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SERIES I: INTEGRATED SERVICES DIGITAL NETWORK

Overall network aspects and functions – Protocol layer requirements

## AAL type 2 service specific convergence sublayer for narrow-band services

ITU-T Recommendation I.366.2

(Formerly CCITT Recommendation)

ITU-T I-SERIES RECOMMENDATIONS

**INTEGRATED SERVICES DIGITAL NETWORK**



*For further details, please refer to the list of ITU-T Recommendations.*

**ITU-T Recommendation I.366.2**

# AAL type 2 service specific convergence sublayer for narrow-band services


## Summary

This Recommendation defines a Service Specific Convergence Sublayer that operates above the Common Part Sublayer of an AAL type 2 connection. The purpose of the SSCS is to convey narrow-band channels consisting of voice, voiceband data, or circuit mode data.

The SSCS specifies packet formats and procedures to encode the different information streams for bandwidth-efficient transport by AAL type 2. It accommodates known techniques of low rate audio encoding, silence compression, and facsimile demodulation/remodulation. It makes provision for in-band signalling of narrow-band calls and for control of the SSCS operating state.

The SSCS anticipates that multiplexing in the Common Part Sublayer will be used to carry multiple narrow-band channels over individual ATM connections. Additional details needed to configure and manage trunk groups or access configurations, however, are beyond the scope of this Recommendation. The focus is on normative aspects of SSCS coding and behaviour within a single AAL type 2 connection.



## Source

ITU-T Recommendation I.366.2 was revised by ITU-T Study Group 13 (2001-2004) and approved under the WTSA Resolution 1 procedure on 24 November 2000.

## FOREWORD

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

The World Telecommunication Standardization Assembly (WTSA), which meets every four years, establishes the topics for study by the ITU-T study groups which, in turn, produce Recommendations on these topics.

The approval of ITU-T Recommendations is covered by the procedure laid down in WTSA Resolution 1.

In some areas of information technology which fall within ITU-T's purview, the necessary standards are prepared on a collaborative basis with ISO and IEC.

## NOTE

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

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As of the date of approval of this Recommendation, ITU had received notice of intellectual property, protected by patents, which may be required to implement this Recommendation. However, implementors are cautioned that this may not represent the latest information and are therefore strongly urged to consult the TSB patent database.

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# AAL type 2 service specific convergence sublayer for narrow-band services

## 1 Scope

The SSCS specifies packet formats and procedures for the following kinds of information:

– voice encodings;
– silence insertion descriptors;
– circuit mode digital information;
– frame mode data units;
– dialled digits (multifrequency tones);
– channel associated signalling bits;
– demodulated facsimile;
– alarm indications;
– user state control operations;
– rate control;
– synchronization of change in SSCS operation;
– channel loopback.

In general, if an application chooses to implement one or more of the information types, its default option should be to adhere to the corresponding parts of this Recommendation. But the use of alternative packet formats and procedures, standard or proprietary, is not precluded.

The SSCS is defined from the point of view of one instance operating in a single AAL type 2 connection. Other AAL type 2 connections routed over the same ATM connection may employ the same or different SSCS. Any coordination between events that occur on different AAL type 2 connections is outside the scope of this Recommendation.

An instance of the SSCS is characterized by the parameters of operation summarized in clause 18. Some are complex data structures, like the agreed profile of encoding formats in effect. Values of the parameters shall be agreed between the transmitter and the receiver SSCS to be the same for both directions of communication.

It is outside the scope of this Recommendation to state how an instance of the SSCS with specific parameter values should be selected for a particular AAL type 2 connection.

NOTE – Some possibilities are:

1) A configuration may be agreed in advance and imposed manually.
2) The identity of the SSCS and its parameters may be communicated through signalling [Q.2630.1, Q.2630.2].

## 2 References

This Recommendation does not define algorithms for encoding audio streams. It references existing algorithms and specifies how the bits that they output are conveyed within a packet structure. It also makes provision for multiple audio algorithms to be composed into an operational profile, so that the encoding can be varied rapidly among them during a call.

## 2.1 Normative references

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

– ITU-T G.704 (1998), *Synchronous frame structures used at 1544, 6312, 2048, 8448 and 44 736 kbit/s hierarchical levels*.

– ITUT-T G.711 (1988), *Pulse code modulation (PCM) of voice frequencies*.

– ITU-T G.722 (1988), *7 kHz audio-coding within 64 kbit/s*.

– ITU-T G.723.1 (1996), *Dual rate speech coder for multimedia communications transmitting at 5.3 and 6.3 kbit/s*.

– ITU-T G.723.1 Annex A (1996), *Silence compression scheme*.

– ITU-T G.726 (1990), *40, 32, 24, 16 kbit/s adaptive differential pulse code modulation (ADPCM)*.

– ITU-T G.727 (1990), *5-, 4-, 3- and 2-bits/sample embedded adaptative differential pulse code modulation (ADPCM)*.

– ITU-T G.728 (1992), *Coding of speech at 16 kbit/s using low-delay code excited linear prediction*.

– ITU-T G.728 Annex H (1999), *Variable bit rate LD-CELP operation mainly for DCME at rates less than 16 kbit/s*.

– ITU-T G.729 (1996), *Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear prediction (CS-ACELP)*.

– ITU-T G.729 Annex A (1996), *Reduced complexity 8 kbit/s CS-ACELP speech codec*.

– ITU-T G.729 Annex B (1996), *A silence compression scheme for G.729 optimized for terminals conforming to Recommendation V.70*.

– ITU-T G.729 Annex D (1998), *6.4 kbit/s CS-ACELP speech coding algorithm*.

– ITU-T G.729 Annex E (1998), *11.8 kbit/s CS-ACELP speech coding algorithm*.

– ITU-T I.231.1 (1988), *Circuit-mode bearer service categories: Circuit-mode 64 kbit/s unrestricted, 8 kHz structured bearer service*.

– ITU-T I.231.10 (1992), *Circuit-mode bearer service categories: Circuit-mode multiple-rate unrestricted 8 kHz structured bearer service category*.

– ITU-T I.361 (1999), *B-ISDN ATM layer specification*.

– ITU-T I.363.2 (2000), *B-ISDN ATM Adaptation Layer specification: Type 2 AAL*.

– ITU-T I.366.1 (1998), *Segmentation and Reassembly Service Specific Convergence Sublayer for the AAL type 2*.

– ITU-T I.610 (1999), *B-ISDN operation and maintenance principles and functions*.

– ITU-T M.20 (1992), *Maintenance philosophy for telecommunications networks*.

– ITU-T Q.23 (1988), *Technical features of push-button telephone sets*.

– ITU-T Q.24 (1988), *Multifrequency push-button signal reception*.

– ITU-T Q.320 (1988), *Signal code for register signalling*.

– ITU-T Q.322 (1988), *Multifrequency signal sender*.

– ITU-T Q.323 (1988), *Multifrequency signal receiving equipment*.

– ITU-T Q.441 (1988), *Signalling code*.

– ITU-T T.4 (1999), *Standardization of Group 3 facsimile terminals for document transmission*.

– ITU-T T.30 (1999), *Procedures for document facsimile transmission in the general switched telephone network*.

– ITU-T V.17 (1991), *A 2-wire modem for facsimile applications with rates up to 14 400 bit/s*.

– ITU-T V.21 (1988), *300 bits per second duplex modem standardized for use in the general switched telephone network*.

– ITU-T V.27 *ter* (1988), *4800/2400 bits per second modem standardized for use in the general switched telephone network*.

– ITU-T V.29 (1988), *9600 bits per second modem standardized for use on point-to-point 4-wire leased telephone-type circuits*.

– ITU-T V.33 (1988), *14 400 bits per second modem standardized for use on point-to-point 4-wire leased telephone-type circuits*.

– ETSI TS 126 071 V3.0.1 (2000-01), *Universal Mobile Telecommunications System (UMTS); Mandatory Speech Codec speech processing functions – AMR Speech Codec – General Description (3G TS 26.071 version 3.0.1 Release 1999)*.

– ETSI TS 126 101 V3.0.0 (2000-01), *Universal Mobile Telecommunications System (UMTS); Mandatory Speech Codec speech processing functions; AMR Speech Codec Frame Structure (3G TS 26.101 version 3.0.0 Release 1999)*.

### 2.2 Bibliography


– ITU-T Q.2630.1 (1999), *AAL type 2 signalling protocol (Capability Set 1)*.

– ITU-T Q.2630.2 (2000), *AAL type 2 signalling protocol (Capability Set 2)*.


## 3 Definitions

This Recommendation defines the following terms:

**3.1 active voice**: A sampled audio interval that has been determined to contain speech as opposed to silence. The classification is made by a Voice Activity Detection algorithm. It enables discontinuous transmission, whereby the bit rate of the signal is reduced during silent intervals.

**3.2 AAL type 2 connection**: The logical concatenation of one or more AAL type 2 links between two AAL type 2 service endpoints.

**3.3 AAL type 2 link**: The logical user plane communication facility between two adjacent AAL type 2 switching points or service endpoints. An AAL type 2 link is designated by a single CID value.

**3.4 AAL type 2 service endpoint**: A termination point of an AAL type 2 connection.

**3.5 audio**: In the context of this Recommendation, audio is used as a general term for signals of the audible medium, including voice and voiceband data.

**3.6 channel-associated signalling bits**: Bits dedicated for connection control across a 1544 kbit/s or 2048 kbit/s interface that carries 64 kbit/s channels. Procedures are based on the state of up to four signalling bits (A, B, C, D) that are allocated per channel per multiframe. See 3.1.3.2/G.704 and 5.1.3.2/G.704.

**3.7 circuit mode data**: A continuous stream of digital information at N*64 kbit/s having an 8 kHz structure.

**3.8 dialled digits**: Multifrequency audio tones typically used for inter-register signalling of addresses during call set-up or for end-to-end device control during an established call. Depending on the system, codes are defined for the digits 0-9 of a telephone keypad and other auxiliary signals.

**3.9 encoding data unit**: An octet-aligned concatenation of one or more frames of an audio algorithm, entailing a specific format of the bits. Every audio packet and SDU contains an integral number of EDUs. Predefined profiles reference the EDUs that are defined in Annexes B through I.

**3.10 facsimile demodulation/remodulation**: The process of detecting facsimile traffic, extracting digital information from the incoming analogue modulated signal, transporting this across a trunk in packet formats, and reproducing the facsimile control and image information by remodulation at the other end.

**3.11 frame mode data**: An intermittent data stream containing delimited units of information, possibly of varying size, with idle intervals between them.

**3.12 packet**: In the context of this Recommendation, a packet is an AAL type 2 CPS-PDU.

**3.13 packet time**: This concept applies to packets containing voice and to SIDs. For voice, it is the total audio interval represented by the encoded data units. For SIDs, it is the minimum duration of silence that is indicated by the corresponding profile entry; the silence extends indefinitely, because SIDs need not be sent at a regular interval.

**3.14 profile**: A profile is a set of entries, where each entry specifies an encoding format (see Annex A) with a UUI range and length. This set defines a mapping that informs the receiver of a type 1 packet how to interpret the packet contents, i.e. which encoding format in the profile is being used. Once a profile is adopted between a transmitter and a receiver, the transmitter can select any entry of the adopted profile and the receiver shall accept any entry selected by the transmitter.

**3.15 sequence number interval**: The time interval for incrementing sequence numbers in the packets that convey an audio stream. This interval is specified as part of the definition of each entry in a profile.

**3.16 service data unit**: This is the data unit that is passed in primitives between the User and the SSCS. In the context of the audio service provided by the SSCS, an SDU represents an audio signal of a certain duration, encoded as allowed by the adopted profile.

**3.17 silence insertion descriptor**: A compressed representation of the audio background noise that can be sent during silent intervals. SIDs may not be continuous and may only be sent when there is a change in noise characteristics. Playing out received SIDs is known as Comfort Noise Generation.

**3.18 SSCS state**: This state variable takes one of three values: Audio, Circuit Mode, or Facsimile Demodulation. Its effect is to establish the corresponding information stream as primary. It is the responsibility of the Users at the two ends of an AAL type 2 connection to set their local SSCS states consistently, in each direction, for the primary information stream being transported. The concept of SSCS state does not apply to N*64 kbit/s services for N > 1 but does apply to 64 kbit/s services.

**3.19 user state**: This state variable takes one of four values: Voice, Voiceband Data, Circuit Mode, or Facsimile Demodulation. It is separate from but can be mapped to an SSCS state. The interpretation of User states is beyond the scope of the SSCS protocol. The concept of User state does not apply to N*64 kbit/s services for N > 1 but does apply to 64 kbit/s services.

## 4 Abbreviations

This Recommendation uses the following abbreviations:

| | |
|---|---|
| AAL | ATM Adaptation Layer |
| AIS | Alarm Indication Signal |
| AMR | Adaptive Multi-Rate |
| CAS | Channel-Associated Signalling |
| CEP | Connection End-Point |
| CID | Channel Identifier |
| CPS | Common Part Sublayer |
| CRC | Cyclic Redundancy Check |
| DTMF | Dual-Tone Multi-Frequency |
| EDU | Encoding Data Unit |
| EPT | Echo Protection Tone |
| HDLC | High-level Data Link Control |
| LI | Length Indicator |
| MF-R1 | Multi-Frequency tones for signalling system R1 |
| MF-R2 | Multi-Frequency tones for signalling system R2 |
| MSC | Mobile Switching Centre |
| NT | Network Termination |
| PA | Protocol Analysis |
| PDU | Protocol Data Unit |
| RAI | Remote Alarm Indication |
| RAN | Radio Access Network |
| RDI | Remote Defect Indication |
| SAP | Service Access Point |
| SDU | Service Data Unit |
| SID | Silence Insertion Descriptor |
| SSCS | Service-Specific Convergence Sublayer |
| UUI | User-to-User Indication |
| WA | Waveform Analysis |

## 5 Conventions

This Recommendation adopts the conventions of 2.1/I.361 regarding the order and significance of bits within a field, whether contained within a single octet or spanning more than one octet.

Statements are made herein about the desired behaviour of Users of the SSCS, e.g. the signal processing functions that control audio encoding and decoding. The purpose of these statements is to motivate and explain SSCS features and to encourage interoperable use of services provided by the SSCS. User behaviour is not a normative part of the specification of the SSCS protocol.

## 6 Reference model

Figure 6-1 is the reference model of the SSCS.

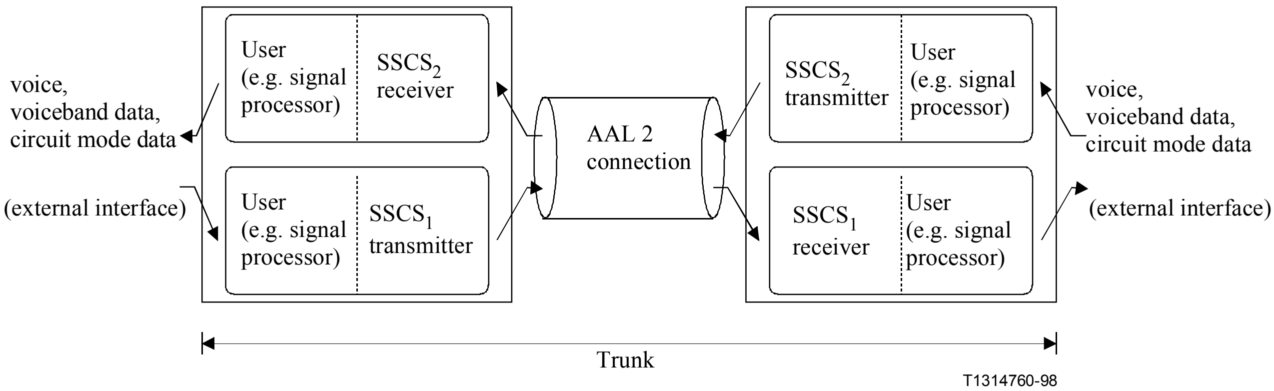



**Figure 6-1/I.366.2 – Reference model of the SSCS for narrow-band services**

The intended use of the SSCS is to carry the information content of one narrow-band call over each AAL type 2 connection – with the bearer capability indicating voice, voiceband data, circuit mode data or frame mode data – as determined by the external input. But secondary messaging, such as dialled digits, channel-associated signalling bits, alarms and loopback may be interleaved on the same AAL type 2 connection.

NOTE – If frame mode data is presented alone at an external interface, not in combination with other information like audio, it can be handled directly by the segmentation SSCS defined in ITU-T I.366.1.

The encoding of the information content can vary dynamically during a connection via User state change, rate control or signalling. For example, upon detecting voiceband data traffic, e.g. 2100 Hz tone, the encoding rate may be increased from its nominal to a higher rate in order to accommodate the voiceband data. If facsimile traffic is detected and facsimile demodulation is supported, the encoding may be switched to facsimile demodulation.

The SSCS directly supports the transport of channel-associated signalling bits and makes available frame mode data for the transport of common-channel signalling messages.

At the access interfaces on either end of a trunk, external inputs could come from private narrow-band switching systems implemented using ISDN or analogue technology. Figures 6-2a to 6-2c give examples of how this Recommendation might be deployed.

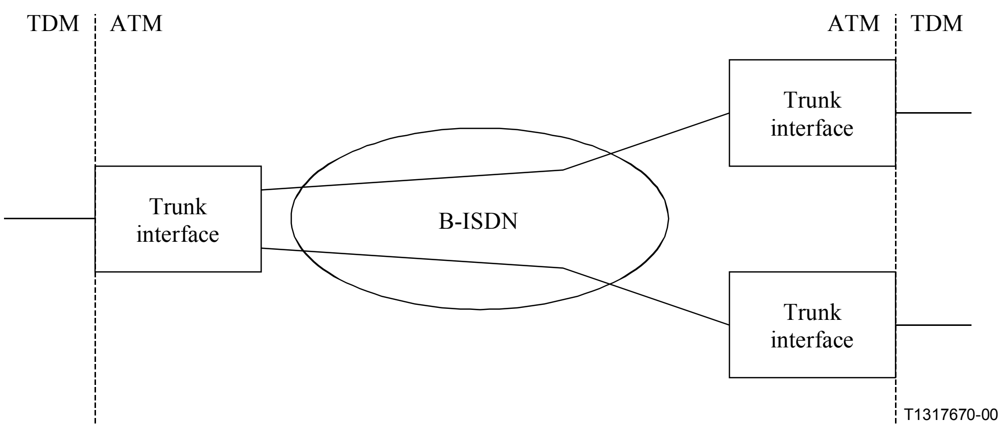



**Figure 6-2a/I.366.2 – Example deployment of narrow-band trunking**

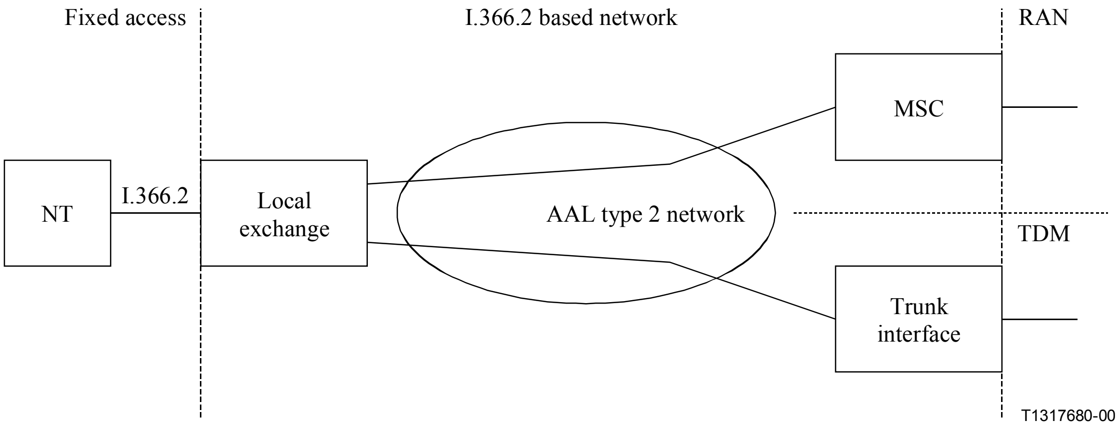



**Figure 6-2b/I.366.2 – Example deployment of narrow-band trunking**

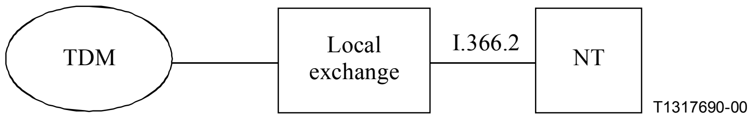



**Figure 6-2c/I.366.2 – Example deployment for fixed access**

ATM connections that are used for AAL type 2 trunking could be either switched or permanent VCs. Depending on the specific application, the mapping of narrow-band information streams (e.g. time slots on a primary rate interface) to AAL type 2 connections could be either static or dynamic. Narrow-band connections could even be switched at a trunk interface to one of several outgoing ATM connections, based on an analysis of the destination address. These are possible applications of the SSCS and are outside the scope of this Recommendation.

## 7 Functional description

At each end of an AAL type 2 connection, the operations of the SSCS are coordinated by the User, e.g. signal processing. As shown in the reference model, these are distinct entities. This clause defines the boundary between them.

### 7.1 Transmitter functions

The following functions, if supported, are considered a responsibility of the User at the transmitter:

a) Encoding of audio samples into a sequence of bits.

b) Selection of audio encoding algorithm based on the characteristics of a call and resource conditions, e.g. congestion indications.

c) Silence compression by voice activity detection and the discontinuous transmission of silence insertion descriptors.

d) Pass-through of circuit mode data as one 8 kHz octet stream per time slot.

e) Extraction of data frames and the removal of flags, bit stuffing, and CRC, if relevant.

f) Detection and preferential treatment of facsimile and modem traffic, e.g. higher-fidelity encoding.

g) Extraction of dialled digit codes from multifrequency tones.

h) Extraction of channel-associated signalling bits and analysis of their transitions.

i) Demodulation of facsimile into baseband bits for page control and image data.
j) Detection of alarms.
k) Synchronized transfer of processed signals to the SSCS.
l) Requests and responses of user state control operations.
m) Rate control.
n) Synchronization of change in SSCS operation.
o) Loopback.

These corresponding functions, if supported, are a responsibility of the SSCS at the transmitter:

a) Insertion of encoded audio bits into a packet structure.
b) Indication of the algorithm used through fields of the packet header (e.g. UUI codepoint and length indicator) or packet payload.
c) Insertion of SID bits and indication of the SID used, just like any audio algorithm.
d) Insertion of octet streams into a packet structure based on time slots.
e) Segmentation, with error protection, of data frames into a sequence of packets.
f) Insertion of encoded bits for voiceband data and indication of the algorithm used, just like any other audio.
g) Insertion of dialled digit codes into a distinguished packet structure.
h) Insertion of channel associated signalling bit transitions into a distinguished packet structure.
i) Insertion of facsimile baseband bits into packet structures distinguished for this purpose.
j) Insertion of alarms into a distinguished packet structure.
k) Sequence numbering of packets to assist isochronous reconstruction of information streams at the receiver.
l) Generation of user state control messages.
m) Insertion of rate control commands into a distinguished packet structure.
n) Insertion of synchronization of change in SSCS operation into a distinguished packet structure.
o) Insertion of loopback into a distinguished packet structure.

### 7.2 Receiver functions

The following functions, if supported, are a responsibility of the SSCS at the receiver:

a) Identification of incoming packet types, determined by fields of the packet header or packet payload.
b) Buffering of time-sensitive packets to reduce delay variation (build-out for dejittering).
c) Attention to sequence numbers in the timely release of packet contents to the User, e.g. discard of late packets.
d) Extraction of algorithm identification and encoded audio bits from packet structure.
e) Indication of any unrecoverable gaps in the bit stream.
f) Extraction of octet streams from a packet structure based on time slots.
g) Reassembly, with error detection, of data frames from a sequence of packets.
h) Extraction of dialled digits codes.
i) Extraction of channel-associated signalling bit transitions.
j) Extraction of facsimile baseband bits.

k) Extraction of alarms.

l) Interpretation of user state control messages.

m) Extraction of rate control commands from packet structure.

n) Extraction of synchronization of change in SSCS operation from packet structure.

o) Extraction of loopback from packet structure.

These corresponding functions, if supported, are considered a responsibility of the User at the receiver:

a) Recognition of the encodings applied to an information stream.

b) Removal of any delay variation introduced by User decoding.

c) Synchronized transfer of encoded information from the SSCS.

d) Decoding of audio bits into a sequence of audio samples, including comfort noise generation as directed by silence insertion descriptors.

e) Attempting to mask the error perceptually if expected audio bits are missing.

f) Regeneration of circuit mode data as one 8 kHz octet stream per time slot.

g) Regeneration of data frames and the restoration of flags, bit stuffing, and CRC, if relevant.

h) Regeneration of multifrequency tones from the dialled digit codes.

i) Regeneration of channel-associated signalling from the bit transitions.

j) Remodulation of facsimile from the baseband bits.

k) Interpretation of alarms.

l) Indications and confirms of user state control operations.

m) Rate control.

n) Synchronization of change in SSCS operation.

## 8 Services provided

The services offered by the SSCS are delivered through two Service Access Points (SAPs) as shown in Figure 8-1.

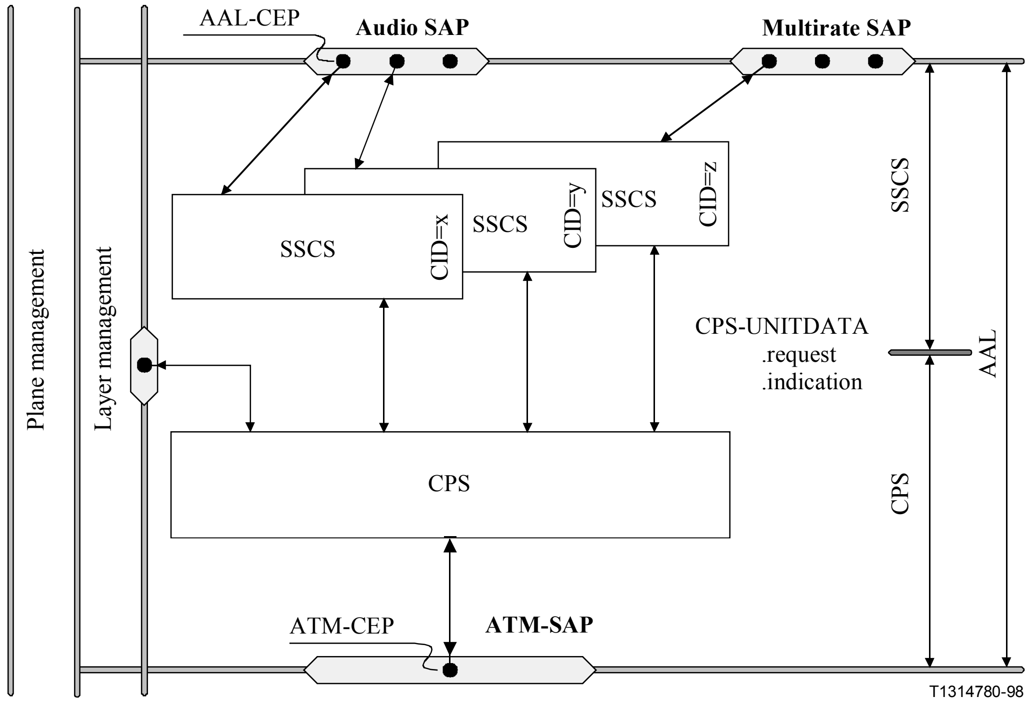



**Figure 8-1/I.366.2 – Functional model of the AAL type 2 sublayers**

The Audio SAP delivers 64 kbit/s services, with audio being the default. The Multirate SAP delivers circuit mode N*64 kbit/s services, N ≥ 1. The services delivered at each SAP are shown in Table 8-1.

**Table 8-1/I.366.2 – Services delivered at the two SSCS SAPs**

| Service category | Services delivered | Mandatory/Optional |
|---|---|---|
| Audio service category (through Audio SAP) | Audio | C |
| | Circuit mode data for 64 kbit/s only | O |
| | Frame mode data | O |
| | Dialled digits | O |
| | Channel-associated signalling | O |
| | Facsimile demodulation/remodulation | O |
| | Alarms | M |
| | State control | O |
| | Rate control | O |
| | Synchronization of change in SSCS operation | O |
| | Loopback | O |
| Multirate service category (through Multirate SAP) | Circuit mode data for N*64 kbit/s, N ≥ 1 | C |
| | Frame mode data | O |
| | Alarms | M |
| | Loopback | O |

**Table 8-1/I.366.2 – Services delivered at the two SSCS SAPs *(concluded)***

| | |
|---|---|
| M | Mandatory |
| C | Conditional |
| O | Optional |
| NOTE 1 – Either one service category or the other or both shall be implemented. The Alarms service is mandatory in all cases. | |
| NOTE 2 – If the audio service category is implemented, the audio service is mandatory. There is no requirement to implement specific encoding algorithms, except for the mandatory profile specified in 13.4. | |
| NOTE 3 – If the multirate service category is implemented, the circuit mode data service is mandatory. There is no requirement to implement this for any specific value of N. | |

To implement each service, primitives are passed through the SAP. The primitives and their parameters are described, by service, in the clauses that follow.

The audio, circuit mode data, and facsimile demodulation/remodulation services represent primary information streams of the audio service. Only one of these streams can be transported on an AAL type 2 connection at a given time. The primary information stream is determined by the SSCS state, which is set as described under the state control service.

The dialled digits service is a secondary information stream. It could be transported simultaneously with one of the primary streams, but it is anticipated that the primary stream will be made idle during the transport of dialled digits. Channel-associated signalling, rate control, synchronization of change in SSCS operation, alarms and loopback services are secondary information streams that can be transported simultaneously with one of the primary information streams.

### 8.1 Audio service

The service provided is the transfer of audio signals (voice, voiceband data, and facsimile).

The data unit traversing the SAP (SSCS SDU) contains either a voice encoding or a silence encoding. Silences are conveyed across the SAP either explicitly by a Silence Insertion Descriptor (SID) data unit, or implicitly by a Null data unit. Supplementary information about the semantic content of the data unit accompanies the transfer of each non-null data unit across the SAP. This specifies the encoding algorithm for voice signals and the generic or algorithm-specific SID for silence, allowing the User to instantaneously alter the algorithm in response to changing characteristics of a call and resource conditions.

Only non-null data units are sent by the transmitting SSCS entity to the receiving SSCS entity. The receiving SSCS entity is able to regenerate the missing null data units. The audio service is a real-time service – the temporal spacing between any two consecutive data units at the transmit SAP is reproduced by the SSCS at the receive SAP.

The SSCS provides no error protection for the data unit itself and error detection for the supplementary information.

The primitives are **Audio** *Request* and *Indication*. The next primitive is transferred across the SAP *k* ms after the last transfer, where *k* is temporal duration associated with the data unit of the primitive.

There is a basic clock associated with the Audio SAP which is used to define the duration of the *k* ms interval. The service is synchronous in the sense that the SSCS transmitting and receiving entities are frequency-locked to a common clock or to separate clocks each of which is traceable to a Primary Reference Source. The clock used by the transmitting User is defined by the transmitting SSCS entity.

The parameters for both primitives are as shown in Table 8-2:

- **Service Data Unit**: The data unit consists of the audio signal appropriately encoded and formatted, consisting of one or more EDUs, as determined by the adopted profile. The data unit can be Null, which is an implicit indication of silence.
- **Data Type**: This parameter provides the semantic information needed by the receiving User to interpret the content of the data unit. Example parameter values are 64 kbit/s G.711 A-law, 16 kbit/s G.728, 12.8 kbit/s G.728, Generic SID, 8 kbit/s G.729, G.729 SID, Null.

**Table 8-2/I.366.2 – Primitives and parameters of the audio service**

| Parameter | Audio request | Audio indication |
|---|---|---|
| Service Data Unit | m | m |
| Data Type | m | m |
| m mandatory | | |

The **Audio** primitives apply only at the Audio SAP.

### 8.2 Circuit mode data service

The service provided is the transfer of circuit mode data.

The circuit mode data service is an emulation of the N*64 kbit/s (N = 1, 2, ..., 30) unrestricted, 8 kHz structured circuit mode service of ITU-T I.231.1 and ITU-T I.231.10. The value N = 31 is also allowed. The value of N does not change during the lifetime of a connection.

When the receiving SSCS entity discovers a momentary gap in the incoming data units (resulting from lost or excessively delayed packets), it substitutes a Null data unit and delivers it to the User. The appropriate fill pattern, if any, to cover the gap is application-specific and is the responsibility of some layer above the SSCS.

The SSCS provides no error protection for the data unit.

The primitives are **Circuit_Mode** *Request* and *Indication*. The next primitive is transferred across the SAP *k* ms after the last transfer, where *k* is temporal duration associated with the data unit of the primitive. The value of *k* does not change during the lifetime of the connection.

There is a basic clock associated with the SAP at which this service is offered, and it is used to define the duration of the *k* ms interval. The service is synchronous in the sense that the SSCS transmitting and receiving entities are frequency-locked to a common clock or to separate clocks each of which is traceable to a Primary Reference Source. The clock used by the transmitting User is defined by the transmitting SSCS entity.

The parameters for both primitives are as shown in Table 8-3:

- **Service Data Unit**: The data unit consists of P octets, where P is an integral multiple of N. The data unit can be Null for the Indication primitive, in which case it indicates a gap in the data stream resulting from an SSCS error condition.

**Table 8-3/I.366.2 – Primitives and parameters of the circuit mode data service**

| Parameter | Circuit_Mode request | Circuit_Mode indication |
|---|---|---|
| Service Data Unit | m | m |
| m mandatory | | |

## 8.3 Frame mode data service

The service provided is the transport of data units as specified in the Transmission Error Detection service of ITU-T I.366.1. There is no assurance of data unit delivery, but if delivered, the relative sequence and the bit integrity of data units are assured. This service corresponds to the service provided by the Common Part of AAL type 5, ITU-T I.363.5, except that the option of corrupted data delivery is not available.

The primitives are **Frame_Mode** *Request* and *Indication* with the following parameter as shown in Table 8-4:

- **Info**: User data up to a maximum of 65 535 octets.

**Table 8-4/I.366.2 – Primitives and parameters of the frame mode data service**

| Parameter | Frame_Mode request | Frame_Mode indication |
|---|---|---|
| Info | m | m |
| m mandatory | | |

## 8.4 Dialled digits service

The service provided is the transfer of dialled digits, namely, the transfer of five descriptors: digit type, digit character, start time, end time, and power level of the dual frequency pulse.

The start time and end time are not conveyed explicitly as a parameter within a primitive across the SAP; rather, each is indicated implicitly by the time instant at which the primitive is conveyed across the SAP. The start time and end time are transferred by the SSCS with an accuracy of 1 ms.

The range of power levels transferred by the SSCS is –31 to 0 dBm0; values outside this range are clipped by the transmitting user.

The primitives are **Dialled_Digits** *Request* and *Indication* with the following parameters as shown in Table 8-5:

- **Digit Type**: DTMF, Signalling System R1, Signalling System R2;
- **Character**: 0,1,2,3,4,5,6,7,8,9,*,#,A,B,C,D, and Tone-off for DTMF
  0,1,2,3,4,5,6,7,8,9,KP,ST, and Tone-off for Signalling System R1
  1,2,3,4,5,6,7,8,9,10,11,12,13,14,15, and Tone-off for Signalling System R2;
- **Power Level**: −31, −30, …, −1,0 dBm0.

The start time of the dual frequency pulse is indicated by a primitive containing three parameters: Digit Type, Character and Power Level. The end time is indicated by a primitive containing the Tone-off character.

**Table 8-5/I.366.2 – Primitives and parameters of the dialled digits service**

| Parameter | Dialled_Digits request | Dialled_Digits indication |
|---|---|---|
| Digit Type | m | m |
| Character | m | m |
| Power Level | m | m |
| m mandatory | | |

The **Dialled_Digits** primitives apply only at the Audio SAP.

### 8.5 Channel-associated signalling service

The service provided is the transfer of channel-associated signalling information, i.e. the transfer of ABCD bits. Typically, the value of the (A,B,C,D) vector does not change for extended periods of time. In order to significantly improve transmission efficiency, the SSCS transmitting entity identifies such quiescent periods during which it transmits only refresh information to the receiving SSCS entity. The User perceives continuous service, i.e. an (A,B,C,D) vector is transferred across the SAP with exact periodicity.

The service is assured in the sense that the SSCS provides error detection capability and, during periods when the signalling bits are changing, transfers the (A,B,C,D) vector three times to achieve forward error correction.

The primitives are **CAS** *Request* and *Indication*. A primitive is transferred across the SAP every 2 ms or every 3 ms. The clock used to derive the 2- or 3-ms interval is the basic clock used at the Audio SAP. Both primitives have one parameter as shown in Table 8-6:

- **Bit Vector**: A vector (A,B,C,D) consisting of four bits.

**Table 8-6/I.366.2 – Primitives and parameters of the channel associated signalling service**

| Parameter | CAS request | CAS indication |
|---|---|---|
| Bit Vector | m | m |
| m mandatory | | |

The **CAS** primitives apply only at the Audio SAP.

### 8.6 Facsimile demodulation/remodulation service

The service provided is the transfer of demodulated facsimile image information and control information, from the demodulating User to the remodulating User.

The demodulating User passes a block of image information to the SSCS nominally every 20 ms. The SSCS conveys this block to the remodulating user without error protection. The SSCS compensates for packet delay variation by using sequence numbers, so that successive transfers of image information experience equal delay.

For control information, the SSCS transfer service is assured in the sense that the SSCS provides bit error detection, transfers information three times to achieve forward error correction, and identifies (unrecoverable) gaps in T.30 data transfer caused by packet loss. The SSCS also compensates for the packet delay variation by using time stamps, so that successive transfers of control information experience equal delay.

There are two primitives: **Fax_Demod** *Request* and *Indication*. Both primitives have at least two parameters as shown in Table 8-7. The first parameter is:

- **Info type**: Image, Control.

For image information, there is exactly one accompanying parameter:

- **Image Data**: 6, 12, 18, 24, 30, 36 octets.

For control information, there is at least one accompanying parameter:

- **Control type**: T.30 Preamble, EPT (Echo Protection Tone), Training, Fax Idle, T.30 Data.

Accompanying parameters for each control type are as follows.

T.30 Preamble and Fax Idle have no accompanying parameters.

EPT is accompanied by the parameter:

- **EPT Frequency**: 1700 Hz, 1800 Hz.

Training is accompanied by the two parameters:

- **Modulation Type**: V.27 *ter*, V.29, V.17 long training, V.17 short training, V.33.
- **Modulation Rate**: Unknown, 2400, 4800, 7200, 9600, 12 000, 14 400 bit/s.

T.30 Data is accompanied by the two parameters:

- **Data Framing**: Continue, End.
- **Data Bits**: N bit values.

  N = 8 if Continue.

  $1 \leq N \leq 8$ if End.

**Table 8-7/I.366.2 – Primitives and parameters of the facsimile demodulation/remodulation service**

| Parameter | Fax_Demod request | Fax_Demod indication |
|---|---|---|
| Info Type | m | m |
| Image Data | c | c |
| Control Type | c | c |
| EPT Frequency | c | c |
| Modulation Type | c | c |
| Modulation Rate | c | c |
| Data Framing | c | c |
| Data Bits | c | c |
| m mandatory<br>c conditional (see 8.6) | | |

The **Fax_Demod** primitives apply only at the Audio SAP.

### 8.7 Alarms service

The service provided is the transfer of external and internal alarm indications between the two peer Users. The primitives are **Alarm** *Request* and *Indication*. Both have two parameters as shown in Table 8-8:

- **Alarm Type**: External AIS, External RAI, AAL type 2 connection AIS, AAL type 2 connection RDI.
- **Alarm Status**: On, Off.

**Table 8-8/I.366.2 – Primitives and parameters of the alarms service**

| Parameter | Alarm request | Alarm indication |
|---|---|---|
| Alarm Type | m | m |
| Alarm Status | m | m |
| m mandatory | | |

## 8.8 State control service

The service provided is the transfer of User state information between the two peer Users. The service is assured in the sense that the SSCS provides error detection and transfers information three times to achieve forward error correction.

The primitives are **State_Control** *Request*, *Indication*, *Response* and *Confirm*. These contain at least one parameter as shown in Table 8-9:

- **User State**: Voice, Voiceband Data, Circuit Mode, Facsimile Demodulation.

If the User State is Facsimile Demodulation, the subset of supported modulations is reported through the additional parameter:

- **Modulations**: zero or more of V.17, V.27 *ter*, V.29, V.33.

Each User asserts its own demodulation/remodulation capabilities in the respective Request or Response. Its peer's capabilities are received in the associated Confirm or Indication. Each User thereby obtains enough information to compute which modulations are common to both.

In addition, the Response and Confirm primitives also contain the parameter:

- **Ack**: Accept, Reject.

In addition, the User can set the transmit or receive state of the local SSCS. The primitive is **Set_SSCS_State** *Request* with the parameters as shown in Table 8-9:

- **Direction**: Transmit, Receive.
- **SSCS State**: Audio, Circuit Mode, Facsimile Demodulation.

**Table 8-9/I.366.2 – Primitives and parameters of the state control service**

| Parameter | State_Control request | State_Control indication | State_Control response | State_Control confirm | Set_SSCS_State request |
|---|---|---|---|---|---|
| User State | m | m | m | m | – |
| Modulations | c | c | c | c | – |
| Ack | – | – | m | m | – |
| Direction | – | – | – | – | m |
| SSCS State | – | – | – | – | m |
| m mandatory<br>c conditional (see 8.8)<br>– absent | | | | | |

The **State_Control** and **Set_SSCS_State** primitives apply only at the Audio SAP.

The initial User state is Voice, and the initial SSCS state is Audio.

## 8.9 Rate control service

The service provided is the transfer of requests from one SSCS user to its peer to operate using an indicated set of entries of the profile agreed for the connection.

The primitives are **Rate_Control** *Request* and *Indication* with the following parameter shown in Table 8.10:

- **Profile_Entry_Index**: index of a profile entry.

**Table 8-10/I.366.2 – Primitives and parameters of the rate control service**

| Parameter | Rate_Control request | Rate_Control indication |
|---|---|---|
| Profile_Entry_Index | m | m |

### 8.10 Change SSCS operation service

The service provided is the transfer of requests from one SSCS user to its peer to modify the SSCS attributes (e.g. profile number, DTMF support, etc).

The primitives are **SSCS_Change** *Request* and *Indication* with the following parameter shown in Table 8.11:

- **Correlation_Identifier**: 1 octet.

**Table 8-11/I.366.2 – Primitives and parameters of the SSCS change synchronization service**

| Parameter | SSCS_Change_Synchro request | SSCS_Change_Synchro indication |
|---|---|---|
| Correlation_Identifier | m | m |

### 8.11 Loopback service

The service provided is the transfer of loopback request from one SSCS user to the remote SSCS entity. The primitives are **Loopback** *Request* and *Response* and do not contain any parameters.

## 9 Means to achieve isochrony

A major influence on the SSCS is the requirement for isochrony – that stimuli occurring at the transmitter should be reproduced with the same interval between them at the receiver. Equivalently, the end-to-end delay of the information stream should be a constant.

This is important for voiceband data, because modems will sense abnormal phase shifts if the delay varies. It is also important for voice. Annoying distortions can result if the end-to-end delay varies from one stimulus to another and brief silence periods are thereby shortened or lengthened. This is of particular concern with newer algorithms like G.723.1 and G.729, because they contain a bit-exact specification of silence compression, which allows no explicit control over the parameters of voice activity detection and hangover time (minimal duration of silence).

Because the information streams are packetized for transport, isochrony rests on correct scheduling of the playout time for each packet. To keep the end-to-end delay constant, a receiver must have enough timing information to remove packet delay variation, up to the maximum anticipated for an AAL type 2 connection.

### 9.1 Delay variation in user processing

The primitives of clause 8 adopt a synchronous model of User and SSCS interaction. In this model, an audio primitive is transferred across the SAP *k* ms after the last transfer, where *k* is the temporal duration associated with the data unit of the primitive. The same principle applies to circuit mode data and facsimile image data. The SSCS takes responsibility for generating sequence numbers for these packets or time stamps in other control packets, based on the time that the User requests a transmission by invoking the corresponding primitive.

This model requires the transmitting User to compensate for and remove any delay variation due to its own actions in processing different units of information. Variation can arise in audio encodings, as an example, because algorithms may operate with different frame sizes and look-aheads and algorithmic complexity may require different computing times. Variation can also arise in facsimile demodulation, due to the time it takes to acquire and analyse different incoming signals during the successive phases of a connection.

The method available to a User (transmitting or receiving) to eliminate such variations is to insert extra delay in those operations that complete quickly – exactly enough delay to bring its processing of each real-time information stream to a constant maximum delay.

In order to maintain a constant delay, the User must anticipate the kinds of operations it will invoke. Bounds can be set on the extent of processing variations by the encoding profile in effect for audio and by the range of other options employed, such as facsimile demodulation. These are parameters of the SSCS that shall be agreed for both directions of communication and should be known to the User.

Constant delay applies to both the transmitting and the receiving User. Each one best understands its own processing elements and can insert a precise amount of delay where needed. This makes good use of the isochrony that the SSCS provides. It is a more efficient approach than the alternative of permitting delay variation in the transmitting User and compensating for it with extra build-out at the receiving User.

### 9.2 Delay variation below the SSCS sublayer

An SSCS transmitter passes information from its User to the Common Part Sublayer (CPS) with no delay variation.

In the receiving direction, the SSCS introduces calculated packet delays, through extra build-out and analysis of sequence numbers or time stamps, whose effect is to cancel delay variation incurred by transport through the CPS and thereby provide an isochronous service to the User.

In addition to the cell delay variation of the underlying ATM connection that leads to packet delay variation, the other major factor in packet delay variation is queuing at the AAL type 2 CPS transmitter during periods of talk spurt overload, i.e. when voice, instead of silence, from too many AAL type 2 connections is directed simultaneously onto the same ATM connection. If not controlled through effective policies of connection admission, the queuing effect can easily lead to packet delay variation that greatly exceed the cell delay variation.

Packet queuing can be reduced if feedback is given to the User and the encoding format profile permits a rapid switch to algorithms that provide greater compression during periods of congestion. However, this entails some loss of fidelity; it applies to many algorithms but not to all; and it cannot be expected to eliminate every overload, e.g. a burst of modem traffic.

When operating in a regime where packet delay variation is a significant factor, sequence numbering of packets can help an SSCS receiver to detect and to recover from anomalies – lost, early, or late packets – and to make fewer reconstruction errors than if sequence numbers were absent or ignored.

The use of sequence numbers is predicated on the assumption that the basic units of packetization are generated at some fixed frequency by the transmitter, which is known to the receiver, and that the frequency is reflected in the way that sequence numbers are incremented. If these conditions are true, sequence numbers can be interpreted as relative time stamps, with a limited resolution and a limited range before wrapping around.

NOTE 1 – Some studies suggest that for discontinuous transmission, i.e. using silence compression, the modulus of sequence numbering should be related to the receiver's build-out delay for dejitter buffering plus a targeted quantile of the packet delay variation, this sum divided by the sequence number interval. Appendix II contains a simple derivation of this guideline.

Where the SSCS specifies that a given field represents a sequence number, it is mandatory for the transmitter to supply values that are incremented correctly according to clause 14. It is optional for a receiver to act on sequence numbers, and the algorithms that it may use are not standardized.

For example, if a silence interval is short, a receiver may use sequence numbers to position the start of the next talk spurt accurately in relation to the preceding talk spurt. But if it considers the silence interval to be sufficiently long, it may choose instead to reinitiate build-out.

Despite the additional margin created by a proper use of sequence numbers, congestion caused by too many simultaneous talk spurts should still be avoided, either by limiting the number of contributing Users (connection admission control) or by increasing compression (shifting to a lower rate encoding) at the onset of congestion.

NOTE 2 – The detection and treatment of congestion and its notification to the User are not specified in this Recommendation.

## 10 Packet format types

Protocol data units of the SSCS are transported as CPS-Packets over one AAL type 2 connection using the primitives and parameters defined in Table 2/I.363.2.

The SSCS makes explicit use of the CPS-UUI field and implicit use of the Length Indicator in the CPS-Packet header. CPS-INFO, the payload, is variable length up to a maximum of 45 octets.

The CPS-Packet header provides error control over all its fields, including UUI and LI, but the CPS-Packet payload has no built-in protection. By specifying additional error control over some or all of a payload, the SSCS defines three packet types.

Frame mode data uses the packet format of the SSTED-PDU defined in 8.3/I.366.1.

### 10.1 Type 1 – Unprotected

Figure 10-1 defines the format of packet type 1. The payload is unprotected. This format type is used by default unless an alternative type is explicitly specified in this Recommendation.

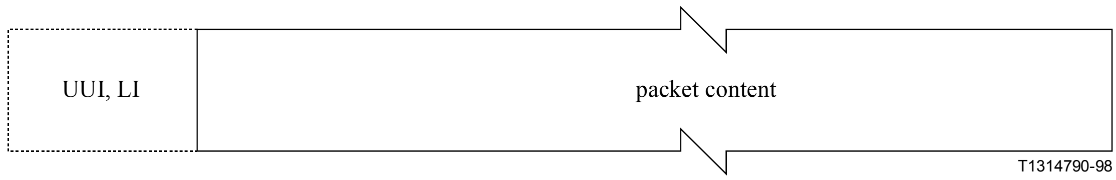
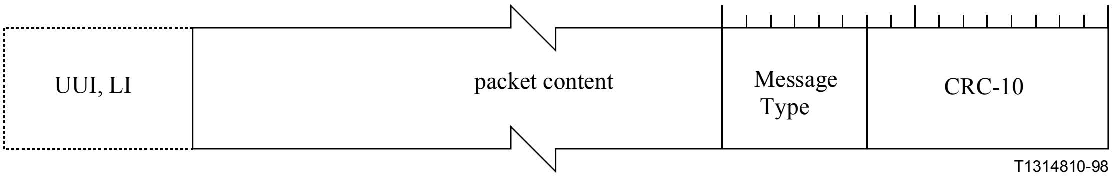



**Figure 10-1/I.366.2 – Packet format type 1 – Unprotected**

### 10.2 Type 3 – Fully protected

Figure 10-2 defines the format of packet type 3. The entire payload is protected by a 10-bit CRC.




Message Type 6 bits
CRC-10 10 bits

**Figure 10-2/I.366.2 – Packet format type 3 – Fully protected**

The CRC-10 field is computed as for OAM cells, as defined in 7.1/I.610, using the polynomial $x^{10} + x^9 + x^5 + x^4 + x + 1$.

The remaining 6 bits of the two-octet trailer constitute a Message Type field.

Type 3 packets are used for the following information streams:

- Dialled digits.
- Channel associated signalling bits.
- Facsimile demodulation control data.
- Alarms.
- User state control operations.
- Rate control.
- Synchronization of change in SSCS operation.
- Loopback.

The Message Type field is coded according to Table 10-1.

**Table 10-1/I.366.2 – Message Type codes for packet format type 3**

| Information stream | Message Type code | Packet format | Reference |
|---|---|---|---|
| Dialled digits | 000010 | Dialled digits | Figure K.1 |
| Channel associated signalling | 000011 | CAS bits | Figure L.1 |
| Facsimile demodulation control | 100000 | T.30_Preamble | Figure M.1 |
| | 100001 | EPT | Figure M.2 |
| | 100010 | Training | Figure M.3 |
| | 100011 | Fax_Idle | Figure M.4 |
| | 100100 | T.30_Data | Figure M.5 |
| OAM | 000000 | Alarm | Figure N.1 |
| | | Loopback | Figure N.2 |
| User state control | 000001 | User state control | Figure O.1 |
| Rate control | 000100 | Rate control | Figure R.1 |
| Synchronization of change in SSCS operation | 000101 | Synchronization of change in SSCS operation | Figure S.1 |

## 11 Common facilities for type 3 packets

Some, not all, type 3 packets share the additional structure shown in Figure 11-1. It applies to dialled digits, channel-associated signalling bits, facsimile demodulation control, and user state control packets. Alarms are patterned on OAM cells and do not use the common facilities for type 3 packets.

| 8 | 7 | 6 | 5 | 4 | 3 | 2 | 1 |
|---|---|---|---|---|---|---|---|
| Redundancy | | Time Stamp | | | | | |
| | | | | | | | |
| Message-dependent information | | | | | | | |
| ... | | | | | | | |
| Message Type = xxxxxx | | | | | | | |
| CRC-10 | | | | | | | |

**Figure 11-1/I.366.2 – Common facilities for type 3 packets**

The message-dependent information in Figure 11-1 represents packet content that varies, depending on the Message Type. It is not part of the common facilities.

### 11.1 Relative event timing

The Time Stamp field serves to counter packet delay variation and allows a receiver to reproduce accurately the relative timing of successive events that are separated by a short interval. Events that are separated by a long interval, e.g. many times the maximum packet delay variation, do not normally require precise timing.

The Time Stamp field is 14 bits. The most significant bit is bit 6 of the first octet, and the least significant is bit 1 of the second octet. The transmitter begins time stamping at an arbitrary value and increments by one every millisecond. After reaching the maximum unsigned count, the time stamp wraps around to zero again. A full cycle takes slightly less than 16.4 seconds.

Having received two type 3 packets that designate events, E1 and E2, with respective time stamps TS1 and TS2, a receiver should decide whether the interval between reception of the packets is short enough to require precise timing of the events. If so, the receiver shall schedule their play-out times, PT1 and PT2, so that PT2 – TS2 = PT1 – TS1.

### 11.2 Triple redundancy and refresh

The common facility for type 3 packets requiring error correction is triply redundant transmission. Such packets are sent three times, with a fixed interval between transmissions.

The redundancy interval depends on the information stream. It is 5 ms for dialled digits and channel associated signalling bits. It is 20 ms for facsimile demodulation control packets and user state control packets.

Each copy of a redundant packet contains the same content, except in the Redundancy field. The three copies of a packet can be correlated because they all have the same time stamp.

The Redundancy field is set to values 0, 1 and 2 respectively for a packet's first, second, and third transmission under triple redundancy.

Redundancy value 3 indicates no use of triple redundancy, whereby some messages with the same format are sent singly, as specified in the corresponding annex referenced by Table 10-1. One use of this is for a long-term refresh of state information, such as the values of the CAS bits for an AAL type 2 connection. These messages may occur periodically, but at a much longer interval. The receiver shall not expect three copies of them to be spaced at the redundancy interval.

NOTE – It is for further study whether an option should be provided to turn off triple redundancy when AAL type 2 transport is known to be operating with negligible error and loss rates.

## 12 UUI codepoint assignments

Table 12-1 defines how the SSCS uses UUI codepoints.

**Table 12-1/I.366.2 – UUI codepoint assignments**

| UUI codepoints | Packet content | Reference |
|---|---|---|
| 0-15 (Note 1) | Encoding formats for audio, circuit mode data, and demodulated facsimile image data using type 1 packets (Notes 2 and 3) | Annexes A-I, Annex J, Annex M |
| 16-23 | Reserved for future assignment | – |
| 24 | Type 3 packets, except OAM packets | Clause 10.2 |
| 25 | Non-standard extension (Note 4) | – |
| 26 | Framed mode data, final packet | Clause 16 |
| 27 | Framed mode data, more to come | Clause 16 |
| 28-30 | Reserved (ITU-T I.363.2) | – |
| 31 | OAM packets | Annex N |

NOTE 1 – The least significant bits of UUI codepoints 0-15 may be used for sequence numbering. The number of bits used for this purpose depends on the profile of encoding formats, as explained in clause 14.

NOTE 2 – For audio, the profile of encoding formats is an SSCS parameter of operation to be agreed between the transmitter and the receiver. Annexes A through I define how the bits output by ITU-T audio algorithms are formatted into type 1 packets. These annexes are referenced by the predefined profiles of Annex P. They may also be referenced by other custom profiles.

NOTE 3 – Annexes J and M define how circuit mode data and demodulated facsimile image data are formatted into type 1 packets.

NOTE 4 – The codepoint for non-standard extension may be used to encode vendor or operator proprietary features. If the use of a non-standard extension is not understood or not agreed, the receiver shall discard such packets and take no further action.

When the SSCS is accessed through the Audio SAP, the transmission and reception of type 1 packets (UUI codepoints 0-15) is dedicated to one primary information stream at any time – either audio, circuit mode data, or demodulated facsimile image data. This is determined by the SSCS state in effect for that direction. The two directions of an AAL type 2 connection can be set to different states, either transiently or for a sustained period.

When the SSCS is accessed through the Multirate SAP, the transmission and reception of type 1 packets is dedicated exclusively to circuit mode data, using the formats of Annex J.

## 13 Encoding format profiles

### 13.1 The function of a profile

This clause characterizes encoding format profiles for UUI codepoints 0-15. Profiles make reference to audio encoding formats like those defined in the annexes. A specific profile must be an agreed operating parameter between the SSCS transmitter and receiver for both directions of an AAL type 2 connection.

The agreed profile applies only to one AAL type 2 connection. The same or different profiles may be agreed for other AAL type 2 connections.

A profile is a mapping that informs the receiver of a type 1 audio packet how to interpret the packet content.

The domain of this mapping is a set of pairs (UUI, Length). The first element of each pair is a UUI codepoint in the range 0-15. The second element is the packet length for one of the encoding formats included in the profile.

The result of a profile mapping is an explicit packet format plus a value for the sequence number interval. Predefined profiles reference the explicit packet formats defined in Annexes A through I. Custom profiles may have additional methods to define mutually agreed packet formats.

Thus, a profile determines the set of valid pairs (UUI, Length). Pairs that do not occur in the agreed profile are invalid and shall not be transmitted as type 1 packets.

Contiguous UUI codepoints that have the same profile mapping for all respective lengths shall be considered to form a sequence numbering subrange. The lowest UUI codepoint in such a subrange represents sequence number 0 and the remainder represent successive sequence numbers, up to a modulus equal to the size of the subrange. Thus, if a profile specifies that UUI codepoints 0-7 are to have one interpretation and codepoints 8-15 a different interpretation, then the sequence numbering modulus is 8 in each subrange.

For continuity of sequence numbering as the encoding format shifts from one subrange to another, all subranges of a profile shall have the same size and therefore a uniform modulus. Furthermore, this value shall be a power of two. The valid moduli are: 1, 2, 4, 8 and 16.

Each profile is an attempt to balance two contending interests: The number of encodings should be enough to accommodate a significant range of narrow-band traffic types with acceptable efficiency. But the number of encodings should be small enough to allow the use of sequence numbers, unless the operating environment is so benign that sequence numbers are not needed.

The inclusion of an encoding format into an agreed profile constitutes the receiver's permission for the transmitter to select that format dynamically at any time with no further preliminaries. This is another reason to limit the diversity of algorithms in a profile, because the resources of signal processing could be overwhelmed if too many algorithms had to be loaded in advance, not knowing what to expect in the next packet.

NOTE – A change of encoding formats can affect the bandwidth used by an AAL type 2 connection, either increasing or decreasing it. Such variations must be taken into account by connection admission control. However, such matters are outside the scope of this Recommendation.

### 13.2 Relationship of service data unit and sequence number interval

Service data units (SDUs) for audio are defined in relation to the profile of encoding formats adopted on a given AAL type 2 connection. Each algorithm that occurs in a given profile may appear in multiple entries corresponding to packets of different lengths. These packet lengths shall align in a simple sequence, whereby each is an integral multiple of the smallest packet length that occurs for the algorithm (at a given bit rate). The smallest packet length is the SDU of the algorithm, in relation to the given profile.

Any other entry that occurs in the profile for the same algorithm is an integral multiple M of the SDU. The value M = 1 corresponds to the SDU itself. To be well formed, if a profile includes multiple M of the SDU for a particular algorithm, it shall also include multiple M−1. It follows that, for each algorithm, a profile will contain all multiples of its SDU from one up to some maximum value of M.

The example profiles of the following clause and the predefined profiles of Annex P contain, for ease of understanding, a column labelled "M" which indicates the multiples of the SDU for each algorithm that occur in the profile.

Each packet has a sequence number that is incremented by some integral value from the preceding packet, according to the sequence number interval set by the profile entry. The smallest audio packet that can be sent is one SDU. It follows that the time spanned by an SDU is an integral multiple of the sequence number interval.

The essential relationships can be expressed as follows:

Audio Packet = M × SDU, SDU = $N_1$ × EDU, SDU time = $N_2$ × Sequence Number Interval.

There is no direct relationship between the EDU and the sequence number interval.

NOTE – In the example profiles of 13.3, these relationships hold with the values: $N_1 = 5$ for voice encodings, except $N_1 = 1$ for G.729-8, $N_1 = 1$ for SIDs; and $N_2 = 2$ for G.729-8, $N_2 = 1$ otherwise.

### 13.3 Examples of profile structure

This clause provides a small number of informative examples to illustrate some of the possible structures for an encoding format profile.

These examples are not warranted to be generally useful and are not guaranteed to be maintained for future reference. Those needs are met by the predefined profiles of Annex P.

- Example A in Table 13-1 shows a profile that makes maximum use of the UUI field for sequence numbering since the packet length alone provides discrimination between entries.
- Example B in Table 13-2 shows a profile that splits the UUI range in half in order to discriminate between algorithms that use the same packet length.
- Example C in Table 13-3 shows a profile that uses three UUI ranges of 4 each. As a result, it does not use the UUI values 12-15.

The "Profile entry index" in each example is used by the rate control service.

**Table 13-1/I.366.2 – Example profile A**

| Profile entry index | UUI codepoint range | Packet length (octets) | Encoding format reference | Description of algorithm | M | Packet time (ms) | Sequence number interval (ms) |
|---|---|---|---|---|---|---|---|
| 0 | 0-15 | 40 | Figure B.1 | G.711-64 generic | 1 | 5 | 5 |
| 1 | 0-15 | 25 | Figure F.1 | G.727 (5,2) | 1 | 5 | 5 |
| 2 | 0-15 | 20 | Figure F.2 | G.727 (4,2) | 1 | 5 | 5 |
| 3 | 0-15 | 15 | Figure F.3 | G.727 (3,2) | 1 | 5 | 5 |
| 4 | 0-15 | 10 | Figure F.4 | G.727 (2,2) | 1 | 5 | 5 |
| – | 0-15 | 1 | Figure I.1 | Generic SID | 1 | 5 | 5 |

**Table 13-2/I.366.2 – Example profile B**

| Profile entry index | UUI codepoint range | Packet length (octets) | Encoding format reference | Description of algorithm | M | Packet time (ms) | Sequence number interval (ms) |
|---|---|---|---|---|---|---|---|
| 0 | 0-7 | 40 | Figure B.1 | G.711-64 A-law | 1 | 5 | 5 |
| 1 | 0-7 | 35 | Figure B.2 | G.711-56 A-law | 1 | 5 | 5 |
| – | 0-7 | 1 | Figure I.1 | Generic SID | 1 | 5 | 5 |
| 3 | 8-15 | 40 | Figure B.1 | G.711-64 μ-law | 1 | 5 | 5 |
| 4 | 8-15 | 35 | Figure B.2 | G.711-56 μ-law | 1 | 5 | 5 |
| – | 8-15 | 1 | Figure I.1 | Generic SID | 1 | 5 | 5 |

**Table 13-3/I.366.2 – Example profile C**

| Profile entry index | UUI codepoint range | Packet length (octets) | Encoding format reference | Description of algorithm | M | Packet time (ms) | Sequence number interval (ms) |
|---|---|---|---|---|---|---|---|
| 0 | 0-3 | 40 | Figure B.1 | G.711-64 A-law | 1 | 5 | 5 |
| 1 | 0-3 | 35 | Figure B.2 | G.711-56 A-law | 1 | 5 | 5 |
| 2 | 4-7 | 40 | Figure E.2 | G.726-32 | 2 | 10 | 5 |
| 3 | 4-7 | 20 | Figure E.2 | G.726-32 | 1 | 5 | 5 |
| 4 | 8-11 | 40 | Figure H.1 | G.729-8 | 4 | 40 | 5 |
| 5 | 8-11 | 30 | Figure H.1 | G.729-8 | 3 | 30 | 5 |
| 6 | 8-11 | 20 | Figure H.1 | G.729-8 | 2 | 20 | 5 |
| 7 | 8-11 | 10 | Figure H.1 | G.729-8 | 1 | 10 | 5 |

### 13.4 Mandatory profile support

If the Audio service category is implemented, it is a requirement for conformance to this Recommendation that the predefined profile of Table P.1 shall be implemented. It shall be implemented for either the A-law or the μ-law option of generic PCM and may be implemented for both.

## 14 Sequence numbering

### 14.1 Basic principles

Audio encoding packets are accompanied by a sequence number field that is embedded within the UUI codepoint range 0-15 for type 1 packets. It is mandatory for a transmitter to increment the designated bits at the frequency corresponding to the sequence number interval specified in the profile entry that defines the last packet transmitted.

It is optional for a receiver to act on sequence numbers, and the algorithms that it may use are not standardized.

Sequence numbers shall begin at an arbitrary value, such as zero. Within the number of least significant bits designated, sequence numbers shall wrap around from all ones back to zero.

A transmitter shall update and transmit the full field. By extracting from what it receives a smaller number of least significant bits, a receiver can obtain the effect of a reduced modulus, if desired.

The sequence number of a packet shall correspond to the beginning time of the first service data unit in the packet. The time spanned by an audio packet may be greater than the specified sequence number interval. The sequence number of the following packet shall be incremented by a value equal to their ratio.

### 14.2 Embedding in UUI codepoints

How sequence numbers are embedded within the UUI codepoint range 0-15 is one aspect of the adopted profile of encoding formats. Some of the predefined profiles of Annex P, like Example A of Table 13-1, take the simple approach of using this range entirely for sequence numbers modulo 16.

Other profiles may divide 0-15 into subranges, say 0-7 and 8-15, that designate different families of encoding formats within the overall profile. This would be necessary if there were a conflict of packet lengths, so that two subranges were needed to distinguish between two different encoding formats having the same value of LI. That value of LI plus a UUI codepoint 0-7 signifies one format, while the same value of LI plus a UUI codepoint 8-15 signifies a different format. Sequence numbering would be modulo 8 in this case, using the three least significant bits of each UUI codepoint. Shifts between the two encoding families are permitted, from packet to packet, while still maintaining the continuity of sequence numbers in the least significant bits of the UUI.

For continuity of sequence numbering, it is necessary that each UUI subrange contain the same number of codepoints. If the range 0-15 is so divided, the modulus must be a power of two, and only five divisions are possible: (maximum number of subranges, sequence number modulus) = (1, 16), (2, 8), (4, 4), (8, 2), (16, 1). The extreme of modulus 1 corresponds to a suppression of sequence numbering, which is permitted but is not usually advised.

### 14.3 Incrementing during silence

Audio sequence numbers shall increment during periods of silence according to the sequence number interval of the last packet transmitted, which could be either a voice encoding or a silence insertion descriptor. This incrementing, when no packets are being transmitted, maintains sequence numbers as relative time stamps. Audio sequence numbers shall not be reset at the beginning or end of a talk spurt.

The reason for incrementing audio sequence numbers through a silence period is to position the playout of the next talk spurt accurately with respect to the end of the preceding talk spurt. This is a way to eliminate variations in the duration of silence between transmitter and receiver that might otherwise occur. The shortening or lengthening of brief silences, such as between syllables, is likely to be perceived as a distortion in audio quality.

### 14.4 Change of sequence number interval

Switching between different profile entries that use the same sequence number interval has no impact on the continuity of sequence numbering and does not interrupt the benefits. However, if the sequence number interval changes when switching between entries, the meaning of the increment in sequence numbers differs before and after. Some receivers may not be able to reconcile such changes. In this case the benefits of sequence numbering would be lost at such a juncture. The benefits would also be lost if there were any ambiguity in when the interval changed – due to a lost packet, for example. The benefits will be regained if a new interval is maintained for the full modulus.

In general, changes of the sequence number interval within a profile of encoding formats are allowed but should be minimized.

For the non-audio encodings that use type 1 packets – circuit mode data and demodulated facsimile image data – the frequency for incrementing sequence numbers is specified in the corresponding Annex J or Annex M and cannot be varied. In both cases, the sequence number modulus is fixed at 16.

## 15 Circuit mode data

Circuit mode data is digital information at N*64 kbit/s with an 8 kHz structure. The 8 kHz octet timing is derived from a synchronous clock reference signal. The value of N is a parameter that is fixed for the duration of a narrow-band connection and is a parameter of the encoding format.

The entire UUI codepoint range 0-15 shall be used to encode circuit mode data with sequence numbers modulo 16. A large sequence number modulus is an important factor in enhancing the integrity of circuit mode data.

Since the packet stream is continuous, modulus 16 should suffice to counteract anomalies for small values of N. Sequence numbers for circuit mode data need not be large enough to constrain packet delay variations that may span a silence interval, because there are no silence intervals. The only need is to detect late and lost packets.

However, as N grows larger, the inter-packet interval grows shorter. This makes circuit mode data prone to disruption for runs of several successive packets when congestion occurs due to statistical fluctuations in discontinuous traffic sharing the same ATM connection. If a large amount of circuit mode data is to be transported, the capacity of the underlying ATM connections and connection admission policies should be engineered with such demands in mind.

Packet formats for circuit mode data are specified in Annex J.

## 16 Frame mode data

Frame mode data units shall be octet aligned.

NOTE 1 – If flags or other means are used externally to mark the boundaries between data units, they should be removed by the User on input and should be restored on output. If bit stuffing is used externally for flag transparency, the bit stuffing should be removed by the User on input and should be restored on output.

Frame mode data shall use UUI codepoints 26 and 27 as specified in ITU-T I.366.1 to delineate a sequence of packets whose reassembly is the data unit.

The transmission error detection capability defined in ITU-T I.366.1 shall be used for both external frame mode data streams and internally generated logical information streams.

NOTE 2 – Frame mode data units within external information streams are expected to contain their own style of error protection. Their CRC or other such fields should be validated and discarded. The error protection should not be passed through segmentation transparently.

Permission to send frame mode data on a given AAL type 2 connection shall be an SSCS parameter of operation agreed between the transmitter and receiver.

Frame mode data can be carried on an AAL type 2 connection simultaneously with one of the primary information streams (audio, circuit mode data, demodulated facsimile image data).

NOTE 3 – The simultaneous use of frame mode data with a primary information stream will require traffic mechanisms that currently do not exist.

## 17 Facsimile demodulation/remodulation

### 17.1 Functional requirements

Facsimile demodulation and remodulation is a more efficient way to transport facsimile traffic over an AAL type 2 connection.

The basic function of facsimile demodulation is to detect facsimile traffic, demodulate the facsimile signals, and transmit the demodulated image data and associated control signals to the remote facsimile module using the facsimile packet formats and procedures of Annex M. At the remote facsimile module, the baseband signal is remodulated to voiceband for transmission to the peer facsimile terminal. Facsimile traffic that cannot be demodulated is transmitted via audio encodings suitable for voiceband data, e.g. 40 kbit/s ADPCM or 64 kbit/s PCM.

Table 17-1 summarizes the requirements for facsimile demodulation.

**Table 17-1/I.366.2 – Facsimile demodulation requirements**

| | |
|---|---|
| Facsimile traffic demodulated | ITU-T Group 3 facsimile T.30 and T.4 standard facilities; optionally, T.30 non-standard facilities |
| Facsimile traffic not demodulated<br>(e.g. handled by G.726-40 or G.711-64) | ITU-T Group 1 and Group 2;<br>some or all T.30 non-standard facilities |
| Image data high-speed modulations (Note) | V.17 (14 400, 12 000, 9600, 7200 bit/s);<br>V.29 (9600, 7200 bit/s);<br>V.27 *ter* (4800, 2400 bit/s)<br>V.33 (14 400) |
| Control signals demodulated | V.21 (300 bit/s) |
| Remodulated signal level | –17 dBm0 |
| Facsimile terminal types | Automatic and manual |
| Facsimile demodulation capability | Enable/Disable |
| NOTE – The support of V.34 modulation schemes is for further study. | |

### 17.2 Two methods of analysis

Two approaches exist to handle facsimile demodulation: Protocol Analysis and Waveform Analysis.

The packet formats and procedures of Annex M support both approaches as well as interoperability between the approaches (as does ITU-T G.766).

#### 17.2.1 Protocol analysis

The protocol analysis (PA) approach is based on decoding and interpreting the procedural signals exchanged between the facsimile terminals. A minimum amount of signal analysis, such as activity detection and low/high speed discrimination, is also performed in this approach.

Using the T.30 message information, protocol analysis keeps track of the state of standard facility traffic and obtains the necessary information to control the demodulators.

In a non-standard mode of operation, protocol analysis depends on recognizing the non-standard T.30 protocol identification code, interpreting the meaning of the information exchanged between the facsimile terminals, and demodulating or remodulating the facsimile signals accordingly.

#### 17.2.2 Waveform analysis

The waveform analysis (WA) approach is based on analysis and classification of the modulated waveforms transmitted by the facsimile terminals.

Waveform analysis does not interpret the T.30 facsimile protocol and, therefore, does not keep track of the progress of facsimile traffic through its different states.

In waveform analysis there is no difference between standard and non-standard facilities. This approach can handle the demodulation of both kinds of facsimile traffic.

### 17.3 Optional support of non-standard facilities

The packet formats and procedures of Annex M support the optional demodulation of facsimile traffic with non-standard T.30 facilities.

During the facsimile set-up, the called facsimile terminal may identify its non-standard capabilities. The calling facsimile terminal may then instruct the called terminal to operate in non-standard T.30 mode. This scenario typically occurs when both terminals are of the same manufacturer.

### 17.4 Transparency for T.30 data

T.30 is an end-to-end handshaking protocol between facsimile machines. The demodulated T.30 HDLC framed signals are passed transparently (including HDLC flags) between the near end and far end SSCS even when errors are detected (e.g. cyclic redundancy check failure). HDLC zero stuffing is also passed transparently. A mechanism is provided so that the extra bits that might be produced due to non-octet alignment need not be transmitted to the far end facsimile terminal.

The Preamble Flags are not passed but are indicated and should be regenerated at the far end.

For protocol analysis, some information fields of the HDLC frame may optionally be altered to control the protocol, e.g. disabling non-standard facilities. This is an exception to the usual transparency of T.30 data and requires the module making the intervention to recalculate HDLC framing.

### 17.5 Timing requirements

It is important that the output signal of a facsimile remodulator follow the timing requirements specified in ITU-T T.30.

In the T.30 protocol, there are time gaps between certain consecutive signals that are required to be kept within a specified tolerance. Specifically, there is a requirement of a gap of 75 ms ± 20 ms between the end of certain low-speed signals (e.g. DCS) and the start of the following high-speed signal (e.g. EPT or training). There is the same requirement between the end of certain high-speed signals (e.g. page data) and the start of the following low-speed signal (e.g. end-of-procedure EOP).

Another requirement of the facsimile transmission protocol is that there should be no gap between the end of the training sequence and the beginning of the data. The preceding gap between the EPT and the training sequence is 20 to 25 ms.

The packet formats and procedures of Annex M ensure the capability to reconstruct these gaps by setting a time stamp in facsimile demodulation control packets as the information they contain is passed from the demodulating User to the SSCS for transmission. A time stamp denotes the beginning of the corresponding event.

NOTE – As explained in 9.1, the demodulating User interacts synchronously with the SSCS and shall ensure constant delay for different events. This is measured from the beginning of an input signal, which is then analysed and classified, to the request that the SSCS generate a control packet. Time stamps in facsimile demodulation control packets thereby maintain a constant offset from the underlying event, and intervals between events are accurately conveyed.

The remodulating User need not obey received time stamps as the only guide to its behaviour. Under severely errored line conditions, constant delay in time stamping at the demodulator might not be maintained, and the receiver may choose to reconstruct T.30 signals at the remodulator to ensure that timing requirements are met.

T.30 timing tolerances may be extended using optional techniques like the insertion of extra low-speed or high-speed flag sequences.

### 17.6 Starting and stopping facsimile demodulation

A connection is classified as facsimile traffic upon detection of a T.30 preamble. Optionally, a connection may be classified as facsimile traffic based upon the detection of CNG and CED tones as specified in ITU-T T.30. However, one or both of these tones may be omitted in the case of manually operated facsimile equipment.

The User shall enter and exit the Facsimile Demodulation state by coordination with the peer User. If either User determines that a connection cannot be handled by facsimile demodulation, they may revert to Voiceband Data state and remain in that state until the transfer is concluded.

User state control messages instructing the peer User to enter and exit the Facsimile Demodulation state are described in Annex O.

The SSCS is locally controlled by its User to operate in Audio or Facsimile Demodulation mode. The transmit and receive directions of the SSCS can be in different states.

### 17.7 Facsimile demodulation packets

Facsimile demodulation uses the following packet types, whose format is defined in Annex M:

- Modulation control messages and T.30 data are transported using type 3 packets (with the payload protected by a 10-bit CRC).
- Image data is transported using type 1 packets (with unprotected payload).

Packet flows for typical facsimile demodulation scenarios are shown in Appendix III.

## 18 SSCS parameters of operation

Values for the SSCS parameters of operation listed in Table 18-1 must be determined before this SSCS can be used on an AAL type 2 connection. Such determination may be made via provisioning or signalling in a manner outside the scope of this Recommendation. In the absence of provisioning or signalling (at the ATM level or the AAL type 2 level) for a given parameter, the default value of that parameter shall apply. The values of these SSCS parameters may differ from one AAL type 2 connection to another.

**Table 18-1/I.366.2 – SSCS parameters of operation**

| SSCS parameter | | Audio service category | | Multirate service category |
|---|---|---|---|---|
| | | Permitted values | Default value | Permitted values |
| 1 | Service category (Note 1) | Audio | Audio | Multirate |
| 2 | Transport of audio information | enabled | enabled | N/A |
| 3 | Source of the encoding format profile | ITU-T predefined, other predefined, custom | ITU-T predefined | N/A |
| 3a | ITU-T predefined profile (Annex P, Figure P.1) | 1 .. 255 | 1 | N/A |
| 3b | Other predefined profile | 1 .. 255 | N/A | N/A |
| 3c | Custom profile: description of its content | For further study | N/A | N/A |
| 4 | Interpretation of the generic PCM encoding format defined in Annex B | A-law, μ-law | A-law | N/A |
| 5 | Transport of demodulated facsimile data (Note 2) | enabled, disabled | disabled | N/A |
| 6 | Transport of channel-associated signalling bits | enabled, disabled | disabled | N/A |
| 7 | Transport of DTMF dialled digits | enabled, disabled | disabled | N/A |
| 8 | Transport of MF-R1 dialled digits | enabled, disabled | disabled | N/A |
| 9 | Transport of MF-R2 dialled digits | enabled, disabled | disabled | N/A |
| 10 | Transport of circuit mode data (Note 2) | enabled, disabled | disabled | enabled |
| 10a | Multiplier N in N*64 kbit/s circuit mode data | 1 | 1 | 1 .. 31 |
| 11 | Transport of frame mode data | enabled, disabled | disabled | enabled, disabled |
| 11a | Maximum length of a frame mode data unit | 1 .. 65535 | N/A | 1 .. 65535 |
| 12 | Transport of rate control | enabled, disabled | disabled | N/A |
| 13 | Transport of synchronization of change in SSCS operation | enabled, disabled | disabled | N/A |
| 14 | Loopback | enabled, disabled | disabled | enabled, disabled |

N/A Not applicable

NOTE 1 – The default service category is the Audio Service category.

NOTE 2 – If the value of this parameter is "disabled", the User shall not invoke Set_SSCS_State.request to change to the corresponding SSCS state, Circuit Mode or Facsimile Demodulation, respectively.

## ANNEX A

## Specification of audio encoding formats

Encoding formats for ITU-T audio algorithms, including silence insertion, are defined in Annexes B through I.

NOTE – Custom encoding formats different from those defined in Annexes B through I are allowed in custom profiles. The definition of custom formats and profiles is outside the scope of this Recommendation.

In each case of voice encoding, an encoding data unit (EDU) is specified. In some cases this is defined to be a concatenation of multiple algorithmic frames in order that the result be octet aligned.

EDUs may be additionally concatenated, in the order of earliest first, to form a packet as specified by an entry in an encoding format profile. The profile entry specifies the resulting length in octets. This must be an integral multiple of the EDU size for the referenced encoding format. All the data in a single packet shall be of the same encoding format, i.e. the same audio algorithm and bit rate.

The extent to which EDUs can be multiplied in a single packet is determined by the range of entries that make up the adopted profile. The number of EDUs allowed may be less than the maximum derived by considering only the limitation of maximum CPS packet size.

A well-formed profile obeys the principle stated in 13.2 and allows service data units (SDUs) to be identified in all multiples from 1 up to some maximum value of M. An SDU is an integral multiple of an EDU.

Silence insertion descriptors are treated differently. SIDs are never multiplied within a single packet and are not mixed with other data units. Each SID is an SDU by itself.

## ANNEX B

## Encoding format for audio algorithm G.711

### B.1 General

G.711 Pulse Code Modulation (PCM) is a coder that produces one 8-bit value every 125 μs representing the sign and amplitude of an audio sample. Two encoding laws are recommended, referred to as A-law and μ-law.

Also useful is the concept of a generic PCM encoding format. This signifies that the choice of A-law or μ-law is not made explicitly as part of an encoding profile. It is known instead through a separate SSCS parameter of operation.

Encoded values are represented in the SSCS with the polarity (sign) bit as the most significant (see Tables 1/G.711 and 2/G.711). Bit numbering in G.711 is the reverse of the I.361 convention adopted here.

Because they correspond linearly to the amplitude of an audio sample, one or more least significant bits of an encoded value may be dropped, and the remaining bits still provide useful information. In fact, in the case of CAS on a 1544 kbit/s interface, the least significant bit is normally corrupted in intermediate switching and transmission systems. A precedent for dropping one or two bits and transmitting G.711 at rates of 56 and 48 kbit/s can be found in Annex A/H.221.

Informative reference: ITU-T H.221 (1999), *Frame structure for a 64 to 1920 kbit/s channel in audiovisual teleservices*.

## B.2 Encoding data unit

The data unit format requires that G.711 outputs be accumulated over an interval of 1 ms to yield a sequence of 8 encoded values. These are concatenated in chronological order, with the earliest positioned at the most significant bit of the first octet.

Formats for 64, 56 and 48 kbit/s are shown in Figures B.1 through B.3. They are the same for A-law, μ-law, and generic PCM.

<table>
<tr><td>8</td><td>7</td><td>6</td><td>5</td><td>4</td><td>3</td><td>2</td><td>1</td><td></td></tr>
<tr><td colspan="8">PCM 1</td><td>1</td></tr>
<tr><td colspan="8">PCM 2</td><td>2</td></tr>
<tr><td colspan="8">PCM 3</td><td>3</td></tr>
<tr><td colspan="8">PCM 4</td><td>4</td></tr>
<tr><td colspan="8">PCM 5</td><td>5</td></tr>
<tr><td colspan="8">PCM 6</td><td>6</td></tr>
<tr><td colspan="8">PCM 7</td><td>7</td></tr>
<tr><td colspan="8">PCM 8</td><td>8</td></tr>
</table>

**Figure B.1/I.366.2 – G.711-64 EDU format**

<table>
<tr><td>8</td><td>7</td><td>6</td><td>5</td><td>4</td><td>3</td><td>2</td><td>1</td><td></td></tr>
<tr><td colspan="7">PCM 1</td><td></td><td>1</td></tr>
<tr><td colspan="6">PCM 2</td><td colspan="2"></td><td>2</td></tr>
<tr><td colspan="5">PCM 3</td><td colspan="3"></td><td>3</td></tr>
<tr><td colspan="4">PCM 4</td><td colspan="4">PCM 5</td><td>4</td></tr>
<tr><td colspan="3"></td><td colspan="5">PCM 6</td><td>5</td></tr>
<tr><td colspan="2"></td><td colspan="6">PCM 7</td><td>6</td></tr>
<tr><td></td><td colspan="7">PCM 8</td><td>7</td></tr>
</table>

**Figure B.2/I.366.2 – G.711-56 EDU format**

<table>
<tr><td>8</td><td>7</td><td>6</td><td>5</td><td>4</td><td>3</td><td>2</td><td>1</td><td></td></tr>
<tr><td colspan="6">PCM 1</td><td colspan="2"></td><td>1</td></tr>
<tr><td colspan="4">PCM 2</td><td colspan="4">PCM 3</td><td>2</td></tr>
<tr><td colspan="2"></td><td colspan="6">PCM 4</td><td>3</td></tr>
<tr><td colspan="6">PCM 5</td><td colspan="2"></td><td>4</td></tr>
<tr><td colspan="4">PCM 6</td><td colspan="4">PCM 7</td><td>5</td></tr>
<tr><td colspan="2"></td><td colspan="6">PCM 8</td><td>6</td></tr>
</table>

**Figure B.3/I.366.2 – G.711-48 EDU format**

ANNEX C

## Encoding format for audio algorithm G.722

### C.1 General

G.722 Sub-Band Adaptive Pulse Code Modulation (SB-ADPCM) is a coder that produces one 8-bit value every 125 µs and represents audio samples with higher fidelity than G.711 PCM. G.722 operates by splitting the frequency band 50-7000 Hz into two sub-bands and coding each separately using ADPCM to yield 2 bits for the higher sub-band and 6 bits for the lower sub-band.

Encoded values are represented in the SSCS with the higher sub-band ADPCM codeword in the most significant bits followed by the lower sub-band ADPCM codeword in the least significant bits (see 1.4.4/G.722). Bit numbering in G.722 is the reverse of the I.361 convention adopted here.

ITU-T G.722 provides three modes of operation, whereby 0, 1, or 2 least significant bits are dropped from the encoded value of the lower sub-band. Accordingly, 64, 56, or 48 kbit/s is used to transfer the audio information. From an algorithmic point of view, the mode may be changed after any audio sample.

ITU-T G.722 does not define an intrinsic SID and may be used with the generic SID of Annex I. If this is the case, the audio coder and decoder shall be reset synchronously at the beginning of each talk spurt, as described in I.3.

### C.2 Encoding data unit

The data unit format requires that G.722 outputs be accumulated over an interval of 1 ms to yield a sequence of 8 encoded values. These are concatenated in chronological order, with the earliest positioned at the most significant bit of the first octet.

Formats for 64, 56 and 48 kbit/s are shown in Figures C.1 through C.3.

| 8 7 6 5 4 3 2 1 | |
|---|---|
| SB-ADPCM 1 | 1 |
| SB-ADPCM 2 | 2 |
| SB-ADPCM 3 | 3 |
| SB-ADPCM 4 | 4 |
| SB-ADPCM 5 | 5 |
| SB-ADPCM 6 | 6 |
| SB-ADPCM 7 | 7 |
| SB-ADPCM 8 | 8 |

**Figure C.1/I.366.2 – G.722-64 EDU format**

<table>
<tr><th>8</th><th>7</th><th>6</th><th>5</th><th>4</th><th>3</th><th>2</th><th>1</th><th></th></tr>
<tr><td colspan="7">SB-ADPCM 1</td><td></td><td>1</td></tr>
<tr><td colspan="6">SB-ADPCM 2</td><td colspan="2"></td><td>2</td></tr>
<tr><td colspan="5">SB-ADPCM 3</td><td colspan="3"></td><td>3</td></tr>
<tr><td colspan="4">SB-ADPCM 4</td><td colspan="4">SB-ADPCM 5</td><td>4</td></tr>
<tr><td colspan="3"></td><td colspan="5">SB-ADPCM 6</td><td>5</td></tr>
<tr><td colspan="2"></td><td colspan="6">SB-ADPCM 7</td><td>6</td></tr>
<tr><td></td><td colspan="7">SB-ADPCM 8</td><td>7</td></tr>
</table>

**Figure C.2/I.366.2 – G.722-56 EDU format**

| 8 | 7 | 6 | 5 | 4 | 3 | 2 | 1 | |
|---|---|---|---|---|---|---|---|---|
| SB-ADPCM 1 | | | | | | | | 1 |
| SB-ADPCM 2 | | | | SB-ADPCM 3 | | | | 2 |
| | | SB-ADPCM 4 | | | | | | 3 |
| SB-ADPCM 5 | | | | | | | | 4 |
| SB-ADPCM 6 | | | | SB-ADPCM 7 | | | | 5 |
| | | SB-ADPCM 8 | | | | | | 6 |

**Figure C.3/I.366.2 – G.722-48 EDU format**

# ANNEX D

## Encoding format for audio algorithm G.723.1

### D.1 General

G.723.1 operates at either 5.3 or 6.4 kbit/s. Both rates are a mandatory part of the encoder and decoder. Every 30 ms, G.723.1 emits either 160 or 192 bits, respectively, that characterize a voice sample. It is possible to switch between the two rates at any 30 ms boundary.

### D.2 Encoding data unit

The bits of a G.723.1 frame are formatted as shown in Figures D.1 and D.2 (see Tables 5/G.723.1 and 6/G.723.1). Within the fields of a data unit, later octets are more significant. This is based on H.324 bit order assignment and is the reverse of the I.361 convention.

| 8 | 7 | 6 | 5 | 4 | 3 | 2 | 1 | |
|---|---|---|---|---|---|---|---|---|
| LPC_B5...B0 | | | | | | 0 | 0 | 1 |
| LPC_B13...B6 | | | | | | | | 2 |
| LPC_B21...B14 | | | | | | | | 3 |
| ACL0_B5...B0 | | | | | | LPC_B23...B22 | | 4 |
| ACL2_B4...B0 | | | | | ACL1_B1...B0 | | ACL0_B6 | 5 |
| GAIN0_B3...B0 | | | | ACL3_B1...B0 | | ACL2_B6...B5 | | 6 |
| GAIN0_B11...B4 | | | | | | | | 7 |
| GAIN1_B7...B0 | | | | | | | | 8 |
| GAIN2_B3...B0 | | | | GAIN1_B11...B8 | | | | 9 |
| GAIN2_B11...B4 | | | | | | | | 10 |
| GAIN3_B7...B0 | | | | | | | | 11 |
| GRID3 | GRID2 | GRID1 | GRID0 | GAIN3_B11...B8 | | | | 12 |
| MSBPOS_B6...B0 | | | | | | | 0 | 13 |
| POS0_B1...B0 | | MSBPOS_B12...B7 | | | | | | 14 |
| POS0_B9...B2 | | | | | | | | 15 |
| POS1_B1...B0 | | POS0_B15...B10 | | | | | | 16 |
| POS1_B9...B2 | | | | | | | | 17 |
| POS2_B3...B0 | | | | POS1_B13...B10 | | | | 18 |
| POS2_B11...B4 | | | | | | | | 19 |
| POS3_B3...B0 | | | | POS2_B15...B12 | | | | 20 |
| POS3_B11...B4 | | | | | | | | 21 |
| PSIG0_B5...B0 | | | | | | POS3_B13...B12 | | 22 |
| PSIG2_B2...B0 | | | PSIG1_B4...B0 | | | | | 23 |
| PSIG3_B4...B0 | | | | | PSIG2_B5...B3 | | | 24 |

**Figure D.1/I.366.2 – G.723.1-6.4 EDU format**

| 8 | 7 | 6 | 5 | 4 | 3 | 2 | 1 | |
|---|---|---|---|---|---|---|---|---|
| LPC_B5...B0 | | | | | | 0 | 1 | 1 |
| LPC_B13...B6 | | | | | | | | 2 |
| LPC_B21...B14 | | | | | | | | 3 |
| ACL0_B5...B0 | | | | | | LPC_B23...B22 | | 4 |
| ACL2_B4...B0 | | | | | ACL1_B1...B0 | | ACL0_B6 | 5 |
| GAIN0_B3...B0 | | | | ACL3_B1...B0 | | ACL2_B6...B5 | | 6 |
| GAIN0_B11...B4 | | | | | | | | 7 |
| GAIN1_B7...B0 | | | | | | | | 8 |
| GAIN2_B3...B0 | | | | GAIN1_B11...B8 | | | | 9 |
| GAIN2_B11...B4 | | | | | | | | 10 |
| GAIN3_B7...B0 | | | | | | | | 11 |
| GRID3 | GRID2 | GRID1 | GRID0 | GAIN3_B11...B8 | | | | 12 |
| POS0_B7...B0 | | | | | | | | 13 |
| POS1_B3...B0 | | | | POS0_B11...B8 | | | | 14 |
| POS1_B11...B4 | | | | | | | | 15 |
| POS2_B7...B0 | | | | | | | | 16 |
| POS3_B3...B0 | | | | POS2_B11...B8 | | | | 17 |
| POS3_B11...B4 | | | | | | | | 18 |
| PSIG1_B3...B0 | | | | PSIG0_B3...B0 | | | | 19 |
| PSIG3_B3...B0 | | | | PSIG2_B3...B0 | | | | 20 |

**Figure D.2/I.366.2 – G.723.1-5.3 EDU format**

### D.3 Silence insertion descriptor

Annex A/G.723.1 defines a voice activity detector and comfort noise generator for use with G.723.1. It classifies each 30 ms sample as either active voice or background noise.

Active voice is encoded according to Figures D.1 and D.2. Background noise is encoded as a Silence Insertion Descriptor according to Figure D.3 (see Table A.1/G.723.1). SIDs are sent only intermittently, when an appreciable change is detected in the nature of the background noise.

| 8 | 7 | 6 | 5 | 4 | 3 | 2 | 1 | |
|---|---|---|---|---|---|---|---|---|
| LPC_B5...B0 | | | | | | 1 | 0 | 1 |
| LPC_B13...B6 | | | | | | | | 2 |
| LPC_B21...B14 | | | | | | | | 3 |
| GAIN_B5...B0 | | | | | | LPC_B23...B22 | | 4 |

**Figure D.3/I.366.2 – G.723.1 SID packet format**

## ANNEX E

## Encoding format for audio algorithm G.726

### E.1 General

G.726 Adaptive Pulse Code Modulation (ADPCM) supports bit rates of 40, 32, 24 and 16 kbit/s. The encoding produces 5, 4, 3, or 2 bits, respectively, every 125 µs.

Encoded values are represented in the SSCS with the sign bit as the most significant (see Tables 7/G.726 through 10/G.726). Bit numbering in G.726 is the reverse of the I.361 convention adopted here.

The principal application of the 24 and 16 kbit/s rates is to handle temporary overloads in voice multiplexing equipment. The principal application of the 40 kbit/s is to carry voiceband modem signals operating at data rates greater than 4.8 kbit/s.

ITU-T G.726 does not define an intrinsic SID and may be used with the generic SID of Annex I. If this is the case, the audio coder and decoder shall be reset synchronously at the beginning of each talk spurt, as described in I.3.

## E.2 Encoding data unit

The data unit format requires that G.726 outputs be accumulated over an interval of 1 ms to yield a sequence of 8 encoded values. These are concatenated in chronological order, with the earliest positioned at the most significant bit of the first octet.

Formats for the coding rates of 40, 32, 24 and 16 kbit/s are shown in Figures E.1 through E.4.

<table>
<tr><th>8</th><th>7</th><th>6</th><th>5</th><th>4</th><th>3</th><th>2</th><th>1</th><th></th></tr>
<tr><td colspan="5">ADPCM 1</td><td colspan="3">ADPCM 2</td><td>1</td></tr>
<tr><td colspan="2"></td><td colspan="5">ADPCM 3</td><td></td><td>2</td></tr>
<tr><td colspan="4">ADPCM 4</td><td colspan="4">ADPCM 5</td><td>3</td></tr>
<tr><td></td><td colspan="5">ADPCM 6</td><td colspan="2"></td><td>4</td></tr>
<tr><td colspan="3">ADPCM 7</td><td colspan="5">ADPCM 8</td><td>5</td></tr>
</table>

**Figure E.1/I.366.2 – G.726-40 EDU format**

<table>
<tr><th>8</th><th>7</th><th>6</th><th>5</th><th>4</th><th>3</th><th>2</th><th>1</th><th></th></tr>
<tr><td colspan="4">ADPCM 1</td><td colspan="4">ADPCM 2</td><td>1</td></tr>
<tr><td colspan="4">ADPCM 3</td><td colspan="4">ADPCM 4</td><td>2</td></tr>
<tr><td colspan="4">ADPCM 5</td><td colspan="4">ADPCM 6</td><td>3</td></tr>
<tr><td colspan="4">ADPCM 7</td><td colspan="4">ADPCM 8</td><td>4</td></tr>
</table>

**Figure E.2/I.366.2 – G.726-32 EDU format**

<table>
<tr><th>8</th><th>7</th><th>6</th><th>5</th><th>4</th><th>3</th><th>2</th><th>1</th><th></th></tr>
<tr><td colspan="3">ADPCM 1</td><td colspan="3">ADPCM 2</td><td colspan="2">ADPCM 3</td><td>1</td></tr>
<tr><td></td><td colspan="3">ADPCM 4</td><td colspan="3">ADPCM 5</td><td></td><td>2</td></tr>
<tr><td colspan="2">ADPCM 6</td><td colspan="3">ADPCM 7</td><td colspan="3">ADPCM 8</td><td>3</td></tr>
</table>

**Figure E.3/I.366.2 – G.726-24 EDU format**

<table>
<tr><th>8</th><th>7</th><th>6</th><th>5</th><th>4</th><th>3</th><th>2</th><th>1</th><th></th></tr>
<tr><td colspan="2">ADPCM 1</td><td colspan="2">ADPCM 2</td><td colspan="2">ADPCM 3</td><td colspan="2">ADPCM 4</td><td>1</td></tr>
<tr><td colspan="2">ADPCM 5</td><td colspan="2">ADPCM 6</td><td colspan="2">ADPCM 7</td><td colspan="2">ADPCM 8</td><td>2</td></tr>
</table>

**Figure E.4/I.366.2 – G.726-16 EDU format**

ANNEX F

## Encoding format for audio algorithm G.727

### F.1 General

G.727 Embedded Adaptive Pulse Code Modulation (EADPCM) is a family of variable bit rate coding algorithms with the capability of bit dropping outside the encoder and decoder blocks. G.727 produces code words which contain enhancement bits and core bits. Enhancement bits can be discarded during network congestion. The number of core bits must remain the same to avoid mistracking of the adaptation state between transmitter and receiver.

Encoded values are represented in the SSCS with the core bits as the most significant followed by the enhancement bits as the least significant (see Tables 8/G.727 through 11/G.727). Bit numbering in G.727 is the reverse of the I.361 convention adopted here.

Algorithms of the G.727 family are referred to by $(x,y)$ pairs, where $x$ is the number of core plus enhancement bits and $y$ is the number of core bits.

ITU-T G.727 provides coding rates of 40, 32, 24 and 16 kbit/s with core rates of 16, 24 and 32 kbit/s. This corresponds to the $(x,y)$ pairs: (5,2), (4,2), (3,2), (2,2); (5,3), (4,3), (3,3); (5,4), (4,4).

ITU-T G.727 does not define an intrinsic SID and may be used with the generic SID of Annex I. If this is the case, the audio coder and decoder shall be reset synchronously at the beginning of each talk spurt, as described in I.3.

### F.2 Encoding data unit

The data unit format requires that G.727 outputs be accumulated over an interval of 1 ms to yield a sequence of 8 encoded values. These are concatenated in chronological order, with the earliest positioned at the most significant bit of the first octet.

Formats for coding rates of 40, 32, 24 and 16 kbit/s are shown in Figures F.1 through F.4.

<table>
<tr><th>8</th><th>7</th><th>6</th><th>5</th><th>4</th><th>3</th><th>2</th><th>1</th><th></th></tr>
<tr><td colspan="5">EADPCM 1</td><td colspan="3">EADPCM 2</td><td>1</td></tr>
<tr><td colspan="2"></td><td colspan="5">EADPCM 3</td><td></td><td>2</td></tr>
<tr><td colspan="4">EADPCM 4</td><td colspan="4">EADPCM 5</td><td>3</td></tr>
<tr><td></td><td colspan="5">EADPCM 6</td><td colspan="2"></td><td>4</td></tr>
<tr><td colspan="3">EADPCM 7</td><td colspan="5">EADPCM 8</td><td>5</td></tr>
</table>

**Figure F.1/I.366.2 – G.727 (5,2), (5,3) and (5,4) EDU format**

<table>
<tr><th>8</th><th>7</th><th>6</th><th>5</th><th>4</th><th>3</th><th>2</th><th>1</th><th></th></tr>
<tr><td colspan="4">EADPCM 1</td><td colspan="4">EADPCM 2</td><td>1</td></tr>
<tr><td colspan="4">EADPCM 3</td><td colspan="4">EADPCM 4</td><td>2</td></tr>
<tr><td colspan="4">EADPCM 5</td><td colspan="4">EADPCM 6</td><td>3</td></tr>
<tr><td colspan="4">EADPCM 7</td><td colspan="4">EADPCM 8</td><td>4</td></tr>
</table>

**Figure F.2/I.366.2 – G.727 (4,2), (4,3) and (4,4) EDU format**

<table>
<tr><td>8</td><td>7</td><td>6</td><td>5</td><td>4</td><td>3</td><td>2</td><td>1</td><td></td></tr>
<tr><td colspan="3">EADPCM 1</td><td colspan="3">EADPCM 2</td><td colspan="2">EADPCM 3</td><td>1</td></tr>
<tr><td></td><td colspan="3">EADPCM 4</td><td colspan="3">EADPCM 5</td><td></td><td>2</td></tr>
<tr><td colspan="2">EADPCM 6</td><td colspan="3">EADPCM 7</td><td colspan="3">EADPCM 8</td><td>3</td></tr>
</table>

**Figure F.3/I.366.2 – G.727 (3,2) and (3,3) EDU format**

<table>
<tr><td>8</td><td>7</td><td>6</td><td>5</td><td>4</td><td>3</td><td>2</td><td>1</td><td></td></tr>
<tr><td colspan="2">EADPCM 1</td><td colspan="2">EADPCM 2</td><td colspan="2">EADPCM 3</td><td colspan="2">EADPCM 4</td><td>1</td></tr>
<tr><td colspan="2">EADPCM 5</td><td colspan="2">EADPCM 6</td><td colspan="2">EADPCM 7</td><td colspan="2">EADPCM 8</td><td>2</td></tr>
</table>

**Figure F.4/I.366.2 – G.727 (2,2) EDU format**

# ANNEX G

## Encoding format for audio algorithm G.728

### G.1 General

G.728 Low Delay Code Excited Linear Prediction (LD-CELP) is a coder that produces a group of four codewords every 2.5 ms. Each group of codewords is referred to as an adaptation cycle or frame.

Shape codevector indexes and gain indexes are represented in the SSCS according to the conventions of I.361, whereby earlier octets and higher-numbered bits are more significant.

The basic algorithm of G.728 runs at 16 kbit/s. Annex H expands operation by adding two lower rates of 12.8 and 9.6 kbit/s.

Recommendation G.728 does not define an intrinsic SID and may be used with the generic SID of Annex I. If this is the case, the audio coder and decoder shall be reset synchronously at the beginning of each talk spurt, as described in I.3.

### G.2 Encoding data unit

The formats for G.728 at 16, 12.8 and 9.6 kbit/s are shown in Figures G.1 through G.3 (see 5.11/G.728, H.3.1.1/G.728 and H.4.1.1/G.728). Within the fields of a data unit, bit and octet significance follows the I.361 convention adopted here.

<table>
<tr><td>8</td><td>7</td><td>6</td><td>5</td><td>4</td><td>3</td><td>2</td><td>1</td><td></td></tr>
<tr><td colspan="7">shape codevector index 1</td><td></td><td>1</td></tr>
<tr><td colspan="2">gain index 1</td><td colspan="6">shape codevector index 2</td><td>2</td></tr>
<tr><td></td><td colspan="3">gain index 2</td><td colspan="4">shape codevector index 3</td><td>3</td></tr>
<tr><td colspan="3"></td><td colspan="2">gain index 3</td><td colspan="3"></td><td>4</td></tr>
<tr><td colspan="4">shape codevector index 4</td><td colspan="4">gain index 4</td><td>5</td></tr>
</table>

**Figure G.1/I.366.2 – G.728-16 EDU format**

<table>
<tr><th>8</th><th>7</th><th>6</th><th>5</th><th>4</th><th>3</th><th>2</th><th>1</th><th></th></tr>
<tr><td colspan="6">shape codevector index 1</td><td colspan="2">gain index 1</td><td>1</td></tr>
<tr><td colspan="6">shape codevector index 2</td><td colspan="2">gain index 2</td><td>2</td></tr>
<tr><td colspan="6">shape codevector index 3</td><td colspan="2">gain index 3</td><td>3</td></tr>
<tr><td colspan="6">shape codevector index 4</td><td colspan="2">gain index 4</td><td>4</td></tr>
</table>

**Figure G.2/I.366.2 – G.728-12.8 EDU format**

<table>
<tr><th>8</th><th>7</th><th>6</th><th>5</th><th>4</th><th>3</th><th>2</th><th>1</th><th></th></tr>
<tr><td colspan="4">shape codevector index 1</td><td colspan="2">gain index 1</td><td colspan="2">shape code-</td><td>1</td></tr>
<tr><td colspan="2">vector index 2</td><td colspan="2">gain index 2</td><td colspan="4">shape codevector index 3</td><td>2</td></tr>
<tr><td colspan="2">gain index 3</td><td colspan="4">shape codevector index 4</td><td colspan="2">gain index 4</td><td>3</td></tr>
</table>

**Figure G.3/I.366.2 – G.728-9.6 EDU format**

# ANNEX H

## Encoding format for audio algorithm G.729

### H.1 General

The basic algorithm of G.729 runs at 8 kbit/s. Every 10 ms it emits 80 bits that encode a voice frame.

Encoded values are represented in the SSCS according to the conventions of ITU-T I.361, whereby earlier octets and higher-numbered bits are more significant.

G.729 Annex A defines a reduced complexity coder that is interoperable with basic G.729. G.729 Annex C defines an interoperable floating point version of G.729 (main body) and G.729 Annex A. The format of the encoded values is the same for G.729, G.729 Annex A and G.729 Annex C. Any combination of G.729, G.729 Annex A and G.729 Annex C transmitter and receiver can be used together.

G.729 Annex B defines a voice activity detector and comfort noise generator for use with G.729 or G.729 Annex A. It classifies each 10 ms sample as either active voice or background noise.

G.729 Annex D defines a 6.4 kbit/s rate extension of G.729 for momentary reduction in channel capacity, e.g. to handle overload conditions. G.729 Annex E provides a 11.8 kbit/s extension of G.729 for better performance with a wide range of input signals, such as speech with background noise and music.

### H.2 Encoding data unit

The bits of a G.729 frame are formatted as shown in Figure H.1 (see Table 8/G.729). Within the fields of a data unit, bit and octet significance follows the I.361 convention adopted here.

<table>
<tr><td>8</td><td>7</td><td>6</td><td>5</td><td>4</td><td>3</td><td>2</td><td>1</td><td></td></tr>
<tr><td>L0</td><td colspan="7">L1</td><td>1</td></tr>
<tr><td colspan="5">L2</td><td colspan="3">L3</td><td>2</td></tr>
<tr><td colspan="2"></td><td colspan="6">P1</td><td>3</td></tr>
<tr><td colspan="2"></td><td>P0</td><td colspan="5"></td><td>4</td></tr>
<tr><td colspan="8">C1</td><td>5</td></tr>
<tr><td colspan="4">S1</td><td colspan="3">GA1</td><td></td><td>6</td></tr>
<tr><td colspan="3">GB1</td><td colspan="5">P2</td><td>7</td></tr>
<tr><td colspan="8">C2</td><td>8</td></tr>
<tr><td colspan="5"></td><td colspan="3">S2</td><td>9</td></tr>
<tr><td></td><td colspan="3">GA2</td><td colspan="4">GB2</td><td>10</td></tr>
</table>

**Figure H.1/I.366.2 – G.729-8 EDU format**

## H.3 Silence insertion descriptor

Active voice is encoded according to Figure H.1. Background noise is encoded as a Silence Insertion Descriptor according to Figure H.2 (see Table B.2/G.729). SIDs are sent only intermittently, when an appreciable change is detected in the nature of the background noise.

<table>
<tr><td>8</td><td>7</td><td>6</td><td>5</td><td>4</td><td>3</td><td>2</td><td>1</td><td></td></tr>
<tr><td>SPI</td><td colspan="5">First stage vector</td><td colspan="2"></td><td>1</td></tr>
<tr><td colspan="2">Second stage vector</td><td colspan="5">Gain (energy)</td><td>RES</td><td>2</td></tr>
</table>

RES Reserved (set to zero)
SPI Switched predictor index of LSF quantizer

**Figure H.2/I.366.2 – G.729 SID packet format**

## H.4 G.729-6.4 encoding data unit

The bits of a G.729-6.4 frame are formatted as shown in Figure H.3 (see Table D.1/G.729). Within the fields of a data unit, bit and octet significance follows the I.361 convention adopted here.

<table>
<tr><td>8</td><td>7</td><td>6</td><td>5</td><td>4</td><td>3</td><td>2</td><td>1</td><td></td></tr>
<tr><td>L0</td><td colspan="7">L1</td><td>1</td></tr>
<tr><td colspan="5">L2</td><td colspan="3">L3</td><td>2</td></tr>
<tr><td colspan="2"></td><td colspan="6">P1</td><td>3</td></tr>
<tr><td colspan="2"></td><td colspan="6">C1</td><td>4</td></tr>
<tr><td colspan="3"></td><td colspan="2">S1</td><td colspan="3">GA1</td><td>5</td></tr>
<tr><td colspan="3">GB1</td><td colspan="4">P2</td><td></td><td>6</td></tr>
<tr><td colspan="8">C2</td><td>7</td></tr>
<tr><td colspan="2">S2</td><td colspan="3">GA2</td><td colspan="3">GB2</td><td>8</td></tr>
</table>

**Figure H.3/I.366.2 – G.729-6.4 encoding data unit format**

## H.5 G.729-12 encoding data unit

The bits of a G.729-12 frame are formatted as shown in Figure H.4 (see Tables E.3a/G.729 and E.3b/G.729). Figure H.4 parts A and B describe the fields respectively for the forward adaptive mode and the backward adaptive mode of the G.729 Annex E algorithm. The net bit rate for the G.729 Annex E algorithm is 11.8 kbit/s, which is encapsulated in octet boundaries by using two stuffing bits UB, thus increasing the gross bit rate to 12 kbit/s. The value of the UB bits is not defined. Within the fields of a data unit, bit and octet significance follows the I.361 convention adopted here.

<table>
<tr><th>8</th><th>7</th><th>6</th><th>5</th><th>4</th><th>3</th><th>2</th><th>1</th><th></th></tr>
<tr><td>0</td><td>0</td><td>L0</td><td colspan="5">L1</td><td>1</td></tr>
<tr><td colspan="2"></td><td colspan="5">L2</td><td></td><td>2</td></tr>
<tr><td colspan="4">L3</td><td colspan="4">P1</td><td>3</td></tr>
<tr><td colspan="4"></td><td>P0</td><td colspan="3">C0_1</td><td>4</td></tr>
<tr><td colspan="4"></td><td colspan="4">C1_1</td><td>5</td></tr>
<tr><td colspan="3"></td><td colspan="5">C2_1</td><td>6</td></tr>
<tr><td colspan="2"></td><td colspan="6">C3_1</td><td>7</td></tr>
<tr><td></td><td colspan="7">C4_1</td><td>8</td></tr>
<tr><td colspan="3">GA1</td><td colspan="4">GB1</td><td></td><td>9</td></tr>
<tr><td colspan="4">P2</td><td colspan="4">C0_2</td><td>10</td></tr>
<tr><td colspan="3"></td><td colspan="5">C1_2</td><td>11</td></tr>
<tr><td colspan="2"></td><td colspan="6">C2_2</td><td>12</td></tr>
<tr><td></td><td colspan="7">C3_2</td><td>13</td></tr>
<tr><td colspan="7">C4_2</td><td></td><td>14</td></tr>
<tr><td colspan="2">GA2</td><td colspan="4">GB2</td><td colspan="2">UB</td><td>15</td></tr>
</table>

UB Unused Bit

**Figure H.4/I.366.2 – G.729-12 encoding data unit format for the forward adaptive mode *(part A)***

<table>
<tr><th>8</th><th>7</th><th>6</th><th>5</th><th>4</th><th>3</th><th>2</th><th>1</th><th></th></tr>
<tr><td>1</td><td>1</td><td colspan="6">P1</td><td>1</td></tr>
<tr><td colspan="2"></td><td>P0</td><td colspan="5"></td><td>2</td></tr>
<tr><td colspan="8">C0_1</td><td>3</td></tr>
<tr><td colspan="8">C1_1</td><td>4</td></tr>
<tr><td colspan="2"></td><td colspan="6">C2_1</td><td>5</td></tr>
<tr><td></td><td colspan="7">C3_1</td><td>6</td></tr>
<tr><td colspan="7">C4_1</td><td></td><td>7</td></tr>
<tr><td colspan="2">GA1</td><td colspan="4">GB1</td><td colspan="2"></td><td>8</td></tr>
<tr><td colspan="3">P2</td><td colspan="5"></td><td>9</td></tr>
<tr><td colspan="8">C0_2</td><td>10</td></tr>
<tr><td colspan="8">C1_2</td><td>11</td></tr>
<tr><td colspan="2"></td><td colspan="6">C2_2</td><td>12</td></tr>
<tr><td></td><td colspan="7">C3_2</td><td>13</td></tr>
<tr><td colspan="7">C4_2</td><td></td><td>14</td></tr>
<tr><td colspan="2">GA2</td><td colspan="4">GB2</td><td colspan="2">UB</td><td>15</td></tr>
</table>

UB Unused Bit

**Figure H.4/I.366.2 – G.729-12 encoding data unit format for the backward adaptive mode *(part B)***

ANNEX I

**Encoding format for generic silence insertion descriptor**

## I.1 General

ITU-T G.711, G.722, G.726, G.727 and G.728 do not contain provisions for voice activity detection, discontinuous transmission, and comfort noise generation tailored to the specific algorithm. Such procedures may be added in a generic way by the transmitter and receiver.

## I.2 Packet format

For this purpose, a generic SID, as shown in Figure I.1, is used.

| 8 | 7 | 6 | 5 | 4 | 3 | 2 | 1 | |
|---|---|---|---|---|---|---|---|---|
| RES | Noise Level | | | | | | | 1 |

RES Reserved (set to zero)

**Figure I.1/I.366.2 – Generic SID packet format**

The Noise Level field is coded according to Table I.1. It represents the total noise power level that the transmitter wishes to convey to the receiver. Other noise characteristics, such as the spectral distribution, are not specified.

**Table I.1/I.366.2 – Noise level codes**

| Noise Level | Meaning |
|---|---|
| 0-29 | reserved |
| 30 | –30 dBm0 |
| 31 | –31 dBm0 |
| . . . | . . . |
| 77 | –77 dBm0 |
| 78 | –78 dBm0 |
| 79-126 | reserved |
| 127 | Idle Code (no noise) |

NOTE – Table I.1 provides codepoints for measured noise levels, and the full or partial use of these codepoints is up to the implementation.

## I.3 Procedures

A generic SID packet shall be sent immediately after the last active voice packet of a talk spurt, i.e. at the first opportunity consistent with the proper operation of sequence numbers. It marks the beginning of silence and alerts the receiver to expect an absence of active voice packets. The SID may also be sent at arbitrary times during a silence period, either with the same or different noise level.

Since other characteristics of the noise, aside from its level, are not specified, they are to be chosen by the receiver. If a receiver is not capable of generating the total power level specified, it may generate a different level or may apply the idle code, i.e. no noise. Otherwise, the specified level should be considered a guideline.

If the first active voice packet following a generic SID packet selects an adaptive audio algorithm – such as G.722, G.726, G.727 or G.728 – encoding and decoding of that packet shall be performed starting from an audio coder state that has been reset to its specified initial values.

NOTE – This eliminates glitches that could otherwise occur if the transmitter's state were to diverge during the silence, when active voice packets are not being sent and the receiver's state is not being updated. Resetting both encoder and decoder in this way maintains a synchronized state and initiates a fresh adaptation for each talk spurt, unbiased by the previous one.

# ANNEX J

# Encoding format for circuit mode data at N*64 kbit/s

## J.1 Packet format

The number of time slots, N, is a parameter of the packet encoding format. From the value of N a packing multiple, M, is derived as specified in Table J.1.

**Table J.1/I.366.2 – Circuit mode data packet format parameters**

| Number of time slots N | Multiples per packet M | Sequence number interval (ms) | Number of packets per 5 ms |
|---|---|---|---|
| 1 | 40 | 5.000 | 1 |
| 2 | 20 | 2.500 | 2 |
| 3-4 | 10 | 1.250 | 4 |
| 5 | 8 | 1.000 | 5 |
| 6-8 | 5 | 0.625 | 8 |
| 9-10 | 4 | 0.500 | 10 |
| 11-20 | 2 | 0.250 | 20 |
| 21-31 | 1 | 0.125 | 40 |

These two parameters, N and M, then determine the format of each packet as shown in Figure J.1.

The packet payload consists of M multiples of an N-octet time slot interleaved group. Each group consists of one contemporaneous octet from each time slot in the same order that the time slots occur in the narrow-band call. The first group of N octets is the earliest in time, and the Mth group of N octets is the latest in time.

The most significant bit of each octet is aligned with the most significant bit of its time slot.

| 8 7 6 5 4 3 2 1 | |
|---|---|
| Octet 1 of time slot 1 | 1 |
| Octet 1 of time slot 2 | 2 |
| . . . | ... |
| Octet 1 of time slot N | N |
| Octet 2 of time slot 1 | N+1 |
| Octet 2 of time slot 2 | N+2 |
| . . . | ... |
| Octet 2 of time slot N | 2*N |
| . . . | ... |
| Octet M of time slot 1 | M*N–N+1 |
| Octet M of time slot 2 | M*N–N+2 |
| . . . | ... |
| Octet M of time slot N | M*N |

**Figure J.1/I.366.2 – Circuit mode data packet format**

ANNEX K

## Packet format and procedures for dialled digits

### K.1 General

The dialled digits packet format can be used to transport Dual-Tone Multi-Frequency (DTMF) signals, Multi-Frequency tones for signalling system R1 (MF-R1), or Multi-Frequency tones for signalling system R2 (MF-R2) across an AAL type 2 connection for reproduction at the other side. ITU-T Q.23 defines the frequency coding of DTMF, and ITU-T Q.24 gives tolerances for signal reception by different administrations. ITU-T Q.320, Q.322 and Q.323 define register signalling for MF-R1. ITU-T Q.441 defines interregister signalling for MF-R2.

Dialled digit packets constitute a separate, secondary information stream that avoids dependence on the audio encoding profile in effect. Some low bit rate audio encodings, such as G.723.1, do not convey multifrequency tones with acceptable fidelity. Other audio encodings that have higher fidelity may not require the support of dialled digit packets but can still find savings in bandwidth by the use of the dialled digits procedure.

The transmission of dialled digit packets is optional. SSCS parameters of operation separately enable the transport of DTMF, MF-R1, and MF-R2 dialled digits.

Dialled digits may be used during call set-up to convey destination address information. They may also be used in the middle of a call. Dialled digits are one way to convey User commands to a device at the far end of a connection, such as an automatic voice message recording system.

Dialled digit and audio encoding packets may occur at the same time. They are independent streams and may experience differential delay in reconstruction and play-out. In general, dialled digit signals are generated to be recognized by machines and during the intervals of operation other audio in the same direction is ignored. For the most reliable operation, transmitters should stop sending audio packets while detecting and sending dialled digits. However, specification of the behaviour of transmitting Users is beyond the scope of this Recommendation. For this reason, receivers should discard any audio while playing out dialled digits instead of striving to merge the two streams.

## K.2 Packet format

Dialled digit packets are format type 3 and benefit from CRC-10 error detection. They make use of the common facilities for type 3 packets defined in clause 11, including triple redundancy. A time stamp provides accurate relative timing for the duration of dialled digits and pauses.

The format of dialled digit packets is shown in Figure K.1.

<table>
<tr><td>8</td><td>7</td><td>6</td><td>5</td><td>4</td><td>3</td><td>2</td><td>1</td><td></td></tr>
<tr><td colspan="2">Redundancy</td><td colspan="6">Time Stamp</td><td>1</td></tr>
<tr><td colspan="8"></td><td>2</td></tr>
<tr><td colspan="3">RES</td><td colspan="5">Signal Level</td><td>3</td></tr>
<tr><td colspan="3">Digit Type</td><td colspan="5">Digit Code</td><td>4</td></tr>
<tr><td colspan="6">Message Type = 000010</td><td colspan="2"></td><td>5</td></tr>
<tr><td colspan="8">CRC-10</td><td>6</td></tr>
</table>

RES Reserved (set to zero)

**Figure K.1/I.366.2 – Dialled digits packet format**

The Time Stamp and Redundancy fields are coded according to the common facilities for type 3 packets defined in clause 11.

The Signal Level field is coded with a binary value from 0 to 31, signifying a total power level of 0 to −31 dBm0. Levels of −31 dBm0 and below are indicated by the value 31, and levels of 0 dBm0 and above are indicated by the value 0.

The Digit Type field is coded according to Table K.1.

**Table K.1/I.366.2 – Dialled digits type codes**

| Digits Type | Meaning |
|---|---|
| 000 | DTMF |
| 001 | MF-R1 |
| 010 | MF-R2 Forward |
| 011 | MF-R2 Backward |
| 100-111 | Reserved |

The Digit Code field is coded according to Table K.2 for DTMF, Table K.3 for MF-R1, and Table K.4 for MF-R2.

**Table K.2/I.366.2 – DTMF dialled digit codes**

| Digit Code | Meaning |
|---|---|
| 00000 | 0 |
| 00001 | 1 |
| 00010 | 2 |
| 00011 | 3 |
| 00100 | 4 |
| 00101 | 5 |
| 00110 | 6 |

**Table K.2/I.366.2 – DTMF dialled digit codes *(concluded)***

| Digit Code | Meaning |
|---|---|
| 00111 | 7 |
| 01000 | 8 |
| 01001 | 9 |
| 01010 | * |
| 01011 | # |
| 01100 | A |
| 01101 | B |
| 01110 | C |
| 01111 | D |
| 10000-11110 | Reserved |
| 11111 | Tone-off |

**Table K.3/I.366.2 – MF-R1 dialled digit codes**

| Digit Code | Meaning |
|---|---|
| 00000 | 0 |
| 00001 | 1 |
| 00010 | 2 |
| 00011 | 3 |
| 00100 | 4 |
| 00101 | 5 |
| 00110 | 6 |
| 00111 | 7 |
| 01000 | 8 |
| 01001 | 9 |
| 01010 | KP |
| 01011 | ST |
| 01100 | Spare<br>(700 + 1700) |
| 01101 | Spare<br>(900 + 1700) |
| 01110 | Spare<br>(1300 + 1700) |
| 01111-11110 | Reserved |
| 11111 | Tone-off |

**Table K.4/I.366.2 – MF-R2 dialled digit codes**

| Digit Code | Meaning |
|---|---|
| 00000 | Reserved |
| 00001 | 1 |
| 00010 | 2 |
| 00011 | 3 |
| 00100 | 4 |
| 00101 | 5 |
| 00110 | 6 |
| 00111 | 7 |
| 01000 | 8 |
| 01001 | 9 |
| 01010 | 10 |
| 01011 | 11 |
| 01100 | 12 |
| 01101 | 13 |
| 01110 | 14 |
| 01111 | 15 |
| 10000-11110 | Reserved |
| 11111 | Tone-off |

## K.3 Transmitter procedures

When a transmitter wishes to convey to the receiver the beginning of a dialled digit or pause, the dialled digits packet shall be sent with triple redundancy at intervals of 5 ms.

If a tone persists, every 500 ms thereafter the dialled digits packet shall be sent to refresh the play-out (with the Redundancy field coded as value 3).

If a new event is conveyed before the triple redundancy of a previous event has completed, the transmitter shall stop sending packets for the previous event, in order to avoid the interleaving of two different time stamps.

A User transmitting dialled digits should ensure that no more than 20 ms of DTMF, MF-R1, or MF-R2 tones are allowed to pass through the encoded audio path, so that differential delays between the two streams do not cause false double signals at the far end receiver.

If a transmitter detects multifrequency tones but is not capable of determining their level, it shall set the Signal Level field to a preset value.

NOTE – When both DTMF and MF-R1 dialled digits are enabled, discrimination conflicts may occur where frequency tolerances overlap for the symbol pairs {DTMF 2, MF-R1 4} and {DTMF 3, MF-R1 7}. The User may use the context of a call to resolve such ambiguities.

## K.4 Receiver procedures

A User receiving dialled digits is expected to regenerate dialled digit signals according to the parameters conveyed to the best of its ability. At most one signal shall be played at any given time.

Transitions are explicitly indicated, and a new dialled digit signal implicitly turns off the old one. Transitions to silence are explicit and triply redundant, like all others.

When regenerated, the two frequencies of a tone pair and their relative levels should be within the tolerances for the local environment. The total power level should be as indicated.

In order to make full use of triple redundancy without introducing extra delay variation, the SSCS receiver should wait before indicating dialled digits or pauses until such time as it should have received all three copies of a transition. Although transmitted three times, it only requires one packet to be received correctly for a new dialled digit or silence transition to be recognized.

A User receiving dialled digits should not filter the duration of transitions before reproducing them.

If despite triple redundancy one or more transitions are lost, a User shall continue to play out the preceding tone. It is the SSCS's option to indicate the end of a tone if no further dialled digit packets are received within a period of 2 seconds.

ANNEX L

**Packet format and procedures for channel-associated signalling bits**

## L.1 General

This annex defines the packet format and procedures that shall be used to transport Channel Associated Signalling (CAS) bits over an AAL type 2 connection as a separate, secondary information stream.

The concepts of CAS are defined in ITU-T G.704 – see 3.1/G.704 for the 1544 kbit/s interface and 5.1/G.704 for the 2048 kbit/s interface.

The transmission of CAS packets is optional and is enabled by an SSCS parameter of operation.

## L.2 Packet format

CAS packets are format type 3 and benefit from CRC-10 error detection. They make use of the common facilities for type 3 packets defined in clause 11, including triple redundancy. A time stamp provides accurate relative timing of transitions in the state of the CAS bits.

The format of CAS packets is shown in Figure L.1.

| 8 | 7 | 6 | 5 | 4 | 3 | 2 | 1 | |
|---|---|---|---|---|---|---|---|---|
| Redundancy | | Time Stamp | | | | | | 1 |
| | | | | | | | | 2 |
| RES | | | | A | B | C | D | 3 |
| Message Type = 000011 | | | | | | | | 4 |
| CRC-10 | | | | | | | | 5 |

RES Reserved (set to zero)

**Figure L.1/I.366.2 – Channel-associated signalling bits packet format**

The Time Stamp and Redundancy fields are coded according to the common facilities for type 3 packets defined in clause 11.

The fields designated A, B, C and D contain the current value of the corresponding CAS bits.

## L.3 Transmitter procedures

When a transmitter wishes to convey to the receiver a change in state of the ABCD bits, the CAS packet shall be sent with triple redundancy at intervals of 5 ms.

Every 5 seconds thereafter the CAS packet is sent to refresh the ABCD state (with the Redundancy field coded as value 3).

Procedures to debounce CAS bits at the transmitter are outside the scope of this Recommendation. Within certain limits, transient changes in the state of the CAS bits can be considered insignificant.

If a new state change is conveyed before the triple redundancy of a previous state change has completed, the transmitter shall stop sending packets for the previous state, in order to avoid the interleaving of two different time stamps.

If an external interface supplies fewer than four independent CAS bits, e.g. 1544 kbit/s with the 12-frame multiframe, the transmitting User shall aggregate and map the supplied bits to the four ABCD bits that the SSCS transfers. For example, the sequence {A, B, A', B'} should be transferred as C = A', D = B' and the sequence {A, A', A'', A'''} as B = A', C = A'', D = A'''.

## L.4 Receiver procedures

If a User is interpreting the semantics of signalling, it should filter out (debounce) insignificant transient changes in the state of the CAS bits.

In order to make full use of triple redundancy without introducing extra delay variation, the SSCS receiver should wait before indicating changes in the state of the CAS bits until such time as it should have received all three copies of a transition. Although transmitted three times, it only requires one packet to be received correctly for a change in the state of the CAS bits to be recognized.

# ANNEX M

# Packet format and procedures for facsimile demodulation

Facsimile demodulation is a more efficient way to transport the effective content of voiceband data calls that a signal classifier has concluded represent facsimile transmissions.

This annex defines packet formats and procedures to convey Group 3 facsimile traffic at bit rates up to 14.4 kbit/s.

NOTE – The support of V.34 modulation schemes is for further study.

In this annex, "low-speed" refers to the V.21 modulation of T.30 control information, and "high-speed" refers to the V.17, V.27 *ter*, V.29 or V.33 modulation of facsimile image data.

## M.1 Facsimile demodulation control concepts

### M.1.1 Message types

Modulation control messages and T.30 data are transported using type 3 packets (with the payload protected by a 10-bit CRC).

Codepoints for the following messages types are defined in Table 10-1:

- T.30_Preamble;
- EPT;
- Training;

• Fax_Idle;
• T.30_Data.

The packet formats for these messages are defined in M.2.

### M.1.2 Modulation control messages

T.30_Preamble, EPT, Training, and Fax_Idle are used to control transitions in the facsimile demodulation and remodulation process between the near-end and far-end Users of the SSCS.

### M.1.3 T.30 Data

T.30_Data is used to transport V.21-demodulated HDLC-framed facsimile control data bits between the near-end and far-end Users.

Whole HDLC frames, including zero-bit stuffing and interframe flags, are transmitted via the T.30_Data packet format. There is no HDLC flag removal after the leading T.30 preamble flags.

HDLC framed data is passed transparently even when errors are present (e.g. CRC failure); no CRC calculation is required. HDLC zero-bit stuffing is also passed transparently.

For protocol analysis, some information fields of the HDLC frame may optionally be altered to control the protocol, e.g. disabling non-standard facilities. This is an exception to the usual transparency of T.30 data and requires the module making the intervention to recalculate HDLC framing.

### M.1.4 Common facilities

Modulation control packets and T.30 data make use of the common facilities for type 3 packets defined in clause 11. In particular, a time stamp provides accurate relative timing for transition events.

Modulation control messages are sent with triple redundancy at an interval of 20 ms. Although transmitted three times, it only requires one modulation control packet to be received correctly for its content to be recognized and appropriate action taken.

T.30 data messages have their own unique redundancy scheme up until the end of low-speed modulation. Only the ending T.30 data octet is sent using the common facility, with triple redundancy at an interval of 20 ms. Although transmitted three times, it only requires one ending T.30 data packet to be received correctly for its content to be recognized and appropriate action taken.

### M.1.5 Time stamps

Modulation control and T.30 Data packets begin with a time stamp. The Time Stamp field of 14 bits is coded in units of milliseconds, with the most significant bits in the first octet. It represents the relative timing of events on the input to the demodulator. The time stamp is driven by the synchronous clock frequency associated with the Audio SAP. It wraps around after it reaches the maximum count.

### M.1.6 Sequence numbers

T.30_Data packets contain a 4-bit sequence number. The Sequence Number field is used to determine if a T.30_Data packet has been lost.

The first T.30_Data packet after a T.30_Preamble shall start with sequence number 0. The sequence number is incremented in every subsequent T.30_Data packet and wraps around to zero after it reaches the maximum count.

### M.1.7 Robustness against packet loss

Modulation control messages achieve robustness against packet loss by being sent three times. The time interval between the repeated packets is nominally 20 ms. The second and third transmission shall use the time stamp of the first packet transmitted. The Redundancy field is set to the values 0, 1 and 2 respectively in these three transmissions.

T.30 data robustness is achieved by the following octet roll-over method: each T.30_Data packet contains three data octets: the current octet (n), the previous octet (n–1) and the octet before the previous (n–2). In this way, the data octets are sent with triple redundancy. Each data octet appears at a shifted position in three successive packets. The time interval between T.30_Data packets is nominally 26.7 ms (the time to transmit 8 bits at a rate of 300 bit/s).

The ending T.30 data octet is sent without roll-over and using the common facility for triple redundancy at an interval of 20 ms.

## M.2 Facsimile demodulation control packets

### M.2.1 T.30_Preamble

This message is sent to the far end upon detection of T.30 preamble HDLC flags (provided that the User is in Facsimile Demodulation state).

The far end shall start to regenerate the preamble HDLC Flags upon reception of this message.

The T.30_Preamble packet format is shown in Figure M.1.

| 8 | 7 | 6 | 5 | 4 | 3 | 2 | 1 | |
|---|---|---|---|---|---|---|---|---|
| Redundancy | | Time Stamp | | | | | | 1 |
| | | | | | | | | 2 |
| Message Type = T.30_Preamble | | | | | | | | 3 |
| CRC-10 | | | | | | | | 4 |

**Figure M.1/I.366.2 – T.30_Preamble packet format**

### M.2.2 EPT

The Echo Protection Tone (EPT) signal may be sent by the facsimile terminal in front of high-speed modulation in order to disable echo cancellers. This signal has a duration of 185-200 ms and one of two frequencies, 1700 Hz or 1800 Hz.

The near-end User shall send an EPT tone-on message to the far end upon detection of an EPT tone. The EPT tone is turned off a specified time (20-25 ms) before the play-out of Training.

The far-end User shall reconstruct the EPT signal upon reception of this message.

The EPT packet format is shown in Figure M.2.

| 8 | 7 | 6 | 5 | 4 | 3 | 2 | 1 | |
|---|---|---|---|---|---|---|---|---|
| Redundancy | | Time Stamp | | | | | | 1 |
| | | | | | | | | 2 |
| RES | | | | EPT Frequency | | | | 3 |
| Message Type = EPT | | | | | | | | 4 |
| CRC-10 | | | | | | | | 5 |

RES Reserved (set to zero)

**Figure M.2/I.366.2 – EPT packet format**

The EPT Frequency field is coded according to Table M.1.

**Table M.1/I.366.2 – EPT frequency codepoints**

| EPT frequency | Meaning |
|---|---|
| 0000 | 1700 Hz |
| 0001 | 1800 Hz |
| 0010-1111 | Reserved |

### M.2.3 Training

The training signal is used by facsimile terminals to indicate the start of high-speed modulation.

Upon detection of training received from the facsimile terminal, the near-end User shall send a Training message to the far-end User.

Upon reception of a Training message, the far-end User shall start to generate the respective training sequence at its remodulator output.

The Training packet format is shown in Figure M.3.

| 8 | 7 | 6 | 5 | 4 | 3 | 2 | 1 | |
|---|---|---|---|---|---|---|---|---|
| Redundancy | | Time Stamp | | | | | | 1 |
| | | | | | | | | 2 |
| Modulation Type | | | | Modulation Rate | | | | 3 |
| Message Type = Training | | | | | | | | 4 |
| CRC-10 | | | | | | | | 5 |

**Figure M.3/I.366.2 – Training packet format**

The Modulation Type and Modulation Rate fields are coded according to Tables M.2 and M.3, respectively.

**Table M.2/I.366.2 – Modulation type codepoints**

| Modulation Type | Meaning |
|---|---|
| 0000 | V.27 *ter* |
| 0001 | V.29 |
| 0010 | V.17 long training |
| 0011 | V.17 short training |
| 0100 | V.33 |
| 0101-1111 | Reserved |

**Table M.3/I.366.2 – Modulation rate codepoints**

| Modulation Rate | Meaning (bit/s) |
|---|---|
| 0000 | Unknown rate |
| 0001 | 2 400 |
| 0010 | 4 800 |
| 0011 | 7 200 |
| 0100 | 9 600 |
| 0101 | 12 000 |
| 0110 | 14 400 |
| 0111-1111 | Reserved |

For V.17 modulation, two types of training sequence exist. The training sequence is long before the training check (TCF) and short before page data. The modulation rate and modulation type (short or long) cannot be deduced from the beginning of the training sequence itself.

Protocol analysis can predict the V.17 training sequence, based on its understanding of T.30 protocol. But waveform analysis needs additional time before acquiring the full details of type and rate.

Therefore, upon detecting the beginning of a V.17 training sequence, a V.17 long Training message with Modulation Rate set to "unknown rate" may be sent. Signal analysis shall determine the type and the rate of the training sequence later and shall generate an additional short or long Training message with a specific Modulation Rate.

If a short Training message is received by the far-end User while generating the V.17 long training sequence with unknown rate, it shall change the training to a short sequence, followed by scrambled ones using the Modulation Rate indicated in the message.

### M.2.4 Fax_Idle

The Fax_Idle message shall be sent upon detecting that the high-speed modulation data from the local facsimile terminal has terminated.

The Fax_Idle packet format is shown in Figure M.4.

| 8 | 7 | 6 | 5 | 4 | 3 | 2 | 1 | |
|---|---|---|---|---|---|---|---|---|
| Redundancy | | Time Stamp | | | | | | 1 |
| | | | | | | | | 2 |
| Message Type = Fax_Idle | | | | | | | | 3 |
| CRC-10 | | | | | | | | 4 |

**Figure M.4/I.366.2 – Fax_Idle packet format**

### M.2.5 T.30_Data

T.30 data consists of successive octets of V.21-demodulated HDLC frames. The demodulated octets are transferred without undoing HDLC framing. Interframe flags, zero-bit stuffing, and Frame Check Sequence are transferred intact.

The T.30_Data packet format is shown in Figure M.5.

| 8 | 7 | 6 | 5 | 4 | 3 | 2 | 1 | |
|---|---|---|---|---|---|---|---|---|
| Redundancy | | Time Stamp | | | | | | 1 |
| | | | | | | | | 2 |
| End | BILO | | | Sequence Number | | | | 3 |
| Data(n) | | | | | | | | 4 |
| Data(n−1) | | | | | | | | 5 |
| Data(n−2) | | | | | | | | 6 |
| Message Type = T.30_Data | | | | | | | | 7 |
| CRC-10 | | | | | | | | 8 |

BILO Bits in last octet

**Figure M.5/I.366.2 – T.30_Data packet format**

Each packet contains three successive octets as follows:

– Data(n) = current data octet;

– Data(n−1) = preceding data octet;

– Data(n−2) = data octet before preceding data octet.

In each octet, the first bit transmitted is the most significant bit.

The End field is coded according to Table M.4.

**Table M.4/I.366.2 – End marker for T.30 data**

| End | BILO | Significance | Redundancy |
|---|---|---|---|
| 0 | Reserved | More T.30 data is to come | 3 |
| 1 | Bits in last octet – 1 | Data(n) is the last octet of T.30 data | 0, 1, 2 |

The BILO field indicates how many bits, beginning with the most significant bit, are valid in the last octet of T.30 data:

BILO = (number of valid bits in the Data(n) octet of the End packet) – 1

This field may be considered optional because the addition of several garbage bits to the end of the V.21 signal does not have any impact on facsimile terminals. When not used the BILO field shall be set to the value 7 (all bits valid).

Receipt of a T.30_Data packet with the End bit set to one signifies to the far-end User that V.21 modulation should be turned off after the last valid data bit has been played out.

The Sequence Number field shall be set to zero in the first T.30_Data packet after T.30_Preamble. In the first packet transmitted the Data(n−1) and Data(n−2) octets shall be set to 0111 1110.

The Sequence Number is incremented in following T.30_Data packets and wraps to zero around after it reaches the maximum count.

The End packet, which contains the last octet of the V.21 signal in Data(n), shall increment its sequence number as usual from the preceding T.30_Data packet transmitted. This packet shall then be frozen and be transmitted three times for redundancy at an interval of 20 ms. The Sequence Number and the End indication shall be kept constant during these three transmissions. The Redundancy field shall have values 0, 1 and 2 respectively for these three transmissions.

The Redundancy field shall have value 3 in all but the End packet.

## M.3 Facsimile image data packets

Facsimile image data is sent in type 1 packets. The time interval between the packets is nominally 20 ms. The number of data octets in a packet depends on the high-speed modulation rate.

The training check sequence (TCF) is transmitted end-to-end in the same way as ordinary page data.

The image data packet does not include an explicit sequence number. Instead, the sequence number is derived from the UUI field in the packet header (codepoints 0-15). The sequence number interval is 20 ms and the modulus is 16.

As image data bits are output by the demodulator, the near-end User shall packetize them in 20 ms (nominal) intervals and send them to the far-end User. The actual packet time is derived from the incoming facsimile bit rate and may vary slightly due to tolerances in the modulation. The number of octets in the transmitted packets depends on the modulation type and is shown in Table M.5.

**Table M.5/I.366.2 – Image data packet lengths**

| Facsimile bit rate (bit/s) | Packet length (octets) |
|---|---|
| 2 400 | 6 |
| 4 800 | 12 |
| 7 200 | 18 |
| 9 600 | 24 |
| 12 000 | 30 |
| 14 400 | 36 |

NOTE – For V.34 facsimile demodulation, a nominal packet size of 10 ms would be required; the packet length for 33 600 bit/s is then 42 octets. Support of V.34 facsimile demodulation is for further study.

Upon detecting that the high-speed modulation data from the local facsimile terminal has terminated, the near-end User shall conclude image data packetization and shall send the last image data packet even if the packet length has not reached the values given in Table M.5. Any bits remaining in the last octet to be sent shall be filled with ones. Following this, a Fax_Idle message shall be sent (with triple redundancy) to the far-end User.

The facsimile image data packet format is shown in Figure M.6. The first bit transmitted is the most significant bit of the first octet.

| 8 7 6 5 4 3 2 1 | |
|---|---|
| Data 1 | 1 |
| Data 2 | 2 |
| . . . | . . . |
| Data N | N |

**Figure M.6/I.366.2 – Facsimile image data packet format**

ANNEX N

## Packet format and procedures for OAM (alarms and loopback)

## N.1 General

Alarm indications and loopback are defined by Table N.1.

**Table N.1/I.366.2 – OAM Signals**

| Signal | Description | Reference |
|---|---|---|
| External AIS | Alarm Indication Signal – a signal, associated with a maintenance alarm detected on a defective maintenance span, that is transmitted in the direction of the defect as a substitute for the normal signal. Its purpose is to show downstream entities that a defect has been identified, so that other maintenance alarms consequent to this first defect can be inhibited. The external bit stream representation of AIS may be an all-1s signal. | 5.4.2 a)/M.20 |
| External RAI | Remote Alarm Indication – a signal transmitted upstream from a terminal that has detected defects persisting long enough to constitute a received signal failure. Its purpose is to report in the backward direction that there is an interruption of service in the forward direction. | 2.1.3.1.3/G.704; Table 5A/G.704 Note 3; Table 14/G.704 Note 4 |
| AAL type 2 connection AIS (internal) | Alarm Indication Signal – a signal transmitted in the downstream direction from the AAL type 2 connecting point that first detects a defect affecting the AAL type 2 connection; this includes defects indicated by lower layers. | 6.2.2.1.1.1/I.610 |
| AAL type 2 connection RDI (internal) | Remote Defect Indication – a signal transmitted upstream by an AAL type 2 endpoint that is in an alarm state as the result of having received an AAL type 2 connection AIS or having detected a defect that affects the AAL type 2 connection. | 6.2.2.1.1.2/I.610 |
| Loopback | For on-demand connectivity monitoring.<br>For fault localization.<br>For pre-service connectivity verification. | N/A |
| NOTE 1 – Requirements are being studied for OAM flows within AAL type 2. The material of this annex may in the future be moved to a separate Recommendation, maintaining compatibility with the formats and procedures described here.<br>NOTE 2 – Other potentially useful features of future OAM flows for AAL type 2 may be: continuity check, performance management in terms of AAL type 2 packets, and activation/deactivation. | | |

## N.2 Packet format

OAM packets are sent using UUI codepoint value 31. The OAM Type and Function Type fields are coded according to Table N.2.

NOTE – UUI and LI fields are carried in the AAL type 2 CPS-Packet header as shown in Figure 10-2. In the future, OAM-related packets could be made variable length up to 45 octets.

**Table N.2/I.366.2 – Coding of OAM type and function type fields**

| OAM type | Coding | Function type | Coding |
|---|---|---|---|
| External alarms | 1100 | External AIS | 0000 |
| | | External RAI | 0001 |
| Fault management | 0001 | AAL type 2 connection AIS | 0000 |
| | | AAL type 2 connection RDI | 0001 |
| | | AAL type 2 loopback (LB) | 1000 |
| Reserved | other values | Reserved | other values |

### N.2.1 Alarm Packet format

Alarm indications are conveyed in a type 3 packet whose format is defined by Figure N.1. They do not make use of the common facilities for type 3 packets.

| 8 | 7 | 6 | 5 | 4 | 3 | 2 | 1 |
|---|---|---|---|---|---|---|---|
| OAM Type | | | | Function Type | | | |
| Message Type = 000000 | | | | | | CRC-10 | |
| CRC-10 | | | | | | | |

**Figure N.1/I.366.2 – Alarm packet format**

### N.2.2 Loopback Packet format

Loopback packets are conveyed in a type 3 packet whose format is defined by Figure N.2. They do not make use of the common facilities for type 3 packets.

– Loopback indication (LBI) (1 bit): This bit provides a boolean indication as to whether or not the CPS-Packet has already been looped back. The field confirms that the loopback has occurred at the CID and avoids the problem of infinite loopback. The source point encodes this field as 1. The loopback point changes the encoding to 0.

| 8 | 7 | 6 | 5 | 4 | 3 | 2 | 1 |
|---|---|---|---|---|---|---|---|
| OAM Type | | | | Function Type | | | |
| LBI | Reserved | | | | | | |
| Message Type = 000000 | | | | | | CRC-10 | |
| CRC-10 | | | | | | | |

**Figure N.2/I.366.2 – Loopback packet format**

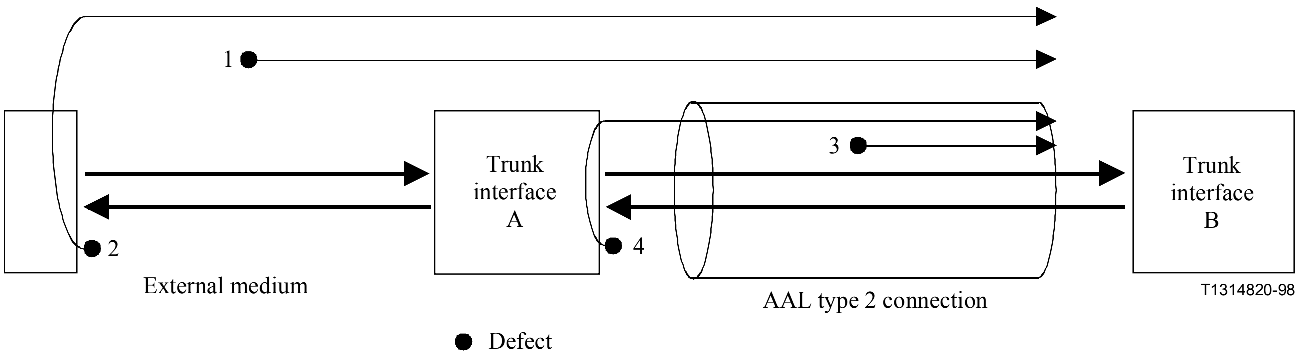
## N.3 Procedures

### N.3.1 Alarm procedures

An alarm condition is asserted when the corresponding alarm packet is generated.

For as long as a condition persists, an alarm packet shall be transmitted at least once per second. If a time interval of 3.5 seconds elapses at a receiver without reassertion of a signal, this shall be interpreted as removal of the corresponding condition. In addition, receipt of any packet except an AAL type 2 connection AIS alarm packet shall signify removal of an AAL type 2 connection AIS alarm condition.

Figure N.3 illustrates that a trunk interface A, at one end of an AAL type 2 connection, has the ability to send alarm packets to trunk interface B at the other end of the AAL type 2 connection. Such packets may be sent as a consequence of failures occurring in one of the three other streams appearing at trunk interface A. The coding of the OAM type and Function type fields indicates in which direction the defect occurred and whether it was external or internal to the AAL type 2 connection.




| **Defect at:** | **Correlates with OAM Type and Function Type:** |
|---|---|
| 1 | External AIS |
| 2 | External RAI |
| 3 | AAL type 2 connection AIS |
| 4 | AAL type 2 connection RDI |

**Figure N.3/I.366.2 – Defects correlated with alarm indications**

In Figure N.3, defects at **1** and **2** may only be detected on the basis of an entire trunk group, which may map to multiple AAL type 2 connections. In that case, alarm packets for the external AIS or RAI shall be sent on each individual AAL type 2 connection that is affected.

Defects at **3** are detected and the corresponding alarm packets are generated by an AAL type 2 connecting point, not by the trunk interface, which is an AAL type 2 endpoint.

### N.3.2 Loopback procedures

A loopback may be inserted within an AAL type 2 access network on individual AAL type 2 connections from the AAL type 2 local exchange toward the a remote AAL type 2 NT as shown in Figure N.4. The loopback initiating AAL type 2 Local exchange sends a LB packet with the LBI set to 0. Upon receiving the LB packet, the AAL type 2 NT sends a LB packet to the initiating AAL type 2 local exchange with the LBI set to 1.

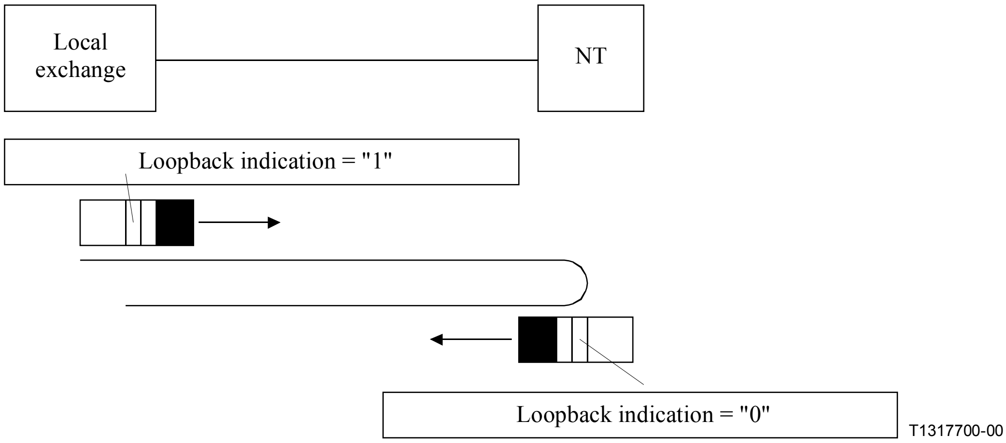



**Figure N.4/I.366.2 – CID Loopback with LBI function**

NOTE – The loopback procedures in this annex are intended for Non-Switched access applications of AAL type 2 with a fixed relationship between individual local exchanges and NTs as defined in Annex A/Q.2630.1. Switched access applications are beyond the scope of this annex.

The waiting time between the transmission of successive LB packets on a connection shall be 5 seconds. It shall be considered unsuccessful if the loopback cell is not returned to the originating point within 5 seconds.

# ANNEX O

# Packet format and procedures for user state control

## O.1 General

The SSCS supports the following narrow-band telephone services at the Audio SAP:

- Voice;
- Voiceband data;
- Circuit Mode (64 kbit/s);
- Facsimile demodulation.

The User will apply different procedures and encoding algorithms to each of these services. User states are needed in order to distinguish between voice, voiceband data, circuit mode, and facsimile demodulation handling.

User state control messages are used to communicate state control to the peer User. The SSCS provides reliable transfer by using the common facility for type 3 packets of triple redundancy.

User states are determined locally by signal classifier decisions and by considering communication from the peer User in the form of state control messages. Symmetry and collisions of User state changes are the User's responsibility.

## O.2 Packet format

User state control packets are format type 3 and benefit from CRC-10 error detection. They make use of the common facilities for type 3 packets defined in clause 11, including triple redundancy.

In this case, the time stamp is not needed for accurate relative timing of requests and responses. It is used instead as a basis for filtering out and suppressing redundant primitive indications and confirms to the receiving User.

The format of user state control packets is shown in Figure O.1.

<table>
<tr><td>8</td><td>7</td><td>6</td><td>5</td><td>4</td><td>3</td><td>2</td><td>1</td><td></td></tr>
<tr><td colspan="2">Redundancy</td><td colspan="6">Time Stamp</td><td>1</td></tr>
<tr><td colspan="8"></td><td>2</td></tr>
<tr><td>Type</td><td colspan="4">User State</td><td>Ack</td><td colspan="2">RES</td><td>3</td></tr>
<tr><td colspan="5">RES</td><td>V.17</td><td>V.27 ter</td><td>V.29</td><td>4</td></tr>
<tr><td colspan="6">Message Type = 000001</td><td colspan="2"></td><td>5</td></tr>
<tr><td colspan="8">CRC-10</td><td>6</td></tr>
</table>

RES Reserved (set to zero)

**Figure O.1/I.366.2 – User state control packet format**

The Time Stamp and Redundancy fields are coded according to the common facilities for type 3 packets defined in clause 11. Redundancy value 3 is not used.

The Type, User State, Ack, V.17, V.27 *ter* and V.29 fields are coded according to Tables O.1 through O.4.

**Table O.1/I.366.2 – Type codes**

| Type | Meaning |
|---|---|
| 0 | Request |
| 1 | Response |

**Table O.2/I.366.2 – User state codes**

| User State | Meaning |
|---|---|
| 0000 | Voice |
| 0001 | Voiceband data |
| 0010 | Circuit mode |
| 0011 | Facsimile demodulation |
| 0100-1111 | Reserved |

The Ack field is significant only within a response packet, i.e. Type = 1.

**Table O.3/I.366.2 – Ack codes**

| Ack | Meaning |
|---|---|
| 0 | Reject |
| 1 | Accept |

The V.17, V.27 *ter* and V.29 fields are significant only within a state change to facsimile demodulation, i.e. User State = 0011. These bits are set independently of one another to indicate the capability to demodulate and remodulate facsimile image data using the corresponding modulation. Each User shall declare its own capabilities in the request or response packet, respectively. At the end of a state change, each User will possess the same information and can deduce which capabilities are common to both Users.

**Table O.4/I.366.2 – V.17, V.27 *ter* and V.29 codes**

| Value | Meaning |
|---|---|
| 0 | Not supported |
| 1 | Supported |

## O.3 Procedures

Robustness against packet loss is achieved by repeating each user state control message three times. The time interval between the repeated packets is 20 ms. The second and third transmission shall use the time stamp of the first packet transmitted. The time stamp of a response is unrelated to the time stamp of the request.

The receiving SSCS shall filter out repetitions and extend the appropriate primitive type (indication or confirm) to its User.

The procedures are described by Figure O.2.

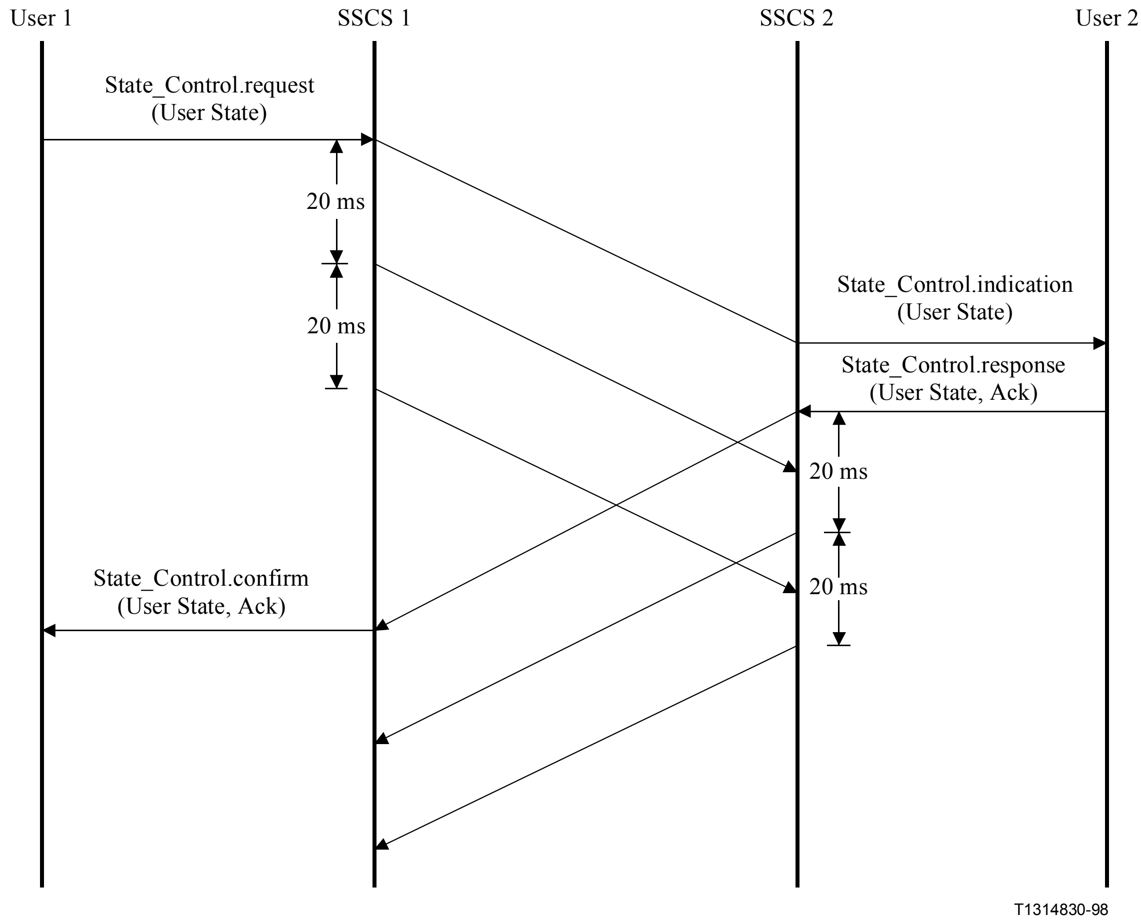



**Figure O.2/I.366.2 – User state control message flow**

## ANNEX P

### Predefined encoding format profiles

This annex defines a number of ITU-T predefined profiles for use by audio information streams, which use UUI codepoints 0-15 with type 1 packets. By making reference to the identifiers of these profiles, the transmitter and receiver can agree on one of the major operating parameters of the SSCS.

This Recommendation does not include procedures for the use of these identifiers. Such procedures may be the subject of other Recommendations which should also allow for the use of non-ITU-T predefined profiles.

Inclusion in this annex does not imply that all implementations are required to support every profile. An implementation may choose to support any or none of the profiles defined here, except for the mandatory profile specified in 13.4. In addition, an implementation may support one or more profiles defined elsewhere.

Figure P.1 lists the assigned ITU-T standard codes to be used for identification of predefined profiles, and the remaining tables define the individual profiles. The definition of each profile includes the following information for each entry:

- profile entry index;
- UUI codepoint range;
- packet length;
- reference to a figure depicting the encoding data unit format;
- description of algorithm;
- value of M, the number of service data units in a packet;
- packet time;
- sequence number interval.

| Identifier | Description of Profile | Reference |
|---|---|---|
| 0 | Not used | – |
| 1 | PCM-64 | Table P.1 |
| 2 | PCM-64 and silence | Table P.2 |
| 3 | ADPCM and silence | Table P.3 |
| 4 | G.728 with higher efficiency | Table P.4 |
| 5 | G.728 with lower delay | Table P.5 |
| 6 | G.729 with higher efficiency and G.726 for voiceband data | Table P.6 |
| 7 | G.729 with lower delay | Table P.7 |
| 8 | G.729 with lower delay and G.726-32 for voiceband data at lower rates | Table P.8 |
| 9 | G.729 with lower delay and G.726-40 for voiceband data at higher rates | Table P.9 |
| 10 | G.729 with full variable bit rates | Table P.10 |
| 11 | AMR | Table P.11 |
| 12 | G.723 | Table P.12 |
| 13 | PCM 64 kbits/s and ADPCM 32 kbits/s | Table P.13 |
| 14-255 | Reserved for future ITU-T assignment | – |

**Figure P.1/I.366.2 – Identifiers for ITU-T predefined profiles**

**Table P.1/I.366.2 – Profile using PCM-64**

| Profile entry index | UUI codepoint range | Packet length (octets) | Encoding format reference | Description of algorithm | M | Packet time (ms) | Sequence number interval (ms) |
|---|---|---|---|---|---|---|---|
| 0 | 0-15 | 40 | Figure B.1 | PCM, G.711-64, generic | 1 | 5 | 5 |

**Table P.2/I.366.2 – Profile using PCM-64 and silence**

| Profile entry index | UUI codepoint range | Packet length (octets) | Encoding format reference | Description of algorithm | M | Packet time (ms) | Sequence number interval (ms) |
|---|---|---|---|---|---|---|---|
| 0 | 0-15 | 40 | Figure B.1 | PCM, G.711-64, generic | 1 | 5 | 5 |
| – | 0-15 | 1 | Figure I.1 | Generic SID | 1 | 5 | 5 |

**Table P.3/I.366.2 – Profile using ADPCM and silence**

| Profile entry index | UUI codepoint range | Packet length (octets) | Encoding format reference | Description of algorithm | M | Packet time (ms) | Sequence number interval (ms) |
|---|---|---|---|---|---|---|---|
| 0 | 0-15 | 40 | Figure B.1 | PCM, G.711-64, generic | 1 | 5 | 5 |
| 1 | 0-15 | 25 | Figure E.1 | ADPCM, G.726-40 | 1 | 5 | 5 |
| 2 | 0-15 | 20 | Figure E.2 | ADPCM, G.726-32 | 1 | 5 | 5 |
| 3 | 0-15 | 15 | Figure E.3 | ADPCM, G.726-24 | 1 | 5 | 5 |
| 4 | 0-15 | 10 | Figure E.4 | ADPCM, G.726-16 | 1 | 5 | 5 |
| – | 0-15 | 1 | Figure I.1 | Generic SID | 1 | 5 | 5 |

**Table P.4/I.366.2 – Profile using G.728 with higher efficiency**

| Profile entry index | UUI codepoint range | Packet length (octets) | Encoding format reference | Description of algorithm | M | Packet time (ms) | Sequence number interval (ms) |
|---|---|---|---|---|---|---|---|
| 0 | 0-15 | 40 | Figure B.1 | PCM, G.711-64, generic | 1 | 5 | 5 |
| 1 | 0-15 | 20 | Figure G.1 | LD-CELP, G.728-16 | 2 | 10 | 5 |
| 2 | 0-15 | 16 | Figure G.1 | LD-CELP, G.728-12.8 | 2 | 10 | 5 |
| 3 | 0-15 | 12 | Figure G.1 | LD-CELP, G.728-9.6 | 2 | 10 | 5 |
| 4 | 0-15 | 10 | Figure G.1 | LD-CELP, G.728-16 | 1 | 5 | 5 |
| 5 | 0-15 | 8 | Figure G.2 | LD-CELP, G.728-12.8 | 1 | 5 | 5 |
| 6 | 0-15 | 6 | Figure G.3 | LD-CELP, G.728-9.6 | 1 | 5 | 5 |
| – | 0-15 | 1 | Figure I.1 | Generic SID | 1 | 5 | 5 |

**Table P.5/I.366.2 – Profile using G.728 with lower delay**

| Profile entry index | UUI codepoint range | Packet length (octets) | Encoding format reference | Description of algorithm | M | Packet time (ms) | Sequence number interval (ms) |
|---|---|---|---|---|---|---|---|
| 0 | 0-15 | 40 | Figure B.1 | PCM, G.711-64, generic | 1 | 5 | 5 |
| 1 | 0-15 | 10 | Figure G.1 | LD-CELP, G.728-16 | 1 | 5 | 5 |
| 2 | 0-15 | 8 | Figure G.2 | LD-CELP, G.728-12.8 | 1 | 5 | 5 |
| 3 | 0-15 | 6 | Figure G.3 | LD-CELP, G.728-9.6 | 1 | 5 | 5 |
| – | 0-15 | 1 | Figure I.1 | Generic SID | 1 | 5 | 5 |

**Table P.6/I.366.2 – Profile using G.729 with higher efficiency and G.726 for voiceband data**

| Profile entry index | UUI codepoint range | Packet length (octets) | Encoding format reference | Description of algorithm | M | Packet time (ms) | Sequence number interval (ms) |
|---|---|---|---|---|---|---|---|
| 0 | 0-15 | 40 | Figure B.1 | PCM, G.711-64, generic | 1 | 5 | 5 |
| 1 | 0-15 | 25 | Figure E.1 | ADPCM, G.726-40 | 1 | 5 | 5 |
| 2 | 0-15 | 20 | Figure H.1 | CS-ACELP, G.729-8 | 2 | 20 | 5 |
| 3 | 0-15 | 16 | Figure H.3 | CS-ACELP, G.729-6.4 | 2 | 20 | 5 |
| 4 | 0-15 | 10 | Figure H.1 | CS-ACELP, G.729-8 | 1 | 10 | 5 |
| 5 | 0-15 | 8 | Figure H.3 | CS-ACELP, G.729-6.4 | 1 | 10 | 5 |
| – | 0-15 | 2 | Figure H.2 | G.729 SID | 1 | 10 | 5 |

**Table P.7/I.366.2 – Profile using G.729 with lower delay**

| Profile entry index | UUI codepoint range | Packet length (octets) | Encoding format reference | Description of algorithm | M | Packet time (ms) | Sequence number interval (ms) |
|---|---|---|---|---|---|---|---|
| 0 | 0-15 | 40 | Figure B.1 | PCM, G.711-64, generic | 1 | 5 | 5 |
| 1 | 0-15 | 10 | Figure H.1 | CS-ACELP, G.729-8 | 1 | 10 | 5 |
| – | 0-15 | 2 | Figure H.2 | G.729 SID | 1 | 10 | 5 |

**Table P.8/I.366.2 – Profile using G.729 with lower delay and G.726-32 for voiceband data at lower rates**

| Profile entry index | UUI codepoint range | Packet length (octets) | Encoding format reference | Description of algorithm | M | Packet time (ms) | Sequence number interval (ms) |
|---|---|---|---|---|---|---|---|
| 0 | 0-15 | 40 | Figure B.1 | PCM, G.711-64, generic | 1 | 5 | 5 |
| 1 | 0-15 | 20 | Figure E.2 | ADPCM, G.726-32 | 1 | 5 | 5 |
| 2 | 0-15 | 10 | Figure H.1 | CS-ACELP, G.729-8 | 1 | 10 | 5 |
| – | 0-15 | 2 | Figure H.2 | G.729 SID | 1 | 10 | 5 |

**Table P.9/I.366.2 – Profile using G.729 with lower delay and G.726-40 for voiceband data at higher rates**

| Profile entry index | UUI codepoint range | Packet length (octets) | Encoding format reference | Description of algorithm | M | Packet time (ms) | Sequence number interval (ms) |
|---|---|---|---|---|---|---|---|
| 0 | 0-15 | 40 | Figure B.1 | PCM, G.711-64, generic | 1 | 5 | 5 |
| 1 | 0-15 | 25 | Figure E.1 | ADPCM, G.726-40 | 1 | 5 | 5 |
| 2 | 0-15 | 10 | Figure H.1 | CS-ACELP, G.729-8 | 1 | 10 | 5 |
| 3 | 0-15 | 8 | Figure H.3 | CS-ACELP, G.729-6.4 | 1 | 10 | 5 |
| – | 0-15 | 2 | Figure H.2 | G.729 SID | 1 | 10 | 5 |

**Table P.10/I.366.2 – Profile using G.729 with full variable bits rates**

| Profile entry index | UUI codepoint range | Packet length (octets) | Encoding format reference | Description of algorithm | M | Packet time (ms) | Sequence number interval (ms) |
|---|---|---|---|---|---|---|---|
| 0 | 0-15 | 40 | Figure B.1 | PCM, G.711-64, generic | 1 | 5 | 5 |
| 1 | 0-15 | 30 | Figure H.4 | CS-ACELP, G.729-12 | 2 | 20 | 5 |
| 2 | 0-15 | 20 | Figure H.1 | CS-ACELP, G.729-8 | 2 | 20 | 5 |
| 3 | 0-15 | 16 | Figure H.3 | CS-ACELP, G.729-6.4 | 2 | 20 | 5 |
| 4 | 0-15 | 15 | Figure H.4 | CS-ACELP, G.729-12 | 1 | 10 | 5 |
| 5 | 0-15 | 10 | Figure H.1 | CS-ACELP, G.729-8 | 1 | 10 | 5 |
| 6 | 0-15 | 8 | Figure H.3 | CS-ACELP, G.729-6.4 | 1 | 10 | 5 |
| – | 0-15 | 2 | Figure H.2 | G.729 SID | 1 | 10 | 5 |

**Table P.11/I.366.2 – Profile using AMR**

| Profile entry index | UUI codepoint range | Packet length (octets) | Encoding format reference | Description of algorithm | M | Packet time (ms) | Seq. No. Interval (ms) |
|---|---|---|---|---|---|---|---|
| 0 | 0-7 | 31 | Figure Q.1 | AMR 12.2 | 1 | 20 | 20 |
| 0 | 8-15 | 31 | Figure Q.1 | AMR 12.2 (errored) | 1 | 20 | 20 |
| 1 | 0-7 | 26 | Figure Q.2 | AMR 10.2 | 1 | 20 | 20 |
| 1 | 8-15 | 26 | Figure Q.2 | AMR 10.2 (errored) | 1 | 20 | 20 |
| 2 | 0-7 | 21 | Figure Q.3 | AMR 7.95 | 1 | 20 | 20 |
| 2 | 8-15 | 21 | Figure Q.3 | AMR 7.95 (errored) | 1 | 20 | 20 |
| 3 | 0-7 | 19 | Figure Q.4 | AMR 7.4 | 1 | 20 | 20 |
| 3 | 8-15 | 19 | Figure Q.4 | AMR 7.4 (errored) | 1 | 20 | 20 |
| 4 | 0-7 | 18 | Figure Q.5 | AMR 6.7 | 1 | 20 | 20 |
| 4 | 8-15 | 18 | Figure Q.5 | AMR 6.7 (errored) | 1 | 20 | 20 |
| 5 | 0-7 | 16 | Figure Q.6 | AMR 5.9 | 1 | 20 | 20 |
| 5 | 8-15 | 16 | Figure Q.6 | AMR 5.9 (errored) | 1 | 20 | 20 |
| 6 | 0-7 | 14 | Figure Q.7 | AMR 5.15 | 1 | 20 | 20 |
| 6 | 8-15 | 14 | Figure Q.7 | AMR 5.15 (errored) | 1 | 20 | 20 |
| 7 | 0-7 | 13 | Figure Q.8 | AMR 4.75 | 1 | 20 | 20 |
| 7 | 8-15 | 13 | Figure Q.8 | AMR 4.75 (errored) | 1 | 20 | 20 |
| – | 0-15 | 2 | Figure Q.9 | AMR SID_First | 1 | – | 20 |
| – | 0-7 | 6 | Figure Q.10 | AMR SID_Update | 1 | 160 | 160 |
| – | 8-15 | 6 | Figure Q.10 | AMR SID_Update (errored) | 1 | 160 | 160 |

**Table P.12/I.366.2 – Profile using G.723**

| Profile entry index | UUI codepoint range | Packet length (octets) | Encoding format reference | Description of algorithm | M | Packet time (ms) | Sequence number interval (ms) |
|---|---|---|---|---|---|---|---|
| 0 | 0-15 | 24 | Figure D.1 | G.723.1-6.4 | 1 | 30 | 5 |
| 1 | 0-15 | 20 | Figure D.2 | G.723.1-5.3 | 1 | 30 | 5 |
| 2 | 0-15 | 4 | Figure D.3 | G.723.1 SID | 1 | 30 | 5 |

**Table P.13/I.366.2 – Profile using PCM 64 kbits/s and ADPCM 32 kbits/s**

| Profile entry index | UUI codepoint range | Packet length (octets) | Encoding format reference | Description of algorithm | M | Packet time (ms) | Sequence number interval (ms) |
|---|---|---|---|---|---|---|---|
| 0 | 0-7 | 40 | Figure B.1 | PCM, G.711-64, generic | 1 | 5 | 5 |
| 1 | 8-15 | 40 | Figure E.2 | ADPCM, G.726-32 | 2 | 10 | 5 |
| 2 | 8-15 | 20 | Figure E.2 | ADPCM, G.726-32 | 1 | 5 | 5 |
| – | 0-15 | 1 | Figure I.1 | Generic SID | 1 | 5 | 5 |

ANNEX Q

**Encoding format for audio algorithm AMR**

## Q.1 General

The AMR speech coder consists of the multi-rate speech coder, a source controlled rate scheme including a voice activity detector and a comfort noise generation system, and an error concealment mechanism to combat the effects of transmission errors and lost packets.

The multi-rate speech coder is a single integrated speech codec with eight-source rates from 4.75 kbit/s to 12.2 kbit/s, and a low bit rate background noise encoding mode. The speech coder is capable of switching its bit-rate every 20 ms speech frame upon command.

Encoded values are represented in the SSCS according to the conventions of I.361, whereby earlier octets and higher-numbered bits are more significant.

## Q.2 Encoding Data Unit

The detailed allocation of the bits in the adaptive multi-rate speech encoder is shown for each mode in Figures Q.1 to Q.8 based on the information in Tables Q.1a/b to Q.8a/b respectively. Tables Q.1a to Q.8a ("a" suffixes) show the order of the bits produced by the speech encoder. In these tables the MSB is always sent first. Tables Q.1b to Q.8b ("b" suffixes) are used to reorder the bit sequence generated by the speech encoder. The ordering algorithm is described in pseudo-code as:

*for* $j = 0$ *to* $K - 1$

$d(j) := s(table_m(j) + 1);$

where $table_m(j)$ refers to the table relevant to specific AMR mode $m$=0..7. An AMR mode directly maps to the "profile entry index" in profile P11. Tables Q.1b to Q.8b should be read line by line from left to right. The first element of the table has the index 0.

The size of the speech frames of the AMR is not octet aligned for all modes of operation. For this reason bit stuffing is used in order to achieve octet structure for the AMR frame. Stuffing bits are referred as UB (unused bits) in the tables and figures below.

**Table Q.1a/I.366.2 – Source encoder output parameters in order of occurrence and bit allocation within the speech frame of 244 bits/20 ms, 12.2 kbit/s mode**

| Bits (MSB-LSB) | Description |
|---|---|
| s1-s7 | index of 1st LSF submatrix |
| s8-s15 | index of 2nd LSF submatrix |
| s16-s23 | index of 3rd LSF submatrix |
| s24 | sign of 3rd LSF submatrix |
| s25-s32 | index of 4th LSF submatrix |
| s33-s38 | index of 5th LSF submatrix |
| Subframe 1 | |
| s39-s47 | adaptive codebook index |
| s48-s51 | adaptive codebook gain |
| s52 | sign information for 1st and 6th pulses |
| s53-s55 | position of 1st pulse |
| s56 | sign information for 2nd and 7th pulses |
| s57-s59 | position of 2nd pulse |
| s60 | sign information for 3rd and 8th pulses |
| s61-s63 | position of 3rd pulse |
| s64 | sign information for 4th and 9th pulses |
| s65-s67 | position of 4th pulse |
| s68 | sign information for 5th and 10th pulses |
| s69-s71 | position of 5th pulse |
| s72-s74 | position of 6th pulse |
| s75-s77 | position of 7th pulse |
| s78-s80 | position of 8th pulse |
| s81-s83 | position of 9th pulse |
| s84-s86 | position of 10th pulse |
| s87-s91 | fixed codebook gain |
| Subframe 2 | |
| s92-s97 | adaptive codebook index (relative) |
| s98-s141 | same description as s48-s91 |
| Subframe 3 | |
| s142-s194 | same description as s39-s91 |
| Subframe 4 | |
| s195-s244 | same description as s92-s141 |

**Table Q.1b/I.366.2 – Ordering of the speech encoder bits for the 12.2 kbit/s mode: $table_7(j)$**

| | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|
| 0 | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 | 9 |
| 10 | 11 | 12 | 13 | 14 | 23 | 15 | 16 | 17 | 18 |
| 19 | 20 | 21 | 22 | 24 | 25 | 26 | 27 | 28 | 38 |
| 141 | 39 | 142 | 40 | 143 | 41 | 144 | 42 | 145 | 43 |
| 146 | 44 | 147 | 45 | 148 | 46 | 149 | 47 | 97 | 150 |
| 200 | 48 | 98 | 151 | 201 | 49 | 99 | 152 | 202 | 86 |
| 136 | 189 | 239 | 87 | 137 | 190 | 240 | 88 | 138 | 191 |
| 241 | 91 | 194 | 92 | 195 | 93 | 196 | 94 | 197 | 95 |
| 198 | 29 | 30 | 31 | 32 | 33 | 34 | 35 | 50 | 100 |
| 153 | 203 | 89 | 139 | 192 | 242 | 51 | 101 | 154 | 204 |
| 55 | 105 | 158 | 208 | 90 | 140 | 193 | 243 | 59 | 109 |
| 162 | 212 | 63 | 113 | 166 | 216 | 67 | 117 | 170 | 220 |
| 36 | 37 | 54 | 53 | 52 | 58 | 57 | 56 | 62 | 61 |
| 60 | 66 | 65 | 64 | 70 | 69 | 68 | 104 | 103 | 102 |
| 108 | 107 | 106 | 112 | 111 | 110 | 116 | 115 | 114 | 120 |
| 119 | 118 | 157 | 156 | 155 | 161 | 160 | 159 | 165 | 164 |
| 163 | 169 | 168 | 167 | 173 | 172 | 171 | 207 | 206 | 205 |
| 211 | 210 | 209 | 215 | 214 | 213 | 219 | 218 | 217 | 223 |
| 222 | 221 | 73 | 72 | 71 | 76 | 75 | 74 | 79 | 78 |
| 77 | 82 | 81 | 80 | 85 | 84 | 83 | 123 | 122 | 121 |
| 126 | 125 | 124 | 129 | 128 | 127 | 132 | 131 | 130 | 135 |
| 134 | 133 | 176 | 175 | 174 | 179 | 178 | 177 | 182 | 181 |
| 180 | 185 | 184 | 183 | 188 | 187 | 186 | 226 | 225 | 224 |
| 229 | 228 | 227 | 232 | 231 | 230 | 235 | 234 | 233 | 238 |
| 237 | 236 | 96 | 199 | | | | | | |

| 8 | 7 | 6 | 5 | 4 | 3 | 2 | 1 | |
|---|---|---|---|---|---|---|---|---|
| d(7) | … | … | … | … | … | … | d(0) | 1 |
| d(15) | … | … | … | … | … | … | d(8) | 2 |
| . . . . . . . . . . | | | | | | | | … |
| UB | UB | UB | UB | d(243) | … | … | d(240) | 31 |

**Figure Q.1/I.366.2 – AMR 12.2 EDU format**

**Table Q.2a/I.366.2 – Source encoder output parameters in order of occurrence and bit allocation within the speech frame of 204 bits/20 ms, 10.2 kbit/s mode**

| Bits (MSB-LSB) | Description |
|---|---|
| s1-s8 | index of 1st LSF subvector |
| s9-s17 | index of 2nd LSF subvector |
| s18-s26 | index of 3rd LSF subvector |
| Subframe 1 | |
| s27-s34 | adaptive codebook index |
| s35 | sign information for 1st and 5th pulses |
| s36 | sign information for 2nd and 6th pulses |
| s37 | sign information for 5th and 7th pulses |
| s38 | sign information for 4th and 8th pulses |
| s39-s48 | position for 1st, 2nd, and 5th pulses |
| s49-s58 | position for 3rd, 6th, and 7th pulses |
| s59-s65 | position for 4th and 7th pulses |
| s66-s72 | codebook gains |
| Subframe 2 | |
| s73-s77 | adaptive codebook index (relative) |
| s78-s115 | same description as s35-s72 |
| Subframe 3 | |
| s116-s161 | same description as s27-s72 |
| Subframe 4 | |
| s162-s204 | same description as s73-s115 |

**Table Q.2b/I.366.2 – Ordering of the speech encoder bits for the 10.2 kbit/s mode: *table*$_6$(*j*)**

| | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|
| 7 | 6 | 5 | 4 | 3 | 2 | 1 | 0 | 16 | 15 |
| 14 | 13 | 12 | 11 | 10 | 9 | 8 | 26 | 27 | 28 |
| 29 | 30 | 31 | 115 | 116 | 117 | 118 | 119 | 120 | 72 |
| 73 | 161 | 162 | 65 | 68 | 69 | 108 | 111 | 112 | 154 |
| 157 | 158 | 197 | 200 | 201 | 32 | 33 | 121 | 122 | 74 |
| 75 | 163 | 164 | 66 | 109 | 155 | 198 | 19 | 23 | 21 |
| 22 | 18 | 17 | 20 | 24 | 25 | 37 | 36 | 35 | 34 |
| 80 | 79 | 78 | 77 | 126 | 125 | 124 | 123 | 169 | 168 |
| 167 | 166 | 70 | 67 | 71 | 113 | 110 | 114 | 159 | 156 |
| 160 | 202 | 199 | 203 | 76 | 165 | 81 | 82 | 92 | 91 |
| 93 | 83 | 95 | 85 | 84 | 94 | 101 | 102 | 96 | 104 |
| 86 | 103 | 87 | 97 | 127 | 128 | 138 | 137 | 139 | 129 |
| 141 | 131 | 130 | 140 | 147 | 148 | 142 | 150 | 132 | 149 |
| 133 | 143 | 170 | 171 | 181 | 180 | 182 | 172 | 184 | 174 |
| 173 | 183 | 190 | 191 | 185 | 193 | 175 | 192 | 176 | 186 |
| 38 | 39 | 49 | 48 | 50 | 40 | 52 | 42 | 41 | 51 |
| 58 | 59 | 53 | 61 | 43 | 60 | 44 | 54 | 194 | 179 |
| 189 | 196 | 177 | 195 | 178 | 187 | 188 | 151 | 136 | 146 |
| 153 | 134 | 152 | 135 | 144 | 145 | 105 | 90 | 100 | 107 |
| 88 | 106 | 89 | 98 | 99 | 62 | 47 | 57 | 64 | 45 |
| 63 | 46 | 55 | 56 | | | | | | |

| 8 | 7 | 6 | 5 | 4 | 3 | 2 | 1 | |
|---|---|---|---|---|---|---|---|---|
| d(7) | … | … | … | … | … | … | d(0) | 1 |
| d(15) | … | … | … | … | … | … | d(8) | 2 |
| | | | . . . . . . . . . . | | | | | … |
| UB | UB | UB | UB | d(203) | … | … | d(200) | 26 |

**Figure Q.2/I.366.2 – AMR 10.2 EDU format**

**Table Q.3a/I.366.2 – Source encoder output parameters in order of occurrence and bit allocation within the speech frame of 159 bits/20 ms, 7.95 kbit/s mode**

| Bits (MSB-LSB) | Description |
|---|---|
| s1-s9 | index of 1st LSF subvector |
| s10-s18 | index of 2nd LSF subvector |
| s19-s27 | index of 3rd LSF subvector |
| Subframe 1 | |
| s28-s35 | adaptive codebook index |
| s36-s38 | position of 1st pulse |
| s39-s41 | position of 2nd pulse |
| s42-s44 | position of 3rd pulse |
| s45-s48 | position of 4th pulse |
| s49 | sign information for 1st pulse |
| s50 | sign information for 2nd pulse |
| s51 | sign information for 3rd pulse |
| s52 | sign information for 4th pulse |
| s53-s56 | adaptive codebook gain |
| s57-s61 | fixed codebook gain |
| Subframe 2 | |
| s62-s67 | adaptive codebook index (relative) |
| s68-s93 | same description as s36-s61 |
| Subframe 3 | |
| s94-s127 | same description as s28-s61 |
| Subframe 4 | |
| s128-s159 | same description as s62-s93 |

**Table Q.3b/I.366.2 – Ordering of the speech encoder bits for the 7.95 kbit/s mode: *table*$_5$(*j*)**

| | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|
| 8 | 7 | 6 | 5 | 4 | 3 | 2 | 14 | 16 | 9 |
| 10 | 12 | 13 | 15 | 11 | 17 | 20 | 22 | 24 | 23 |
| 19 | 18 | 21 | 56 | 88 | 122 | 154 | 57 | 89 | 123 |
| 155 | 58 | 90 | 124 | 156 | 52 | 84 | 118 | 150 | 53 |
| 85 | 119 | 151 | 27 | 93 | 28 | 94 | 29 | 95 | 30 |
| 96 | 31 | 97 | 61 | 127 | 62 | 128 | 63 | 129 | 59 |
| 91 | 125 | 157 | 32 | 98 | 64 | 130 | 1 | 0 | 25 |
| 26 | 33 | 99 | 34 | 100 | 65 | 131 | 66 | 132 | 54 |
| 86 | 120 | 152 | 60 | 92 | 126 | 158 | 55 | 87 | 121 |
| 153 | 117 | 116 | 115 | 46 | 78 | 112 | 144 | 43 | 75 |
| 109 | 141 | 40 | 72 | 106 | 138 | 36 | 68 | 102 | 134 |
| 114 | 149 | 148 | 147 | 146 | 83 | 82 | 81 | 80 | 51 |
| 50 | 49 | 48 | 47 | 45 | 44 | 42 | 39 | 35 | 79 |
| 77 | 76 | 74 | 71 | 67 | 113 | 111 | 110 | 108 | 105 |
| 101 | 145 | 143 | 142 | 140 | 137 | 133 | 41 | 73 | 107 |
| 139 | 37 | 69 | 103 | 135 | 38 | 70 | 104 | 136 | |

| 8 | 7 | 6 | 5 | 4 | 3 | 2 | 1 | |
|---|---|---|---|---|---|---|---|---|
| d(7) | … | … | … | … | … | … | d(0) | 1 |
| D(15) | … | … | … | … | … | … | d(8) | 2 |
| . . . . . . . . . . | | | | | | | | … |
| UB | D(158) | … | … | … | … | … | d(152) | 20 |

**Figure Q.3/I.366.2 – AMR 7.95 EDU format**

**Table Q.4a/I.366.2 – Source encoder output parameters in order of occurrence and bit allocation within the speech frame of 148 bits/20 ms, 7.40 kbit/s mode**

| Bits (MSB-LSB) | Description |
|---|---|
| s1-s8 | index of 1st LSF subvector |
| s9-s17 | index of 2nd LSF subvector |
| s18-s26 | index of 3rd LSF subvector |
| Subframe 1 | |
| s27-s34 | adaptive codebook index |
| s35-s37 | position of 1st pulse |
| s38-s40 | position of 2nd pulse |
| s41-s43 | position of 3rd pulse |
| s44-s47 | position of 4th pulse |
| s48 | sign information for 1st pulse |
| s49 | sign information for 2nd pulse |
| s50 | sign information for 3rd pulse |
| s51 | sign information for 4th pulse |
| s52-s58 | codebook gains |
| Subframe 2 | |
| s59-s63 | adaptive codebook index (relative) |
| s64-s87 | same description as s35-s58 |
| Subframe 3 | |
| s88-s119 | same description as s27-s58 |
| Subframe 4 | |
| s120-s148 | same description as s59-s87 |

**Table Q.4b/I.366.2 – Ordering of the speech encoder bits for the 7.4 kbit/s mode: $table_4(j)$**

| | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|
| 0 | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 | 9 |
| 10 | 11 | 12 | 13 | 14 | 15 | 16 | 26 | 87 | 27 |
| 88 | 28 | 89 | 29 | 90 | 30 | 91 | 51 | 80 | 112 |
| 141 | 52 | 81 | 113 | 142 | 54 | 83 | 115 | 144 | 55 |
| 84 | 116 | 145 | 58 | 119 | 59 | 120 | 21 | 22 | 23 |
| 17 | 18 | 19 | 31 | 60 | 92 | 121 | 56 | 85 | 117 |
| 146 | 20 | 24 | 25 | 50 | 79 | 111 | 140 | 57 | 86 |
| 118 | 147 | 49 | 78 | 110 | 139 | 48 | 77 | 53 | 82 |
| 114 | 143 | 109 | 138 | 47 | 76 | 108 | 137 | 32 | 33 |
| 61 | 62 | 93 | 94 | 122 | 123 | 41 | 42 | 43 | 44 |
| 45 | 46 | 70 | 71 | 72 | 73 | 74 | 75 | 102 | 103 |
| 104 | 105 | 106 | 107 | 131 | 132 | 133 | 134 | 135 | 136 |
| 34 | 63 | 95 | 124 | 35 | 64 | 96 | 125 | 36 | 65 |
| 97 | 126 | 37 | 66 | 98 | 127 | 38 | 67 | 99 | 128 |
| 39 | 68 | 100 | 129 | 40 | 69 | 101 | 130 | | |

| 8 | 7 | 6 | 5 | 4 | 3 | 2 | 1 | |
|---|---|---|---|---|---|---|---|---|
| d(7) | … | … | … | … | … | … | d(0) | 1 |
| d(15) | … | … | … | … | … | … | d(8) | 2 |
| . . . . . . . . . . | | | | | | | | … |
| UB | UB | UB | UB | d(147) | … | … | d(144) | 19 |

**Figure Q.4/I.366.2 – AMR 7.4 EDU format**

**Table Q.5a/I.366.2 – Source encoder output parameters in order of occurrence and bit allocation within the speech frame of 134 bits/20 ms, 6.70 kbit/s mode**

| Bits (MSB-LSB) | Description |
|---|---|
| s1-s8 | index of 1st LSF subvector |
| s9-s17 | index of 2nd LSF subvector |
| s18-s26 | index of 3rd LSF subvector |
| Subframe 1 | |
| s27-s34 | adaptive codebook index |
| s35-s37 | position of 1st pulse |
| s38-s41 | position of 2nd pulse |
| s42-s45 | position of 3rd pulse |
| s46 | sign information for 1st pulse |
| s47 | sign information for 2nd pulse |
| s48 | sign information for 3rd pulse |
| s49-s55 | codebook gains |
| Subframe 2 | |
| s56-s59 | adaptive codebook index (relative) |
| s60-s80 | same description as s35-s55 |
| Subframe 3 | |
| s81-s109 | same description as s27-s55 |
| Subframe 4 | |
| s110-s134 | same description as s56-s80 |

**Table Q.5b/I.366.2 – Ordering of the speech encoder bits for the 6.7 kbit/s mode: *table*$_3$(*j*)**

| | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|
| 0 | 1 | 4 | 3 | 5 | 6 | 13 | 7 | 2 | 8 |
| 9 | 11 | 15 | 12 | 14 | 10 | 28 | 82 | 29 | 83 |
| 27 | 81 | 26 | 80 | 30 | 84 | 16 | 55 | 109 | 56 |
| 110 | 31 | 85 | 57 | 111 | 48 | 73 | 102 | 127 | 32 |
| 86 | 51 | 76 | 105 | 130 | 52 | 77 | 106 | 131 | 58 |
| 112 | 33 | 87 | 19 | 23 | 53 | 78 | 107 | 132 | 21 |
| 22 | 18 | 17 | 20 | 24 | 25 | 50 | 75 | 104 | 129 |
| 47 | 72 | 101 | 126 | 54 | 79 | 108 | 133 | 46 | 71 |
| 100 | 125 | 128 | 103 | 74 | 49 | 45 | 70 | 99 | 124 |
| 42 | 67 | 96 | 121 | 39 | 64 | 93 | 118 | 38 | 63 |
| 92 | 117 | 35 | 60 | 89 | 114 | 34 | 59 | 88 | 113 |
| 44 | 69 | 98 | 123 | 43 | 68 | 97 | 122 | 41 | 66 |
| 95 | 120 | 40 | 65 | 94 | 119 | 37 | 62 | 91 | 116 |
| 36 | 61 | 90 | 115 | | | | | | |

| 8 | 7 | 6 | 5 | 4 | 3 | 2 | 1 | |
|---|---|---|---|---|---|---|---|---|
| d(7) | … | … | … | … | … | … | d(0) | 1 |
| d(15) | … | … | … | … | … | … | d(8) | 2 |
| . . . . . . . . . . | | | | | | | | … |
| UB | UB | d(133) | … | … | … | … | d(128) | 17 |

**Figure Q.5/I.366.2 – AMR 6.7 EDU format**

**Table Q.6a/I.366.2 – Source encoder output parameters in order of occurrence and bit allocation within the speech frame of 118 bits/20 ms, 5.90 kbit/s mode**

| Bits (MSB-LSB) | Description |
|---|---|
| s1-s8 | index of 1st LSF subvector |
| s9-s17 | index of 2nd LSF subvector |
| s18-s26 | index of 3rd LSF subvector |
| Subframe 1 | |
| s27-s34 | adaptive codebook index |
| s35-s38 | position of 1st pulse |
| s39-s43 | position of 2nd pulse |
| s44 | sign information for 1st pulse |
| s45 | sign information for 2nd pulse |
| s46-s51 | codebook gains |
| Subframe 2 | |
| s52-s55 | adaptive codebook index (relative) |
| s56-s72 | same description as s35-s51 |
| Subframe 3 | |
| s73-s97 | same description as s27-s51 |
| Subframe 4 | |
| s98-s118 | same description as s52-s72 |

**Table Q.6b/I.366.2 – Ordering of the speech encoder bits for the 5.9 kbit/s mode: *table$_2$(j)***

| | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|
| 0 | 1 | 4 | 5 | 3 | 6 | 7 | 2 | 13 | 15 |
| 8 | 9 | 11 | 12 | 14 | 10 | 16 | 28 | 74 | 29 |
| 75 | 27 | 73 | 26 | 72 | 30 | 76 | 51 | 97 | 50 |
| 71 | 96 | 117 | 31 | 77 | 52 | 98 | 49 | 70 | 95 |
| 116 | 53 | 99 | 32 | 78 | 33 | 79 | 48 | 69 | 94 |
| 115 | 47 | 68 | 93 | 114 | 46 | 67 | 92 | 113 | 19 |
| 21 | 23 | 22 | 18 | 17 | 20 | 24 | 111 | 43 | 89 |
| 110 | 64 | 65 | 44 | 90 | 25 | 45 | 66 | 91 | 112 |
| 54 | 100 | 40 | 61 | 86 | 107 | 39 | 60 | 85 | 106 |
| 36 | 57 | 82 | 103 | 35 | 56 | 81 | 102 | 34 | 55 |
| 80 | 101 | 42 | 63 | 88 | 109 | 41 | 62 | 87 | 108 |
| 38 | 59 | 84 | 105 | 37 | 58 | 83 | 104 | | |

| 8 | 7 | 6 | 5 | 4 | 3 | 2 | 1 | |
|---|---|---|---|---|---|---|---|---|
| d(7) | … | … | … | … | … | … | d(0) | 1 |
| d(15) | … | … | … | … | … | … | d(8) | 2 |
| . . . . . . . . . . | | | | | | | | … |
| UB | UB | d(117) | … | … | … | … | d(112) | 15 |

**Figure Q.6/I.366.2 – AMR 5.90 EDU format**

**Table Q.7a/I.366.2 – Source encoder output parameters in order of occurrence and bit allocation within the speech frame of 103 bits/20 ms, 5.15 kbit/s mode**

| Bits (MSB-LSB) | Description |
|---|---|
| s1-s8 | index of 1st LSF subvector |
| s9-s16 | index of 2nd LSF subvector |
| s17-s23 | index of 3rd LSF subvector |
| Subframe 1 | |
| s24-s31 | adaptive codebook index |
| s32 | position subset |
| s33-s35 | position of 1st pulse |
| s36-s38 | position of 2nd pulse |
| s39 | sign information for 1st pulse |
| s40 | sign information for 2nd pulse |
| s41-s46 | codebook gains |
| Subframe 2 | |
| s47-s50 | adaptive codebook index (relative) |
| s51-s65 | same description as s32-s46 |
| Subframe 3 | |
| s66-s84 | same description as s47-s65 |
| Subframe 4 | |
| s85-s103 | same description as s47-s65 |

**Table Q.7b/I.366.2 – Ordering of the speech encoder bits for the 5.15 kbit/s mode: $table_1(j)$**

| | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|
| 7 | 6 | 5 | 4 | 3 | 2 | 1 | 0 | 15 | 14 |
| 13 | 12 | 11 | 10 | 9 | 8 | 23 | 24 | 25 | 26 |
| 27 | 46 | 65 | 84 | 45 | 44 | 43 | 64 | 63 | 62 |
| 83 | 82 | 81 | 102 | 101 | 100 | 42 | 61 | 80 | 99 |
| 28 | 47 | 66 | 85 | 18 | 41 | 60 | 79 | 98 | 29 |
| 48 | 67 | 17 | 20 | 22 | 40 | 59 | 78 | 97 | 21 |
| 30 | 49 | 68 | 86 | 19 | 16 | 87 | 39 | 38 | 58 |
| 57 | 77 | 35 | 54 | 73 | 92 | 76 | 96 | 95 | 36 |
| 55 | 74 | 93 | 32 | 51 | 33 | 52 | 70 | 71 | 89 |
| 90 | 31 | 50 | 69 | 88 | 37 | 56 | 75 | 94 | 34 |
| 53 | 72 | 91 | | | | | | | |

| 8 | 7 | 6 | 5 | 4 | 3 | 2 | 1 | |
|---|---|---|---|---|---|---|---|---|
| d(7) | … | … | … | … | … | … | d(0) | 1 |
| d(15) | … | … | … | … | … | … | d(8) | 2 |
| . . . . . . . . . . | | | | | | | | … |
| UB | d(102) | … | … | … | … | … | d(96) | 13 |

**Figure Q.7/I.366.2 – AMR 5.15 EDU format**

**Table Q.8a/I.366.2 – Source encoder output parameters in order of occurrence and bit allocation within the speech frame of 95 bits/20 ms, 4.75 kbit/s mode**

| Bits (MSB-LSB) | Description |
|---|---|
| s1-s8 | index of 1st LSF subvector |
| s9-s16 | index of 2nd LSF subvector |
| s17-s23 | index of 3rd LSF subvector |
| Subframe 1 | |
| s24-s31 | adaptive codebook index |
| s32 | position subset |
| s33-s35 | position of 1st pulse |
| s36-s38 | position of 2nd pulse |
| s39 | sign information for 1st pulse |
| s40 | sign information for 2nd pulse |
| s41-s48 | codebook gains |
| Subframe 2 | |
| s49-s52 | adaptive codebook index (relative) |
| s53-s61 | same description as s32-s40 |
| Subframe 3 | |
| s62-s65 | same description as s49-s52 |
| s66-s82 | same description as s32-s48 |
| Subframe 4 | |
| s83-s95 | same description as s49-s61 |

**Table Q.8b/I.366.2 – Ordering of the speech encoder bits for the 4.75 kbit/s mode: $table_0(j)$**

| | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|
| 0 | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 | 9 |
| 10 | 11 | 12 | 13 | 14 | 15 | 23 | 24 | 25 | 26 |
| 27 | 28 | 48 | 49 | 61 | 62 | 82 | 83 | 47 | 46 |
| 45 | 44 | 81 | 80 | 79 | 78 | 17 | 18 | 20 | 22 |
| 77 | 76 | 75 | 74 | 29 | 30 | 43 | 42 | 41 | 40 |
| 38 | 39 | 16 | 19 | 21 | 50 | 51 | 59 | 60 | 63 |
| 64 | 72 | 73 | 84 | 85 | 93 | 94 | 32 | 33 | 35 |
| 36 | 53 | 54 | 56 | 57 | 66 | 67 | 69 | 70 | 87 |
| 88 | 90 | 91 | 34 | 55 | 68 | 89 | 37 | 58 | 71 |
| 92 | 31 | 52 | 65 | 86 | | | | | |

| 8 | 7 | 6 | 5 | 4 | 3 | 2 | 1 | |
|---|---|---|---|---|---|---|---|---|
| d(7) | … | … | … | … | … | … | d(0) | 1 |
| d(15) | … | … | … | … | … | … | d(8) | 2 |
| . . . . . . . . . . | | | | | | | | … |
| UB | d(94) | … | … | … | … | … | d(88) | 12 |

**Figure Q.8/I.366.2 – AMR 4.75 EDU format**

## Q.3 Silence Descriptor

The voice activity detection algorithm for AMR is specified in [3GPP TS26.094]. This is a transmitter option. If enabled, silence descriptor (SID) frames are sent during silence periods. The first SID for a silence period is basically empty and only indicates that start of a silence period. The coding of a SID_First frame is indicated in Figure Q.9. The actual content in the two octets of a SID_First frame is irrelevant and should be ignored by the receiver.

| 8 | 7 | 6 | 5 | 4 | 3 | 2 | 1 | |
|---|---|---|---|---|---|---|---|---|
| Reserved (set to zero) | | | | | | | | 1 |
| Reserved (set to zero) | | | | | | | | 2 |

**Figure Q.9/I.366.2 – AMR SID_First EDU format**

Subsequent SID frames (SID_Update frames) contain comfort noise bits. The AMR specific comfort noise generator is specified in [3GPP 20.092]. The bit allocation and sequence of the bits from comfort noise encoding is shown in Figure Q.10 based on Table Q.9.

**Table Q.9/I.366.2 – Source encoder output parameters in order of occurrence and bit allocation for SID_Update (AMR comfort noise encoding)**

| Bits (MSB-LSB) | Description |
|---|---|
| s1-s3 | index of reference vector |
| s4-s11 | index of 1st LSF subvector |
| s12-s20 | index of 2nd LSF subvector |
| s21-s29 | index of 3rd LSF subvector |
| s30-s35 | index of logarithmic frame energy |

| 8 | 7 | 6 | 5 | 4 | 3 | 2 | 1 | |
|---|---|---|---|---|---|---|---|---|
| s8 | s7 | s6 | s5 | s4 | s3 | s2 | s1 | 1 |
| s16 | s15 | s14 | s13 | s12 | s11 | s10 | s9 | 2 |
| s24 | s23 | s22 | s21 | s20 | s19 | s18 | s17 | 3 |
| s32 | s31 | s30 | s29 | s28 | s27 | s26 | s25 | 4 |
| UB | UB | UB | UB | UB | s35 | s34 | s33 | 5 |

**Figure Q.10/I.366.2 – AMR SID_Update EDU format**

SID_Update frames are generated every 160 ms during silence periods. Between SID_Update frames the AMR encoder generates No_Data frames every 20 ms. No_Data frames are "null data units" and are not transmitted.

ANNEX R

## Packet formats and procedures for rate control

### R.1 General

The rate control packet format can be used to convey requests from an SSCS user to its peer to operate using only certain set of entries of the profile used for the connection.

### R.2 Packet format

Rate control packets are format type 3 and benefit from CRC-10 error detection. They make use of the common facilities for type 3 packets defined in clause 11, including triple redundancy.

The format of rate control packets is shown in Figure R.1.

<table>
<tr><td>8</td><td>7</td><td>6</td><td>5</td><td>4</td><td>3</td><td>2</td><td>1</td><td></td></tr>
<tr><td colspan="2">Redundancy</td><td colspan="6">Time Stamp</td><td>1</td></tr>
<tr><td colspan="8">Time Stamp</td><td>2</td></tr>
<tr><td colspan="8">Profile Entry Index</td><td>3</td></tr>
<tr><td colspan="6">Message Type = 000100</td><td colspan="2">CRC-10</td><td>4</td></tr>
<tr><td colspan="8">CRC-10</td><td>5</td></tr>
</table>

**Figure R.1/I.366.2 – Rate control packet format**

The "Profile Entry Index" is an 8-bit field which is a binary representation of the to "Profile Entry Index" in the profile table. Bit 8 is the MSB and bit 1 is the LSB.

### R.3 Transmitter procedures

A transmitter receiving rate control from the remote SSCS user is expected to use the requested Profile_Entry Index to decide the actual profile entry in which it will continue operating, which must be either the requested one or one that operates at equal or lower bit rate.

In order to be able to adapt rate as fast as possible, the SSCS transmitter should convey the **Rate_Control.indication** to the user upon reception of the first rate control, without waiting for such time as it should have received all three copies of a rate control command.

### R.4 Receiver procedures

When a receiver wishes to convey to the transmitter a rate control request, the rate control packet shall be sent with triple redundancy at intervals of 5 ms.

If a new event is conveyed before the triple redundancy of a previous event has completed, the receiver shall stop sending packets for the previous event, in order to avoid the interleaving of two different time stamps.

ANNEX S

**Packet formats and Procedures for synchronization of change in SSCS operation**

## S.1 General

The Change SSCS operation packet format can be used to convey requests from an SSCS user to its peer to reconfigure the SSCS attributes (e.g. profile number, DTMF support, etc) to previously agreed values by the two peers. External mechanisms, e.g. signalling, are used between the two peers to agree upon the new attributes of the SSCS and to exchange correlation identifiers (one per SSCS user) that identify the SSCS change operation.

## S.2 Packet format

Change SSCS operation packets are type 3 format and benefit from CRC-10 error detection. They make use of the common facilities for type 3 packets defined in clause 11, including triple redundancy.

The format of Change SSCS operation packets is shown in Figure S.1.

<table>
<tr><th>8</th><th>7</th><th>6</th><th>5</th><th>4</th><th>3</th><th>2</th><th>1</th><th></th></tr>
<tr><td colspan="2">Redundancy</td><td colspan="6">Time Stamp</td><td>1</td></tr>
<tr><td colspan="8">Time Stamp</td><td>2</td></tr>
<tr><td colspan="8">Correlation ID</td><td>3</td></tr>
<tr><td colspan="6">Message Type = 000101</td><td colspan="2">CRC-10</td><td>4</td></tr>
<tr><td colspan="8">CRC-10</td><td>5</td></tr>
</table>

**Figure S.1/I.366.2 – Change SSCS operation packet format**

## S.3 Transmitter procedures

When a transmitter wishes to convey to the receiver a Change SSCS operation request, the Change SSCS operation packet shall be sent with triple redundancy at intervals of 5 ms.

If a new event is conveyed before the triple redundancy of a previous event has completed, the transmitter shall stop sending packets for the previous event, in order to avoid the interleaving of two different time stamps.

## S.4 Receiver procedures

A User receiving Change operation shall immediately switch to the new SSCS configuration in which it will continue operating.

In order to be able to synchronize as fast as possible, the SSCS receiver should convey the **SSCS_Change.indication** to the user upon reception of the first Change SSCS operation packet, without waiting for such time as it should have received all three copies of a Change SSCS operation command.

APPENDIX I

**AAL Type 1 interworking between N-ISDN and B-ISDN**

**I.1** AAL type 1 is the recommended adaptation layer for universal support of 64 kbit/s PCM voice within or between B-ISDN networks (I.363.1) and for interworking between N-ISDN and B-ISDN (I.580).

**I.2** AAL type 2 and this Recommendation are defined for network specific use and may be used on an internetwork interface only when agreed between service providers.

### I.3 Informative references

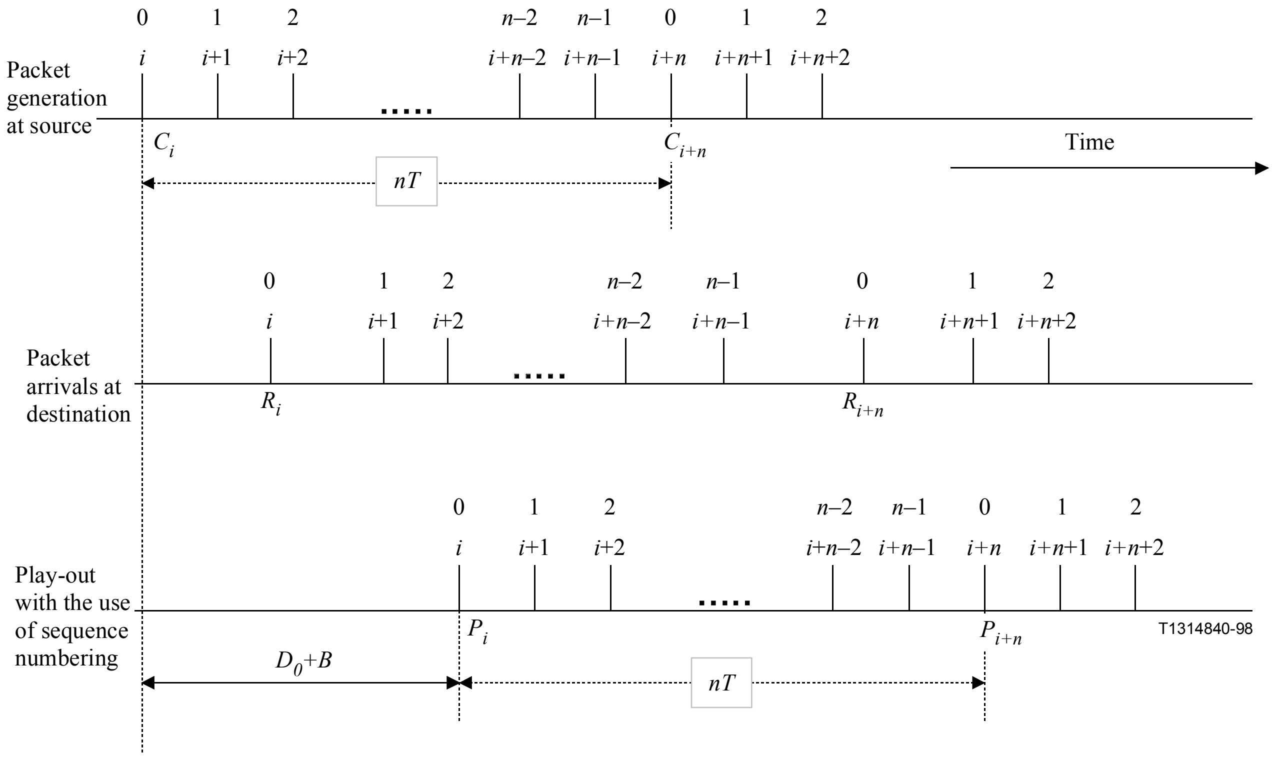
– ITU-T I.363.1 (1996), *B-ISDN ATM Adaptation Layer (AAL) specification: Type 1 AAL*.

– ITU-T I.580 (1995), *General arrangements for interworking between B-ISDN and 64 kbit/s based ISDN*.

## APPENDIX II

## Simple derivation of the modulus for sequence numbering

### II.1 Packet flow timing notation and relationships




*B* Build-out delay

$D_0$ Total network delay of the first packet

Let the first packet be defined as the 0th packet. Packet *i* refers to the Ith packet.

$T$ sequence number interval;

$C_i$ source generation time of packet *i*;

$R_i$ destination receive time of packet *i*;

$P_i$ scheduled playout time of packet *i*;

$D_i$ total network delay of packet *i*;

With the above definitions, these equations follow:

$$C_i = C_0 + iT$$

$$R_i = C_i + D_i = C_0 + iT + D_i$$

$$P_0 = R_0 + B = C_0 + D_0 + B$$

$$P_i = P_0 + iT = C_0 + D_0 + B + iT$$

## II.2 Simple derivation of the modulus

In this analysis, $n$ denotes the tentative modulus for sequence numbering. This is the point at which sequence numbers wrap around to zero again. That is, it is impossible to distinguish between $i$ and $i + n$ using the limited number of bits that are allocated to the sequence number field.

The derivation proceeds from these two objectives:

1) With high probability, packet $i$ should arrive before its playout time: $R_i \leq P_i$;
2) But it should *not* arrive so early as to confound the previous playout time modulo $n$: $P_{i-n} \leq R_i$.

The rationale for 1) is clear: If a packet arrives late, its place will need to be filled by some error concealment strategy. The most that sequence numbers can do is detect and discard a late packet, so that it does not cause a permanent ripple. But the error is still deplorable. It can be made rare by setting the build-out delay large enough.

Concern about 2) applies mostly at the beginning of a new talk spurt (although it could also apply immediately after the rare loss of a full cycle of $n$ packets). A new talk spurt should be started on time, not a full cycle too early. Choosing a value of $n$ large enough can make such errors rare.

The two inequalities can be expanded in terms of the relationships above and simplified:

$$P_{i-n} \leq R_i \leq P_i$$

$$C_0 + D_0 + B + (i-n)T \leq C_0 + iT + D_i \leq C_0 + D_0 + B + iT$$

$$D_0 + B - nT \leq D_i \leq D_0 + B$$

From the right half, derive one lower bound: $D_i - D_0 \leq B$

From the left half, derive another: $(B + D_0 - D_i)/T \leq n$

The difference $D_i - D_0$ (or $D_0 - D_i$ equivalently) is a sample of the *Packet Delay Variation*. To ensure that the two original objectives are met with whatever probability is considered adequate, e.g. 99% or 99.9% or 99.99% etc., one can substitute for this difference a corresponding quantile of the distribution of the PDV.

Therefore, to ensure that packets get played out at the right time, one should take:

- $B \geq \text{PDV}$
- $n \geq (B + \text{PDV})/T$

These inequalities determine the receiver build-out delay $B$ and the sequence number modulus $n$ based on the sequence number interval $T$ and the packet delay variation experienced at maximum anticipated load.

On the other hand, if the sequence number modulus *n* is already fixed, these inequalities may be used to determine how much packet delay variation can be tolerated, as a prelude to setting corresponding limits on call admissions:

- PDV ≤ $n\ T/2$

## APPENDIX III

## Example facsimile demodulation scenarios

This appendix covers three scenarios:

- III.1 describes a typical facsimile call completed with normal demodulation.
- III.2 describes a fallback to voiceband data due to no support for non-standard T.30 facilities.
- III.3 describes a fallback to voiceband data due to a failure to demodulate.

**List of abbreviations**

| | |
|---|---|
| CED | Called Terminal Identification |
| CFR | Confirmation To Receive |
| CNG | Calling Tone |
| DCN | Disconnect |
| DCS | Digital Command Signal |
| DIS | Digital Identification Signal |
| EOP | End Of Procedure |
| EPT | Echo Protection Tone |
| FTT | Failure to Train |
| MCF | Message Confirmation |
| NSF | Non-Standard Facilities |
| NSS | Non-Standard Set-up |
| TCF | Training Check |
| vbd | Voiceband data |

### III.1 Typical facsimile demodulation scenario – Normal completion

Figure III.1 shows a typical V.29 facsimile call, demodulated and transmitted using the packet formats defined in Annex M.

The following is a short description of the facsimile control and data flows:

a) *Called side*

At the end of the first received V.21 signal (DIS) from the facsimile terminal, the User issues a **State_Control(facsimile_demodulation)** request to the far-end User. The User state is changed to Facsimile Demodulation at both ends only if the calling side positively acknowledges with a **State_Control(facsimile_demodulation, accept)** response, and only then is a demodulator/remodulator pair allocated to the AAL type 2 connection.

b) *Calling side*

The User responds to the received **State_Control(facsimile_demodulation)** indication by issuing a **State_Control(facsimile_demodulation, accept)** response. Receiving the next V.21 signal (usually DCS) from the facsimile terminal, the User detects HDLC flags and sends a T30_Preamble message to the far-end User. Later it receives demodulated data from the demodulator and sends T30_Data messages transparently to the far-end User.

c) *Calling side*

Upon detecting an Echo Protection Tone (EPT) from the facsimile terminal, the near-end User sends an EPT message (including EPT Frequency) to the far-end User. The far-end User reproduces a corresponding tone towards the called side facsimile terminal.

d) *Calling side*

Upon detecting a training signal from the facsimile terminal, the near-end User sends a Training message to the far-end User. The far-end User turns off the EPT tone 20 ms before starting to reproduce the Training signal.

e) *Calling side*

As TCF image data bits are output by the demodulator, the near-end User packs them in 20 ms nominal intervals and sends them to the far end. The far-end User transfers the received data to the remodulator which transmits the remodulated data to the facsimile terminal.

f) *Calling side*

At the end of the high-speed signal, the near-end User sends a Fax_Idle message to the far-end User.

g) *Called side*

Following the first high-speed signal (usually TCF), the called side facsimile terminal sends an acknowledgment V.21 signal (CFR). The Users handle it in the same way as other V.21 signals – see item b).

h) *Calling side*

The EPT, Training and image data blocks transmitted by the facsimile terminal after receipt of CFR, are handled by the Users in the same way as described above for the EPT, Training, and TCF signals – see items c) through f).

i) *Calling side*

The next V.21 signal (EOP) is handled by the Users in the same way as other V.21 signals.

j) *Called side*

The facsimile terminal acknowledges the reception of the image using a V.21 control signal (MCF). The Users handle it the in same way as other V.21 signals.

k) *Calling side*

Following the image transfer, the calling side facsimile terminal sends a DCN signal to the called side terminal. The near-end User recognizes the DCN code from the facsimile terminal and after its transmission issues a **State_Control(voice)** request to the far-end User. The far-end User issues a **State_Control(voice, accept)** response. At this point, the Users on both sides are back in the voice state.

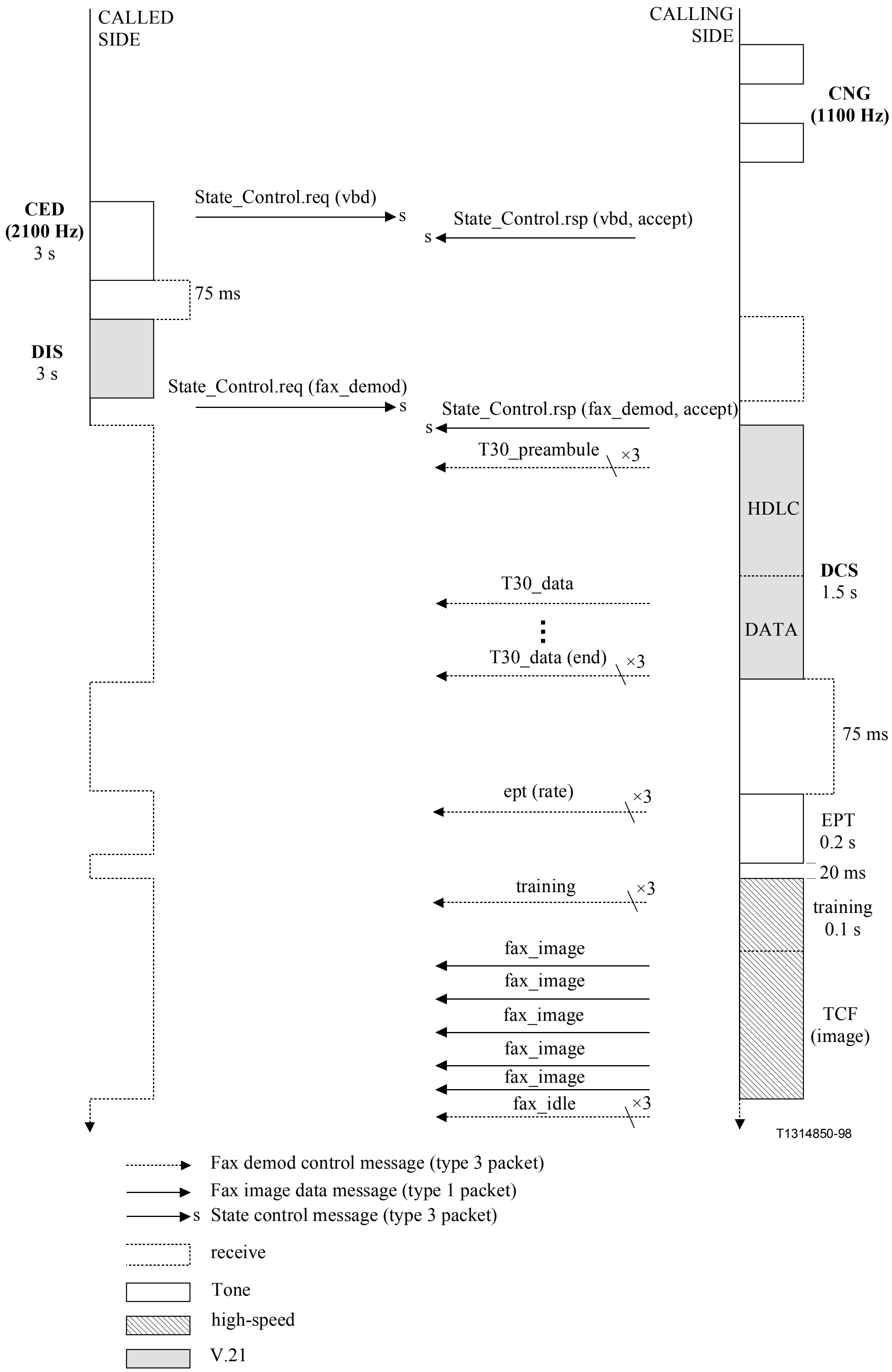



**Figure III.1/I.366.2 – Typical facsimile demodulation scenario – Normal completion *(part 1 of 2)***

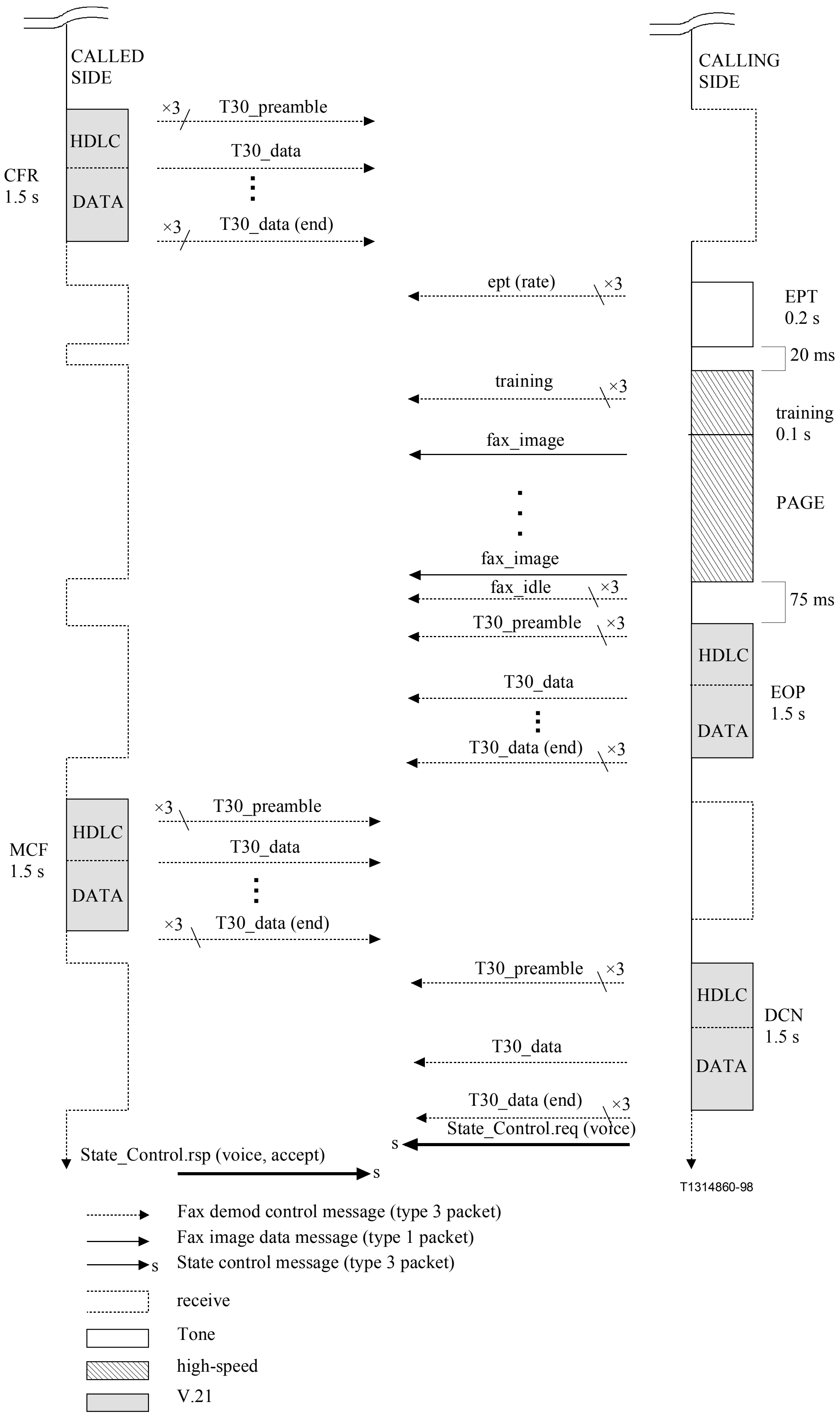



**Figure III.1/I.366.2 – Typical facsimile demodulation scenario – Normal completion *(part 2 of 2)***

### III.2 Fallback to voiceband data – NSS not supported

Figure III.2 shows a fallback to voiceband data due to no support for non-standard T.30 facilities.

The following is a short description of the facsimile control and data flows:

a) *Called side*

At the end of the first received V.21 signal (DIS) from the facsimile terminal, the User issues a **State_Control(facsimile_demodulation)** request to the far-end User. The User state is changed to Facsimile Demodulation at both ends only if the calling side positively acknowledges with a **State_Control(facsimile_demodulation, accept)** response, and only then is a demodulator/ remodulator pair allocated to the AAL type 2 connection.

b) *Calling side*

The User responds to the received **State_Control(facsimile_demodulation)** indication by issuing a **State_Control(facsimile_demodulation, accept)** response. Receiving the next V.21 signal (usually DCS) from the facsimile terminal, the User detects HDLC flags and sends a T30_Preamble message to the far-end User. Later it receives demodulated data from the demodulator and sends T30_Data messages transparently to the far-end User.

c) *Calling side*

The User analyses the T30_Data and finds that it is an NSS message invoking non-standard T.30 facilities which cannot be demodulated. The User issues a **State_Control(voiceband_data)** request to the far-end User.

d) *Called side*

The User responds to the received **State_Control(voiceband_data)** indication by issuing a **State_Control(voiceband_data, accept)** response.

e) *Both sides*

From this point until the end of the facsimile transfer, both Users remain in the voiceband data state. The receivers on both sides treat type 1 packets according to the audio encoding profile in effect, paying attention to the sequence number (UUI field) and length of each packet.

f) *Calling side*

Following the image transfer, the calling side facsimile terminal sends a DCN signal to the called side terminal. The near-end User recognizes the DCN code from the facsimile terminal and after its transmission issues a **State_Control(voice)** request to the far-end User. The far-end User issues a **State_Control(voice, accept)** response. At this point, the Users on both sides are back in the voice state.

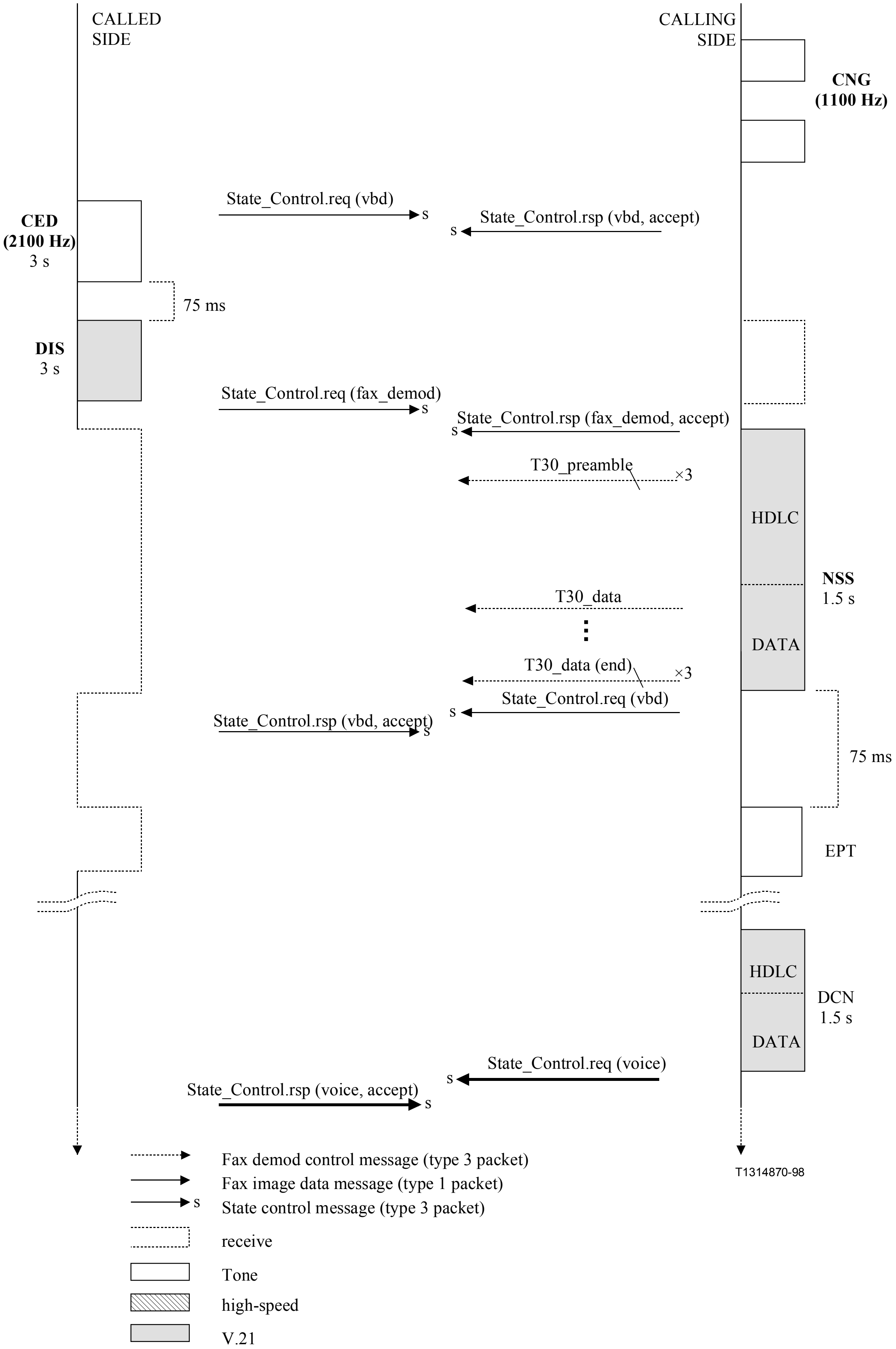



**Figure III.2/I.366.2 – Fallback to voiceband data – NSS not supported**

## III.3 Fallback to voiceband data – demodulation failure

Figure III.3 shows a fallback to voiceband data due to a failure to demodulate.

The following is a short description of the facsimile control and data flows:

a) *Called side*

At the end of the first received V.21 signal (DIS) from the facsimile terminal, the User issues a **State_Control(facsimile_demodulation)** request to the far-end User. The User state is changed to Facsimile Demodulation at both ends only if the calling side positively acknowledges with a **State_Control(facsimile_demodulation, accept)** response, and only then is a demodulator/ remodulator pair allocated to the AAL type 2 connection.

b) *Calling side*

The User responds to the received **State_Control(facsimile_demodulation)** indication by issuing a **State_Control(facsimile_demodulation, accept)** response. Receiving the next V.21 signal (usually DCS) from the facsimile terminal, the User detects HDLC flags and sends a T30_Preamble message to the far-end User. Later it receives demodulated data from the demodulator and sends T30_Data messages transparently to the far-end User.

c) *Calling side*

Upon detecting an Echo Protection Tone (EPT) from the facsimile terminal, the near-end User sends an EPT message (including EPT Frequency) to the far-end User. The far-end User reproduces a corresponding EPT tone towards the called side facsimile terminal.

d) *Calling side*

The User cannot classify the training signal received from the facsimile terminal. The near-end User issues a **State_Control(voiceband_data)** request to the far-end User.

e) *Calling side*

The User responds to the received **State_Control(voiceband_data)** indication by issuing a **State_Control(voiceband_data, accept)** response.

f) *Both sides*

From this point until the end of the facsimile transfer, both Users remain in the voiceband data state. The receivers on both sides treat type 1 packets according to the audio encoding profile in effect, paying attention to the sequence number (UUI field) and length of each packet.

g) *Called side*

Because falling back to voiceband data during an active transmission can be disruptive, the called User may receive an invalid training check sequence (TCF).To this it responds negatively by sending FTT.

h) *Calling side*

Upon receiving FTT, the calling User repeats the DCS and the training check sequence.

i) *Called side*

The called User receives a valid training check sequence and responds CFR.

j) *Both sides*

While both Users remain in the voiceband data state, the facsimile terminals on either side progress through a normal sequence of further exchanges as described in scenario III.1.

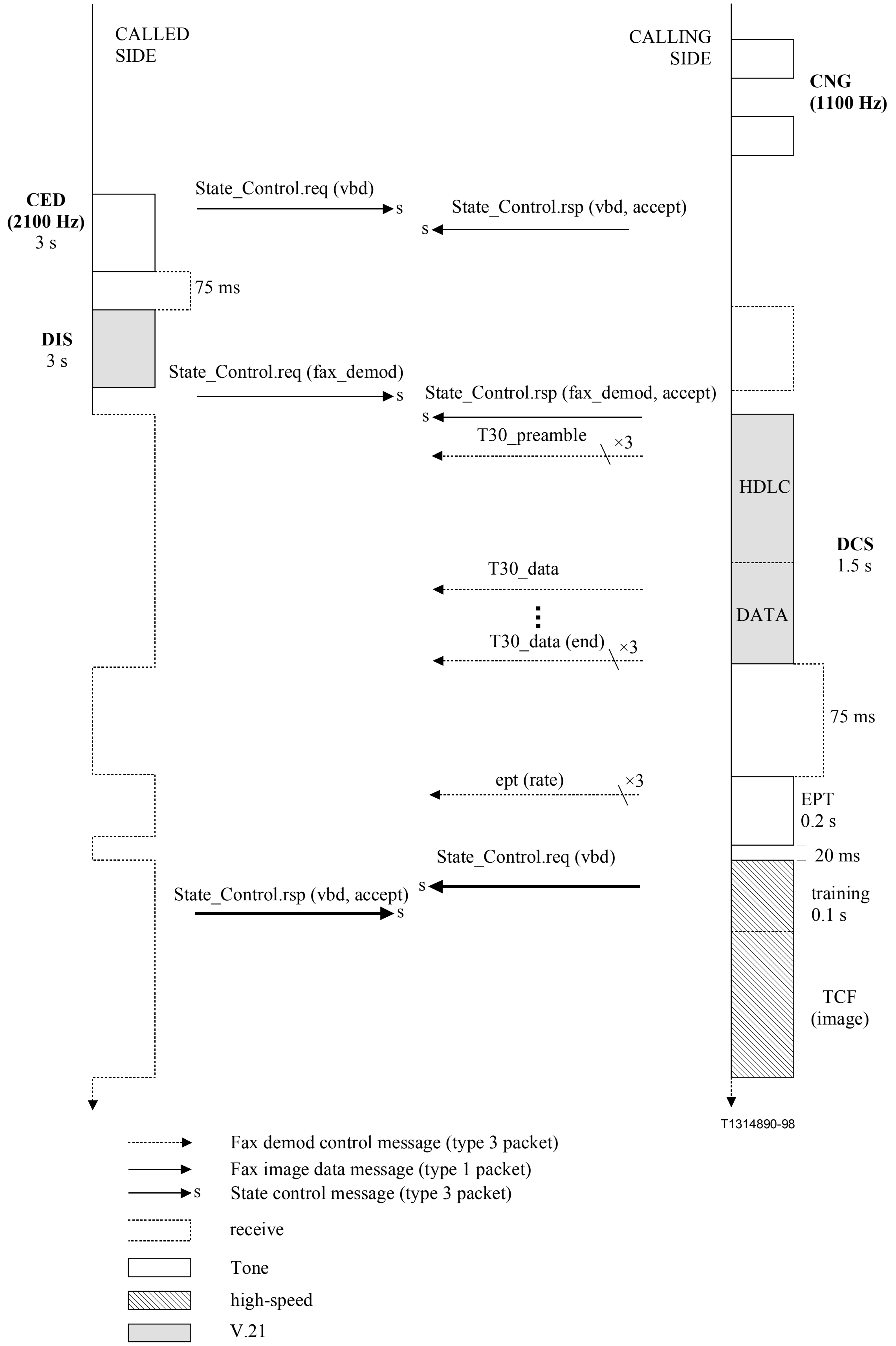



**Figure III.3/I.366.2 – Fallback to voiceband data – Demodulation failure *(part 1 of 2)***

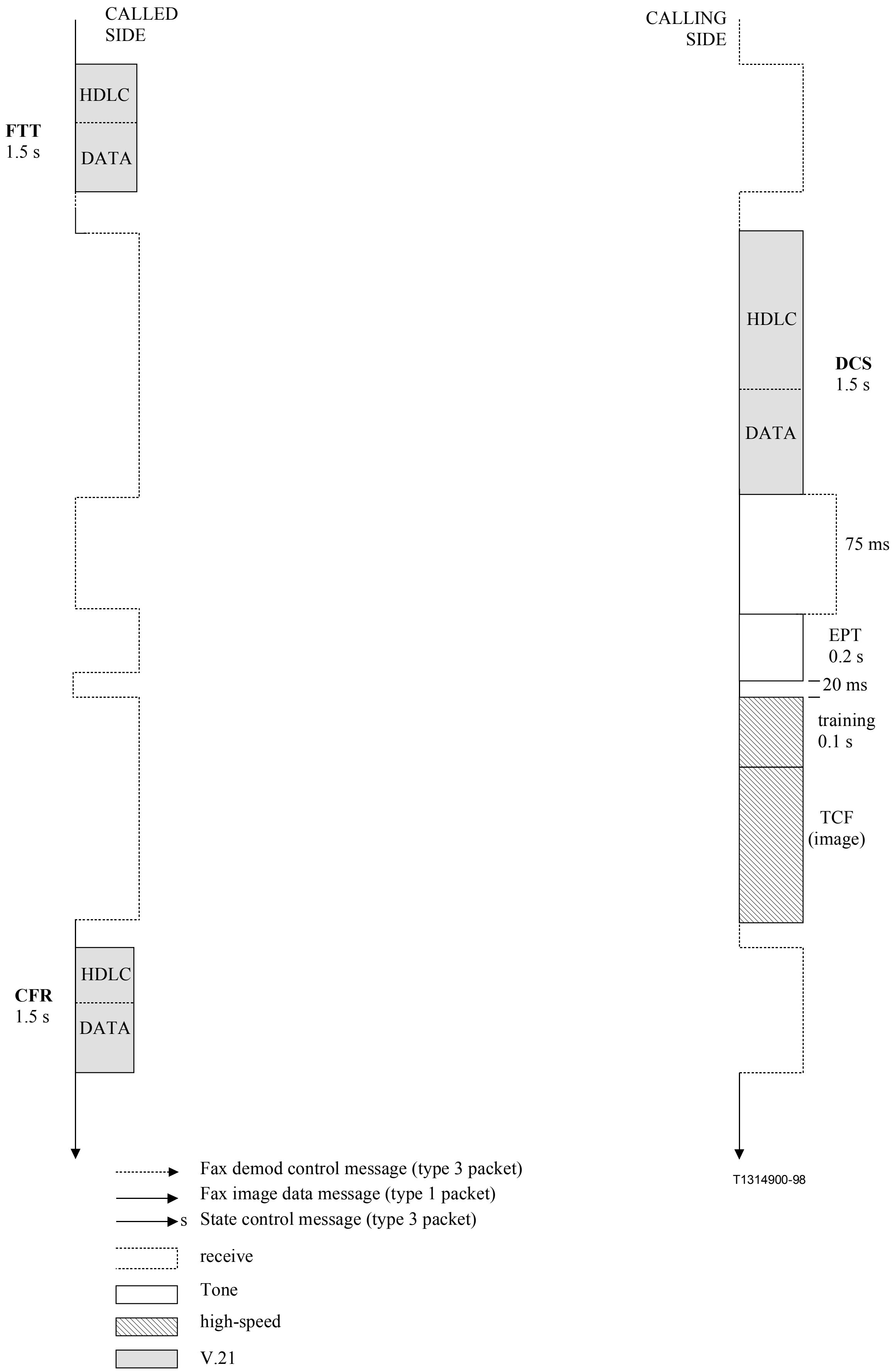



**Figure III.3/I.366.2 – Fallback to voiceband data – Demodulation failure** ***(part 2 of 2)***

## APPENDIX IV

## Example facsimile demodulation V.17 training

Clause M.2.3 describes how V.17 short or long training sequences affect facsimile demodulation. The following examples illustrate the different behaviour of waveform analysis and protocol analysis in regard to training and their interoperation.

- **Waveform Analysis – Demodulator Side**

Upon detecting the beginning of a V.17 training sequence, WA sends a V.17 long Training message with Modulation Rate set to "unknown rate". Signal analysis determines the type and the rate of the training sequence later, and WA then sends one additional short or long Training message with a specific Modulation Rate.

- **Protocol Analysis – Demodulator Side**

Upon detecting the beginning of a V.17 training sequence, PA sends a V.17 short or long Training message with a specific Modulation Rate. PA determines the type and rate from previously analysed T.30 data. PA has no need to use the value "unknown rate".

- **Waveform Analysis – Remodulator Side**

WA obtains the Modulation Type and Modulation Rate from a received Training message.

Upon receipt of a V.17 long Training message with "unknown rate", WA starts generating the V.17 long training sequence.

If a V.17 long Training message with a specific Modulation Rate is received while generating V.17 long training at unknown rate, WA continues the long training sequence, with bridge signal, ending with scrambled ones at the rate indicated in the message.

If a V.17 short Training message with a specific Modulation Rate is received while generating V.17 long training at unknown rate, WA changes the training to a short sequence, without bridge signal, ending with scrambled ones at the rate indicated in the message.

Upon receipt of a V.17 Training message with specific rate and type information, WA generates the V.17 short or long training sequence as specified. This occurs when a WA remodulator is opposite a PA demodulator.

- **Protocol Analysis – Remodulator Side**

PA determines the type and rate of the training sequence from previously analysed T.30 data.

Receipt of a Training message simply stimulates PA to start the training sequence.

While generating the training sequence, PA ignores the receipt of additional Training messages. This occurs when a PA remodulator is opposite a WA demodulator.

APPENDIX V[1]

## Protocol implementation conformance statement (PICS) proforma

(This appendix does not form an integral part of this Recommendation. This appendix is only normative in the sense that if a Protocol Implementation Conformance Statement is made, this proforma shall be used.)

### V.1 Introduction

Prior to the conformance testing and the interoperability testing of Implementations Under Test (ITUs), it is necessary to have the PICS (Protocol Implementation Conformance Statement) document for an implementation.

This particular PICS deals with the implementation of the AAL Type 2 Service Specific Convergence Sublayer for Narrow-Band Services.

#### V.1.1 Scope

This appendix provides the PICS proforma for the AAL Type 2 Service Specific Convergence Sublayer for Narrow-Band Services, in compliance with the relevant requirements, and in accordance with the relevant guidelines, given in ITU-T X.296 [2].

#### V.1.2 Normative references


[1] ITU-T X.290 (1995), *OSI conformance testing methodology and framework for protocol Recommendations for ITU-T applications – General concepts*.

[2] ITU-T X.296 (1995), *OSI conformance testing methodology and framework for protocol Recommendations for ITU-T applications – Implementation conformance statements*.


#### V.1.3 Abbreviations

IUT Implementation Under Test

M Mandatory

N/A Not applicable

NOT Item not supported; absence of item

O Optional

PICS Protocol Implementation Conformance

SUT System Under Test

X Prohibited (excluded)

O.<n> Optional, but, if chosen, support is required for either at least one or only one of the options in the group labelled by the same numeral <n>

#### V.1.4 Conformance statement

The supplier of a protocol implementation which is claimed to conform to the B-ISDN ATM Adaptation Layer Type 2 Specification is required to complete a copy of the PICS proforma provided in V.2 and is required to provide the information necessary to identify both the supplier and the implementation.

---

[1] Users of this Recommendation may freely reproduce the PICS proforma in this appendix so that it can be used for its intended purpose and may further publish the completed PICS.

## V.2 PICS proforma

### V.2.1 Identification of the PICS proforma corrigenda

| | |
|---|---|
| Identification of corrigenda applied to this PICS proforma | Rec. I.366.2 (2000)<br>Cor.:<br>Cor.: |
| | |

### V.2.2 Instructions for completing the PICS proforma

The PICS proforma is a fixed-format questionnaire. Answers to the questionnaire should be provided in the rightmost columns, either by simply indicating a restricted choice (such as Yes or No), or by entering a value or a set of range of values.

A supplier may also provide additional information, categorized as exceptional or supplementary information. An exception item should contain the appropriate rational.

The supplementary information is not mandatory and the PICS is complete without such information. The presence of optional supplementary or exception information should not affect test execution, and will in no way affect interoperability verification.

NOTE – Where an implementation is capable of being configured in more than one way, a single PICS may be able to describe all such configurations. However, the supplier has the choice of providing more than one PICS, each covering some subset of the implementation's configuration capabilities, in case this makes for easier or clearer presentation of the information.

### V.2.3 Identification of the implementation

**Implementation under test (IUT)**

*Identification*

IUT Name: ______________________________________________________________

IUT Version: ____________________________________________________________

________________________________________________________________________

*System Under Test*

SUT Name: _____________________________________________________________

________________________________________________________________________

Hardware Configuration: _________________________________________________

________________________________________________________________________

________________________________________________________________________

Operating System:

*Product Supplier*

Name: ____________________________________________________________________

Address: __________________________________________________________________

___________________________________________________________________________

Telephone Number: __________________________________________________________

Facsimile Number: __________________________________________________________

Email Address (optional): ____________________________________________________

Additional Information: ______________________________________________________

*Client*

Name: ____________________________________________________________________

Address: __________________________________________________________________

___________________________________________________________________________

Telephone Number: __________________________________________________________

Facsimile Number: __________________________________________________________

Email Address (optional): ____________________________________________________

Additional Information: ______________________________________________________

*PICS Contact Person*

Name: ____________________________________________________________________

Address: __________________________________________________________________

___________________________________________________________________________

Telephone Number: __________________________________________________________

Facsimile Number: __________________________________________________________

Email Address (optional): ____________________________________________________

Additional Information: ______________________________________________________

*Identification of the protocol*

This PICS proforma applies to the following Recommendation:

– ITU-T I.366.2, *AAL type 2 service specific convergence sublayer for narrow-band services*.

### V.2.4 Global statement of conformance

The implementation described in this PICS meets all of the mandatory requirements of the reference protocol.

___Yes

___No

NOTE – Answering "No" indicates non-conformance to the specified protocol. Non-supported mandatory capabilities are to be identified in the PICS, with an explanation of why the implementation is non-conforming.

### V.2.4.1 Major capabilities

| Item number | Item description | Reference | Status | Predicate | Support |
|---|---|---|---|---|---|
| MC1 | Is the Audio service category supported? | 8 | O.1 | | |
| MC2 | Is the Multirate service category supported? | 8 | O.1 | | |
| MC3 | Is Audio (voice and voiceband data) supported? | 8.1, 3 | M | MC1 | |
| MC4 | Is Circuit mode data for 64 kbit/s only supported? | 8.2, 3 | O | MC1 | |
| MC5 | Is Circuit mode data for N*64 kbit/s, N ≥ 1 supported? | 8.2, 3 | M | MC2 | |
| MC6 | Is Frame mode data supported? | 8.3, 3 | O | MC1 or MC2 | |
| MC7 | Are Dialled digits supported? | 8.4, 3 | O | MC1 | |
| MC8 | Is Channel-associated signalling supported? | 8.5, 3 | O | MC1 | |
| MC9 | Is Facsimile demodulation/remodulation supported? | 8.6, 3 | O | MC1 | |
| MC10 | Is OAM (Alarms) supported? | 8.7 | M | MC1 or MC2 | |
| MC11 | Is OAM (Loopback) supported? | 8.11 | O | MC1 or MC2 | |
| MC12 | Is User State Control supported? | 8.8, 3 | M<br>O | MC4 or MC9<br>MC3 | |
| MC13 | Is Rate Control supported? | 8.9 | O | MC1 | |
| MC14 | Is Synchronization of change in SSCS operation supported? | 8.10 | O | MC1 | |
| O.1 It is mandatory to support at least one of these options. | | | | | |

### V.2.4.2 Audio (voice and voiceband data)

| Item number | Item description | Reference | Status | Predicate | Support |
|---|---|---|---|---|---|
| AUD1 | Is the type 1 packet format used? | 10.1 | M | MC3 | |
| AUD2 | Are the UUI and length indication fields used to determine the profile entry? | 13.1 | M | AUD1 | |
| AUD3 | Are the sequence numbering procedures of clause 14 implemented? | 14 | M | AUD1 | |
| AUD4 | Is the Profile using PCM-64 supported? | 13.4, Table P.1 | M | MC3 | |
| AUD4.1 | Is it implemented for A-law? | 13.4 | O.1 | AUD4 | |
| AUD4.2 | Is it implemented for μ–law? | 13.4 | O.1 | AUD4 | |
| AUD5 | Is the Profile using PCM-64 and silence supported? | Table P.2 | O | MC3 | |
| AUD6 | Is the Profile using ADPCM and silence supported? | Table P.3 | O | MC3 | |

| Item number | Item description | Reference | Status | Predicate | Support |
|---|---|---|---|---|---|
| AUD7 | Is the Profile using G.728 with higher efficiency supported? | Table P.4 | O | MC3 | |
| AUD8 | Is the Profile using G.728 with lower delay supported? | Table P.5 | O | MC3 | |
| AUD9 | Is the Profile using G.729 with higher efficiency and G.726 for voiceband data supported? | Table P.6 | O | MC3 | |
| AUD10 | Is the Profile using G.729 with lower delay supported? | Table P.7 | O | MC3 | |
| AUD11 | Is the Profile using G.729 with lower delay and G.726-32 for voiceband data at lower rates supported? | Table P.8 | O | MC3 | |
| AUD12 | Is the Profile using G.729 with lower delay and G.726-40 for voiceband data at higher rates supported? | Table P.9 | O | MC3 | |
| AUD13 | Is the Profile using G.729 with full variable bit rates supported? | Table P.10 | O | MC3 | |
| AUD14 | Is the profile using AMR supported? | Table P.11 | O | MC3 | |
| AUD15 | Is the profile using G.723 supported? | Table P.12 | O | MC3 | |
| AUD16 | Is the profile using PCM 64 kbits/s and ADPCM 32 kbits/s supported? | Table P.13 | O | MC3 | |
| O.1 It is mandatory to support at least one of these options. | | | | | |

### V.2.4.3 Circuit mode data for 64 kbit/s only

| Item number | Item description | Reference | Status | Predicate | Support |
|---|---|---|---|---|---|
| CMD1 | Is the type 1 packet format used? | 10.1 | M | MC4 | |
| CMD2 | Is the UUI field used as a sequence number modulo 16? | 15 | M | CMD1 | |
| CMD3 | Is the packet encoding format for N = 1 used? | Annex J | M | MC4 | |

### V.2.4.4 Circuit mode data for N*64 kbit/s N ≥ 1

| Item number | Item description | Reference | Status | Predicate | Support | Values | |
|---|---|---|---|---|---|---|---|
| | | | | | | Allowed | Supported |
| CMN1 | What values of N are supported? | 8.2 | M | MC5 | | 1-31 | |
| CMN2 | Is the type 1 packet format used? | 10.1 | M | MC5 | | | |
| CMN3 | Is the UUI field used as a sequence number modulo 16? | 15 | M | CMN2 | | | |
| CMN4 | Is the packet encoding format used corresponding to the values of N supported? | Annex J | M | CMN1 | | | |

### V.2.4.5 Frame mode data

| Item number | Item description | Reference | Status | Predicate | Support | Values | |
|---|---|---|---|---|---|---|---|
| | | | | | | Allowed | Supported |
| FMD1 | Is data octet aligned? | 16 | M | MC6 | | | |
| FMD2 | What is the maximum length of the frame mode data unit supported? | Table 18-1 | M | MC6 | | 1-65535 | |
| FMD3 | Is the transmission error detection capability (defined in ITU-T I.366.1) used? | 16 | M | MC6 | | | |
| FMD4 | Is the packet format of the SSTED-PDU defined in 8.3/I.366.1 used? | 10 | M | MC6 | | | |
| FMD5 | Are the UUI codepoints 26/27 used to delineate the sequence of data? | 16 | M | MC6 | | | |

### V.2.4.6 Dialled digits

| Item number | Item description | Reference | Status | Predicate | Support |
|---|---|---|---|---|---|
| DDG1 | Is DTMF supported? | 8.4, K.1, K.2 | O.1 | MC7 | |
| DDG2 | Is MF-R1 supported? | 8.4, K.1, K.2 | O.1 | MC7 | |
| DDG3 | Is MF-R2 supported? | 8.4, K.1, K.2 | O.1 | MC7 | |
| DDG4 | Is the type 3 packet format used with UUI codepoint 24? | 10.2, Table 12-1, Figure K.1 | M | MC7 | |
| DDG5 | Is the packet format for dialled digits (DTMF) supported? | K.2, Table K.2 | M | DDG1 | |
| DDG6 | Is the packet format for dialled digits (MF-R1) supported? | K.2, Table K.3 | M | DDG2 | |
| DDG7 | Is the packet format for dialled digits (MF-R2) supported? | K.2, Table K.4 | M | DDG3 | |
| DDG8 | Are type 3 packets sent using triple redundancy? | 11.2, K.3 | M | DDG4 | |
| DDG9 | Is a fixed interval of 5 ms used between transmissions? | 11.2, K.3 | M | DDG8 | |
| DDG10 | If a tone persists, is a refresh sent every 500 ms? | K.3 | M | MC7 | |
| DDG11 | Does the receiver allow for all three copies of a transition to be received before indicating the dialled digit? | 11.2, K.4 | O | MC7 | |
| DDG12 | Is the relative event timer used to schedule playout? | 11.1 | M | MC7 | |
| O.1 It is mandatory to support at least one of these options. | | | | | |

### V.2.4.7 Channel-associated signalling

| Item number | Item description | Reference | Status | Predicate | Support |
|---|---|---|---|---|---|
| CAS1 | Is the type 3 packet format used with UUI codepoint 24? | 10.2, Table 12-1, Figure L.1 | M | MC8 | |
| CAS2 | Is the packet format for CAS supported? | L.2 | M | MC8 | |
| CAS3 | Are type 3 packets sent using triple redundancy? | 11.2, L.3 | M | CAS1 | |
| CAS4 | Is a fixed interval of 5 ms used between transmissions? | 11.2, L.3 | M | CAS3 | |
| CAS5 | Is a refresh sent every 5 seconds? | L.3 | M | MC8 | |
| CAS6 | Does the receiver allow for all three copies of a transition to be received before indicating the CAS bits? | 11.2, L.4 | O | MC8 | |
| CAS7 | Is the relative event timer used to schedule playout? | 11.1 | M | MC8 | |

### V.2.4.8 Facsimile demodulation/remodulation

| Item number | Item description | Reference | Status | Predicate | Support |
|---|---|---|---|---|---|
| FDR1 | Is the modulation type V.17 supported? | 17.1 | O | MC9 | |
| FDR2 | Is the modulation type V.27 *ter* supported? | 17.1 | O | MC9 | |
| FDR3 | Is the modulation type V.29 supported? | 17.1 | O | MC9 | |
| FDR4 | Is the modulation type V.33 supported? | 17.1 | O | MC9 | |
| FDR5 | Do facsimile image data packets use the type 1 packet format? | 10.1, 17.7, M.3 | M | MC9 | |
| FDR6 | Is the UUI field for type 1 packets used as a sequence number modulo 16 | M.3 | M | FDR4 | |
| FDR7 | Do facsimile demodulation control packets use the type 3 packet format with UUI codepoint 24? | 10.2, Table 12-1, 17.7, M.2 | M | MC9 | |
| FDR8 | Are type 3 packets sent using triple redundancy? | 11.2, M.1.4, M.1.7 | M | FDR6 | |
| FDR9 | Is a fixed interval of 20 ms used between transmissions? | 11.2, M.1.4, M.1.7 | M | FDR7 | |
| FDR10 | Are non-standard T.30 facilities supported? | 17.3 | O | MC9 | |

### V.2.4.9 OAM

| Item number | Item description | Reference | Status | Predicate | Support |
|---|---|---|---|---|---|
| OAM1 | Is the type 3 alarm packet format used with UUI codepoint 31? | 10.2, Table 12-1, Figure N.1 | M | MC10 | |
| OAM2 | Is the type 3 loopback packet format used with UUI codepoint 31? | 10.2, Table 12-1, Figure N.2 | M | MC11 | |
| OAM3 | Is the OAM alarm packet format used? | N.2.1 | M | MC10 | |
| OAM4 | Is the OAM loopback packet format used? | N.2.2 | M | MC11 | |
| OAM5 | While an alarm condition persists, is an alarm packet transmitted at least once per second? | N.3.1 | M | MC10 | |
| OAM6 | If 3.5 seconds elapses without signal reassertion, is the condition removed? | N.3.1 | M | MC10 | |
| OAM7 | If 5 seconds elapses without a loopback packet being return, is the loopback removed? | N.3.2 | M | MC11 | |
| OAM8 | Is the type 2 AIS condition removed on receipt of packets other than OAM packets? | N.3 | M | MC10 | |

### V.2.4.10 User state control

| Item number | Item description | Reference | Status | Predicate | Support |
|---|---|---|---|---|---|
| USC1 | Is the initial User state set to Voice on each AAL type 2 connection? | 8.8 | M | MC11 | |
| USC2 | Is the type 3 packet format used with UUI codepoint 24? | 10.2, Table 12-1, Figure O.1 | M | MC11 | |
| USC3 | Is the User state control packet format used? | O.2 | M | MC11 | |
| USC4 | Are packets sent using triple redundancy? | 11.2, O.3 | M | MC11 | |
| USC5 | Is a fixed interval of 20 ms used between transmissions? | 11.2, O.3 | M | USC4 | |
| USC6 | Does the receiver act on the first packet received and filter out any repetitions? | 11.2, O.3 | M | MC11 | |

### V.2.4.11 UUI codepoints

| Item number | Item description | Reference | Status | Predicate | Support |
|---|---|---|---|---|---|
| UCP1 | Are the reserved UUI codepoints 16-23 used? | Table 12-1 | X | MC1 or MC2 | |
| UCP3 | Is the UUI codepoint 25 used for non-standard extensions? | Table 12-1 | O | MC1 or MC2 | |
| UCP4 | Are the reserved UUI codepoints 28-30 used? | Table 12-1 | X | MC1 or MC2 | |

### V.2.4.12 SSCS parameters of operation

| Item number | Item description | Reference | Status | Predicate | Support |
|---|---|---|---|---|---|
| SPO1 | Are the values of the SSCS parameters of operation determined before this SSCS is used on an individual type 2 connection? | 18 | M | MC1 or MC2 | |
| SPO2 | In the absence of determination of SSCS parameters of operation via signalling or provisioning are the default values used? | Table 18-1 | M | MC1 | |
| SPO3 | If the service category of an individual AAL type 2 connection is audio, is the transport of circuit mode data with N > 1 prohibited? | Table 8-1, Table 18-1 | M | MC1 and MC2 | |
| SPO4 | If the service category of an individual AAL type 2 connection is multirate, is the transport of audio, dialled digits, channel associated signalling, facsimile demodulation/ modulation, and user state control prohibited? | Table 8-1, Table 18-1 | M | MC1 and MC2 | |

#### V.2.4.13 Rate control

| Item number | Item description | Reference | Status | Predicate | Support |
|---|---|---|---|---|---|
| RC1 | Is the type 3 packet format used with UUI codepoint 24? | 10.2, Table 12-1, Figure R.1 | M | MC13 | |
| RC2 | Is the rate control packet format used? | R.2 | M | MC13 | |
| RC3 | Are packets sent using triple redundancy? | R.2 | M | MC13 | |
| RC4 | Is a fixed interval of 5 ms used between transmissions? | R.4 | M | RC3 | |
| RC5 | Does the receiver act on the first packet received and filter out any repetitions? | R.3 | M | MC13 | |

#### V.2.4.14 Synchronization of change in SSCS operation

| Item number | Item description | Reference | Status | Predicate | Support |
|---|---|---|---|---|---|
| SYN1 | Is the type 3 packet format used with UUI codepoint 24? | 10.2, Table 12-1, Figure S.1 | M | MC14 | |
| SYN2 | Is the synchronization of change in SSCS operation packet format used? | S.2 | M | MC14 | |
| SYN3 | Are packets sent using triple redundancy? | S.2 | M | MC14 | |
| SYN4 | Is a fixed interval of 5 ms used between transmissions? | S.3 | M | SYN3 | |
| SYN5 | Does the receiver act on the first packet received and filter out any repetitions? | S.4 | M | MC14 | |

## APPENDIX VI

## Use of this Recommendation for mobile applications

This appendix provides guidelines on the use of this Recommendation to support mobile applications. Table VI.1 is an adaptation of Table 18-1 showing the level of functionality recommended to support current mobile applications.

**Table VI.1/I.366.2 – SSCS parameters of operation for mobile applications**

| SSCS parameter | | Audio Service category | | Multirate Service category |
|---|---|---|---|---|
| | | **Permitted values** | **Default value** | **Permitted values** |
| 1 | Service category | Audio | Audio | N/A |
| 2 | Transport of audio information | enabled | enabled | N/A |
| 3 | Source of the encoding format profile | ITU-T predefined, other predefined, custom | ITU-T predefined | N/A |
| 3a | ITU-T predefined profile (Annex P, Figure P.1) | 1 … 13 | None | N/A |
| 3b | Other predefined profile | 1 … 255 | N/A | N/A |
| 3c | Custom profile: description of its content | for further study | N/A | N/A |
| 4 | Interpretation of the generic PCM encoding format defined in Annex B | N/A | N/A | N/A |
| 5 | Transport of demodulated facsimile data | N/A | N/A | N/A |
| 6 | Transport of channel associated signalling bits | N/A | N/A | N/A |
| 7 | Transport of DTMF dialled digits | enabled, disabled | disabled | N/A |
| 8 | Transport of MF-R1 dialled digits | N/A | N/A | N/A |
| 9 | Transport of MF-R2 dialled digits | N/A | N/A | N/A |
| 10 | Transport of circuit mode data | N/A | N/A | N/A |
| 10a | Multiplier N in N*64 kbit/s circuit mode data | N/A | N/A | N/A |
| 11 | Transport of frame mode data | N/A | N/A | N/A |
| 11a | Maximum length of a frame mode data unit | N/A | N/A | N/A |
| 12 | User Rate control | enabled, disabled | enabled | N/A |

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