INTERIM EUROPEAN TELECOMMUNICATION STANDARD

I-ETS 300 131

April 1992

Source: ETSI TC-RES

Reference: CV/RES-0001

ICS: 33.060, 33.060.20

Key words: CT2, CAI

Radio Equipment and Systems (RES); Common air interface specification to be used for the interworking between cordless telephone apparatus in the frequency band 864,1 MHz to 868,1 MHz, including public access services

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New presentation - see History box

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Foreword

This Interim European Telecommunication Standard (I-ETS) has been prepared by the Radio Equipment and Systems (RES) Technical Committee of the European Telecommunications Standards Institute (ETSI) and has been adopted having undergone the ETSI standards approval procedure.

Parts of this I-ETS were prepared by the British Standards Institution (BSI), by the United Kingdom Department of Trade and Industry (DTI) and by those companies listed in Annex H. It contains information and copyright material, the property of these organisations. The connection of equipment specified here to the Public Switched Telephone Network (PSTN) is covered in prETS 300 001 [1].

Every I-ETS prepared by ETSI is a voluntary standard. This I-ETS contains text concerning the type approval of the equipment to which it relates. This text should be considered solely as guidance, and does not make the I-ETS mandatory.

The standard includes two types of requirement: those that are required in all units; and those that are optional in a unit, but shall be implemented in the specified manner if provided. The specifications of parts one and two are requirements unless otherwise stated. The specifications of part three are optional unless otherwise stated. The specifications of part four are requirements unless otherwise stated. The tests specified in part five shall be passed by all units where the tested feature is provided.

Annexes A to D and Annex J of this standard are normative. Annexes E to H and K to L are informative.

Introduction

This specification covers the minimum performance requirements for fixed and portable radio units used with the second generation cordless telephone (common air interface) CT2 (CAI) service operating in the band 864,100 MHz to 868,100 MHz.

This specification is intended to allow a user to migrate from one cordless telephone environment (public or private; telepoint, domestic or PBX) to another without having to change, or having to purchase additional, radio equipment.

The specification is divided into five main parts:

Part 1: The radio interface (Clause 4): This part covers the minimum radio frequency performance requirements including channel frequencies, modulation and channel selection.

Part 2: Signalling layers one and two (Clauses 5 and 6): There are three layers of signalling requirements for the radio units. The first two layers are detailed in this part.

Signalling layer one covers aspects such as time-division duplexing, data multiplexing, link initiation and handshaking. This layer allows systems to obtain mutual synchronisation over a digital synchronisation channel and provides bi-directional data channels for digital signalling data and digital speech data.

Signalling layer two covers the signalling channel protocols, message formats, error detection, error correction and message acknowledgement. This layer allows systems to communicate over an established link using data and signalling channels which are established and maintained free from interference where possible.

Part 3: Signalling layer three (Clause 7): This part defines the structure of and attaches meanings to messages. Part of the message space is undefined in order to accommodate future expansion of services and facilities.

Part 4: Speech coding and transmission (Clause 8): This part specifies the requirements for the digital coding and transmission of analogue speech information.

Part 5: Parametric and system tests (Clauses 9 to 11): This part specifies the tests required to verify that a system conforms to this specification.

1 Scope

This standard specifies the technical requirements for equipment known generically as common air interface second generation cordless telephones or CAI CT2. This equipment is intended to convey digitally-encoded speech with associated digital signalling, via a radio frequency channel, to and from the Public Switched Telephone Network (PSTN), possibly via a private network. The transmission of non-speech data, other than by voice-band audio signals conveyed in the digital channel allocated to voice data, is not permitted by this standard.

This standard specifies the essential requirements for:

- generation and interpretation of digital speech;
- generation and interpretation of digital control signalling;
- the means by which the two ends of a cordless link become and remain synchronised; and
- the means by which the specified data structures are modulated onto one of a number of specified radio frequency carriers.

The speech performance characteristics defined in this document typically conform to ETS 300 085 [7], which specifies the overall performance between the handset acoustic interface and a PCM digital network interface. The deviations from ETS 300 085 [7] are limited to the consequences of non-PCM coding and transmission delay.

These additional features are not included in ETS 300 085 [7], but are likely to occur in a CT2 system: analogue interface, loudspeaking and hands-free operation, tandeming with a mobile radio network. Headsets are not covered by the present specifications.

For the CT2 systems which connect to the PSTN via an analogue interface, the document includes the basics on which national specifications can be built, referring to prETS 300 001 [1] (which specifies the connection of terminal equipment to the PSTN via a 2-wire analogue interface).

This standard covers the technical requirements considered necessary to ensure that:

- minimum interference is created to other users of the RF spectrum; and
- there exists a defined minimum degree of interworking between the portable and fixed parts of a cordless telephone apparatus, allowing users, in possession of a compliant Cordless Portable Part (CPP) and if so authorised, to make and receive telephone calls from any compliant public or private Cordless Fixed Part (CFP).

Equipment covered by this standard is of the following types:

- apparatus intended for voice telephony use;
- cordless portable parts (CPPs) and cordless fixed parts (CFPs) with internal or external antenna, and with or without an external RF connector;
- CPPs containing one or more RF transceivers; and
- CFPs containing one or more RF transceivers and capable of operating simultaneously with one or more CPPs.

2 Normative references

This Interim European Telecommunications Standard incorporates, by dated or undated reference, provisions from other publications. These normative references are cited at the appropriate places in the text and the publications are listed hereafter. For dated references, subsequent amendments to, or revisions of, any of these publications apply to this Interim European Telecommunications Standard only when incorporated in it by amendment or revision. For undated references the latest edition of the publication referred to applies.

[1] prETS 300 001: "Attachments to Public Switched Telephone Network (PSTN); General technical reqirements for equipment connected to an analogue subscriber interface in the PSTN (Candidate NET 4)".

[2] CCITT Recommendation G.823 (1988): "The control of jitter and wander within digital networks which are based on the 2048 kbit/s hierarchy"

[3] CCITT Recommendations I.441 and I.451 (1988): "Integrated Services Digital Network (ISDN) user-network interface, data-link layer specification" and "Integrated Services Digital Network (ISDN) user-network interface layer 3 specification for basic call control"

[4] CCITT Recommendation Q.931 (1988): "Digital Access Signalling System"

[5] CCITT Recommendation T.50 (1984): "International Alphabet No.5"

[6] CCITT Recommendation G.721 (1988): "32 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM)"

[7] ETS 300 085: "Integrated Services Digital Network (ISDN): "3,1 kHz telephony teleservice; Attachment requirements for handset terminals (T/TE 10-06) (Candidate NET 33)"

[8] CCITT Recommendation G.122 (1988): "Influence of national systems on stability, talker echo, and listener echo in international connections"

[9] CCITT Recommendation G.132 (1988): "Attenuation Distortion"

[10] CCITT Recommendation G.223 (1964 with amendments): "Assumptions for the calculation of noise on hypothetical reference circuits for telephony"

[11] CCITT Recommendation G.714 (1988): "Separate performance characteristics for the encoding and decoding sides of PCM channels applicable to 4-wire voice frequency interfaces"

[12] CCITT Recommendation P.51 (1988): "Artificial ear and artificial mouth"

[13] CCITT Recommendation P.64 (1988): "Determination of sensitivity/frequency characteristics of local telephone systems to permit calculation of their loudness ratings"

[14] CCITT Recommendation P.79 (1988): "Calculation of loudness ratings"

[15] CCITT Recommendation G.711 (1972 with amendments): "Pulse Code Modulation (PCM) of voice frequencies"

[16] CCITT Recommendation P.76 (1988): "Determination of loudness ratings; fundamental principles"

[17] CCITT Recommendation G.113 (1988): "Transmission Impairments"

[18] ISO 3 - 1973: "Preferred numbers - Series of preferred numbers"

[19] CCITT Blue Book (1988), Volume V, Supplement 13: "Noise Spectra"

[20] ISO 2022: "Information processing - ISO 7-bit and 8-bit coded character sets - Code extension techniques"

3 Definitions and abbreviations

For the purpose of this standard, the following definitions and abbreviations apply.

ACW: Address Code Word

ADPCM: Adaptive Differential Pulse Code Modulation

AFC: Automatic Frequency Control

AGC: Automatic Gain Control

AM: Amplitude Modulation

AUTH_KEY: Authentication function identifier

AUTH_NO: Authentication function to be used

AUTH_PREF: Preferred CPP Authentication function

AUTH_REQ: Authentication Request Information Element

AUTH_RES: Authentication Response Information Element

AUTH2_REQ: Alternative Authentication Request Information Element

AUTH2_RES: Alternative Authentication Response Information Element

B channel: 32 kbit/s Speech or Data Channel (in CT2)

BAS_CAP: Base Capabilities Information Element

BASET: Base Type

BER: Bit Error Ratio

BID: Base Identity code

CAI: Common Air Interface

CC: Channel Control Information Element

CFP: Cordless Fixed Part

CHAR: Character Information Element

CHM: Channel Marker bit pattern

CHMF: Channel Marker bit pattern sent by CFP

CHMP: Channel Marker bit pattern sent by CPP

C/I: Carrier/Interference Ratio

CIC: Codec Identity Code

CKEY: Encrypted KEY

Code Word No: Indicates Number of Octets in Last Code Word

CPP: Cordless Portable Part. Note that CPP=CRE if the CPP has no TIM capability and that CPP=CRE+TIM if the CPP has a TIM capability.

CRC: Cyclic Redundancy Check

CRE: Cordless Radio Equipment

CT2: Second Generation Cordless Telephone

CTA: Cordless Telephone Apparatus

DCAP: Display capability

D channel: Signalling channel (in CT2)

DCW: Data Code Word

DISP: Display Information Element

DL_EST_IND: Data Establish Indication

DL_EST_REQ: Data Establish Request

E-CKEY: Encrypted E-KEY

E-KEY: Expected Key

Endwrd: Indicates Last Code Word of a Packet

ERP: Ear Reference Point

f_c: Nominal Channel Centre Frequency

FA: Feature Activation Information Element

FI: Feature Indication Information Element

FSK: Frequency Shift Keying

FT:	Format Type bit
HIC:	Handset Identity Code
HSSC:	High Speed Signalling Capability
IA5:	International Alphabet Number 5 [5]
ICOM:	CFP Intercom Capability
ID:	Identity Code (generic name for PID,LID)
IDLE_D:	D channel Preamble pattern
ID_LOST:	Handshake-lost code word
ID_OK:	Handshake-ok code word
IE:	Information Element
INCZ:	Increment ZAP field control bit
INIT:	Initialisation Information Element
I/S:	Information/Supervisory Type Packet
ISDN:	Integrated Services Digital Network
ISO:	International Standards Organisation
KEY:	Personal Key Number
KP:	Keypad Information Element
LID:	Link Identification Code
LINK_GRANT:	CFP Acknowledgement Assigning Call Reference ID
LINK_REQUEST:	Link Status Request to seize Link
Lmest:	Sidetone Path Loss
LRGP:	Loudness Rating Guard-ring Position
LSB:	Least Significant bit
LSTR:	Listener SideTone Ratio
LS(0/1):	Link Status bits
L3_end:	Indicates Last Packet of a Layer 3 Message
MANIC:	Manufacturer Identity for CFP and CPP
MB:	Message Buffer Size

MF:	Multiple Frequency
MIC:	Manufacturer Identity Code
MMI:	Man-Machine Interface
MODEL:	Manufacturers CPP model identity code
MRP:	Mouth Reference Point
MSB:	Most Significant bit
MUX1.2:	Signalling Multiplex Mode 1 (two-bit signalling)
MUX1.4:	Signalling Multiplex Mode 1 (four-bit signalling)
MUX2:	Signalling Multiplex Mode 2
MUX3:	Signalling Multiplex Mode 3
N(r):	Receive Sequence Number
N(s):	Send Sequence Number
NO_POLL:	Number of polled CPPs Information Element
NTTA:	Network Terminating and Testing Apparatus
OARAC:	On Air (de-)Registration Acknowledge Information Element
OPSIC:	Operators Identification Code
PABX:	Private Automatic Branch Exchange
PACKET:	Layer two entity comprising ACW and any following DCWs transmitted as a single unit over which the signalling protocol may operate
PAR_REQ:	Parameter Request Information Element
PAR_RES:	Parameter Response Information Element
PAR_SET:	Parameter Set Information Element
PCM:	Pulse Code Modulation
P/F:	Poll/Final bit (0 = Unacknowledged Operation; 1 = Acknowledged Operation)
PI:	Protocol Identifier
PID:	Portable (CPP) Identification Code (=MIC + HIC)
PLL:	Phase-Locked Loop
PSTN:	Public Switched Telephone Network
QDU:	Quantization Distortion Unit

RAND:	Random Number used in generating CKEY
REJ:	Indicates Rejection of a Received Packet
Rem:	Indicates Number of Octets in Last Code Word
RF:	Radio Frequency
RFP:	Radio Fixed Part
RLR:	Receiving Loudness Rating
RLRH:	Handset Receiving Loudness Rating
SABM:	Set Asynchronous Balanced Mode; Layer two link protocol initialisation command.
SABM_ACK:	Layer two acknowledgment to SABM
SCA:	Standard Control Authority
SIG:	Signal Information Element
SLR:	Sending Loudness Rating
SLRH:	Handset Sending Loudness Rating
SR:	Signalling Rate bit
SRc:	Signalling Rate Capability
SRr:	Signalling Rate Request
STMR:	Sidetone Masking Rating
SUP:	Supervisory Packet
SYN channel:	Synchronisation channel
SYNC:	Synchronisation word (in SYN channel)
SYNCD:	Synchronisation word for D channel
SYNCF:	Synchronisation word from CFP (in SYN channel)
SYNCP:	Synchronisation word from CPP (in SYN channel)
Tbid:	CPP BID detect from CHM time (19 ms)
Tcfp:	CFP processing time (18 ms max)
Tclr:	Timer awaiting acknowledgement to a clear down request (1,00 s to 1,04 s)
TCOS:	Telepoint Class Of Service
TCLw:	Weighted Terminal Coupling Loss

Tcpp:	CPP processing time (6,2 ms max)
TDD:	Time-Division Duplexing
TERM_CAP:	Terminal Capabilities Information Element
Tf:	CFP Handshake Period
Tfcyc:	CFP MUX2 minimum transmit time (1,4 s)
Tfdetect:	CFP Link grant transmit to ID_OK detection time from CPP (100 ms)
Tfmax:	CFP Link Establishment Timeout (5 s)
Tfpres:	CFP poll response timer (1,00 s to 1,04 s)
Tftx:	CFP link set up LINK GRANT Transmission time (56 ms to 84 ms)
tgain:	Transmit Power Level Control bit
Thlost:	Link Re-establish time (10 s)
Thrx:	ID handshake receive time (1,00 s to 1,04 s)
Thtx:	ID handshake transmit time (400 ms)
TIM:	Telepoint Subscriber Identification Module. The TIM contains subscriber related information and authentication algorithms to gain access to a telepoint network.
Tp:	CPP Handshake Period
Tpcyc:	CPP MUX3 minimum transmit time (750 ms)
Tpid:	CPP PID detect from BID time (384 ms)
Tpmax:	CPP call set up time (5 s)
Tpoll:	CPP poll detection re-trigger time (1,00 s to 1,04 s)
Trate:	Code word transmit rate time (50/100 ms)
TRD:	Telepoint Registration Data
Trtx:	Retransmission time (66/320/600 ms)
U:	Locking/Non-locking Shift Indicator
UI:	Unnumbered Information message
V(r):	Local State Variable (Receive)
V(s):	Local State Variable (Send)
ZAP:	"Zap" field used with INC ZAP to disable handsets

4 Radio frequency interface

4.1 General

Clause 4 covers the minimum RF performance and RF system requirements for cordless telephone equipment which permit, by radio means, some or all of the functions of a normal telephone apparatus and comprises one or more single PSTN line fixed parts, one or more antenna systems, and one or more cordless portable parts.

NOTE: In Clause 4, communication is taken to be the CFP and CPP interchanging either control, or speech, or both.

4.2 Channel frequencies

4.2.1 Channel centre frequencies

In countries where the frequency band is available, the channel centre frequencies for the forty CT2 channels shall be 864,050 MHz + (0,100 x n) MHz, where n is the channel number, lying in the range 1 to 40 inclusive. The first channel (channel number one) lies at 864,150 MHz and the last (channel number forty) at 868,050 MHz.

4.2.2 Channel frequency accuracy

The channel frequency accuracy required of both the CFP and CPP transmitters shall be ±10 kHz maximum difference between the nominal and actual channel centre frequencies over supply voltage and temperature ranges. AFC may be used in the receiver at both CFP and CPP but may only be linked to control the transmitter centre frequency at the CPP.

4.2.3 Rate of change of transmit centre frequency

The maximum rate of change of transmit centre frequency at both CFP and CPP shall not exceed 1 kHz/ms, except for the specific cases of switching of the CPP transmitter from MUX3 to MUX2 and for channel changing.

4.2.4 CTA access

The CTA shall have access to all radio channels defined in subclause 4.2.1.

4.3 Signalling strategy

The supplier shall declare that the signalling strategy complies with each subclause of 4.3.

4.3.1 CTA access

The CTA shall have access to the full number of allocated channels and make use of any free channel when signalling to establish a communication channel.

4.3.2 Signalling during communication

Signalling during communication shall be limited to the same radio channel as is used for communication.

4.3.3 Signalling outside communication

Signalling outside the communication state shall only be allowed for the purposes of subclause 4.4 and limited in duration by the requirements of subclause 4.9.2.

4.4 Dynamic channel allocation strategy

The supplier shall declare that the signalling strategy complies with each subclause of 4.4.

4.4.1 Incoming calls

When an incoming call is detected by the CFP it shall choose a free channel over which to signal, using its handshake, to the CPP. The CPP upon detection and recognition of this handshake shall respond on this chosen channel with a signal using its handshake. The CFP, upon detection and recognition of this response, shall in conjunction with the CPP establish the communication link.

If the above channel acquisition is unsuccessful then the CFP may make re-attempts, sequentially, on the subsequent free channels. These re-attempts shall be restricted to using a maximum of five free channels and shall be constrained by the requirements of subclauses 4.9.2.2 and 4.4.4.

4.4.2 Outgoing calls

When a CPP is requested to make an outgoing call it shall choose a free radio channel over which to signal for a maximum period of 5 s, using its handshake, to the CFP. The CFP, upon detection of this matching handshake shall respond on this chosen channel with a signal using its handshake. The CPP, upon detection and recognition of this response, shall in conjunction with the CFP establish the communications link.

If the above channel acquisition is unsuccessful then the CPP may make re-attempts, sequentially, on the subsequent free channels. These re-attempts shall be restricted to using a maximum of five free channels and shall be constrained by the requirements of subclauses 4.9.2.1 and 4.4.4.

4.4.3 Channel selection strategies

Manufacturers shall use such selection strategies as to ensure random utilization of the radio channels defined in subclause 4.2.1.

4.4.4 Free channel

The decision as to whether a channel is free shall be made on the basis of intermittent or continuous monitoring for a period of time between 200 ms and 2 s. If intermittent monitoring is used the decision shall be based upon a minimum of five distributed samples. Monitoring shall be performed to a nominal resolution of 6 dB or better over this period. The decision on whether a channel is free shall be considered valid only during the period of 2 s immediately following the end of the monitoring period.

A free channel is defined as the following:

i) any channel with a local field strength below an absolute maximum of 40 dB relative to 1 μV/m; or

ii) where all channels are above 40 dB relative to 1 μV/m, then any channel which has the lowest field strength of all channels defined in subclause 4.2.1 as measured, by intermittent or continuous monitoring, to a nominal resolution of 6 dB or better,

but may exclude any channels on which an unsuccessful attempt has been made to establish communications for that call.

4.5 Radio transmitters

4.5.1 RF power

4.5.1.1 Maximum RF power

The normal carrier output power or effective radiated power (see subclause 9.3.1) under normal test conditions and under extreme test conditions shall not exceed 10 mW.

4.5.1.2 Minimum RF power

At nominal design operating voltage the normal carrier output power or effective radiated power (see subclause 9.3.1) under normal test conditions shall not be less than 5 mW for a CPP or CFP, and not less than 6,3 mW for a public access CFP. Informative Annex E contains recommendations concerning interim arrangements for minimum RF power.

4.5.1.3 Output power at low power setting

Output power at the low power setting (see subclause 4.5.3.4) shall be lower than the output power level at the normal power setting. The difference in output power between the two power levels shall be 16 dB ±4 dB and shall be independent of the absolute power level chosen for normal power output.

4.5.2 Modulation

The modulation employed shall be 2-level FSK shaped by an approximately Gaussian filter to meet the requirements of subclause 4.5.5. The peak frequency deviation under all possible data patterns shall lie between 14,4 kHz and 25,2 kHz.

A binary 1 shall be encoded as a frequency higher than the carrier frequency (f_c+ f); a binary 0 shall be encoded as a frequency lower than the carrier frequency (f_c - f). f_c is the RF carrier frequency and f is the deviation.

The designation of the specified emission, according to article 4 of the Radio Regulations is 100KF7WECT.

4.5.3 Adaptive transmitter power strategy

The supplier shall declare that the signalling strategy complies with each subclause of 4.5.3.

4.5.3.1 General

The adaptive transmitter power strategy during communications may vary the transmitter power by up to 20 dB in a CPP and any CFP. This shall only be used when the received field strength is above 90 dB relative to 1 μV/m and shall not alter the operation of any other part of the CTA.

4.5.3.2 CPP only

If the adaptive transmitter power strategy is used only in the CPP, then both a reduction and increase in transmitter power shall be allowed.

4.5.3.3 CPP and CFP

If the adaptive transmitter power strategy is used in both the CPP and CFP, then transmitter power shall not be increased during a call without re-establishing that the radio channel is free in accordance with subclause 4.4.4.

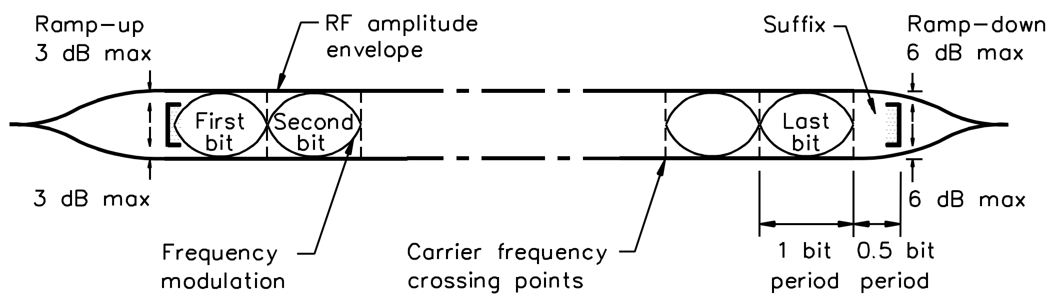
4.5.3.4 Power level changes

The CPP shall be capable of switching its RF output power level between two settings: normal and low power. On power-up, on the origination of a call, or on link re-establishment, the CPP shall set its output power level to normal. At all other times the CPP shall only change its RF output power level in response to a power control message (detailed in subclause 6.5.6) transmitted by the CFP. All RF output level changes shall conform to the requirements of subclause 4.5.3.

4.5.4 Transmitter burst envelope

4.5.4.1 Amplitude

The amplitude of the RF envelope at the start of the first valid bit to be transmitted shall be within 3 dB of the final amplitude of the burst, as shown in figure 1.



NOTE: The signal frequency during ramp-up, suffix and ramp-down may lie anywhere

Figure 1: Data packet within RF envelope

4.5.4.2 Dispersion effects

In order to allow for the effects of dispersion in channel filters, the amplitude of the transmitted signal during the period of 0,5 data bit after the end of the normal transmitted data shall be maintained to within 6 dB of the amplitude obtaining during the transmission of normal data, as shown in figure 1.

4.5.4.3 Guard time

During the guard time between transmitting and receiving, no additional information is present. The signal frequency during ramp-up, suffix and ramp-down may lie anywhere between the specified deviation limits.

4.5.5 Adjacent channel power (narrow band)

The adjacent channel power under either normal or extreme test conditions shall not exceed 10 μW when integrated within a bandwidth of 80 kHz ± 5%.

4.5.6 Out of band power arising from transmitter transients

The power level of any modulated products at a frequency separated by 100 kHz from the nominal frequency shall not exceed 2,5 μW and those by 500 kHz from the nominal frequency shall not exceed 1 nW.

4.5.7 Intermodulation attenuation

This requirement applies to transmitters/receivers to be contained (nested) in a single enclosure or a single unit containing two or more transmitters/receivers which are not separable.

The effective radiated power of the intermodulation products measured in a 10 kHz bandwidth shall not exceed 4 nW.

4.6 Radio receivers

4.6.1 General

4.6.1.1 CTA including three or more RFPs

The manufacturer shall declare that the radio receivers in any CFP comprising part of a CTA which includes three or more RFPs which are contained within the same enclosure and/or which utilise a common antenna or antennas or antenna connection port or ports comply with subclauses 4.6.3, 4.6.4 and 4.6.5.

4.6.1.2 Other CTAs

The manufacturer shall declare that the radio receivers in the CPP or CPPs and in any CFP (other than those described in subclause 4.6.1.1) comprising the CTA comply with subclauses 4.6.3 and 4.6.4.

4.6.1.3 CTAs without integral or supplied antenna

In the case of equipments without integral or supplied antennas, such declaration may be based upon measurements made by application of signals to the termination point provided for non-integral antennas. The termination point shall have an impedance of nominally 50 ohms. In this case, the conversion factor of 0 dB relative to 1 μV/m (0 dBμV/m) being equivalent to -134 dBm (referring to a lambda/2 dipole with 2,2 dB of gain over the isotropic radiator) shall be used to convert the stated field strengths to absolute signal levels.

4.6.2 Sensitivity

The receiver sensitivity (see subclause 9.5.7) shall be defined at a bit error ratio of 1 in 1000 or better in both the B (speech data) and D (signalling data) channels (see subclause 5.2). Informative Annex E contains recommendations concerning interim arrangements for receiver sensitivity.

4.6.2.1 Receiver sensitivity for CFP or CPP using an integral or supplied antenna

The radio receiver sensitivity shall be at least 40 dBμV/m. It is recommended that this be achieved by ensuring that the radio receiver sensitivity is typically 34 dBμV/m or better.

4.6.2.2 Receiver sensitivity with a 50 ohm connector

The radio receiver sensitivity shall be -100 dBm or better at the antenna connector. A sensitivity of - 94 dBm at the antenna connector shall be allowed for a system employing a multi-channel passive combiner/splitter.

4.6.3 Interference rejection

4.6.3.1 Unmodulated interfering carrier signal

The communications state, once established between the CFP and CPP shall be maintained when the receiver of the CFP or CPP is receiving a signal from its associated CPP or CFP at a signal strength equal to that specified in subclause 4.6.2 for the receiver sensitivity plus 5 dB and when an unmodulated interfering carrier signal is introduced at any frequency within the ranges and at the corresponding field strengths (dBμV/m) or carrier to interference ratios (dBc) listed below in table 1.

Table 1: Unmodulated Interfering Signals

Frequency Range(s)	Extreme conditions	Nominal conditions
25 MHz to 800 MHz	120 dBµV/m	123 dBµV/m
800 MHz to 850 MHz 890 MHz to 4 GHz	110 dBµV/m	113 dBµV/m
850 MHz to 860 MHz 872 MHz to 890 MHz	100 dBµV/m	103 dBµV/m
860 MHz to f_c-300 kHz f_c+300 kHz to 872 MHz	35 dBc	38 dBc
f_c-300 kHz to f_c-200 kHz f_c+200 kHz to f_c+300 kHz	30 dBc	33 dBc
f_c-200 kHz to f_c-100 kHz f_c+100 kHz to f_c+200 kHz	20 dBc	20 dBc
f_c-100 kHz to f_c+100 kHz	-20 dBc	-20 dBc

where f_c is the nominal frequency of operation.

The signal from the associated CPP or CFP and the interfering carrier wave are assumed to have the same polarisation.

4.6.3.2 Modulated asynchronous interfering signal

The communications state, once established between the CFP and CPP shall be maintained at a BER of 1 in 1000 or better when the receiver of the CFP or CPP is receiving a signal from its associated CPP or CFP at a signal strength equal to that specified in subclause 4.6.2 for the receiver sensitivity plus 10 dB and when a modulated asynchronous interfering signal is introduced, modulated in any manner that meets this specification, at any channel and at the corresponding field strengths (dBμV/m) or the carrier to interference ratios (dBc) listed below in tables 2 to 4. In these tables, W is the wanted channel. The signal from the associated CPP or CFP and the interfering carrier are assumed to have the same polarisation.

Table 2: Modulated Interfering Signals

Frequency Range(s)	Extreme conditions	Nominal conditions
860 MHz to W - 4 W + 4 to 872 MHz	80 dBμV/m	83 dBμV/m

Table 3: Adjacent Channel Rejection

Channels	Extreme conditions	Nominal conditions
W + 3 and W - 3	30 dBc	33 dBc
W + 2 and W - 2	25 dBc	28 dBc
W + 1 and W - 1	0 dBc	0 dBc

Table 4: Co-channel Rejection

Channel	Extreme conditions	Nominal conditions
W	-20 dBc	-20 dBc

4.6.4 Blocking due to spurious responses

If any part of the CTA fails to meet the requirements of subclauses 4.6.3.1 or 4.6.3.2 only due to a maximum, in each case, of ten spurious responses to, respectively, unmodulated or modulated carrier wave signals at discrete frequencies in the range 25 MHz to 4 GHz, of which two shall be at a field strength of not less than 80 dB relative to 1 μV/m and a remainder at a field strength of not less than 100 dB relative to 1 μV/m, then the CTA shall be deemed to meet the requirements of this standard.

For the purposes of this subclause a spurious response is the failure of the communications state between a CFP or CPP and its corresponding CPP or CFP due to the introduction of interfering radio signals, at any frequency within a continuous band of width 1 MHz or less and of which the centre frequency varies with the channel of operation selected by the CTA, and at a lower field strength than that given as the limiting value in subclauses 4.6.3.1 or 4.6.3.2 for that frequency equal to the centre frequency of this band.

4.6.5 Intermodulation response rejection

The communications state, once established between a CFP and one of its associated CPPs, shall be maintained without interruption when the CFP is receiving a signal from this CPP at a signal strength of 45 dB relative to 1 μV/m and when two interfering signals are introduced, each of which generates a signal strength of 85 dB relative to 1 μV/m at the antenna of the CFP (each modulated as described below) in each of the following cases (where f_c is the frequency of operation):

i) at frequencies of f_c+ 400 kHz and f_c+ 800 kHz;

ii) at frequencies of f_c - 400 kHz and f_c - 800 kHz;

iii) at frequencies of f_c+ 400 kHz and f_c - 400 kHz.

The interfering signals shall each bear continuous data modulation similar to that produced by the CPP. The data modulating the interfering signals shall not, either separately or when combined in any way, simulate the handshake code required to maintain the communications state between the CPP and CFP.

4.7 Combined radio transmitter/receivers

4.7.1 Adverse power supply conditions

The adjacent channel power and spurious emission limits specified shall not be exceeded under normal and under adverse power supply condition.

4.7.2 Spurious emissions of the combined transmitter/receiver

The power of any spurious emission in the specified range of frequencies when the equipment is in the active mode, shall not exceed the value of 20 nW in the frequency bands:

-	41,0 MHz	to	68,0 MHz
-	87,5 MHz	to	118,0 MHz
-	162,0 MHz	to	230,0 MHz
-	470,0 MHz	to	862,0 MHz
-	10,7 GHz	to	12,75 GHz

and shall not exceed a value of 250 nW on other frequencies below 1000 MHz.

On frequencies above 1000 MHz the power of any spurious emission shall not exceed a value of 1 μW.

The power of any spurious emission in the specified range of frequencies, when the equipment is in the idle mode shall not exceed 0,2 nW in the range 864 MHz to 868 MHz (when measured in a 1 kHz bandwidth), 2 nW in the range 100 kHz to 1000 MHz, 4 nW in the range 10,7 GHz to 12,75 GHz, and shall not exceed 20 nW at other frequencies in the range 1000 MHz to 12,75 GHz.

4.8 Termination of the communication state

The supplier shall declare that the signalling strategy complies with each subclause of 4.8.

4.8.1 Clear down signal sequence

Any action for the deliberate termination of the communication state shall initiate an interchange, over the RF link, of a clear down signal sequence.

4.8.2 Cessation of RF activity

Any action for the deliberate termination of the communication state shall, within 1 s, cause the cessation of RF activity in the CPP and that part of the CFP with which it is in communication.

4.8.3 Off-line timing

Where only one CPP is in communication with the PSTN, any action for the deliberate termination of the communication state shall cause that part of the CFP with which it is in communication, and which was on-line to the PSTN, to go off-line to the PSTN within 1 s.

4.9 Channel scanning

The supplier shall declare that the signalling strategy complies with each subclause of 4.9.

4.9.1 Available channels

The CTA shall have access to all radio channels defined in subclause 4.2.1.

4.9.2 Response times

The response times given in subclauses 4.9.2.1 and 4.9.2.2 shall be for the following conditions:

i) full availability of free radio channels to the service;

ii) any possible group of three adjacent radio channels, below a field strength of 40 dB relative to 1 μV/m and the remaining channels carrying speech traffic, with the same modulation format as the CTA at field strengths of 50 dB relative to 1 μV/m.

4.9.2.1 Outgoing

The interval between the CPP initiating the connection to the PSTN and the CFP being on-line shall not exceed 5 s.

4.9.2.2 Incoming

The interval between call arrival indication at the CFP and the CPP's ringing indicator responding, if enabled, shall be less than 5 s. The CFP shall not answer the call arrival indication until the CPP has achieved the communication state.

4.10 In-communication channel switching

4.10.1 Capability

The supplier shall declare whether the CTA is capable of in-communication channel switching.

4.10.2 Channel change delay

If the equipment is capable of in-communication channel switching, an in-communication CTA shall not change channels before 3 s of handshake has been lost (see subclauses 5.4.5 and 5.5) due to poor RF link conditions.

NOTE: Subclause 5.5.1.6 still applies during in-communication channel switching.

4.11 Controls

Those controls, which if maladjusted might increase the interfering potentialities of the equipment, shall not be easily accessible, in particular any control which may cause the equipment to operate outside the permitted frequency limits specified in the other parts of Clause 4.

4.12 Synthesisers and PLL systems

Where synthesisers and/or phase-locked loop systems are utilised for carrier generation, precautions shall be taken to ensure that any lack of synchronisation does not cause deviation outside the permitted frequency limits specified in the remaining parts of Clause 4.

5 Signalling layer one

CT2 signalling layer one specifies methods for the following:

i) the time-division duplexing of data;

ii) the multiplexing of data onto the TDD structure;

iii) two-way digital link initiation;

iv) handshaking.

Operation of CT2 CAI layer one is primarily designed for point to point communication, but multi-point operation is allowed, during the call set up phase for multiple ringing of CPPs. Multiple response resolution from CFPs providing the same service is also provided.

5.1 Data structure and timing

Four different burst structures (MUX3, MUX2, MUX1.4 and MUX1.2) are defined in subclause 5.2. Briefly, however, the standard supports the use of a transmission burst length during speech communication mode of either 68 bits or 66 bits at 72 kbit/s. This is repeated every 144 bits (2 ms) with an interleaved receiving window of up to 68 bits. Since the speech channel (B channel) occupies 64 of the bits in a burst, systems equipped to cater for the 68-bit burst length can signal, in the presence of the speech channel, at twice the rate of those that handle 66-bit bursts only. Eight or twelve bits of guard time remain per burst frame for burst lengths of 68 bits and 66 bits respectively.

5.1.1 Data rate

The nominal data rate shall be 72 kbit/s with a tolerance of ±50 ppm at the CPP and ±50 ppm at the CFP. The data clock may have drift and jitter up to the limits specified for the CCITT Recommendation G.823 [2] table 1, 64 kbit/s interface, but this does not override the long term frequency accuracy specified above.

5.1.2 Time-division duplexing

Normal speech and data communication takes place in time-division duplex (TDD) mode with a guard band between transmit and receive periods (figure 2). The guard band ensures that sufficient time remains at the beginning and end of a transmitted burst for the transmitted power to rise and decay without violating the AM splash requirements of subclause 4.5.6, and for the receiver to be enabled sufficiently in advance of the expected return burst.

Time-division duplex using the frame shown in figure 2 (in MUX1.2, MUX1.4 or MUX2) is the normal operational state except in the case where the CPP is initiating the link, when transmission from the CPP is not constrained to the burst frame structure (see subclause 5.2 for full details).

5.1.3 Master-slave relationship

In the communication state, the CFP shall act as master for frame, burst and bit synchronisation. The CPP shall synchronise its frame, burst and bit clocks to the signals received from the CFP.

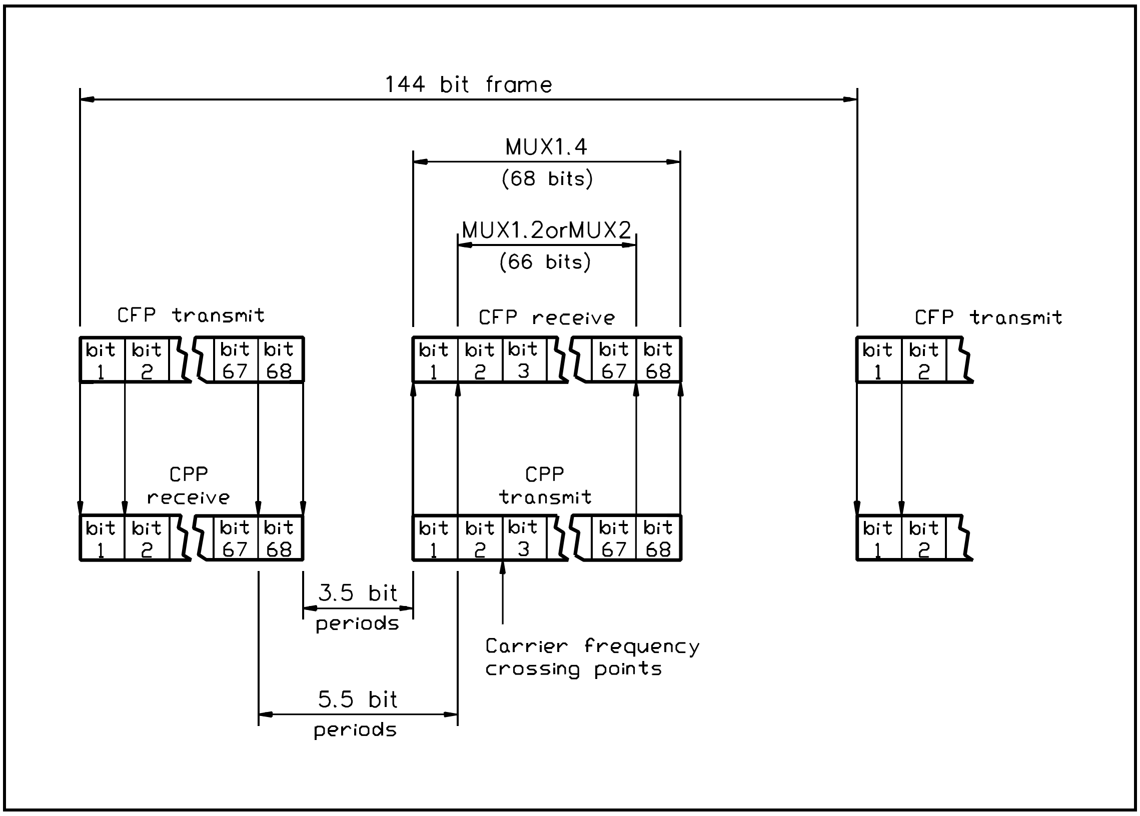


Figure 2: Transmission duplexing

5.2 Sub-channel multiplexes

A CT2 CTA exchanges data in a time-division duplex mode with an instantaneous data rate of 72 kbit/s. Under various situations up to three sub-channels have to be multiplexed within the available data bandwidth. These are:

i) a signalling channel (the D channel);

ii) a voice/data channel (the B channel);

iii) a burst synchronisation channel (SYN channel) used for bit and burst synchronisation.

According to the requirements of a particular situation or function, the proportion of the main channel bandwidth allocated to each of the above sub-channels may vary, and a sub-channel may be absent in some circumstances. Each different allocation of sub-channel bandwidths is termed a multiplex. Three multiplexes (known as multiplexes one, two and three) are used. They are illustrated in figures 3, 4 and 5 respectively and described in subclauses 5.2.2 to 5.2.4 below.

5.2.1 Channel Markers (CHM) and synchronism markers (SYNC)

Multiplexes two and three are used in situations where a CTA may have not yet gained burst synchronism. Special bit patterns are used in the SYN channel in order to mark an RF channel where a link set up attempt is being made and/or to mark a particular time within the multiplex period in order to allow CTAs to gain burst synchronism. The patterns are all of 24-bit length and chosen to yield low auto-correlation and low cross correlation with other frequently-occurring bit patterns.

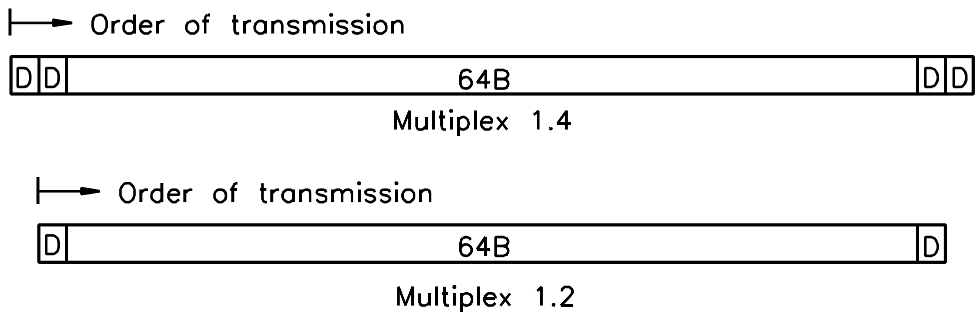
Patterns called CHM are used to mark transmissions within a CTA which is attempting to initialise a radio link and also to mark a particular time within the multiplex. Patterns called SYNC are used when a link has already been established. Details of the usage of CHM and SYNC are covered further in subclauses 5.3 and 5.4.

The CHM bit pattern sent by the CFP (specifically CHMF) is the bit-wise inverse of the CHM pattern sent by the CPP (specifically CHMP). Similarly SYNC sent by the CPP (SYNCP) is the bit-wise inverse of that sent by the CFP (SYNCF). CPPs which expect to see CHMF or SYNCF are therefore unable to recognise marker patterns from other CPPs, and CFPs are unable to recognise marker patterns from other CFPs. The bit patterns for SYNCP, SYNCF, CHMP and CHMF are given below:

```
         msb                                  lsb
        (sent last)                        (sent first)
CHMF     1011 . 1110 . 0100 . 1110 . 0101 . 0000  (BE4E50H)
              .      .      .      .      .
CHMP     0100 . 0001 . 1011 . 0001 . 1010 . 1111  (41B1AFH)
              .      .      .      .      .
SYNCF    1110 . 1011 . 0001 . 1011 . 0000 . 0101  (EB1B05H)
              .      .      .      .      .
SYNCP    0001 . 0100 . 1110 . 0100 . 1111 . 1010  (14E4FAH)
```

5.2.2 Multiplex one

Multiplex one is invoked from multiplex two by means of a layer three channel control message (see subclause 7.2.6). Multiplex one (figure 3) is used bi-directionally over an established link to carry the D and B-channels. The SYN channel is nonexistent in this multiplex and therefore should burst synchronisation be lost, it cannot be recovered without re-initialising the link (see subclauses 5.4.4 and 5.4.5). If the B channel source is disconnected, the input to the B channel scrambler (see below) shall be all zeros.



NOTE: The burst is repeated every 144 bits (2 ms).

Figure 3: Multiplex one

Multiplex one supports both the 68-bit and 66-bit burst structure (referred to as MUX1.4 and MUX1.2 for signalling with four bits or two bits respectively). The raw data rates in this multiplex are 2,0 kbit/s (MUX1.4) or 1,0 kbit/s (MUX1.2) for the D channel and 32,0 kbit/s for the B channel. All CTAs shall support at least MUX1.2. Support for MUX1.4 shall be optional, it shall be used only if both ends of the CTA support the higher signalling rate.

Data bytes in the D channel are aligned in this multiplex so that bytes always start on a frame boundary (figure 6). The alignment of the data in the B channel specified in subclause 8.3.

In MUX1.4 the valid transmitted data bits in a burst are numbered 1 to 68. At the CPP antenna the start of transmission of valid data bit 1 of MUX1.4 shall occur 3,5 bit periods ±0,25 bit periods after the end of valid data bit 68 has been received (also measured at the CPP antenna).

In MUX1.2 the valid transmitted data bits in a burst are numbered 2 to 67. At the CPP antenna the start of transmission of valid data bit 2 of MUX1.2 shall occur 5,5 bit periods ±0,25 bit periods after the end of valid data bit 67 has been received (also measured at the CPP antenna).

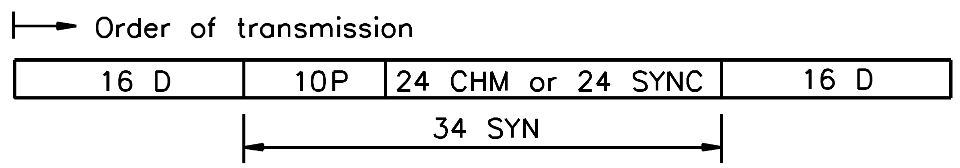
In order to ensure reasonably random bit sequences in MUX1.2 and MUX1.4, the following bits should be inverted in the 64 bits of the B channel within each 2 ms frame:

03, 04, 06, 09, 14, 16, 18, 19, 20, 22, 23, 27, 28, 29, 30, 31,
34, 35, 37, 40, 45, 47, 49, 50, 51, 53, 54, 58, 59, 60, 61, 62.

Note that these bit numbers refer to the 64 bits of the B channel and not the burst bit numbers (bit 1 is the first transmitted bit of the B channel). All other bits should remain non-inverted.

5.2.3 Multiplex two

MUX2 (figure 4) is used to carry the D and SYN channels for link establishment and re-establishment. The B channel is nonexistent in MUX2. MUX2 is used prior to a switch to MUX1 by means of a layer three message (subclause 7.2.6).



NOTE: The burst is repeated every 144 bits (2 ms).

Figure 4: Multiplex two

MUX2 carries the D channel at a data rate of 16,0 kbit/s and the SYN channel at an overall rate of 17,0 kbit/s. The SYN channel consists of ten bits of preamble (one-zero transitions) followed by a channel marker (CHMF) or synchronism marker (SYNC: either SYNCF or SYNCP).

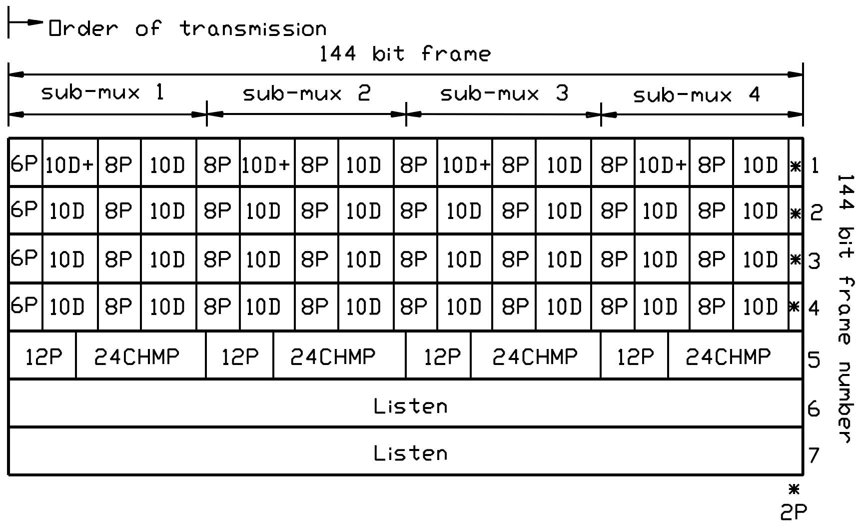
The use of CHM or SYNC in the SYN channel is covered in subclauses 5.2.1, 5.3 and 5.4.

Data bytes in the D channel are aligned in this multiplex so that the D channel synchronisation word, SYNCD, (see subclause 6.3) occurs as the first 16 bits in the D channel after the SYN channel (figure 6). The alignment of the data in the SYN channel is specified in figure 4.

In MUX2 the valid transmitted data bits in a burst are numbered 2 to 67. At the CPP antenna the start of transmission of valid data bit 2 of MUX2 shall occur 5,5 bit periods ±0,25 bit periods after the end of valid data bit 67 has been received (also measured at the CPP antenna).

5.2.4 Multiplex three

MUX3 (figure 5), carrying the D and SYN channels, is used for link establishment and re-establishment in the direction CPP to CFP. The B channel is nonexistent in MUX3.



NOTE 1: Transmission is continuous for five frames, then off for two.

NOTE 2: The 20 bits of D channel preamble are repeated in each of the four sub multiplexes before D changes.

NOTE 3: P is preamble in the SYN channel.

NOTE 4: The D channel synchronisation word (SYNCD) always begins at the start of slots marked +.

Figure 5: Mutliplex three

MUX3 repetitively transmits for 10 ms and receives for 4 ms. Responses from CFPs in MUX2 are detected during the receive slot. During transmission each sub-channel (D and SYN) is further sub-multiplexed by repetition four times over. This scheme enables the CFP (which in multiple-link systems is constrained to fixed receive slots) to be able to receive one of the sub-multiplexes of the SYN and D channels from the CPP.

The sub-multiplexed D channel comprises 20-bit words split into 10-bit sections surrounded by preamble in order to avoid CHM/SYNC emulation. The SYN channel contains a 12-bit preamble followed by a 24-bit CHMP.

Once the CFP has locked to the SYN channel of one sub-multiplex and recognised the correct ID in the corresponding D channel, the CFP shall then attempt to re-initialise the link (from its own end), using MUX2 with SYNCF in the SYN channel.

Data bytes in the D channel are aligned in this multiplex so that the D channel synchronisation word, SYNCD, (see subclause 6.3) occurs as the first sixteen bits in the D channel after the listen window. The alignment of the data in the SYN channel is specified in figure 5.

The MUX3 transmission shall be in accordance with the transmit envelope specified in figure 1.

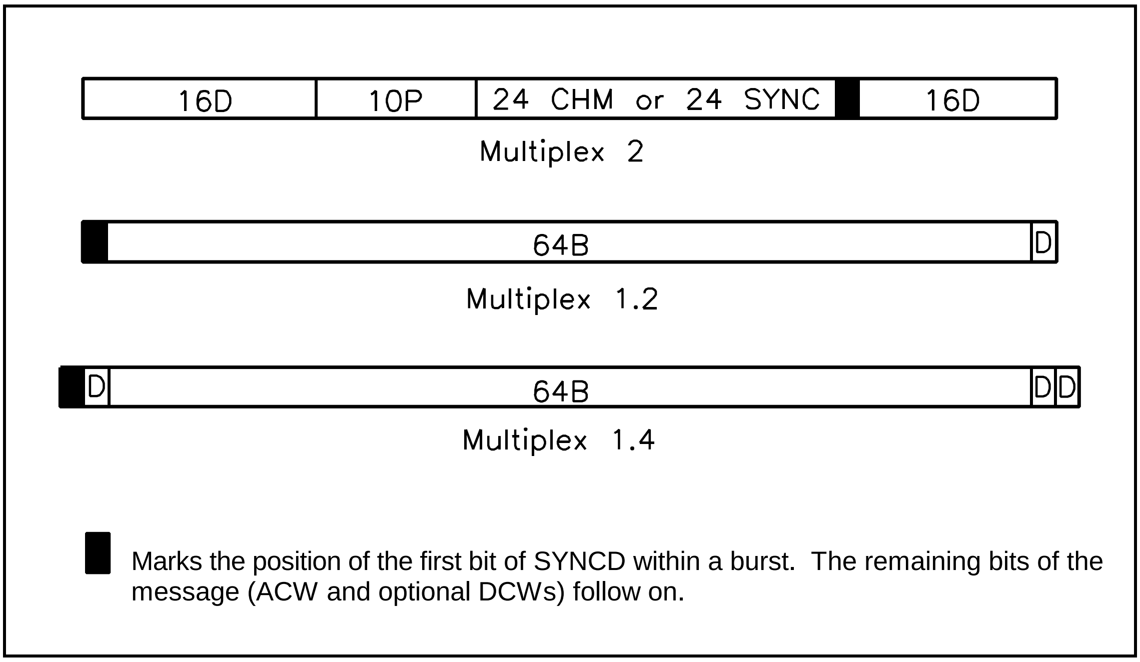


Figure 6: Signalling channel byte alignment

5.3 Calling channel detection

Calling channel detection operates in one of two ways depending upon the unit which originates the call request.

The two possibilities are that the CFP (master) originates the call which is then detected at the CPP (slave); or that the call originates at the CPP (slave) and is detected at the CFP (master). The CPP shall track the CFP bit timing clock once a link is established.

5.3.1 Calling channel detection at the CPP

The CPP expects to receive information formatted in MUX2. The process of calling channel detection on a single channel is shown in figure 7 and the sequence is as follows:

i) On command from the CPP control system, the RF synthesiser begins to switch to the new channel and eventually settles on the new centre frequency and begins receiving.

ii) The presence of D (or SYN) channel data from the CFP allows any AGC system to settle. During preamble the system gains bit synchronisation and then detects the CHM in the SYN channel of MUX2 from the CFP (CHMF in this case). D channel decoding then starts. Should CHMF not be detected within a suitable receive window then another channel may be examined.

iii) Only when CHMF has been correctly detected and a recognised ID found in the D channel, (LID and PID fields, see subclauses 6.4.3, 6.4.4 and 6.4.5), may a response be transmitted using MUX2 with SYNCP in the SYN channel and the contents of the LID and PID fields reflected back to the CFP.

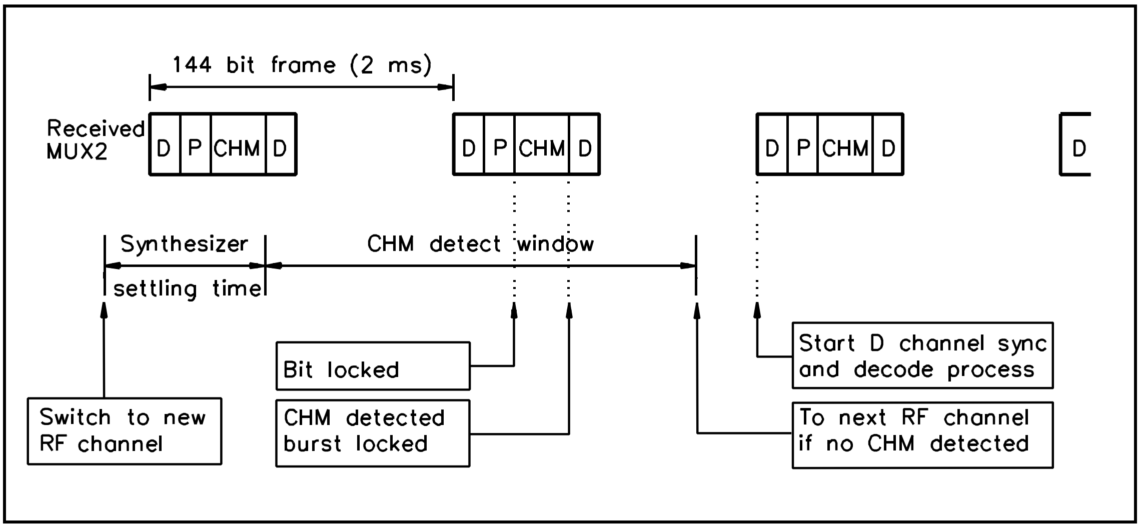


Figure 7: Calling channel detection and burst sync. at the CPP

Channel scanning continues until either a valid incoming call is detected and established or the user initiates an outgoing call.

5.3.2 Calling channel detection at the CFP

The CFP expects to receive information formatted in MUX3. The process of CHM detection is similar to that above except that the receive window is known, a priori, since the CFP acts as the frame timing master:

i) On command from the CFP control system, the RF synthesiser begins to switch to the new channel and eventually settles on the new centre frequency and begins receiving.

ii) The presence of D (or SYN) channel data from the CPP allows any AGC system to settle. During preamble the system gains bit synchronisation (temporarily acting as a slave for bit timing) and then attempts to detect CHMP in the SYN channel of MUX3. This is transmitted every seventh 144-bit cycle. D channel decoding then starts. Should CHMP not be detected within a suitable receive window then another channel may be examined.

iii) Only when CHMP has been correctly detected and a recognised link end point identification code found in the D channel LID field (see subclause 6.4.5), may the link be re-initialised from the CFP using MUX2 with SYNCF in the SYN channel, and with the CPP's own PID and a call reference value (assigned by the CFP) in the LID field. At this point the CFP re-gains master status for bit timing. The processing time for the CFP to decode the MUX3 information field and start sending SYNCF in MUX2 shall be less than Tcfp (18 ms, and recommended to be less than 4 ms to minimise the risk of response collision).

Channel scanning may continue until either a valid outgoing call is detected or the PSTN initiates an incoming call.

5.4 Link set up and establishment

Upon identification of the RF channel upon which a call is taking place and verification of compatible link-end identification, digital link set up can occur.

5.4.1 Link set up from CFP to CPP

Action at the CFP: When a link needs to be established to a CPP (e.g. in response to an incoming call indication), the CFP shall acquire a free RF channel and begin transmitting (and receiving) in MUX2 over that channel (with CHMF in the SYN channel and with PID and BID or LID in the D channel). The transmission shall last for a minimum of 1,4 s (Tfcyc) unless either a response from a recognised CPP is received (see subclause 6.6.7), or the link establishment timeout expires (Tfmax), or the link establishment request ceases. The three possible eventualities are:

i) the link is established when the CFP accepts a response from the target CPP with the correct ID code before the expiry of the transmit time;

ii) no recognised response is received before the expiry of the transmit time. In this case a new RF channel may be selected and a further link set up attempt may be made on the new channel;

iii) the link establishment is subject to a timeout period of sufficient length to cover the specified maximum quiet period of ringing cadence (see prETS 300 001, [1]). If no incoming ringing is detected for this timeout period, then link establishment shall cease. All link establishment requests other than those initiated by incoming ringing detection shall be subject to a single timeout period (Tfmax), or until the link establishment request ceases, whichever is shorter.

Action at the CPP: The CPP should normally check every channel periodically for the presence of CHMF. On detection of CHMF in the SYN channel and a matching PID in the D channel, the CPP shall respond using MUX2 with SYNCP in the SYN channel and either:

i) the PID and LID fields reflected back in the D channel; or

ii) the PID field reflected back and the poll decline LID in the D channel (see subclause 6.4.5, NOTE 3).

5.4.2 Link set up from CPP to CFP

Action at the CPP: When a link needs to be established to a CFP, the CPP shall acquire a free RF channel and begin transmitting in MUX3 (with CHMP in the SYN channel and PID, LID in the D channel) and listening for MUX2 over that channel. The transmission cycle period shall last for a minimum of 750 ms (Tpcyc) unless either a response from a recognised CFP is received (see subclause 6.6.6), the link establishment timeout expires (Tpmax), or the link establishment request ceases. The three possible eventualities are:

i) A one-way link is established when the CFP has detected the CHMP from the CPP and checks PID, LID.

The CFP then replies using MUX2 with SYNCF in the SYN channel, for a period between 56 ms and 84 ms (Tftx).

The CPP synchronises to the SYNCF of MUX2 during its receive slots in multiplex three. Immediately the CPP has detected SYNCF from a CFP, it ceases transmission of MUX3 and continues to receive the transmission from the CFP in MUX2. If the PID is subsequently detected, the CPP then responds in MUX2 with SYNCP in the SYN channel, otherwise the CPP repeats the link set up attempt from free channel acquisition. The processing time in the CPP to stop transmission of MUX3 when MUX2 has been detected shall be less than Tcpp (6,2 ms).

ii) No recognised response is received before the expiry of the transmit time. In this case a new RF channel may be selected and a further link set up attempt may be made on the new channel. This may be repeated for a total of five channel attempts.

iii) If a period of 5,0 s (Tpmax) expires before a link is successfully established, no further link set up shall be attempted until the initiation of a new call attempt.

Action at the CFP: The CFP should normally check every channel periodically for the presence of CHMP. On detection of CHMP in the SYN channel and a suitable end point ID in the LID field of the D channel, the CFP shall respond using MUX2 with SYNCF in the SYN channel and PID plus a link reference ID in the LID field of the D channel, for a period of between 56 ms and 84 ms (Tftx).

5.4.3 Set up collision resolution

There are situations under which set up collisions may occur. These are:

i) Call deadlock: A situation can exist where the CFP can be trying to establish a link to an associated CPP on RF channel X when the CPP is also trying to establish a link to the CFP on channel Y. Each end therefore may never get a response for a period of up to Tpmax. This may be overcome by the CPP reverting to channel scanning.

ii) Response collision: A CPP which has requested a given service will cause responses from all CFPs which provide this service and are in range. The probability of both initial and repeated collision of such responses is minimised by all CFPs starting their channel scanning algorithms at a random channel number and so generating a random time at which the CPP's call will be detected.

    If a collision does occur, the CPP will continue to call using MUX3 for the normal calling period and then select a new channel. CFPs, alerted by the failure of the CPP to respond in MUX2 to their MUX2 broadcasts shall re-scan for the CPP and in doing so shall re-start their channel scanning algorithms from randomly determined channel numbers.

5.4.4 Link re-establishment on the existing channel

Link re-establishment on the same RF channel may occur upon request from either end in an existing link. The method used is to cause the CPP to transmit in MUX3 with CHMP in the SYN channel in a similar fashion to normal call set up from the CPP. Link re-establishment is permitted only after a period of at least 300 ms for MUX1.4 and MUX2, and at least 600 ms for MUX1.2, from a previous link establishment or re-establishment.

After transmission or reception of a link re-establishment message, the CPP immediately switches to MUX3 and, using the last received link reference in the LID field, continues to transmit fixed format link request code words (see subclause 6.4) in MUX3 until either the link is re-established, or the 10 second handshake timeout (Thlost) matures (see subclause 5.5.1.6).

5.4.5 Link re-establishment on a different channel

After three seconds loss of handshake, link re-establishment may be attempted by the CPP on a different channel (see subclause 4.10). The action at the CPP may be to acquire a free channel (in accordance with the requirements of subclause 4.4) and transmit in MUX3 with CHMP in the SYN channel, but using the last received link reference in the LID field, in a similar fashion to link re-establishment on the existing channel from the CPP (see subclause 5.4.4). The action at the CFP shall then be to channel-scan continuously looking for CPP transmissions. Link re-establishment attempts shall cease when the 10 second handshake timeout (Thlost) matures (see subclause 5.5.1.6).

5.5 ID handshaking

5.5.1 General

The supplier shall declare that the ID handshaking strategy of the CTA complies with each subclause of subclause 5.5.1.

5.5.1.1 Handshake code series

This shall be a series of at least 2.5 million discrete codes.

5.5.1.2 Code allocation

Manufacturers shall allocate CPP codes either randomly from the range available and without duplication until the range is exhausted or sequently starting at a random point in the range. It shall not be possible for the user to program the handshake code at the CPP.

5.5.1.3 Code matching

The CFP and CPP shall use matching codes for handshake purposes.

NOTE: A CFP may be programmable to match the code(s) allocated to CPP(s). It is recommended that any user programming be a secure procedure.

5.5.1.4 Code recognition

The CFP and CPP shall establish mutual recognition by means of the handshake code before permitting communication.

5.5.1.5 Communication state

The handshake code shall be transmitted both ways between the CFP and CPP at least once per second during communication.

5.5.1.6 Lack of in-communication handshake: RF activity

If RF link conditions are such that greater than 10 s has elapsed without any successful handshake then the CTA shall cease RF activity.

5.5.1.7 Lack of in-communication handshake: off-line state

Following cessation of RF activity in a part of the CFP that was in the on-line state then that part of the CFP shall return to the off-line state within 1 s.

5.5.2 ID handshake operation

The CFP and the CPP both transmit ID handshake code words at a maximum rate of once every 400 ms subsequent to link set up (see subclauses 6.6.6 and 6.6.7), and a minimum rate of once every second. The CFP and CPP transmissions are asynchronous.

Note that the code words ID_OK, ID_LOST, LINK_REQUEST and LINK_GRANT (see subclause 6.4) all constitute valid ID handshakes.

5.5.3 Handshake protocol

The CPP(CFP) prepares to transmit a valid handshake after the handshake interval (Thtx) has elapsed since the start of the last transmitted handshake. If the CPP(CFP) has received a valid handshake from the CFP(CPP) within the last 1 second period, the CPP(CFP) transmits an ID_OK code word, otherwise it transmits an ID_LOST code word.

On reception of an ID_OK code word from the CFP(CPP), the CPP(CFP) shall restart a 10 second timer (Thlost).

The CFP(CPP) shall maintain a 1 second timer (Thrx) that is restarted on each reception of a valid handshake from the CPP(CFP). If the 1 second (Thrx) timer matures without reception of a valid handshake from the CPP(CFP), the CFP(CPP) shall transmit an ID_LOST code word (see subclause 6.4) instead of an ID_OK code word, and shall not restart its 10 s timer (Thlost).

If the CFP(CPP) subsequently detects a valid handshake from the CPP(CFP), the CFP(CPP) shall restart its 1 s timer (Thrx) and revert to sending an ID_OK code word (instead of ID_LOST).

If the handshake is lost for at least 3 seconds, the CFP or CPP may attempt link re-establishment on this or any other channel following the requirements of subclause 4.4.

5.5.4 Reception of valid handshakes

Reception of any valid handshake instead of an ID_OK handshake shall prevent the restarting of the Thlost timer at the receiver (i.e. CFP or CPP). The receiver shall not update its Thlost timer on further handshake transmissions, and the transmitter shall not update its Thlost timer unless an ID_OK handshake is received. If the timer Thlost expires, the link shall be terminated and the call lost. (Free channel acquisition for call re-establishment is permitted after 3 seconds loss of handshake, see subclauses 4.10 and 5.4.5).

Table 5: Handshake Timers

CFP	CPP	Timer	Timed Period
0,4 s to 1,0 s	0,4 s to 1,0 s	Thtx	Handshake interval
1,0 s	1,0 s	Thrx	Handshake loss period
10,0 s	10,0 s	Thlost	Handshake lost timer

There will always be a timing skew between the timer periods in the CFP and the CPP due to the transmission time of a message and software processing time at each end. This will normally be less than 500 ms.

5.5.5 ID handshake mechanism

The loss of handshake may be due to two reasons:

i) ID message to CPP is lost;

ii) ID message to CFP is lost.

The scenarios for both these cases are shown in the following diagram (figure 8):

```
              1. Message to CFP lost               2. Message to CPP lost

            CPP            CFP                  CPP            CFP
                |   ID_OK |                                  |  ID_OK   |
         /      |-------->|Tf:=10 s                   /      |-------->|Tf:=10 s
         |      |         |                           |      |         |
   h/s   |      | ID_OK   |                        h/s   |      | ID_OK   |
period <Tp:=10 s|<--------|                      period <Tp:=10 s|<--------|
         |      |         |                           |      |         |
         |      | ID_OK   |                           |      |   ID_OK |
         \      |--->|    |(Tf=9 s)                   \      |-------->|Tf:=10 s
                |         |ID_OK lost                        |         |
                | ID_LOST |\_________/               | ID_OK   |
       (Tp=9 s) |<--------| <-.                 (Tp=9 s) |    |<---|
                | ID_OK   |   .                ID_OK lost|         |
                |--->|    |(Tf=8 s)            \________/| ID_LOST |
                |         |   .                      .-> |-------->|(Tf=9 s)
                | ID_LOST |   .                      .   |         |
       (Tp=8 s) |<--------| <-.                      .   |   ID_OK |
                | ID_OK   |   .                 (Tp=8 s) |    |<---|
                |--->|    |(Tf=7 s)                  .   |         |
                |         |                          .   | ID_LOST |
                |         |                          .-> |-------->|(Tf=8 s)
       (Tp=7 s) |.   .   .|                          .   |         |
                |.   .   .|  .                           |   ID_OK |
                |.   .   .|  .                  (Tp=7 s) |.   |<---|
        TRANSMIT|.   .   .|  .                           |.   .   .|(Tf=7 s)
        MUX3    |>--->   .|  RECEIVE                     |>--->   .|
           .    |>--->   >|  MUX3               TRANSMIT |>--->   >| RECEIVE
           .    |>--->   >|  .                  MUX3     |>--->   >| MUX3
```

NOTE 1: Tf is the 10 s timer (Thlost) in the CFP.

NOTE 2: Tp is the 10 s timer (Thlost) in the CPP.

NOTE 3: ID handshake interval (Thtx) = 0,4 s in the absence of signalling messages (range is 0,4 s to 1,0 s).

NOTE 4: ID_OK lost timer (Thrx) = 1,0 s.

NOTE 5: Link re-establishment allowed when Tf <= 7 s.

Figure 8: Handshake loss scenarios

6 Signalling layer two

The services provided by signalling layer two are:

i) acknowledged and unacknowledged information transfer over the link (air interface);

ii) error detection;

iii) error correction by re-transmission and the correct ordering of messages (acknowledged operation only);

iv) identification of link endpoints (ID code handling);

v) link maintenance.

6.1 Code word usage

A code word shall be sent at least once every Trate (100 ms in MUX1.2, and every 50 ms in MUX1.4 and MUX2).

Implementations of this standard shall ensure that no more than 1 in 10^7 correctly transmitted code words is misinterpreted by the receiver when operating at a BER of better than 1 in 50.

6.2 General message format

Messages are passed between layer two and layer three by virtue of the DL_DATA and DL_UNIT_DATA primitives (see subclause 6.7). A layer three message may contain one or more layer three information elements (see subclauses 7.1 and 7.2). The breakdown of layer three messages into packets, and packets into code words is illustrated below (figure 9):

```
                +-----++- - -++- - -+                 +- - -+
                | ACW |. DCW .. DCW .                 . DCW .
                |     |.  1  ..  2  .  .  .  .  .  .  .  5  .
                +-----++- - -++- - -+                 +- - -+
                \_________________ _____.  .  .  .  ._______/
                                  \/
                 +------------------------------+     Maximum of six
LAYER TWO        |    PACKET containing 1-6     |     code words for
PACKET:          |        code words            |     each packet.
                 +------------------------------+

                  ________________________________. . . . ____________
                 /                                                    \
                 +-----------+      +-----------+        +-----------+
LAYER THREE      |  PACKET 1 |      |  PACKET 2 |        |  PACKET N |
MESSAGE:         |           |      |           |        |           |
                 +-----------+ . .  +-----------+ . . . . +-----------+

                 ^                  ^                    ^
                 |                  |                    |
                 ACW                ACW                  ACW
                 L3_end=1           L3_end=1             L3_end=0
                                                        (last packet)
```

Figure 9: General message format

6.3 General packet format

A packet is composed of one or more code words each of eight octets. In order to meet the handshake requirements of subclause 5.5, layer three messages are formatted into one or more packets each of up to six code words in length. Layer three messages greater than one packet in length use the L3_end bit (see subclause 6.5.2). The first code word in a packet is an "address" code word and any subsequent code words are "data" code words (if required for longer packets). The address code word is always preceded by a 16-bit synchronisation word, SYNCD. Each code word contains 16 check bits for error control. IDLE_D is sent only when there is no D channel packet to send. The sequence IDLE_D, SYNCD, address code word is shown below.

Bit:	8	7	6	5	4	3	2	1	octet
IDLE_D	0	1	0	1	0	1	0	1	n-2
	0	1	0	1	0	1	0	1	n-1
	0	1	0	1	0	1	0	1	n
SYNCD	0	0	1	0	0	0	1	1	1
	1	1	1	0	1	0	1	1	2
ADDRESS CODE WORD								1	1
									2
									3
									etc.

6.3.1 Order of transmission and field mapping convention

The octets are transmitted in ascending numerical order starting with octet one. Within an octet, bit 1 is the first to be transmitted followed by the remaining bits in ascending order.

A data value is contained within a field which may not correspond with an octet in size. When a field is contained within a single octet, the bit within the field having the lowest bit number (i.e. the first of the bits in the field to be transmitted) represents the least significant bit of the data value. When the field spans more than one octet, the least significant part of the data value is contained within the lowest numbered octet (i.e. the octet which is transmitted first). As an example, a 12-bit field is shown below.

Bit:	8	7	6	5	4	3	2	1	octet
	2^7							2^0	n
					2^{11}			2^8	n+1

The exception to this convention is the check field. In this case bit 1 of the first octet is the high order bit and bit 8 of the second octet is the low order bit (i.e. the bits arrive in descending order of significance).

6.3.2 IDLE_D

IDLE_D (one-zero reversals) is transmitted in the D channel when there is no signalling packet or supervisory fill-in packet (see subclause 6.5.7) to be sent. Note that IDLE_D does not necessarily map onto preamble in MUX2 or MUX3 (figures 4 and 5). The final bit of an IDLE_D sequence (the bit before the signalling channel synchronisation word SYNCD) shall always be a binary zero.

6.3.3 Synchronisation word (SYNCD)

Every packet begins with a 16-bit synchronisation word to enable the receiver to establish code word framing. The synchronisation word is recognised when all 16 bits have been received correctly.

6.3.4 Code Words - (address and data code words)

Packets are transmitted in 64-bit code words (8 octets), and each packet may occupy a number of code words. The first code word of a packet is an address code word and any subsequent code words are data code words. Bit 1, octet 1 of the code words are encoded 1 and 0 for address code words and data code words respectively. Two packet formats exist as indicated by the format type bit, FT (octet 1, bit 2). The format type; when set to 0 indicates a fixed length link set up and handshake ACW type used for link end point addressing and service requests; when set to 1 indicates the variable length packet format which is used to transfer link supervisory and layer three messages. (FT is only significant in ACWs and should be set to 1 in DCWs).

An example code word is shown in Annex G. The general code word format is shown below:

Bit: 8	7	6	5	4	3	2	1	octet
						FT=0/1 FT=1	1=ACW 0=DCW	1
								2
								3
								4
								5
								6
CHECK (octet 1)								7
CHECK (octet 2)								8

6.3.5 Code word transmission sequence.

If, in MUX2, it is intended to transmit two ACWs from the same logical transmitter source, the second ACW immediately following the first ACW (neither of which is a fill-in ACW), then, unless it is known that the two ACWs are different, the second ACW shall be preceded by either the fill-in ACW or by a minimum of 48 bits of IDLE_D.

If, in MUX1, it is intended to transmit two ACWs from the same logical transmitter source, the second ACW immediately following the first ACW (neither of which is a fill-in ACW), then, unless it is known that the two ACWs are different, the second ACW shall be preceded by the fill-in ACW.

6.3.6 Check field encoding (octets 7 and 8)

The 16 check bits (octets 7 and 8) are calculated in three steps:

i) 15 check bits are appended to the 48 information bits (octets 1 to 6) by encoding them in a (63,48) cyclic code. For encoding the 48 information bits may be considered to be the coefficients of a polynomial having terms from x^{62} down to x^{15}. This polynomial is divided by the generating polynomial:

$$x^{15} + x^{14} + x^{13} + x^{11} + x^{4} + x^{2} + 1.$$

The 15 check bits (octet 7 and octet 8 bits 1 to 7), correspond to the coefficients of the terms from x^{14} to x^{0} in the remainder polynomial found at the completion of the division.

ii) Bit 7 of octet 8 is inverted.

iii) Bit 8 of octet 8 is added such that the whole 64-bit code word has even parity.

Bit: 8	7	6	5	4	3	2	1	octet
							CRC msb	7
CHECK PARITY	CRC lsb\							8

6.4 Fixed length packet format (FT = 0)

The following fixed length format code word is defined for use in MUX2 and MUX3 for link initiation, and in MUX1 and MUX2 for handshaking.

Bit: 8	7	6	5	4	3	2	1	octet
HIC			0/1 SR	0/1 LS1	0/1 LS0	0 FT	1 ACW	1
HIC								2
HIC								3
MIC								4
LID								5
LID								6
CHECK (octet 1)								7
CHECK (octet 2)								8

6.4.1 SR (octet 1 (part))

Signalling rate request/response bit: Used by the call initiator to request a signalling rate (two or four bits per burst), and by the call receptor to set the signalling rate (prior to communication in MUX1.2 or MUX1.4) at the maximum rate compatible with both their capabilities. SR = 1 corresponds to MUX1.4 and SR = 0 to MUX1.2 (i.e. the default is MUX1.2 if either end requests MUX1.2).

6.4.2 LS (octet 1 (part))

Link status is a two-bit field that is used during call set up and handshaking. It defines the type of fixed format code word as follows:

Table 6: Coding of Link Status bits in handshakes

LINK STATUS LS1	LS0	MEANING
0	0	LINK_REQUEST
0	1	LINK GRANT
1	0	ID_OK
1	1	ID_LOST

LINK_REQUEST is transmitted as a request to "seize" the link. It is sent from CPP to CFP either in MUX3 as the first packet during CPP call set up and link re-establishment, or returned as a poll response in MUX2 from CPP to CFP during CFP call set up, when the CPP is attempting to answer the call.

LINK_GRANT is transmitted by the CFP subsequent to receiving a LINK_REQUEST from the CPP. It contains the link reference in the LID field, and this link reference is subsequently used in the LID field transmitted by the CPP. Reception of LINK_GRANT causes the handshake timers Thrx, and Thlost to be started (and restarted on further receptions of LINK_GRANT). In the CFP LINK_GRANT is transmitted continuously until either ID_OK is received or Tftx expires.

ID_OK is used as the handshake packet that is transmitted (at the handshake rate defined in subclause 5.5) when a valid handshake has been received within the last 1 second period. It is also used in link establishment from a CFP as the poll packet and as the poll response packet from a CPP.

ID_LOST is used as the handshake packet that is transmitted (at the handshake rate defined in subclause 5.5) when no valid handshake has been received within the last 1 second period.

For the duration of a link, the LID transmitted by the base and handset in ID_OK and ID_LOST handshakes; that in the LINK_REQUEST code word transmitted by the handset during link re-establishment; and that in the LINK_GRANT code word transmitted by the base during link re-establishment shall be the LID value originally transmitted by the base in the

LINK_GRANT code word at the time of link set up. LINK_REQUEST and LINK_GRANT code words shall only be transmitted during link establishment and link re-establishment.

6.4.3 HIC (octets 1 (part), 2, 3)

The handset identification code is a 19-bit field. Manufacturers are free to allocate the codes according to the procedures in subclause 5.5.1.2.

6.4.4 MIC (octet 4)

The manufacturer identification code is an 8-bit field. HIC plus MIC together form a 27-bit field PID (portable identity code) which acts as a link end point address. Allocation of the code is controlled by and registered with the Standard Control Authority body. On complete allocation of the 19-bit HIC field, the manufacturer shall obtain and use from the Standard Control Authority a further unique MIC to avoid duplication of PIDs.

Specific code allocations are detailed in a separate document (obtainable from the SCA).

6.4.5 LID (octets 5 and 6)

The link identification code is used for the following purposes:

i) End point identification for CPP call set up. In this instance the LID identifies a specific CFP or a requested service;

ii) Link reference for associating CPP and CFP call during handshake exchanges and link re-establishment so that communication is maintained with the initial end point only. For this reason it is recommended that the link reference value be different from the BID;

iii) Base Identifier (BID). This is a ringing address to which one or more CPPs, when appropriately programmed, will respond (during multiple ringing).

Table 7: Usage of the LID field

Link set up direction	Message direction	MUX mode	LID content
CPP to CFP	CPP -> CFP	MUX3	End point ID
	CFP -> CPP	MUX2	Link reference ID
	both ways	MUX2 MUX1	Link reference ID
CFP to CPP	CFP -> CPP	MUX2	Base ID (BID)
	CPP -> CFP	MUX2	Base ID (BID)
	both ways	MUX2 MUX1	Link reference ID
Link re-establish	CPP -> CFP	MUX3	Link reference ID (last received value)
	both ways	MUX2 MUX1	Link reference ID

LID values are allocated as follows:

Table 8: Values used in the LID field

LID Value, hex	CONTENT	NOTE
0000 to 03EF	Specific telepoint access values as defined in a separate document (obtainable from the SCA).	1
03F0 to 03FE	Reserved for future emergency access, allocated by the SCA.	-
03FF	Emergency access (CAI).	2
0400	Poll decline request.	3
0401 to FFFE	Link reference, BID.	4
FFFF	ID registration.	5

NOTE 1: Specific telepoint access. LID values for telepoint access are assigned in pairs, designated 2N and 2N+1. 2N values are used to specifically identify a target network, from which a CPP expects a response. 2N+1 values are used for roaming access, where the 2N part of the 2N+1 identifies the network with which the CPP is registered. This allows a receiving network to respond or not respond to a roaming request by determining whether or not the 2N part of the 2N+1 value is acceptable to the receiving network.

"Targeted" roaming is possible where a 2N LID value is used to target a specific telepoint network, even though the network is not the "home" network. In this situation, the home operator or network can be obtained by examining the OPSIC field in the AUTH_RES (subclause 7.2.9) or AUTH2_RES (subclause 7.2.18) layer three messages.

NOTE 2: Emergency access (CAI). This LID value shall be known to telepoint CFPs. CPPs attempting set up a call using this LID value shall not be obstructed by authentication or registration requirements and shall be connected to the default emergency service (see subclause 7.3.3).

NOTE 3: Poll decline request. This LID value may be used by a CPP which is being polled in order to "drop out" of the polling, i.e. not answer the call (see subclause 6.6.5).

NOTE 4: Link reference, BID. The BID (base ID) is a LID value which identifies a CFP. The link reference is a value which occupies the same range as BID and is used to identify an established link.

NOTE 5: ID registration. This LID is allocated for use by the CPP to enable it to set up a link for on-air registration to a CFP.

6.5 Variable length packet format (FT = 1)

A layer three message is formatted into a number of packets depending on the length of the message. In "single frame acknowledge" (CCITT Recommendations I.441 and I.451 [3]) mode, only one packet can be sent before the corresponding acknowledgement has been received. Received packets in one direction are acknowledged by using the N(r) bit in transmitted packets in the other direction. For transmission of acknowledged messages, DL_ESTABLISH_IND shall be set. DL_ESTABLISH_IND is set after receipt of SABM or SABM_ACK (see subclause 6.5.6). SABM may be sent at any time after link set up, and is independent of multiplex mode.

Address code words:

Bit: 8	7	6	5	4	3	2	1	octet
0	0/1 L3_end	0/1 endwrd	code word no. or rem.			1 FT	1 ACW	1
control								2
content								3
								4
								5
								6
CHECK (octet 1)								7
CHECK (octet 2)								8

Data code words (in packets only where appropriate to message length):

Bit: 8	7	6	5	4	3	2	1	octet
0	0	0/1 endwrd	code word no. or rem.			1 FT	0 DCW	1
content								2
								3
								4
								5
								6
CHECK (octet 1)								7
CHECK (octet 2)								8

6.5.1 PI (octet 1)

PI is the protocol identifier. This shall be set to 0 for basic operation, and set to 1 for enhanced operation. The protocol identifier shall be 0 for this application, (i.e. basic) meaning that the protocol used in communication is that defined in this document in conjunction with the control octet defined below. When PI is set to 1 then octet 2 is reserved for future applications.

6.5.2 L3_end (octet 1)

The L3_end bit is set to 1 to indicate that further layer three information follows in subsequent packets (ACW and any subsequent DCWs), and set to 0 for the last packet. If a message is transmitted in more than one packet then it is mandatory that acknowledged (numbered) packets be used. Supervisory messages are limited to one packet.

6.5.3 Endwrd and code word no/rem encoding (octet 1)

Bit 6, octet 1 (endwrd) is set to 0 in each code word except for the last code word of the packet (when endwrd is 1).

When endwrd is 0 then bits 3 to 5 of octet 1 (code word number) indicate (in binary) the number of data code words remaining in the packet; thus for a packet of three code words total, the first (address) code word has a code word no. of two, and so on.

When a code word has an endwrd value of 1, then bits 3 to 5 of octet 1 (rem) indicate the number of content octets remaining in that code word; thus if 1 octet only of the last code word contains significant supervisory or layer three information then rem is 1 (and any following octets in the code word are ignored).

6.5.4 Control (octet 2)

The control octet identifies the packet type and sequence number (where applicable). The use of this field is described below:

The control octet is used for link establishment and packet transmission for both unacknowledged and acknowledged packets with error correction by re-transmission.

Acknowledged (numbered) operation ensures that the layer two protocol delivers packets, numbered modulo 2 with the N(s) bit, in the correct order and without loss.

Acknowledgment (i.e. numbered operation) is via the N(r) bit in received packets from the other end of the link. The N(r) bit indicates the sequence number of the next expected numbered packet. If a re-transmission is required, the receiver may set the REJ bit, and sets the N(r) bit equal to the N(s) of the requested packet. Re-transmission using REJ may be used if, for instance, a DCW in a message is found to be corrupt (CRC failure) or if an unexpected DCW is received. (i.e. not preceded by an ACW). Note that both layer three and supervisory packets share the same (layer two) numbering sequence.

Unacknowledged (unnumbered) operation delivers packets directly between peer layers without invoking the layer two re-transmission protocol. For example, unacknowledged operation must be used when alerting (multiple) CPPs.

NOTE: Receipt of one or more unacknowledged (un-numbered) packets between the numbered packets of a multi-packet layer three message shall not disrupt the reception of the numbered message. The reception of an unacknowledged (un-numbered) packet cannot be guaranteed.

Two packet types are identified; a link supervisory type and an information type. The bit allocation shall be:

Bit: 8	7	6	5	4	3	2	1	octet
0	0 QUALIFIER	0	I/S	N(s)	P/F	N(r)	REJ	2

Qualifier: Set to 0 (reserved for future use).

I/S: 1 = information type (layer three),

0 = supervisory type (layer two or one).

This bit when set to 1 indicates that this packet contains layer three information and when set to 0 indicates that the packet contains link supervisory data.

N(s): Send sequence number. N(s) is don't care when P/F = 0.

P/F: Poll/final bit (CCITT Recommendation I.441 [3]). When set to 0 this indicates unacknowledged operation, i.e. no response required. When set to 1 this indicates acknowledged operation (N(s) is significant).

N(r): Receive sequence number. Used for acknowledgment of received packets.

REJ: Reject bit. When set to 1 it indicates that a received packet was rejected. N(r) is equal to the N(s) value of the rejected packet.

Table 9: Coding of Control Octet values

P/F	I/S	PACKET CONTENT
0	1	Unacknowledged layer three packet
1	1	Acknowledged layer three packet
0	0	Unacknowledged supervisory packet
1	0	Acknowledged supervisory packet

N(s) and N(r) are used for packet acknowledgment. N(s) refers to the sequence number of the packet being sent, and N(r) indicates the sequence number of the next expected numbered packet.

6.5.5 Content

The octets marked content in the ACW and DCWs of packets are destined to carry layer three information as described in Clause 7 but are also capable of carrying link supervisory data. Unused octets in ACWs and DCWs shall be filled with 0F0H. The link supervisory messages are described below.

6.5.6 Link supervisory "messages"

The link supervisory messages that follow are defined. All others are reserved for future allocation by the Standard Control Authority. Link supervisory messages have the following format:

Bit: 8	7	6	5	4	3	2	1	octet
Message type			Message content					X

The following link supervisory messages are defined:

i) Transmit power level control

This acknowledged message sets the transmit power level of the recipient's link end. It may be ignored by the CFP.

Bit: 8	7	6	5	4	3	2	1	octet
0	0	0	0	0	0	0	tgain	X

Table 10: Coding of Transmit Power Control

tgain	TRANSMIT POWER LEVEL
0	Full power
1	Low Power

ii) Link re-establish on a given channel

This message may be sent from CPP or CFP. It shall cause an immediate attempt at link re-establishment on a given channel number.

If the channel number is 0, this message shall cause link re-establishment on the same (current) channel. The message is sent unacknowledged from either end.

A channel number in the range 1 to 40 shall cause link re-establishment attempts on the specified channel where values 1 to 40 correspond to the lowest to highest RF frequencies (see subclause 4.2). The message is sent unacknowledged and from the CFP only, subject to the provisions of subclauses 4.4.3, 4.4.4 and 4.10.2.

Channel numbers 41 to 63 are invalid and have no effect.

NOTE: This effectively uses two link supervisory message types.

Bit: 8	7	6	5	4	3	2	1	octet
0	1	Channel Number						X

iii) Protocol initialisation (SABM)

The SABM message is an unacknowledged message that is sent from the CPP to cause protocol initialisation at the receiving end of the link, during link establishment only. The CFP shall not send SABM.

NOTE: Where a new CFP accepts a re-established link, it is the responsibility of the network to correctly set the state variables in the new CFP so that the acknowledged messaging is correctly resumed. In this situation, the new CFP will have to be informed of the LID (link reference) value to expect for the re-establishment to occur, and the new CFP must in addition be informed of the state variable values and any unfinished layer three business.

SABM messages are transmitted continuously from the CPP until a SABM_ACK message is received. V(r) shall be initialised to zero before SABM is sent. Upon receiving a SABM message at either CFP or CPP, the local state variables V(s) and V(r) shall be initialised to zero, DL_ESTABLISH_IND shall be set and an acknowledgment returned in the form of a SABM_ACK message. It is mandatory for a CPP to transmit SABM messages. The N(r) bit in the SABM message shall be set to 0, to prevent possible mis-interpretation of the received message. Protocol initialisation is shown below (figure 10):

```
CPP                                  CFP
===                                  ===
V(r) := 0       SABM
          ----------------->
                                  V(s) := 0
                                  V(r) := 0
              SABM_ACK            DL_ESTABLISH_IND := true
          <-----------------
V(s) := 0
DL_ESTABLISH_IND := true
```

Figure 10: CFP to CPP link set up

V(s) is the send state variable; this denotes the sequence number of the next packet to be transmitted i.e. N(s) of transmitted packets := V(s).

V(r) is the receive state variable; the sequence number of the next packet expected. N(r) of transmitted packets := V(r). V(r) is checked against the N(s) field of incoming packets to make sure they are in sequence. An out of sequence packet shall be ignored.

Bit: 8	7	6	5	4	3	2	1	octet
1	1	0	0	0	0	0	0	X

iv) Protocol acknowledgment (SABM_ACK)

This is an unacknowledged message that is used to acknowledge the link initialisation (SABM) message. It shall be transmitted back to the other end of the link after a SABM message has been received. On reception of a SABM_ACK message, transmissions of SABM are ceased, the local state variable V(s) is initialised and DL_ESTABLISH_IND is set (see the diagram above).

Bit: 8	7	6	5	4	3	2	1	octet
1	0	1	0	0	0	0	0	X

6.5.7 Fill-in

This is an unacknowledged supervisory message of zero length, contained in a variable-format ACW, that is used as fill-in when there are no signalling messages to send. It maintains traffic on the D channel for bit error ratio monitoring. This message is sent in preference to the IDLE_D pattern. The full fill-in code word is therefore as follows:

Bit: 8	7	6	5	4	3	2	1	octet
0 PI	0 L3_end	1 endwrd	0 rem.	0	0	1 FT	1 ACW	1
0 QUALIFIER	0	0	0 I/S	X N(s)	0 P/F	1/0 N(r)	1/0 REJ	2
1	1	1	1	0	0	0	0	3
1	1	1	1	0	0	0	0	4
1	1	1	1	0	0	0	0	5
1	1	1	1	0	0	0	0	6
CHECK (octet 1)								7
CHECK (octet 2)								8

6.6 Link establishment and re-establishment

Link set up uses sequences of single address code words. The minimum transmission time for a single ACW + SYNCD (which precedes every ACW) is three bursts, i.e. 6 ms but subject to the conditions of subclause 6.3.5 may be greater. This is the basis of the timings in the following text.

NOTE: The LID field for group call identification is as explained in the third note of subclause 6.4.5.

6.6.1 Link set up from the CPP

The CPP shall transmit LINK_REQUEST code words (LS = 00). Each code word contains the PID and end point ID code (in the LID field) necessary to access the required service. This fixed format ACW shall be transmitted in MUX3. The SR bit shall be set (=SRr) to request the maximum signalling bit rate of which the CPP is capable.

This fixed format ACW is repeated continuously for Tpmax (5 s) maximum with the CPP monitoring for a CFP response in MUX2.

The CFP detects the MUX3 CHMP from the calling CPP during its calling channel scan, and subsequently receives the ACW described above. If the received PID and LID are valid the CFP responds to the ACW from the CPP. The CFP examines the SR bit received from the CPP, and if it is capable of signalling in multiplex 1 at the requested rate, it sets SR (=SRc) equal to SRr, otherwise it sets SRc=0. The CFP then transmits LINK_GRANT code words (LS = 01), for the period Tftx, containing a link reference in the LID field for the CPP to use in future communications. If the received PID and LID are not valid, the CFP shall not respond and may return to calling channel scanning.

The CPP detects the CFP MUX2 transmission, and receives the LINK_GRANT code word. The CPP then starts its handshake timers Thtx, Thrx, and Thlost, and transmits an ID_OK handshake code word, echoing back the link reference (in the LID field) received from the CFP. On receipt of the ID_OK handshake code word, the CFP starts its handshake timers Thtx, Thrx, and Thlost, and stops transmission of LINK_GRANT code words. The CPP shall transmit an ID_OK code word on each reception of a LINK_GRANT code word, subject to the conditions of subclause 5.5.3.

The CFP waits a minimum of Tfdetect (100 ms) for a CPP ID_OK handshake code word (before reverting to channel scanning). It checks the received LID against its own link reference value, and if they are different the CFP returns to calling channel detection mode. If the received LID matches, then the link is set up, and protocol initialisation may be attempted. Protocol initialisation is performed by the CPP transmitting SABM packets. When acknowledgment (SABM_ACK) is received, the link state variables V(s) and V(r) are initialised, and DL_ESTABLISH_IND is set.

6.6.2 Link re-establishment

Link re-establishment is always performed by the CPP transmitting in MUX3, and the CFP listening in MUX3, and the mechanism is similar to call set up from the CPP.

Link re-establishment may be due to the following reasons:

i) command to re-establish on same channel has been transmitted;

ii) command to re-establish on same channel has been received;

iii) command to re-establish on a given channel has been received;

iv) ID handshake loss.

The CPP, when transmitting in MUX3, inserts CHMP in the SYN channel and the last received LID from the CFP in the D channel. This value of LID distinguishes a link re-establishment from a new link set up.

6.6.3 Link set up from the CFP

Ringing can be activated to either a single CPP or multiple CPPs within a registered group. The mechanism is identical in either case in that a single CPP is treated as a "multiple" group limited to one. Multiple ringing is a point to multi-point (or single point for a group of one) link set up attempt that reverts to a point to point link only when the CFP detects a response from a CPP when the CPP is requesting the link.

There are five requirements for multiple ringing:

i) packets from CFP to CPP during ringing shall be restricted to single ACW packets;

ii) poll packets from the CFP containing the BID or LID shall be repeated every other packet (ACW), with a single ACW (UI) or the equivalent time of IDLE_D between them to enable the CPP to detect the poll. Successive poll packets shall not occur more frequently than once every 12 ms;

iii) the poll response from the CPP shall be a single ACW;

iv) the poll response ACW shall be transmitted from the CPP in a transmit burst following the end of the receive burst containing the poll packet. The start of the poll response (the burst with 16 bits of IDLE_D and SYNCD) shall occur between 2 ms and 8 ms after the end of the poll (to allow time to set up the poll response ACW for the transmit burst). The end of the response occurs 5 ms after the start, i.e. within 13 ms of the end of the poll;

```
     ______    ______    ______    ______     ______    ______
    /CPP RX\__/CPP TX\../CPP RX\__/CPP TX\.../CPP TX\../CPP RX\..
    \______/  \______/  \______/  \______/   \______/| \______/
           |\__...___   ___...__/  |                 |
End of poll^          \/           ^Start of poll    ^End of poll
                  2 ms...8 ms       response          response
              \__________________  __________________/
                                 \/
                            13 ms maximum.
```

Figure 11: Poll response timing

v) the polling from the CFP is performed on a cyclic basis, with only one poll per CPP per cycle, and with the relative order of poll position maintained.

For incoming public access to handsets, the public access LID values (0000H to 03EFH) may be used by the base station. If it is intended to subsequently authenticate the handset, the LID value used by the base shall be the LID value that is used by the handset to access that public access service.

6.6.4 CFP polling

The CFP transmits ID_OK code words to each CPP that is required to be called within a group. This poll serves a dual purpose:

i) to provide a "group calling" address using a BID or LID to enable all CPPs registered with that CFP to recognise this BID and so wait on the channel for reception of an individual CPP poll. This poll waiting timeout is Tpid (384 ms);

ii) to address, or poll each CPP in the group calling list individually with a PID so that each CPP can respond to its poll in the appropriate response slot.

The CFP transmits a fixed format poll ACW and an optional variable format packet (e.g. alternately SIG (ring on/off) and FI) in MUX2 on a continuous repeated basis within a Tpid (384 ms) period (e.g. POLL1, SIG, POLL2, FI, POLL3, SIG, ..., POLLn, FI). A CPP shall act on unacknowledged layer three (e.g. SIG and FI) packets received only after it has recognised its own PID from a poll.

If a CFP transmits UIs between polls (instead of FILL_IN), and the UIs include SIG, FI and NO_POLL, then the recommended sequencing of the UI messages between polls is SIG, FI, SIG, NO_POLL, SIG, FI, SIG, NO_POLL. This is to allow rapid ring cadence following by polled CPPs.

The CFP transmits poll,(SIG, FI or NO_POLL) packets continuously (using CHMF in the SYN channel) so that all registered CPPs in the ringing list are polled, and monitors for CPP responses. The timeout for call set up is given in subclause 5.4.1. The CPP transmission of responses is constrained by a transmit window that is the period between poll packets from the CFP; a particular CPP's response window starts 2 ms after it has been polled, to avoid CPP poll response collisions.

The ID_OK poll contains the BID or LID in the LID field (for group call identification at the CPP) and a PID (for one of the group registered CPPs). The SR bit is set to indicate the MUX1 signalling rate capability (SR=SRr).

The CPP detects the MUX2 CHMF from the CFP during its calling channel scan. It waits for Tbid (19 ms) maximum to receive an ID_OK poll (ACW). If a SIG or FI packet is received before a poll, it is ignored. If a poll is received, the PID in the poll is compared with the stored PID. If they match, the CPP starts a 1 second poll timer (Tpoll), and transmits an ID_OK poll response back to the CFP within the response "time-slot", and may act on subsequently received layer three UI packets (e.g. SIG or FI). If the PID does not match, the received BID or LID in the LID field is checked with the stored BID or LID, and if they match, this "arms" the CPP to continue to monitor the channel for up to Tpid (384 ms) to detect its own PID (if it does not detect its own PID it continues to channel scan). If the CPP subsequently receives a poll containing its PID, this enables the CPP to decode all subsequent UI packets (e.g. SIG, to start ringing). A CPP which has not received further polls within 1 second (Tpoll) shall revert to channel scanning.

6.6.5 CPP poll response

The CPP has three possible responses to a poll from the CFP:

i) the normal CPP poll response shall be an ID_OK code word, transmitted using SYNCP in the SYN channel and containing the PID and LID field. The LID field contains the value of the LID field received from the CFP (i.e. the CPP reflects the LID field back to the CFP);

ii) a LINK_REQUEST code word (LS=00) which is sent after the CPP user takes the call, or a CPP determines it can accept the link as it is the only CPP being polled (see subclause 7.2.19);

iii) a poll decline request. This is an ID_OK code word with the LID field containing a value of 0400H.

The CFP monitors for MUX2 (with SYNCP in the receive SYN channel) to detect CPP responses to its polling. If the CPP did not answer the call, but transmitted an ID_OK code word, the CFP logs the response (for call establishment not due to incoming ringing, the CFP halts its Tfmax (5 s) call set up timer here, see subclause 5.4.1. For call establishment due to incoming ringing, Tfmax is halted when the correct layer three feature activation (FA) information element (subclause 7.2.4) is received by the CFP). If all CPPs on the current group ringing list have responded to their polls, the CFP replaces CHMF by SYNCF in the transmit SYN channel to prevent unnecessary wake-up of other CPPs in range.

If a CPP user attempts to decline the link, the CPP sends a "poll decline" request every time it is polled; the CPP should remain on the channel until Tpoll expires, to ensure it is no longer being polled by the CFP.

Whenever the CFP detects an ID_OK poll response, the CFP starts a 1 second timer (Tfpres); if this timer expires, the CFP is required to scan for a free channel, and restart the polling process as if starting a new link set up (for incoming ringing), or terminate the link set up attempt (when not due to incoming ringing). When the CFP accepts a LINK_REQUEST packet, the TFPRES timer shall be halted, and the timers Tftx and Tfdetect started.

If the CPP attempts to answer the call by transmitting a LINK_REQUEST code word, the SR bit in the LINK_REQUEST code word indicates the MUX1 rate selected (commanded) by the CPP (SR=SRc). The CFP subsequently transmits LINK_GRANT code words for a period of Tftx, and starts the Tfdetect timer, the period during which it expects to receive an ID_OK code word from the CPP. If Tfdetect matures, the CFP reverts to starting a new link set up to the CPPs (for incoming ringing), or terminates the link set up (when not due to incoming ringing). The CFP is then armed to signal at the commanded MUX1 rate. MUX1 is entered by means of a layer three message.

The CPP shall respond to LINK_GRANT code words by transmitting ID_OK code words, and shall start or restart its handshake timers Thrx, Thtx and Thlost.

The CPP only sends LINK_REQUEST code words on receipt of ID_OK polling code words from a CFP, containing the CPP's PID.

6.6.6 CPP link set up and re-establishment diagram

```
                           CPP                          CFP
          T3R2:            ===                          ===              R3:
                             | (CHMP,MUX3)PID,LID,LS=00  | MUX3 detected, PID
                            A| ==== LINK_REQUEST ======> |A' & LID validated.
                             |                           |
                             | (SYNCF,MUX2)PID,LID,LS=01 |             T2R2:
          MUX2 detected.   B| <=== LINK_GRANT ========= |
          T2R2: Set SR.      |                           |
                             | (SYNCP,MUX2)PID,LID,LS=10 |
                            C| ===== ID_OK ============> |B'
                             .                           .

The protocol shall be initialised from the CPP (during link establishment only):
                             .                           .
                             |      SABM  (SUP)          |
         DL_ESTABLISH_REQ:   | ========================> | DL_ESTABLISH_IND
                             |      SABM_ACK (SUP)       | Set V(s):=V(r):=0
         DL_ESTABLISH_IND,   | <======================== |
         Set V(s):=V(r):=0   |                           |
                             .                           .

A :  Start Tpcyc (min=750 ms, see subclause 5.4.2), for link establishment or re-
establishment on another channel.

A':  Start Tfdetect, Tftx.

B :  Halt Tpcyc and Tpmax, Start Thlost, Thrx.

B':  Start Thrx, Thtx and Thlost. Halt Tfdetect, Tftx.  The link is set up at this
point.

C :  Start Thtx.
```

Figure 12: CPP link set up

NOTE 1: After points B, B' the transmitter must start transmission of ID_OK handshakes such that an ID_OK is sent within 400 ms to 1 s of the last transmitted handshake.

NOTE 2: Up to point A', the LID field contains a link endpoint ID code, and from point A', the CFP replaces the link endpoint ID with a link reference ID (see subclause 6.4).

NOTE 3: During call re-establishment LID contains the last used link reference value.

NOTE 4: At point A during link re-establishment on the same channel, the CPP and CFP continue to transmit and receive respectively until the (minimum) 3 s loss of handshake timer matures, where-upon re-establishment on a different channel may occur as described above, i.e. Tpcyc is not active during link re-establishment on the same channel.

6.6.7 CFP link set up diagram

```
                       CFP                           CPP
     T2R2:             ===                           ===               R2:
                        |                             |    CPP1,2 out of range
                        |*(CHMF,MUX2)PID1,LID,LS=10   |   +--------------------+
           Poll CPP1:  A| ========= ID_OK ==========> |   |If ALL CPPs respond |
                        |                             |   |to POLL message send|
                        |*(CHMF,MUX2),UI              |   |SYNC not CHM. *     |
           Ring ALL:    | ---------------------------> |   +--------------------+
                        |                             |
                        |*(CHMF,MUX2)PID2,LID,LS=10   |    NOTE: Assumes CFP has
           Poll CPP2:   | ========= ID_OK ==========> |    ID codes for 2 CPPs
                        |                             |    programmed.
                        |*(CHMF,MUX2),UI ++           |
           Ring ALL  -> | ---------------------------> |    ++ UI e.g.: SIG/FI
                        . . . . . . . . . . . . . . . .    CPP1 in range
                        .                             .
                        |*(CHMF,MUX2)PID1,LID,LS=10   |                T2R2:
                        | ========= ID_OK ==========> |A'  CPP1 detects poll
                        |                             |    from CFP.-------+
                        |(SYNCP,MUX2)PID1,LID,LS=10   |                    |
        Poll response  B| <======== ID_OK =========== | <- CPP1 in range <-+
        detected.       . . . . . . . . . . . . . . . .
                        .                             .
                        |*(CHMF,MUX2)PID1,LID,LS=10   |    CPP1 LINE key pressed
           Poll CPP1:   | ========= ID_OK ==========> |A'           |
                        |                             |             |
                        |(SYNCP,MUX2)PID1,LID,LS=00   |             |
        Call seized    C| <==== LINK_REQUEST ======== | <-----------+
        by CPP1.        |                             |
                        |(SYNCF,MUX2)PID1,LID,LS=01   |
                        | ===== LINK_GRANT =========> |D'
                        |                             |
                        |(SYNCP,MUX2)PID1,LID.LS=10   |
                       D| <===== ID_OK ============== |
                        .                             .


The protocol shall be initialised from the CPP (during link establishment only):


                        .       SABM  (SUP)           .
     DL_ESTABLISH_IND,  | <========================== | <--- DL_ESTABLISH_REQ:
     Set V(s):=V(r):=0. |       SABM_ACK (SUP)        |
                        | ==========================> | ---> DL_ESTABLISH_IND-->
                        |                             |      Set V(s):=V(r):=0
                        |                             |

A :  Start Tfcyc (min=1,4 s, see subclause 5.4.1)

A':  Start (and re-start) Tpoll (see subclause 6.6.4)

B :  Halt Tfcyc, Start (and re-start) Tfpres.  For Tfmax actions, see subclause 6.6.5.

C :  Start Tftx and Tfdetect.  Halt Tfpres.

D':  Start Thrx, Thtx. Start (and re-start) Thlost, Halt Tpoll

D :  Start Thrx, Thtx and Thlost.  Halt Tftx and Tfdetect.  The link is set up at this
point.
```

Figure 13: CFP link set up

NOTE 1: After point D the transmitter must start transmission of ID_OK handshakes such that an ID_OK is sent within 400 ms to 1 s of the last transmitted handshake.

NOTE 2: Up to point C, the LID field contains the BID, and from point C, the CFP replaces BID with the link reference ID (see subclause 6.4).

6.7 Layer two link protocol set up and control

The layer two set up and control is performed using the control octet of the address code word. The protocol is based on CCITT Recommendation I.441 LAPD single frame operation [3]. The frames referred to as SI0/1 shall be used for information transfer and link control. The acknowledged mode of information transfer shall require acknowledgment. The content of the four primitives DL_DATA_REQUEST, DL_UNIT_DATA_REQUEST, DL_DATA_INDICATION and DL_UNIT_DATA_INDICATION shall be a layer three message. This may contain one or more layer three information elements (see Clause 7).

NOTE: This list is intended for guidance with individual manufacturers free to add to at their discretion, to achieve refinements to functions (including clear down), as required.

DL_ESTABLISH_REQUEST	Layer three request for protocol initialisation.
DL_ESTABLISH_INDICATION	Layer two acknowledgement of protocol initialisation.
DL_DATA_REQUEST	For sending acknowledged data from layer three to layer two.
DL_DATA_INDICATION	For sending acknowledged data from layer two to layer three.
DL_UNIT_DATA_REQUEST	For sending unacknowledged data from layer three to layer two.
DL_UNIT_DATA_INDICATION	For sending unacknowledged data from layer two to layer three.
DL_RELEASE_INDICATION	For sending release indication to layer three from layer two.
DL_RELEASE_REQUEST	For sending release request to layer two from layer three.

6.7.1 Layer two link protocol timings

There shall only ever be one outstanding packet waiting for acknowledgement in the acknowledged transfer mode (i.e. k=1 in CCITT Recommendations I.441 and I.451 [3]). A timeout (Trtx) shall be maintained for re-transmission and this shall be not greater than:

i) 600 ms in MUX1.2; or

ii) 320 ms in MUX1.4; or

iii) 66 ms in MUX2.

The timer is started on completion of transmission of the packet. Receipt of a packet with REJ set shall cause this timer to mature immediately, and be re-started after re-transmission of the packet.

6.7.2 Layer two link operation

i) V(r) is the receive state variable which signifies the next expected N(s);

ii) V(s) is the send state variable which signifies the next N(s) to be transmitted;

iii) N(r) is used for acknowledgement of received packets.

6.7.2.1 Link initialisation (see subclause 6.5.6 (iii))

Set up the link by setting V(r) := 0 and transmitting SABM continuously with:

I/S := 0, N(s) := 0, N(r) := 0, P/F := 0.

a) If first SABM_ACK received:

   stop transmitting SABM, V(s) := 0.

   set DL_ESTABLISHED_IND.

   start transmitting in normal mode;

b) or if SABM received:

   V(s) := 0, V(r) := 0.

   set DL_ESTABLISHED_IND.

   transmit SABM_ACK;

c) or if second or subsequent SABM_ACK received:

   no action.

6.7.2.2 Packet transmission

1) Check:

   a) if DL_UNIT_DATA_REQUEST is set:

      send packet: P/F := 0, N(s) := X, N(r) := V(r);

   b) or if DL_DATA_REQUEST is set:

      send packet: P/F := 1, N(s) := V(s), N(r) := V(r);

      start re-transmission timer (Trtx).

2) If re-transmission timer (Trtx) matures:

   send packet: P/F := 1, N(s) := V(s), N(r) := V(r);

   re-start re-transmission timer (Trtx).

6.7.2.3 Packet reception

1) IF PACKET NOT RECOGNISED (PI=1):

Ignore packet, no action.

2) IF PACKET RECOGNISED (PI=0):

i) First:

a) if I/S = 1, P/F = 0:

set DL_UNIT_DATA_INDICATION;

un-numbered packet (no acknowledgment);

pass to layer three;

b) or if I/S = 0, P/F = 0:

un-numbered supervisory packet;

terminate at layer two;

c) or if I/S = 1/0, P/F = 1:

numbered packet.

IF DL_ESTABLISH_INDICATION is set

THEN

IF N(s) = V(r)

THEN

increment V(r);

send next packet with REJ := 0.

IF I/S = 1

THEN

IF L3_end = 0

THEN set DL_DATA_INDICATION

ELSE await further packet

ELSE

terminate at layer two.

send acknowledgement

with N(s) := V(s), N(r) := V(r).

ELSE (i.e. N(s) <> V(r))

send normal response

with N(s) := V(s), N(r) := V(r).

ELSE

Ignore packet.

ii) Second:

IF REJ = 0

THEN

IF V(s) <> N(r) (in sequence N(r): acknowledgement)

THEN

increment V(s).

stop re-transmission timer (Trtx);

ELSE (Out of sequence N(r))

do nothing, no action.

ELSE (i.e. REJ = 1)

IF DL_ESTABLISH_INDICATION AND V(s) = N(r)

THEN

re-transmission request, so complete transmission of the packet and then re-transmit the packet immediately (subclause 6.7.2.2).

ELSE

no action

iii) Third: Bad packet received (see subclause 6.5.4):

to request re-transmission of the last received packet without the transmitter waiting for Trtx to mature, the REJ bit in the ACW control field is set. This is received at the other end of the link and causes a repeated transmission of the last packet. Send next packet with REJ := 1.

7 Signalling layer three

Layer three carries the signalling messages destined for the PSTN and vice versa, plus call control messages within the CT2 internal network. The stimulus mode signalling procedures described below are appropriate for both circuit switched voice calls and circuit switched data calls. The signalling channel is not intended to carry packet data.

Stimulus mode signalling is the most appropriate mode for the CT2 application and is described in CCITT Recommendation Q.931 [4] as follows: "signalling messages sent by stimulus mode terminals to the network are usually generated as a direct result of actions by the terminal user (e.g. handset lifted) and in general do little more than describe the event which has taken place at the man-machine interface. Similarly, signalling messages sent by the network to terminals operating in stimulus mode contain explicit instructions regarding the operations to be performed by the terminal (e.g. connect B channel, start alerting, etc.)".

The signalling system specified below is a simplified version of that described in CCITT Recommendation Q.931 [4]. The simplification involves the removal of fields not required in the CT2 application; viz. the protocol discriminator, call reference value and message types. The CT2 system uses a subset of the information fields of CCITT Recommendation Q.931 [4] (which have been re-coded), plus new information fields not found in CCITT Recommendation Q.931 [4]. The information element principles of CCITT Recommendation Q.931 [4] have been preserved and exist in both single octet and variable length form. A layer three message is defined as a group of information elements delivered error free by layer two to the far end. Unless explicitly stated, messages may be sent either acknowledged or unacknowledged at layer two. The information element grouping in a message is determined by actions at the user interface at the CPP and the call state at the CFP (see Annexes A and F). The maximum layer three message length is 29 octets in equipment conforming to this standard. Single octet and variable length information elements are defined below.

It is not mandatory to provide all of the detailed signals defined in this document but any equipment implementing implied functions that are covered here shall use the appropriate signals as defined. Those layer three messages that are not implemented or not recognised shall be ignored and shall not cause mis-operation.

7.1 Single octet information elements

The single octet information element is encoded as shown in the diagram below:

Bit: 8	7	6	5	4	3	2	1	octet
1	information element identifier			contents of information element				X

One single octet information element is encoded to allow a shift into alternative signal code sets:

Bit: 8	7	6	5	4	3	2	1	octet
1	0	0	1	U	new code set identifier			X

When U (bit 4) is set to 0 this indicates locking shift and the specified code set is activated for all information elements in the rest of the message. When set to 1 this indicates non-locking action and that only the next information element is taken from the specified code set. The default code set is that specified in this document. Other code sets should follow the same rules for single octet and variable length information formats as contained in this document.

Code set identifier:

Bit:	3 2 1	Significance
	0 0 0	Default code set.
	0 0 1 . . . 1 1 1	Code sets 1 to 7 are reserved for future use: allocation of these codes is controlled by and registered with the Standard Control Authority body. These code sets are reserved for future non-mandatory applications.

NOTE: The information element 7 is reserved for use as detailed in subclause 6.5.5. Other information elements are reserved for future allocation. Allocation is controlled by and registered with the SCA.

7.2 Variable length information elements

The variable length information element is encoded as shown in the diagram below:

```
Bit:  8      7      6      5      4      3      2      1   octet
    ┌──────┬─────────────────────────────────────────────┐
    │  0   │       information element identifier        │   1
    ├──────┴─────────────────────────────────────────────┤
    │         length of information element (octets)     │   2
    ├────────────────────────────────────────────────────┤
    │           contents of information element          │   3
    ┼                                                    ┼
    │           etc.                                     │   4
    ┼                                                    ┼
                                                            etc.
```

NOTE: Any additional information fields added within the content part of variable length information elements are controlled and registered with the Standard Control Authority. All information elements, whose length is explicitly defined in this standard, may have qualifying information appended to them, subject to the approval by the Standard Control Authority, but it is mandatory to act on the defined information element content.

```
Bit:  8      7      6      5      4      3      2      1   octet
    ┌──────┬─────────────────────────────────────────────┐
    │  0   │      information element identifier         │   1
    ├──────┴─────────────────────────────────────────────┤
    │        length of information element (m + n)       │   2
    ├────────────────────────────────────────────────────┤
    │      contents of information element (m octets)    │   3
    ├────────────────────────────────────────────────────┤
    │                         .                          │
                              .                             etc.
    │                                                    │
    ├────────────────────────────────────────────────────┤
    │         (qualifying information) (n octets)        │m+n+2
    └────────────────────────────────────────────────────┘
```

Layer three data fields are subject to the following interpretation:

The least significant bit in any field is the lower bit number and the least significant octet in any multi octet field is the lower octet number unless explicitly stated otherwise.

```
Bit:  8      7      6      5      4      3      2      1   octet
    ┌────────────────────────────────────────────────────┐
    │                                               lsb  │  LS
    ┼                                                    ┼
    │                         .                          │
                              .
    │                                                    │
    ┼                                                    ┼
    │  msb                                               │  MS
    └────────────────────────────────────────────────────┘
```

A list of the information element fields carried in layer three messages required to be transmitted over the D signalling channel is given below:

Table 11: Variable-length Information Elements

Mnemonic	Bit: 87654321	Dir'n CPP CFP	Function
KP	00000000	------>	Keypad
DISP	00000010	<------	Display
SIG	00000011	<------	Signal
FA	00000001	------>	Feature activation
FI	00000100	<------	Feature indication
CC	00000101	<------	Channel control
INIT	00000110	<------	Initialise
AUTH_REQ	00000111	<------	Authentication request
AUTH_RES	00001000	------>	Authentication response
TERM_CAP	00001001	------>	Terminal (CPP) capabilities
BAS_CAP	00001010	<------	Base (CFP) capabilities
CHAR	00001011	<----->	IA5 character
OARAC	00001100	<------	On air (de-)registration acknowledge
PAR_REQ	00001101	<----->	Parameter request
PAR_RES	00001110	<----->	Parameter response
PAR_SET	00001111	<----->	Parameter set
AUTH2_REQ	00010000	<------	Alternative AUTH_REQ
AUTH2_RES	00010001	------>	Alternative AUTH_RES
NO_POLL	00010010	<------	Number of handsets being polled

Experimental information elements are detailed in a separate document (obtainable from the SCA).

7.2.1 Keypad information element (KP)

The purpose of the keypad information element is primarily to convey IA5 characters entered at the CPP keypad for dialling purposes and is coded as shown below:

```
Bit:  8      7      6      5      4      3      2      1   octet
     ┌──────┬─────────────────────────────────────────────┐
     │  0   │  0      0      0      0      0      0      0 │   1
     │      │       keypad information element identifier │
    ─┼──────┴─────────────────────────────────────────────┼─
     │             length of keypad information element   │   2
    ─┼────────────────────────────────────────────────────┼─
     │             keypad information (IA5 characters)    │   3
    ─┼────────────────────────────────────────────────────┼─
                                                            etc.
```

The keypad information (octet 3 etc.) is capable of carrying the following IA5 characters. The coding allows for digit by digit sending (repeated keypad information elements with length field value 1) or en bloc sending (e.g. repertory numbers with one KP element and length as appropriate). IA5 characters occupy seven bits; the eighth bit shall be set to 0. The eighth bit when set to 1 denotes manufacturer specific information. Keypad information may also be used by manufacturers for on-air registration of CPPs.

KP elements carry dialling character information with characters presented in the order in which they were keyed by an operator. For example, the telephone number, 01 234 5678 is transported with the 0 in the least significant (lower numbered) octet of the KP element and 8 in the most. The terms most significant and least significant octets (as defined in subclause 7.2) therefore do not apply to KP elements.

Allocation of keypad information codes:

Codes 0 to 31 (decimal)

These codes are used for CAI specific control purposes as follows:

Bit: 8 7 6 5 4 3 2 1	Value	Meaning
0 0 0 0 0 0 0 0	NUL	Ignore.
0 0 0 0 0 0 1 0	STX	Return home.
0 0 0 0 0 0 1 1	ETX	Return end.
0 0 0 0 0 1 0 1	ENQ	Pause. The duration of pause is determined by the CFP.
0 0 0 0 1 0 0 0	BS	Move back one column.
0 0 0 0 1 0 0 1	HT	Move forward one column.
0 0 0 0 1 0 1 0	LF	Move down one row.
0 0 0 0 1 0 1 1	VT	Move up one row.
0 0 0 0 1 1 0 0	FF	Clear display (and return home).
0 0 0 0 1 1 0 1	CR	Return to beginning of the current row.
0 0 0 0 1 1 1 0	SO	Flash off.
0 0 0 0 1 1 1 1	SI	Flash on (all following displayed characters until flash off).
0 0 0 1 0 0 0 1	DC1	Resume transmission (XON).
0 0 0 1 0 0 1 0	DC2	Go to decadic.
0 0 0 1 0 0 1 1	DC3	Stop transmission (XOFF).
0 0 0 1 0 1 0 0	DC4	Go to MF.
0 0 0 1 0 1 0 1	NAK	Fast flash on (all following characters until flash off).
0 0 0 1 0 1 1 0	SYN	Toggle scroll lock.
0 0 0 1 1 0 0 1	EM	Clear to end of display (maintain cursor position).
0 0 0 1 1 0 1 0	SUB	Clear to end of line (maintain cursor position).
0 0 0 1 1 0 1 1	ESC	Escape as in IA5. The usage of ESC to be as in ISO 2022 [20].

All other codes in the range 0 to 31 inclusive are reserved for future allocation. Allocation is controlled by and registered with the Standard Control Authority body.

Codes 32 to 127 (decimal)

These codes are used in the standard IA5 sense. Values '0' through '9' are mandatory for decadic dialling with '*' and '#' additionally mandatory for MF dialling. The characters 'a' through 'd' are optional additional dialling information in MF mode.

Bit: 8 7 6 5 4 3 2 1	Value
0 0 1 1 0 0 0 0	'0'
0 0 1 1 0 0 0 1	'1'
0 0 1 1 0 0 1 0	'2'
0 0 1 1 0 0 1 1	'3'
0 0 1 1 0 1 0 0	'4'
0 0 1 1 0 1 0 1	'5'
0 0 1 1 0 1 1 0	'6'
0 0 1 1 0 1 1 1	'7'
0 0 1 1 1 0 0 0	'8'
0 0 1 1 1 0 0 1	'9'
0 0 1 0 1 0 1 0	'*'
0 0 1 0 0 0 1 1	'#'
0 1 1 0 0 0 0 1	'a'
0 1 1 0 0 0 1 0	'b'
0 1 1 0 0 0 1 1	'c'
0 1 1 0 0 1 0 0	'd'

NOTE: '*','#','a','b','c' and 'd' may be ignored by a CFP working in decadic mode.

Codes 128 to 255 (decimal)

These codes are reserved for manufacturer specific applications. The codes are F0 to F127. F0 to F11 are recommended for general use.

Bit: 8 7 6 5 4 3 2 1	Value
1 0 0 0 0 0 0 0	F0
. .	
. .	
1 0 0 0 1 0 0 1	F9
1 0 0 0 1 0 1 0	F10
1 0 0 0 1 0 1 1	F11
. .	
. .	
1 1 1 1 1 1 1 1	F127

7.2.2 Display information element (DISP)

The purpose of the display information element is to convey IA5 characters to the CPP display and is coded as shown below:

Bit: 8	7	6	5	4	3	2	1	octet
0	0	0	0	0	0	1	0	1
	display information element identifier							
length of display information element								2
display information (IA5 characters)								3
								etc.

The display information (octet 3 etc.) is capable of carrying any IA5 character. CPPs not equipped with a display can ignore normal display parameters.

Information shall be displayed in the order in which it is received e.g. a message containing display information "1234" followed by a message "5678" shall appear on a suitably large display as "12345678"

Allocation of display information codes:

Codes 0 to 31 (decimal)

These codes are used for CAI specific control purposes as follows:

Bit: 8 7 6 5 4 3 2 1	Value	Meaning
0 0 0 0 0 0 0 0	NUL	Ignore.
0 0 0 0 0 0 1 0	STX	Return home.
0 0 0 0 0 0 1 1	ETX	Return end.
0 0 0 0 0 1 0 1	ENQ	Pause. The duration of pause is determined by the CFP.
0 0 0 0 1 0 0 0	BS	Move back one column.
0 0 0 0 1 0 0 1	HT	Move forward one column.
0 0 0 0 1 0 1 0	LF	Move down one row.
0 0 0 0 1 0 1 1	VT	Move up one row.
0 0 0 0 1 1 0 0	FF	Clear display (and return home).
0 0 0 0 1 1 0 1	CR	Return to beginning of current row.
0 0 0 0 1 1 1 0	SO	Flash off.
0 0 0 0 1 1 1 1	SI	Flash on (all following displayed characters until flash off).
0 0 0 1 0 0 0 1	DC1	Resume transmission (XON).
0 0 0 1 0 0 1 0	DC2	Go to decadic.
0 0 0 1 0 0 1 1	DC3	Stop transmission (XOFF).
0 0 0 1 0 1 0 0	DC4	Go to MF.
0 0 0 1 0 1 0 1	NAK	Fast flash on (all following characters until flash off).
0 0 0 1 0 1 1 0	SYN	Toggle scroll lock.
0 0 0 1 1 0 0 1	EM	Clear to end of display (maintain cursor position).
0 0 0 1 1 0 1 0	SUB	Clear to end of line (maintain cursor position).
0 0 0 1 1 0 1 1	ESC	Escape as in IA5. The usage of ESC to be as in ISO 2022 [20].

The code FF clear display is required as a minimum for all displays.

All other codes in the range 0 to 31 inclusive are reserved for future allocation. Allocation is controlled by and registered with the Standard Control Authority body.

Codes 32 to 127 (decimal)

These codes are used in the standard IA5 sense. Values '0' through '9','*' and '#' are the minimum recommended for decadic and MF dialling. The characters 'a' through 'd' are optional additional dialling information in MF mode.

Bit: 8 7 6 5 4 3 2 1	Value
0 0 1 1 0 0 0 0	'0'
0 0 1 1 0 0 0 1	'1'
0 0 1 1 0 0 1 0	'2'
0 0 1 1 0 0 1 1	'3'
0 0 1 1 0 1 0 0	'4'
0 0 1 1 0 1 0 1	'5'
0 0 1 1 0 1 1 0	'6'
0 0 1 1 0 1 1 1	'7'
0 0 1 1 1 0 0 0	'8'
0 0 1 1 1 0 0 1	'9'
0 0 1 0 1 0 1 0	'*'
0 0 1 0 0 0 1 1	'#'
0 1 1 0 0 0 0 1	'a'
0 1 1 0 0 0 1 0	'b'
0 1 1 0 0 0 1 1	'c'
0 1 1 0 0 1 0 0	'd'

Codes 128 to 255 (decimal)

These codes are reserved for manufacturer specific applications.

7.2.3 Signal information element (SIG)

The purpose of the signal information element is to convey indications to prompt the CPP to generate alerting (tone caller), warning and error audible signals as coded below:

Bit: 8	7	6	5	4	3	2	1	octet
0	0 signal information element identifier	0	0	0	0	1	1	1
0 length of signal information element	0	0	0	0	0	0	1	2
signal class				value				3

The signal content field (octet 3) is encoded as follows:

Signal class 0 (stop audible signals):

	Value 4 3 2 1	Significance
Bit:		
	x x x x	Stop all audible signals.

Signal class 1 (alerting signals):

Signal class 1 may cause one of several (according to value) ringing signals to be emitted at the CPP. The alerting class assumes a monostable action of 350 ms to 450 ms. The monostable action is re-triggered on receipt of further identical signals.

	Value 4 3 2 1	Significance
Bit:		
	0 0 0 0	General alerting signal.
	0 0 0 1	Line alerting signal.
	0 0 1 0	Intercom alerting signal.
	0 0 1 1	Enumerated alerting signal 3.
	. .	
	. .	
	. .	
	1 1 1 1	Enumerated alerting signal 15.

Signal class 2 (warning signals):

Signal class 2 may cause one of several (according to value) warning signals to be emitted at the CPP. The warning class signal event causes an audible warning signal, the duration of which is determined by the CPP.

	Value 4 3 2 1	Significance
Bit:		
	0 0 0 0	General warning signal.
	0 0 0 1	Enumerated warning signal 1.
	. .	
	. .	
	. .	
	1 1 1 1	Enumerated warning signal 15.

Signal class 3 (error signals):

Signal class 3 may cause one of several (according to value) error signals to be emitted at the CPP. The error class signal event causes an audible error signal, the duration of which is determined by the CPP.

Bit:	Value 4 3 2 1	Significance
	0 0 0 0	General error signal.
	0 0 0 1	Enumerated error signal 1.
	. .	
	. .	
	. .	
	1 1 1 1	Enumerated error signal 15.

NOTE: All other classes are reserved for future allocation. Allocation is controlled by and registered with the Standard Control Authority body.

7.2.4 Feature activation information element (FA)

The purpose of the feature activation information element is to convey information on actions at the man-machine interface at the CPP and is coded as shown below:

Bit: 8	7	6	5	4	3	2	1	octet
0	0	0	0	0	0	0	1	1
	FA information element identifier							
0	0	0	0	0	0	0	1	
length of FA information element								2
feature class			value					3

The feature activator content field (octet 3) is encoded as follows:

Feature class 0 (line selection):

Feature class 0 allows the potential selection of one of 31 directly addressed "outside" lines on private CFPs. The value of 0 is used for general line select.

Bit:	Value 5 4 3 2 1	Significance
	0 0 0 0 0	General line select.
	0 0 0 0 1	Line select 1.
	. .	
	. .	
	. .	
	1 1 1 1 1	Line select 31.

Feature class 1 (system selection):

Feature class 1 allows the potential selection of one of 31 directly addressed "systems" within a private CFP environment. The value of 0 is used for general system select.

Bit:	Value 5 4 3 2 1	Significance
	0 0 0 0 0	General system select.
	0 0 0 0 1	System select 1.
	. .	
	. .	
	. .	
	1 1 1 1 1	System select 31.

Feature class 2 (local intercom selection):

Feature class 2 allows the potential selection of one of 31 directly addressed CPPs within the intercom network of a CTA. The value of 0 is used for general intercom select and following KP digits may indicate the ("dialled") intercom address.

	Value	
Bit:	5 4 3 2 1	Significance
	0 0 0 0 0	general local intercom select.
	0 0 0 0 1	named local intercom select 1.
	. .	
	. .	
	. .	
	1 1 1 1 1	named local intercom select 31.

Feature class 3 ("public" access selection - "telepoint"):

Feature class 3 allows the potential selection of one of 31 public services. The value of 0 is used for general service select.

	Value	
Bit:	5 4 3 2 1	Significance
	0 0 0 0 0	General service select.
	0 0 0 0 1	service 1.
	. .	
	. .	
	. .	
	1 1 1 0 1	service 29.
	1 1 1 1 0	CPP local de-registration request.
	1 1 1 1 1	CPP local registration request.

CPP local registration and de-registration are to be used to locally register or de-register a CPP with a telepoint CFP; once registered, a CPP could receive incoming calls from the telepoint network via the CFP, and de-registering prevents the network from directing further calls for the CPP to the specific telepoint CFP.

The following diagram (figure 14) illustrates the usage of the above four feature classes within the CT2 environment:

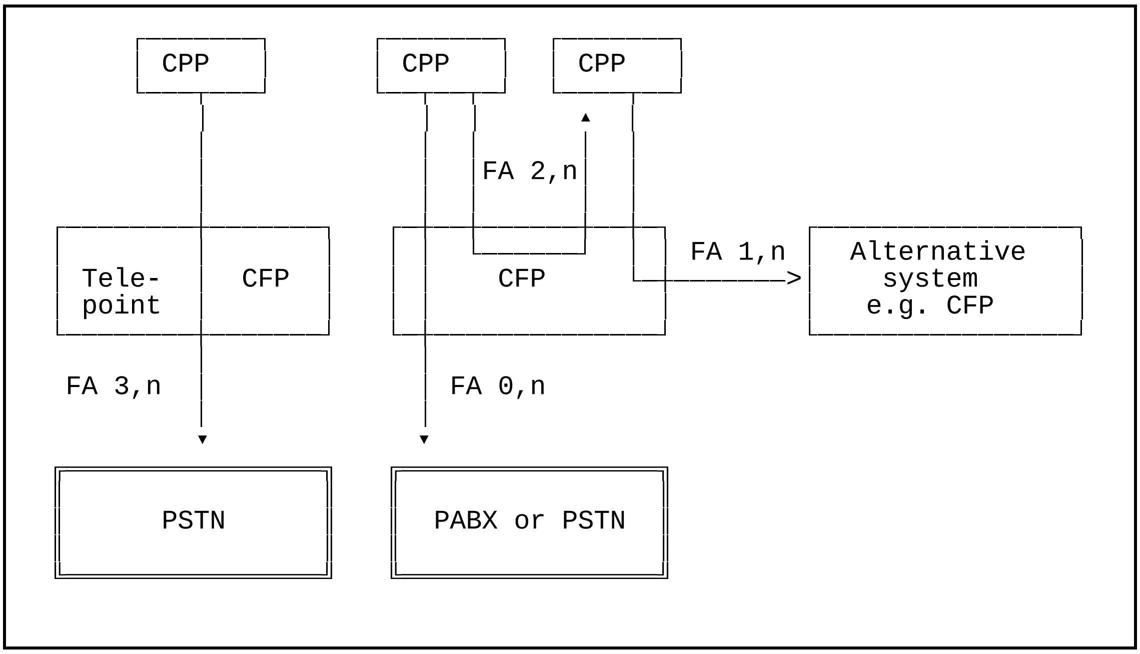


Figure 14: Feature class usage diagram

Feature class 4 (emergency access selection):

Feature class 4 allows the potential selection of one of 32 emergency services.

Bit:	Value 5 4 3 2 1	Significance
	0 0 0 0 0	default emergency service.
	0 0 0 0 1	emergency service 1.
	.	.
	.	.
	.	.
	1 1 1 1 1	emergency service 31.

NOTE: Emergency access may be via dedicated lines or via uncommitted PSTN lines. In the latter case it will be necessary for the CFP to dial the appropriate emergency service number.

Feature classes 5 and 6 (reserved for future use.)

Allocation of these classes is controlled by and registered with the Standard Control Authority body. These classes are reserved for future applications.

Feature class 7 (auxiliary function selection):

Auxiliary functions for the purposes of call control are activated via feature class 7. "link communication" is defined here as "a maintained link at layer two", and a "session" is defined as meaning "the requesting, granting and use of a service."

Bit:	Value 5 4 3 2 1	Significance
	0 0 0 0 0	Clear.
	0 0 0 0 1	Full release.
	0 0 0 1 0	Register recall.
	0 0 0 1 1	Partial release.
	0 0 1 0 0	Hold.
	0 0 1 0 1	Alternative network function (e.g. Mercury in the UK).
	0 0 1 1 0	Supervisory mode.
	0 0 1 1 1	On air local registration (non-telepoint).
	0 1 0 0 0	On air local de-registration (non-telepoint).
	0 1 0 0 1	Call transfer.
	1 1 1 1 1	Null feature.

0) Clear causes termination of all sessions and link communications. This message shall be acknowledged at layer two.

1) Full release is a request to terminate the current session, and if no other sessions, then CFP originated clear down shall ensue. This message shall be acknowledged at layer two.

2) Register recall is totally equivalent to a wired telephone register recall switch.

3) Partial release is a request to terminate the current session. This message shall be acknowledged at layer two. After 20 s ±1 s, if no other session is requested by the CPP and granted by the CFP, then the CFP sends an acknowledged INIT, which terminates the link.

4) Hold causes the current session to be suspended. The RF link shall be retained in anticipation of starting or resuming another session.

5) Alternative network function which shall request a predefined call routing procedure to be actioned by the CFP.

6) Supervisory mode.

7) On air registration causes connection of the CPP to the CFP registration service.

NOTE: This is the local CFP registration service and is not a telepoint registration service.

8) On air de-registration causes connection of the CPP to the CFP de-registration service.

NOTE: This is the local CFP de-registration service and is not a telepoint de-registration service.

9) Call transfer. This feature activation invokes the CFP call transfer facility (if provided).

NOTE: Call transfer may be provided by the CFP without the use of this FA but by the use of other allocated information elements.

15) Null feature. A feature activation that requests a null session (i.e. the link is maintained).

NOTE: All other values are reserved for future allocation. Allocation is controlled by and registered with the Standard Control Authority body.

7.2.5 Feature indication information element (FI)

The purpose of the feature indication information element is to convey information which may activate a "feature indicator" (e.g. an appropriate display character(s) or icon) at the CPP to show the call state and is coded as shown below:

Bit: 8	7	6	5	4	3	2	1	octet
0	0	0	0	0	1	0	0	1
	FI information element identifier							
0	0	0	0	0	0	1	0	
length of FI information element								2
feature class			value					3
state parameter								4

The feature class and value (octet 3) are encoded in the same way as the feature activation information element octet 3.

The significance of the state parameter depends on the associated feature class, as follows:

Feature class 0 (line selection):

Feature class 0 FIs identify the state of "outside" lines on private CFPs, with the significance of the state parameters as follows:

Bit:	State parameter 8 7 6 5 4 3 2 1	Significance
	0 0 0 0 0 0 0 0	Line not seized/line call ceased.
	0 0 0 0 0 0 0 1	Line seized.
	0 0 0 0 0 0 1 0	Incoming line call.
	0 0 0 0 0 0 1 1	Line on hold.
	0 0 0 0 0 1 0 0	Line busy.
	0 0 0 0 0 1 0 1	Line unobtainable.

Feature class 1 (system selection):

Feature class 1 FIs identify the state of interconnections to "systems" within a private CFP environment, with the significance of the state parameters as follows:

Bit:	State parameter 8 7 6 5 4 3 2 1	Significance
	0 0 0 0 0 0 0 0	System not selected/system call ceased.
	0 0 0 0 0 0 0 1	System selected.
	0 0 0 0 0 0 1 0	Incoming call from a system.
	0 0 0 0 0 0 1 1	Call to system on hold.
	0 0 0 0 0 1 0 0	Interconnection to system busy.
	0 0 0 0 0 1 0 1	System unobtainable.

Feature class 2 (local intercom selection):

Feature class 2 FIs identify the state of connection to one of the directly addressed CPPs within the intercom network of a CTA (where "connect" may , but not necessarily, be the RF link to the CPP). The significance of the state parameter is as follows:

Bit:	State parameter 8 7 6 5 4 3 2 1	Significance
	0 0 0 0 0 0 0 0	Connection not made/ceased.
	0 0 0 0 0 0 0 1	Connection made.
	0 0 0 0 0 0 1 0	Incoming intercom call.
	0 0 0 0 0 0 1 1	Intercom connection on hold.
	0 0 0 0 0 1 0 0	CPP busy (engaged).
	0 0 0 0 0 1 0 1	CPP unobtainable (e.g. out of range or no RF link available).

Feature class 3 ("public" access selection ("telepoint"))

Feature class 3 FIs identify the state of "outside" lines on telepoint base stations, with the significance of the state parameters as follows:

Bit:	State parameter 8 7 6 5 4 3 2 1	Significance
	0 0 0 0 0 0 0 0	Line not seized/line call ceased.
	0 0 0 0 0 0 0 1	Line seized.
	0 0 0 0 0 0 1 0	Incoming line call.
	0 0 0 0 0 0 1 1	Line on hold.
	0 0 0 0 0 1 0 0	Line busy (line already in use).
	0 0 0 0 0 1 0 1	Line unobtainable.

The significance of the FI state parameter when associated with the CPP local de-registration and registration request (FA) is as follows:

Bit:	State parameter 8 7 6 5 4 3 2 1	Significance
	0 0 0 0 0 0 0 0	Service ceased.
	0 0 0 0 0 0 0 1	Service accepted.
	0 0 0 0 0 0 1 0	Not applicable.
	0 0 0 0 0 0 1 1	Not applicable.
	0 0 0 0 0 1 0 0	Not applicable.
	0 0 0 0 0 1 0 1	Service unobtainable.

Feature class 4 (emergency access selection)

Feature class 4 FIs identify the state of lines reserved for emergency access or temporarily in use as emergency access lines. The significance of the state parameters is as follows:

Bit:	State parameter 8 7 6 5 4 3 2 1	Significance
	0 0 0 0 0 0 0 0	Line not seized/line call ceased.
	0 0 0 0 0 0 0 1	Line seized.
	0 0 0 0 0 0 1 0	Incoming call on dedicated emergency line.
	0 0 0 0 0 0 1 1	Line on hold.
	0 0 0 0 0 1 0 0	Line busy.
	0 0 0 0 0 1 0 1	Line unobtainable.

Feature class 7 (auxiliary function selection)

Feature indications may not meaningfully be attached to all auxiliary functions. It is recommended that FIs for the following functions are not used (but if used the state parameter should be set to idle (00000000)): clear, full release, register recall, partial release, hold, call transfer, null feature.

1) Alternative network function

Bit:	State parameter 8 7 6 5 4 3 2 1	Significance
	0 0 0 0 0 0 0 0	Line not seized/line call ceased.
	0 0 0 0 0 0 0 1	Line seized.
	0 0 0 0 0 0 1 0	Incoming line call.
	0 0 0 0 0 0 1 1	Line on hold.
	0 0 0 0 0 1 0 0	Line busy.
	0 0 0 0 0 1 0 1	Line unobtainable.

2) Supervisory mode

Bit:	State parameter 8 7 6 5 4 3 2 1	Significance
	0 0 0 0 0 0 0 0	Supervisory service request not set up or ceased.
	0 0 0 0 0 0 0 1	Supervisory service request accepted.
	0 0 0 0 0 0 1 0	Not used.
	0 0 0 0 0 0 1 1	Not used.
	0 0 0 0 0 1 0 0	Supervisory service busy.
	0 0 0 0 0 1 0 1	Supervisory service unobtainable.

3) On air registration

Bit: State parameter 8 7 6 5 4 3 2 1	Significance
0 0 0 0 0 0 0 0	Registration service request not set up or ceased.
0 0 0 0 0 0 0 1	Registration service request accepted.
0 0 0 0 0 0 1 0	Not used.
0 0 0 0 0 0 1 1	Not used.
0 0 0 0 0 1 0 0	Registration service busy.
0 0 0 0 0 1 0 1	Registration service unobtainable.

4) On air de-registration

Bit: State parameter 8 7 6 5 4 3 2 1	Significance
0 0 0 0 0 0 0 0	De-registration service request not set up or ceased.
0 0 0 0 0 0 0 1	De-registration service request accepted.
0 0 0 0 0 0 1 0	Not used.
0 0 0 0 0 0 1 1	Not used.
0 0 0 0 0 1 0 0	De-registration service busy.
0 0 0 0 0 1 0 1	De-registration service unobtainable.

NOTE: All other parameters are reserved for future allocation. Allocation is controlled by and registered with the Standard Control Authority body.

7.2.6 Channel control information element (CC)

The purpose of the channel control information element is to convey information from the CFP to cause the CPP to control the B channel connection and multiplex as coded below:

Bit: 8	7	6	5	4	3	2	1	octet
0	0	0	0	0	1	0	1	
	CC information element identifier							1
0	0	0	0	0	0	0	1	
length of CC information element								2
B channel control parameter								3

The B channel control parameter (octet 3) is encoded as follows:

Bit: Parameter value 8 7 6 5 4 3 2 1	Significance
0 0 0 0 0 0 0 0	Use MUX1 with disconnected B channel.
0 0 0 0 0 0 1 0	Use MUX1 and connect B channel (no local sidetone).
0 0 0 0 0 0 1 1	Use MUX1 and connect B channel (with local sidetone).
0 0 0 0 0 1 0 0	Use MUX2 and disconnect B channel.

NOTE: All other parameters are reserved for future allocation. Allocation is controlled by and registered with the Standard Control Authority body.

It shall the responsibility of the CFP to ensure that the link returns to the correct multiplex mode under circumstances such as link re-establishment during multiplex mode changes.

7.2.7 Initialisation information element (INIT)

The initialise information element is used by the CFP to instruct the CPP to return to standby state whereby the B channel is disconnected, and all displays, indicators and tone generators are returned to an idle condition. This message shall be acknowledged at layer two.

Bit: 8	7	6	5	4	3	2	1	octet
0	0 INIT information element identifier	0	0	0	1	1	0	1
0 length of INIT information element	0	0	0	0	0	0	1	2
0 INIT parameter	0	0	0	0	0	0	0	3

NOTE: All other parameters are reserved for future allocation. Allocation is controlled by and registered with the Standard Control Authority body.

This message may be sent at any time by the CFP to terminate a link. It is not necessary for the CPP to first send a CLEAR or equivalent feature activation from class 7.

The following diagrams explain the use of the CLEAR, FULL RELEASE and INIT messages used to perform the clear down procedure:

```
                          CPP                        CFP
Start Tclr(1 s) timeout
and send CLEAR.            |                          |
           |               | CLEAR                    |
           └─────────>     | ─────────────────────>   | CLEAR received. Shut down
                           |                ack       | CPP with INIT.  Start
IF Tclr(1 s) timer         | < - - - - - - - - - - -  | Tclr(1 s) timer awaiting
expires or INIT            |                          | acknowledgement to a
received:                  |                          | clear down request
       |                   |                   INIT   |           |
       ├─────<──────────   |<──────────────────────   | <─────────┘
       |          |        |  ack                     |
shut down layers  └───>    | - - - - - - - - - - - >  | ───> Shut down layers
one, two and three.        |                          | one, two and three if
                           |                          | ack or timeout.
                           |                          |
```

Figure 15: Clear Down procedure with CLEAR

```
                           CPP                     CFP
Start Tclr(1 s) timeout
and send FULL RELEASE.|                        |
                      | FULL RELEASE           | FULL RELEASE received,
                      | ---------------------> | shut down CPP with INIT
                      |               ack      | (only if no sessions).
                      | < - - - - - - - - - -  |              |
IF Tclr(1 s) timer    :                        :              |
expires:              :  EITHER 1:             :              |
      |               :                        :              |
      |               |                   INIT | Start Tclr (1 s) timer.
      |-----<------   | <--------------------- |<-------------
      |          |    |   ack                  |
      v          L--> | - - - - - - - - - - >  | ---> IF ack received OR
Shut down layers      |                        | Tclr(1 s) timer expires
one, two and three.   |                        | shut down layers one,
                      :                        : two and three.
                      :  OR 2:                 :
                      :                        :
                      |                        | FULL RELEASE received.
                      |  ANY layer three,      | IF session(s) exist
Stop Tclr(1 s) <----- | <--------------------  | DO NOT shut down.
timer.                |  (but NOT INIT)        |
                      |                        |
```

Figure 16: Clear Down procedure with FULL RELEASE

7.2.8 Authentication request information element (AUTH_REQ)

The authentication request information element is issued by a telepoint CFP to initiate the call authentication process. The authentication request information element has two parameters, RAND and INCZ, and causes the CPP to respond with the authentication response information element (AUTH_RES).

Bit: 8	7	6	5	4	3	2	1	octet
0	0 AUTH_REQ information element identifier	0	0	0	1	1	1	1
0	0 length of AUTH_REQ information element	0	0	0	1	1	0	2
AUTH_NO								3
RAND								4
								5
								6
								7
0	0	0	0	0	0	0	INCZ	8

AUTH_NO is used to indicate to the CPP which of the authentication algorithms offered by the CPP is to be used, even if the CPP is only capable of performing one authentication process.

RAND is a 32-bit random number generated by the CFP to be used by the CPP in the call authentication process.

INCZ, when set to 1, causes the CPP to increment (modulo 16) the contents of a 4-bit ZAP field within the appropriate registration slot. When INCZ is set to 0 the field is left unchanged. The ZAP field is transferred to a telepoint CFP in the AUTH_RES information element and may be used by the telepoint operator as a part of the overall CPP authentication process.

7.2.9 Authentication response information element (AUTH_RES)

The authentication response information element is sent by the CPP in response to an authentication request information element and conveys telepoint registration and authentication parameters to the telepoint CFP.

Bit: 8	7	6	5	4	3	2	1	octet
0	0 AUTH_RES information element identifier	0	0	1	0	0	0	1
length of AUTH_RES information element								2
CKEY								3
								4
								5
								6
OPSIC							lsb	7
ZAP				TCOS			msb	8
TRD								9
								10
								11
								12
								13
								14
								15
								16
								17
								18

CKEY is the 32-bit result of the call authentication process, calculated by the CPP and returned to the telepoint CFP for checking.

OPSIC is the operators identification code. Specific code allocations for this field are detailed in a separate document (obtainable from the SCA).

TCOS (telepoint class of service) field is used to transport telepoint class of service details from the CPP. The use and interpretation of the data in this field is proprietary to the telepoint operators.

TRD (telepoint registration data) field is used to transport the CPP details of the telepoint account being used. An unused BCD location in the TRD field shall contain binary 1111. The least significant account digit is placed in the right hand half of the octet immediately following the octet containing ZAP, with the next most significant digit in the left hand half, and so on. Where an unused location is filled with binary 1111, it shall always be in the left hand half of the most significant octet. Specific code allocations for this field are defined by and are the responsibility of the telepoint operators.

ZAP is the contents of the 4-bit ZAP field stored in the appropriate registration slot of a CPP as described in subclause 7.2.8.

7.2.10 Terminal capabilities information element (TERM_CAP)

The terminal capabilities information element is issued by the CPP following link establishment and before connection of the B channel.

TERM_CAP is a statement of the capabilities of the CPP. The CFP shall operate within those capabilities or shall terminate the link.

Bit: 8	7	6	5	4	3	2	1	octet
0	0	0	0	1	0	0	1	
	TERM_CAP information element identifier							1
0	0	0	0	0	1	1	1	
length of TERM_CAP information element								2
HSSC	DCAP			MB	CIC			3
MANIC								4
MODEL								5
AUTH_PREF								6
							lsb	7
AUTH_KEY								8
msb								9

CIC is a 3-bit field signifying the codec type used by the CPP. The CAI ADPCM voice codec (see subclause 8.1.4) has a CIC value of 000. Other values shall only be allocated by the SCA.

MB = 0 signifies a CPP with a message buffer size limited to 29 octets at layer three (6 code words max.) and MB = 1 for CPP with full 128 octet capability.

DCAP identifies the CPPs display capabilities as below:

Bit:	3 2 1	Significance
	0 0 0	No display.
	0 0 1	Numeric only display.
	0 1 0	7 segment display.
	0 1 1	Full alphanumeric display.

HSSC indicates high speed signalling capability. HSSC set to 1 denotes the ability to revert to MUX2 signalling from MUX1 without the need for link re-establishment:

Bit:	1	Significance
	0	No high speed signalling capability.
	1	High speed signalling capability available.

MANIC identifies the CPP manufacturer, the value of zero to indicate anonymous. The assignment of MANIC field values is controlled by and registered with the Standard Control Authority body. Specific code allocations are detailed in a separate document (obtainable from the SCA).

MODEL is used to identify the CPP model number associated with the manufacturer ID in the MANIC field. Values assigned to this field are at manufacturer's discretion except where MANIC value is 0 when MODEL shall also be 0.

AUTH_PREF is used by the CPP to indicate to a CFP which of the authentication algorithms offered by the CPP is the CPP's preferred algorithm. If only one algorithm is offered, this shall be indicated as the preferred algorithm in this field.

AUTH_KEY is a bit field used to indicate to a CFP which authentication algorithms the CPP is capable of performing. If no telepoint authentication algorithms are offered, all bits in this field shall be set to 0. A bit if set to 1 indicates that the CPP is capable of performing the associated algorithm, and if the bit is set to 0, then the CPP is not capable of performing the associated algorithm. The assignment of bits in this field is controlled by and registered with the Standard Control Authority. Specific bit allocations are detailed in a separate document (obtainable from the SCA).

NOTE: The relationship between the AUTH_NO (in AUTH_REQ and AUTH2_REQ), AUTH_PREF and AUTH_KEY fields is as follows (illustrated by way of example, all numbers in hexadecimal):

If the following allocations for telepoint authorisation mechanisms X, Y and Z are assumed (for example only):

AUTH_KEY	Significance
000000	No telepoint ability.
000001	Bit 1 indicates ability X.
000002	Bit 2 indicates ability Y.
000004	Bit 3 indicates ability Z.

then the following example combinations may be interpreted:

AUTH_KEY	Significance
000003	indicates abilities X and Y.
000005	indicates abilities X and Z.
000006	indicates abilities Y and Z.
000007	indicates abilities X, Y and Z.

If a CPP has ability X only, then the AUTH_KEY bit field is coded as 000001; in this case, the AUTH_PREF field in TERM_CAP is coded 01 (bit 1), and AUTH_NO in AUTH_REQ is coded as 01. For a CPP indicating the ability to perform the X and Z algorithms, AUTH_KEY is 000005, and AUTH_PREF will be either of 03 (bit 3, prefers to use Z), or 01 (prefers to use X). AUTH_NO will be either 03 (instructing the CPP to use Z), or 01, (instructing the CPP to use X).

If the CPP is capable of performing an algorithm which is requested using AUTH2_REQ and AUTH2_RES, the value placed in the AUTH_NO field of AUTH2_REQ shall indicate which algorithm to use; if for example algorithm Z is requested by AUTH2_REQ, then the AUTH_NO value shall be 03 in the example above.

7.2.11 Base capabilities information element (BAS_CAP)

The base capabilities information element is issued by the CFP following link establishment and before connection of the B channel. BAS_CAP is a statement of the capabilities of the CFP. The CPP shall operate within those capabilities or shall terminate the link. TERM_CAP and BAS_CAP operate independently and do not form a capabilities negotiating mechanism.

Bit: 8	7	6	5	4	3	2	1	octet
0	0	0	0	1	0	1	0	
	BAS_CAP information element identifier							1
0	0	0	0	0	0	1	1	
length of BAS_CAP information element								2
HSSC	DCAP			MB	ICOM			3
MANIC								4
BASET								5

ICOM identifies the intercom capability of the CFP:

Bit: 3 2 1	Significance
0 0 0	No intercom.
0 0 1	Tone only page (both directions).
0 1 0	CFP and CPP speech intercom.
0 1 1	CPP and CPP speech intercom.
1 0 0	CFP and CPP, plus CPP and CPP speech intercom.

MB = 0 signifies a CFP with a message buffer size limited to 29 octets at layer three (6 code words max.) and MB = 1 for CFP with full 128 octet capability.

DCAP identifies the CFPs display capabilities as below:

Bit:	3 2 1	Significance
	0 0 0	No display.
	0 0 1	Numeric only display.
	0 1 0	7 segment display.
	0 1 1	Full alphanumeric display.

HSSC indicates high speed signalling capability. HSSC set to 1 denotes the ability to revert to MUX2 signalling from MUX1 without the need for link re-establishment:

Bit:	1	Significance
	0	No high speed signalling capability.
	1	High speed signalling capability available.

MANIC identifies the CFP manufacturer, the value of zero to indicate anonymous. The assignment of MANIC field values is controlled by and registered with the Standard Control Authority body. Specific code allocations are detailed in a separate document (obtainable from the SCA).

BASET indicates the CFP type, as follows:

Bit:	8 7 6 5 4 3 2 1	Significance
	0 0 0 0 x 0 0 0	Domestic CFP.
	0 0 0 0 x 0 0 1	Telepoint CFP.
	0 0 0 0 x 0 1 0	Plan/key system CFP.
	0 0 0 0 x 0 1 1	PBX.

The x (bit 4) indicates ISDN type:

Bit:	4	Significance
	0	non-ISDN.
	1	ISDN.

7.2.12 Character information element (CHAR)

The purpose of the character information element is to convey IA5 characters between the CFP and CPP in both directions. The interpretation of the characters is manufacturer specific.

```
Bit:  8      7      6      5      4      3      2      1    octet
    ┌──────┬─────────────────────────────────────────────────┐
    │  0   │  0      0      0      1      0      1      1   │  1
    │      │    character information element identifier    │
    ├──────┴─────────────────────────────────────────────────┤
    │            length of character information element     │  2
    ├────────────────────────────────────────────────────────┤
    │            IA5 character(s)                            │  3
    ├────────────────────────────────────────────────────────┤
                                                               etc.
```

The IA5 characters carried by CHAR are ordered such that a character sequence within CHAR may equivalently be transmitted as a number of successive CHAR messages carrying the same constituent parts, i.e. CHAR('A','B','C') would be equivalent to CHAR('A'), CHAR('B'), CHAR('C').

7.2.13 On-air (de-)registration acknowledge information element (OARAC)

OARAC is transmitted by the CFP on successful completion of the on-air registration/de-registration process (initiated by FA 7,7/8) and conveys the BID field (see subclause 6.4.5). The BID field is set to FFFFH for successful de-registration (or null registration).

Bit: 8	7	6	5	4	3	2	1	octet
0	0	0	0	1	1	0	0	1
	registration information element identifier							
0	0	0	0	0	0	1	0	
length of registration information element								2
registration base identifier (BID)								3
								4

7.2.14 Parameter set information element (PAR_SET)

PAR_SET is a bi-directional information element used by either the CFP of CPP to change the identified parameter type to the given parameter value at the far end of the link. The parameter type and value are defined in the content field of this information element. Only one parameter can be sent per PAR_SET information element.

Bit: 8	7	6	5	4	3	2	1	octet
0	0	0	0	1	1	1	1	1
	PAR_SET information element identifier							
length of PAR_SET information element								2
parameter type								3
parameter value								4
								etc.

Parameter type:

Type = 0; class of service:

Bit: 8	7	6	5	4	3	2	1	octet
0	0	0	e	d	c	b	a	
parameter value								4

Value	Significance
a = 1	intercom permitted.
b = 1	local calls permitted.
c = 1	national calls permitted.
d = 1	international calls permitted.
e = 1	one (predefined) hotline number is accessed automatically by the CFP.

Type = 1; extension number (BCD):

Bit: 8	7	6	5	4	3	2	1	octet
MSBCD digit				LSBCD digit				
parameter value								4

Specific code allocations are detailed in a separate document (obtainable from the SCA).

NOTE: All other types (2 to 255) are reserved for future allocation. Allocation is controlled by and registered with the Standard Control Authority body.

7.2.15 Parameter request information element (PAR_REQ)

PAR_REQ is a bi-directional information element used by either the CFP of CPP to request additional parameters from the other end of the link. The requested parameter type is defined in the content field of this information element. Only one parameter can be requested per PAR_REQ information element.

Bit: 8	7	6	5	4	3	2	1	octet
0	0	0	0	1	1	0	1	1
	PAR_REQ information element identifier							
0	0	0	0	0	0	0	1	
length of PAR_REQ information element								2
requested parameter type								3

Requested parameter type:

Types as defined in information element PAR_SET.

Specific code allocations are detailed in a separate document (obtainable from the SCA).

NOTE: All other types (2 to 255) are reserved for future allocation. Allocation is controlled by and registered with the Standard Control Authority body.

7.2.16 Parameter response information element (PAR_RES)

PAR_RES is bi-directional information element which forms a response to PAR_REQ and whose content field contains the requested parameter. The content field size is parameter dependent.

Bit: 8	7	6	5	4	3	2	1	octet
0	0	0	0	1	1	1	0	1
	PAR_RES information element identifier							
length of PAR_RES information element								2
response parameter								3
								etc.

The requested parameter type is identical to the associated parameter type in PAR_REQ. If the requested parameter is not recognised, a PAR_RES information element of length one shall be returned.

Specific code allocations are detailed in a separate document (obtainable from the SCA).

7.2.17 Alternative authentication request information element (AUTH2_REQ)

This alternative authentication request information element is issued by a telepoint CFP to initiate the alternative call authentication process.

Bit: 8	7	6	5	4	3	2	1	octet
0	0	0	1	0	0	0	0	1
	AUTH2_REQ information element identifier							
length of AUTH2_REQ information element								2
AUTH_NO								3
								etc.

AUTH_NO is used to indicate to the CPP which of the authentication algorithms offered by the CPP is to be used, even if the CPP is only capable of performing one authentication process.

7.2.18 Alternative authentication response information element (AUTH2_RES)

This alternative authentication response information element is sent by the CPP in response to an alternative authentication request information element and conveys alternative telepoint registration and authentication parameters to the telepoint CFP.

Bit: 8	7	6	5	4	3	2	1	octet
0	0 AUTH2_RES information element identifier	0	1	0	0	0	1	1
length of AUTH2_RES information element								2
OPSIC							lsb	3
							msb	4
								etc.

OPSIC is the operator's identification code. Specific code allocations for this field are detailed in a separate document (obtainable from the SCA).

7.2.19 Number of CPPs polled information element (NO_POLL)

This information element may be used during link establishment from a CFP to CPPs to indicate the number of CPPs the CFP is currently polling. During link establishment, this information element can be transmitted unacknowledged by the CFP between fixed format poll code words (subclause 6.6.4). If the CPP is being polled, and if the RES (respond) bit in the information element parameter is set, the CPP is able to establish a link to the CFP before any user action to initiate the link request. (The CPP sends a LINK_REQUEST before the user has accepted the call, and the NULL FA (7,31) after TERM_CAP and BAS_CAP exchange, in order to maintain the correct syntax at layer three. The appropriate FA (e.g. LINE or ICOM) is only sent once the user has pressed the appropriate key on the CPP.

Bit: 8	7	6	5	4	3	2	1	octet
0	0 NO_POLL information element identifier	0	1	0	0	1	0	1
0 length of NO_POLL information element	0	0	0	0	0	0	1	2
RES	number of CPPs being polled							3

RES = 0 prevents the CPP from establishing a link, except by user action, even if the CPP is the only CPP being polled.

RES = 1 allows a CPP to establish a link without user action. The RES bit shall only be set by the CFP when the number of CPPs being polled is one.

7.3 Layer three mandatory syntax

7.3.1 Access to layer three

At a CPP, access to layer three shall require that layer two has been initialised; that, subsequently, TERM_CAP and BAS_CAP information elements have been exchanged; and that the current BAS_CAP is acceptable.

At a CFP, access to a session at layer three shall require that layer two has been initialised; that, subsequently, TERM_CAP and BAS_CAP information elements have been exchanged; that the current TERM_CAP is acceptable and that an acceptable FA has been received from the CPP.

7.3.2 Exit from layer three

At a CPP, exit from layer three shall occur in response to the termination of layer two; to the receipt of INIT from the CFP; or to an attempt to access layer three with an unacceptable BAS_CAP.

At a CFP, exit from all sessions at layer three shall occur in response to the termination of layer two; to the receipt of CLEAR from the CPP; to an attempt to access a session at layer three with an unacceptable TERM_CAP; or to an attempt to access a session at layer three with an unacceptable FA when no previous session exists.

7.3.3 Emergency access

If emergency access is provided in a CFP then the CFP shall accept access to layer two via an emergency LID followed by access to layer three via an emergency FA.

The CFP may refuse to accept an emergency access FA at layer three initialisation if access to layer two was not via an emergency LID.

If the CFP provides emergency access from the "call in progress" state (see Annex A) then it shall accept the sequence PARTIAL RELEASE followed by an emergency FA.

NOTE: Permitting non-emergency follow-on calls after emergency access to layer three may cause a breach of security.

7.3.4 On-air registration

If on-air registration is provided in a CFP then the CFP shall, as a minimum, accept access to layer two via the ID registration LID followed by access to layer three via the on-air registration FA and shall provide confirmation of successful completion of the registration via the on-air registration acknowledge information element.

8 Speech coding and telephony

This Clause specifies the requirements for cordless telephone apparatus capable of transmission of analogue information over the CT2 common air interface B channel.

In order to ensure satisfactory interworking of different handsets and base units it is necessary to specify the performance of the analogue information transmitted via the B channel over the digital link. This requires not only use of a common speech algorithm, but standardisation of frequency responses, notional reference speech levels (or loudness) at the air interface, and various other parameters.

A single transmission plan is proposed where characteristics are lumped into ideal elements.

This Clause proposes a method using reference transceivers for frequency sensitivity measurements suitable for a common air interface meeting the requirements of the CT2 specifications.

There are two speech sub paths which need to be defined: these are send and receive between the handset and the air interface. Send and receive sub-paths, in the CFP, between the telephone line and the air interface may be derived by subtraction from the two specified sub-paths and the speech paths specified in the appropriate published national standard.

8.1 Definitions

8.1.1 Cordless Portable Part (CPP)

A portable part of the cordless telephone apparatus which, by integral radio and aerial means and in conjunction with an associated cordless fixed part, permits some or all of the functions of normal telephone apparatus including manual initiation and termination of calls by a deliberate act.

8.1.2 Fixed geometry CPP

A CPP in which the electro-acoustic transducers and their associated acoustic components are held in fixed relative positions and/or orientations during all on-line conditions of the CPP.

8.1.3 Variable geometry CPP

A CPP that allows the position and/or orientation of its electro-acoustic transducers and their associated components to be changed during all on-line conditions of the CPP.

8.1.4 CAI ADPCM voice codec (CIC = 0)

The ADPCM voice codec algorithm shall be in accordance with CCITT Recommendation G.721 [6]. The codec in a handset and in a fixed part which provides an analogue speech connection may be of reduced specification in that the functions in G.721 associated with the compressed PCM interfaces (PCM format conversion and synchronous tandem coding adjustment) may be omitted. In a fixed part which provides a digital speech connection the full algorithm in G.721 shall be used. Informative Annex E contains recommendations concerning interim arrangements for portable part codecs.

8.2 Speech transmission algorithm

8.2.1 Speech coding algorithm

The speech coding algorithm corresponding to CIC = 0 (see subclause 7.2.10) shall comply with the definitions in subclause 8.1.4.

Compliance shall be by supplier's declaration.

8.2.2 Codec for telepoint CFP

A telepoint CFP shall contain a codec that conforms to CCITT Recommendation G.721 [6].

Compliance shall be by supplier's declaration.

8.3 Bit transmission sequence

The sixteen complete ADPCM words comprising each burst shall be transmitted in chronological order, and with the most significant bit transmitted first within each word.

Compliance shall be by supplier's declaration.

8.4 Frequency responses

To allow total compliance with national standard for any combination of handset and base unit, the sum of the handset to CAI and CAI to base unit telephone line frequency responses shall meet the appropriate published national requirements.

8.4.1 Sending frequency response

The handset sending frequency response (from MRP to CAI) shall be below the upper limit and above the lower limit defined in table 12 and shown in figure 17.

Table 12: Co-ordinates of sending response limit curves

Limit curve	Frequency Hz	Sending response dB(arbitrary level)
Upper limit	100	-18
	200	- 5
	3400	+ 2
	4000	0
Lower limit	300	-inf
	300	-12
	1000	- 5
	3000	- 6
	3400	- 8
	3400	-inf

The limit curves shall be determined by straight lines joining successive co-ordinates given in the table, when frequency response is plotted on a linear scale against frequency on a logarithmic scale.

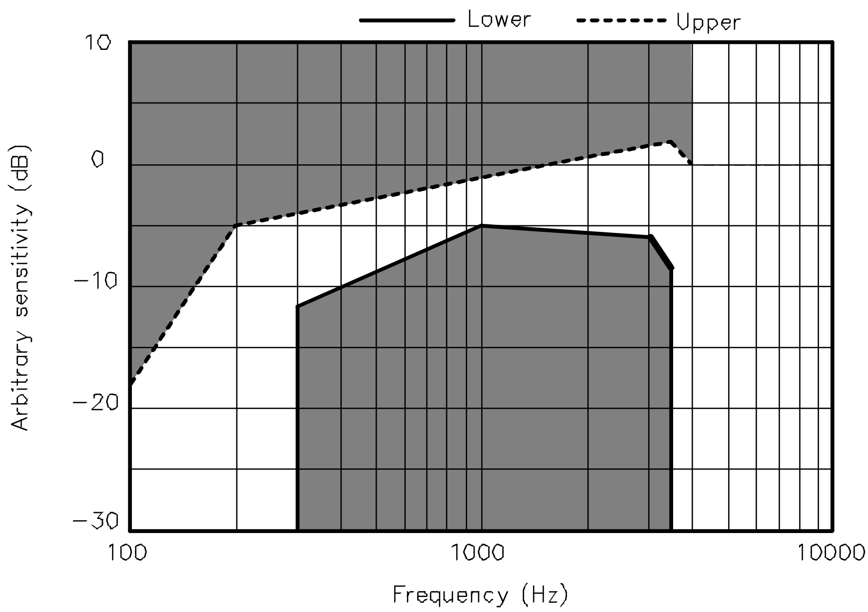


Figure 17: Sending frequency response mask

Compliance shall be checked by the test described in subclause 11.4.

8.4.2 Receiving frequency response

The handset receiving frequency response (from CAI to ERP) shall be below the upper limit and above the lower limit defined in table 13 and shown in figure 18.

Table 13: Co-ordinates of receiving response limit curves

Limit curve	Frequency Hz	Sending response dB(arbitrary level)
Upper limit	100	-18
	160	- 5
	300	0
	1000	0
	3000	+ 2
	4000	+ 2
Lower limit	300	-inf
	300	- 8
	500	- 4
	3000	- 4
	3400	- 8
	3400	-inf

The limit curves shall be determined by straight lines joining successive co-ordinates given in the table, when frequency response is plotted on a linear scale against frequency on a logarithmic scale.

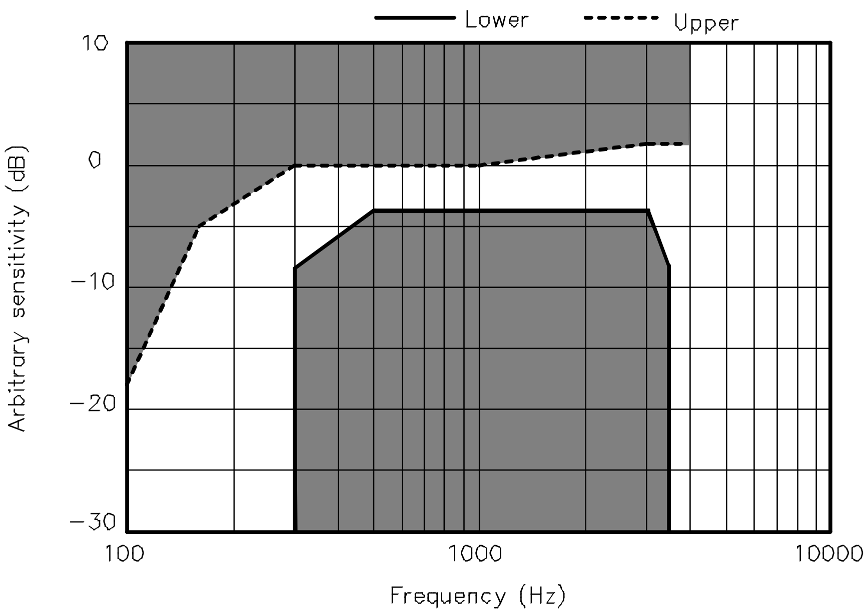


Figure 18: Receiving frequency response mask

Compliance shall be checked by the test described in subclause 11.5.

8.5 Digital signal level

The "level" of the digital signal at the uniform PCM interface is defined in dBm0. A 1 kHz sine wave whose peak signal corresponds with the maximum PCM code is assigned a level of +3,14 dBm0 (CCITT Recommendation G.711 [15]). The "impedance" associated with signals at a digital interface shall be 600 ohms. Hence 0 dBm0 corresponds to 775 mV rms or -2,2 dBV.

8.6 Sending and receiving loudness ratings

Two loudness ratings are defined: the handset loudness ratings (SLRH, RLRH) between the acoustic reference points and the uniform PCM interface.

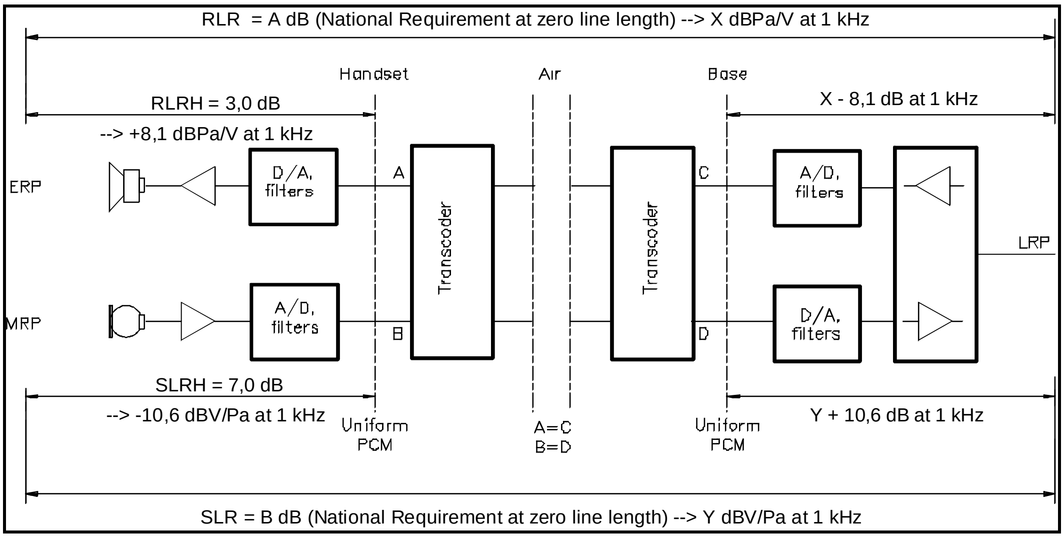
The loudness ratings shall be within the following limits:

i) the handset send loudness rating (SLRH) shall be 7,0 dB ± 3,0 dB; and

ii) if the handset is not fitted with a user-controlled receiving volume control, the handset receive loudness rating (RLRH) shall be 3,0 dB ± 3,0 dB;

iii) if the handset is fitted with a user-controlled receiving volume control, the maximum handset receive loudness rating (RLRH) shall not exceed -7,0 dB ± 3,0 dB. For at least one position of the receiving volume control, the RLRH shall meet the specification for a handset not fitted with a user-controlled receiving volume control. Compliance with specifications shall be checked with the volume control set at maximum unless otherwise stated.

The SLRH and RLRH shall be checked as described in subclauses 11.6 and 11.7 of this specification.



NOTE 1: Line interface gains are for zero line length.

NOTE 2: Any companding/CODEC functions are contained within the "A/D, filters" blocks.

NOTE 3: National frequency response adjustments may be made in the "A/D, filters" blocks in the CFP.

Figure 19: CAI high-level transmission plan

For guidance, an equivalent high level transmission plan, derived from the specified RLRH and SLRH and a national specification, showing nominal 1 kHz levels, is shown in figure 19. The nominal send and receive sensitivities from acoustic reference points to CAI are:

i) send sensitivity at nominally 1 kHz (MRP to CAI) of -10,6 dBV/Pa; and

ii) receive sensitivity at nominally 1 kHz (CAI to ERP) of +8,1 dBPa/V.

8.7 Sidetone loudness ratings

8.7.1 Sidetone for analogue telephony

When the speech path between the handset and the analogue telephone line connection at the associated base unit is established, negligible local sidetone shall be provided within the handset. Sidetone shall be provided by the base unit (CFP).

8.7.2 Sidetone for digital telephony

For operation of the handset with a "digital" base unit, sidetone shall be provided locally within the handset. This shall be under control of the layer three signalling message as defined in subclause 7.2.6.

8.7.2.1 Talker sidetone

The value of the sidetone masking rating (STMR) shall be within the range 13 dB ± 5 dB.

Where a user-controlled receiving volume control is provided, the STMR shall meet the above requirement at the volume control setting where the RLRH is at the nominal value.

Compliance shall be checked by the test described in subclause 11.8.

NOTE: It is recommended that the sidetone level is independent of the receiving volume control.

8.7.2.2 Listener sidetone

The value of the listener sidetone ratio shall not be less than 15 dB.

Where a user-controlled receiving volume control is provided, the LSTR shall meet the above requirement at the volume control setting where the RLRH is at the nominal value.

Compliance shall be checked by the test described in subclause 11.22.

8.8 Clipping

The apparatus shall be designed such that clipping occurs by "saturation" of the digital speech encoding function.

8.9 Distortion

NOTE: CT2 apparatus containing ADPCM processing will by its nature cause some distortion of signals transmitted over the speech channel. It cannot therefore be expected that such apparatus meet the same requirements for distortion as apparatus which does not utilise ADPCM processing.ETS 300 085 [7] expects only 1 QDU of distortion (CCITT Recommendation G.711 [15]), rather than 3,5 QDU (CCITT Recommendation G.113 [17] and G.721 [6]) which CT2 apparatus will in reality produce. Similarly apparatus which includes ADPCM processing may not be able to meet the linearity requirements for apparatus designed to meet CCITT Recommendation G.711 [15] only. To ensure there is no gross distortion contributed by the analogue circuitry, the following tests are included.

8.9.1 Sending distortion

For a pure tone of nominal frequency 1 kHz and level -4,7 dBPa applied at the MRP, the ratio of signal to total distortion (harmonic and quantising) shall be not less than 35 dB.

Compliance shall be checked by the test of subclause 11.9.

8.9.2 Receiving distortion

For a pure tone of nominal frequency 1 kHz applied at the reference CFP so as to produce a level of -10 dBm0 at the uniform PCM interface, the ratio of signal to total distortion (harmonic and quantising) measured at the ERP shall be not less than 35 dB.

Compliance shall be checked by the test of subclause 11.10.

8.9.3 Sidetone distortion

The third harmonic distortion generated by the terminal equipment shall not be greater than 10%.

Compliance shall be checked by the test of subclause 11.23.

8.10 Noise

8.10.1 Sending

The noise produced by the apparatus in the sending direction shall not exceed -68 dBm0p (psophometric weighting).

Compliance shall be checked by the test of subclause 11.11.

8.10.2 Sending (narrow-band noise)

The narrow-band noise produced by the apparatus in the sending direction and contained within any 10 Hz bandwidth between the frequency limits 300 Hz to 3400 Hz, shall not exceed -73 dBm0.

Compliance shall be checked by the test of subclause 11.12.

8.10.3 Receiving

The noise produced by the apparatus under test and measured at the ERP shall not exceed -60 dBPa A-weighted. If the apparatus is fitted with a user-controlled receiving volume control, the requirement shall be met for the volume control setting for which the RLRH is equal to the nominal value.

Compliance shall be checked by the test of subclause 11.13.

8.11 Delay

Due to the burst nature of the CT2 transmission, the total "loop" delay is a combination of conventional analogue delays of transducers, filters etc. and the digital delay caused by the burst structure. In order to ensure different types of CPP can interwork with different types of CFP, it is necessary to define the delay of the CPP and the delay in the CFP.

8.11.1 Handset delay

The delay between commencement of an acoustic input at the CPP and its arrival at the receiving reference point of the CPP when operating with a reference CFP, shall not exceed 4,0 ms when averaged over the frequency range 500 Hz to 2500 Hz.

NOTE: This includes the delay caused by the reference CFP in looping back the ADPCM (digital) signal.

Compliance shall be checked by the method given in subclause 11.14.

8.11.2 Base delay

When measured with the worst-case handset permitted under 8.11.1 the base station shall meet the following requirement. The sum of the delay from MRP to the line interface and the delay from the line interface to the ERP shall not exceed 5,0 ms when averaged over the frequency range 500 Hz to 2500 Hz.

Compliance shall be checked by manufacturer's declaration.

8.11.3 Network echo from a CFP with a 2-wire analogue interface

Informative Annex L contains information relating to network echo from a CFP with a 2-wire analogue interface.

8.12 Terminal coupling loss

8.12.1 Weighted terminal coupling loss (TCLw)

Weighted terminal coupling loss shall be measured with the handset suspended in free air (free field). The handset shall be positioned at least 0,5 m away from the nearest part of the test chamber. TCLw measured from the digital input to digital output shall be at least 34 dB. Informative Annex E contains recommendations concerning interim arrangements for weighted terminal coupling loss.

Compliance shall be checked by the test of subclause 11.15.

8.12.2 Stability loss (fixed geometry CPPs)

With the handset lying on and the transducers facing a hard surface, the attenuation from the digital input to the digital output shall be at least 6 dB at all frequencies in the range 200 Hz to 4000 Hz. If the apparatus is fitted with a user-controlled receiving volume control, the requirement shall be met for the volume control setting for which the RLRH is equal to the nominal value.

Compliance shall be checked by the test of subclause 11.16.

8.12.3 Stability loss (variable geometry CPPs)

If the apparatus is fitted with a user-controlled receiving volume control, the requirement shall be met for the volume control setting for which the RLRH is equal to the nominal value. The apparatus shall be capable of meeting at least one of the following conditions i) or ii).

i) If it is possible to position the earpiece directly in front of the mouthpiece with a distance of 150 mm between the front planes of each, the attenuation from the digital input to the digital output shall be at least 6 dB at all frequencies in the range 200 Hz to 4000 Hz: (a) in this relative position; and (b) in the just off-hook position.

ii) If the relative movement and orientation of the acoustic and electro-acoustic elements are limited by means of a hinge or similar mechanism, the attenuation from the digital input to the digital output shall be at least 6 dB at all frequencies in the range 200 Hz to 4000 Hz with the transducers in any relative position and orientation that can be achieved whilst the CPP is in the active condition (i.e. the B channel is established between the CPP and an associated reference CFP).

Compliance shall be checked by the test of subclause 11.17.

8.12.4 CFP with a 4-wire interface

Informative Annex K contains information relating to artificial echo loss for CFPs with a 4-wire connection.

8.13 Out of band signals

8.13.1 Discrimination against out-of-band input signals (sending)

With any sine-wave signal in the range 4,6 kHz to 8,0 kHz applied at the MRP at a level of -4,7 dBPa, the level of any image frequency produced at the digital interface shall be below the following limit. For sine-wave signals of 4,6 kHz and 8,0 kHz the reference limit levels shall be -30 dB and -40 dB respectively below a reference level obtained at 1 kHz and the same acoustic input level.

The image frequency limit shall be defined by a straight line joining the two reference limit levels when drawn on a logarithmic (frequency) - linear (dB sensitivity) scale.

Compliance shall be checked by the test of subclause 11.18.

8.13.2 Spurious out-of-band signals (receiving)

For a sine wave in the frequency range 300 Hz to 3400 Hz applied at the reference base unit so as to cause a level of 0 dBm0 at the digital interface, the level of spurious out-of-band image signals measured selectively at the ERP shall be lower than the in-band acoustic level obtained by a digital input signal of -35 dBm0 at 1 kHz.

Compliance shall be checked by the test of subclause 11.19.

8.14 Sampling frequency level (receiving)

The level of any 8 kHz acoustic signal at the ERP shall be less than -70 dBPa. If the apparatus is fitted with a user-controlled receiving volume control, the requirement shall be met for the volume control setting for which the RLRH is equal to the nominal value.

Compliance shall be checked by the test of subclause 11.20.

8.15 Acoustic shock

8.15.1 Maximum intended sound pressure level

With a digitally encoded signal representing the maximum possible signal at the digital interface, the sound pressure level at the ERP shall not exceed +24 dBPa (rms unweighted).

Compliance shall be checked by the test of subclause 11.21.

8.15.2 Maximum possible sound pressure level

The sound output from the receiver shall be limited by the power output capability of the receiver drive amplifier to give a peak sound pressure at the ERP not greater than 36 dBPa under any continuous or transient conditions.

Compliance shall be by supplier's declaration.

8.16 Audible incoming call indication

8.16.1 Provided on CPP: sound pressure level

If audible incoming call indication is provided anywhere on the CPP, the sound pressure level at the ERP shall not exceed 24 dBPa.

NOTE: The initial sound pressure in 8.16.1 should not exceed 0 dBPa and should rise in increments no greater than 6 dB, at a rate not greater than 6 dB/s, to a maximum within not less than 6 s.

8.16.2 Generated other than through the earpiece: maximum sound pressure level

If audible incoming call indication on the CPP is generated other than through the earpiece, the sound pressure at the commencement of such indication shall not exceed 50 dB A-weighted at 1 m free field in any direction, and shall also comply with 8.16.1.

NOTE: The initial level in 8.16.2 should rise in increments no greater than 6 dB, at a rate not greater than 6 dB/s, to a maximum within not less than 6 s.

9 Radio frequency parametric and system tests

9.1 Test conditions, power sources and ambient temperatures

9.1.1 Normal and extreme test conditions

Type approval tests shall be made under normal test conditions, and also, where stated, under extreme test conditions.

9.1.2 Test power source

During the approval tests, the power source of the equipment shall be replaced by a test power source, capable of producing normal and extreme test voltages as specified in subclauses 9.1.3.2 and 9.1.4.2. The internal impedance of the test power source shall be low enough for its effect on the test results to be negligible. For the purpose of tests, the voltage of the power source shall be measured at the input terminals of the equipment.

If the equipment is provided with a permanently connected power cable, the test voltage shall be measured at the point of connection of the power cable to the equipment.

In equipment with incorporated batteries, the test power source shall be applied as close to the battery terminals as practicable.

During tests, the power source voltages shall be maintained within a tolerance of ±3% relative to the voltage at the beginning of each test.

9.1.3 Normal test conditions

9.1.3.1 Normal temperature and humidity

The normal temperature and humidity conditions for tests shall be any convenient combination of temperature and humidity within the following ranges:

temperature +15°C to +35°C,

relative humidity 20% to 75%.

It should be noted that when it is impracticable to carry out the tests under these conditions, a statement giving the actual temperature and relative humidity during the tests, shall be added to the test report.

9.1.3.2 Normal test power source

9.1.3.2.1 Mains voltage

The normal test voltage for equipment to be connected to the mains shall be the nominal mains voltage. For the purpose of this specification, the nominal voltage shall be the voltage or voltages for which the equipment was designed as declared by the manufacturer. The frequency of the test power source corresponding to the AC mains shall be between 49 Hz and 51 Hz.

9.1.3.2.2 Regulated lead acid battery power sources

When the radio equipment is intended for operation from the usual type of regulated lead acid battery source, the normal test source voltage shall be 1,1 times the nominal voltage of the battery (6 volts, 12 volts etc.).

9.1.3.2.3 Nickel cadmium battery

When the equipment is intended for operation from the usual type of nickel cadmium battery, the normal test voltage shall be the nominal voltage of the battery (1,2 volt per cell).

9.1.3.2.4 Other power sources

For operation from other power sources or types of battery, either primary or secondary, the normal test source voltage shall be that declared by the equipment manufacturer.

9.1.4 Extreme test conditions

9.1.4.1 Extreme temperatures

For tests at extreme temperatures, measurements shall be made in accordance with the procedures specified in subclause 9.1.5 at an upper value of +40°C and at a lower value of 0°C.

9.1.4.2 Extreme test source voltages

9.1.4.2.1 Mains voltage

The extreme test source voltages for equipment to be connected to an AC mains source shall be the nominal mains voltage ±10%. The frequency of the test power source shall be between 49 Hz and 51 Hz.

9.1.4.2.2 Regulated lead acid battery power sources

When the equipment is intended for operation from the usual type of regulated lead acid battery source, the extreme test voltages shall be 1,3 and 0,9 times the nominal voltage of the battery.

9.1.4.2.3 Nickel cadmium battery

When the equipment is intended for operation from the usual type of nickel cadmium battery, the extreme test voltages shall be 1,25 and 0,85 times nominal voltage of the battery.

9.1.4.2.4 Other power sources

The lower extreme test voltage for equipment with power sources using primary batteries shall be as follows:

i) for Leclanché type of battery - 0,85 times the nominal voltage;

ii) for other types of primary battery - the end point voltage declared by the equipment manufacturer.

For equipment using other power sources, or capable of being operated from a variety of power sources, or designed for operation within extreme voltage limits not in accordance with those quoted above the extreme test voltages shall be those agreed between the equipment manufacturer and the testing authority and shall be recorded with the test results.

9.1.5 Procedure for tests at extreme temperatures

Before measurements are made the equipment shall have reached thermal balance in the test chamber. The equipment shall be switched off during the temperature stabilising period. If the thermal balance is not checked by measurements, a temperature stabilising period of at least one hour, or such period as may be decided by the testing authority, shall be allowed. The sequence of measurements shall be chosen, and the humidity content in the test chamber shall be controlled, so that excessive condensation does not occur.

Before tests at the upper temperature the equipment shall be placed in the test chamber and left until thermal balance is attained. The equipment shall then be switched on in the active condition for a period of half an hour after which the equipment shall meet the specified requirements.

9.2 Electrical test conditions

9.2.1 Arrangements for signals to be applied to the fixed and portable receivers

The cordless telephone equipment utilises radio frequency link control protocols involving the transmission of a handshake code between the fixed and portable parts to maintain the radio frequency communication link. Subclause 5.5.1.6 contains a requirement for the radio frequency link to cease operation if a time greater than 10 seconds has elapsed without a successful handshake taking place.

In order to carry out the radio frequency tests contained in this specification it is necessary to arrange for transmission of the relevant handshake code to be maintained for the duration of the tests. This handshake shall be obtained by coupling the fixed or portable part under test to its associated portable or fixed part such that reliable handshaking is established. If the equipment is fitted with dynamic radio frequency output power control it should operate at its maximum power.

In the case of equipment with an integral antenna, the required level of coupling shall be achieved by, placing the associated fixed part (with if necessary an antenna connected) or portable part, at a distance such as to produce the signal required for link establishment. In the case of equipment with antenna terminals, or when an equipment with an integral antenna is being tested in the test fixture, a radio frequency coupling network shall apply the correct signal level.

Care should be taken to ensure that the coupling method employed causes the minimum effect on the test results.

9.2.2 Artificial antenna

Tests on the transmitter shall be carried out with a substantially non-reactive non-radiating 50 ohm load connected to the terminals, or in the case of equipments with integral antenna, to the test fixture terminal.

9.2.3 Test fixture for integral antenna

In the case of equipment intended for use with an integral antenna, the manufacturer shall supply a test fixture suitable to allow relative measurements to be made on the submitted sample.

This test fixture shall provide a 50 ohm radio frequency terminal at the working frequencies of the equipment.

The test fixture shall provide means of making external connection to at least the radio frequency input and output and of replacing the power source by external power supply.

The performance characteristics of this test fixture under normal and extreme conditions will be subject to the approval of the testing authority.

The characteristics of interest to the testing authority will be that:

i) the coupling loss shall not be excessive, that is not greater than 20 dB; and

ii) the variation of coupling loss with frequency shall not cause errors exceeding 2 dB in measurements using the test fixture; and

iii) the coupling device shall not include any non-linear elements.

The testing authority may provide its own test fixture.

9.2.4 Test site and general arrangements for measurements involving the use of radiated fields

9.2.4.1 Test site

The test site shall be on a reasonably level surface or ground.

At one point on the site, a ground plane of at least 5 metres diameter shall be provided. In the middle of this ground plane, a non-conducting support, capable of rotation through 360° in the horizontal plane, shall be used to support the test sample at 1,5 metres above the ground plane. The test site shall be large enough to allow the erection of a measuring or transmitting antenna at a distance of lambda/2 or 3 metres whichever is the greater. The distance actually used shall be recorded with the results of the tests carried out on the site.

Sufficient precautions shall be taken to ensure that reflections from extraneous objects adjacent to the site and ground reflections do not degrade the measurement results.

9.2.4.2 Test antenna

When the site is used for radiation measurements the test antenna shall be used to detect the radiation from both the test sample and the substitution antenna.

This antenna shall be mounted on a support to allow the antenna to be used in either horizontal or vertical polarisation and for the height of its centre above ground to be varied over the range 1 to 4 metres. Preferably a test antenna with pronounced directivity should be used. The size of the test antenna along the measurement axis shall not exceed 20% of the measuring distance.

The test antenna shall be connected to a test receiver capable of being tuned to any frequency under investigation and of measuring accurately the relative levels of signals at its input.

9.2.4.3 Substitution antenna

The substitution antenna shall be a lambda/2 dipole, resonant at the frequency under consideration, or a shortened dipole, calibrated to the lambda/2 dipole. The centre of this antenna shall coincide with the reference point of the test sample it has replaced. This reference point shall be the volume centre of the sample when its antenna is mounted inside the cabinet, or the point where an external antenna is connected to the cabinet.

The distance between the lower extremity of the dipole and the ground shall be at least 0,3 metre.

The substitution antenna shall be connected to a calibrated signal generator.

The signal generator and the receiver shall be operated at the frequencies under investigation and shall be connected to the antenna through suitable matching and balancing networks.

9.2.4.4 Optional additional indoor site

When the frequency of the signals being measured is greater than 80 MHz, use may be made of an indoor site. If this alternative site is used, this shall be recorded in the test report.

The measurement site may be a laboratory room with a minimum area of 6 metres by 7 metres and at least 2,7 metres in height.

Apart from the measuring apparatus and the operator, the room shall be as free as possible from reflecting objects other than the walls, floor and ceiling.

The site arrangement is shown in principle in figure 20.

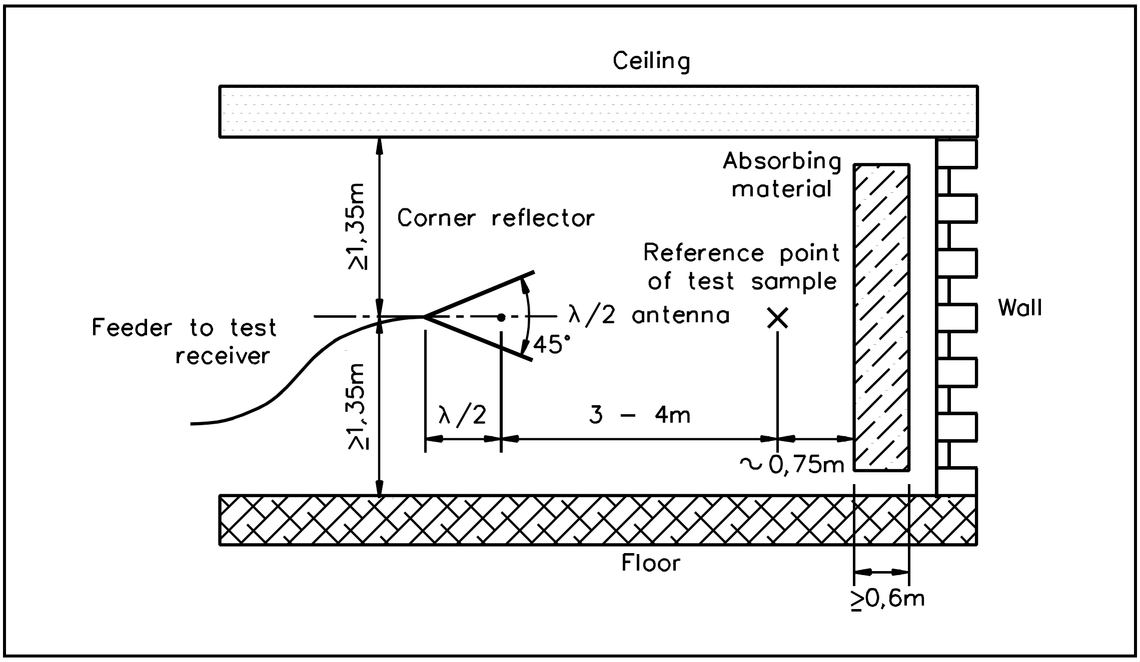


Figure 20: Indoor site arrangement

The potential reflections from the wall behind the equipment under test are reduced by placing a barrier of absorbent material in front of it. The corner reflector around the test antenna is used to reduce the effect of reflections from the opposite wall and from the floor and ceiling in the case of horizontally polarised measurements.

Similarly, the corner reflector reduces the effects of reflections from the side walls vertically polarised measurements.

For the lower part of the frequency range (below approximately 175 MHz) no corner reflector or absorbent barrier is needed.

For practical reasons, the lambda/2 antenna in figure 20 may be replaced by an antenna of constant length, provided that this length is between lambda/4 and lambda at the frequency of measurement and the sensitivity of the measuring system is sufficient. In the same way the distance of lambda/2 to the apex may be varied.

The test antenna, test receiver, substitution antenna and calibrated signal generator are used in a way similar to that of the general method.

To ensure that errors are not caused by the propagation path approaching the point at which phase cancellation between direct and the remaining reflected signals occurs, the substitution antenna shall be moved through a distance of ±0,1 metre in the direction of the test antenna as well as in the two directions perpendicular to this first direction. If these changes of distance cause a signal change of greater than 2 dB, the test sample should be resited until a change of less than 2 dB is obtained.

9.2.5 Transceiver test facility

The manufacturer shall supply facilities to enable control of those functions of the equipment which are associated with the parameters measured in Clause 9.

Adequate operating instructions relevant to the equipment submitted for test shall be provided.

Control shall be provided of switching between the normal active mode (at maximum rated transmitter power) and the idle mode.

Connections shall be provided to enable external access to the equipment power supply.

The manufacturer shall ensure that the control and connection facilities provided do not have a significant effect on the measured results.

9.3 Transmitter

The tests in this subclause require the testing authority to provide a reference part (CPP and CFP), or the manufacturer to supply a complete CTA.

9.3.1 Transmitter carrier power

9.3.1.1 Definition

The transmitter carrier power is the mean power delivered to the artificial antenna during a radio frequency cycle or, in the case of equipment with integral antenna, the effective radiated power in the direction of maximum field strength under specified conditions of measurement (subclause 9.2.4) if possible in the absence of modulation. The rated output power is the carrier power declared by the manufacturer.

9.3.1.2 Method of measurement for equipment with an antenna connection

The handshake code between the fixed and portable parts is established as described in subclause 9.2.1.

The transmitter shall be connected to an artificial antenna (subclause 9.2.2), and the power delivered to this artificial antenna shall be measured.

The mean power measured shall be multiplied by 2 to obtain the carrier power.

The measurements shall be made under normal test conditions (subclause 9.1.3) and extreme test conditions (subclause 9.1.4.1 and 9.1.4.2 applied simultaneously).

9.3.1.3 Method of measurement for equipment with an integral antenna.

9.3.1.3.1 Method of measurement under normal test conditions

On a test site, fulfilling the requirements of subclause 9.2.4 the sample shall be placed on the support in the following position:

i) for equipment with an internal antenna it shall stand so that the axis, of the equipment which in its normal use is closest to the vertical, shall be vertical;

ii) for equipment with a rigid external antenna, the antenna shall be vertical;

iii) for equipment with a non-rigid external antenna, the antenna shall be extended vertically upwards by a non-conducting support.

The handshake code between the fixed and portable parts is established as described in subclause 9.2.1. The test receiver shall be tuned to the frequency of the signal being measured.

The test antenna shall be orientated for vertical polarisation and shall be raised or lowered through the specified height range until a maximum signal level is detected on the test receiver. The transmitter shall then be rotated through 360° until the maximum signal is received.

It should be noted that the maximum may be a lower value than the value obtainable at heights outside the specified limits.

The transmitter shall be replaced by the substitution antenna, as defined in subclause 9.2.4.3 and the test antenna raised or lowered as necessary to ensure that the maximum signal is still received. The input signal to the substitution antenna shall be adjusted in level until an equal or a known related level to that detected from the transmitter is obtained in the test receiver.

The carrier power is equal to the power supplied to the substitution antenna, increased by the known relationship if necessary.

The measurement shall be repeated for any alternative antenna supplied by the manufacturer.

A check should be made at other planes of polarisation to ensure that the value obtained above is the maximum. If larger values are obtained, this fact should be recorded in the test report.

9.3.1.3.2 Method of measurement under extreme test conditions

The equipment shall be placed in the test fixture (subclause 9.2.3). The handshake code between the fixed portable parts is established as described in subclause 9.2.1. The power delivered to the artificial antenna shall be measured. The measurements shall be made under normal test conditions (subclause 9.1.3) and extreme test conditions (subclauses 9.1.4.1 and 9.1.4.2 applied simultaneously).

The mean power measured shall be multiplied by 2 to obtain the carrier power.

9.3.1.4 Limits

The limits shall be those specified in subclause 4.5.1.

9.3.2 Adjacent channel power (narrow-band measurement)

9.3.2.1 Definition

The adjacent channel power is that part of the total power output of a transmitter under defined conditions of modulation, which falls within a specified pass-band centred on the nominal frequency of either of the adjacent channels. This power is the sum of the mean power produced by the modulation, hum and noise of the transmitter.

Adjacent channel power for the purpose of this specification will be deemed to include performance of the transmitter under extreme test conditions.

9.3.2.2 Method of measurement

The adjacent channel power shall be measured with a spectrum analyser which conforms to subclause 9.3.2.3.

Equipment with an antenna terminal shall have the terminal connected to a spectrum analyser by a coupling device which provides the appropriate input level to the spectrum analyser. Equipment with an integral antenna shall be placed in the test fixture (subclause 9.2.3) and the radio frequency output of the test fixture shall be applied to the spectrum analyser at the appropriate input level. The handshake code between the fixed and portable parts is established as described in subclause 9.2.1. The transmitter shall be operated at the measured carrier power (subclause 9.3.1) under normal test conditions (subclause 9.1.3) such as to produce a modulated output representative of normal active use (subclause 9.6).

The spectrum analyser shall be adjusted so that the spectrum of the transmitter output, including that part which falls in the adjacent channels, is displayed.

For the purpose of this test the integration bandwidth used in this measurement shall be 80 kHz with a tolerance of ±5%.

The centre frequency of the bandwidth within which measurements are to be made shall have a 100 kHz separation from the nominal carrier frequency of the transmitter.

The adjacent channel power is the sum of the power levels of each of the discrete components and of the noise falling in the appropriate bandwidth.

This sum may be automatically calculated by the spectrum analyser, or an automatic power level integrating device may be used to obtain it (see subclause 9.3.2.4).

In the latter case, the relative power level of the modulated transmitter is initially measured by integration over the appropriate bandwidth, centred on the nominal frequency. The measurement is repeated with this bandwidth centred on the nominal frequency of the adjacent channel and the input level to the integrating device is increased until the same power level at the output of the device is obtained.

The difference between the input levels, in dB, gives the ratio of the adjacent channel power to the carrier power.

The adjacent channel power, expressed as an effective radiated power, is calculated by applying this ratio to the carrier power as determined in subclause 9.3.1.

The measurement shall be repeated for the other adjacent channel.

9.3.2.3 Characteristics of the spectrum analyser

Characteristics of the spectrum analyser shall meet at least the following requirements:

It shall be possible to measure the amplitude of a signal or noise at a level 3 dB or more above the noise level of the spectrum analyser, as displayed on the screen, to an accuracy of ±2 dB in the presence of a signal separated in frequency by 10 kHz at a level 90 dB above that of the signal to be measured.

The reading accuracy of the frequency marker shall be within ±2 kHz.

The accuracy of relative amplitude measurements shall be within ±1 dB.

It shall be possible to adjust the spectrum analyser to allow the separation on its screen of two components with a frequency difference of 1 kHz.

The video bandwidth should be relatively low e.g. 1 kHz.

9.3.2.4 Integrating and power summing device

This device would only be used if the sum of the components and the noise has not been calculated automatically. It is connected to the video output of the spectrum analyser, described in subclause 9.3.2.3. It shall be possible to sum the effective power of all discrete components and the noise power falling in the selected bandwidth and to measure this as a ratio relative to the carrier power.

The position and the width of the integration range selected can be indicated on the spectrum analyser by brightening the trace.

When measuring power levels as low as 50 nW, the output of the device should exceed the integral noise level by at least 10 dB. The dynamic range shall permit measurement of the values required under subclause 9.3.2.5 with a reserve of at least 10 dB.

The measurement shall be repeated under extreme conditions (subclauses 9.1.4.1 and 9.1.4.2 applied simultaneously).

9.3.2.5 Limits

The limits shall be those specified in subclause 4.5.5.

9.3.3 Out of band power arising from transmitter transients

9.3.3.1 Definition

The out-of-band power arising from transmitter transients is the peak power of the modulation products, arising from the rapid switching on and off of the transmitter, which fall within a defined frequency band either side of the nominal frequency.

9.3.3.2 Method of measurement

If the transmitter is equipped with an antenna terminal it shall be connected to a spectrum analyser by a coupling device which provides the appropriate input level to the spectrum analyser. If the transmitter is equipped with an integral antenna it shall be placed in the test fixture (subclause 9.2.3) and the radio frequency output of the test fixture applied to the spectrum analyser at the appropriate input level.

The handshake code between the fixed and portable parts is established as described in subclause 9.2.1. The transmitter shall be operated at the measured carrier power (subclause 9.3.1) under normal test conditions (subclause 9.1.3) such as to produce a modulated output representative of normal active use (subclause 9.6).

The spectrum analyser shall be adjusted so that the spectrum of the transmitter output, including the part which falls in the bands 0,5 MHz either side of the nominal frequency is displayed.

9.3.3.3 Characteristics of the spectrum analyser

Characteristics of the spectrum analyser shall meet at least the following requirements:

- the spectrum analyser shall be suitable for making measurements on signals resulting from switching transients;

- the spectrum analyser shall be provided with a 4 pole synchronously tuned intermediate frequency filter;

- the spectrum analyser shall be operated in the peak hold mode;

- the resolution bandwidth shall be set to 10 kHz and the video bandwidth to 3 MHz;

- the levels displayed on the spectrum analyser at frequencies 100 kHz and 0,5 MHz above and below the nominal signal frequency shall be recorded.

9.3.3.4 Limits

The limits shall be those specified in subclause 4.5.6.

9.3.4 Intermodulation attenuation

This requirement applies to transmitter/receivers to be contained (nested) in a single enclosure or a single unit containing two or more transmitters/receivers which are not separable.

9.3.4.1 Definition

For the purpose of this specification the intermodulation attenuation is a measure of the capability of a transmitter to inhibit the generation of signals in its non-linear elements caused by the presence of the carrier and an interfering signal.

9.3.4.2 Method of measurement

Two transmitters/receivers of the type which will be contained (nested) in a single enclosure shall be operated in the enclosure immediately adjacent to each other. Where the transmitter/receivers are equipped with antenna terminals, these shall be connected to the antenna combining system and the antenna which will be employed with the commercial product.

On a test site, fulfilling the requirements of subclause 9.2.4 the sample shall be placed on the support in the following position:

i) For equipment with an internal antenna, it shall stand so that the axis of the equipment which is in its normal use is closest to vertical shall be vertical;

ii) For equipment with a rigid external antenna, the antenna shall be vertical;

iii) For equipment with a non-rigid external antenna, the antenna shall be extended vertically upwards by a non-conducting support.

The handshake codes for the two systems is established as described in subclause 9.2.1.

The transmitters shall be operated at the power levels measured under subclause 9.3.1.

Radiation of any third order intermodulation products shall be detected by the test antenna and a spectrum analyser with a resolution bandwidth of 10 kHz and a video bandwidth of 30 kHz.

At the frequencies at which products are detected, the equipment under test shall be rotated to obtain the maximum response, and the effective radiated power of that product determined by a substitution measurement.

The measurement shall be repeated with the test antenna in the orthogonal polarisation plane.

9.3.4.3 Limits

The limits shall be those specified in subclause 4.5.7.

9.3.5 Prevention of mis-operation due to adverse power supply conditions

9.3.5.1 Definition

For the purpose of this specification mis-operation shall be defined as the generation of emissions outside the specified limits due to a reduction of power supply voltages.

9.3.5.2 Method of measurement

i) the transmitter/receiver under test shall be placed in the test fixture or connected to a suitable artificial load. The handshake code between the fixed and portable parts is established as described in subclause 9.2.1. The emission shall be monitored on a spectrum analyser;

ii) the radiated spectrum shall be monitored whilst the supply voltage (AC or DC) shall be slowly reduced from the normal value to zero at the rate recommended by the equipment manufacturer;

iii) the levels of adjacent channel power and spurious emissions shall be measured and recorded.

9.3.5.3 LimitsThe limits shall be those specified in subclause 4.7.1.

NOTE 1: If a back up power supply is provided in the base unit (i.e. a rechargeable battery) the test shall be repeated with the battery replaced by a variable DC power supply.

NOTE 2: Any non-repetitive transient condition (of duration less than 50 ms) shall be ignored.

9.4 Spurious emissions

The tests in this subclause require the testing authority to provide a reference part (CPP and CFP), or the manufacturer to supply a complete CTA.

9.4.1 Spurious emissions of the combined transmitter/receiver

9.4.1.1 Definition

Spurious emissions are emissions at frequencies other than those of the carrier and sidebands associated with normal modulation.

The level of spurious emissions shall be measured as:

i) their power level in a transmission line or antenna and

ii) their effective radiated power when radiated by the cabinet and structure of the equipment. This is also known as "cabinet radiation".

For equipment which can only be used with an integral antenna, only the measurement mentioned under (ii) applies.

9.4.1.2 Method of measuring the power level (i)

Spurious emissions shall be measured as the power level of any discrete signal delivered into a 50 ohm load. This may be done by connecting the transmitter/receiver output through an attenuator to a spectrum analyser with a resolution bandwidth of 100 kHz and a video bandwidth of 300 kHz or by monitoring the relative levels of the spurious signals delivered to an artificial antenna (subclause 9.2.2)

The handshake code between the fixed and portable parts is established as described in subclause 9.2.1. The measurements shall be made over the frequency range 100 kHz to 12,75 GHz, except for the channel on which the transmitter/receiver is operating and its adjacent channels.

The measurement shall be repeated with the transmitter/receiver in the idle mode.

9.4.1.3 Method of measuring the effective radiated power (ii)

On a test site, fulfilling the requirements of subclause 9.2.4, the sample shall be placed at the specified height on a non-conducting support. The handshake code between the fixed and portable parts is established as described in subclause 9.2.1.

The transmitter/receiver shall be operated with the carrier power delivered to an artificial antenna (subclause 9.2.2), except in the case of testing equipment with an integral antenna.

Radiation of any spurious components shall be detected by the test antenna and a spectrum analyser with a resolution bandwidth of 100 kHz and a video bandwidth of 3 MHz over the frequency range 25 MHz to 12,75 GHz, except for the channel on which the transmitter is intended to operate and its adjacent channels.

At each frequency at which a component is detected, the sample shall be rotated to obtain the maximum response and the effective radiated power of that component determined by a substitution measurement.

The measurement shall be repeated with the test antenna in the orthogonal polarisation plane.

The measurements shall be repeated with the transmitter in the idle mode.

9.4.1.4 Limits

The limits shall be those specified in subclause 4.7.2.

9.5 Radio frequency system operation

The tests in this subclause require the testing authority to provide a reference part (CPP and CFP).

9.5.1 Definitions

Ability to receive: The reference part shall transmit 10 acknowledged (numbered) packets each of 4 code words to the test receiver. The number of acknowledged (numbered) packets that are indicated as having been correctly received (as detected by acknowledgements received at the reference part) shall be counted.

If four or more are correct, then the receiver has the ability to receive. If three or fewer are correct, then the receiver does not have this ability.

9.5.2 Channel frequencies

9.5.2.1 Ability to transmit on each of the 40 channels

This test is covered by subclause 9.5.2.2.

9.5.2.2 Ability to receive on each of the 40 channels

The reference part shall set up link on the top and bottom channels and at least one in between.

9.5.2.3 Ability to receive when the carrier frequency is up to ±10 kHz from nominal

The reference part transmitter carrier frequency shall be adjusted over this range at a defined level of sensitivity (20 dB above the minimum sensitivity requirement).

9.5.2.4 Ability to receive when carrier frequency is varying at a rate of up to 1 kHz/ms

The reference part transmitter carrier frequency shall be adjusted at this rate at a defined level of sensitivity (20 dB above the minimum sensitivity requirement).

9.5.3 Dynamic RF channel allocation strategy

9.5.3.1 No channel is occupied

The channel selection is random when no channel is occupied. For both CPP and CFP by manufacturer's declaration supported by details of the implementation of "random".

9.5.3.2 One channel only below the threshold

That with one channel only below the threshold, that it is selected. For both CPP and CFP an interferer shall occupy all channels except, in turn, channels 1 and 40 which shall be subject to a field strength below the threshold value.

9.5.3.3 All channels occupied

That with all channels occupied, the channel with the lowest signal strength is selected. For both CPP and CFP this shall be as in subclause 9.5.3.2 but with the power level raised so that, in turn, channels 1 and 40 have the lowest signal strength, but above the threshold.

9.5.4 Adaptive CPP transmitter power control

For CPP transmitters only, when commanded, to switch to a low power setting in the range 16 dB ±4 dB down on the effective radiated power. As in subclause 9.3.1, but at the specified reduced power.

9.5.5 RF modulation

9.5.5.1 Peak frequency deviation: transmission

The peak frequency deviation under all possible data patterns shall be in the range 14,4 kHz to 25,2 kHz.

For a CPP, a pure tone acoustic test signal at a frequency between 1004 Hz and 1025 Hz is applied with a sound pressure level of -4,7 dBPa at the mouth reference point.

For a CFP, a pure tone electrical test signal at a frequency between 1004 Hz and 1025 Hz is applied at the CFP so as to produce a level of -10 dBm0 at the uniform PCM interface.

Confirm that for CPP and CFP the peak deviation of the transmitter is greater than 14,4 kHz and is less than 25,2 kHz.

9.5.5.2 Peak frequency deviation: reception

Ability to receive when peak deviation is anywhere in the range 14,4 kHz to 25,2 kHz. The reference part transmitter to be configured to generate peak deviation at 14,4 kHz and 25,2 kHz with the link maintained.

9.5.6 RF envelope

9.5.6.1 Transmitter output: ramp-down

Transmitter to maintain output power within 6 dB of the amplitude obtained during the transmission (figure 1) of normal data, for not less than 0,5 bit periods after the end of the last bit of normal data. By inspection of the envelope using MUX1.2, MUX1.4 (if applicable), MUX2, and MUX3 (if applicable).

9.5.6.2 Transmitter output: ramp-up

Transmitter to attain output power within 3 dB of the amplitude obtained during the transmission (figure 1) of normal data by the start of the first bit of normal data. By inspection of the envelope using MUX1.2, MUX1.4 (if applicable), MUX2, and MUX3 (if applicable).

9.5.7 Radio receiver sensitivity

9.5.7.1 Raw bit error rate

The raw bit error rate shall not exceed 1 in 1000 at a field strength which is specified in subclause 4.6.2. The following procedure is to be adopted to distinguish between those receivers with a BER performance worse than 3,2 x 10^{-3} and those with a performance better than 1 x 10^{-3}:

i) the signal strength at the receiver's antenna shall be set to the field strength specified in subclause 4.6.2;

ii) the reference part transmits, with the maximum peak frequency deviation which is specified in subclause 4.5.2, ten acknowledged packets of 4 code words to the test receiver. The number of acknowledged packets that are indicated as having been correctly received (as detected by acknowledgements received at the reference part) are counted:

   a) if 9 or 10 are correct, then accept the receiver;

   b) if 3 or fewer are correct, then reject the receiver;

   c) if 4 to 8 packets are correct, go to the next stage;

iii) 20 acknowledged packets of 4 code words are sent to the test part:

   a) if 15 or more of this 20 are correct, then accept the receiver;

   b) if 10 or fewer packets are correct, then reject the receiver;

   c) if 11 to 14 packets are correct, go to the next stage;

iv) another 20 acknowledged packets of 4 code words are sent to the test part:

   a) if 14 or more of this second 20 are correct, then accept the receiver;

   b) if 11 or fewer packets are correct, then reject the receiver;

   c) if 12 or 13 packets are correct, go to the next stage;

v) another 20 acknowledged packets of 4 code words are sent to the test part:

   a) if 13 or more are correct, then accept the receiver;

   b) if 12 or fewer packets are correct, then reject the receiver.

The above procedure results in: the probability of rejecting a CPP that gives 1 x 10^{-3} BER is 1,23%; and the probability of accepting a CPP that gives 3 x 10^{-3} BER is 1,21%

9.5.8 Radio receiver blocking performance

9.5.8.1 Ability to receive in the presence of unmodulated interfering signals

Manufacturers shall supply their characteristics for all the ranges. Tests may then be carried out as follows (using a continuously transmitting interferer):

i) select a channel for co-channel interference tests;

ii) select a channel for adjacent channel interference tests (one interferer only);

iii) select up 10 other channels at random for co-channel and adjacent channel tests.

The test will be to ensure that the link is maintained with no change of channel.

9.5.8.2 Ability to receive in the presence of asynchronous modulated interfering signals

As in subclause 9.5.8.1, but using asynchronous modulated interferers.

9.5.9 Blocking due to spurious responses

9.5.9.1 Blocking requirements

As defined in subclause 4.6.4. Additionally, manufacturers shall supply their results and frequency generation plan for the RF parts and also state the frequencies of any other supplies or sources used.

Tests on up to 10 spot frequencies may be carried out.

9.5.9.2 Intermodulation response rejection

As defined in subclause 4.6.5. Additionally, manufacturers shall supply their results and frequency generation plan for the RF parts and also state the frequencies of any other supplies or sources used.

NOTE: Subclause 4.6.5 applies to a CFP only.

Tests on up to 10 spot frequencies may be carried out.

9.6 Transmitter modulation

The equipment presented for type approval shall be set up such that, when the radio frequency link between the fixed and portable parts is established, the modulation of the transmitter is representative of normal active use.

9.7 Power supply units

The fixed and portable parts shall be operated with their appropriate power supply units which shall be submitted with the equipment at the time of type approval. Means for connecting an external power supply to portable equipment shall be provided.

9.8 Declarations by the manufacturer

When submitting an equipment for type testing, the manufacturer shall supply the following information:

i) transmitters:

   a) the oscillator frequency and carrier generation formula or, the technique of frequency generation;

ii) receivers:

   a) the oscillator frequency and local oscillator generation formula;

iii) power supply:

   a) the nominal supply voltage;

b) the type of battery where applicable;

c) the battery end-point voltage where applicable.

9.9 Labelling

The equipment shall be provided with a clear indication of the type number and description under which it is submitted for type testing.

Each type number shall be unique and in the event that the testing authority finds two manufacturers have used a similar type number, one manufacturer will be asked to change the type number.

10 Signalling system tests

The tests in this Clause require the testing authority to provide a reference part (CPP and CFP).

10.1 Multiplex alignment and timing

10.1.1 Alignment of D and B channels in MUX1

Covered by link establishment.

10.1.2 Alignment of D and SYN channels in MUX2

Covered by link establishment.

10.1.3 Transmit/listen timing of MUX3

Covered by link establishment.

10.1.4 Alignment of P, D and SYN channels in MUX3

Covered by link establishment. The CFP manufacturer shall declare his implementation to ensure that all sub-multiplexes are detectable.

10.2 Calling channel detection at the CPP

When stimulated by valid MUX2 transmissions of ID_OK, from a reference CFP, the CPP shall respond with MUX2 transmission of ID_OK, with SYNCP in the SYN channel, and the contents of the LID field reflected back.

This is covered by link establishment, additionally noting that the LID field shall be checked.

Timers: Thtx, Thrx, Thlost.

10.3 Calling channel detection at the CFP

When stimulated by valid MUX3 transmissions of LINK_REQUEST, from a reference CPP, the CFP shall respond with MUX2 transmissions of LINK_GRANT, with SYNCF in the SYN channel, and reflecting back the CPP PID together with a valid link reference value in the LID field.

This is covered by link establishment. Manufacturers shall declare their method of deriving link reference (to ensure that link references are always in the correct range).

Timers: Tftx, Tfdetect.

10.4 Link set up from CFP to CPP

10.4.1 For a CFP

10.4.1.1 To acquire a free RF channel and generate MUX2 Transmissions

To acquire a free RF channel and generate MUX2 transmissions of ID_OK. This is covered by link establishment.

Timers: Tfmax, Tfdetect.

10.4.1.2 Generation of LINK_GRANT

If a valid MUX2 response from a reference CPP is received, the CFP shall generate LINK_GRANT. This is covered by link establishment.

Timer: Tcfp.

10.4.1.3 If no valid MUX2 response is received

If no valid MUX2 response from a CPP is received, if the CFP selects a new RF channel then it shall repeat the operations in subclause 10.4.1.1 to the expiry of Tfmax with transmissions of ID_OK. This test shall confirm selection of a different channel under the above circumstances.

Timers: Tfcyc, Tfmax.

10.4.2 For a CPP

When stimulated by valid MUX2 transmissions of ID_OK, from a reference CFP, the CPP shall respond as the test in subclause 10.2.1. This is covered by subclause 10.2.1.

10.5 Link set up from CPP to CFP

10.5.1 For a CPP

10.5.1.1 To acquire a free RF channel and generate MUX3 transmissions of LINK_REQUEST

This is covered by link establishment.

Timers: Tpcyc, Tpmax.

10.5.1.2 Valid MUX2 response

If a valid MUX2 response of LINK_GRANT, from a reference CFP is received, the CPP shall synchronise to SYNCF and generate MUX2 transmissions of ID_OK, echoing back the LID value as received.

Also, if an invalid MUX2 response is received from the reference CFP, it shall be confirmed that the CPP ceases MUX3 transmissions but later resumes them on a different channel.

10.5.1.3 No valid MUX2 response

If no valid MUX2 response from a CFP is received, the CPP shall select a new RF channel and generate MUX3 transmissions of LINK_REQUEST. Repeat the test in subclause 10.5.1.1 but to expiry of Tpmax or up to 5 channels attempted.

10.5.2 For a CFP

When stimulated by valid MUX3 transmissions of LINK_REQUEST, from a reference CPP, the CFP shall respond as in the test in subclause 10.3.1.

10.6 Set up collision resolution

Channel scanning algorithm selection of start channel number shall demonstrate a random distribution. This shall be checked by manufacturer's declaration and implementation description (this applies to the CFP only).

10.7 Link re-establishment on the existing channel

10.7.1 For a CPP

10.7.1.1 Valid link re-establishment message

When stimulated by a valid link re-establishment message from a reference CFP, the CPP shall respond with MUX3 transmission of LINK_REQUEST, using the last received link reference value in the LID field.

Also, with two link re-establishments within 300 ms (for MUX1.4, MUX2) or 600 ms (for MUX1.2), it shall be confirmed that the second is either not actioned, or is not actioned until after the expiry of this time.

10.7.1.2 Valid MUX2 response

If a valid MUX2 response of LINK_GRANT, from the reference CFP is transmitted, the CPP shall respond as in the test in subclause 10.5.1.2.

10.7.1.3 No valid MUX2 response

If no valid MUX2 response from a CFP is received, the CPP shall not transmit beyond the expiry of Thlost. This shall be tested by measurement of Thlost.

If a CPP claims to offer the feature of generating link re-establishment messages, then this feature shall be demonstrated by manufacturer's declaration.

10.7.2 For a CFP

10.7.2.1 Valid link re-establishment message

When stimulated by a valid link re-establishment message from a reference CPP, the CFP shall respond by ceasing transmissions (implying that the CFP is listening for MUX3). Testing will be by generating link re-establishment messages with valid and invalid link references.

10.7.2.2 Valid MUX3 LINK REQUEST

If a valid MUX3 LINK_REQUEST from the reference CPP is received, the CFP shall respond as in the test in subclause 10.3.1.

When stimulated by valid MUX3 transmissions of LINK_REQUEST from a reference CPP, the CFP shall respond with MUX2 transmissions if the LID was correct.

10.7.2.3 No valid MUX3 LINK REQUEST

If no valid MUX3 LINK_REQUEST from a CPP is received, the CFP shall timeout on the expiry of Thlost. If a CFP claims to offer the feature of generating link re-establishment messages, then this feature is to be demonstrated by manufacturer's declaration.

10.8 Link re-establishment on a different channel

10.8.1 For a CPP

If the CPP attempts re-establishment, it shall be at least 3 s after loss of handshake. It should generate MUX3 transmissions of LINK_REQUEST, using the last received link reference value in the LID field. Re-establishment attempts shall cease within Thlost (10 s).

The reference CFP shall cease transmission of handshake. After not less than 3 s the CPP shall generate MUX3 transmissions.

10.8.2 For a CFP

No earlier than after 3 s loss of handshake, the CFP shall respond as in the tests in subclauses 10.7.2.1, 10.7.2.2 and 10.7.2.3. This shall be tested as in the above subclauses, but stimulated by loss of handshake for no less than 3 s.

10.9 Generation and reception of valid handshakes

10.9.1 Handshake intervals

By call logging over a period of 50 s, this test shall confirm the minimum and maximum handshake intervals for MUX1.2, MUX1.4, and MUX2 if a high-speed signalling capability is claimed.

10.9.2 For a CPP

10.9.2.1 Response to loss of valid handshakes

When interworking with a reference CFP which ceases transmission of valid handshakes check expiry of Thrx and transmission of ID_LOST.

10.9.2.2 Re-acquisition of valid handshakes

In the event of a valid handshake's being transmitted from the reference CFP within these times, Thlost to be reset and normal communications shall be resumed. Confirm ID_LOST is replaced by ID_OK on receipt of a valid handshake.

10.9.3 For a CFP

10.9.3.1 Response to loss of valid handshakes

When interworking with a reference CPP which ceases transmission of valid handshakes check expiry of Thrx and transmission of ID_LOST.

10.9.3.2 Re-acquisition of valid handshakes

In the event of a valid handshake's being transmitted from the reference CPP within these times, Thlost to be reset and normal communications resumed. Confirm ID_LOST is replaced by ID_OK on receipt of a valid handshake.

10.10 Layer two parameters

Following are the layer two parameters:

1) Code words to be transmitted at least once every Trate. By the use of a logging device and inspection.

2) Less than 1 in 10^7 code words to be misinterpreted at a BER of 1 in 50. By manufacturer's declaration of the implementation of the CRC algorithm.

3) Confirm general message format. Covered by link establishment.

4) Confirm general packet format. Covered by link establishment.

5) Confirm non-repetition of two identical ACWs. By manufacturer's declaration.

6) Confirm CRC calculation. Covered in 2. above.

7) Confirm code word formats (ACWs, DCWs). Covered by link establishment.

8) Confirm correct allocation of HIC/MIC. Manufacturer to declare his value of MIC and this to be confirmed by inspection of logged transmissions. The allocation of HIC is to be declared by the manufacturer.

9) Confirm operation of MUX1.2, and of MUX1.4 if claimed. Covered elsewhere.

10) Confirm handshake operation. Covered elsewhere.

11) Confirm correct allocation of LID field. Emergency access is to be tested, and also any other claimed accesses.

12) Confirm PI=0. PI=0 checking to be built in to the reference parts.

13) Confirm ability to receive layer 3 messages spread over more than one packet. Test reception by message composition in the reference part. Where the manufacturer declares an ability to transmit layer three messages over more than one packet, this feature is to be tested by manufacturer's declaration.

14) Confirm operation of endwrd and code word number/remainder encoding for reception, and where declared for transmission. Generate incorrect Endwrd/code word number remainder messages and confirm operation.

15) Confirm numbered/unnumbered message operation. Generate messages by reference part to both CPP and CFP to confirm acceptance of numbered and unnumbered messages.

16) Confirm unused octets filled with F0(hex). By inspection of logged data.

17) Confirm transmit power control operation. See subclause 9.5.4.

18) Confirm Fill-in generated when required, and of the correct format. By inspection of logged data confirm IDLE_D or FILL_IN is transmitted.

19) If the manufacturer declares an ability to support transmit or receive layer three buffer sizes of 128 bytes, this feature is to be tested by manufacturer's declaration.

10.11 Layer one and layer two timers

1) Tbid: Manufacturer's declaration

2) Tcfp (CFP processing time (18 ms, recommended 4 ms)): Timings to be declared by the manufacturer for the worst case, i.e. the processing time for the largest possible number of stored PIDs and LIDs.

3) Tcpp (CPP processing time (6,2 ms)): Timings to be declared by the manufacturer for the worst case, i.e. the processing time for the largest possible number of stored LIDs.

4) Tfcyc: Simple measurement of the CFP minimum MUX2 transmission time.

5) Tfdetect: Manufacturer's declaration.

6) Tfmax: Simple measurement of the CFP link establishment timeout.

7) Tftx: Simple measurement of transmit time after transmission of LINK_GRANT.

8) Thlost: Establish link, turn off transmitter at other end and note time to cease RF.

9) Thrx: See Handshaking, below.

10) Thtx: By inspection of the logged data stream.

11) Tpcyc: Simple measurement of the CPP MUX3 minimum transmission time.

12) Tpid: Manufacturer's declaration.

13) Tpmax: Simple measurement of the CPP call set up time with no response from the CFP.

14) Tpoll: Manufacturer's declaration.

15) Trate: By inspection of the logged data stream for each MUX.

16) Trtx: By inspection of the logged data stream for each MUX to ensure that should Trtx expire re-transmission occurs.

17) Link Re-establishments (300/600 ms): Already covered.

10.12 Acknowledged message protocol validation

Correct operation to and from both CPP and CFP to be checked by controlled insertion of errors into the reference CPP and CFP transmitted and received messages as follows:

10.12.1 CPP response to received packets

i) reference CFP transmits a packet with a single-bit error inserted in the Content (octets 3 to 6 inclusive) of the ACW. The CPP shall respond with re-transmission request or no action;

ii) reference CFP re-transmits a correct packet to simulate a lost/corrupted acknowledgement. The CPP shall respond with correct acknowledgement;

iii) reference CFP transmits correct packet. The CPP shall respond with correct acknowledgement.

10.12.2 CPP transmit actions

i) reference CFP inserts a single-bit error in the received signal from the CPP and responds with a re-transmission request. The CPP shall respond with a re-transmission;

ii) reference CFP responds to a transmission from the CPP with the correct acknowledgement. The CPP shall respond by transmitting the next packet;

iii) reference CFP makes no response to a transmission from the CPP. The CPP shall respond by re-transmitting the packet at least once on expiry of the Trtx timeout. This does not preclude the possibility of the CPPs re-transmitting a packet before the expiry of Trtx.

10.12.3 CFP response to received packets

i) reference CPP transmits a packet with a single-bit error inserted in the Content (octets 3 to 6 inclusive) of the ACW. The CFP to respond with re-transmission request or no action;

ii) reference CPP re-transmits a correct packet to simulate a lost/corrupted acknowledgement. The CFP to respond with correct acknowledgement;

iii) reference CPP transmits correct packet. The CFP to respond with correct acknowledgement.

10.12.4 CFP transmit actions

i) reference CPP inserts a single-bit error in the received signal from the CFP and responds with a re-transmission request. The CFP to respond with a re-transmission;

ii) reference CPP responds to a transmission from the CPP with the correct acknowledgement. The CFP to respond by transmitting the next packet;

iii) reference CPP makes no response to a transmission from the CPP. The CFP to respond by re-transmitting the packet at least once on expiry of the Trtx timeout. This does not preclude the possibility of the CFPs re-transmitting a packet before the expiry of Trtx.

10.13 Handshake operation

Additional to the basic requirement of exchange of handshake as defined for subclause 5.5, the following special conditions to be checked for both CFP and CPP when working with a reference CPP and a reference CFP, respectively:

i) response to not receiving the correct handshake from the reference part.

The test part shall respond by transmitting ID_LOST on the expiry of Thrx;

ii) response to receiving the correct handshake from the reference part.

The test part shall respond by transmitting ID_OK;

iii) response to receiving from the reference part a valid handshake which is not that allocated to the test part.

The test part shall respond by transmitting ID_LOST on expiry of Thrx;

iv) Thtx to be tested: (a) With handshakes from the reference part at a period of less than 1 s, confirm that no ID_LOSTs are transmitted by the test part; and (b) With handshakes from the reference part at a period over 1,040 s, ID_LOSTs are transmitted (intermixed with ID_OKs).

10.14 Layer three parameters

10.14.1 The receiving end

By manufacturer's declaration, confirm that the receiving end:

i) responds correctly to all mandatory messages appropriate to the territory in which approval is sought;

ii) responds correctly to all non-mandatory CAI messages for which operation of the feature is claimed; and

iii) ignores all other messages, without spurious operation or mis-operation of any valid feature.

10.14.2 The transmitting end

By manufacturer's declaration, confirm that the transmitting end:

i) generates correctly all the mandatory messages appropriate to the territory in which approval is sought;

ii) generates correctly all non-mandatory CAI messages for which operation of the feature is claimed; and

iii) generates no other messages.

10.15 Layer three timers

Tclr: manufacturer's declaration.

Partial release timer: manufacturer's declaration.

10.16 Declarations by the manufacturer

Where parameters, capabilities, etc., are subject to manufacturer's declaration and not to a specific test, it shall be the manufacturer's responsibility to:

i) supply the measured value of the parameter, characteristic, etc.;

ii) be prepared to submit full details as to how the item was tested, test results, and in general full documentary evidence that the testing was valid;

iii) be prepared, if necessary, to reproduce the tests on demand.

10.16.1 Information

The manufacturer shall supply the following information:

1) Details of the blocking performance of the receiver for all of the specified frequency ranges in subclauses 4.6.3, 4.6.4 and 4.6.5.

2) Details of the means by which RF channel selection is random from RF channels which are all free.

3) Details of the MUX1 B channel scrambling and de-scrambling algorithms.

4) Details of the algorithm for selection of link reference values.

5) The value of MIC used in the unit.

6) Details of emergency access methods and registration access methods.

10.16.2 Declarations

The manufacturer shall make the following declarations:

1) That the transmitter RF carrier frequency complies with subclause 4.2.

2) That the signalling strategy complies with subclause 4.3.

3) That the dynamic channel allocation strategy complies with subclause 4.4.

4) That the modulation of the transmitted RF carrier complies with subclause 4.5.2.

5) That the unit complies with subclause 4.6.1.

6) That the unit complies with subclause 4.8.

7) That the unit complies with subclause 4.9.

8) Whether or not subclause 4.10 is applicable.

9) That the absolute data rate, and the drift and jitter of the data rate lie within the limits of subclause 5.1.1.

10) That the unit has the ability to receive under all conditions of the absolute data rate, and the drift and jitter limits of subclause 5.1.1.

11) That the unit complies with subclause 5.1.3.

12) That if the unit is a CPP, the sub-multiplexes of MUX3 are all identical.

13) That if the unit is a telepoint CFP, the RF channel scanning starts at a random RF channel number.

14) That if the unit is a CPP whether it is or is not capable of generating link re-establishment messages.

15) That code word usage complies with subclause 6.1.

16) That the code word transmission sequence complies with subclause 6.3.5.

17) That the unit is or is not capable of transmitting later three messages over more than one packet.

18) That the unit does or does not support transmit and receive buffer sizes of 128 bytes.

19) That the timer Tbid lies within the required range.

20) That the timer Tcfp lies within the required range under the worst case, i.e. the processing time for the largest possible number of stored PIDs and LIDs.

21) That the timer Tcpp lies within the required range under the worst case, i.e. the processing time for the largest possible number of stored LIDs.

22) That the timer Tfdetect lies within the required range.

23) That the timer Tpid lies within the required range.

24) That the timer Tpoll lies within the required range.

25) That the timer Tclr lies within the required range.

26) That the partial release timer lies within the required range.

10.17 Additional test requirements

10.17.1 Specifics for telepoint to UKF1

Authentication is to be tested by offering a series of random challenges to the CPP and noting the response. Sufficient details of the authentication process for testing purposes are contained in a document to be made available through the United Kingdom Department of Trade and Industry (DTI).

In addition, the following shall apply:

i) manufacturers shall declare their list of features and supply details as to how registration data is to be included in CPPs;

ii) specific links are to be established for:

  a) general telepoint accesses in the following sequence:

    1) with an unregistered CPP, that the test CFP does not permit the call to proceed;

    2) with a registered CPP, that the link is established.

  The reference CFP should then generate 16 AUTH_REQs with INCZ set to 1, in each case by inspection noting that the ZAP field in AUTH_RES is incremented and that eventually it cycles round to its initial value.

  b) emergency accesses for both registered and unregistered CPPs, to ensure that in both instances the call proceeds;

iii) CPP manufacturers to declare that no additional mechanisms have been provided to modify the ZAP field other than those approved by the network operators;

iv) manufacturers to declare their CFP and CPP capabilities, and these are to be checked against the logged data stream;

v) Manufacturers to declare the compliance of their CFPs and CPPs with the mandatory layer three syntax requirements of Annex A.

10.17.2 Reserved for future use

10.18 Characteristics of the reference test set

To perform the tests described, the approvals test system requires the following capabilities:

1) A full implementation of the CAI radio and layers one to three specifications for both CFP and CPP.

2) A capability of monitoring and logging the data bit streams.

3) An ability to select a defined channel for operation.

4) An ability to vary its transmission characteristics to permit:

  - a variation of the transmit carrier frequency by up to ±10 kHz;

  - a variation of the transmit carrier frequency at a rate of 1 kHz/ms;

- full variation of the frequency stability, drift and jitter;

- a variation of the modulation index to generate minimum and maximum peak frequency deviations.

5) An ability to indicate link continuity status. Specifically when a link re-establishment takes place.

6) An ability to observe minimum and maximum frequency deviations as a receiver.

7) An ability to conduct transmitter output power measurements.

8) An ability to conduct field strength measurements.

9) An ability to observe envelope shaping.

10) Easy substitution of alternative codec types.

11) An ability to count transmitted and received code words and re-transmissions.

12) An ability to insert bit-errors into the data stream.

13) An ability to adjust the handshake transmission rate to less than once every 1,04 s.

14) An ability to confirm that PI=0.

15) The ability to control signal strength.

16) The ability to measure jitter and drift.

17) An ability to cater for reception and transmission of 128-octet layer three messages.

11 Speech and telephony tests

11.1 Measurement philosophy

The performance of the cordless handset (CPP) when operating into the common air interface is measured by means of a reference base unit (CFP).

The reference CFP shall provide the equivalent of true air interface measurements and therefore shall not contain circuitry which will modify the true air interface speech frequency performance. To meet these requirements measurements shall be referred to a uniform PCM interface. The CCITT Recommendation G.721 [6] algorithm requires such a uniform interface although it may be embedded within an IC in any particular implementation and thus not physically available.

The transcoding algorithms are specified such that encoding and decoding are symmetrical, i.e. with an encoder and decoder connected in tandem, the "levels" of the digital signals at the uniform PCM input to the encoder and output from the decoder are identical. Once the speech channel signals are in the digital domain they are essentially loss-less and hence the level at the common air interface can be related to any digital interface.

Ideally, to measure the send signals from the handset at the air interface a PCM level meter would be connected to the reference decoder uniform PCM output, and to generate receive signals for the handset at the air interface a PCM signal generator would be connected to the reference encoder uniform PCM input.

A more practical means of measuring the speech channel performance may be achieved by converting the uniform PCM to standard μ or A law PCM and then using a standard PCM test set and applying the appropriate correction factor as defined in CCITT Recommendations G.711 [15] and G.721 [6], (although this could have a deleterious effect on some parameters such as distortion).

A reference CFP is shown in figure 21 and incorporates the specified transcoder algorithm.

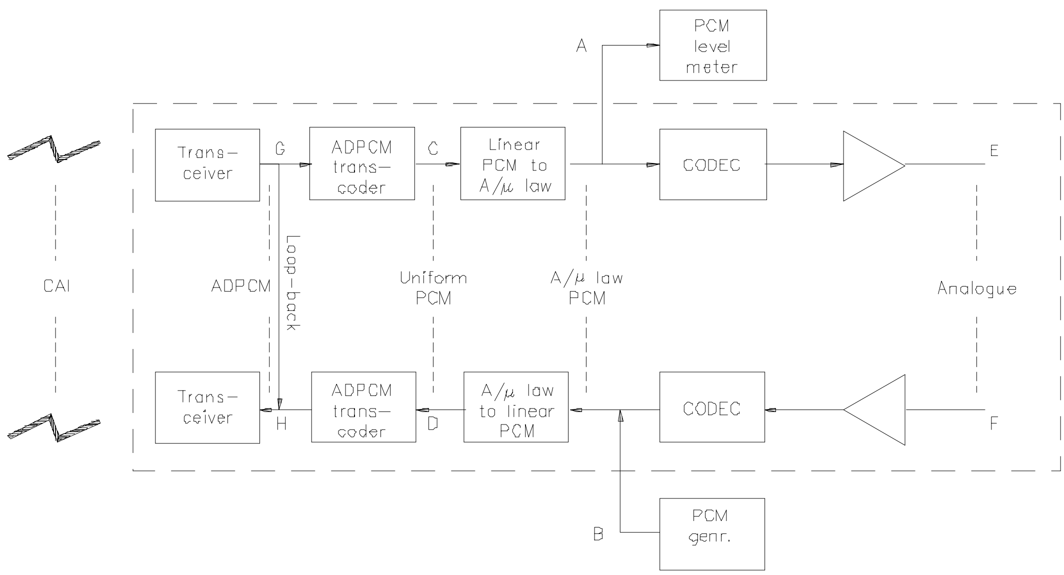


Figure 21: Reference CFP

The uniform PCM interfaces (Points C and D in figure 21) are those designated SL and SR in CCITT Recommendation G.721 [6].

11.2 Digital signal level

The digital signal levels in a transmission plan are defined in subclause 8.5.

11.3 General conditions of test

Unless otherwise stated in this standard, the tests are made under the normal operating conditions specified as follows:

- ambient temperature of +20°C ± 5°C;

- relative humidity of 45% to 75%; and

- air pressure of 86 kPa to 106 kPa.

The CPP under test shall be tested in conjunction with the reference CFP separated by some unobstructed distance, between 2 m and 10 m, in a substantially RF interference and reflection free environment, and such that the normal handshaking between CFP and CPP is maintained.

Unless otherwise stated in a particular test, where the mouthpiece of the CPP under test is fixed relative to the earpiece, the CPP is placed in the LRGP as described in CCITT Recommendation P.76, Annex A [16]. Where the mouthpiece of the CPP under test is not fixed relative to the earpiece, the front plane of the mouthpiece is mounted 15 mm in front of the lip ring and coaxial with the artificial mouth.

Unless otherwise stated in a particular test, the volume control on a CPP with a user-controlled receiving volume control shall be set to maximum.

The PCM level meter used with the reference CFP and shown in figure 21 uses a codec to convert the companded digital bit-stream to the equivalent analogue values, so that existing test equipment and procedures may be used. This codec should be a high quality codec whose characteristics are close to ideal. The codec should have characteristics such as attenuation/frequency distortion, idle channel noise, quantising distortion etc. which exceed the requirements specified in CCITT Recommendation G.714 [11] so as not to mask the corresponding parameters of the apparatus under test. The linear A/D and D/A convertors used by the codec should have at least 14-bit resolution, and the filter response should lie within the upper and lower limits given in table 14.

Table 14: Frequency response of the reference codec

Limit curve	Frequency Hz	Loss dB
Upper limit	0	0,0
	80	0,0
	80	-0,25
	3600	-0,25
	3600	0,0
	4000	0,0
Lower limit	100	+40,0
	100	+0,25
	3000	+0,25
	3000	+0,9
	3400	+0,9
	3400	+40,0

The limit curves shall be determined by straight lines joining successive co-ordinates given in the table, when loss is plotted on a linear scale against frequency on a logarithmic scale.

11.4 Sending sensitivity frequency response (subclause 8.4.1)

A pure tone signal of -4,7 dBPa is applied at the MRP as described in CCITT Recommendation P.64 [13], using an artificial mouth conforming to CCITT Recommendation P.51 [12].

A digital measuring instrument, or high quality digital decoder followed by an analogue level measuring set, is connected to point A of the reference CFP as shown in figure 21.

Measurements are made at one twelfth-octave intervals as given by the R40 series of preferred numbers in ISO 3 [18] for frequencies from 100 Hz to 4 kHz inclusive. At each frequency the level for an input sound pressure of -4,7 dBPa is measured.

11.5 Receiving sensitivity frequency response (subclause 8.4.2)

A digital signal generator is connected to point B of the reference CFP as shown in figure 21, and the level adjusted to produce a level of -16 dBm0 at the uniform PCM interface.

Measurements are made at one twelfth-octave intervals as given by the R40 series of preferred numbers in ISO 3 [18] for frequencies from 100 Hz to 4 kHz inclusive. At each frequency, the sound pressure level in the artificial ear is measured.

11.6 Handset sending loudness rating (subclause 8.6 (i))

The sending sensitivity is measured at each of the 14 frequencies given in table 2 of CCITT Recommendation P.79 [14], bands 4-17.

The sensitivity is expressed in terms of dBV/Pa and the loudness rating is calculated according to the formula 4.19b of CCITT Recommendation P.79 [14] over bands 4-17, and using the sending weighting factors from table 2, adjusted according to table 3 of the recommendation.

11.7 Handset receiving loudness rating (subclause 8.6 (ii))

The receiving sensitivity is measured at each of the 14 frequencies given in table 2 of CCITT Recommendation P.79 [14], bands 4-17.

The sensitivity is expressed in terms of dBPa/V and the loudness rating is calculated according to the formula 4.19c of CCITT Recommendation P.79 [14] over bands 4-17, and using the receiving weighting factors from table 2 of CCITT Recommendation P.79 [14], adjusted according to table 3. The artificial ear sensitivity shall be corrected using the real ear correction of table 4 of the Recommendation.

11.8 Handset sidetone masking rating (subclause 8.7.2.1)

A pure tone signal of -4,7 dBPa is applied at the MRP as described in CCITT Recommendation P.64 [13], using an artificial mouth conforming to CCITT Recommendation P.51 [12].

The reference CFP is arranged to send the appropriate control signal (via the D channel) to the CPP to enable the local sidetone path.

Measurements are made at one twelfth-octave intervals as given by the R40 series of preferred numbers in ISO 3 [18] for frequencies from 100 Hz to 4 kHz inclusive. At each frequency the level at the artificial ear for an input sound pressure of -4,7 dBPa is measured.

The sidetone path loss (Lmest) is expressed in dB and the STMR is calculated from the formula 8-4 of CCITT Recommendation P.79 [14], using the weighting factors of column 3 in table 6 (unsealed), and values of real ear correction in accordance with table 4.

11.9 Sending distortion (subclause 8.9.1)

A pure tone signal of -4,7 dBPa and a frequency in the range 1004 Hz to 1025 Hz is applied at the MRP as described in CCITT Recommendation P.64 [13], using an artificial mouth conforming to CCITT Recommendation P.51 [12].

A digital measuring instrument, or high quality digital decoder followed by an analogue level measuring set, is connected to point A of the reference CFP as shown in figure 21.

The ratio of the signal to total distortion power of the digital signal output is measured with the psophometric noise weighting (see CCITT Recommendations G.714 [11] and G.223 [10] table 4).

11.10 Receiving distortion (subclause 8.9.2)

A digital signal generator is connected to point B of the reference CFP as shown in figure 21, and the level adjusted to produce a digitally simulated sine-wave of frequency in the range 1004 Hz to 1025 Hz at a level of -10 dBm0 at the uniform PCM interface.

The ratio of signal to total distortion power of the digital signal output in the artificial ear is measured with the psophometric noise weighting (see CCITT Recommendations G.714 [11] and G.223 [10] table 4).

11.11 Sending noise (subclause 8.10.1)

The handset shall be mounted at the LRGP and the earpiece sealed to the knife-edge of the artificial ear in an acoustically quiet environment (ambient noise less than 30 dB A-weighted).

A digital measuring instrument, or high quality digital decoder followed by an analogue level measuring set, is connected to point A of the reference CFP as shown in figure 21.

The noise level at the uniform PCM interface is measured using psophometric weighting to CCITT Recommendation G.223 [10], table 4.

11.12 Sending noise (narrow band) (subclause 8.10.2)

The handset shall be mounted at the LRGP and the earpiece sealed to the knife-edge of the artificial ear in an acoustically quiet environment (ambient noise less than 30 dB A-weighted).

A high quality digital decoder followed by a selective measuring set or spectrum analyser with an effective bandwidth of 10 Hz is connected to point A of the reference CFP as shown in figure 21.

The rms voltage of the 10 Hz band limited signal is measured within the frequency range 305 Hz to 3395 Hz.

11.13 Receiving noise (subclause 8.10.3)

A digital signal generator is connected to point B of the reference CFP, and is set to provide a signal corresponding to decoder value number 1 at the uniform PCM interface (point D of figure 21).

With an ambient noise level not exceeding 30 dB A-weighted, the noise level in the artificial ear is measured.

11.14 Handset delay (subclause 8.11.1)

1) The CPP is mounted in the LRGP and the earpiece is sealed to the knife-edge of the artificial ear. A frequency response analyser is connected to the artificial ear and voice as shown in figure 22 configuration A.

2) The reference CFP is arranged to provide loop-back of the ADPCM signal as shown between points G and H of figure 21.

3) The reference CFP is arranged to send the appropriate control signal (via the D channel) to the CPP to disable the local sidetone path.

4) For each of the nominal frequencies (f0) given in table 15 in turn, the delay is derived from the measurements at the corresponding values of f1 and f2.

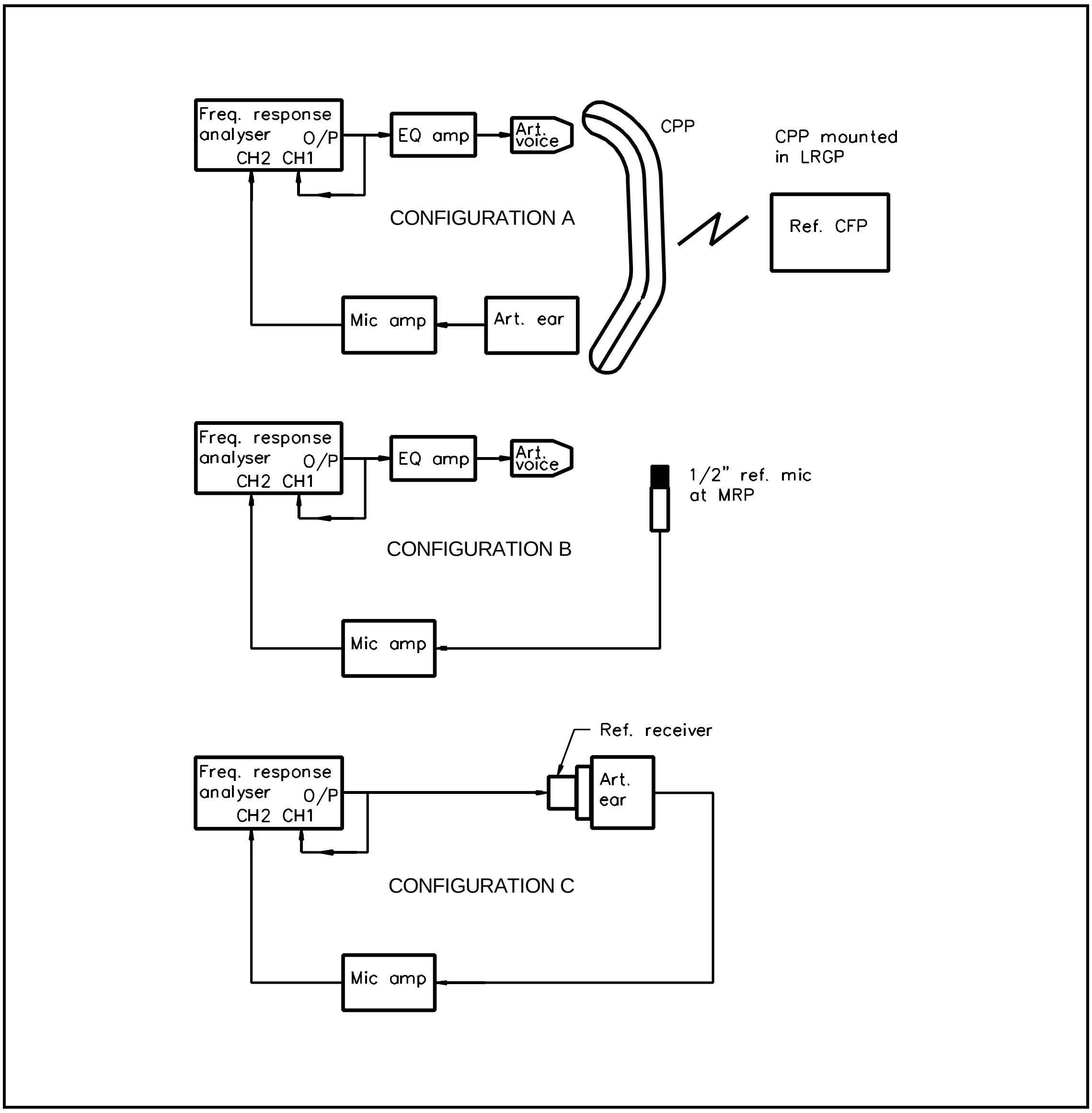


Figure 22: Handset delay test configuration

Table 15: Frequencies for delay measurement

f0 (Hz)	f1 (Hz)	f2 (Hz)
500	475	525
630	605	655
800	775	825
1000	975	1025
1250	1225	1275
1600	1575	1625
2000	1975	2025
2500	2475	2525

5) For each value of f0, the delay is evaluated as follows:

i) output the frequency f1 from the frequency response analyser;

ii) measure the phase shift in degrees between CH1 and CH2 (P1);

iii) output the frequency f2 from the frequency response analyser;

iv) measure the phase shift in degrees between CH1 and CH2 (P2);

v) compute the delay in milliseconds using the formula:

D = {1000 x (P2 - P1)}/{360 x (f2 - f1)}

vi) Average the 8 delay values to find the delay value D1.

6) Arrange a reference microphone at the MRP of the artificial voice as shown in configuration B of figure 22.

7) Repeat the tests in subclause 11.14.4 to find the average delay D2.

8) Arrange a close coupled reference receiver at the ERP of the artificial ear as shown in configuration C of figure 22.

9) Repeat the tests in subclause 11.14.4 to find the average delay D3.

10) The handset delay is calculated using the expression:

D = D1 - (D2 + D3)

The above test assumes the reference decoder to have a loop-back delay of 1,25 ms ± 1/4 bit.

In the reference CFP, the sum of two delays: (a) the delay from the received CAI bit n of the B channel to the corresponding sample in the 32 kbit/s ADPCM stream and (b) the delay to the transmitted CAI bit n of the B channel from the corresponding sample in the 32 kbit/s ADPCM stream shall be 1,25 ms for all n in the range 1-64.

This figure assumes that:

i) there is zero propagation delay; and

ii) the transmit response time of the CPP under test (as defined in subclause 6.4 is exactly 3,5 or 5,5 bit periods.

In practice, CPP transmit response times may vary by ±1/4 bit period. For any CFP whose speech input and output are connected together, the sum of the delays (a) and (b) will vary as a result.

If the CFP used in the test has a different loop-back delay, then the difference in delay from the nominal 1,25 ms shall be added or subtracted from the value of D1 as appropriate.

11.15 Weighted terminal coupling loss (subclause 8.12.1)

1) A digital signal generator is connected to point B of the reference CFP, and a PCM level meter is connected to point A as shown in figure 21.

2) The digital signal generator is set to provide a signal level of 0 dBm0 at the uniform PCM point (point D of figure 21).

3) The reference CFP is arranged to send the appropriate control signal (via the D channel) to the CPP to disable the local sidetone path.

4) The level at the uniform PCM interface (point C of figure 21) is evaluated using the level meter for one twelfth-octave intervals as given by the R40 series of preferred numbers in ISO 3 [18] for frequencies 300 Hz to 3400 Hz.

5) The echo loss is calculated according to CCITT Recommendation G.122 [8].

11.16 Stability loss - fixed geometry (subclause 8.12.2)

1) With the digital signal generator set to provide a signal level of 0 dBm0 at the uniform PCM point (point D of figure 21), the attenuation from digital input to digital output (point C of figure 21) is measured at one-twelfth octave intervals for frequencies in the range 200 Hz to 4000 Hz under the following conditions:

2) The handset shall be placed on one inside surface of three perpendicular plane smooth hard surfaces forming a corner. Each corner shall extend 0,5 m from the apex of the corner. One surface shall be marked with a diagonal line extending from the corner and a reference position marked on the line 250 mm from the corner.

   The handset shall be positioned centrally along the diagonal line with the earcap nearer to the apex of the corner such that:

   i) the mouthpiece and earcap shall face towards the surface; and

   ii) the extremity of the handset shall coincide with the normal to the reference point.

11.17 Stability loss - variable geometry (subclause 8.12.3)

With the digital signal generator set to provide a signal level of 0 dBm0 at the uniform PCM point (point D of figure 21), the attenuation from digital input to digital output is measured at one-twelfth octave intervals for frequencies in the range 200 Hz to 4000 Hz.

11.18 Out of band (sending) (subclause 8.13.1)

1) A digital level meter is connected to point A of the reference CFP as shown in figure 21.

2) A pure sine wave of level -4,7 dBPa is applied at the MRP.

3) For frequencies of 4,65 kHz, 5,0 kHz, 6,0 kHz, 6,5 kHz, 7,0 kHz, 7,5 kHz, the level of any image frequency is measured.

11.19 Out of band (receiving) (subclause 8.13.2)

A digital signal generator is connected to point B of the reference CFP, and is set to provide a signal level of 0 dBm0 at the uniform PCM interface (point D of figure 21).

For input signals at the frequencies 500 Hz, 1000 Hz, 2000 Hz, and 3150 Hz, the level of any out-of-band signals at frequencies up to 8 kHz is measured at the ERP.

11.20 Sampling frequency level (receiving) (subclause 8.14)

A digital signal generator is connected to point B of the reference CFP, and is set to provide a signal corresponding to decoder value number 1 at the uniform PCM interface (point D of figure 21).

With an ambient noise level not exceeding 30 dB A-weighted, the level of any 8 kHz signal in the artificial ear is measured.

11.21 Acoustic shock (subclause 8.15)

A digital signal generator is connected to point B of the reference CFP as shown in figure 21, and the level adjusted to produce a level of +3,14 dBm0 (Tmax) at the uniform PCM interface.

Measurements are made at one-third octave intervals as given by the R10 series of preferred numbers in ISO 3 [18] for frequencies from 200 Hz to 4 kHz inclusive. At each frequency, the sound pressure level in the artificial ear is measured.

11.22 Listener sidetone (subclause 8.7.2.2)

The sound field is calibrated in the absence of any local obstacles. The averaged field shall be uniform to within +4 dB/-2 dB within a radius of 0,15 m of the MRP, when measured in one-third octave bands from 100 Hz to 8 kHz (bands 1 to 20).

A calibrated half-inch microphone is mounted at MRP. The sound field is measured in one-third octave bands. The spectrum shall be "Hoth" CCITT Volume V, Supplement 13 [19] to within ±1 dB and the level shall be adjusted to 50 dB A-weighted (-44 dBPa A-weighted). The tolerance on this level is ±1 dB.

NOTE: Where adaptive techniques or voice switching circuits are not used, it is recommended to increase the sound level to 60 dB A-weighted (-34 dBPa A-weighted) to help measurement accuracy.

The artificial mouth and ear are placed in the correct position relative to MRP, and the handset is mounted at LRGP and the earpiece is sealed to the knife-edge of the artificial ear.

Measurements are made in one-third octave bands for the 14 bands centred at 200 Hz to 4000 Hz (bands 4-17). For each band the sound pressure in the artificial ear is measured by connecting a suitable measuring set to the artificial ear.

The listener sidetone path loss is expressed in dB and the LSTR is calculated from the formula 8-4 of CCITT Recommendation P.79 [14], using the weighting factors in column (3) in table 6/P.79, and the values of L_E; in accordance with table 4/P.79.

11.23 Sidetone distortion (subclause 8.9.3)

The handset is mounted at LRGP and the earpiece is sealed to the knife-edge of the artificial ear. An instrument capable of measuring the third harmonic distortion of signals with fundamental frequencies in the range of 315 Hz to 1000 Hz is connected to the artificial ear.

A pure-tone signal of -4,7 dBPa is applied at the MRP at frequencies of 315 Hz, 500 Hz and 1000 Hz. For each frequency, the third harmonic distortion is measured in the artificial ear.

11.24 Subjective speech quality

11.24.1 Overall requirements

The CTA shall give acceptable speech performance.

This requirement shall be deemed to be fulfilled when either the conditions of subclause 11.24.2 or those of subclauses 11.24.3 and 11.24.4 are met, the criteria of subclause 11.24.5 being applied in each case to the results obtained by following the appropriate option (A or B respectively) in the test described in Annex J.

In either case, the test as described in Annex J may be carried out with a wire connection instead of the normal radio connection from the encoder outputs to the corresponding decoder inputs (notwithstanding the fact that in such a case the apparatus tested cannot strictly be called cordless), provided that the radio connection which replaces the wire connection, when tested in accordance with the general conditions specified in 11.3, does not introduce any effective degradation of the digital signal. Compliance with this condition shall be by a supplier's declaration.

11.24.2 CTA tested as an entirety

When the CTA is to be treated as an entirety it shall meet the requirements of a), b), c), d) and e).

a) The standard of performance specified in 11.24.5 shall be attained for the entire CTA.

b) The frequency response characteristics shall be such that the spectrum of the speech originating from the microphone of the CPP, when it reaches the input to the sending path encoder, is the same as the expected average spectrum of the speech coming from the network when it reaches the input to the receiving path encoder; given that it may be assumed for this purpose that speech arriving directly from the network has in general had imposed upon it a frequency-response characteristic between the limits shown in figure 17.

c) The gains or losses before the two encoders shall be such that, if a 1000 Hz sine wave at -5 dBPa at the MRP produces an electrical level x dBV at the sending encoder input, and a 1000 Hz sine wave at -20 dBV applied at the NTTA produces an electrical level y dBV at the receiving encoder input, then x and y shall not differ by more than 3 dB.

d) The same process of analogue-to-digital conversion, encoding, decoding and digital-to-analogue conversion shall be used in both directions.

e) Any adaptive process shall operate identically in both directions.

Compliance with the requirements of (a) shall be checked by the test described in Annex J, option A. Compliance with the requirements of (b), (c), (d) and (e) shall be by manufacturer's declaration.

11.24.3 Codec to be treated in isolation

Where the codec is to be treated in isolation, the standard of performance specified in subclause 11.24.5 shall be attained for the codec pair.

Compliance shall be checked by the test described in Annex J option B.

For the purposes of this Clause, the supplier shall designate an active speech level, measured in dBV, in accordance with the method described in Annex J.1.4, to be known as the nominal optimum level. This level shall then be used as the central value of the three input levels applied to the codec in the test described in Annex J.

NOTE 1: The term "codec" for the purposes of this subclause is taken to mean the complete apparatus as tested in accordance with option B of Annex J, in the same operating conditions as those in which it was tested. For clarity, this means that the coder-decoder pair together with any associated signal conditioning circuits (to effect gain, loss, impedance changing, anti-alias filtering, frequency-response adjustments, and similar modifications), and any non-linear processors (such as voice switching devices or companders), are all in the same state as they were during the said test.

NOTE 2: The codec input and output have to be actual electrically accessible points for the test described in Annex J, but may be purely notional points in the complete CTA. That is, the above requirements do not preclude the combining together of circuit elements that were partly within and partly outside the codec in the sense defined above, provided that the overall sensitivity-frequency responses of the analogue paths (up to the analogue-to-digital interfaces and from the digital-to-analogue interfaces) are not altered.

11.24.4 CTA incorporating codec complying with 11.24.3

Where a CTA incorporates a codec in accordance with 11.24.3, and is not to be treated as an entirety, it shall meet the requirements of (a), (b) (c) and (d).

a) No non-linear processing shall be incorporated in the CTA except:

   i) non-linear signal processes normally accepted in simple telephones;

   ii) non-linear processes forming part of the apparatus complying with the other requirements of this subclause.

b) Any CTA in which the codec is eventually incorporated shall be designed to have the nominal sensitivity-frequency responses of its component parts within the following limits:

   i) sending side of the CPP (from MRP up to the input of the codec);

   between the lower limit and the upper limit of figure 17, but with an additional loss or gain, the same at all frequencies, sufficient to ensure that a 1000 Hz sinusoidal input at a level of - 5 dBPa at the MRP results in an electrical input to the codec within the range: nominal optimum level ±1 dB;

ii) receiving side of the CFP (from the NTTA to the input of the codec);

a gain or loss, the same at all frequencies, sufficient to ensure that a 1000 Hz sinusoidal signal at a level of -20 dBV applied at the NTTA results in an input to the codec in the range: nominal optimum level ±1 dB.

c) The codec used for the CTA sending direction and the codec used for the CTA receiving direction shall both be identical in function and performance with the one approved as a result of undergoing the test described in Annex J option B.

d) When a codec is replaced by a purely analogue having the same sensitivity-frequency response, the resulting system shall meet the appropriate national requirements. Compliance shall be by supplier's declaration.

11.24.5 Standard of performance

The following criteria shall be used when assessing the values of Q_r that are tabulated as a result of the tests described in Annex J.

a) For the combinations of input and output levels marked with an x in table 16, there shall be not less than 6 codes in category A, not more than 1 code in category C and none in category D.

b) For the remaining combinations of input and output levels, there shall be not more than a total of 7 codes in categories C and D combined, of which not more than 3 codes are in category D.

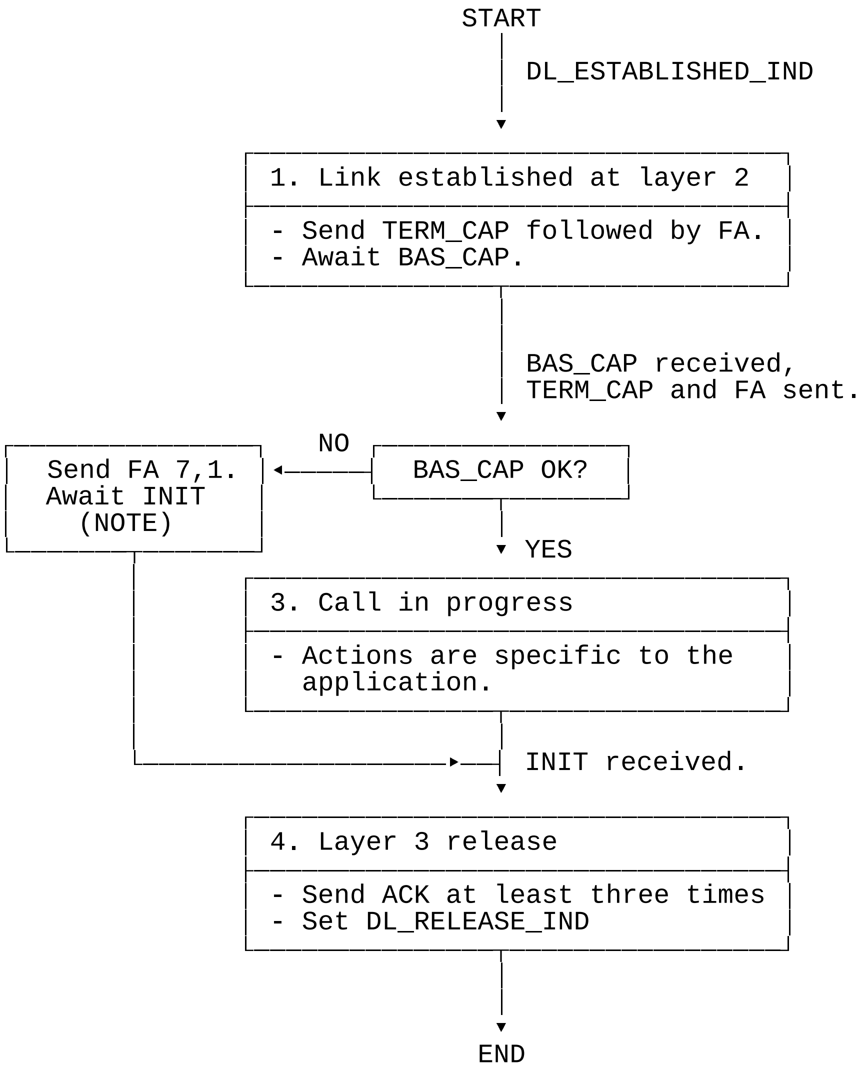
Table 16: Classification of input-output level combinations

Condition	Male Speech				Female Speech			
	Listening Level, dBv				Listening Level, dBv			
	-12	-22	-32	-42	-12	-22	-32	-42
1 High input level		x	x			x	x	
2 Medium input level		x	x			x	x	
3 Low input level		x	x			x	x	
4 Medium input level + noise								

Annex A (normative)

Layer three mandatory syntax diagrams

A.1 CPP mandatory layer three initialisation syntax



NOTE: A one second timeout applies if acknowledgement is not received.

Figure A.1: CPP mandatory layer three initialisation syntax

A.2 CFP mandatory layer three initialisation syntax

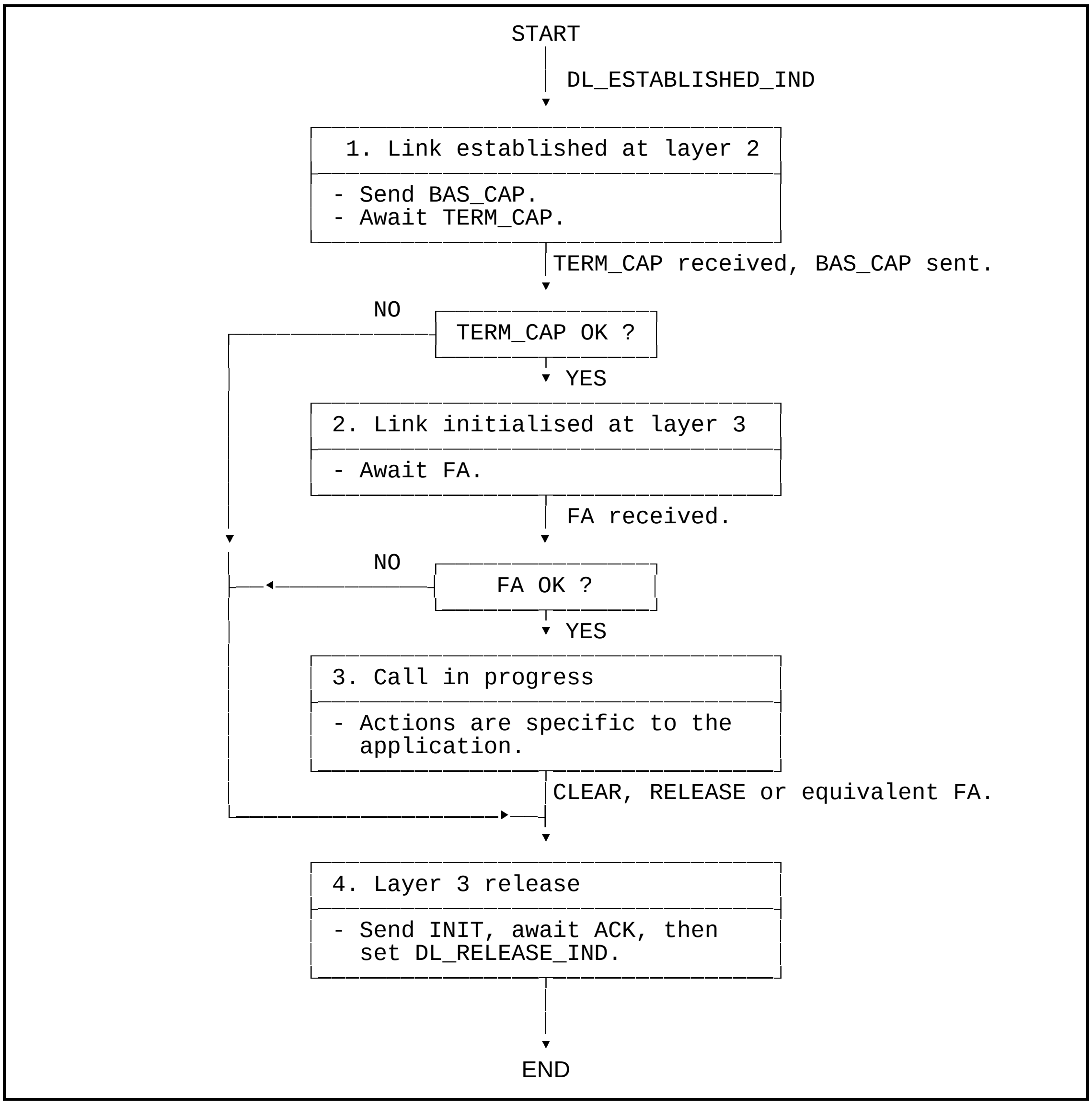


Figure A.2: CFP mandatory layer three initialisation syntax

A.3 Telepoint CPP mandatory layer three syntax

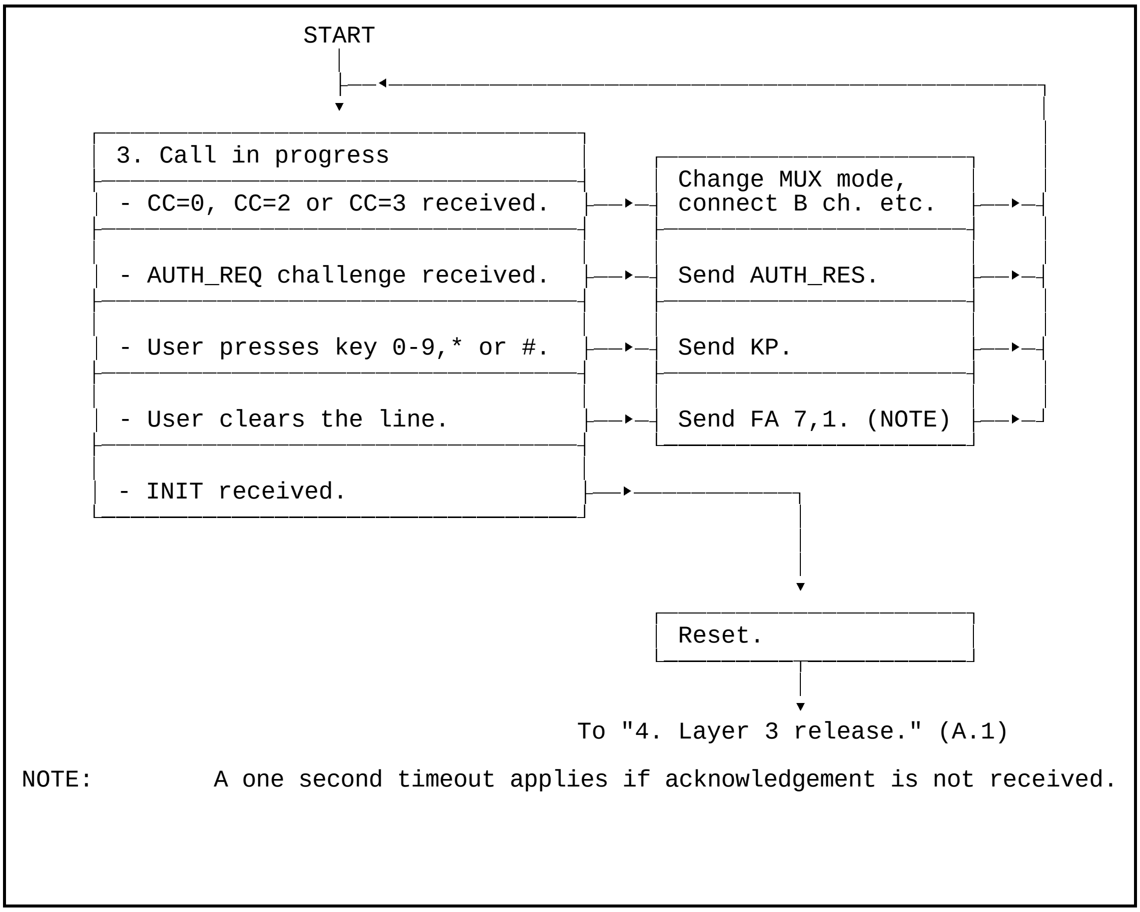


Figure A.3: Telepoint CPP mandatory layer three syntax

A.4 Telepoint CFP mandatory layer three syntax

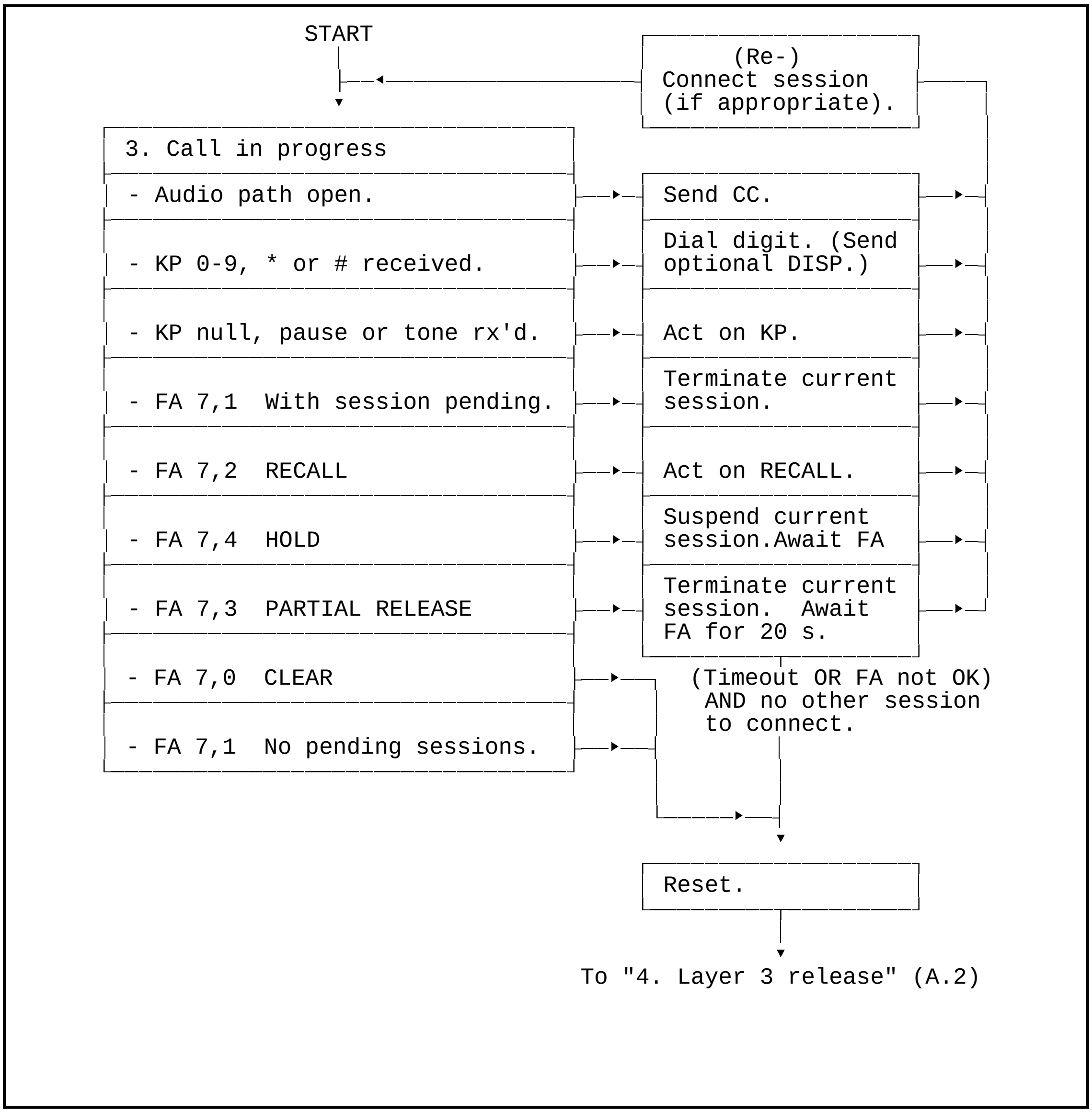


Figure A.4: Telepoint CFP mandatory layer three syntax

Annex B (normative):

Telepoint operation

B.1 Minimum telepoint handset configuration

Table B.1: Minimum telepoint handset configuration

Information Element	Direction CPP CFP	CPP transmits	CPP receives & processes
KP	-------->	0-9,*,#	
DISP	<--------		- (NOTE)
SIG	<--------		-
FA	-------->	Class 3 (telepoint) value 00000 Class 4 (emergency) value 00000 Class 7 (auxiliary) 00001 (full release)	
FI	<--------		-
CC	<--------		00000000 (release B) 00000010 (con B, no ls) 00000011 (con B, ls)
INIT	<--------		All
BAS_CAP	<--------		All
CHAR	--------> <--------	-	 -
OARAC	<--------		-
PAR_SET	--------> <--------	-	 -
PAR_REQ	--------> <--------	-	 -
PAR_RES	--------> <--------	-	 -
NO_POLL	<--------		-
AUTH_REQ	<--------		All
AUTH_RES	-------->	All	
AUTH2_REQ	<--------		-
AUTH2_RES	-------->	-	
TERM_CAP	-------->	All	

NOTE: "-" means non-mandatory

B.2 Minimum telepoint base station configuration

Table B.2: Minimum telepoint base station configuration

Information Element	Direction CPP CFP	CFP transmits	CFP receives & processes	
KP	-------->	- (1)	0-9,*,#,NUL,pause,go to MF	
DISP	<--------	-		
SIG	<--------	-		
FA	-------->		Class 3 (telepoint) values 00000-11111 Class 4 (emergency) values 00000-11111 Class 7 (auxiliary) 00000 (clear) 00001 (full release) 00010 (recall) 00011 (partial release) 00100 (hold)	
FI	<--------	-		
CC	<--------	00000010 (con B, - ls) or 00000011 (con B, + ls) (2)		
INIT	<--------	All		
BAS_CAP	<--------	All		
CHAR	--------> <--------	 -	- 	
OARAC	<--------	-	-	
PAR_SET	--------> <--------	 -	- 	
PAR_REQ	--------> <--------	 -	- 	
PAR_RES	--------> <--------	 -	- 	
NO_POLL	<--------		-	
		transmits	receives (3)	processes (3)
AUTH_REQ	<--------	All		
AUTH_RES	-------->		All	-
AUTH2_REQ	<--------	- (4)		
AUTH2_RES	-------->		- (4)	-
TERM_CAP	-------->		All	-

NOTE 1: "-" means non-mandatory

NOTE 2: Although not mandatory, it is recommended that CC value 00000000 (use MUX1, no B) be sent before sending either CC value 00000010 (use MUX1, con B, no LS), or CC value 00000011 (use MUX1, con B, with LS). This avoids an audible switch from MUX2 to MUX1.

NOTE 3: Some Information elements may be processed, if required, outside a public access (telepoint) CFP.

NOTE 4: Informative Annex E contains recommendations concerning the possible future use of AUTH2_REQ and AUTH2_RES.

B.3 Telepoint authentication

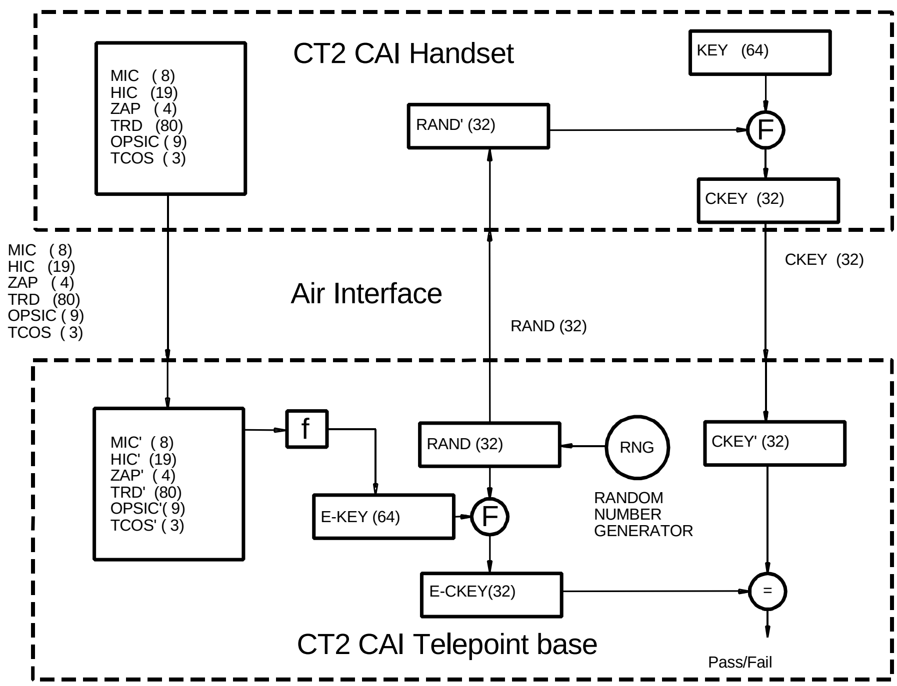
B.3.1 Introduction

This annex specifies the mechanisms of basic authentication to be used in CT2 CAI handsets, offering UKF1 authentication facilities. This mechanism is also designed for use worldwide as a system of authentication for roaming customers.

This annex does not seek to justify the need for security, or to establish a required level of security.

Authentication as described below is one part of the overall user validation processes which will be undertaken by the telepoint operators; the remainder are outside the scope of this specification and have no implications for roaming product (i.e. handset) design.

B.3.2 Basis of operation



NOTE 1: The relationship shown **f** is proprietary to individual telepoint operators.

NOTE 2: TRD is shown at it's maximum length.

Figure B.1: Telepoint authentication scheme UKF1The basic mechanism of authentication is described below and is also shown in figure B.1.

B.3.2.1 Identification information

A registration slot in a handset (CPP) contains identificationinformation which is transmitted to the telepoint base station (CFP) during the set up and authentication phases of a telepoint call. This information is sufficient to identify uniquely both the handset making the call and also the telepoint account to which the call should be charged.

The information comprises the following fields:

- MIC 8 bits Manufacturer identity code;

- HIC 19 bits Handset identity code;

-	OPSIC	9	bits	Home service identity;
-	TCOS	3	bits	Telepoint class of service;
-	ZAP	4	bits	ZAP field;
-	TRD	80	bits	Telepoint registration data.

The HIC and MIC fields (see subclauses 6.4.3 and 6.4.4) are sufficient to identify uniquely the calling handset but not necessarily the telepoint account). They are transmitted to the base in signalling layer two fixed-format address code words (ACWs) during link initiation and subsequently during handshaking.
OPSIC identifies the users "home" service; i.e. the service with which the user has registered. This main use for this is during roaming when it identifies to the operator accepting the call the service through which the call should be charged. OPSIC codes will be allocated by the Standard Control Authority (SCA).

TCOS gives a telepoint class of service indication; i.e. the level of service that should be made available to the user. The values used will be agreed (for roaming service) and published by the SCA. Individual operators may wish to supplement this field (for example using TRD) for their "home" subscribers.

The TRD field provides additional information as necessary to ensure that the telepoint account is uniquely identified. For roaming use the format of the data will be standardised (and published by the SCA), but for "home" use the format of the data will be proprietary to individual operators.

The ZAP field, which is intended to allow a telepoint operator to temporarily or permanently bar a handset registration from that specific service, is defined in subclause B.3.3.

OPSIC, TCOS, TRD and ZAP are transmitted from the handset to the base in layer three information element AUTH_RES (see subclause 7.2.9).

It shall not be possible for the user to display, or otherwise obtain from the handset the values in any of the above fields, with the exception of OPSIC. The value of the LID field may also be made available to the user.

B.3.2.2 KEY number

The handset also stores internally a KEY number which is transmitted to the telepoint base station during the set up and authentication phases of a telepoint call. Authentication is achieved in the telepoint system by comparing the KEY received from the handset with the correct KEY for that handset.

To avoid problems of fraud (arising from the monitoring of the air-interface and the cloning of valid handsets) the KEY will be ciphered (encrypted) before transmission over the air-interface.

The process by which the content of the KEY field is interrogated by the base and the handset authenticated is:

i) the base transmits to the handset a 32-bit random challenge (RAND) in the layer three information element AUTH_REQ where it is received as RAND';

ii) The handset encrypts the 64-bit KEY using an encryption function "F", and using RAND' as the key, to produce the 32-bit ciphered-KEY (CKEY);

iii) The handset then transmits CKEY to the base in the layer three information element AUTH_RES where it is received as CKEY';

iv) The base determines the expected-KEY (E-KEY) for the handset using the identification information (see subclause 7) and using the same function "F", with RAND as the key, calculates the expected value of CKEY (E-CKEY);

v) The base compares the received CKEY (CKEY') with the expected value (E-CKEY). If the values match the handset is judged to be valid.

B.3.2.3 Function "F"

The function "F" used in the cipher process described above is common to all telepoint systems using UKF1, and shall be incorporated into all telepoint CPPs.

B.3.2.4 Assignment of KEY numbers

The sections above have assumed that the telepoint base station, given the identity of the handset and account, will "know" the value of the KEY expected from the handset (E-KEY). The means by which this is achieved is proprietary to individual telepoint operators.

B.3.3 ZAP facility

Each handset will have a 4-bit ZAP field associated with the telepoint registration which is intended to allow the telepoint operator to temporarily or permanently bar that handset registration from access to the service.

The contents of the ZAP field are sent to the telepoint base station during the authentication phase of a telepoint call in the layer three information element AUTH_RES. The contents of the ZAP field can only ever be changed by one mechanism - this is that the telepoint base station can instruct the handset to increment the contents of the field by sending a layer three AUTH_REQ information element with the INCZ field set (see subclause 7.2.8). The field is incremented modulo-16 such that the binary value 1111 is incremented to 0000.

The use of the ZAP field in achieving user validation is proprietary to individual telepoint operators, and in some cases the field may not be used. The ZAP facility shall however be provided by all CAI-compatible handsets.

B.3.4 Registration data fields

The data used in the authentication process described above must be programmed (e.g. by the user) into the handset. The exceptions to this are HIC and MIC which are programmed into the handset at manufacture, and ZAP which cannot be programmed.

In addition the handset must be programmed with a link identification code (LID); this is the value used in the LID field when the handset establishes a call to the "home" telepoint service (see subclause 6.4.5). It therefore allows the handset to target one specific service. The LID value for a given service will be allocated to the telepoint operator by the Standard Control Authority body.

The data to be programmed is supplied, at registration, by the telepoint operator. The data is:

- LID 16 bits link identification code;
- KEY 64 bits KEY number assigned to the handset;
- OPSIC 9 bits home service identification code;
- TCOS 3 bits telepoint class of service;
- TRD 80 bits telepoint registration data.

The LID, KEY, OPSIC and TCOS fields are mandatory and fixed-length but the TRD field may be used or omitted at the discretion of the telepoint operator. The TRD field is optional and is variable-length up to a maximum of 20 BCD digits. The number of digits used is determined by the individual telepoint operator.

B.3.5 Entry of registration data

It is essential for telepoint operators that the method by which registration data is programmed into a handset is basically identical for all handset types.

The required entry mechanism is described below. This requires that every handset has keys 0-9,'*' and '#'.

B.3.5.1 Basic data entry

For coding simplicity fields which are specified in straight binary will be input as series of octal digits (0-7). Fields which are specified in BCD digits will be input as series of decimal digits (0-9). Thus for the information to be input as part of the registration:

- LID 16 bits denoted ss....ss input in octal
- KEY 64 bits denoted pp....pp input in octal
- OPSIC 9 bits denoted oo....oo input in octal
- TCOS 3 bits denoted ccc input in octal
- TRD 80 bits denoted tt....tt input in decimal

Data will be input in blocks of eight characters where each character represents data bits as:

	Ch1	Ch2	Ch3	Ch4	Ch5	Ch6	Ch7	Ch8
Block 0:	sss	sss	sss	sss	sss	spp	ppp	ppp
Block 1:	ppp	ppp	ppp	ppp	ppp	ppp	ppp	ppp
Block 2:	ppp	ppp	ppp	ppp	ppp	ppp	ppp	ppp
Block 3:	ppp	ppp	ppo	ooo	ooo	ooc	cc?	???
Block 4:	tttt	tttt	tttt	tttt	tttt	tttt	tttt	tttt
Block 5:	tttt	tttt	tttt	tttt	tttt	tttt	tttt	tttt
Block 6:	tttt	tttt	tttt	tttt				

NOTE 1: Characters are entered left to right (on the above table), blocks are entered in ascending numerical order.

NOTE 2: Bits are entered as defined in subclause 6.3.1; thus the least significant bit of each field is input first (and is shown on the top-left hand side of the field in the above table) and the most significant bit of the field is input last (and is shown on the bottom-right hand side of the field in the table).

NOTE 3: Bits shown "?" are used for padding only and are not allocated; the bits entered can take any value and are ignored.

NOTE 4: The value of the digit entered to represent the 3- or 4-bit group is calculated where the left-hand bit of the group as shown in the table (i.e. the bit entered first) is the least significant bit and has a binary weighting of 1.

NOTE 5: The TRD field is variable length and is shown in the above table at its maximum length of 20 BCD characters; where TRD is either omitted or used at less than its maximum length then fewer blocks may need to be input.

NOTE 6: Where the final block consists of less than 8 digits then the final data digit shall be followed by a '*' character to terminate entry and then the block should be padded as necessary with random digits (0-9) to fill the block; where the final block is of exactly 8 digits (i.e. where TRD is of length 0, 8 or 16 digits) then no '*' character will be used.

B.3.5.2 Check digits

Each block of eight characters (0-9 or '*') will be followed by a ninth character (0-9 or '*') as a check character. This character is calculated as the modulo-11 sum of the block number (0-6) plus the value of each character entered multiplied by its position in the block.

NOTE 1: The first character entered in a block is at position 1, the final character is a position 8.

NOTE 2: When the modulo-11 sum has value 10 (decimal) then the '*' character is used as the check character.

NOTE 3: In calculation of the check character the '*' terminating character has value 10 (decimal) and the random padding digits are included.

B.3.5.3 Termination of data entry

After the final data block a terminating block is required. This takes the form of 4 characters; a 2-digit modulo-100 sum (00-99) of all the previously entered data digits followed by a single modulo-11 check character of these two digits (calculated as in subclause B.3.5.2) and a '#' terminating character.

Note that the '*' terminating character (if present), the random padding digits and the check character in each block are ignored for the purposes of the calculation of the modulo-100 sum.

B.3.5.4 Example

By way of example consider the case when the data to be input takes the values shown below. In this example TRD is used with a length of 4 digits.

LID	(msb)	0000	0000	1000	1000			(lsb)
KEY	(msb)	0000	0001	0010	0011	0100	0101	
		0110	0111	1000	1001	1010	1011	
		1100	1101	1110	1111			(lsb)
OPSIC	(msb)	1	0000	1111				(lsb)
TCOS	(msb)	000						(lsb)
TRD	(msb)	1001	0111	0101	0011			(lsb)

This data would be coded as:

	Ch1	Ch2	Ch3	Ch4	Ch5	Ch6	Ch7	Ch8
Block 0:	000	100	010	000	000	011	110	111
Block 1:	101	100	111	101	010	110	010	001
Block 2:	111	001	101	010	001	011	000	100
Block 3:	100	000	001	111	000	010	00?	???
Block 4:	1100 1010	1110 1001						

The 4 bits marked "?" in the above table will be filled with random data. If, for example, they are filled with binary "1s" the data becomes:

	Ch1	Ch2	Ch3	Ch4	Ch5	Ch6	Ch7	Ch8
Block 0:	000	100	010	000	000	011	110	111
Block 1:	101	100	111	101	010	110	010	001
Block 2:	111	001	101	010	001	011	000	100
Block 3:	100	000	001	111	000	010	001	111
Block 4:	1100 1010	1110 1001						

This data would then be input with the following strings of digits:

Block 0:	012006370
Block 1:	517523242
Block 2:	745246015
Block 3:	104702478
Block 4:	3579*3451
Block 5:	268#

Note that block 4 contains the terminating character '*' and three random padding digits ('345' in this instance).

B.3.6 Man-machine interface (MMI) for registration

The procedure for entry of telepoint registration data should be common to all handset types. The required procedure is:

i) the user selects the correct mode for entry of registration information and, where more than one registration "slot" is provided, the correct "slot" identity. This procedure may vary between different handset types. Handset emits short (confidence) "beep" and clears display if present;

ii) the user enters the 9 characters (0-9,'*') comprising the first/next block of data. Characters echoed to display if present. Normal use may be made of <Backspace> or other editing keys;

iii) after the final character of the block has been input the handset checks for a digit error in the block by performing the modulo-11 calculation. If no error is discovered the handset emits a short (confidence) beep, clears the display if present and continues from ii with the next data block;

iv) if an error is encountered in the modulo-11 check, the handset emits a warning/error beep and requires the user to re-enter the digits of the block from step i;

v) when the terminating '#' is detected the handset checks that it is correctly positioned and that the modulo-11 check for that line is correct. If an error is detected the handset emits a warning/error beep and required that the digits of that line be re-entered;

vi) if the modulo-11 character is correct then the handset performs the overall modulo-100 check. If an error is discovered the handset will abort the registration procedure;

vii) if no error is found the handset emits a short (confidence) beep and completes the registration procedure by writing the registration data to semi-permanent storage;

viii) if any other key is pressed (or event occurs) which would normally terminate such a procedure, or if the handset is switched off, the registration procedure should be aborted and all data entered should be discarded;

ix) any other key depression should be ignored (may emit a warning/error beep).

Note that if a registration procedure is terminated early or aborted because of an entry error then the data entered should be discarded. At the very least, steps should be taken to ensure that the "incorrect" data can never be used in a call set up request, and, ideally, the previous "valid" data in the "slot" should remain intact.

B.3.7 Handset registration capacity

The registration information described above is specific to one telepoint registration. Each registration therefore requires storage as follows:

- LID 16 bits
- KEY 64 bits
- OPSIC 9 bits
- TCOS 3 bits
- TRD 80 bits
- ZAP 4 bits

---
176 bits

This means that a registration "slot" will require 22 bytes of non-volatile storage. As a minimum, a CAI-compatible CT2 handset shall have two independent registration "slots".

Annex C (normative):

Serial number format

The standard serial number format for CT2 CAI handsets shall be as follows:

- MMM mmm HHHHHH xx...xx

where:

MMM	is the decimal representation of the MIC, in the range 000 to 255 (2^8 - 1);
mmm	is the decimal representation of the MODEL number in the handset (and is the decimal equivalent of the MODEL field in TERM_CAP), and lies in the range 000 to 255 (2^8 - 1);
HHHHHH	is the decimal representation of the HIC, and lies in the range 000000 to 524287 (2^{19} - 1); and
xx...xx	is free-format, manufacturer-specific data, and thus may be alphanumeric.

The serial number in this format shall be accessible to the user of a CPP (e.g. on a display and accessed by a specified key sequence, or on a printed label in the battery compartment).

Annex D (normative):

Accuracy of measurement

D.1 Radio frequency parametric and system tests

The overall accuracy of measurement shall be as follows:

Table D.1: Accuracy of RF paramter measurements

Item	Accuracy
DC voltage	±3 %
AC mains voltage	±3 %
AC mains frequency	±0,5 %
Radio frequency	±50 Hz
Radio frequency voltage	±2 dB
Radio frequency field strength	±3 dB
Radio frequency carrier power	±10 %
Adjacent channel power	±3 dB
Impedance of artificial loads, combining units, cables, plugs, attenuators etc.	±5 %
Source impedance of generators and input impedance of measuring receivers	±10 %
Attenuation of attenuators	±0,5 dB
Temperature	±1 °C
Humidity	±5 %

D.2 Signalling system tests

The overall accuracy of measurement shall be as follows:

Table D.2: Accuracy of signalling system measurements

Item	Accuracy
Time	±5 %

D.3 Speech and telephony tests

The overall accuracy of measurement shall be as follows:

Table D.3: Accuracy of speech and telephony measurements

Item	Accuracy
Electrical signal power (for signals ≥ -50 dBm)	± 0,2 dB
Electrical signal power (for signals < -50 dBm)	± 0,4 dB
Sound Pressure	± 0,6 dB
Time	± 5 %
Frequency	± 2 % *

NOTE: * When measuring sampled systems, it is advisable to avoid measuring at sub-multiples of the sampling frequency. There is a tolerance of ±2% on the frequencies, which may be used to avoid this problem, except for 4 kHz where only the -2% tolerance may be used.

Annex E (informative)

Interim arrangements

E.1 Minimum RF power

Notwithstanding the provisions of subclause 4.5.1.2, the ETSI Radio Equipment and Systems Technical Committee recommends that, for an interim period ending on 31st December 1992, national authorities accept the use of CFPs and CPPs meeting a reduced specification for minimum RF output power (as defined in subclause 9.3.1), provided that the following is met:

- at nominal design operating voltage the normal carrier output power or effective radiated power under normal test conditions shall not be less than 1 mW.

E.2 Radio receiver sensitivity

Notwithstanding the provisions of subclause 4.6.2, the ETSI Radio Equipment and Systems Technical Committee recommends that, for an interim period ending on 31st December 1992, national authorities accept the use of CFPs and CPPs meeting a reduced specification for radio receiver sensitivity (as defined in subclause 9.5.7), provided that the following is met:

- the radio receiver sensitivity shall be at least 45 dBμV/m. It is recommended that this be achieved by ensuring that the radio receiver sensitivity is typically 40 dBμV/m or better. The receiver sensitivity (see subclause 9.5.7) shall be defined at a bit error ratio of 1 in 1000 or better in both the B (speech data) and D (signalling data) channels (see subclause 5.2).

E.3 Portable part ADPCM voice codec

Notwithstanding the provisions of subclause 8.1.4, the ETSI Radio Equipment and Systems Technical Committee recommends that, for an interim period ending on 31st March 1993, national authorities accept the use of a reduced specification codec in the portable part, provided that it meets the following requirement:

- any design variations to enable compliance with alternative relevant national standards shall be confined to the CFP in order to allow for maximum roaming. When working to a full G.721 codec in the fixed part (as defined in subclause 8.1.4), the codec in the portable part may be of reduced specification, but shall interwork with a full CCITT Recommendation G.721 [6] codec to meet the objective and subjective tests contained in subclause 11.24.

E.4 Weighted terminal coupling loss

Notwithstanding the provisions of subclause 8.12.1, the ETSI Radio Equipment and Systems Technical Committee recommends that, for an interim period ending on 31st March 1993, national authorities accept portable parts with a reduced TCLw (weighted terminal coupling loss), provided that it meets the following requirement:

- with the earpiece sealed to an artificial ear, the weighted terminal coupling loss (TCLw) measured from the digital input to the digital output shall be at least 25 dB. Compliance shall be checked by the test of subclause 11.15.

E.5 Alternative authentication algorithms

Implementors of public access (telepoint) CFPs should be aware that discussions are taking place between telepoint service operators that may lead to a change of status from non-mandatory to mandatory for AUTH2_REQ and AUTH2_RES information elements in the minimum telepoint CFP configuration.

Annex F (informative)

Message sequence diagrams

The following diagrams represent typical layer three message sequences used in CT2 applications. Where multiple messages are shown, the convention is adopted that the message closest to the head of the arrow is the first to be sent and the first to arrive.

Examples of the following situations are shown:

- call set up from CPP to telepoint CFP;
- call set up from CPP to private CFP;
- group call from private CFP to CPPs;
- call set up from telepoint CFP to CPP;
- call clear down;
- on-air registration from CPP to private CFP.

Note that although not shown in these diagrams, messages may, on occasion, overlap. That is, one may be in the course of being sent from one end of a CTA whilst another, shown subsequently in the diagram, is in the course of being received at that end. This is usually true of TERM_CAP, FA and BAS_CAP at the transition from LINK ESTABLISHED to CALL IN PROGRESS. Some messages cannot overlap, such as AUTH_REQ and AUTH_RES, where the complete reception of the first is essential before sending of the second can be started.

F.1 Call set up to a telepoint CFP

INITIAL STATE: CPP idle.

ACTION: The user initiates a call to a telepoint system. The user, after the telepoint authentication procedure (UKF1), dials 7654, using overlap sending.

CPP	MESSAGE	CFP
Call request (telepoint) > DL_EST_REQ -------> DL_EST_IND <-------	==================MUX3=========> <====bidirectional MUX2=========> layer one, two initialise (subclause 5 and subclause 6)	---------> DL_EST_IND
LINK ESTABLISHED >	FA=3;n TERM_CAP=CIC,MB --------------------------------> BAS_CAP <--------------------------------	**LINK ESTABLISHED** n recognised ? Codec etc compatible? <
CALL IN PROGRESS Display cleared Show "telepoint active"	DISP=FF,<MESSAGE>(NOTE) FI=3;n,1 <--------------------------------	**CALL IN PROGRESS**
	AUTH_REQ(RAND,INCZ=0) <--------------------------------	Authentication req. (no "zap")
CPP sends authent- ication data.	AUTH_RES(CKEY,TRD,ZAP) -------------------------------->	Authentication OK?
	CC=0 (transmitted in MUX2) <--------------------------------	T2R1 (Tx in MUX2, Rx in MUX1)
T1R1 (Tx in MUX1, Rx in MUX1)	ack (transmitted in MUX1) - - - - - - - - - - - - - - - - - >	
	CC=2 (transmitted in MUX1) <--------------------------------	T1R1 (Tx in MUX1, Rx in MUX1)
Connect B channel	ack - - - - - - - - - - - - - - - - - >	Connect B channel to line (dial tone)
	<====bidirectional MUX1=========>	
User keys first address digit.	KP='7' -------------------------------->	(in band dial tone off)
Display echo char. User keys remaining digits.	DISP=FF,'7' <-------------------------------- KP='6' --------------------------------> DISP='6' <-------------------------------- KP='5' --------------------------------> DISP='5' <-------------------------------- KP='4' -------------------------------->	
	DISP='4' <-------------------------------- <====call in progress===========>	(in band exchange ring back tone - far end answers - call in progress)

NOTE: <MESSAGE> may be any indication of the availability of the telepoint service.

F.2 Call set up to a private CFP

INITIAL STATE: CPP idle.

ACTION: The user initiates a call to a private system using feature activator 0,0. The user dials 7654 from the number store (the digits are sent en bloc).

CPP	MESSAGE	CFP
Call request > DL_EST_REQ -------> DL_EST_IND <-------	=================MUX3======> <===bidirectional MUX2======> layer one, two initialise (subclause 5 and subclause 6)	---------> DL_EST_IND
LINK ESTABLISHED ——————————————>	FA=0,0 TERM_CAP ---------------------------->	**LINK ESTABLISHED**
	BAS_CAP <----------------------------	<————————————————
CALL IN PROGRESS		**CALL IN PROGRESS**
Display cleared.	FI=0,0;1 DISP=FF <----------------------------	
	CC=0 (transmitted in MUX2) <----------------------------	T2R1 (Tx in MUX2, Rx in MUX1)
T1R1 (Tx in MUX1, Rx in MUX1)	ack (transmitted in MUX1) - - - - - - - - - - - - - - - >	
	CC=2 (transmitted in MUX1) <----------------------------	T1R1 (Tx in MUX1, Rx in MUX1)
Connect B channel	ack - - - - - - - - - - - - - - - >	Connect B channel to line (dial tone)
	<===bidirectional MUX1======>	
User retrieves number from store.	KP='7654' ---------------------------->	(in band dial tone off)
Display echo chars.	DISP='7654' <----------------------------	(in band exchange ring back tone - far end answers - call in progress)
	<===call in progress========>	

F.3 Private CFP incoming (group) call

INITIAL STATE: CPP idle.

ACTION: An outside call arrives at the CFP. Multiple ringing is initiated to two CPPs registered as a single ringing group. Both CPP are in range and respond to layer two messages. User at CPP 1 answers.

CFP	MESSAGE	CPP
Incoming line call detected.	<===poll, responses MUX2====>	
Repeat ring signal until answer, following cadence.	SIG = 1,0 or FI=0,n;2 (UI) ---------------------------> SIG = 0,X or FI=0,n;2 (UI) --------------------------->	Incoming ringing on Incoming ringing off
		User at CPP 1 selects line.
DL_EST_IND <--------	layers one, two and protocol initialised (subclauses 5 & 6).	---------> DL_EST_IND
LINK ESTABLISHED	TERM_CAP, FA = 0,m <---------------------------	**LINK ESTABLISHED**
	SIG = 0,X, FI = 0,n;1 --------------------------->	Stop ringing
	BAS_CAP --------------------------->	
CALL IN PROGRESS		**CALL IN PROGRESS**
T2R1 (Tx in MUX2, Rx in MUX1)	CC=0 (transmitted in MUX2) --------------------------->	
	ack (transmitted in MUX1) < - - - - - - - - - - - - - -	T1R1 (Tx in MUX1, Rx in MUX1)
T1R1 (Tx in MUX1, Rx in MUX1)	CC=2 (transmitted in MUX1) --------------------------->	
	ack < - - - - - - - - - - - - - -	Connect B channel
Connect B channel to line	<===bidirectional MUX1======>	
	<=== call in progress ======>	

F.4 Telepoint CFP incoming call

INITIAL STATE: CPP idle.

ACTION: An outside call arrives at the telepoint CFP. Ringing is initiated to a registered CPP. The CPP is in range and responds to layer two messages. The user at the CPP answers.

CFP	MESSAGE	CPP
Incoming line call detected.		
>	<=====poll, responses MUX2======>	
Repeat ring signal until answer, following cadence.	SIG = 1,0 or FI=3,n;2 (UI) ------------------------------->	Incoming ringing on
	SIG = 0,X or FI=3,n;2 (UI) ------------------------------->	Incoming ringing off
		User at CPP 1 selects line.
		<
DL_EST_IND <--------	layers one, two and protocol initialised (subclauses 5 & 6).	---------> DL_EST_IND
LINK ESTABLISHED		**LINK ESTABLISHED**
	TERM_CAP, FA = 3;m <-------------------------------	<
>	SIG = 0,X, FI = 3,n;1 ------------------------------->	Stop ringing
>	BAS_CAP ------------------------------->	
CALL IN PROGRESS	FI=3;n,1 DISP=FF,<MESSAGE> (NOTE) ------------------------------->	**CALL IN PROGRESS** Display cleared. Show "telepoint active".
Authentication request (no "zap").	AUTH_REQ(RAND,INCZ=0) ------------------------------->	
Authentication OK?	AUTH_RES(CKEY,TRD,ZAP) <-------------------------------	CPP sends authentication data.
T2R1 (Tx in MUX2, Rx in MUX1)	CC=0 (transmitted in MUX2) ------------------------------->	
	ack (transmitted in MUX1) < - - - - - - - - - - - - - - - - -	T1R1 (Tx in MUX1, Rx in MUX1)
T1R1 (Tx in MUX1, Rx in MUX1)	CC=2 (transmitted in MUX1) ------------------------------->	
Connect B channel to line	ack < - - - - - - - - - - - - - - - - -	Connect B channel
	<=====bidirectional MUX1========>	
	<===== call in progress ========>	

NOTE: <MESSAGE> may be any indication of the availability of the telepoint service.

F.5 Call clear down

INITIAL STATE: Call in progress.

ACTION: User clears the call.

CPP	MESSAGE	CFP
CALL IN PROGRESS	<===call in progress========>	**CALL IN PROGRESS**
Call release req.	FA=7,1 ----------------------------> ack < - - - - - - - - - - - - - - -	No outstanding sessions. Exchange line released.
B chan off,disp idle, transducers off.	INIT <---------------------------- ack - - - - - - - - - - - - - - - >	Terminate call
	<=== end of call ===========>	
	Layers one and two released (not shown)	

F.6 On air registration from CPP to private CFP

INITIAL STATE: CPP idle.

ACTION: The user enables on-air registration at the CFP and then activates the registration sequence from the CPP. A four-digit pin is requested, which the user enters ('3*09'). It is echoed as '----' for security.

CPP	MESSAGE	CFP
The user invokes the registration feature.	= LID=11...11 === MUX3 =====> <== bidirectional MUX2 =====>	The user presses the registration enable button.
> DL_EST_REQ -------> DL_EST_IND <-------	layer one, two initialise (subclause 5 and subclause 6)	----------> DL_EST_IND
LINK ESTABLISHED	FA=7,7 TERM_CAP ----------------------------->	**LINK ESTABLISHED**
	BAS_CAP <-----------------------------	<
CALL IN PROGRESS	FI=7,7;1 DISP=FF <-----------------------------	**CALL IN PROGRESS**
Display cleared. On-air registration icon illuminated.	DISP = 'EntEr Pin ' <-----------------------------	
User enters pin.	KP = '3' ----------------------------->	
	DISP = FF,'-' <-----------------------------	
	KP = '*' ----------------------------->	
	DISP = '-' <-----------------------------	
	KP = '0' ----------------------------->	
	DISP = '-' <-----------------------------	
	KP = '9' ----------------------------->	
	DISP = '-' <-----------------------------	
	OARAC(BID) <-----------------------------	
	DISP = FF DISP = 'donE' <-----------------------------	
Display idle, icons cleared	INIT <-----------------------------	
	ack - - - - - - - - - - - - - - - >	
	<=== end of registration ===>	

Annex G (informative):

Code word example

An example code word with CRC is shown below. This is a fill-in code word with N(s), N(r) and REJ all set to 0.

Bit: 8	7	6	5	4	3	2	1	octet
0	0	1	0	0	0	1	1	1
0	0	0	0	0	0	0	0	2
1	1	1	1	0	0	0	0	3
1	1	1	1	0	0	0	0	4
1	1	1	1	0	0	0	0	5
1	1	1	1	0	0	0	0	6
0	1	0	1	0	0	0	0	7
1	1	1	0	1	1	1	1	8

The CRC is calculated as follows:

i) division of x^{15} times the data in octets 1 to 6 above by the generating polynomial of 6.3.6 (i) gives a 15-bit remainder of:

$x^{10} + x^8 + x^6 + x^5 + x^4 + x^3 + x^1$,

that is 000010101111010. The division may be carried out by normal long division techniques but using exclusive-or as the digit-by-digit subtraction method.

ii) the last bit (the coefficient of x^0) is then inverted (6.3.6 (ii)).

iii) the parity bit is then added to provide even parity for the 64-bit code word (6.3.6 (iii)). The check word thus becomes 0000101011110111 as shown above (note the ordering of the bits).

Annex H (informative):

Intellectual property rights

The following eight companies have signed an intellectual property rights agreement concerning the manufacture of equipment which complies with the standard for the CT2 (CAI) telephone service, of which this specification forms a part.

The companies have patent and other intellectual property rights which may relate to the specification and have agreed to make available certain specified licensing rights to users and manufacturers in specific territories. In certain cases these rights will be available free of charge.

For information, the CAI secretariat should be contacted at the following address:

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For further information, the patent department of the following United Kingdom companies should be contacted at the address provided:

British Telecommunications PLC
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Annex J (normative):

Subjective speech quality tests

NOTE: the references cited in this annex are listed in subclause J.6.

Throughout this annex, two options are contemplated, as referred to in subclause 11.24, namely:

Option A whereby the CTA is tested as an entirety; and

Option B whereby the codec (as defined in subclause 11.24.3) is tested in isolation, with a view to approval for incorporation in a variety of CTAs subject to the conditions of subclause 11.24.2.

Where the requirements of the two options diverge, they are stated and headed separately. Otherwise the same requirements apply for both options, and the phrase "apparatus under test" is used to denote either the complete CTA or the codec as the case may be.

The test comprises the following stages.

1) preparing a master recording of group of sentences;

2) a) Replaying the sentences in a defined manner through the apparatus under test and re-recording the output; and

 b) re-recording the sentences through a certain controlled-distortion system;

3) replaying the output from 2) to 20 subjects, and collecting from each subject a response to each group of sentences;

4) processing the results so obtained to derive ratings, to which the criteria of subclause 11.24.3 are then applied.

Stages 1) and 2b) can be carried out once for all, and the resulting recordings used in all subsequent tests. Alternatively, as explained below, the re-recording part of stage 2a) or 2b) or both may be omitted if the appropriate apparatus is available for real-time use during the listening sessions themselves.

J.1 Preparation of master recordings

J.1.1 Speech material

The speech material shall consist of 440 simple, meaningful, short sentences, all different, and chosen at random as being easy to understand (e.g. from current non-technical literature or newspapers). Very short and very long sentences shall be avoided, the aim being that each sentence when spoken should have a duration of about 2 s.

These sentences shall be made up into lists in random order, in such a way that there is no obvious connection of meaning between one sentence and the next. There shall be 22 such lists, each having 20 sentences in 5 groups of 4. Suitable sentences can be found in appendix 3 of reference [J.1], and can be rearranged to constitute the required number of lists and groups.

Four talkers are required. They shall be native speakers of English without obvious speech defects; two male and two female.

Let the lists of sentences be denoted by the letters A to T, Y and Z. Let the talkers be designated 1, 2, 3 and 4. Then talkers shall record the following lists respectively (Table J.1).

Table J.1: Lists to be recorded by talkers

Talker	Lists
1 (male)	A B C D E Y
2 (male)	F G H I J
3 (female)	K L M N O Z
4 (female)	P Q R S T

J.1.2 Apparatus and environment

The talkers shall be seated (one at a time) in a room with reverberation time less than 500 ms, and room noise level below 30 dB A-weighted with no dominant peaks in the spectrum.

Speech shall be recorded from a linear microphone and low-noise amplifier with a flat frequency response as specified in IEC Publication 581-5. The microphone shall be positioned between 140 mm and 200 mm from the talker's lips. A wind-screen shall be used if breath puffs from the talker are noticed.

The same speech shall also be recorded simultaneously from the sending output of a telephone equivalent in sending performance to the Intermediate Reference System [J.2], with the handset held in the normal manner.

The recording system [1)] shall be a high-quality two-channel cassette system with a frequency response that is flat within ±1 dB from 50 Hz to 8000 Hz and a dynamic range (peak signal to average noise ratio) of at least 60 dB. Two separate recording systems shall be used simultaneously: one for recording the wideband speech in one channel, and the other for recording the telephone speech in the corresponding channel. The other channel of each recording system shall be used for recording control signals as explained below.

[1)] The PCM-F1 Digital Audio Processor, manufactured by Sony, used with the 14-bit option, in conjunction with the Sony SL-F1UB or SL-F1E videocassette recorder, has been found suitable.

J.1.3 Recording procedure

The talkers shall enunciate each list according to the following pattern.

"Talker no.1"

"Beginning of list A"

"Beginning of group 1"

sentence 1 from list A

sentence 2 from list A

sentence 3 from list A

sentence 4 from list A

"End of group 1"

"Beginning of group 2"

sentence 5 from list A

sentence 6 from list A

sentence 7 from list A

sentence 8 from list A

"End of group 2"

and so on to the end of the list

"End of list 1"

and similarly for other talkers and lists.

Talkers should pronounce the sentences fluently but not dramatically.

Each announcement shall be followed by a pause lasting up to 5 s, and the sentences within each group shall be timed to begin at regular intervals of 4 s.

This can be achieved by giving a sequence of lamp signals (3 s on, 1 s off) to indicate to the talker when to read the next sentence from a script. However, the following method is far superior, and is strongly recommended, in order to satisfy the timing requirements, to provide synchronization and control signals suitable for automating the subsequent processing and replay of the recordings, and to avoid noise from sources such as the rustling of paper.

The announcement or sentences are presented to the talker one by one on visual display. The display of each sentence is immediately preceded by a control tone (e.g. 1 kHz at a conveniently high level) lasting 0,5 s, recorded in the second channel. A computer program co-ordinates these events according to the following time sequence (Table J.2).

Table J.2: Required sequence of events

Duration (s)	On display	In second channel
3	"Talker" announcement	
2	blank	
3	"List" announcement	
2	Blank	
3	"Beginning of group" announcement	
0,5	blank	
0,5	blank	control tone
3	sentence	
0,5	blank	
0,5	blank	control tone
3	sentence	
0,5	blank	
0,5	blank	control tone
3	sentence	
0,5	blank	
0,5	blank	control tone
3	sentence	
0,5	blank	
0,5	blank	control tone
3	"End of group" announcement	
2	blank	
3	"Beginning of group" announcement	
0,5	blank	
0,5	blank	control tone

and so on to the end of the list.

J.1.4 Calibration signals and speech levels

The active speech level shall be observed during recording. Care shall be taken during the recording process that the active speech level in both recording systems is between 20 dB and 30 dB below the overload point of recording system for each sentence measured separately. Any group of sentences for which this does not hold shall be re-recorded.

The recordings when completed shall be played back, and the active speech level of each sentence shall be measured. The lists (announcements, sentences and control tones) shall then be re-recorded on to a second system with the necessary gain adjustments, so as to bring each group of sentences to the standardised active speech level specified below, and still preserve the proper time relationships between the sentences and the tone signals in the other channel. At the beginning of each resulting recording, 20 s of 1 kHz tone shall be inserted at the re-recording stage (for calibration purposes) at the level specified below.

For the narrowband speech, the standardised level shall be derived by measuring and adjusting the narrowband recorded signal directly, and shall be -20 dB (±0,5 dB) relative to the peak overload level of the recording system. The 1 kHz calibration tone shall have its r.m.s. level equal to the mean active level of the re-recorded speech. The resulting standardised recordings shall be called "telephone-band input recordings".

For the wideband speech, the individual target speech levels shall be such that equality is maintained at the output of an electric-acoustic-electric replay chain consisting of:

a) a bandpass filter, passing only the one-third octave bands from that centred at 200 Hz to that centred at 4 kHz inclusive; and

b) an artificial mouth; and

c) an IRS sending end, with the handset mounted as follows:

   Where the mouthpiece of the apparatus is fixed relative to the earpiece, the handset is placed in the LRGP as described in Annex A of CCITT Recommendation P.76 [16]. Where the mouthpiece of the apparatus is not fixed relative to the earpiece, the front plane of the mouthpiece is mounted 15 mm from the front of the lip ring and coaxial with the artificial mouth. The earcap is sealed to the knife-edge of the artificial ear.

The speech shall be measured at the output of the IRS sending end, but re-recorded from the output of the bandpass filter, with the gain adjustment before the bandpass filter calculated to give a level of -17 dBV (±0,5 dB) at the measurement point. In order to avoid overloading the artificial mouth, care shall be taken during this process to keep speech level at the mouth reference point low enough, even before adjustment, to give an active level not exceeding -5 dBPa for any sentence. The 1 kHz calibration tone shall have its r.m.s. level at the re-recording point such that when measured through the electric-acoustic-electric replay chain it is at -12 dBV. The resulting standardised recordings shall be called "4 kHz input recordings".

J.2 Processing of recordings

J.2.1 General

The input recordings shall be played through various conditions to generate ten sets of output cassettes as follows. The input gain of the re-recording system shall be so adjusted for each condition that the level of the calibration tone on the output cassettes is the same for all conditions.

In all cases the second channel of each input recording, containing the control tones, shall be directly copied on to the second channel of the corresponding output cassette simultaneously with the processing described below, so as to maintain the time relationships on the output cassettes.

The re-recording part of this processing may be dispensed with in subclause J.2.2 if the apparatus under test is available for use in real time during the listening sessions, and in subclause J.2.3 if the Modulated Noise Reference Unit is available for use in real time during the listening sessions, provided that the resulting signal delivered to the telephone receiving end meets the condition specified in subclause J.3.1.

J.2.2 Processing through the apparatus under test

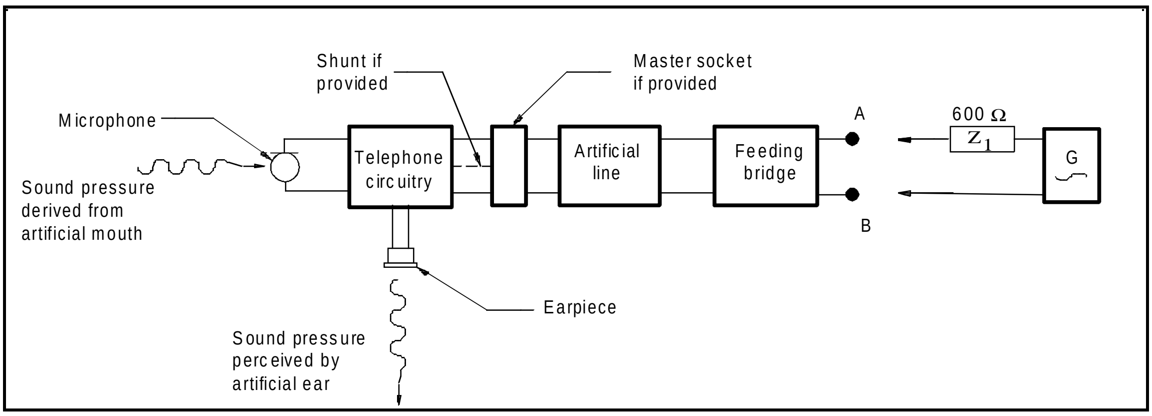
Option A. The CTA shall have the 4 kHz input recordings for lists A to T inclusive re-recorded through its sending path in the following conditions:

1) CTA sending with high vocal level (0 dBPa);

2) CTA sending with median vocal level (-8 dBPa);

3) CTA sending with low vocal level (-16 dBPa);

4) CTA sending with median vocal level and added noise.



NOTE 1: The artificial line and the feeding bridge shall meet the appropriate national requirements.

NOTE 2: The artificial mouth and the artificial ear are described in CCITT Recommendation P.51[16]

Figure J.1: The local telephone system (LTS)

This re-recording shall be carried out with the arrangement of apparatus as specified in CCITT Recommendation P.64 (§ 6 and Annex B § 1, upper envelope method [13]) with a line length of 1 km. The cassette replaying apparatus shall be connected through a level-control attenuator to the artificial mouth, in place of the signal generator. Hardware equalization interposed after the level-control attenuator shall ensure that the frequency response from cassette-replaying apparatus to the mouth reference point is flat within ±2 dB (a measurement of pink noise input and output in one-third octave bands suffices to check this requirement). The acoustic level shall be adjusted until the calibration tone is at the sound pressure level specified for condition 1 above. All the lists shall be played through, and re-recordings (to include all the announcements and the calibration tones) shall be made on a identical recording system connected to points A and B of figure J.1, with the same precautions regarding the relationship between signal levels and peak overload level as were observed in making the original recordings. The same process shall be repeated for conditions 2, 3, and 4. The different acoustic levels required for these conditions shall be achieved by varying the level-control attenuator. In condition 4 the noise (which may be added either on replay of the input recordings or at recording time on to a special version of the input recording) shall be white noise, band-limited (300 Hz to 3400 Hz), at an average level (psophometrically weighted) 35 dB below the active speech level.

Option B. The telephone-band input recordings for lists A to T inclusive shall be re-recorded through the codec in the following conditions:

1) codec with high input level (8 dB above nominal optimum level);

2) codec with median input level (nominal optimum level);

3) codec with low input level (8 dB below nominal optimum level);

4) codec with median input level and added noise.

The cassette-replaying apparatus shall be connected to the codec input through a level-control attenuator. The input level shall be adjusted until the calibration tone is at the r.m.s. voltage level specified for condition 1 above. All the lists shall be played through, and re-recordings (to include all the announcements and the calibration tones) shall be made on an identical recording system connected to points A and B of figure J.1, with the same precautions regarding the relationship between signal levels and peak overload level as were observed in making the original recordings. The same process shall be repeated for conditions 2, 3, and 4. The different input levels required for these conditions shall be achieved by varying the level-control attenuator. In condition 4 the noise (which may be added either on replay of the input recordings or at recording time on to a special version of the input recording) shall be white noise, band-limited (300 Hz to 3400 Hz), at an average level (psophometrically weighted) 35 dB below the active speech level.

J.2.3 Processing through controlled distortion

The telephone-band input recordings of lists A to T inclusive shall be played at constant level into the Modulated Noise Reference Unit [J.4], and re-recorded at the output (correctly terminated) in the following five conditions:

5) normal telephone speech via MNRU at Q = 10;

6) normal telephone speech via MNRU at Q = 15;

7) normal telephone speech via MNRU at Q = 20;

8) normal telephone speech via MNRU at Q = 25;

9) normal telephone speech via MNRU at Q = 30.

The telephone-band input recordings will also be used in the following condition, which does not require any processing (i.e. the output cassettes are simple copies of the input recordings):

10) normal telephone speech without MNRU (or Q = infinity).

Lists Y and Z will be used for practice for the listeners, in a special condition 0, generated from the telephone-band input recording of these lists with noise added (as in condition 4) at 20 dB below the active speech level to the second and fourth groups only. The output cassettes for this condition, like those for conditions 5 to 10, can be prepared once for all.

J.3 Conduct of listening test

J.3.1 Apparatus, calibration and environment

The output recordings shall be replayed via a level-control attenuator into a telephone with receiving performance equivalent to that of the Intermediate Reference System [J.2]. There shall be five predetermined settings of the attenuator such as to deliver the following calibration tone levels (and hence nominal active speech levels) in dBV to the listening end for all conditions: -12, -22, -32, -42, -52. The listening room shall meet the same conditions as the recording room.

Band-limited white noise, (300 Hz to 3400 Hz), with an average level equivalent to -70 dBmp measured at the 0 dB RLR input, shall be injected into the receiving end of the Intermediate Reference System at all times when the subject is listening, whether any speech is being transmitted to the subject or not.

J.3.2 Selection of subjects.

Subjects shall be native speakers of English without any obvious defects in hearing. The following shall be excluded: persons who admit to specialist knowledge of the transmission, processing, broadcasting or recording of speech; and persons familiar with the sentence lists, such as those who compiled or recorded them. Subjects should ideally never have heard the lists of sentences before, but in any case no subject shall participate in a listening test using the same lists of sentences more than once in 12 months.

J.3.3 Procedure

Subjects shall undergo the listening test in two sessions, with an interval of not less than 10 minutes. Recordings shall be played back to subjects according to the following experimental design (based on two 10 x 10 graeco-latin squares interleaved, Tables J.3 and J.4).

Table J.3: First Session

Column: Row	P	1	2	3	4	5	6	7	8	9	10
1	0 Y	6 A	1 K	2 I	7 R	1 J	6 S	7 C	5 T	9 G	10 L
2	0 Z	9 I	8 Q	1 G	2 L	2 A	1 R	10 J	7 S	4 C	6 T
3	0 Y	4 F	9 P	10 C	8 K	9 E	3 M	5 A	2 R	2 B	1 S
4	0 Z	10 H	2 T	3 A	10 P	4 B	9 K	8 I	8 M	6 D	5 O
5	0 Y	8 J	4 S	5 B	3 T	6 C	10 Q	9 H	9 L	1 E	8 N
6	0 Z	1 C	6 R	6 F	5 S	5 H	4 T	2 E	10 K	3 J	9 M
7	0 Y	3 E	3 L	9 D	4 M	8 F	5 N	4 G	6 O	7 H	7 P
8	0 Z	5 G	5 M	7 J	6 N	10 D	7 O	6 B	1 P	8 A	2 Q
9	0 Y	7 B	10 O	4 H	9 Q	3 I	8 L	1 D	4 N	10 F	3 R
10	0 Z	2 D	7 N	8 E	1 O	7 G	2 P	3 F	3 Q	5 I	4 K

Table J.4: Second Session

Column: Row	P	11	12	13	14	15	16	17	18	19	20
1	0 Z	8 B	9 N	5 F	8 P	3 H	2 M	4 D	3 O	10 E	4 Q
2	0 Y	5 D	10 M	8 H	9 O	7 F	3 N	6 E	4 P	3 B	5 K
3	0 Z	1 H	7 T	3 D	10 N	6 J	4 O	7 I	5 Q	8 G	6 L
4	0 Y	7 E	4 R	9 J	3 S	2 G	6 Q	1 F	7 L	5 C	1 N
5	0 Z	2 F	6 P	10 I	5 R	4 A	7 K	3 G	1 M	7 D	2 O
6	0 Y	10 G	8 O	7 A	7 Q	8 D	1 L	9 B	2 N	4 I	3 P
7	0 Z	6 I	1 Q	2 C	2 K	10 B	8 R	5 J	9 S	1 A	10 T
8	0 Y	3 C	3 K	4 E	4 L	1 I	9 T	2 H	10 R	9 F	8 S
9	0 Z	9 A	2 S	6 G	1 T	5 E	5 P	8 C	6 K	2 J	7 M
10	0 Y	4 J	5 L	1 B	6 M	9 C	10 S	10 A	8 T	6 H	9 R

Where:

- rows represent listeners;
- columns represent order of presentation;
- numbers represent conditions as described in J.2; and
- letters represent lists of sentences (each with its own talker).

Each row shall be used for two listeners, who may listen and respond simultaneously if this is convenient, provided their responses are recorded distinctly and are not known to each other. Within each row, the specified condition-list combinations shall be played in the given order, each session commencing with the cell in the appropriate column headed "P". Within each list (single cell of the square), each group of four sentences shall be presented at a different one of the five predetermined attenuations (listening levels), the order of these attenuations being freshly randomised each time.

It is strongly recommended that each list in each condition should be recorded on a separate output cassette, so that the condition-list combinations can be administered to each subject in the required order with a minimum of winding time. If this is done, each session can be expected to last between 20 minutes and 30 minutes.

The announcements, "Beginning of list 1" and so on, are recorded for the benefit of the processing laboratory and the experimenter conducting the listening test, but shall not be played back in such a way that they could be heard by the subjects.

For each group of sentences, each listener shall give a response on the following scale.

Opinions based on the effort required to understand the meanings of sentences:

A) complete relaxation possible; no effort required;

B) attention necessary: no appreciable effort required;

C) moderate effort required;

D) considerable effort required;

E) no meaning understood with any feasible effort.

Subjects' responses may be collected by any convenient method, e.g. pencil and paper, press-buttons controlling lamps recorded by the operator, or automatic data-logging equipment. But whatever method is used, subjects shall not be able to observe other subjects' responses, nor shall they be able to see the record of their own previous responses. Apart from the inevitable memory and practise effects, each response must be independent of every other. No information shall be available to the subjects which could identify the circuit conditions or the listening levels, or indicate which of them are the same. The only information available to the subjects must be that which they derive by their own powers of perception in listening to the speech.

Together with each response, information shall be recorded identifying the row and column of the design (hence identifying the condition, list and talker), and the listening level used in each case; the two listeners using the same row of the design shall be distinguished.

On arrival, subjects shall receive the following instructions in writing.

NOTE: For convenience the instructions are here formulated with reference to one particular method of signalling to the subject and recording the responses, but if a different method is used then the wording in italics is to be adapted accordingly.

LISTENING EXPERIMENT
In this experiment you will be listening to short groups of sentences via the telephone handset, and giving your opinion of the speech you hear. On the table in front of you is *a box with five illuminated press buttons*. When *all the lamps go on*, you will hear four sentences. Listen to these, and when *the lamps go out, press the appropriate button* to indicate your opinion on the following scale. Effort required to understand the meanings of sentences. A Complete relaxation possible; no effort required. B Attention necessary; no appreciable effort required. C Moderate effort required. D Considerable effort required. E No meaning understood with any feasible effort. *The button you have pressed will light up for a short time*. Then *the lamp will go out and* there will be a brief pause before *all the lamps go on again* for the next four sentences. There will be a longer pause after every 20 sentences (i.e. after every 5 opinions). There will be a total of 210 sentences (calling for 55 opinions) in this session, and the same number in your second session. Thank you for your help in this experiment.

After having read these instructions, and having had the opportunity to ask for any needed clarification, the subject shall listen to the preliminary list and give a response to each group, in order to become familiar with the procedure. No suggestion should be made that the speech in the preliminary list exhausts the range of qualities that can be expected to be heard. After this the subject should be asked whether he or she is ready to go ahead with the rest of the experiment. Questions about procedure or about the meaning of the instruction shall be answered, but any technical questions must be met with the response "We cannot tell you anything about that until the experiment is finished".

At the beginning of the second session the subject shall hear the other preliminary list, in order to become accustomed again to the procedure.

If substantial errors occur in the administration of the test (e.g. if responses are missed, or the wrong recordings are played back) then all the results from the subject concerned shall be discarded (even though as a matter of courtesy the subject may be allowed to finish the sessions and not informed of the error), and that row of the design shall be repeated with a fresh subject. It is essential that each row should be completed by the same subject throughout and that repeat attempts by the same listener with the same sentences should be avoided.

J.4 Treatment of results

Numerical scores shall be allocated to the responses as follows:

- A = 4; B = 3; C = 2; D = 1; E = 0;

and all further analysis shall be in terms of these numbers (known as listening-effort opinion scores). Data from the two P columns (condition 0, lists Y and Z) shall be omitted from all the calculations, i.e. the expression "all scores" is to be taken as meaning "all scores except those in the P columns". Let suffixes be allocated as follows (Table J.5).

Table J.5: Allocation of suffixes

Suffix	Range	Denoting
i	1-20	Subjects (2 per row)
j	1-20	Columns
p	1-20	Condition-Voice combinations (where for condition t, 1 ≤ t ≤ 10, p = t for male speech and p = t + 10 for female speech)
r	1-20	Lists (A to T inclusive)
n	1-5	Listening levels

With these conventions, let y_{ipn} denote the score for the i th listener in the p th condition-voice combination at the n th listening level, and similarly for other suffixes. The mean over talkers and listeners, denoted by Y_{pn} shall be calculated separately for each combination p at each listening level n.

Analysis of variance shall be carried out to evaluate the following effects and test them for significance: rows (listeners), columns (sequence effects), numbers (conditions), letters (talkers and lists, including the male/female talker contrast), listening levels, and the interaction of listening levels with each of the other factors. The results of these significance tests shall be used to determine whether there are any abnormal features of the data such as might cast any reasonable doubt on their reliability, and shall be produced as evidence in defence against any challenge to the proper conduct of the experiment.

The procedure for carrying out the analysis of variance is as follows.

NOTE 1: In the following formulae, some variables are denoted by symbols consisting of more than one alphanumeric character. Such character strings are not products. The multiplication sign "×" will be inserted explicitly where required, except in matrix products.

NOTE 2: In the above analysis-of-variance computations, it is advisable to carry at least 7 decimal places, and to express Sums of Squares and Mean Squares to 4 decimal places; for the F ratios, 2 decimal places are usually sufficient.

Compute the following subtotals and totals:

$$s_{in} = \sum_{j=1}^{20} y_{ijn} \qquad c_{jn} = \sum_{i=1}^{20} y_{ijn}$$

$$k_{pn} = \sum_{i=1}^{20} y_{ipn} \qquad u_{rn} = \sum_{i=1}^{20} y_{irn}$$

$$L_n = \sum_{i=1}^{20} s_{in}$$

$$S_i = \sum_{n=1}^{5} s_{in}$$

$$C_j = \sum_{n=1}^{5} c_{jn}$$

$$K_p = \sum_{n=1}^{5} k_{pn}$$

$$U_r = \sum_{n=1}^{5} u_{rn}$$

$$m_n = \sum_{r=1}^{10} u_{rn} \qquad f_n = \sum_{r=11}^{20} u_{rn}$$

$$M = \sum_{n=1}^{5} m_n \qquad F = \sum_{n=1}^{5} f_n$$

$G = \sum_i \sum_j \sum_n y_{ijn}$ = sum of all scores, and let $Gq = G^2/2000$.

Compute the "sq" terms, known as "Sums of Squares" to be inserted in the following Analysis of Variance table.

Factor	D.F.	Sum of Squares
Listening levels	4	$sq1 = (\sum_{n=1}^{5} L_n^2)/400 - Gq$
Male/female talkers	1	$sq2 = (M^2 + F^2)/1000 - Gq$
Interaction	4	$sq3 = (\sum_{n=1}^{5} m_n^2 + \sum_{n=1}^{5} f_n^2)/200 - Gq - sq1 - sq2$
Subjects	19	$sq4 = (\sum_{i=1}^{20} S_i^2)/100 - Gq$
Interaction	76	$sq5 = (\sum_{n=1}^{5} \sum_{i=1}^{20} s_{in}^2)/20 - Gq - sq1 - sq4$
Columns	18	$sq6 = (\sum_{j=1}^{20} C_j^2)/100 - Gq - sq2$
Interaction	72	$sq7 = (\sum_{n=1}^{5} \sum_{j=1}^{20} c_{jn}^2)/20 - Gq - sq1 - sq6 - sq2 - sq3$
Conditions	18	$sq8 = (\sum_{p=1}^{20} K_p^2)/100 - Gq - sq2$
Interaction	72	$sq9 = (\sum_{n=1}^{5} \sum_{p=1}^{20} k_{pn}^2)/20 - Gq - sq1 - sq8 - sq2 - sq3$
Lists	18	$sq10 = (\sum_{r=1}^{20} U_r^2)/100 - Gq - sq2$
Interaction	72	$sq11 = (\sum_{n=1}^{5} \sum_{r=1}^{20} u_{rn}^2)/20 - Gq - sq1 - sq10 - sq2 - sq3$
Residual	1625	$sq12 = sq13 - sq1 - sq2 - sq3 - sq4 - sq5 - sq6 - sq7 - sq8 - sq9 - sq10 - sq11$
Total	1999	$sq13 = \sum_{n=1}^{5} \sum_{i=1}^{20} \sum_{j=1}^{20} (y_{ijn})^2 - Gq$

"D.F." stands for "degrees of freedom".

From each sum of squares divided by the degrees of freedom in the same row, compute a "mean square" to be entered on that same row in a further column of the above table.

From each mean square (except the last two) compute the corresponding "F ratio", by dividing by the appropriate denominator. This is the Residual Mean Square in all cases except two, namely:

a) The Male/Female Talkers mean square (sq2/1), for which the denominator is the Lists mean square (sq10/18); and

b) the M/F × Levels Interaction mean square (sq3/4), for which the denominator is the Lists × Levels Interaction mean square (sq11/72).

Test the significance of each factor by comparing its F ratio with the values in a standard table, such as table 18 of reference [J.5]. Each effect can then be judged "not significant", "significant" or "highly significant".

"Significant" in this context is a technical term meaning that the magnitude of the observed effect is such that there is a probability smaller than 0,05 of obtaining an equal or greater observed value "by chance", i.e. by sampling fluctuation only from a population with the amount of inherent variation estimated by the Residual Mean Square: in common-sense terms, one would rather believe in such a case that there is a real effect operating than that the sample is merely an unusual one from a population without such a real effect operating. "Highly significant" is similarly defined but with a probability smaller than 0,01.

In this sense it may be expected that Levels, Subjects, Conditions and usually the interaction between Levels and Conditions will be significant or highly significant; that the Male/Female factor and its interaction with levels may or may not be significant, depending on the nature of the codec; and that the Lists factor sometimes, and the Columns factor and other interactions more often, will be not significant. The Residual Mean Square will almost always lie in the range 0,2 to 0,4. Any major deviation from this pattern will call for explanation, and will generally raise suspicion of errors: in the functioning of the apparatus, in the administration of the experiment, in the collection of the scores, or in the arithmetic. Such errors can often be located by examining the mean scores as a function of the value of an unexpectedly behaving variable. For example, an insignificant Levels factor would almost certainly imply that the listening-level control attenuator was not functioning, while a highly significant Columns factor would indicate some abnormality in the time sequence (cassettes out of order; strong fatigue or practise effects, systematic differences between first and second session, etc.).

A Residual Mean Square exceeding 0,5 is expected to occur with only a very low probability, estimated at less than once in every hundred tests, provided these are carried out strictly according to the foregoing instructions and without abnormally large fluctuations in the functioning of the apparatus or in the general conditions. Therefore, a Residual Mean Square larger than 0,5, unless it can be satisfactorily accounted for, shall be regarded as sufficient reason to discard the results and repeat the test.

The value of the Residual Mean Square (RMSq) enters into the confidence limits to be used for judging whether the function fits are satisfactory (see below).

By standard weighted least-squares regression, an equation shall be derived to express the mean score for conditions 5 to 9 inclusive as a function of the listening level and the signal-to-correlated-noise ratio Q. The equation shall be of the form:

$$\ln (Y/(4-Y)) = a + (b\times L) + (c\times L^2) + (d\times L\times M) + (e\times M) + (f\times M^2)$$

Where

$M = 10^{-0,03Q}$

Y = predicted mean score

L = listening level in dBV

and a, b, c, d, e and f are the coefficients to be determined by the regression and the weight to be applied to each value is (Y/(4-Y))/16.

This shall be done separately for the results arising from male speech and the results arising from female speech, a separate equation being derived for each.

The regression coefficients are evaluated as follows.

From the score totals $\mathbf{k_{pn}}$ (where p ranges from 5 to 9 for male speech and from 15 to 19 for female speech), evaluate the mean scores $\mathbf{Y_{pn}}$ and hence the corresponding transformed values

$$\mathbf{T_{pn}} = \ln (\mathbf{Y_{pn}}/(4-\mathbf{Y_{pn}}))$$

and the weighting coefficients

$$\mathbf{W_{pn}} = \ln (\mathbf{Y_{pn}}\times(4-\mathbf{Y_{pn}}))/16;$$

there are 25 pairs of values ($\mathbf{T_{pn}}$ and $\mathbf{W_{pn}}$) for male speech and 25 for female speech, each having an associated value of Q (related to p) and of L (listening level, related to n).

For each of these values of $\mathbf{T_{pn}}$ in turn:

- form the column vector:

  $$\mathbf{G} = [W\times T, W\times T\times L, W\times T\times L^2, W\times T\times L\times M, W\times T\times M, W\times T\times M^2]';$$

- accumulate successive values of **G** into a column vector **J**;

- form the column vector $\mathbf{E} = [1, L, L^2, L\times M, M, M^2]'$;

- accumulate successive values of the product Wx**EE'** into a matrix $\mathbf{F_m}$ (i.e. each term of **EE'** accumulated into $\mathbf{F_m}$ is multiplied by W).

At the end of this accumulation, calculate the inverse of matrix $\mathbf{F_m}$:

$$\mathbf{H_m} = \mathbf{F_m}^{-1}$$

Premultiply $\mathbf{J_m}$ by $\mathbf{H_m}$ to give a column vector $\mathbf{A_m}$:

$$\mathbf{A_m} = \mathbf{H_m J_m}.$$

The vector $\mathbf{A_m}$ then contains the coefficients a, b, c, d, e, f in that order; in other words,

$$\mathbf{A_m} = [a, b, c, d, e, f]'.$$

By standard least-squares regression, 8 further separate equations shall be derived to express the mean score for each of conditions 1, 2, 3 and 4 as a function of the listening level, separately for the results arising from male speech and the results arising from female speech. These equations shall be of the form.

$$\ln (Y/(4-Y)) = a + b\times L + c\times L^2$$

where

Y = predicted mean score

L = listening level in dBV

and a, b, and c are the coefficients to be determined by the regression.

These equations shall be regarded as satisfactory for the purpose of this standard if, among all the results from conditions 1, 2, 3, and 4, not more than 2 of the observed scores for male speech and not more than 2 of the observed scores for female speech differ from the corresponding predicted scores by more than the appropriate confidence interval, obtained by multiplying 1,96√(RMSq/20) by the following factor depending on L (Table J.6).

Table J.6: Factors for determining the confidence interval

L, dBV	-12	-22	-32	-42	-52
Factor	1,3732	1,1711	1,2189	1,1711	1,3732

The regression coefficients shall be evaluated as follows:

Within each condition (p ranging from 1 to 4 for male speech and from 11 to 14 for female speech), from the score totals k_{pn}, evaluate the mean scores Y_{pn} and hence the corresponding transformed values $T_{pn} = \ln (Y_{pn}/(4-Y_{pn}))$. In each condition-voice combination p, there are 5 of these, each having an associated value of L (listening level, related to n).

For each of these values of T_{pn} in turn:

- form the column vector **G** = [T, T × L, T × L²]';
- accumulate successive values of **G** into column vector $\mathbf{J_p}$;
- form the column vector **E** = [1, L, L²]';
- accumulate successive values of the product **EE'** into a matrix $\mathbf{F_c}$.

At the end of this accumulation, calculate the inverse of matrix $\mathbf{F_c}$:

$$\mathbf{H_c} = \mathbf{F_c}^{-1}$$

Premultiply $\mathbf{J_p}$ by $\mathbf{H_c}$ to give a column vector $\mathbf{A_p}$:

$$\mathbf{A_p} = \mathbf{H_c}\mathbf{J_p}$$

The vector $\mathbf{A_p}$ then contains the coefficients a, b, c in that order; in other words:

$\mathbf{A_p}$ = [a, b, c]' for condition-voice combination p.

NOTE: If the values of L, and the numbers of scores per condition, are all exactly as specified above, then the matrix $\mathbf{F_c}$ will always contain the same values, regardless of p, namely:

$$\mathbf{F_c} = \begin{bmatrix} 5,0000000 & -160,0000000 & 6120,0000000 \\ -160,0000000 & 6120,0000000 & -259840,0000000 \\ 6120,0000000 & -259840,0000000 & 11726880,0000000 \end{bmatrix}$$

The matrix $\mathbf{H_c}$ will also contain the same values, regardless of p, namely:

$$\mathbf{H_c} = \begin{bmatrix} 6,07382857100000 & 0,40868571430000 & 0,00588571428600 \\ 0,40868571430000 & 0,03025714286000 & 0,00045714285710 \\ 0,00588571428600 & 0,00045714285710 & 0,00000714285714 \end{bmatrix}$$

to 14 decimal places.

Normally, therefore, these matrices need not be recalculated. However, any deviation in detail (such as using a different value for one of the listening levels) will necessitate calculating these two matrices afresh.

For each of conditions 1 to 4 in turn, separately for male and female speech at each listening level, the following method shall be used to find the corresponding value of Q_r (Q rating for the condition, listening level and type of voice in question); that is, a value of Q yielding the same value of Y when substituted into the equation for the MNRU conditions with the same listening level and type of voice.

- Evaluate Y for listening level L from the equation for condition-voice combination p;
- substitute the same value of L in the MNRU equation with the set of coefficients determined for male or female speech, as the case may be;
- substitute Q=0 in this equation, evaluate Y and call this value Y_m;
- increase Q in small steps, re-evaluating Y_m each time, until $Y_m \geq Y$ (the step size may be adaptive but should never exceed 1 dB and the boundary finally determined should be accurate within 0,1 dB);
- the value of Q at which Y_m first exceeds Y is designated Q_r (the Q-rating for the condition, listening level and type of voice in question);
- if Y_m remains smaller than Y when Q reaches the value of 40, then Q_r need only be recorded as ">40".

This value of Q_r shall be compared with boundary values, established individually for each approved testing laboratory (as described in Clause J.5), and classed, accordingly, in one of four categories denoted by A, B, C and D as follows (Table J.7):

Table J.7: Categorisation of Q_r

Category	Male speech	Female speech
A	$Q_r \geq Q_{M1}$	$Q_r \geq Q_{F1}$
B	$Q_{M2} \leq Q_r < Q_{M1}$	$Q_{F2} \leq Q_r < Q_{F1}$
C	$Q_{M3} \leq Q_r < Q_{M2}$	$Q_{F3} \leq Q_r < Q_{F2}$
D	$Q_r < Q_{M3}$	$Q_r < Q_{F3}$

NOTE: This description of the method of arriving at the categories is conceptually the simplest. However, an equivalent but computationally simpler method is to substitute into the regression equation the boundary values of Q_r, evaluate the corresponding boundary values of Y_m and compare these with the observed values; for example whenever Y exceeds the value of Y_m obtained with $Q=Q_{M1}$, the category is A for that listening level.

The category letters are tabulated in the form shown in table 5, the rows of which correspond to conditions 1 to 4 and the columns to the first 4 listening levels, separately for male and female speech. The distribution of categories in this table shall determine whether the apparatus under test meets the requirements of subclause 11.24.5.

NOTE: The surface-fitting and curve-fitting computations, described above, require very high-precision calculating facilities when carried out by the foregoing method that, nevertheless, has its own distinct advantages. Traditional methods of calculation make use of "coding", i.e. adjusting the origin and scale of each variable so as to express all quantities in terms of a small range of numbers surrounding their means, then restoring all quantities to their original units at the end. Use of this technique was considered for the test method described above. However, in this test, the only variable suitable for such treatment is L, as the other variables involve non-linear functions, and it is doubtful whether much would be gained by its use. An alternative method, utilizing the above technique may be acceptable if evidence were available that demonstrated its equivalence but in cases of dispute or arbitration, only the specified method should be used to demonstrate compliance.

J.5 Determination of laboratory-specific Q-rating boundaries for the purposes of subclause 11.24.5

In order to determine the classifications specified in subclause 11.24.5, the boundary values applied to the Q-ratings, as described in J.4, shall be established independently for each test laboratory before that laboratory is authorised to carry out definitive tests as described in Annex J. The method of establishing these boundary conditions shall be as follows.

The laboratory in question shall carry out not less than eight "calibration tests", consisting of the complete testing process as specified in Annex J up to and including the evaluation of Q ratings, using, for Conditions 1 to 4, a codec that to the satisfaction of the test-laboratory accreditation authority is shown to conform rigorously to CCITT Recommendation G.721. The same apparatus and the same speech material shall be used each time. The procedure for the selection of subjects shall be the same as will ultimately be used in the definitive tests. However, it is important that the same individual subjects shall not be included more than once, on any account, in this series of tests. Any test for which the Residual Mean Square as defined in J.4 exceeds the value of 0,5 shall not be allowed to count as a calibration test.

Six provisional boundary values (Q_{m1}, Q_{m2} and Q_{m3} applicable to male speech; Q_{f1}, Q_{f2} and Q_{f3} applicable to female speech) shall be located in such a way as to meet all the following conditions simultaneously:

a) all 6 values shall be positive integers.

b) $Q_{m1} \leq 26$.

c) $Q_{m1} - Q_{m2} = Q_{m2} - Q_{m3} = 4$.

d) $Q_{f1} \leq 26$.

e) $Q_{f1} - Q_{f2} = Q_{f2} - Q_{f3} = 4$.

f) when the boundaries Q_{m1}, Q_{m2}, Q_{m3} are applied to male speech and the boundaries Q_{f1}, Q_{f2}, Q_{f3} to female speech, to categorise the Q_r values as explained in J.4, the G.721 codec shall fulfil the criterion of 11.24.5 in at least seven tests out of eight, or a proportionately higher number if more calibration tests are carried out.

Choose the set that results in the highest average of Q_{m1} and Q_{f1} from the alternative sets of values fulfilling the preceding conditions; if more than one set of values meets this condition, then the set that makes Q_{m1} most nearly equal to Q_{f1} shall be chosen. If more than one possibility still remains, then further calibration tests shall be carried out until the combined evidence from all the calibration tests leads to an unambiguous solution.

Each of these provisional values shall then be reduced by 2 dB to give the definitive values (Q_{M1}, Q_{M2}, Q_{M3}, Q_{F1}, Q_{F2} and Q_{F3}) to be used by the test laboratory concerned;

i.e. $Q_{M1} = Q_{m1} - 2$ $\quad Q_{F1} = Q_{f1} - 2$

$Q_{M2} = Q_{m2} - 2$ $\quad Q_{F2} = Q_{f2} - 2$

$Q_{M3} = Q_{m3} - 2$ $\quad Q_{F3} = Q_{f3} - 2$.

J.6 References

[J.1] CCITT Study Group XVIII. Document no. 51R of Rapporteur's Group on Wideband Coding within 64 kbit/s. "Plan for opinion-score tests on 64 kbit/s 7 kHz systems." (1984).

[J.2] CCITT Recommendation P.48 (1989): "Specification for an Intermediate Reference System."

[J.3] CCITT Recommendation P.56 (1989): "The Objective Measurement of Active Speech Level."

[J.4] CCITT Recommendation P.81 (1989): "Modulated Noise Reference Unit (MNRU)." (Geneva, 1989), Blue Book Volume V, pp 198 to 203.

[J.5] E.S. Pearson and H.O. Hartley: "Biometrika tables for Statisticians" (Volume 1, 3rd edition). (Cambridge University Press 1966).

Annex K (informative):

Artificial echo loss for a CFP with a 4-wire interface

An artificial echo loss may be required which simulates the echo from a very good analogue 2-wire telephone. When a public network operator uses an echo canceller in the network (e.g. for a satellite link), the artificial echo loss path provides an in-range echo to ensure that the echo canceller, and its NLP, is active. The NLP cancels the 34 dB handset echo.

It is recommended that it is implemented in the CFP between the line input and the line output, as shown in figure K.1. The loss of the echo path should be 24 dB ±2 dB.

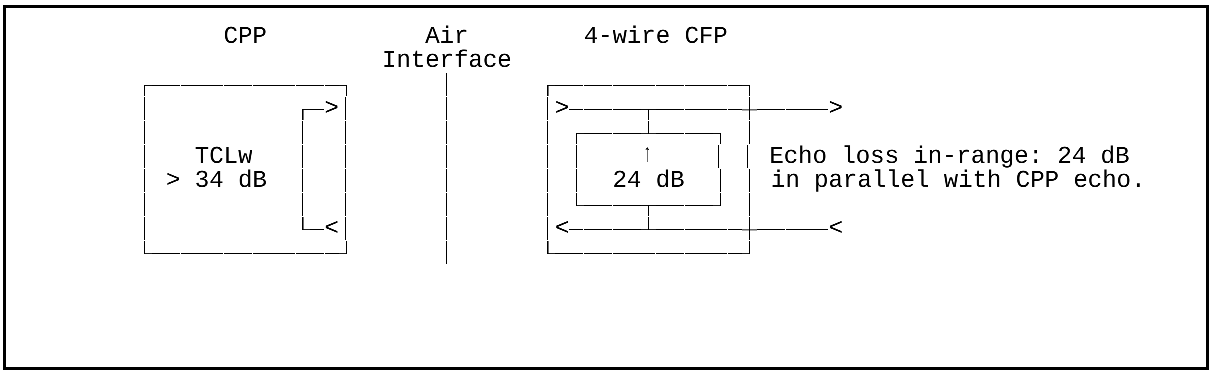


Figure K.1: Artificial echo path in a 4-wire CFP

For connections when it is known that there are no echo cancellers in the public network the artificial echo loss may be disabled.

Annex L (informative):

Network echo from a CFP with a 2-wire analogue interface

It may be required to control the network echo, perceived by the CPP user, by inserting into the receive speech path of the CFP an extra echo loss of X dB.

NOTE 1: The value of X is to be determined, initially, by national administrations.

NOTE 2: X shall be defined by TC/RES in consultation with ETSI sub-technical committees, TE4 and TE5 when this I-ETS is reviewed.

History

Document history	
April 1992	First Edition
March 1996	Converted into Adobe Acrobat Portable Document Format (PDF)