each voice call is transcoded to 64 kbit/s PCM. A law (voice) and data streams are padded to increase the bit rate to 64 kbit/s according to the structure of Recommendation V.110 (data, including facsimile). This allows for interworking with ISDN, PSTN; but, if the BSC and the MSC are not co-located, transmission bandwidth may be wasted.

Thus, the use of PCME between BSCs and MSCS, as well as between MSCS, may be advantageous for the following reasons:

a) They provide voice and data compression on these links. This is important in lowering the costs of operation where 2 Mbit/s links have been leased from the PTT by the cellular operator, when radio links have been set up to interconnect MSCs, or when there is a scarcity of 2 Mbit/s links.

b) They allow for voice networking of the MSCs in the compressed domain.

c) They provide a smooth interface with ISDN, the PSTN, and international gateways in countries that use PCME at their gateways.

Notice that the facsimile demodulation/remodulation function of the PCME cannot be used because facsimile arrives in the form of V.110 stream that is to be given a 64 kbit/s channel. While this reduces the amount of bandwidth available for compressed traffic, the build-out delay can be reduced by about 30 to 40 ms.

Some of the network engineering issues to be studied are:

– GSM specifications mandate that the tail length be set at 64 ms. Phase roll effects in GSM networks should be investigated.

– The effect of tandeming of the GSM algorithm with ADPCM on voiceband traffic of international calls are excluded.

– The worst-case tandeming will be as follows: PCM-(RPE-LTP)-PCM-ADPCM-PCM-ADPCM-PCM-(RPE-LTP)-PCM.

– Additional delay is due to the RPE-LTP encoding in addition to the packetization delay.

I.6.5.3 Summary

The discussion in the appendix was based on a juxtaposition of existing products, without any attempt to optimize the configuration. For example, the packet cross-connect capabilities of the PCME are not used to tie the various cell sites in the compressed domain. No attempt was made to use the facsimile demodulation/remodulation features of the PCME for digital cellular systems. In fact, a protocol may be entertained along the lines of DICE to extract the user's information from the V.100 stream, thereby saving some precious bandwidth (on satellite). Finally, there is no dynamic reconfiguration of PVCs for different types of traffic because this would require changes to the signalling BSC interface to the PCME.

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