Voice over Frame Relay Implementation Agreement

FRF.11

Frame Relay Forum Technical Committee

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Revision History

Version	Change	Date
1.0	Approved	May, 1997

1. Introduction

1.1 Purpose

Frame relay is now a major component of many network designs. The protocol provides a minimal set of switching functions to forward variable sized data payloads through a network. The basic frame relay protocol, described in the Frame Relay Forum User to Network (UNI) and Network to Network (NNI) Implementation Agreements, has been augmented by additional agreements which detail techniques for structuring application data over the basic frame relay information field. These techniques enabled successful support for data applications such as LAN bridging, IP routing, and SNA.

This specification extends frame relay application support to include the transport of digital voice payloads. Frame formats and procedures required for voice transport are described in this Implementation Agreement. This specification addresses the following requirements:

- Transport of compressed voice within the payload of a frame relay frame
- Support a diverse set of voice compression algorithms
- Effective utilization of low-bit rate frame relay connections
- Multiplexing of up to 255 sub-channels on a single frame relay DLCI
- Support of multiple voice payloads on the same or different sub-channel within a single frame
- Support of data sub-channels on a multiplexed frame relay DLCI

Transport of compressed voice is provided with a generalized frame format that supports multiplexing of sub-channels on a single frame relay DLCI. Support for the unique needs of the different voice compression algorithms is accommodated with algorithm-specific "transfer syntax" definitions. These definitions establish algorithm specific frame formats and procedures. Annexes describing different transfer syntax definitions are found at the end of this document.

Transport of supporting information for voice communication, such as signalling indications (e.g., ABCD bits), dialed digits, and facsimile data, is also provided through the use of transfer syntax definitions specific to the information being sent.

1.2 Overview of Agreement

A description of the reference model and service description for the voice over frame relay (VoFR) service is provided in Section 2, along with the concept of a Voice Frame Relay Access Device (VFRAD).

Specification of the frame formats and procedures is provided in Section 3.

Transfer syntax definitions for individual voice compression algorithms as well as generic supporting information (e.g., dialed digits) are provided in Annex sections at the conclusion of the document. Figure 1-1 illustrates some of the transfer syntax definitions used for Voice Over Frame Relay.

Vocoders					Other			
G.729 CS-ACELP	G.728 LD CELP	G.723.1 MP-MLQ	G.726/G.727 ADPCM	G.711 PCM	Dialed Digits	CAS	Data Transfer	Fax Relay

Figure 1-1 Transfer Syntax Examples

1.3 Voice Frame Relay Access Device (VFRAD)

A voice over frame relay access device supports voice services. A VFRAD may be positioned between a PBX or key set and the frame relay network. Alternatively, the VFRAD may be integrated into an end-system that directly supports telephony applications and frame relay. The VFRAD multiplexes voice and fax traffic along with data traffic from a variety of services/sources into a common frame relay connection.

1.4 Definitions

Must, Shall or Mandatory- the item is an absolute requirement of this implementation agreement.

Should - the item is desirable.

May or Optional - the item is not compulsory, and may be followed or ignored according to the needs of the implementor.

1.5 Acronyms

ADPCM	Adaptive Differential Pulse Code Modulation
AIS	Alarm Indication Signal
B_c	Committed Burst
B_e	Excess Burst
BECN	Backward Explicit Congestion Notification
BER	Bit Error Rate
CAS	Channel Associated Signalling
CS-ACELP	Conjugate Structure – Algebraic Code Excited Linear Predictive
CELP	Code Excited Linear Prediction
CID	Channel Identification
CIR	Committed Information Rate
CCS	Common Channel Signalling
DE	Discard Eligibility

DLCI	Data Link Connection Identifier
DTMF	Dual Tone Multi-Frequency
E-ADPCM	Embedded Adaptive Differential Pulse Code Modulation
FAX	Facsimile Group 3
FECN	Forward Explicit Congestion Notification
FRAD	Frame Relay Access Device
HDLC	High Level Data Link Control
IA	Implementation Agreement
I/F	Interface
IWF	Inter-working Function
LD-CELP	Low Delay - Code Excited Linear Prediction
lsb	Least Significant Bit
MP-MLQ	Multi Pulse Maximum Likelihood Quantizer
msb	Most Significant Bit
PCM	Pulse Code Modulation
PVC	Permanent Virtual Connection
SID	Silence Information Descriptor
UNI	User Network Interface
VAD	Voice Activity Detection
VFRAD	Voice Frame Relay Access Device
VoFR	Voice Over Frame Relay
Vocoder	Voice coder/decoder

1.6 Relevant Standards

[1]	FRF.1.1	Frame Relay User-to-Network Implementation Agreement, January 1996
[2]	FRF.3.1	Multiprotocol Encapsulation Implementation Agreement, June 22, 1995
[3]	FRF.12	Frame Relay Fragmentation Implementation Agreement, 1997
[4]	ITU G.711	Pulse Code Modulation of Voice Frequencies, 1988
[5]	ITU G.723.1	Dual Rate Speech Coder for Multimedia Communications Transmitting at 5.3 & 6.3 kbit/s, March 1996
[6]	ITU G.726	40, 32, 24, 16 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM), March 1991
[7]	ITU G.727	5-,4-,3- and 2 bits Sample Embedded Adaptive Differential Pulse Code Modulation, November 1994
[8]	ITU G.728	Coding of Speech at 16 kbit/s Using Low-Delay Code Excited Linear Prediction, November 1994
[9]	ITU G.729	Coding of Speech at 8kbit/s using Conjugate Structure - Algebraic Code Excited Linear Predictive (CS-ACELP) Coding, March 1996
[10]	ITU G.764	Voice Packetization – Packetized voice protocols, December 1990
[11]	ITU T.30	Terminal Equipment and protocol for Telematic Service/Procedure for Facsimile General Switch Networks, March 1993

2. Reference Model and Service Description

2.1 Frame Relay Access

A VFRAD uses the frame relay service at the UNI as a transmission facility for voice, voice signalling, and data. The reference model for voice over frame relay is shown in Figure 2-1. Using the Voice over Frame Relay (VoFR) service, it is possible for any type of VFRAD on the left-hand side of Figure 2-1 to exchange voice and signalling with any type of VFRAD on the right-hand side of Figure 2-1.

Three types of devices are shown in Figure 2-1. The top layer shows end-system devices similar to telephones or FAX machines. The middle layer shows transparent multiplexing devices similar to channel banks. The bottom layer shows switching system devices similar to PBX's.

A VFRAD connects to a frame relay UNI via physical interfaces as defined in [1].

2.1.1 End-System Devices

The top left device in Figure 2-1 could be a PC with FAX or telephony application software using a frame relay network port for connectivity to other VFRAD devices. Such an end-system could use the VoFR protocol stack on a frame relay connection to another end-system (top right). It could also use the VoFR protocol stack on a connection to a transparent channel bank into a private network (middle right) or to a PBX (bottom right).

2.1.2 Transparent-Multiplexing Devices

The middle left device in Figure 2-1 could be a Channel Bank connected via analog trunks to an external PBX (not shown). Such a multiplexing device could use the VoFR protocol stack on a frame relay connection to an end-system (top right), another channel bank (middle right) or a PBX (bottom right).

2.1.3 Switching-System Devices

The bottom left device in Figure 2-1 could be a PBX using a frame relay network for connection to off premise extensions (end-systems) or as trunks to other PBX devices. Such a switching system device could use the VoFR protocol stack on a frame relay connection to an end-system (top right), a channel bank (middle right), or another PBX (bottom right).

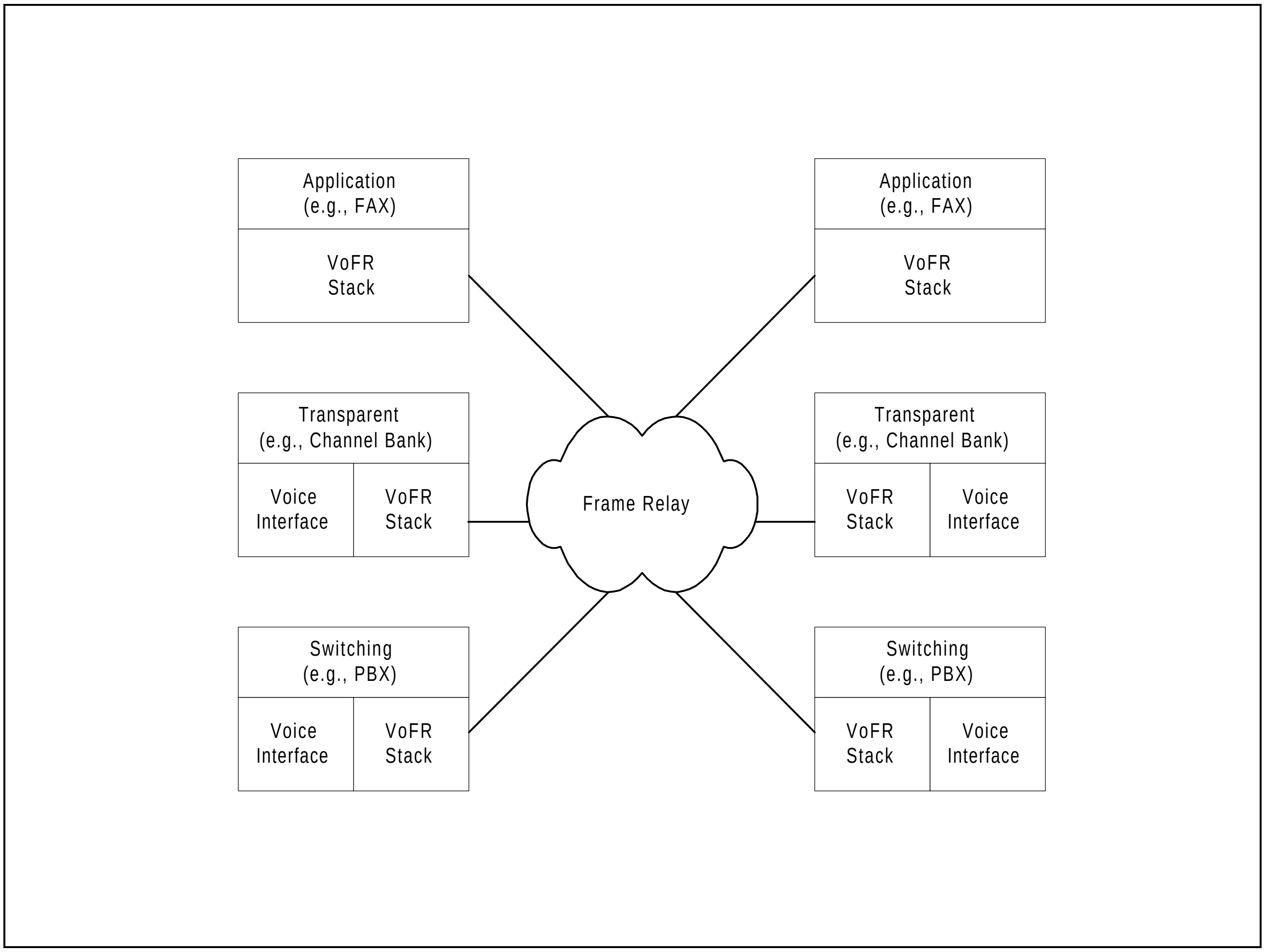

Figure 2-1 Voice over Frame Relay Network Reference Model

2.2 Voice Interfaces

The requirements for implementation of voice interfaces are beyond the scope of this implementation agreement.

2.3 Voice Over Frame Relay Service Description

This implementation agreement defines formats and procedures that support a VoFR service. Elements of the VoFR service support various types of service users that may be performing any of the following voice applications:

1. Call origination and termination — for an end-system

2. Transparent interworking between individual sub-channels on a VoFR interface and sub-channels on another type of voice interface.

3. Call-by-call Switching — for a switching system to terminate an incoming call and originate a call on another voice interface.

To support the VoFR service, the underlying protocol stack must provide a full duplex transport service. The service users can use the following service elements to operate a voice connection. The service elements support the transport of two types of payloads: primary payloads and signalled payloads. Refer to Section 3.1 for a discussion of payload types.

2.3.1 Primary Payloads

2.3.1.1 Encoded Voice

This service element conveys voice information supplied by the service user. The voice information is packaged according to the rules specified by a voice transfer syntax. Voice transfer syntax definitions for various voice compression schemes are described in the annexes of this IA.

2.3.1.2 Encoded FAX or Voice-Band Modem Data

The service users can exchange digital data in a “baseband” format suitable for re-modulation into a FAX or analog modem signal. The VoFR service transports this information between the two service users.

The transmitting service user may locally detect the presence of a FAX or voice-band modem signal for the voice connection and demodulate it before sending it. The receiving service user can detect arriving packets that contain demodulated data and can reconstruct the original modulated signal instead of reconstructing a speech signal.

The encoded FAX or voice-band data payload format is within the scope of this IA. The algorithms used for demodulation and re-modulation of FAX and/or voice-band data are outside the scope of this IA.

The transfer syntax for FAX is described in Annex D.

The transfer syntax for voice band modem data is for further study.

2.3.1.3 Data Frames

This service element conveys data frames supplied by the service user. The frames are packaged according to the rules specified by Annex C.

The content of the data frames is transparent to the VoFR service.

One application of the data frame service element enables transparent tunneling of common channel signalling messages between two compatible end-points (e.g., PBX interfaces). Common channel signalling message formats and procedures are beyond the scope of this agreement.

2.3.2 Signalled Payload

2.3.2.1 Dialed Digits

This service element transparently conveys DTMF, pulse, or other dialed digits supplied by the service user. These digits may be sent during the voice call setup or following call establishment to transfer in-band tones.

2.3.2.2 Signalling Bits (Channel Associated Signalling)

This service element transparently conveys signalling bits supplied by the service user. These bits may indicate seizure and release of a connection, dial pulses, ringing, or other information in accordance with the signalling system in use over the transmission facility.

2.3.2.3 Fault Indication

The service users can use this service to convey an alarm indication signal.

2.3.2.4 Message-Oriented Signalling (Common Channel Signalling)

Refer to section 2.3.1.3

2.3.2.5 Encoded FAX

Refer to Section 2.3.1.2

Encoded FAX may be transmitted on a sub-channel that utilizes a primary payload for encoded voice. In this case, the sub-frames containing the encoded FAX must be sent as a signalled payload.

2.3.2.6 Silence Information Descriptor

Silence Information Descriptor (SID) sub-frames indicate the end of a talk-spurt and convey comfort noise generation parameters. These SID indications support voice activity detection (VAD) and silence suppression schemes.

When VAD is utilized, a SID sub-frame may optionally be transmitted following the last encoded voice sub-frame of a talk-spurt. Reception of a SID sub-frame after a voice sub-frame may be interpreted as an explicit indication of end of talk-spurt. In addition, SID sub-frames may be transmitted at any time during the silence interval to update comfort noise generation parameters.

The SID payload is defined for PCM and ADPCM encoding in the appropriate annexes (Annex F and Annex G). The SID payload definition for other voice encoding algorithms is for further study and can be null.

SID sub-frames should not be sent if VAD is not utilized.

2.4 VFRAD Configuration Requirements

VoFR devices compliant with this implementation agreement are not required to negotiate operational parameters. Negotiation procedures are for further study. Therefore, at the time of provisioning, the network manager must configure end-to-end configuration parameters (e.g., Vocoder.). End-point devices providing the VoFR service are configured with compatible sub-channel assignments, signalling, compression algorithms, and other options.

2.5 VoFR Service Block Diagram

The relationship of the Voice Over Frame Relay service, VoFR Service user and the frame relay service is shown in Figure 2-2.

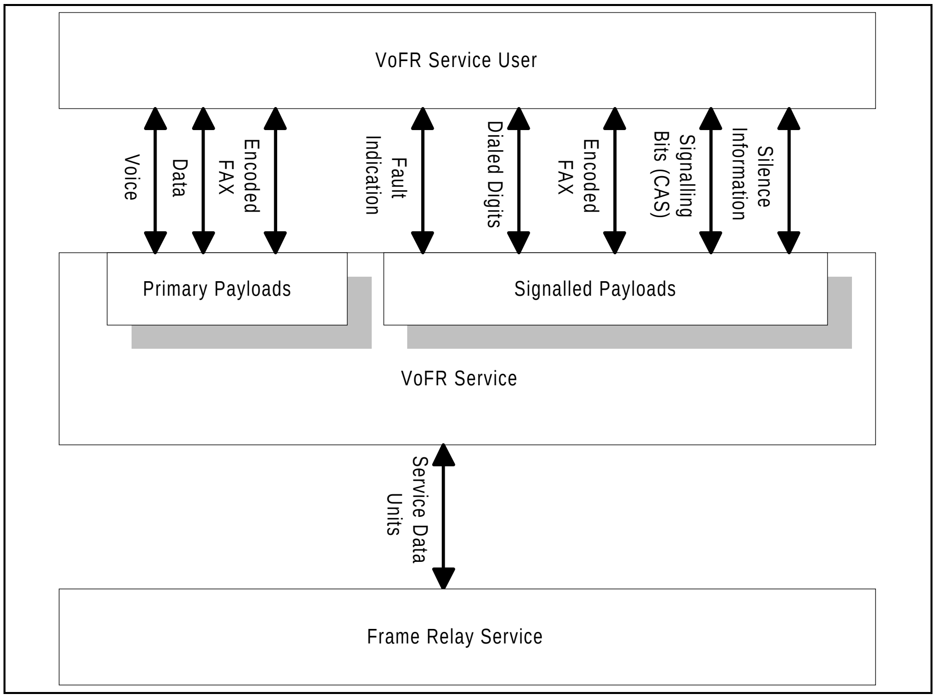

Figure 2-2 VoFR Service Block Diagram.

2.6 Service Multiplexing

The Frame Relay UNI can support multiple PVCs, each of which can provide VoFR service. The VoFR service supports multiple voice and data channels on a single frame relay data link connection. The VoFR service delivers frames on each sub-channel in the order they were sent.

As shown in Figure 2-3 each instance of the voice/data multiplexing layer can support one or more voice connections and data protocol stacks over a single frame relay PVC. The mechanism for separation of the voice and data connections being supported over a single frame relay PVC is within the scope of this IA. The mechanisms and protocol stacks used for data connections are covered in other Frame Relay Forum IA's and relevant standards.

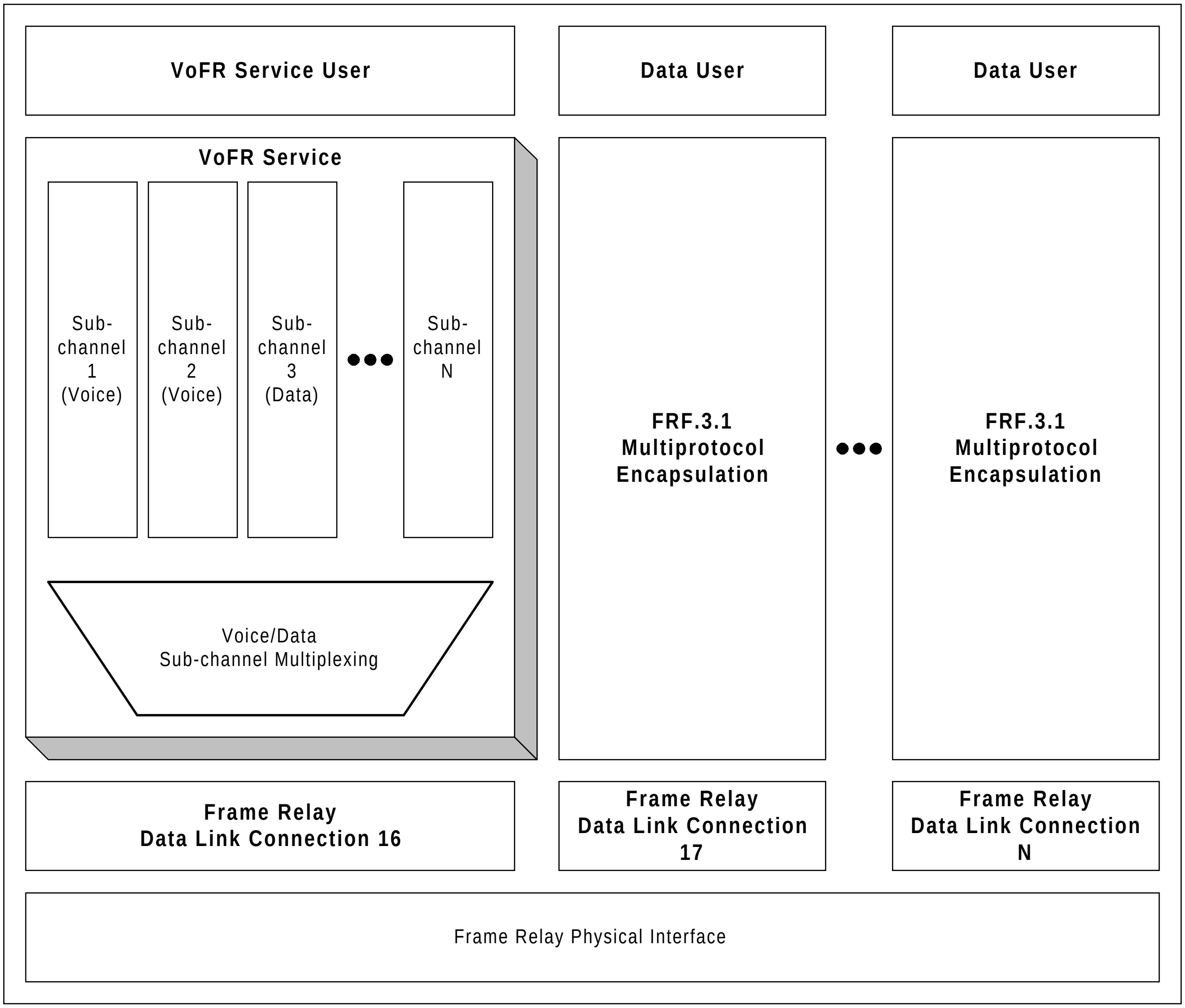

Figure 2-3 Voice Over Frame Relay Multiplexing Model

3. Frame Formats

Voice and data payloads are multiplexed within a voice over frame relay data link connection by encapsulation within the frame format specified in [1]. Each payload is packaged as a sub-frame within a frame's information field. Sub-frames may be combined within a single frame to increase processing and transport efficiencies. Each sub-frame contains a header and payload. The sub-frame header identifies the voice/data sub-channel and, when required, payload type and length. Refer to Figure 3-1 for an illustration of sub-frames. In this example, a single DLCI supports 3 voice channels and 1 data channel. Three voice payloads are packaged in the first frame and a data payload is contained in the second frame.

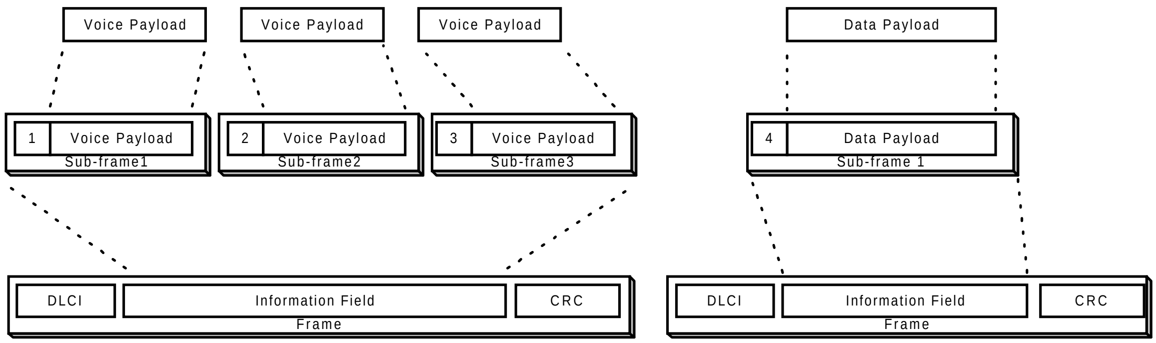

Figure 3-1 Relationship between frames and sub-frames

3.1 Payloads

3.1.1 Primary payload

Each sub-channel of a VoFR connection transports a primary payload. A primary payload contains traffic that is fundamental to operation of a sub-channel. Other payloads may be sent to support the primary payload (e.g., dialed digits for a primary payload of encoded voice). These additional payload types are differentiated from the primary payload by a signalled encoding in the payload type field of the sub-frame. A payload type of all zeros always indicates the primary payload.

Three basic types of primary payloads are utilized: encoded voice payloads, encoded FAX payloads, and data payloads. Refer to the appropriate annex for a description of the transfer syntax which supports these payload types.

3.1.2 Signalled Payload

Payloads containing in-band information, which augment the primary payload flow, are indicated using payload type codings. These signalled payloads include information such as channel-associated signalling, dialed digits, in-band encoded FAX relay, and fault indications. Refer to the appropriate annex for a description of the service elements which support the signalled payloads.

3.2 Sub-frame Format

Each sub-frame consists of a variable length header and a payload. The minimal sub-frame header is a single octet containing the least significant bits of the voice/data channel identification along with extension and length indications. An extension octet containing the most significant bits of the voice/data channel identification and a payload type is present when the Extension Indication is set. A payload length octet is present when the Length Indication is set. Refer to Figure 3-2 and Table 3-1 for a description of the sub-frame structure.

Bits

<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>Octets</td></tr>
<tr><td>EI</td><td>LI</td><td colspan="6">Sub-channel Identification (CID)
(Least significant 6 bits)</td><td>1</td></tr>
<tr><td colspan="2">CID
(msb)</td><td>0
Spare</td><td>0
Spare</td><td colspan="4">Payload Type</td><td>1a
(Note 1)</td></tr>
<tr><td colspan="8">Payload Length</td><td>1b
(Note 2)</td></tr>
<tr><td colspan="8">Payload</td><td>p</td></tr>
</table>

NOTES:
1. When the EI bit is set, the structure of Octet 1a given in Table 3-1 applies.
2. When the LI bit is set, the structure of Octet 1b given in Table 3-1 applies.
3. When both the EI bit and the LI bit are set to 1 both Octet 1a and 1b are used.

Figure 3-2 Sub-frame format

Extension indication (octet 1)

The extension indication (EI) bit is set to indicate the presence of octet 1a. This bit must be set when a sub-channel identification value is > 63 or when a payload type is indicated. Each transfer syntax has an implicit payload type of zero when the EI bit is cleared.

Length indication (octet 1)

The length indication (LI) bit is set to indicate the presence of octet 1b. The LI bit of the last sub-frame contained within a frame is always cleared and the payload length field is not present. The LI bits are set for each of the sub-frames preceding the last sub-frame.

Sub-channel identification (octets 1 and 1a)

The six least significant bits of the sub-channel identification are encoded in octet 1. The two most significant bits of the sub-channel identification are encoded in octet 1a. A zero value in the two most significant bits is implied when octet 1a is not included in the VoFR header (EI bit cleared). Sub-channel identifiers 0000 0000 through 0000 0011 are reserved in both the short and long format.

Payload type (octet 1a)

This field indicates the type of payload contained in the sub-frame.

Bits 4	3	2	1	
0	0	0	0	Primary payload transfer syntax
0	0	0	1	Dialed digit transfer syntax (Annex A)
0	0	1	0	Signalling bit transfer syntax (Annex B)
0	0	1	1	Fax relay transfer syntax (Annex D)
0	1	0	0	Silence Information Descriptor

A zero value for the payload type is implied when octet 1a is not in included in the header (EI bit cleared).

Payload length (octet 1b)

Payload length contains the number of payload octets following the header. A payload length indicates the presence of two or more sub-frames packed in the information field of the frame.

Payload (octet p)

The payload contains octets as defined by the applicable transfer syntax assigned to the sub-channel or as indicated by the payload type octet 1a.

Table 3-1 Sub-frame format

3.3 Sub-frame Examples

The diagrams in this section illustrate some of the possible combinations of sub-frames. Figure 3-3 shows a frame which contains a single voice payload for a low-numbered sub-channel. Octets 1a and 1b are not required. The payload, a CS-ACELP sample, starts after octet 1.

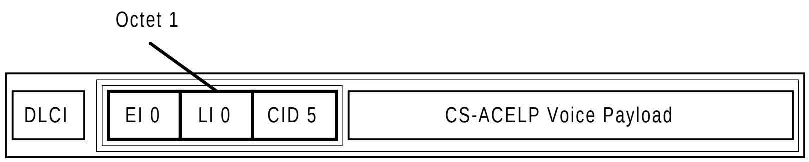

Figure 3-3 Frame containing one sub-frame

Figure 3-4 shows a frame which contains a single voice payload for a high-numbered channel (>63). Octet 1a must be included. Note that the payload type is zero, indicating the transfer syntax that has been configured for the channel. In this example, the transfer syntax is the CS-ACELP syntax.

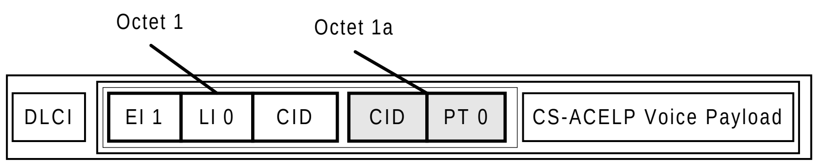

Figure 3-4 Frame containing one subframe for a high numbered channel

Figure 3-5 shows a frame which contains multiple sub-frames for channels 5 and 6. In this case, the payload type is non-zero and octet 1a is required to encode the payload type. The first of the two sub-frames includes octet 1b with the encoding of payload length.

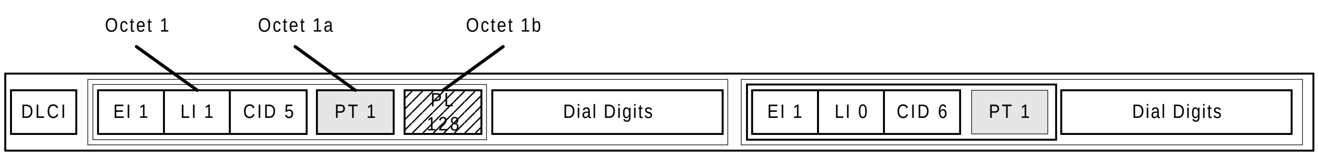

Figure 3-5 Frame containing multiple sub-frames

Figure 3-6 shows a frame which contains multiple sub-frames for channels 5 and 6. In this case, the payload type is zero and the payload length (octet 1b) appears in the first of the two sub-frames.

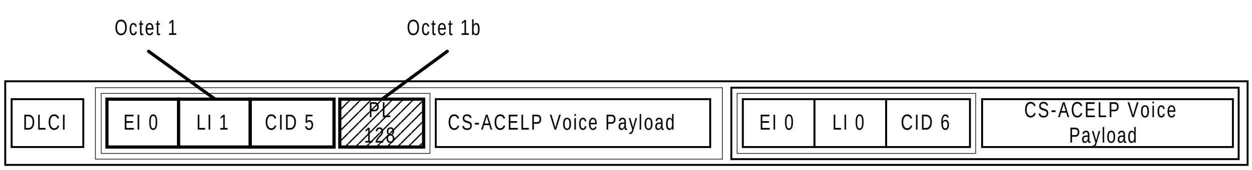

Figure 3-6 Frame containing multiple sub-frames

4. Minimum Requirements for Conformance

This agreement provides support for several optional transfer syntax definitions. Interoperability between VoFR devices is possible only when both devices share support for one or more common transfer syntax definitions. VoFR devices are classified based on the support provided for the common transfer syntax definitions. Class 1 compliant devices support capabilities suitable for high bit-rate interfaces. Class 2 compliant devices support capabilities that enable optimal performance over low bit-rate frame relay interfaces. An implementation is compliant with this agreement if the requirements for at least one of the two classes are met.

4.1 Class 1 Compliance Requirements

4.1.1 Frame Formats

1. Support the frame structure described in section 3.
2. Received optional frames may be discarded.

4.1.2 Primary Payload Types

1. Support of G.727 as described in Annex F is mandatory. Support of other vocoders described in Annex F is optional.
2. A transmit rate of 32Kbps is mandatory.
3. Support for rates of 32kbps, 24kbps, 16kbps are mandatory at the receiver.
4. Support for other primary payload transfer syntax definitions (e.g., FAX) is optional.

4.1.3 Signalled Payload Types

1. Support for the dialed digit signalled payload type is optional.
2. Support for the signalling bits signalled payload type (CAS and AIS) is mandatory.
3. Support for the encoded FAX signalled payload type is optional.

4.2 Class 2 Compliance Requirements

4.2.1 Frame Formats

1. Support the frame structure described in section 3.
2. Received optional frames may be discarded.

4.2.2 Primary Payload Types

1. Support for Annex E CS-ACELP G.729 or G.729A voice transfer syntax is mandatory.

5. Support for other primary payload transfer syntax definitions (e.g., FAX) is optional.

4.2.3 Signalled Payload Types

1. Support for the dialed digit signalled payload type is mandatory.

2. Support for the signalling bits signalled payload type (CAS and AIS) is mandatory.

3. Support for the encoded FAX signalled payload type is optional.

Annex A – Dialed Digit Transfer Syntax

A.1 Reference Documents

None

A.2 Transfer Structure

The Dialed Digit Transfer Syntax is comprised of the Dialed Digit Payload Format and the Dialed Digit Transfer Procedure.

A.3 Dialed Digit Payload Format

At the originating VFRAD the detected digits are inserted into a Dialed Digit Payload by the Dialed Digits Service element. Payload carrying digits will be identified using the VoFR sub-frame payload type field codepoint for the Dialed Digit transfer syntax. The digits will automatically be associated with the corresponding voice traffic based on the Channel ID field.

Each Digit Payload contains three windows of digit transition. The first window represents the current 20ms period [0], the second [-1] is the recent period and the third [-2] is the previous.

Bits

8	7	6	5	4	3	2	1	Octet
Sequence Number								P
reserved 000			Signal Level					P+1
Digit Type [0]			Edge Location [0]					P+2
reserved 000			Digit Code [0]					P+3
Digit-Type[-1]			Edge-Location[-1]					P+4
reserved 000			Digit-Code[-1]					P+5
Digit-Type[-2]			Edge-Location[-2]					P+6
reserved 000			Digit-Code[-2]					P+7

Figure A - 1 Dialed Digit Payload Type

A.3.1 Sequence Number

The sequence field is an 8-bit number that is incremented for every fragment transmitted. The sequence field wraps from all ones to zero in the usual manner of such sequence numbers. Each increment of the sequence represents a period of 20ms.

A.3.2 Signal Level

The power level of each frequency is between 0 to -31 in -dBm0. Power levels above zero dBm0 are coded 00000. In the event that one dialed digit payload contains a transition from one dialed digit to another dialed digit, the signal level field applies to the dialed digit in the "current" 20 ms period.

Code	Power Level dBm0
00000	0
00001	-1
000010	-2
000011	-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	-16
10001	-17
10010	-18
10011	-19
10100	-20
10101	-21
10110	-22
10111	-23
11000	-24
11001	-25
11010	-26
11011	-27
11100	-28
11101	-29
11110	-30
11111	-31

Figure A - 2 Signal Level

A.3.3 Digit Type

Code	**Digit Type**
000	Digit Off
001	DTMF On
010-111	Reserved

Figure A - 3 Digit Types

A 20ms window is used to encode the edge when a digit is turned on and off. This is the delta time, 0ms (00000) to 19ms (10011), from the beginning of the current frame in ms. If there is no transition, the edge location will be set to 0 and the Digit Type of the previous windows will be repeated.

A.3.4 Digit-Code

The following DTMF digit codes are encoded when dialed digit type = DTMF ON.

Digit Code	DTMF Digits
00000	0
00001	1
00010	2
00011	3
00100	4
00101	5
00110	6
00111	7
01000	8
01001	9
01010	*
01011	#
01100	A
01101	B
01110	C
01111	D
10000-11111	Reserved

Figure A - 4 DTMF Digit Codes

A.4 Dialed Digit Transfer Procedures

A.4.1 Procedure for Transmission of Dialed Digit Payloads

When the transmitter detects a validated digit, or has addressing information to send it will start sending a Dialed Digit Payload every 20ms. Since each payload covers 60ms of Digit on/off edge information, there is redundancy of the edge information. The sequence number is incremented by one in each transmitted payload.

When the digit activity is off, the transmitter should continue to send three more Dialed Digit Payloads for 60ms.

A.4.2 Procedure for Interpreting Received Dialed Digit Payloads

When the receiver gets a Dialed Digit Payload or accepts the received addressing information, it will generate digits according to the location of the on and off edges. Silence will be applied to the duration after an off edge and before an on edge. Digits will be generated after an on edge and before an off edge.

If the sequence number is one greater than the last received sequence number, the receiver appends the Current edge information to the previously received information.

If the sequence number is two greater than the last received sequence number, the receiver appends the Recent and Current edge information to the previously received information.

If the sequence number is three greater than the last received sequence number, the receiver appends the previous, recent and current edge information to the previously received information.

If the sequence number is more than three greater than the last received sequence number, the receiver appends the previous, recent and current edge information to the previously received information. It fills in the gap with the static values based on the previously received payload.

On a given sub-channel, if a voice payload is received at any time, an off edge should be appended to the previously received digits on/off edge information.

Annex B – Signalling Bit Transfer Syntax

B.1 Reference Documents

None

B.2 Transfer Structure

The Signalling Bit Transfer Syntax is comprised of payload formats and transfer procedures for alarm indications and channel associated signalling bits.

B.3 Payload Format

Payloads carrying signalling bits will be identified using the payload type field in the VoFR Header. The signalling bits will automatically be associated with the corresponding voice traffic based on the Channel ID field.

The first byte following the VoFR header contains a seven-bit sequence number with the most significant bit assigned as an Alarm Indicator Signal (AIS) bit. A value of 1 signifies an alarm condition.

Bits

	8	7	6	5	4	3	2	1	Octet
	AIS	Sequence Number							P
Previous	D[t-56ms]	C[t-56ms]	B[t-56ms]	A[t-56ms]	D[t-58ms]	C[t-58ms]	B[t-58ms]	A[t-58ms]	P+1
	D[t-52ms]	C[t-52ms]	B[t-52ms]	A[t-52ms]	D[t-54ms]	C[t-54ms]	B[t-54ms]	A[t-54ms]	P+2
	D[t-48ms]	C[t-48ms]	B[t-48ms]	A[t-48ms]	D[t-50ms]	C[t-50ms]	B[t-50ms]	A[t-50ms]	P+3
	D[t-44ms]	C[t-44ms]	B[t-44ms]	A[t-44ms]	D[t-46ms]	C[t-46ms]	B[t-46ms]	A[t-46ms]	P+4
	D[t-40ms]	C[t-40ms]	B[t-40ms]	A[t-40ms]	D[t-42ms]	C[t-42ms]	B[t-42ms]	A[t-42ms]	P+5
Recent	D[t-36ms]	C[t-36ms]	B[t-36ms]	A[t-36ms]	D[t-38ms]	C[t-38ms]	B[t-38ms]	A[t-38ms]	P+6
	D[t-32ms]	C[t-32ms]	B[t-32ms]	A[t-32ms]	D[t-34ms]	C[t-34ms]	B[t-34ms]	A[t-34ms]	P+7
	D[t-28ms]	C[t-28ms]	B[t-28ms]	A[t-28ms]	D[t-30ms]	C[t-30ms]	B[t-30ms]	A[t-30ms]	P+8
	D[t-24ms]	C[t-24ms]	B[t-24ms]	A[t-24ms]	D[t-26ms]	C[t-26ms]	B[t-26ms]	A[t-26ms]	P+9
	D[t-20ms]	C[t-20ms]	B[t-20ms]	A[t-20ms]	D[t-22ms]	C[t-22ms]	B[t-22ms]	A[t-22ms]	P+10
Current	D[t-16ms]	C[t-16ms]	B[t-16ms]	A[t-16ms]	D[t-18ms]	C[t-18ms]	B[t-18ms]	A[t-18ms]	P+11
	D[t-12ms]	C[t-12ms]	B[t-12ms]	A[t-12ms]	D[t-14ms]	C[t-14ms]	B[t-14ms]	A[t-14ms]	P+12
	D[t-8ms]	C[t-8ms]	B[t-8ms]	A[t-8ms]	D[t-10ms]	C[t-10ms]	B[t-10ms]	A[t-10ms]	P+13
	D[t-4ms]	C[t-4ms]	B[t-4ms]	A[t-4ms]	D[t-6ms]	C[t-6ms]	B[t-6ms]	A[t-6ms]	P+14
	D[t]	C[t]	B[t]	A[t]	D[t-2ms]	C[t-2ms]	B[t-2ms]	A[t-2ms]	P+15

Figure B - 1 Signalling Bit Transfer Syntax Payload Format

The sequence number starts at 0 and increments by 1 up through 127 and rolls over back to 0.

The transfer syntax for signalling bits contains 60 milliseconds worth of samples for up to four signalling bits. Each sample has a time resolution of 2.0 milliseconds. Each payload contains ten "new" samples for the current 20 millisecond time interval and a repetition of the ten samples for each of the two immediately preceding 20 millisecond time intervals.

This will result in 15 bytes of packed signalling bit values.

For sixteen state coding, all four bits are independent. For four state coding, the A and B bits are repeated in the C and D bit fields respectively. For two state coding, the A bit is repeated in the B, C, and D bit fields.

B.4 Procedures for Transmission of Payloads

While there are transitions occurring in the signalling bit values, the transmitter sends a signalling bit payload every 20 milliseconds. Since each payload covers 60 milliseconds of signal bit states, there is redundancy of signal bit information. The sequence number is incremented by one in each transmitted payload.

When the signal bit values have been static for 500 milliseconds, the transmitter switches frequency of transmission and sends a signal bit payload only once in every 5 seconds. During this time, the sequence number is not incremented.

When transitions start occurring again, the transmitter resumes incrementing the sequence numbers by one and sending payloads every 20 milliseconds.

The first such payload contains ten static previous and ten static recent values with ten new current samples. The second such payload contains ten static previous values with the ten previous values that were current in the first payload, and ten new current samples. This restarts the overlapping redundancy of information.

The transmitter may debounce the sequence of signalling bit values prior to transmission, but is not required to do so.

B.5 Procedures for Interpreting Received Payloads

When the receiver gets a signalling bit payload, it processes the bits based on the sequence number.

If the sequence number is one larger than the last received sequence number, the receiver appends the Current signal bits to the previously received values.

If the sequence number is two larger than the last received sequence number, the receiver appends the Recent and Current signal bits to the previously received values.

If the sequence number is three larger than the last received sequence number, the receiver appends the Previous, Recent, and Current signal bits to the previously received values.

If the sequence number is more than three larger than the last received sequence number, the receiver appends the Previous, Recent, and Current signal bits to the previously received values. It fills in the gap with static values based on the previously received payload.

If the sequence number is the same as the last received sequence number, the receiver takes the first value and uses it to set its current values for the signalling bits. (The signal bit values are static.)

The transmitter may or may not have debounced the signal bit values before transmission. If the receiving VoFR service user is interpreting the semantics of the signal bits, it should debounce the sequence of bit values received.

Annex C – Data Transfer Syntax

C.1 Reference Documents

[1] FRF.12 Frame Relay Forum Fragmentation Implemention Agreement, March 1997
[2] FRF3.1 Multiprotocol Encapsulation Implementation Agreement, June 1995
[3] RFC 1490 Multiprotocol Interconnect over Frame Relay, 1993

C.2 Data Transfer Structure

This annex describes a transfer syntax to support transport of data frames between two voice over frame relay service users. The contents of the frames are transparent to the voice over frame relay service. Typical applications include the transport of common channel signalling messages, RFC1490 packets [3], and FRF3.1 packets [2].

All data sub-frames contain the fragmentation header.

The payload type is set to primary payload type.

For more information on the fragmentation procedure refer to [1].

C.3 Data Payload Format

Figure C-1 shows the sub-frame payload format.

Bits

8	7	6	5	4	3	2	1	Octet
VoFR Sub-frame Header								1
B	E	0	Sequence number (upper 5 bits)					P
Sequence number (lower 8 bits)								P+1
Payload Fragment (variable length)								P+2 P+N

Figure C - 1 Data Transfer Syntax Payload Format

C.4 Data Procedures

The sub-frame payload will consist of a data frame received from a VoFR service user.

The frame is transmitted on the data link connection in one or more data fragments as defined in [1].

Upon receipt of a sub-frame containing the data transfer syntax, the fragments are re-combined using the procedures of [1], and the frame is delivered to the VoFR service user.

All frames received from the VoFR service user are conveyed without interpretation. Information transmitted using this transfer syntax is transparent to the VoFR service.

The maximum fragment size is governed by the maximum frame size supported by the Q.922 data link connection.

Annex D – Fax Relay Transfer Syntax

D.1 Reference Documents

[1] ITU T.4 Standardization of group 3 facsimile apparatus for document transmission, March 1993

[2] ITU T.30 Terminal Equipment and Protocol for Telematic Service / Procedure for Facsimile General Switch Networks, November 1994

[3] ITU V.17 A 2-wire modem for facsimile applications with rates up to 14,400bit/s, February 1991

[4] ITUV.21 300 bit/s duplex modem standardized for use in general switched telephone network Blue Book Fasc. VIII.1, October 1994

[5] ITU V.27 4800/2400 bit/s modem standardized for use in general switched telephone network Blue Book Fasc. VIII.1, November 1994

[6] ITU V.29 9600 bit/s modem standardized for use in point-to-point 4-wire leased telephone-type circuits Blue Book Fasc. VIII.1, November 1988

[7] ITU V.33 14400 bit/s modem standardized for use in point-to-point 4-wire leased telephone-type circuits Blue Book Fasc. VIII.1, November 1988

D.2 FAX Transfer Structure

The Fax Relay Transfer Syntax is comprised of the Fax Relay Payload Format and the Fax Relay Transfer Procedure. The fax relay transfer syntax provides transfer syntax for fax.

D.3 Fax Relay Payload Format

D.3.1 Modulation Turn-On Payload

The Modulation Turn-On Payload has the modulation types defined in the octet following the time stamp. If Modulation Type is single Frequency Tone, Frequency MS:LS bytes will specify the frequency. Frequency MS:LS bytes should be set to zero if the Modulation Type is not Single Frequency Tone.

Bits

8	7	6	5	4	3	2	1	Octet
EI1=1	Sequence Number				Relay Command=001			P
Time Stamp LS byte								P+1
EI2=0	Time Stamp MS byte							P+2
HDLC	reserved			Modulation Type				P+3
Frequency LS byte								P+4
Frequency MS byte								P+5

***LS=least significant, MS=most significant**

Figure D - 1 Modulation Turn-On Payload

D.3.2 Modulation Turn-Off Payload

The Modulation Turn-Off Payload has the following structure.

Bits

8	7	6	5	4	3	2	1	Octet
EI1=1	Sequence Number				Relay Command=000			P
Time Stamp LS byte								P+1
EI2=0	Time Stamp MS byte							P+2

Figure D - 2 Modulation Turn-Off Payload

D.3.3 T.30 Payload

The T.30 Payload has 3 bytes of demodulated and HDLC de-framed data.

Bits

8	7	6	5	4	3	2	1	Octet
EI1=0	Sequence Number				Relay Command = 010,011or 100			P
Data [I]								P+1
Data[I-1]								P+2
Data[I-2]								P+3

Figure D - 3 T.30 Payload

D.3.4 T.4 Payload

The T.4 Payload should be sent once every 40ms. The Relay Command = 010 (Data). Use of the Relay Command 011 and 100 for T.4 Payload is for further study.

Bits

8	7	6	5	4	3	2	1	Octet
EI1=0	Sequence Number				Relay Command = 010			P
Data[I]								P+1
Data[I-1]								P+2
•								•
Data[I-N-1]								P+N

Figure D - 4 T.4 Payload

Each payload should have the following number of raw demodulated data bytes according to the modulation rate:

Modulation Rate	Bytes per Payload (N)
14400	72
12000	60
9600	48
7200	36
4800	24
2400	12

Figure D - 5 Modulation Rates

D.3.5 Relay Command

The following is a list of code-points for Relay Commands.

Code	Relay Command
000	Modulation-Off
001	Modulation-On
010	Data
011	HDLC-End-Of-Frame
100	HDLC-Frame-Abort
101 - 111	reserved

Figure D - 6 Relay Commands

D.3.6 Modulation Type

The following is an encoding table for the Modulation Type. However, it is optional to support all the types.

Code	Modulation Type
0000	Single Freq Tone
0001	V.21 300bps
0010	V.27ter 2400bps
0011	V.27ter 4800bps
0100	V.29 7200bps
0101	V.29 9600bps
0110	V.33 12000bps
0111	V.33 14400bps
1000	V.17 7200bps
1001	V.17 9600bps
1010	V.17 12000bps
1011	V.17 14400bps
1100 - 1111	reserved

Figure D - 7 Modulation Type

D.3.7 HDLC

HDLC=1 indicates that HDLC de-framing is being applied. HDLC=0 indicates that HDLC de-framing is not used. De-framed packets are the content that remains after removing flags, extra bits inserted for transparency, and the frame check sequence.

D.3.8 Sequence Number

This sequence number is incremented for each new packet independent of the value of Relay Command.

The sequence number is reset at the beginning of each modulation type when the Modulation-On Relay Command is sent. It wraps around when it reaches a maximum count.

D.3.9 Time Stamp

The Time Stamp information represents the relative timing of events on the analog (or equivalent) input to the demodulator. The unit for Time Stamp is 1ms. The accuracy of Time Stamp should be within +/- 5ms.

The Time Stamp is mandatory in the packet header when the Modulation-On or Modulation-Off Relay Commands are sent. It is optional with any other Relay Command.

The Time Stamp clocks free-run on each end and there is no synchronization between them. The Time Stamp wraps around when it reaches the maximum count.

D.3.10 EI1 and EI2

These are header Extension Indicator bits.

EI1=1 indicates that the two Time Stamp bytes exist and immediately follow the first header byte. EI1=0 indicates that there is no Time Stamp bytes.

EI2 is reserved for future use and should be set to 0.

D.3.11 Frequency LS & MS Bytes

These are the least significant (LS) and most significant (MS) bytes for the Single Frequency Tone in unit of Hertz (Hz) within +/- 1.5%.

D.3.12 Data

Data is packed into the packet with the latest byte first and the oldest byte last. Within each byte, the MSB is the most recent bit and LSB is the oldest bit.

D.4 Fax Relay Transfer Procedures

D.4.1 Procedure for Transmission of T.30 Data

When the preamble is detected, at least three identical Modulation Turn-On payloads should be sent with Relay Command = 001 (Modulation-On), Sequence Number = 0, EI1=1, Modulation Type = V.21 and HDLC=1. The same Time Stamp should be use in all three payloads.

When the first byte of HDLC data is being demodulated and deframed, it should be sent with Relay Command = 010 (Data) and Sequence Number =1. Data [I] is the first byte. Data [I-1] and Data [I-2] should be set to all 1's.

When the second byte of HDLC data is being demodulated and deframed, it should be sent with Relay Command = 010 (Data) and Sequence Number =2. Data [I] is the second byte, Data [I-1] is the first byte and Data [I-2] is set to all 1's.

When the third byte of HDLC data is being demodulated and deframed, it should be sent with Relay Command = 010 (Data) and Sequence Number =3. Data [I] is the third byte, Data [I-1] is the second byte and Data [I-2] is the first byte.

Subsequently, a new payload should be sent after every byte being demodulated. The payload is sent with Relay Command = 010 (Data) and the Sequence Number incremented by 1. The most current byte should be immediately after the header, followed by the recent byte and then the previous byte.

At the end of each HDLC frame, if there is no CRC error, the last payload should be sent three times with Relay Command=011 (HDLC-End-Of-Frame). If a CRC error was detected by the sender, the last payload should be sent three times with Command = 100 (HDLC-Frame-Abort). In both cases, all three payloads should have the same three bytes of data as the previous data payload. All three payloads should have the sequence number (N_{EOF}). The previous data payload should have sequence number $N_{EOF} -1$.

The first data payload of the following HDLC frame should have sequence number $N_{EOF} +1$. The data bytes Data [I-1] and Data [I-2} in the first data payload of the following frame are the last two data bytes from the previous frame.

If the modulation turns off, three identical payloads should be sent with Command = 000 (Modulation-Off). All three payloads should have the same sequence number, which is one more than the last data payload, EI=1 and the same two bytes of Time Stamp.

D.4.2 Procedure for Transmission of T.4 Data

When modulation (non-single frequency tone) for the T.4 procedure is detected, at least three Modulation Turn-On payloads should be sent with Relay Command = 001 (Modulation-On), Sequence Number = 0, EI1=1, Modulation Type = the codepoint of the detected modulation type and HDLC=0. Use of HDLC=1 is for further study. The same Time Stamp should be use in all three payloads.

Subsequently, when data is available, a payload should be sent every 40ms with relay Command = 010 (Data) and the sequence number should be incremented by 1. Use of Relay Command 011 and 100 is reserved for further study.

When the modulation turns off, three identical payloads should be sent with Command = 000 (Modulation-Off) and EI1=1. These last three payloads should have the same sequence number and the same Time Stamp. The sequence number should be one larger than the last data.

D.4.3 Handling of Non-Standard Facilities (NSF) Frame

Procedures for disabling the NSF frame are for further study.

ANNEX E – CS-ACELP Transfer Syntax

E.1 Reference Document

[1]	ITU G.729/ ITU G.729 Annex A	Coding of Speech at 8 kbit/s using Conjugate Structure-Algebraic Code Excited Linear Predictive (CS-ACELP) Coding, March 1996

E.2 CS-ACELP Transfer Protocol

When the VoFR service user offers a frame of sampled speech it is immediately transmitted using the transfer structure described below.

E.3 CS-ACELP Transfer Structure

CS-ACELP produces 80 bits for each 10 ms frame of sampled speech. The list of the transmitted parameters used by the CS-ACELP algorithm is provided below. In order to allow the frame relay device to adjust its transmission rate, the CS-ACELP transfer syntax structure will permit multiples of 10 ms frames to be packed into the voice information field. An integer number of 10 ms frames will be packed into the voice information field to form a M*10 ms payload. For each M*10 ms of compressed speech, M*80 bits or M*10 octets will be produced. Support of M=2 is required. A range of 1 to 6 can optionally be supported.

Symbol	Description	Bits
LSP0	Switched predictor index of LSP quantizer	1
LSP1	First stage vector of LSP quantizer	7
LSP2	Second stage lower vector of LSP quantizer	5
LSP3	Second stage lower vector of LSP quantizer	5
P1	Pitch period (Delay)	8
P0	Parity check of pitch period	1
C1	Fixed Code-Book – 1st sub-frame	13
S1	Signs of pulses –1st sub-frame	4
GA1	Gain Code-Book (stage 1) – 1st sub-frame	3
GB1	Gain Code-Book (stage 2) – 1st sub-frame	4
P2	Pitch Period (Delay) - 2nd sub-frame	5
C2	Fixed Code-Book – 2nd sub-frame	13
S2	Signs of pulses –2nd sub-frame	4
GA2	Gain Code-Book (stage 1) – 2nd sub-frame	3
GB2	Gain Code-Book (stage 2) – 2nd sub-frame	4
Total	**Per 10 ms frame**	**80**

*LSP = Line Spectrum Pairs

Figure E – 1 List of Transmitted Parameters

Octet	MSB **Bit Packing** LSB
1	LSP0, LSP1[7...1]
2	LSP2[5..1], LSP3[5..3]
3	LSP3[2,1], P1[7..3]
4	P1[2,1], P0, C1[13..9]
5	C1[8..1]
6	S1[4..1], GA1[3...1], GB1[4]
7	GB1[3...1], P2[5..1]
8	C2[13...6]
9	C2[5...1], S2[4..2]
10	S2[1], GA2[3..1], GB2[4..1]

Figure E - 2 CS-ACELP Bit Packing Structure for Each Frame

7	6	5	4	3	2	1	0	
LSP0	LSP1[7]	LSP1[6]	LSP1[5]	LSP1[4]	LSP1[3]	LSP1[2]	LSP1[1]	Octet P
LSP2[5]	LSP2[4]	LSP2[3]	LSP2[2]	LSP2[1]	LSP3[5]	LSP3[4]	LSP3[3]	Octet P+1
LSP3[2]	LSP3[1]	P1[8]	P1[7]	P1[6]	P1[5]	P1[4]	P1[3]	Octet P+2
P1[2]	P1[1]	P0	C1[13]	C1[12]	C1[11]	C1[10]	C1[09]	
C1[08]	C1[07]	C1[06]	C1[05]	C1[04]	C1[03]	C1[02]	C1[01]	
S1[04]	S1[03]	S1[02]	S1[01]	GA1[03]	GA1[02]	GA1[01]	GB1[04]	
GB1[03]	GB1[02]	GB1[01]	P2[05]	P2[04]	P2[03]	P2[02]	P2[01]	Frame 1
C2[13]	C2[12]	C2[11]	C2[10]	C2[09]	C2[08]	C2[07]	C2[06]	
C2[05]	C2[04]	C2[03]	C2[02]	C2[01]	S2[04]	S2[03]	S2[02]	
S2[01]	GA2[03]	GA2[02]	GA2[01]	GB2[04]	GB2[03]	GB2[02]	GB2[01]	

:

:

LSP0	LSP1[7]	LSP1[6]	LSP1[5]	LSP1[4]	LSP1[3]	LSP1[2]	LSP1[1]	
LSP2[5]	LSP2[4]	LSP2[3]	LSP2[2]	LSP2[1]	LSP3[5]	LSP3[4]	LSP3[3]	
LSP3[2]	LSP3[1]	P1[8]	P1[7]	P1[6]	P1[5]	P1[4]	P1[3]	Frame M
P1[2]	P1[1]	P0	C1[13]	C1[12]	C1[11]	C1[10]	C1[09]	
C1[08]	C1[07]	C1[06]	C1[05]	C1[04]	C1[03]	C1[02]	C1[01]	
S1[04]	S1[03]	S1[02]	S1[01]	GA1[03]	GA1[02]	GA1[01]	GB1[04]	
GB1[03]	GB1[02]	GB1[01]	P2[05]	P2[04]	P2[03]	P2[02]	P2[01]	
C2[13]	C2[12]	C2[11]	C2[10]	C2[09]	C2[08]	C2[07]	C2[06]	Octet P+N-2
C2[05]	C2[04]	C2[03]	C2[02]	C2[01]	S2[04]	S2[03]	S2[02]	Octet P+N-1
S2[01]	GA2[03]	GA2[02]	GA2[01]	GB2[04]	GB2[03]	GB2[02]	GB2[01]	Octet P+N

Where:

P = First octet of payload
M = Number of 10 ms Frames
N = Number of octets in Voice Information Field = M*10

Figure E - 3 CS-ACELP Transfer Structure

E.4 Transfer Characteristics

Packetization Time: **M***10 ms

Algorithm Name	Reference Document	Compression Rate	Frame Size
CS-ACELP	ITU G.729	8 kbit/s	M*10

Figure E - 4 CS-ACELP Transfer Characteristics

E.5 Optional Sequence Number

Transmission of sequence numbers may be configured on a sub-channel basis. When enabled, the voice transfer syntax defined in Figure E - 3 is encapsulated in the Voice Transfer Structure field of the Active Voice Payload shown in Figure F - 2. The Sequence Number of Figure F – 2 shall be incremented every 10 msec. The Coding Type field of Figure F - 2 shall be set to 0000.

Annex F – Generic PCM/ADPCM Voice Transfer Syntax

F.1 Reference Documents

[1]	ITU G.711	Pulse Code Modulation of Voice Frequencies, 1988
[2]	ITU G.726	40, 32, 24, 16 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM), March 1996
[3]	ITU G.727	5-, 4-, 3-, and 2-bits Sample Embedded Adaptive Differential Pulse Code Modulation, November 1994
[4]	ITU G.764	Voice packetization - Packetized voice protocols, December 1990

F.2 Voice Transfer Structure

Encoded voice samples – G.711 (PCM), G.726 (ADPCM), or G.727 (EADPCM) – shall be inserted into the structure defined by Figure F - 1. The transfer of PCM/ADPCM/EADPCM is inspired by ITU-T Recommendation G.764. The following sections define two payload types and the Voice Transfer Structure.

Bit number	8	7		1	
MSB block	MSB/S8	MSB/S7	. . .	MSB/S1	P+ 5
			. .		
	MSB/S40	MSB/S39	. . .	MSB/S33	
MSB-1 block	(MSB-1)/S8	(MSB-1)/S7	. . .	(MSB-1)/S1	
			. .		
	(MSB-1)/S40	(MSB-1)/S39	. . .	(MSB-1)/S33	
. . .			. . .		
LSB block	LSB/S8	LSB/S7	. . .	LSB/S1	
			. . .		
	LSB/S40	LSB/S39	. . .	LSB/S33	Octet N

Figure F - 1 PCM/ADPCM/EADPCM Voice Transfer Structure
(showing case of M = 1)

The voice transfer structure contains blocks arranged according to the significance of the bits. The first block contains the MSBs of all the encoded samples; the second contains the second MSBs and so on. Within a block, the bits are ordered according to their sample number. Since the 5 ms encoding interval corresponds to 40 samples, each block contains 5 octets.

A particular feature of this structure is that non-critical (enhancement) information is placed in locations where it can easily be discarded, without impacting the critical (core) information. For example, if 32 kbit/s EADPCM (G.727 (4,2)) is used, then there will be four blocks corresponding to four bits of varying significance (msb, msb-1, msb-2, lsb). The least significant blocks (msb-2, lsb) are the enhancement blocks and may be discarded under congestion conditions.

Annex G describes a related way of placing the critical and non-critical information into separate frames, so that the enhancement blocks can be marked with Discard Eligibility.

The size of the voice transfer structure depends on the packing factor M and the coding type, as shown in Figure F - 4. The packing factor is a multiple from one to 12. The value of M is configured identically at transmitter and receiver. It is typically, but not necessarily, the same in both directions. Equipment complying with this transfer syntax shall be configurable to support the default value M = 1.

When M is greater than 1, the voice transfer structure contains a first set of blocks, ordered from MSB to LSB, followed by a second set of blocks, ordered likewise, and so on up to the Mth set of blocks.

F.3 Active Voice Payload

When the Payload Type is Primary Payload, other fields in the sub-frame are as shown. The voice transfer structure containing encoded voice samples is defined in section F.2.

Bits

8	7	6	5	4	3	2	1	Octet
Sequence Number				Coding Type				P
Voice Transfer Structure								P+1

Figure F - 2 PCM/ADPCM/EADPCM Steady State Payload

F.3.1 Coding Type

The coding type field indicates the method of encoding PCM/ADPCM/EADPCM voice samples into the voice transfer structure.

The transmitting end-system shall only encode using algorithms for which there is decode support at the receiving end-system. The algorithms supported by the receiver are known by mutual configuration.

Values of the Coding Type field are defined in Figure F - 4.

F.3.2 Sequence number

The sequence number is used to maintain temporal integrity of voice played out by the receiving end-system. For PCM/ADPCM/EADPCM, the underlying encoding interval is 5 ms. Voice samples are processed with this periodicity and the sequence number is incremented by 1. After a count of 15 is reached the sequence number rolls back to 0.

The sequence number is incremented every 5 milliseconds, even when there is no active voice to be sent. This would be the case during a silence insertion period, if voice activity detection were operational.

The peer end-system expects to receive voice samples in sequence and within a certain time period. If voice activity detection is operational and no active voice is received, the peer end-system will continue to increment its expected sequence number every 5 ms.

When multiple voice samples are received in a single subframe (M > 1), the next expected sequence number is incremented by M.

F.4 Silence Insertion Descriptor (SID) Payload

When the Payload Type is Primary Payload with Silence Insertion, other fields in the sub-frame are as shown in Figure F - 3.

Bits

8	7	6	5	4	3	2	1	Octet
Sequence Number				Reserved				P
Reserved		Noise Level						P+1

Figure F - 3 PCM/ADPCM/EADPCM Silence Insertion Descriptor (SID) Payload

F.4.1 Reserved

This field is set to 000000 by the transmitter and is ignored at the receiver.

F.4.2 Sequence number

This field is the same as defined in F.3.2.

F.4.3 Noise Level

The background noise level is expressed in -dBm0. The receiver can use this field to play out an appropriate level of background noise in the absence of active voice.

Additional sub-frames of this type may be sent if the noise level changes or may be sent redundantly to increase the probability of being received.

This payload type should not be sent if voice activity detection is not operational.

F.5 Transfer Characteristics

Encoding interval: 5 ms

Packing factor: M = 1 to 12

Support of M=4 is required. A range of 1 to 12 can optionally be supported.

Coding Type	Algorithm Name	Reference Document	Compression Bit Rate (kbit/s)	Voice Transfer Structure (Octets)
0000	PCM A-law	ITU G.711	64	40*M
0001	“	“	56	35*M
0010	“	“	48	30*M
0011	PCM u-law	“	64	40*M
0100	“	“	56	35*M
0101	“	“	48	30*M
0110	ADPCM	ITU G.726	40	25*M
0111	“	“	32	20*M
1000	“	“	24	15*M
1001	“	“	16	10*M
1010	EADPCM (5,2)	ITU G.727	40	25*M
1011	(4,2)	“	32	20*M
1100	(3,2)	“	24	15*M
1101	(2,2)	“	16	10*M

Figure F - 4 PCM/ADPCM/EADPCM Transfer Characteristics

Annex G – G.727 DISCARD-ELIGIBLE EADPCM VOICE Transfer Syntax

G.1 Reference Documents

[1]	ITU G.727	5-, 4-, 3-, and 2-bits Sample Embedded Adaptive Differential Pulse Code Modulation, November 1994

G.2 Voice Transfer Structure

The voice transfer structure is the same as defined in Annex F.

G.3 Active Voice Payload

The G.727 EADPCM compression algorithm outputs core and enhancement information. This information is separately assembled into blocks. Core information is inserted into frames with low discard eligibility (DE=0), and enhancement information inserted into frames with high discard eligibility (DE=1).

Core and enhancement information, if required by a particular traffic type, may be combined within a single frame with DE = 0.

When the Payload Type is Primary Payload, other fields in the sub-frame are as shown in Figure G - 1. The voice transfer structure containing encoded voice samples is defined in Annex F.

Figure G - 1 shows only two sub-frames, one each for core and enhancement information, but transmitters are explicitly allowed to use the VoFR header to pack multiple sub-frames of the same kind of information into each frame, with DE = 0 or 1, correspondingly.

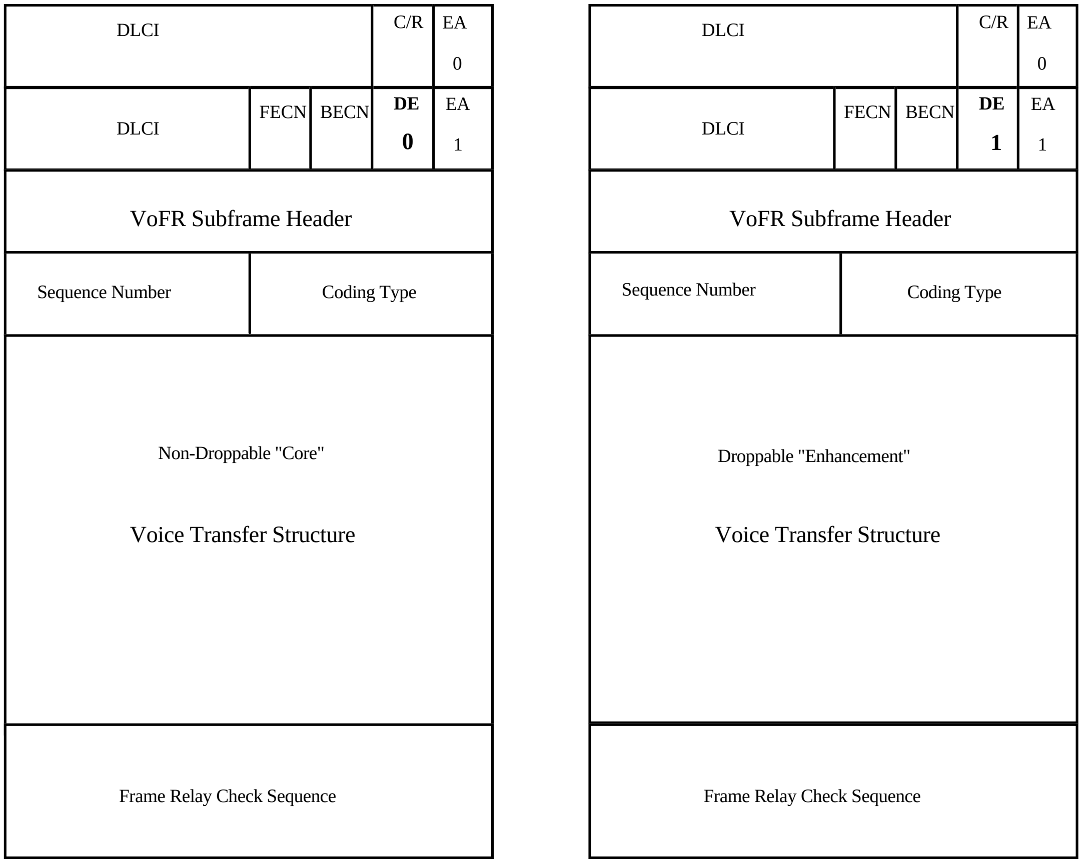

Figure G - 1 Discard-Eligible EADPCM Steady State Payload (showing single sub-frames)

G.3.1 Coding Type

The coding type field indicates the method of encoding EADPCM voice samples into the voice transfer structure.

The transmitting end system shall only encode using algorithms for which there is decode support at the receiving end system. The algorithms supported by the receiver are known by mutual configuration.

Values of the Coding Type field are defined in Figure G - 2.

G.3.2 Sequence number

This field is the same as defined in Annex F.

G.4 Silence Insertion Descriptor (SID) Payload

This payload is the same as defined in Annex F.

G.5 Transfer Characteristics

Encoding interval: 5 ms

Packing factor: M = 1 to 12

Coding Type	Algorithm Name	Type of Information	Compression Bit Rate (kbit/s)	Voice Transfer Structure (Octets)
0000	EADPCM (2,2)	Core	16	10*M
0001	(3,2)	Enhancement	8	5*M
0010	(4,2)	“	16	10*M
0011	(5,2)	“	24	15*M
0100	(3,2)	Combined	24	15*M
0101	(4,2)	“	32	20*M
0110	(5,2)	“	40	25*M
0111	EADPCM (3,3)	Core	24	15*M
1000	(4,3)	Enhancement	8	5*M
1001	(5,3)	“	16	10*M
1010	(4,3)	Combined	32	20*M
1011	(5,3)	“	40	25*M
1100	EADPCM (4,4)	Core	32	20*M
1101	(5,4)	Enhancement	8	5*M
1110	(5,4)	Combined	40	25*M

Figure G - 2 Discard-Eligible EADPCM Transfer Characteristics

ANNEX H – G.728 LD-CELP Transfer syntax

H.1 Reference Documents

[1] ITU G.728 Coding of Speech At 16 kbit/s Using Low-Delay Code Excited Linear Prediction, November 1994

H.2 Voice Transfer Structure

Voice samples that are compressed using 16 kbit/s LD-CELP (G.728) will be inserted into the voice transfer structure defined in Figure H - 1. The LD-CELP compression algorithm produces a 10-bit code-word vector for every 5 samples of input speech from an 8000 sample/sec. stream. The 10 bits are reformatted to fit within the octet structure of the Voice Transfer Structure. Every group of five octets contains four, 10-bit code-words resulting in a 2.5 ms duration sub-frame. Two of these 2.5 ms groups are combined into a 5 ms block for transmission. The MSB of the first 10-bit code-word is aligned with the MSB of the first octet in the block. Subsequent bits of the code-word are placed in descending bit locations of the first octet with the other bits of subsequent code-words being bit packed into the remaining octets. Each block consists of eight 10-bit code-words which are mapped into 10 octets.

H.3 Transfer Protocol

.

9 … 3	2 … 0
MSB 7-bit shape vector LSB	MSB 3-bit gain vector LSB

(MSB) 7-bits of shape vector[0] (LSB)	(MSB) 1-bit of gain[0]		P
2-bits of gain[0] (LSB)	(MSB) 6-bits of shape vector[1]		P+1
1-bit of shape vector[1] (LSB)	(MSB) 3-bits of gain[1] (LSB)	(MSB) 4-bits of shape vector[2]	P+2
3-bits of shape vector[2] (LSB)	(MSB) 3-bits of gain[2] (LSB)	(MSB) 2-bits of shape vector[3]	P+3
5-bits of shape vector[3] (LSB)	(MSB) 3-bits of gain[3] (LSB)		P+4
(MSB) 7-bits of shape vector[4] (LSB)	(MSB) 1-bit of gain[4]		P+5
2-bits of gain[4] (LSB)	(MSB) 6-bits of shape vector[5]		P+6
1-bit of shape vector[5] (LSB)	(MSB) 3-bits of gain[5] (LSB)	(MSB) 4-bits of shape vector[6]	P+7
3-bits of shape vector[6] (LSB)	(MSB) 3-bits of gain[6] (LSB)	(MSB) 2-bits of shape vector[7]	P+8
5-bits of shape vector[7] (LSB)	(MSB) 3-bits of gain[7] (LSB)		P+9

Where: P = VoFR subframe header and optional transfer protocol octets

Figure H - 1 LD-CELP Voice Transfer Structure
(showing case of M = 1)

The size of the voice transfer structure depends on the packing factor M. The packing factor is a multiple from 1 to 12. The value of M is configured identically at transmitter and receiver. It is typically, but not necessarily, the same in both directions. Equipment complying with this transfer syntax shall be configurable to support the value M = 1 to 12.

When M is greater than 1, the voice transfer structure contains multiple blocks, starting with the first encoded voice sample and ending with the last encoded voice sample.

H.4 Transfer Characteristics

Encoding interval: 5 ms

Packing factor: M = 1 to 12

Other Capabilities:

In-Band Tone Handling - Can pass 2400 baud Modem Signals & DTMF

Algorithm Name	Reference Document	Compression Bit Rate	Voice Transfer Structure
LD-CELP	ITU G.728	16 kbit/s	10*M octets

Figure H - 2 LD-CELP Transfer Characteristics

H.5 Optional Sequence Number

Transmission of sequence numbers may be configured on a sub-channel basis. When enabled, the voice transfer syntax defined in Figure H - 1 is encapsulated in the Voice Transfer Structure field of the Active Voice Payload shown in Figure F - 2. The Sequence Number of Figure F – 2 shall be incremented every 10 msec. The Coding Type field of Figure F - 2 shall be set to 0000.

Annex I – G.723.1 MP-MLQ Dual Rate Speech Coder

I.1 Reference Document

[1] ITU G.723.1 Dual Rate Speech Coder for Multimedia Communications Transmitting at 5.3 & 6.3 kbit/s, March 1996

I.2 Transfer Structure

Voice samples that are compressed using the 6.3kbit/s MP-MLQ algorithm (G.723.1 high rate) and 5.3kbit/s ACELP algorithm (G.723.1 low rate) yield a frame of packed parameters for every 240 samples of input speech from a 8000 sample/sec stream. Some of these parameters are based on an analysis of the entire frame; others are based on the analyses of each of the four component 60 sample sub-frames. Figure I - 1 shows of list of transmitted parameters for both MP-MLQ and ACELP.

For MP-MLQ, the resulting 191-bit frame is formatted to fit within the 24 octet structure of the Voice Information Field (one bit is unused) as defined in Figure I - 2. For ACELP, the resulting 160-bit frame is formatted to fit within the 20 octet structure of the Voice Information Field as defined in Figure I - 1. In Figure I - 2 and Figure I - 3, each bit of transmitted parameters is named PAR (x)_By: where PAR is the name of the parameter and x indicates the G.721 sub-frame index if relevant and y stands for the bit position starting from 0 (lsb) to the msb.

The expression PARx_ByPARx_Bz stands for the range of transmitted bits from bit *y* to bit *z*. The unused bit is named UB (value=0). RATEFLAG_B0 tells whether the high rate (0) or the low rate (1) is used for the current frame. VADFLAG_B0 tells whether the current frame is active speech (0) or non-speech (1). The combination of RATEFLAG and VADFLAG both being set to 1 is reserved for future use. Octets are transmitted in the order in which they are listed in Figure I - 2 and Figure I - 3. Within each octet shown, the bits are ordered with the most significant bit on the left.

Name	Transmitted parameters	high rate	low rate # bits
LPC	LSP VQ index	24	24
ACL0	Adaptive Code-Book Lag	7	7
ACL1	Differential Adaptive Code-Book Lag	2	2
ACL2	Adaptive Code-Book Lag	7	7
ACL3	Differential Adaptive Code-Book Lag	2	2
GAIN0	Combination of adaptive and fixed gains	12	12
GAIN1	Combination of adaptive and fixed gains	12	12
GAIN2	Combination of adaptive and fixed gains	12	12
GAIN3	Combination of adaptive and fixed gains	12	12
POS0	Pulse positions index	20*	12
POS1	Pulse positions index	18*	12
POS2	Pulse positions index	20*	12
POS3	Pulse positions index	18*	12
PSIG0	Pulse sign index	6	4
PSIG1	Pulse sign index	5	4
PSIG2	Pulse sign index	6	4
PSIG3	Pulse sign index	5	4
GRID0	Grid index	1	1
GRID1	Grid index	1	1
GRID2	Grid index	1	1
GRID3	Grid index	1	1

Figure I - 1 List of Transmitted Parameters

*Note: The 4 msb of these code-words are combined to form a 13 bit index, msb Position

TRANSMITTED	PARx By, ...
1	LPC B5...LPC B0, VADFLAG B0, RATEFLAG B0
2	LPC B13...LPC B6
3	LPC B21...LPC B14
4	ACL0 B5...ACL0 B0, LPC B23, LPC B22
5	ACL2 B4...ACL2 B0, ACL1 B1, ACL1 B0, ACL0 B6
6	GAIN0 B3...GAIN0 B0, ACL3 B1, ACL3 B0, ACL2 B6, ACL2 B5
7	GAIN0 B11...GAIN0 B4
8	GAIN1 B7...GAIN1 B0
9	GAIN2 B3...GAIN2 B0, GAIN1 B11...GAIN1 B8
10	GAIN2 B11...GAIN2 B4
11	GAIN3 B7...GAIN3 B0
12	GRID3 B0, GRID2 B0, GRID1 B0, GRID0 B0, GAIN3 B11...GAIN3 B8
13	MSBPOS B6...MSBPOS B0, UB
14	POS0 B1, POS0 B0, MSBPOS B12...MSBPOS B7
15	POS0 B9...POS0 B2
16	POS1 B2, POS1 B0, POS0 B15...POS0 B10
17	POS1 B10...POS1 B3
18	POS2 B3...POS2 B0, POS1 B13...POS1 B11
19	POS2 B11...POS2 B4
20	POS3 B3...POS3 B0, POS2 B15...POS2 B12
21	POS3 B11...POS3 B4
22	PSIG0 B5...PSIG0 B0, POS3 B13, POS3 B12
23	PSIG2 B2..PSIG2 B0, PSIG1 B4...PSIG1 B0
24	PSIG3 B4...PSIG3 B0, PSIG2 B5...PSIG2 B3

Figure I - 2 Octet Packing for the 6.3 kbps MP-MLQ codec

TRANSMITTED OCTETS	PARx_By, ...
1	LPC_B5...LPC_B0, VADFLAG_B0, RATEFLAG_B0
2	LPC_B13...LPC_B6
3	LPC_B21...LPC_B14
4	ACL0_B5...ACL0_B0, LPC_B23, LPC_B22
5	ACL2_B4...ACL2_B0, ACL1_B1, ACL1_B0, ACL0_B6
6	GAIN0_B3...GAIN0_B0, ACL3_B1, ACL3_B0, ACL2_B6, ACL2_B5
7	GAIN0_B11...GAIN0_B4
8	GAIN1_B7...GAIN1_B0
9	GAIN2_B3...GAIN2_B0, GAIN1_B11...GAIN1_B8
10	GAIN2_B11...GAIN2_B4
11	GAIN3_B7...GAIN3_B0
12	GRID3_B0, GRID2_B0, GRID1_B0, GRID0_B0, GAIN3_B11...GAIN3_B8
13	POS0_B7...POS0_B0
14	POS1_B3...POS1_B0, POS0_B11...POS0_B8
15	POS1_B11...POS1_B4
16	POS2_B7...POS2_B0
17	POS3_B3...POS3_B0, POS2_B11...POS2_B8
18	POS3_B11...POS3_B4
19	PSIG1_B3...PSIG1_B0, PSIG0_B3...PSIG0_B0
20	PSIG3_B3...PSIG3_B0, PSIG2_B3...PSIG2_B0

Figure I - 3 Octet Packing for the 5.3 kbps ACELP codec

I.3 Transfer Protocol

When the VoFR service user offers a frame of sampled speech it is immediately transmitted using the transfer structure described above in Section I.2.

I.4 Transfer Characteristics

Packetization Time: 30 ms

Other Capabilities:

In-Band Tone Handling - Can pass DTMF

Algorithm Name	Reference Document	Compression Rate	Frame Size
MP-MLQ	ITU G.723.1	6.3 kbit/s	24 octets
ACELP	ITU G.723.1	5.3 kbit/s	20 octets

Figure I - 4 MP-MLQ and ACELP Transfer Characteristics

I.5 Optional Sequence Number

Transmission of sequence numbers may be configured on a sub-channel basis. When enabled, the voice transfer syntax defined in Section I.2 is encapsulated in the Voice Transfer Structure field of the Active

Voice Payload shown in Figure F - 2. The Sequence Number of Figure F – 2 shall be incremented every 10 msec. The Coding Type field of Figure F - 2 shall be set to 0000.