INTERNATIONAL TELECOMMUNICATION UNION

ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

V.150.1

(01/2003)

SERIES V: DATA COMMUNICATION OVER THE TELEPHONE NETWORK

Interworking with other networks

Modem-over-IP networks: Procedures for the end-to-end connection of V-series DCEs

ITU-T Recommendation V.150.1

ITU-T V-SERIES RECOMMENDATIONS

DATA COMMUNICATION OVER THE TELEPHONE NETWORK

General	V.1–V.9
Interfaces and voiceband modems	V.10–V.34
Wideband modems	V.35–V.39
Error control	V.40–V.49
Transmission quality and maintenance	V.50–V.59
Simultaneous transmission of data and other signals	V.60–V.99
Interworking with other networks	**V.100–V.199**
Interface layer specifications for data communication	V.200–V.249
Control procedures	V.250–V.299
Modems on digital circuits	V.300–V.399

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

ITU-T Recommendation V.150.1

Modem-over-IP networks: Procedures for the end-to-end connection of V-series DCEs

Summary

This Recommendation defines the inter-operation of two PSTN to IP network gateways that facilitate the end-to-end connection of V-series DCEs over an IP network. The principal characteristics of these gateways are: a mechanism to allow the transparent transport of modem signals end-to-end, a mechanism to allow the termination of modem signals at the gateways and the transport of the data between gateways, the definition of a transport protocol, which is suitable to relay data between gateways, and procedures to allow gateways to transition between Voice-over-Internet Protocol and Modem-over-Internet Protocol operation.

Source

ITU-T Recommendation V.150.1 was approved on 13 January 2003 by ITU-T Study Group 16 (2001-2004) under the ITU-T Recommendation A.8 procedure. This edition includes the modifications introduced by Corrigendum 1 approved on July 2003 and by Corrigendum 2 approved on March 2004.

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.

Compliance with this Recommendation is voluntary. However, the Recommendation may contain certain mandatory provisions (to ensure e.g., interoperability or applicability) and compliance with the Recommendation is achieved when all of these mandatory provisions are met. The words "shall" or some other obligatory language such as "must" and the negative equivalents are used to express requirements. The use of such words does not suggest that compliance with the Recommendation is required of any party.

INTELLECTUAL PROPERTY RIGHTS

ITU draws attention to the possibility that the practice or implementation of this Recommendation may involve the use of a claimed Intellectual Property Right. ITU takes no position concerning the evidence, validity or applicability of claimed Intellectual Property Rights, whether asserted by ITU members or others outside of the Recommendation development process.

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.

CONTENTS

Page

Page

ITU-T Recommendation V.150.1

Modem-over-IP networks: Procedures for the end-to-end connection of V-series DCEs

1 Scope

This Recommendation specifies the operation between two IP network gateways to facilitate the end-to-end connection of V-series DCEs over an IP network. The gateways are specified herein in terms of their functionality, signals and messages and operating procedures. The principal characteristics of these gateways are as follows:

a) Support of a mechanism to allow the transparent transport of modem signals end-to-end.

b) Support of mechanisms to allow the termination of modem signals at the gateways and the transport of the data between gateways.

c) The definition of a transport protocol, which can be used to relay the data between gateways.

d) The definition of procedures to allow gateways to transition between Voice-over-Internet Protocol and Modem-over-Internet Protocol operation.

This Recommendation includes mandatory requirements, recommendations and options; these are designated by the words "shall," "should," and "may" respectively.

1.1 Recommendation version

For the purposes of forward and backward compatibility, this Recommendation is assigned a version number.

The usage and definition of this version number is defined in ITU-T Rec. V.150.0.

Version: 1

NOTE – The reader is encouraged to check on the ITU-T website for any normative or informative amendments to this Recommendation.

2 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; 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. The reference to a document within this Recommendation does not give it, as a stand-alone document, the status of a Recommendation.

– ITU-T Recommendation G.711 (1988), *Pulse Code Modulation (PCM) of voice frequencies*.

– ITU-T Recommendation H.245 (2003), *Control protocol for multimedia communication*.

– ITU-T Recommendation H.248 (2000), *Gateway control protocol*.

– ITU-T Recommendation H.323, Annex P (2003), *Transfer of modem signals over H.323*.

– ITU-T Recommendation T.38 (2002), *Procedures for real-time Group 3 facsimile communication over IP networks*.

– ITU-T Recommendation V.8 (2000), *Procedures for starting sessions of data transmission over the public switched telephone network*.

– ITU-T Recommendation V.8 *bis* (2000), *Procedures for the identification and selection of common modes of operation between data circuit-terminating equipments (DCEs) and between data terminal equipments (DTEs) over the public switched telephone network and on leased point-to-point telephone-type circuits*.

– ITU-T Recommendation V.14 (1993), *Transmission of start-stop characters over synchronous bearer channels*.

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

– ITU-T Recommendation V.18 (2000), *Operational and interworking requirements for DCEs operating in the text telephone mode*.

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

– ITU-T Recommendation V.22 *bis* (1988), *2400 bits per second duplex modem using the frequency division technique standardized for use on the general switched telephone network and on point-to-point 2-wire leased telephone-type circuits*.

– ITU-T Recommendation V.23 (1988), *600/1200-baud modem standardized for use in the general switched telephone network*.

– ITU-T Recommendation V.24 (2000), *List of definitions for interchange circuits between data terminal equipment (DTE) and data circuit-terminating equipment (DCE)*.

– ITU-T Recommendation V.25 (1996), *Automatic answering equipment and general procedures for automatic calling equipment on the general switched telephone network including procedures for disabling of echo control devices for both manually and automatically established calls*.

– ITU-T Recommendation V.26 *bis* (1988), *2400/1200 bits per second modem standardized for use in the general switched telephone network*.

– ITU-T Recommendation V.26 *ter* (1988), *2400 bits per second duplex modem using the echo cancellation technique standardized for use in the general switched telephone network and on point-to-point 2-wire leased telephone-type circuits*.

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

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

– ITU-T Recommendation V.32 *bis* (1991), *A duplex modem operating at data signalling rates of up to 14 400 bit/s for use on the general switched telephone network and on leased point-to-point 2-wire telephone-type circuits*.

– ITU-T Recommendation V.34 (1998), *A modem operating at data signalling rates of up to 33 600 bit/s for use on the general switched telephone network and on leased point-to-point 2-wire telephone-type circuits*.

– ITU-T Recommendation V.42 (2002), *Error-correcting procedures for DCEs using asynchronous-to-synchronous conversion*.

– ITU-T Recommendation V.42 *bis* (1990), *Data compression procedures for data circuit-terminating equipment (DCE) using error correction procedures*.

– ITU-T Recommendation V.44 (2000), *Data compression procedures*.

– ITU-T Recommendation V.90 (1998), *A digital modem and analogue modem pair for use on a Public Switched Telephone Network (PSTN) at data signalling rates of up to 56 000 bit/s downstream and up to 33 600 bit/s upstream*.

– ITU-T Recommendation V.91 (1999), *A digital modem operating at data signalling rates of up to 64 000 bit/s for use on a 4-wire circuit switched connection and on leased point-to-point 4-wire digital circuits*.

– ITU-T Recommendation V.92 (2000), *Enhancements to Recommendation V.90*.

– ITU-T Recommendation X.680 (2002) | ISO/IEC 8824-1:2002, *Information technology – Abstract Syntax Notation One (ASN.1): Specification of basic notation*.

– ITU-T Recommendation X.691 (2002) | ISO/IEC 8825-2:2002, *Information technology – ASN.1 encoding rules: Specification of Packet Encoding Rules (PER)*.

– ISO/IEC 3309:1993, *Information Technology – Telecommunications and information exchange between systems – High-level data link control (HDLC) procedures – Frame structure*.

– ISO/IEC 4335:1993, *Information Technology – Telecommunications and information exchange between systems – High-level data link control (HDLC) procedures – Elements of procedures*.

– IETF RFC 768 (1980), *User Datagram Protocol*.

– IETF RFC 791 (1981), *Internet Protocol*.

– IETF RFC 2198 (1997), *RTP Payload for Redundant Audio Data*.

– IETF RFC 2327 (1998), *SDP: Session Description Protocol*.

– IETF RFC 2733 (1999), *An RTP Payload format for Generic Forward Error Correction*.

– IETF RFC 2833 (2000), *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*.

– IETF RFC 3407 (2000), *Session Description Protocol (SDP) Simple Capability Declaration*.

3 Definitions and abbreviations

3.1 Definitions

This Recommendation defines the following terms:

3.1.1 call discrimination time-out timer: An optional timer that is started on receipt of a modem signal (such as an answer tone) and stopped on the basis of criteria in the Call Discrimination SDL diagrams in this Recommendation.

3.1.2 control channel: Is the channel provided by the IP-Transport Layer Protocol that is used for the sending of IP-Transport Layer Protocol control messages.

3.1.3 double-trans-compression gateway: Is a gateway that has the resources to support Trans-Compression in both directions of the data flow.

3.1.4 effective data signalling rate: Is the data-signalling rate in bits/s with all format and frame information removed from the data stream (e.g., Start, Stop, Parity bits, protocol headers, CRC, HDLC Flags, pre-ambles, post-ambles, etc.). The effective data signalling rate is always less than or equal to the data signalling rate of the input data stream.

3.1.5 gateway: A gateway converts media provided in one type of network to the format required in another type of network. For example, a gateway could terminate bearer channels from a switched circuit network (e.g., DS0s) and media streams from a packet network (e.g., RTP streams in an IP network).

3.1.6 modem relay: The transport of modem data across a packet network using modem termination at the gateways.

3.1.7 MoIP gateway: A gateway that is compliant with this Recommendation.

3.1.8 no-trans-compression gateway: Is a gateway whose data input in both directions is transferred transparently.

3.1.9 off-ramp gateway: The IP network access point that calls an Answering DCE. (Abbreviated to G2.)

3.1.10 on-ramp gateway: The access point that is called by an originating DCE that interfaces to the IP network. (Abbreviated to G1.)

3.1.11 repeated-break: A break with the same sequence number as a previously received Break.

3.1.12 selective destructive: A transport protocol is "selectively destructive" if the local protocol user can trigger the discard of data that has been given to the transmitter by the local protocol user and not delivered to the peer protocol user by the receiver.

3.1.13 single-trans-compression gateway: Is a gateway that has the ability to perform a Trans-Compression in only one direction of the data flow (serial). For the other direction, the data is passed through transparently with minimum of delay to the outputs.

3.1.14 trans-coding: Translation from one companding law to another (i.e., G.711 A-law to μ-law or vice versa).

3.1.15 trans-compression: A Trans-Compression function consists of a decompression element whose output is connected to the input of a compression element. The input to the Trans-Compression is the input to the decompression element and the output of the Trans-Compression function is the output of the compression element. (See clause 13.)

3.1.16 voiceband data: The transport of modem signals over a voice channel of a packet network with the encoding appropriate for modem signals.

3.1.17 XID/LR: In V.42 LAPM, the XID command or response. In Annex A/V.42, the LR command or response.

The definition of DCE line signals used in this Recommendation is described in Appendix V.

3.2 Abbreviations

This Recommendation uses the following abbreviations:

ANS	V.25 Answer Tone
ASN.1	Abstract Syntax Notation One
ASNam	V.8 Answer Tone
CM	V.8 Call Menu
DE	Destructive and Expedited
DLCI	Data Link Connection Identifier
DS0	Digital Signal level 0
D-TCX	Double Trans-Compression

Cx	Compression Function
Cx'	Disabled Compression Function
DCE	Data Circuit-terminating Equipment (Modem)
DTE	Data Terminal Equipment
Dx	Decompression Function
Dx'	Disabled Decompression Function
FEC	Forward Error Correction
FoIP	Fax over Internet Protocol
G1	On-Ramp Gateway
G2	Off-Ramp Gateway
HDLC	High-level Data Link Control
IP	Internet Protocol
IP-TLP	IP Transport Layer Protocol
JM	V.8 Joint Menu
LAPM	Link Access Protocol for Modems (an error control protocol defined in ITU-T Rec. V.42)
LR	Link Request (Frame defined in Annex A/V.42 (1996))
LRc	Link Request command
LRr	Link Request response
M1	Originating end-point DCE
M2	Answering end-point DCE
MIPS	Millions Instructions Per Second
MNP5	A compression method as defined in Appendix VI.4
MoIP	Modem over Internet Protocol
MR	Modem Relay
MR1	Modem Relay Connection Scenario 1
MR2	Modem Relay Connection Scenario 2
MR3	Modem Relay Connection Scenario 3
MR4	Modem Relay Connection Scenario 4
NDE	Non-Destructive and Expedited
NDNE	Non-Destructive and Non-Expedited
N-TCX	No Trans-Compression
PCM	Pulse Code Modulation
PDU	Protocol Data Unit
PHY	Physical transport Layer of modem connection
PSTN	Public Switched Telephone Network
QoS	Quality of Service

RC	Reason Code
RIC	Reason Identifier Code
RNR	Receiver Not Ready
RR	Receiver Ready
RSC	Reliable Sequenced Channel
RTP	Real-time Transfer Protocol
SPRT	Simple Packet Relay Transport
SPU	Signal Processing Unit
SSE	State Signalling Events
SSRC	Synchronization SouRCe
S-TCX	Single Trans-Compression
TCX	Trans-Compression
UDP	User Datagram Protocol
U-MR	Universal-Modem Relay
USC	Unreliable Sequenced Channel
VBD	Voice Band Data mode
V-MR	V.8-Modem Relay
VoIP	Voice over Internet Protocol
XID	Exchange Identification
XID_C	LAPM XID command
XID_{def}	XID corresponding to the default compression parameters for N-TCX
XID_r	LAPM XID response

4 Conventions

The following convention is used to indicate the elements of PDU defined in this Recommendation.

For the IP-TLP, a message is indicated by its mnemonic, e.g., INIT, CONNECT etc. An element of the message is indicated by mnemonic in parentheses, e.g., MR_EVENT(PHYSUP), CONNECT(NCP).

A SSE message is indicated using SSE:<X>(<code>), where <X> is one of the defined media states and <code> is the applicable reason code.

5 Introduction

Modem over Internet Protocol (MoIP) is the application of V-series DCEs over a Voice-over-Internet Protocol (VoIP) connection. There are two basic models suitable to this application. The first is as shown in Figure 1. This is the generic establishment of a voice connection using a suitable call establishment protocol (e.g., H.323, RFC 2327, etc.). The end-point DCEs attempt to connect without any knowledge that they are using an IP network with an undetermined Quality of Service (QoS). The second model, also shown in Figure 1, is similar; the exception is that the IP network has a well-managed QoS.

Architecturally, this Recommendation only considers the support of a PSTN to IP to PSTN structure. PSTN to IP structures are for further study.

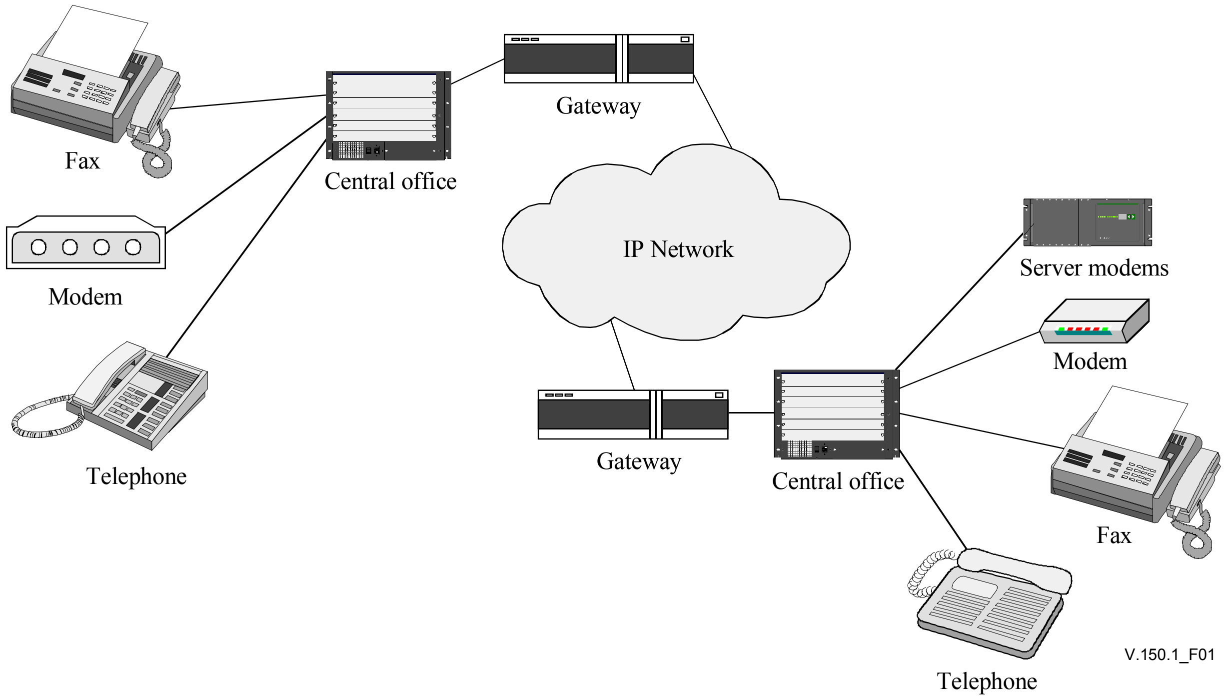


Figure 1/V.150.1 – Typical modem over IP application

5.1 Compliance requirements

The Recommendation does not require behaviour that is inconsistent with other V-series PSTN modem Recommendations, or with national regulatory requirements, and shall be interpreted accordingly. Neither does it preclude the use of proprietary or non-standard modems, however, it does caution that if such devices are used, then care should be taken as not to harm the functionality and procedures defined herein.

In order to be compliant with this Recommendation an implementation must provide functionality that is defined as mandatory.

6 Modem-over-IP gateway functions

Figure 2 provides a conceptual reference model for a MoIP gateway. The model shows two stacks conjoined by the MoIP application. The left-hand stack is that of a typical modem which has a signal converter (modulation), an error correction protocol and a data compression protocol. Apart from the signal converter, the other two protocols may or may not be present. The right-hand stack represents the IP networking functions of a MoIP gateway. The MoIP application as indicated in Figure 2 is defined by the normative contents of this Recommendation. This Recommendation defines a default IP transport layer protocol called SPRT (see Annex B), but does not preclude the use of another protocol that conforms to the requirements as defined in clause 25.

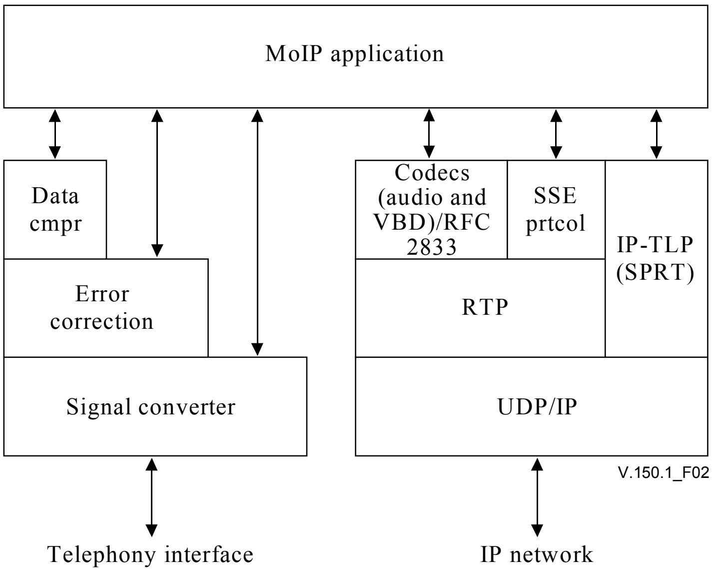


Figure 2/V.150.1 – Gateway reference model

This Recommendation defines two classes of operation for Modem-Over-IP gateways: Voiceband Data and Modem Relay. The two classes of operation are because of the variety of IP networks, DCE feature sets, and gateway capabilities.

7 Audio mode

In this mode, the channel processes speech signals. The mode may include the use of compression algorithms and other processing functions that are not suitable for the transport of modem or facsimile signals.

8 Voiceband data mode

For Voice Band Data (VBD) mode of operation, the path between G1 and G2 remains in a voice configuration. The modem signals are encoded using an appropriate speech codec suitable for the task, and the samples are transported across a packet network.

VBD should:

- Use a voice codec that passes voiceband modulated signals with minimal distortion.
- Have end-to-end constant latency.
- Disable Voice Activity Detection and Comfort Noise Generation during the data transfer phase.
- Disable any DC removal filters that may be integral with the speech encoder used.

VBD should consider the appropriate application of:

- The use of echo cancellers on a VBD channel.
- Forward Error Correction (FEC) (e.g., RFC 2733) or other forms of Redundancy (e.g., RFC 2198).
- Voice packet loss concealment techniques and algorithms that are suitable for modem and facsimile modulations.

8.1 Selection of VBD voice codecs and other enhanced functionality

VBD capabilities are indicated during the Call Set-up phase. See Annexes E and F.

Gateways should undertake appropriate action to avoid more than one trans-coding when in VBD.

8.2 Minimum requirements for VBD

For purposes of interoperability, MoIP gateways shall support at least both G.711 A-law and G.711 μ-law codecs.

9 Modem relay mode

Modem Relay mode of operation is characterized by the termination of both the physical layer and error-correction functions at the gateway. Appendix I provides diagrams of the reference configurations considered by this Recommendation. Modem Relay gateways, depending upon their type demodulate the DCE signals, perform local error correction and pass on the data in one of four ways. These four modes of MR operation are defined in clause 13 and are uniquely defined by the capability and configuration of the gateway Trans-Compression functions.

Figure 3 describes a basic reference model for MR mode of operation. Cx and Dx represent the compression and decompression functions. EC is the Error Correction layer, which also includes any link layer formatting such as HDLC.

NOTE – This reference model shows the functions that may be present depending upon the outcome of the gateway capability exchanges and modem negotiation.

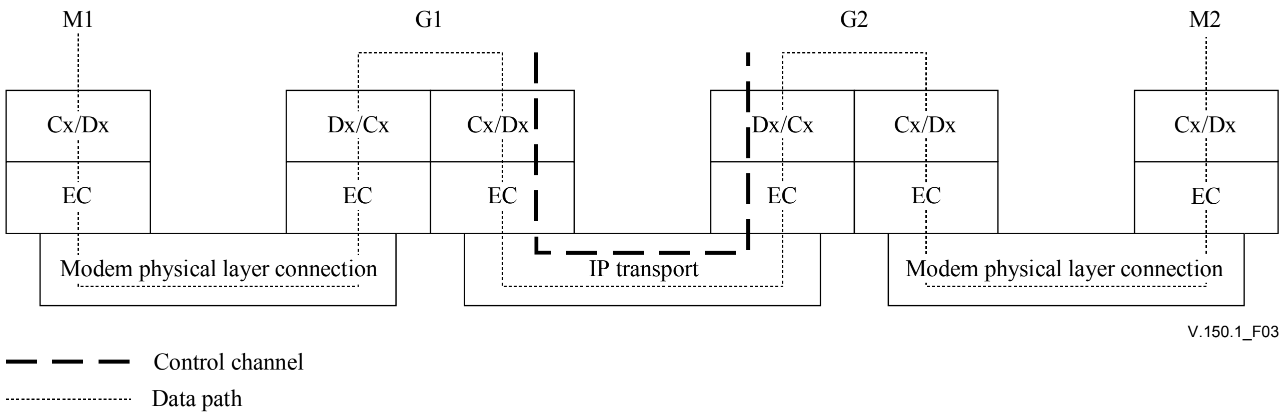


Figure 3/V.150.1 – Modem relay reference model

In addition to the primary data path between the gateways, a secondary control channel is also provided for the exchange of information directly between the gateways. Note the characteristics and format of this control channel are dependent upon the IP transport Protocol being used. The requirements for this control channel are described in 14.7.

Modem Relay defines two gateway types: these are the Universal-Modem Relay (U-MR) gateway and the V.8-Modem Relay (V-MR) gateway.

9.1 Universal-modem relay gateway type

A Universal-Modem Relay gateway (U-MR) provides full termination for a minimum set of V-series modulations, whether negotiated through V.8 or not. This minimum set is defined as:

V.92 Digital, V.90 Digital, V.34, V.32 *bis*, V.32, V.22 *bis*, V.22, V.23 and V.21.

Other modulations may also be supported but are not part of the mandatory set and should be indicated by the gateway during the Call Set-up phase.

9.2 V.8-modem relay gateway type

A V.8-Modem Relay gateway (V-MR) provides full termination for V-series modulations that are selected via the V.8 negotiation procedures. There are no requirements on the set of modulations to be supported by this type of gateway. The modulations that are supported are indicated in the inter-gateway messages exchanged during the Call Set-up Phase.

10 Operational modes of MoIP

A Modem-over-IP (MoIP) gateway has three operational modes. These are Audio Mode, Voice Band Data (VBD) mode and Modem Relay (MR) mode. If available, Audio mode is the initial or default mode of operation for MoIP (see clause 7). If Audio Mode is not an available mode, then a MoIP gateway may begin in either VBD or MR modes.

Other modes of operation that may be optionally provided by MoIP gateways are Fax over IP (FoIP) and Text over IP (ToIP). The procedures for FoIP are defined in ITU-T Rec. T.38. Procedures for ToIP are beyond the scope of this Recommendation. The transitions between MoIP and FoIP/ToIP using the SSE protocol are for further study.

Figure 4 illustrates the relationship between these defined modes. At any given time, a MoIP gateway shall be either in or in transition to one of these modes.

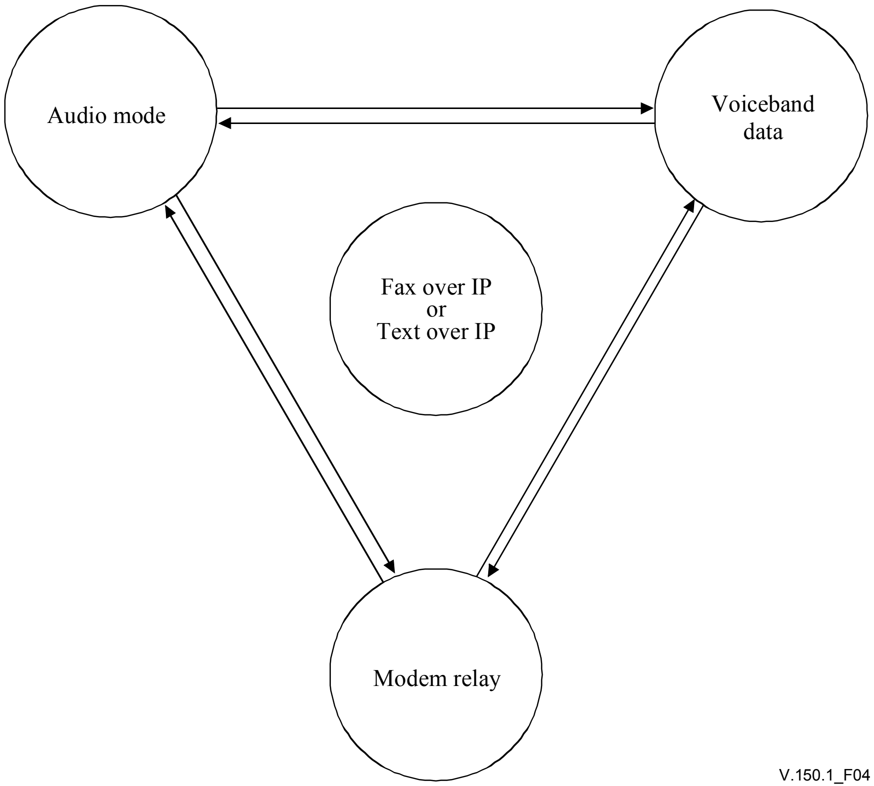


Figure 4/V.150.1 – MoIP operational modes and transition state

11 Modem relay PHY layer functions

This clause describes the functionality and expected behaviour of the Modem Relay PHY. The PHY in this context is defined as the physical layer of the modem to gateway connection and does not include the IP physical layer.

The goal of MoIP is to ensure connectivity of V-series modems on IP networks. The Recommendation does not require nor preclude the use of non-standard modulations. The MoIP procedures take into account that there are two independent PSTN connections to be established in order to provide a single end-to-end connection of the end-point modems. The establishment of the modem physical layer consists of two stages. The first is that of call discrimination and the second is of modem training and capability negotiation. The procedures that define this process are described in clause 20 and Appendix II.

The overall connection physical layer may be selectively transported in either VBD or Modem Relay. The call discrimination and mode selection procedures define the rules for this choice. The behaviour of Modem Relay is determined whether a Universal-Modem Relay (U-MR) or V.8-Modem Relay (V-MR) gateway is being used. Modem Relay supports the ability for a gateway to use different modulation modes on each of the different PSTN links. It is also possible to match the modulations on each PSTN link if necessary. To aid the call discrimination process, gateways exchange their preference for call discrimination and Answer Tone treatment, their supported modulation capability set and whether they are a V-MR- or U-MR-type gateway.

When modems are connected using V.8 negotiation procedures, MoIP defines three cases that are dependent upon the modulation capabilities of the gateways (see 20.5). The gateways are able to determine *a priori* the most appropriate call discrimination procedures to be used when using V.8.

For modulations that allow the selection of multiple data signalling rates, MoIP allows the gateway to either have data signalling rates independent of the two PSTN links or have them matched. Note that the MoIP error control procedures also influence the selection of the link data signalling rates especially in the non-error correcting case.

During the modulation start-up procedures, the gateway shall activate its IP-TLP. This is indicated by transmission of an INIT message (see 15.4.1). The gateways may transmit other IP-TLP messages types once it has received an INIT from the remote gateway.

When the physical layer on the PSTN link has achieved data mode (i.e., the point in time when either V.24 Circuit 106 or 109 may be turned on) the gateway shall transmit the IP-TLP MR_EVENT(PHYSUP) message (see 15.4.8). The selected modulation, data signalling rates and symbol rate of the modem may be used by the remote gateway. The same information (excluding the optional symbol rates) is also made available by the IP-TLP CONNECT message, which is transmitted after the link layer negotiation is complete.

Once established, interruptions to the PSTN PHY caused by retrains and rate-renegotiations may occur and stop the flow of data across the PSTN link. These events shall be reported to the remote gateway using the IP-TLP MR_EVENT message. When the modems return to a stable data mode state the gateway informs its peer by means of the MR_EVENT(PHYSUP) message its current physical layer characteristics. This procedure is described in clause 23.

The only requirement specified for a cleardown of a MoIP session, is that the gateway shall return to its initial state. A gateway shall not initiate a cleardown of the PSTN PHY at its own discretion. Session cleardown and related procedures are described in clause 24.

12 Modem relay error control functions

This clause describes the functionality necessary to support Error Control and No-Error Control modes of Modem Relay.

All the supported connection scenarios for Modem Relay, as described in Appendix I, have the PSTN link error control functions terminated locally by the gateways. Modem Relay may have the same or different error control functions on each of the PSTN links. Figure 5 illustrates possible combinations of error control function configurations. This Recommendation defines procedures that allow the support of links where the end-point modems may either enable or disable their error control functions. For a MoIP gateway the importance of having error control on both the PSTN and the IP links is that it provides full end-to-end flow control. This ability helps to prevent loss of data due to a mismatch in PSTN link characteristics such as different modulation or data signalling rates.

The type of data being transferred across the connection also has relevance to the error control configuration. Much of PSTN modem traffic is start-stop delimited character formatted data and error control is provided by the procedures defined in ITU-T Rec. V.42 which provides the mechanism for conveying start-stop characters over a synchronous data link. If error control is not negotiated, then V.14 procedures provide the method of conveyance as defined in 7.9/V.42. V.14 does not explicitly provide flow control.

Applications that use synchronous data tend to use a link layer protocol that is external to the modem. This is often considered part of an application. From the perspective of a gateway, error control would be disabled intentionally and the link would appear as non-error correction or V.14.

Non-error correction mode of operation of modem relay shall be supported using the procedures defined in this Recommendation. These procedures try to minimize any loss of user data on connections where there is not full end-to-end error control. However, despite a best effort made by a transmitting gateway, there is always a possibility that data may be discarded by a receiving gateway in these circumstances.

12.1 Error control configurations

The following error control configurations are defined because of a link layer negotiation.

12.1.1 Error-controlled

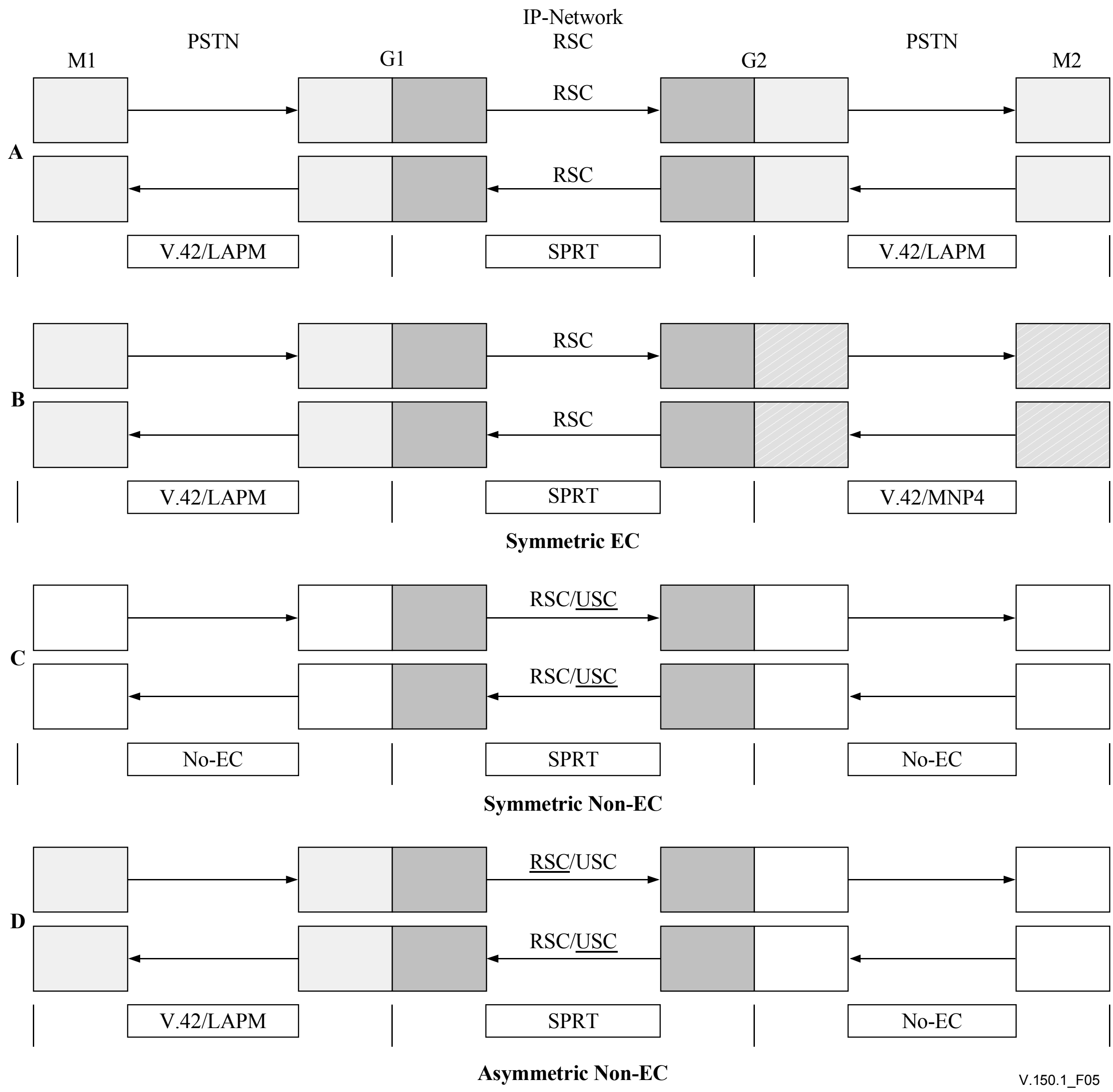
This configuration occurs when both MoIP gateways connect using MR have their error control functions as negotiated between the gateways and the end-point modems (G1/M1 and G2/M2) to be the same. The example depicted in Figure 5A shows both PSTN links selecting V.42/LAPM. Also included are configurations where both MoIP gateways connecting using MR have their error control functions as negotiated between the gateways and the end-point modems (G1/M1 and G2/M2) to be different. The example in Figure 5B shows one PSTN link using V.42/LAPM and the other PSTN link using Annex A/V.42 (1996).

12.1.2 Symmetric-non-error-controlled

Figure 5C illustrates the configuration in which MoIP gateways connecting using MR have both their PSTN link error control functions disabled by the end-point modems (M1 and M2).

12.1.3 Asymmetric-non-error-controlled

For this configuration, the MoIP gateways when connecting using MR have a mixture of enabled and disabled PSTN Error-Control functions. Figure 5D shows the example where one PSTN link uses V.42/LAPM and the other PSTN link does not have any error control.



NOTE – For the Non-EC cases the default transport channels are shown with an underline.
RSC is reliable sequenced channel and USC is unreliable sequenced channel.

Figure 5/V.150.1 – Example of error control configurations

12.1.4 Transport channel selection

For the non-error-controlled configuration, a gateway has the option to select the transport channel it wishes to transmit on, depending upon the remote gateway's capabilities. The selection of these capabilities may be based upon a trade-off between channel reliability and latency, which is implementation-specific. For each of the two configurations, a default transport channel is defined and shall be supported by both gateways. A gateway may indicate its receiving transport channel capability.

Each gateway may indicate it has a capability to receive a non-default transport channel. If a peer gateway has indicated this capability, the transport channel transmitter shall decide whether to use the non-default option for the transport channel prior to initial transmission.

Once a gateway has selected which transport channel it shall use to transmit, it shall not be permitted to use the other channel during the Modem Relay session. This does not apply to the use of the expedited channel.

12.2 Error-controlled PSTN links

This is the typical operational mode of Modem Relay connections. The two PSTN links and the IP-TLP all negotiate their own error control and operational parameters independent of each other. Note this Recommendation does not allow for the exchange of Error Control parameters between the end-to-end modem pair.

12.2.1 Transport channel selection for error-controlled configurations

For error-controlled configurations the IP-TLP shall use a Reliable Sequenced Channel (RSC) (e.g., SPRT Channel 1) for both directions of packet transmission.

12.2.2 Flow control for error-controlled configurations

With an error-controlled link, full end-to-end flow control is present. The flow control for each of the separate error-controlled links is independent of each other. Such data flow control may be performed using V.42 or IP-TLP (e.g., SPRT) procedures. For example, M1/G1 and G2/M2 modem pairs may flow off data from each other by transmitting V.42 RNR: G1 and G2 can flow off by using the window flow control mechanisms of the IP-TLP.

12.3 Non-error-controlled PSTN links

Connections occurring in one of the non-error-controlled configurations may occur:

- Due to an application requiring a synchronous data link; or
- Be disabled to reduce link layer latencies (e.g., electronic gaming); or
- The modems have failed to negotiate error control due to PSTN line conditions.

If a modem fails to establish an error-controlled link, 7.9/V.42 defines that the fallback shall be according to ITU-T Rec. V.14. V.14 is based upon start-stop delimited data.

12.3.1 Transport channel selection for non-error-controlled links

For the non-error-controlled links the selection of the transport channel depends upon whether the links are symmetrically or asymmetrically non-error-controlled.

12.3.1.1 Symmetric No-EC transport channel selection

For the Symmetric No-EC connection, either RSC or an Unreliable Sequenced Channel (USC) (e.g., SPRT Channel 3) may be used for both of the transmission paths of the IP-TLP. The default Transport Channel in this configuration is USC.

12.3.1.2 Asymmetric No-EC transport channel selection

There are independent options for each of the transmit directions for this hybrid mode of operation. For the No-EC to EC transmit direction both the RSC and USC may be used, with the default transport channel being USC. For the EC to No-EC transmit direction again both RSC and USC may be optionally chosen, however, in this case the default is RSC.

12.3.2 Effective data signalling rate matching

The relationship between the physical layer and the link layer for MoIP is very important. It is possible for there to be a mismatch in either modulation or data signalling rate between the two PSTN links. A mismatch in the data signalling rates as selected by the PSTN links requires certain considerations to minimize data loss. In general, modulations may be categorized as being either low-speed or high-speed modulations. For this Recommendation, the term high-speed modulations refers to V.32/V.32 *bis*, V.34, V.90 etc. and low-speed modulations are V.22/V.22 *bis*, V.23, V.21 etc. The general distinction used is that high-speed modulations provide mechanisms to change their data-signalling rate during a connection and low speed modulations do not. For MoIP this is an important characteristic.

12.3.2.1 Rate match rule

The following rule shall be applied where possible to ensure the best possible reduction in data loss due to mismatched data-signalling rates.

12.3.2.1.1 The effective data-signalling rate that a gateway injects into the IP network should be less than the effective data-signalling rate that can be transmitted by the remote gateway onto its telephony leg.

12.3.3 Symmetric non-error-controlled

In this subconfiguration of non-error-controlled links, there are three categories of PSTN link signalling rates to consider: matched high-speed modulations, matched low-speed modulations and mismatched modulations.

12.3.3.1 Matched high-speed modulations

In this category, both the PSTN links have selected a modulation that has the ability to modify their data-signalling rates during the connection, either by retrain or a rate renegotiation procedure. The Rate Match Rule (see 12.3.2.1.1) should be applied during a MR session.

12.3.3.2 Matched low-speed modulations

In this category, both PSTN links have selected to operate in a low-speed modulation. Since there is no ability to modify the data-signalling rate for this type of connection, the gateways should use appropriate buffering to minimize loss of data while avoiding excess latency.

Note that the use of data buffering is implementation specific and is beyond the scope of this Recommendation.

12.3.3.3 Mismatched modulations

This category is where one of the PSTN links has selected a high-speed modulation and, the other link, a low-speed modulation. If possible, the high-speed link should change its data-signalling rate to meet the requirements of the Rate Match Rule. If a match of rate is not possible due to non-overlap in the supported rates in the links, a best effort should be made. As with the matched low-speed modulation case, use of data buffering and reduced latency may reduce data loss.

12.3.3.4 IP-TLP data under-run

For connections where an effective data signalling rate mismatch is unavoidable, there is a possibility that the IP network data being received by the gateway with the higher speed PSTN link will under-run. If appropriate to the data type being transported, this gateway may attempt to fill out the data by inserting the correct idle symbol.

12.3.4 Asymmetric non-error-controlled

This considers the case where the connection has both an error-controlled PSTN link and a non-error-controlled PSTN link. Loss of data due to mismatched data-signalling rates is mitigated to some degree by the presence of one error-controlled entity in the connection. There are four combinations to consider for this type of connection.

12.3.4.1 Low-speed error-controlled link to low-speed non-error-controlled link

For this example, there are also two conditions to consider. Although the modulations selected for the links are considered low-speed, there may still be mismatched data-signalling rates. For example, V.22 *bis* to V.21, which would give 2400 bit/s on one of the links and 300 bit/s on the other. The symmetric case could also occur.

12.3.4.1.1 Error-controlled has higher rate

Since the link with the higher rate is error-controlled, its effective data-signalling rate being input to the IP network can be less than the rate at which it is being delivered from the peer gateway to its local modem. This meets the Rate Match Rule defined in 12.3.2.1.1.

12.3.4.1.2 Non-error-controlled has higher rate

In this type of connection, the Rate Match Rule for the error-controlled to non-error-controlled transmit path is established, so there are no issues with data over-run. However, in the non-error-controlled to error-controlled transmission path, the data-signalling rate from the G2/M2 link is higher than that of the M1/G1 link. It will require appropriate buffering of data to minimize data loss due to over-running of data.

12.3.4.2 High-speed error-controlled link to low-speed non-error-controlled link

This connection type is similar to that described in 12.3.4.1.1. The ability for the high-speed modulated link to modify its data-signalling rate, such that it meets the Rate Match Rule, is an advantage for this case.

12.3.4.3 Low-speed error-controlled link to high-speed non-error-controlled link

This connection type is similar to that described in 12.3.4.1.2. The data-signalling rate from the error-controlled link is less than that of the non-error-controlled link. In the opposite direction, it is possible for the rate from non-error-controlled link to exceed the ability of the error-controlled link to receive it without over-running.

Since the non-error-controlled link has the ability to modify its data-signalling rate to meet the Rate Match Rule, then loss of data can be mitigated. In the case where the Rate Match Rule cannot be met, additional buffering may help, but loss of data may still occur.

12.3.4.4 High-speed error-controlled link to high-speed non-error-controlled link

For this connection type, both links have the ability to modify their data-signalling rate. Applying the Rate Match Rule to adjust the effective data-signalling rates of the two PSTN links will mitigate any potential data loss due to over-run.

12.4 Break handling

This clause introduces how a MoIP gateway supports the transport of break signals that are generated by an end-point DCE. Also described are the methods used to handle the break acknowledgement signal when it is applicable. Clauses 15.4.6 and 15.4.7 define the message formats and 22.1.1 defines the procedures.

The following error-correcting scenarios (as defined in 12.1) are all considered: error corrected; symmetrical non-error corrected and asymmetrical non-error corrected.

Every unique (non-repeated) break signal generated by a local end-point DCE shall be relayed to the remote end-point DCE. Repeated break signals shall not be relayed to the end-point DCE. For the case where the error control protocols are different, or not present on each end of the connections, a break type translation or mapping rule is defined (see Table 38). Also specified are the IP-TLP channels that the break and break acknowledgment messages shall use.

In the symmetric non-error corrected scenario, the transport of the break signal may or may not be transparent, depending upon the data type being used in the MR session.

13 Modem relay trans-compression functions

This clause describes the Trans-Compression functionality. The Trans-Compression Message definitions are described in 15.2.2, 15.4.2 and 15.4.5. The procedures used to select and operate the Trans-Compression functions are defined in 22.2.3 and 22.2.4.

Gateways can be classified by their Trans-Compression (TCX) functionality. TCX is defined as:

An element that consists of a Decompression function of Type A (Dx) and a Compression function of Type B (Cx), where the output of Dx is connected to the input of Cx. Compression and Decompression Types (e.g., V.42 *bis*, V.44) A and B may or may not be the same.

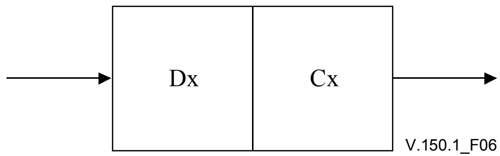


Figure 6/V.150.1 – Trans-compression function

Figure 6 shows the Trans-compression function. Note that Dx, Cx or both can be on or off depending upon the outcome of compression negotiations.

In the situation where a trans-compression element has identical characteristics for both the Dx and Cx, then trans-compression is unnecessary. Gateways may disable trans-compression and revert to a no trans-compression configuration or transparent case for TCX.

This Recommendation defines three basic configurations of TCX. These are described below.

Gateways that implement trans-compression shall support V.42 *bis* compression. Other compression mechanisms are optional.

13.1 No Trans-Compression (N-TCX)

In this configuration as shown in Figure 7, the gateways do not perform any trans-compression but, by means of a proxy-procedure, negotiate on behalf of the modems a common compression mode in terms of algorithm and parameters. The compressed data generated by modems M1 and M2 is transferred end to end.

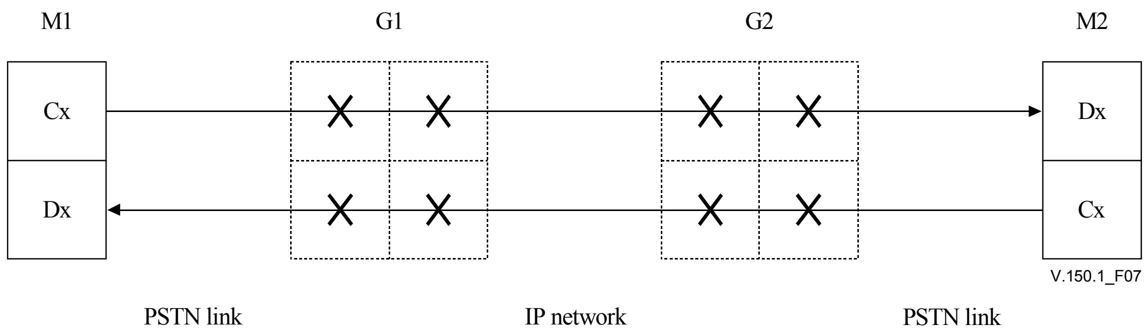


Figure 7/V.150.1 – No trans-compression

13.2 Single Trans-Compression (S-TCX)

In this configuration, the gateways only perform a single Trans-Compression function and the trans-compression functions are equally shared between the gateways.

By convention, the On-Ramp gateway (G1) shall place its trans-compression function in the G1 to M1 transmission path and the Off-Ramp gateway (G2) shall place its trans-compression function in the G2 to M2 transmission path as illustrated in Figure 8.

A Single Trans-Compression gateway shall also provide No Trans-Compression mode of operation.

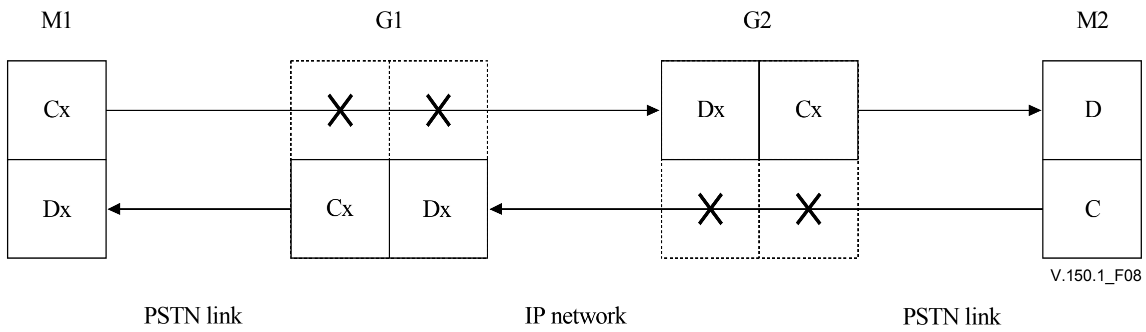


Figure 8/V.150.1 – Single trans-compression

13.3 Double Trans-Compression (D-TCX)

A Double Trans-Compression gateway is one that has two such functions, one in each transmission path. Figures 9 and 10 illustrate the configuration of the Double Trans-Compression gateway. In Figure 9, G2 is the Double Trans-compression and in Figure 10, both G1 and G2 are Double Trans-compression functions.

A Double Trans-Compression gateway shall also support Single Trans-Compression and No Trans-Compression modes of operation.

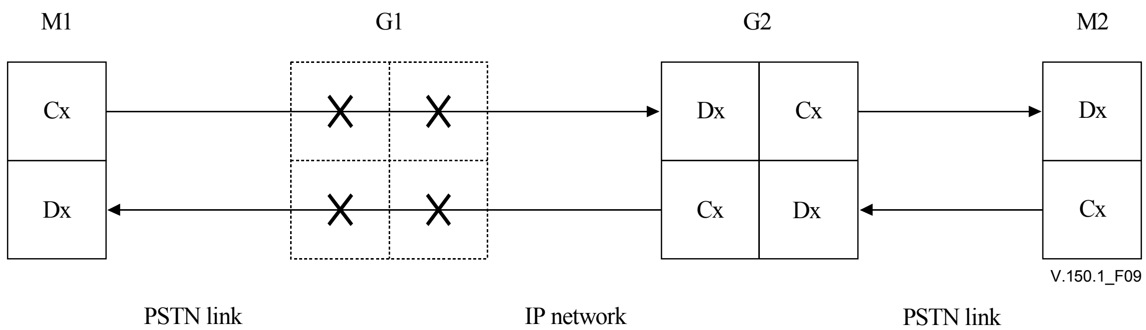


Figure 9/V.150.1 – Double trans-compression (asymmetric)

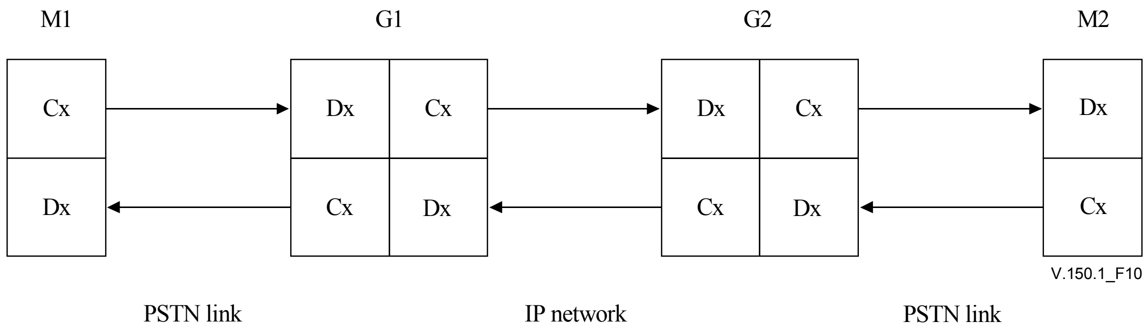
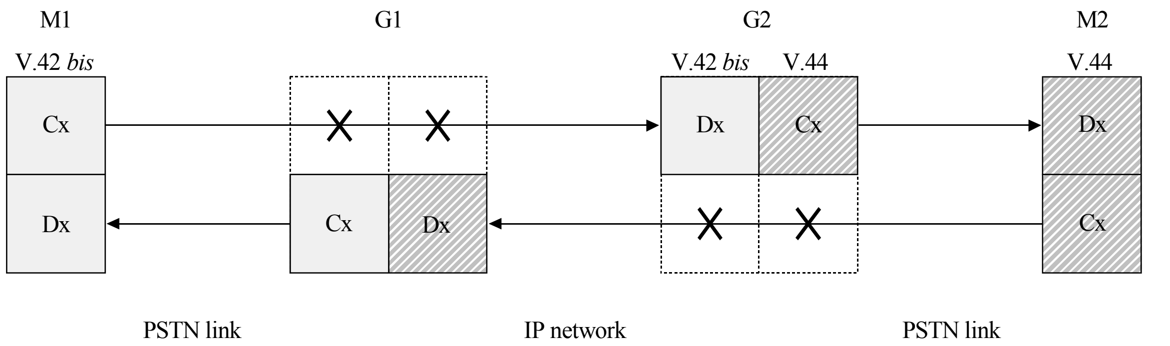


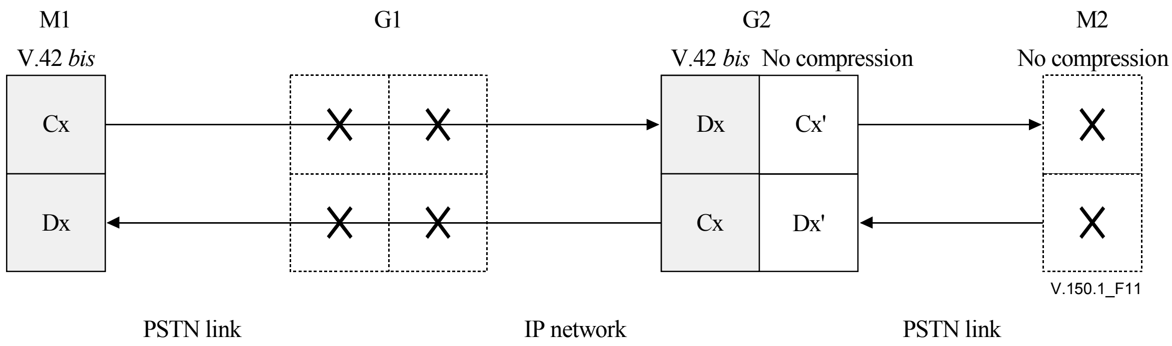
Figure 10/V.150.1 – Double trans-compression (symmetric)

13.4 Mixed-function operation

These previous figures describe the generic reference model for the gateway compression. There are many possible permutations of the trans-compression function along with different compression types. Figure 11 shows two such examples. The first illustrates a trans-compression between V.42 *bis* and V.44 and the second is between V.42 *bis* and a no compression case.



a) Single trans-compression (V.42 *bis* to V.44)



b) Double trans-compression (asymmetric) with M2 disabling compression and M1 using V.42 *bis*

NOTE – Cx' and Dx' represent disabled compression and decompression functions.

Figure 11/V.150.1 – Two examples of mixed-function trans-compression configuration

13.5 Hierarchy for interoperability

To support the ability of a gateway to choose a configuration that best suits its needs (i.e., MIPS, memory and performance optimization), the gateway may support any of the basic configurations of Trans-Compression as described in 13.1, 13.2 and 13.3. However, in order to guarantee interoperability, this clause defines a hierarchy of minimum Trans-Compression functionality that a gateway is required to support.

There are two phases in the process to determine the configuration of the TCX. The first phase is the capability exchange phase. This occurs during the call set-up. These messages are defined in 15.2.2 and are exchanged between the On-Ramp and Off-Ramp gateways. The gateways declare whether they are of the None, Single or Double Trans-Compression types.

For the case where both G1 and G2 declare they are Double Trans-Compression gateways, their initial configuration can be Single-to-Single configuration or if both gateways indicate their desire they can mutually choose the Symmetric Double Trans-Compression configuration. This option is indicated as part of the gateways compression capability message during the call set-up phase (see 15.2.10).

As a result of the Trans-Compression capability exchange, the gateways shall select the mode of operation as defined in Table 1.

Table 1/V.150.1 – Trans-compression mode selection

Trans-compression mode capabilities exchanged		**Resultant trans-compression mode selected by gateways**	
G1	**G2**	**G1**	**G2**
No	No	No	No
No	Single	No	No
No	Double	No	Double
Single	No	No	No
Single	Single	Single	Single
Single	Double	Single	Single
Double	No	Double	None
Double	Single	Single	Single
Double	Double	Single/Double (Note)	Single/Double (Note)
NOTE – If both gateways indicate their ability and desire to support Double-Double configuration it shall do so, otherwise they shall use the Single-Single configuration.			

13.6 XID/LR profiles for use in modem relay connection scenario MR1

Modem relay connection Scenario MR1 requires both M1 and M2 to negotiate identical compression parameters. This clause defines an optional method to provide the ability to predict trans-compression parameters based upon knowledge gained from a previous connection, or by other means. This will allow the gateways to optimize the protocol XID/LR exchange for improved connect time, compression efficiency, and connectivity.

The gateways may exchange their knowledge as described in 22.2.5, and use this information to agree on optimal and consistent XID/LR command and response sequences that each will send when the particular physical connection comes up.

The advantage of using this prior knowledge is that an actual synchronous end-to-end exchange is not necessary, thus minimizing the timing problems associated with such exchanges.

The method and means for the gateways to determine or predict the XID/LR profiles is outside the scope of this Recommendation. Only the procedures to allow the exchange of information are defined.

As indicated, these procedures and messages are optional and are only used when a gateway indicates that it has the ability to receive the new PROF_XCHG (XID/LR profile exchange) message. This indication is provided in the IP-TLP INIT message.

14 Modem relay data transfer

This Recommendation supports the following modem relay data types that are used to exchange user information between the gateways. A set of mandatory data types is defined along with a set of optional types. This clause describes the functionality of the modem relay data types. Their format is described in 15.4.11 and procedures associated with their use in 22.3.

14.1 Data type definitions

Octet Data types may or may not contain formatted data.

The Raw Data types contain data that is in the exact same format and state as it was being transmitted over the PSTN link. No modification to the data is allowed apart from the optional use of compression to reduce the overall amount of data being exchanged between the gateways. The compression defined is lossless and allows for exact representation of the data at a receiver.

14.2 Support of DLCI

The support of DLCI is indicated by a code point in the IP-TLP CONNECT message. This bit when received by a gateway indicates that the peer gateway transmitter is to include a DLCI. If transmitted the DLCI shall be interpreted as specified in 8.2.1.1/V.42. Note that the DLCI may be eight or sixteen-bit fields.

14.3 General statement on the use of start-stop characters

Unless specifically indicated Start-Stop character data shall be sent unpacked, that is one character per octet and one octet per character. MoIP requires implementations to support Start-Stop character formats of one stop bit and whose character may be represented in an eight-bit data field, which may consist of both data and parity. This allows the transparent support of seven-bit data and parity or eight-bit data no parity characters.

14.4 Selection of data types

For a given error-correction mode of operation a default data type is defined. For symmetrical data type operation, both gateway IP transmitters shall use the same data type. Asymmetric data types may be used if both gateways indicate support in their INIT message. A gateway shall use symmetric data types by default.

A gateway shall not change the data type during that modem relay session.

14.5 Mandatory modem relay data types

The choice of mandatory data types is dependent upon the error correction configuration negotiated. For each of these configurations a default is specified. The default data type for error-controlled configurations is octet without format. For asymmetric non-error-controlled, it is also octet without format. In the symmetric non-error-controlled configuration the default is raw octet compressed.

For the error-controlled and asymmetric non-error-controlled configurations, raw octet compressed may not be used; the only choice from the mandatory set is octet without format. For symmetric non-error-corrected configurations, both octet without formatting and raw octet compressed may be chosen. If the configuration is symmetric non-error-controlled octet without format is treated as an optional data type in the data type selection procedures.

14.5.1 Octet without formatting

This data type may be used for error-controlled data or character data mode. When used in character data mode, it shall not include start-stop bits. This data type may include DLCI. The use of the DLCI is indicated by a codepoint in the IP-TLP CONNECT message.

14.5.2 Raw octet compressed

This data type has application with synchronous data streams. This raw data type shall preserve all bits received from the DCE interface. The amount of data exchanged may be reduced by use of simple compression. Repeated octet patterns are coded and removed by the transmitter from the data stream. The receiver decodes the stream and restores it to original form.

14.6 Optional modem relay data types

Gateways shall indicate the capability and support of their receivers for the optional data types defined in this Recommendation. Gateway packet transmitters shall not use an optional data type if the peer gateway receiver does not support it and there is no obligation of the transmitter of an optional data types to use a capability indicated by a receiver.

14.6.1 Character with formatting data type

This data type allows for the transport of Start-Stop characters of different formats. The characters are transmitted unpacked (i.e., one character per octet, one octet per character). This data type shall use the same format rules as defined in Annex B/V.42. The mandatory Start-Stop character capabilities a gateway shall support are Data plus Parity shall equal eight bits and one stop-bit.

There are two forms of this data type; both are indicated as separate options. The first allows for the changing of character format during a MR session and the second does not. Each type has a unique message ID.

14.6.2 Raw bit compressed data type

This data type is used in the same application as the raw octet compressed type. This format codes and decodes N-bit repeated patterns. N may have the value of 8.

14.6.3 Framed data type

This optional data type is for use by a pair of consenting gateways if a gateway is able to determine that the data format is Framed data consistent with ISO/IEC 3309 and 4335. If this data type is selected by the gateways, it may transmit the data with the framing elements removed. These elements if removed shall be replaced by the receiver such that the data remains consistent with ISO/IEC 3309 and 4335. The means and method by which a gateway determines the data type is beyond the scope of this Recommendation.

This data type shall not be used when either or both gateways have negotiated V.42 or Annex A/V.42 (1996).

14.7 Gateway-to-gateway control channel functionality and interfaces

The gateway-to-gateway control channel shall be reliable and provide expedited delivery of control information (IP-TLP messages).

15 Modem-over-IP functionality and interfaces

15.1 Gateway-to-gateway protocol definitions and procedures

15.1.1 General

The gateway-to-gateway communication is specified by the ASN.1 description in Annex A. This description complies with the specification of ASN.1 (see ITU-T Rec. X.680). The ASN.1 encoding in Annex A shall employ the BASIC-ALIGNED version of Packed Encoding Rules (PER) according to ITU-T Rec. X.691. In the case of a conflict between the ASN.1 and the text, the ASN.1 governs.

15.1.2 Bit and octet transmission order

Transmission order is as defined in Internet RFC 791 "Internet Protocol," quoted herein as reference:

– The order of transmission of the header and data described in this Recommendation is resolved to the octet level. Whenever a diagram shows a group of octets, the order of transmission of those octets is the normal order in which they are read in English. For example, in the following diagram the octets are transmitted in the order they are numbered.

0 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19 20 21 22 23 24 25 26 27 28 29 30 31

1	2	3	4
5	6	7	8
9	10	11	12

Figure 12/V.150.1 – Transmission order of octets (based on RFC 791, Figure 10)

– Whenever an octet represents a numeric quantity the left most bit in the diagram is the high order or most significant bit. This is, the bit labelled 0 is the most significant bit. For example, the following diagram represents the value 170 (decimal).

0	1	2	3	4	5	6	7
1	0	1	0	1	0	1	0

Figure 13/V.150.1 – Significance of bit (based on RFC 791, Figure 11)

– Similarly, whenever a multi-octet field represents a numeric quantity the left most bit of the whole field is the most significant bit. When a multi-octet quantity is transmitted the most significant octet is transmitted first.

15.1.3 V.42 I/UI frame preservation

Information is copied between V.42 I/UI frames and the appropriate MoIP gateway PDU without regard to any field structure of the V.42 I/UI frames. It is copied as a sequence of octets, with the order of the octets preserved.

15.2 Gateway capability and call set-up messages

This clause defines the functionality of the messages that are exchanged between gateways during the Call Set-up phase. These definitions are used by the following protocols: Annex P/H.323, H.248 (TBD), H.245 and SIP/SDP. The values shown in this set of messages represent what should be functionally indicated by the signalling protocol. Actual values are outside the scope of this Recommendation.

NOTE – In the following definitions default parameter values, if they exist, are shown in bold.

15.2.1 V.150.1 version

This is the version of this Recommendation that the gateway implements. See 1.1.

15.2.2 TCX definitions

The following definitions are included in the Call Set-up message. These are used as part of the TCX procedures as defined in 22.2.

15.2.2.1 Supported trans-compression capability

(Mnemonic: tcxMode) This message indicates the TCX capability of the gateway. Only one mode should be indicated.

Table 2/V.150.1 – Supported TCX capability values

Parameter	Value
Supported TCX capability	Non-TCX/ Single-TCX/Double-TCX

15.2.2.2 Compression modes available

(Mnemonic: cpxSupport) This bit field indicates the compression capabilities the gateway is able to support.

Table 3/V.150.1 – Optional compression modes available

Optional compression mode	Value
V.44	Available/Not Available
MNP5 mode (See Appendix VI)	Available/Not Available

15.2.2.3 V.42 *bis* parameters

(Mnemonic: v42bisParam) If the gateway supports V.42 *bis* compression this message would indicate the maximum parameters the gateway supports for V.42 *bis*.

Table 4/V.150.1 – V.42 *bis* parameters

Mnemonic	Description
V42bNumCodewords	Proposed number of codewords. Valid values from 512 to 65535.
v42bMaxStringLength	Maximum string length. Valid values from 6 to 250.

15.2.2.4 V.44 parameters

(Mnemonic: v44Param) If the gateway supports V.44 compression as indicated above, this message would indicate the maximum parameters the gateway supports for V.44.

Table 5/V.150.1 – V.44 parameters

Mnemonic	Description
v44NumTxCodewords	Proposed number of codewords in the transmitter. Valid values from 256 to 65535.
v44NumRxCodewords	Proposed number of codewords in the receiver. Valid values from 256 to 65535.
v44MaxTxStringLength	Maximum string length in the transmitter. Valid values from 32 to 255.
v44MaxRxStringLength	Maximum string length in the receiver. Valid values from 32 to 255.
V44LenTxHistory	Proposed size of the transmitter history. Valid values from 512 to 65535.
V44LenRxHistory	Proposed size of the receiver history. Valid values from 512 to 65535.

15.2.3 Modem relay mode

This parameter indicates the gateways Modem Relay type is of type V-MR or U-MR.

Table 6/V.150.1 – Modem relay mode

Parameter	Value
Supported Modem Relay Type	V-MR/U-MR

15.2.4 Modulation support

A list of the V-series modulations supported in Modem Relay mode by the gateway.

The list is comprised of V.17, V.21, V.22/V.22 *bis,* V.23 (duplex and half-duplex), V.26 *bis*, V.26 *ter*, V.27 *ter*, V.32/V.32 *bis*, V.34 (duplex and half-duplex), V.90 (analogue and digital), V.91 and V.92 (analogue and digital).

15.2.5 VBD parameters

The specification of the VBD parameters are beyond the scope of this Recommendation. See 8.1.

15.2.6 Answer tone treatment

As part of the Answer tone processing, all MoIP gateways are required to support and indicate at a minimum, the following RFC 2833 events:

ANS (32), /ANS (33), ANSam (34) and /ANSam (35).

15.2.7 SPRT parameters

If the default IP-TLP SPRT is to be used, then the following parameters are required to be indicated:

Maximum payload size of SPRT channels 0 to 3 and maximum window size for SPRT channels 1 and 2.

15.2.8 FoIP support

This parameter indicates that the gateway has the ability to support T.38 FoIP. It is not necessary for an external signalling mechanism to include specific indication of this ability in the MoIP messaging set, if an alternative mechanism is already defined. However, the information as to the peer gateways ability to support FoIP should be passed to the MoIP application. This will allow the gateway to make appropriate decisions should it determine that a connection is a facsimile call during its discrimination procedures.

Table 7/V.150.1 – FoIP support values

Parameter	Value
T.38 FoIP Supported	Supported/Not Supported

15.2.9 JM delay support

This parameter indicates the ability of a gateway to support the JM delay procedure as defined in 20.7. The default is Not Supported. If both gateways have negotiated connection scenario MR1 and both indicate JM delay procedure support they shall follow the procedures as defined in 20.7 for both On-ramp and Off-ramp gateways.

Table 8/V.150.1 – JM delay support values

Parameter	Value
JM Delay Procedure Supported	Supported/**Not Supported**

15.2.10 Double-to-double TCX preference

This parameter indicates the preference of a D-TCX gateway, when connecting to another D-TCX gateway as to which TCX mode it shall use as the basis for its initial configuration. The default is Single.

Table 9/V.150.1 – Double-to-double TCX support values

Parameter	Value
D-TCX to D-TCX initial preference	**Single**/Double

15.2.11 Call discrimination mode selection

(Mnemonic: *CDSCselect*) This parameter indicates which of the three call discrimination modes a gateway prefers. The three possible mode choices for this parameter are (refer to 20.3): Audio (RFC 2833), VBD-select and Mixed.

Table 10/V.150.1 – Supported TCX capability values

Parameter	Value
Call discrimination mode Preferred	Audio (RFC 2833)/VBD-select/Mixed

15.2.12 ToIP support

The support of ToIP is for future study.

15.3 Gateway call discrimination messages

During this phase of MoIP gateway operation, the following signals may be transmitted or received. The support of both SSE and RFC 2833 events ANS, /ANS, ANSam and /ANSam is required. The use of Tone Packets as defined in section 4 of RFC 2833 shall not be used for transporting the Answer Tone events.

Table 11/V.150.1 – Call discrimination messages

Title	Transport channel	Event codes (decimal)	Functional description of message content	Comments
ANS	RFC 2833	32	G2 detect ANS	Uses section 3 of RFC 2833
/ANS	RFC 2833	33	G2 detect /ANS	Uses section 3 of RFC 2833
ANSam	RFC 2833	34	G2 detect ANSam	Uses section 3 of RFC 2833
/ANSam	RFC 2833	35	G2 detect /ANSam	Uses section 3 of RFC 2833
Audio Mode	SSE (Note)	1	Switch to Audio	Includes Reason Code
VBD mode	SSE (Note)	2	Switch to VBD per capabilities	Includes Reason Code
Modem Relay Mode	SSE (Note)	3	Switch to MR per capabilities	Includes Reason Code plus Information field
Fax Relay	SSE (Note)	4	Switch to FoIP	Procedures for this media switch are for further study
Text Relay	SSE (Note)	5	Switch to ToIP	Procedures used for this media switch are for further study
NOTE – The SSE protocol is defined in Annex C. The Event codes are defined also in the same annex.				

15.3.1 SSE reason identifier codes

The following are the Reason Identifier Codes to be used with the SSE protocol as defined in Annex C.

Table 12/V.150.1 – SSE RIC codes for MoIP

Name	Code (decimal)	Additional informational content
Null	0	None
CM	1	Available modulation modes as indicated in the CM sequence (Format is defined in Table 13)
JM	2	Available modulation modes as indicated in the JM sequence (Format is defined in Table 13)
AA	3	None
AC	4	None
USB1	5	None
SB1	6	None
S1	7	None
V.21 Ch2	8	None
V.21 Ch1	9	None
V.23 High channel	10	None
V.23 Low channel	11	None
Tone (2225 Hz)	12	None
V.21 Ch2 HDLC Flags	13	None
Indeterminate signal	14	None
Silence	15	None

Table 12/V.150.1 – SSE RIC codes for MoIP

Name	Code (decimal)	Additional informational content
CNG	16	None
Voice	17	None
Time-Out	18	Indicates that a timeout has occurred. The timeout event and format is defined in Table 14.
p' State Transition	19	None
Cleardown	20	Indicates reason for clear down. Format is defined in Table 15.
ANS/CED (2100 Hz)	21	None
ANSam	22	None
/ANS	23	None
/ANSam	24	None
QC1a	25	None
QC1d	26	None
QC2a	27	None
QC2d	28	None
Cre	29	None
CRd	30	None
Reserved	31-127	Reserved for use by ITU-T
Vendor Specific	128-255	For use by vendor.

Table 13/V.150.1 – CM and JM additional information format in SSE payload (Bits 16 to 31)

Bit numbers																Modulation availability
16	17	18	19	20	21	22	23	24	25	26	27	28	29	30	31	
X																PCM mode
	X															V.34 duplex
		X														V.34 half-duplex
			X													V.32/V.32 *bis*
				X												V.22/V.22 *bis*
					X											V.17
						X										V.29 half-duplex
							X									V.27 *ter*
								X								V.26 *ter*
									X							V.26 *bis*
										X						V.23 duplex
											X					V.23 half-duplex
												X				V.21
													X			V.90 or V.92 analogue
														X		V.90 or V.92 digital
															X	V.91

Table 14/V.150.1 – SSE timeout reason code definitions in SSE payload (Bits 16 to 31)

Time-out MSB:LSB	Definition
b16:b23	This field indicates the eight-bit code used to identify the timeout event. The values for this field are: 0: NULL 1: Call discrimination timeout 2: IP-TLP timeout 3: SSE explicit acknowledgement timeout
b24:b31	This field provides an eight-bit field which may be used to provide additional vendor specific information regarding the timeout.

Table 15/V.150.1 – SSE clear down reason code definitions in SSE payload (Bits 16 to 40)

Cleardown MSB:LSB of SSE payload	Definition
b16:b23	This field indicates the eight-bit code used to identify the clear down event. The values used for this field are the same as used for bits 0 through 7 of the IP-TLP CLRDOWN message and are defined in Table 28.
b24:b31	Reserved for use by the ITU-T. Each of these shall be set to binary-zero.
b48:b55	This field contains the eight-bit vendor-Tag as defined in clause 8/V.150.0.
b56:b63	This field provides an eight-bit field which may be used to provide additional vendor specific information regarding the clear down.
NOTE – The fields used to optionally indicate vendor-specific information uses the extension capability of the SSE payload as described in C.3.2.	

15.4 Modem relay status messages

Table 16 lists the IP-TLP messages and their message identifier used to transport MoIP control and user data. This clause defines the functionality and format of these messages.

Table 16/V.150.1 – List of modem relay messages

Message Name	Message ID (decimal)	Transport Channel	Description
NULL	0	N/A	Reserved for use by ITU-T
INIT	1	2	Sent upon IP-TLP activation. Indicates: transport channel capabilities, data type and XID profile exchange
XID_XCHG	2	2	Message contains the XID information to be used for compression negotiation for connection scenarios in which either gateway is a N-TCX
JM_INFO	3	2	Contains JM signal information for CM-JM procedure
START_JM	4	2	See JM delay procedures in 20.7
CONNECT	5	2	Sent when gateway is ready to exchange user-data. Message indicates the local dial-up link connection parameters (e.g., modulation, data signalling rates, error control and compression)

Table 16/V.150.1 – List of modem relay messages

Message Name	Message ID (decimal)	Transport Channel	Description
Break	6		Sent when a gateway detects a break signal from its local DCE
Break_Ack	7		Used to indicate to a peer gateway that a break acknowledge signal has been detected
MR_EVENT	8	2	Sent by gateway to indicate a change of data mode state during modem relay. Includes physical layer parameters (modulation, data signalling rate and symbol rate), Retrain and Rate renegotiation event indication
CLEARDOWN	9	2	Indicates cleardown event and reason
PROF_XCHG	10	2	Indicates to the peer gateway the XID profile of the gateway's local modem
Reserved	11-15	N/A	Reserved for use by ITU-T
I_RAW-OCTET	16	1 or 3	Raw Octet compressed
I_RAW-BIT	17	1 or 3	Raw bit compressed (optional)
I_OCTET	18	1 or 3	Octet without formatting
I_CHAR-STAT	19	1 or 3	Character with static formatting (optional)
I_CHAR-DYN	20	1 or 3	Character with dynamic formatting (optional)
I_FRAME	21	1 or 3	Framed Data (optional)
Reserved	22-99	N/A	Reserved for use by ITU-T
VENDOR	100-127	N/A	Vendor specific messages

15.4.1 Initialization message (INIT)

This message is sent immediately upon IP-TLP activation. It indicates that an IP-TLP is active on the gateway sending the message. The message indicates the gateways capability for optional data types, IP-TLP receiver channel and XID profile exchange.

Table 17/V.150.1 – INIT definitions

INIT bits MSB:LSB	Definition
0	***NECRxCH*** if set to binary-one indicates to the gateway receiving the message that if the peer gateway has no Error Control then it prefers to receive on the IP-TLP using the non-default transport channel (RSC). If binary zero then the default channel is used (USC).
1	***ECRxCH*** if set to binary-one indicates to the gateway receiving the message that if the connection ends up in an asymmetric non-EC configuration and if the peer gateway is using an Error Control protocol, then the local gateway prefers to receive on the IP-TLP using the non-default transport channel (USC). If binary zero then the default channel is used (RSC).
2	**XID Profile exchange**. This optional capability is supported if set to a binary one.

Table 17/V.150.1 – INIT definitions

INIT bits MSB:LSB	Definition
3	**Asymmetrical Data Types**. If this bit is set it indicates that the gateway has the optional capability of supporting different data types in its transmitter and receiver simultaneously.
4:15	**Optional MoIP Data types**. This field indicates the optional data types that are supported by the gateway's receiver. Support is indicated by a value of binary ONE in the appropriate bit position. A binary ZERO indicates no support. b4: I_RAW-BIT support b5: I_FRAME support b6: I_CHAR-STAT support b7: I_CHAR-DYN support b8…b15: Reserved for ITU-T; these bits are set to zero by the transmitter and not interpreted by the receiver.

15.4.2 N-TCX XID exchange message (XID_XCHG)

This message is sent by the gateway to indicate the XID parameters for compression negotiation. XID_MSG may only be used in connection scenario MR1.

Table 18/V.150.1 – XID_XCHG definitions

XID_XCHG bits MSB:LSB	Definition
0:7	**Error correcting protocol (ECP)**. The following values are used to indicate the error correction protocol contained in the local link negotiation. 0: No link layer protocol (compression is not negotiated) 1: V.42/LAPM 2: Annex A/V.42 (1996) 3…255: Reserved
8:15	**XID/LR field Octet 1**: These fields are only valid if the ECP field above has the value of 1 or 2. This field contains the XID/LR compression parameters. The XID directional parameters have the same reference as the view of the originating modem (i.e., transmit direction is the direction of flow from the originating modem towards the answering modem). This octet indicates the compression supported, b8: V.42 *bis* b9: V.44 b10: MNP5 b11…b15: Reserved for ITU-T; these bits are set to zero by the transmitter and not interpreted by the receiver.
16:23	**XID/LR field Octet 2**: Indicates V.42 *bis* data compression request (P0)
24:39	**XID/LR field Octet 3 and 4**: V.42 *bis* number of codewords (P1)
40:47	**XID/LR field Octet 5**: V.42 *bis* maximum string length (P2)
48:55	**XID/LR field Octet 6**: V.44 capability (C0)

Table 18/V.150.1 – XID_XCHG definitions

XID_XCHG bits MSB:LSB	Definition
56:63	**XID/LR field Octet 7**: V.44 data compression request (P0)
64:79	**XID/LR field Octet 8 and 9**: V.44 number of codewords in transmit direction (P1T)
80:95	**XID/LR field Octet 10 and 11**: V.44 number of codewords in receive direction (P1R)
96:103	**XID/LR field Octet 12**: V.44 maximum string length in transmit direction (P2T)
104:111	**XID/LR field Octet 13**: V.44 maximum string length in receive direction (P2R)
112:127	**XID/LR field Octet 14 and 15**: V.44 length of history in transmit direction (P3T)
128:143	**XID/LR field Octet 16 and 17**: V.44 length of history in receive direction (P3R)
NOTE – Compression parameter fields that are not selected in the compression-supported fields shall be present and set to zero.	

15.4.3 V.8 JM Information (JM_INFO)

This message is transmitted by G2 to indicate the JM signal it receives from its local modem to G1. This message is used during the CM-JM procedure. JM signal information categories used to indicate modulation (i.e., modulation modes and PCM modem availability) shall be included. The following information categories, if present in the received JM, should also be included: Call Function, Protocol and PSTN Access. Other information categories shall not be included in JM_INFO.

The JM_INFO messages are defined to be of variable length. Each information category consists of 16 bits. The first four bits contain a category identifier and the remaining 12 bits contain the information. If JM_INFO contains multiple categories then bits 0 to 15 are repeated for each instance.

Table 19/V.150.1 – JM_INFO definitions

<table>
<tr><th>JM_INFO bits MSB:LSB</th><th>Definition</th></tr>
<tr><td>0:3</td><td>Category ID: This field indicates the category information being indicated in the next field.
Bits
<u>b0 b1 b2 b3</u>
1 0 0 0 Call Function 1
1 0 1 0 Modulation modes 5
0 1 0 1 Protocols
1 0 1 1 PSTN access
1 1 1 0 PCM modem availability
0 0 0 0 Indicates that this is an extension of the current category</td></tr>
<tr><td>4:15</td><td>Category information: This field contains the JM signal information. If more than 12 bits are used then the extension mechanism is used to complete the fields.
The values defined for each of the allowable categories are defined in Tables 20 through 24.</td></tr>
</table>

Table 20/V.150.1 – Call function values for JM_INFO

Bits 4 5 6	Call function values
1 0 0	PSTN Multimedia Terminal (ITU-T Rec. H.324)
0 1 0	Textphone (ITU-T Rec. V.18)
1 1 0	Videotext (ITU-T Rec. T.101)
0 0 1	Transmit facsimile from call terminal (ITU-T Rec. T.30)
1 0 1	Receive facsimile at call terminal (ITU-T Rec. T.30)
0 1 1	Data (V-series modem Recommendations)

Table 21/V.150.1 – Modulation mode values for JM_INFO

Bit	Modulation modes
4	1 denotes V.34 duplex availability
5	1 denotes V.34 half-duplex availability
6	1 denotes V.32 *bis*/V.32 availability
7	1 denotes V.22 *bis*/V.22 availability
8	1 denotes V.17 availability
9	1 denotes V.29 half-duplex availability (as used in ITU-T Rec. T.30)
10	1 denotes V.27 *ter* availability
11	1 denotes V.26 *ter* availability
12	1 denotes V.26 *bis* availability
13	1 denotes V.23 duplex availability
14	1 denotes V.23 half-duplex availability
15	1 denotes V.21 availability

Table 22/V.150.1 – Protocol values for JM_INFO

Bits 4 5 6	Protocol values
1 0 0	Calls for LAPM protocol according to ITU-T Rec. V.42 NOTE – This value also indicates that the modem is able to bypass the ODP/ADP exchange per 9.3.1/V.92.

Table 23/V.150.1 – PSTN access values for JM_INFO

Bits 4 5 6	PSTN access category values
X	1 denotes that the call DCE is on a cellular connection
X	1 denotes that the answer DCE is on a cellular connection
X	1 denotes a DCE on a digital network connection 0 denotes a DCE on an analogue network connection

Table 24/V.150.1 – PCM modem availability values for JM_INFO

Bits 4 5 6	PCM modem availability (Note)
X	1 denotes V.90 or V.92 analogue modem availability
X	1 denotes V.90 or V.92 digital modem availability
X	1 denotes V.91 modem availability
NOTE – The PCM modem availability bit is not provided in the modulation mode field. The assumption is that if the PCM modem availability category is not provided in the JM_INFO, then the modem M2 does not have PCM modem capability as defined.	

15.4.4 Continue with V.8 JM (Start_JM)

This message is used by gateways when operating in a N-TCX scenario as defined in 20.7. This message does not have any additional informational content. This message is used only when both gateways negotiate the optional delay JM procedures.

15.4.5 CONNECT message

On completion of the link layer procedures, the gateways indicate to each other their local modem parameters using the CONNECT message. This message contains information about the local modem modulation, data-signalling rate, error correction and compression selection and parameters.

Table 25/V.150.1 – CONNECT definitions

CONNECT bits MSB:LSB	Definition
0:5	**Selected Modulation (SELMOD)**: This field indicates the modulation chosen for the local PSTN link. The following values are enumerated: 0: NULL 1: V.92 2: V.91 3: V.90 4: V.34 5: V.32 *bis* 6: V.32 7: V.22 *bis* 8: V.22 9: V.17

Table 25/V.150.1 – CONNECT definitions

CONNECT bits MSB:LSB	Definition
	10: V.29 11: V.27 *ter* 12: V.26 *ter* 13: V.26 *bis* 14: V.23 15: V.21 16: Bell 212 (see VI.2) 17: Bell 103 (see VI.1) 18…30: Vendor-specific modulations 31…63: Reserved for ITU-T: these values are not interpreted by the receiver.
6:7	**Compression Direction**: (Note) If compression was negotiated, this field contains the compression direction parameter from the gateway. The following values are allowed: 0: No compression 1: Only in transmit direction (gateway to modem) 2: Only in receive direction (modem to gateway) 3: Both directions
8:11	**Selected Compression**: This field indicates the compression mode selected by the modem-gateway pair. It can have one of four values. 0: No compression 1: V.42 *bis* 2: V.44 3: MNP5 (See VI.4) 4…15: Reserved for ITU-T; these values are not interpreted by the receiver.
12:15	**Selected Error Correction**: This field indicates the type of error correction selected by the modem-gateway pair. It can have one of three values. 0: V.14 or no error correction protocol 1: V.42/LAPM 2: Annex A/V.42 3…15: Reserved for ITU-T; these values are not interpreted by a receiver.
16:31	**Transmit Data Signalling Rate (TDSR)**: The locally selected transmitter data signalling rate in bit/s. (0…65535)
32:47	**Receive Data Signalling Rate (RDSR)**: The locally selected receiver data signalling rate in bit/s. (0…65535)
48	**DLCI enabled**: This code point indicates to the gateway that the peer transmitter will use the octet without formatting with a DLCI data type.

Table 25/V.150.1 – CONNECT definitions

CONNECT bits MSB:LSB	Definition
49:63	**Available Data Types**: This field indicates the data types available for use by the peer gateway. b49: Octet without format with no DLCI. Indication of this data type is only valid for the symmetrical non-error configuration. b50: I_RAW-BIT b51: I_FRAME b52: I_CHAR-STAT b53: I_CHAR-DYN b54…b63: Reserved for ITU-T; these bits are set to zero by the transmitter and not interpreted by the receiver.
64:79	**Compression Transmit Dictionary size**: If no compression or MNP5 is selected this field is not included in the message. If compression selected is V.42 *bis* or V.44, this indicates the (transmit for V.44) dictionary size. Valid values are 512 to 65535 codewords for V.42 *bis* and 256 to 65535 codewords for V.44.
80:95	**Compression Receive Dictionary size**: If no compression, MNP5 is selected this field is not included in the message. If compression selected is V.42 *bis* this field is set to a value of zero. If compression selected is V.44 this indicates the receiver dictionary size. Valid values are 256 to 65535 codewords. (0 if V.42 *bis*)
96:103	**Compression Transmit String length**: If no compression or MNP5 is selected this field is not included in the message. If compression selected is V.42 *bis* or V.44, this indicates the (transmit for V.44) string length. Valid values are 6 to 250 for V.42 *bis* and 32 to 255 for V.44.
104:111	**Compression Receive String length**: If no compression, MNP5 is selected this field is not included in the message. If compression selected is V.42 *bis* this field is set to a value of zero. If compression selected is V.44 this indicates the receiver string length. Valid values are 32 to 255 (0 if V.42 *bis*).
112:127	**Compression Transmit History size**: If no compression, MNP5 or V.42 *bis* is selected this field is not included in the message. If compression selected is V.44 this indicates the transmitter history size. Valid values are from 512 to 65535.

Table 25/V.150.1 – CONNECT definitions

CONNECT bits MSB:LSB	Definition
128:143	**Compression Receive History size**: If no compression, MNP5 or V.42 *bis* is selected this field is not included in the message. If compression selected is V.44, this indicates the receiver history size. Valid values are from 512 to 65535.
NOTE – The compression parameters are collectively known as the Negotiated Compression Parameter (NCP) set. This includes the selected compression, compression direction, the dictionary, string length and history sizes.	

15.4.6 Break message (BREAK)

This message is used by a gateway to notify its peer gateway that it has received a break signal from its local DCE.

Table 26/V.150.1 – Break message format

BREAK bits MSB:LSB	Definition
0:3	**Break Source Protocol**: This four-bit field indicates the protocol that is generating the break: 0 – V.42/LAPM; 1 – Annex A/V.42 (1996); 2 – V.14; 3 to 15 – Reserved.
4:7	**Break Type**: This four-bit field indicates the type of break that is being generated: 0 – Not Applicable; 1 – Destructive and Expedited; 2 – Non-destructive and Expedited; 3 – Non-destructive and non-expedited; 4 to 15 Reserved
8:15	**Break Length**: This eight-bit field is used to indicate the duration of the break if applicable. The field is encoded in units of 10 ms. The value of "11111111" shall be used to indicate a break longer than 2.54 seconds. The absence of a break length field or a value of zero in the break length field in a received BRK message shall be interpreted as a break of default length.

15.4.7 Break acknowledgment message (BREAKACK)

This message is used by gateways to indicate a break acknowledgement. This message does not have any additional informational content.

15.4.8 Modem relay event (MR_EVENT)

The MR_EVENT message indicates the detection of local events during Modem Relay. These events include changes to the modem's physical layer state, such as retrains and rate renegotiations. Also included is an indication of the modem's physical layer parameters when it enters or returns to data mode.

Table 27/V.150.1 – MR_EVENT definitions

MR_EVENT bits MSB:LSB	Definition
0:7	**Event ID**: This field identifies the event being indicated. The following values are defined: 0: NULL 1: Rate Renegotiation 2: Retrain 3: Physical Layer Ready (PHYSUP) 4…255: Reserved for ITU-T; these values are not interpreted by a receiver.
8:15	**Reason Code**: For event ID values of 1 and 2, the following values are defined: 0: Null – no meaning or not applicable. 1: Initiation – Gateway initiated the retrain or rate renegotiation event. 2: Responding – Gateway is responding to a retrain or rate renegotiation event. If an Event ID of 3 is used, then this field is set Null and bits 16 to 71 are used. For any other event ID bits 16 through 71 are not transmitted.
16:21	**Selected Modulation**: If event ID is 3 then this field indicates the modulation as being used on the local PSTN link. The values of this field are the same as defined for CONNECT (SELMOD). (See Table 25)
22	**Transmit Symbol Rate Enable (TxSEN)**: Used if event ID has a value of 3. If set to a value of 1, this parameter indicates that the optional transmit symbol rate field (TxSR) is being used in the message.
23	**Receiver Symbol Rate Enable (RxSEN)**: Used if event ID has a value of 3. If set to a value of 1, this parameter indicates that the optional receive symbol rate field (RxSR) is being used in the message.
24:39	**Transmit Data Signalling Rate (TDSR)**: Used if event ID has a value of 3. The locally selected transmitter data signalling rate in bit/s. (0…65535). This is the same definition as used in CONNECT (TDSR), (see Table 25).
40:55	**Receive Data Signalling Rate (RDSR)**: Used if event ID has a value of 3. The locally selected receiver data signalling rate in bit/s. (0…65535). This is the same definition as used in CONNECT (RDSR), (see Table 25).

Table 27/V.150.1 – MR_EVENT definitions

MR_EVENT bits MSB:LSB	Definition
56:63	**Physical Layer Transmitter Symbol Rate (TxSR)**: Used if event ID has a value of 3. This optional field indicates the gateway physical layer transmitter symbol rate. The values are only valid if the TxSEN bit is set. The values in symbols/s for this field are: 0: Null (i.e., when a symbol rate is not applicable, e.g., V.21) 1: 600 2: 1200 3: 1600 4: 2400 5: 2743 6: 3000 7: 3200 8: 3429 9: 8000 10…254: Reserved for ITU-T; these values are not interpreted by a receiver 255: Unspecified.
64:71	**Physical Layer Receiver Symbol Rate (RxSR)**: Used if event ID has a value of 3. This optional field indicates the gateway physical layer receiver symbol rate. The values are only valid if the RxSEN bit is set. This field uses the same set of values as described for TxSR above.

15.4.9 Clear down indication (CLRDOWN)

This message may be sent to a remote gateway to notify it of a local clear down event.

Table 28/V.150.1 – CLRDOWN definition

CLRDOWN bits MSB:LSB	Definition
0:7	**Reason Code (Note)**: This field indicates the eight-bit code used to identify the clear down event. The codes defined are: 0: Unknown/unspecified 1: Physical Layer Release (i.e., data pump release) 2: Link Layer Disconnect (i.e., receiving a V.42 DISC frame) 3: Data compression disconnect 4: Abort (i.e., termination due to Abort procedure as specified in SDL) 5: On-hook (i.e., when gateway receives On-hook signal from an end-point device) 6: Network layer termination 7: Administrative (i.e., operator action at gateway).
8:15	This field contains the eight-bit vendor-Tag as defined in clause 8/V.150.0.
16:23	This field provides an eight-bit field which may be used to provide additional vendor specific information regarding the clear down.
NOTE – The clear down reason codes used in this message are the same as used in the SSE clear down indication.	

15.4.10 XID profile exchange for MR1 (PROF_XCHG)

This message defines the format of the optional XID profile exchange procedures.

Table 29/V.150.1 – PROF_XCHG definition

PROF_XCHG bits MSB:LSB	Definition
0:1	V.42/LAPM protocol support (0 = No, 1 = Yes, 2 = unknown) (Note 1)
2:3	Annex A/V.42 (1996) protocol support (0 = No, 1 = Yes, 2 = unknown)
4:5	V.44 compression support (0 = No, 1 = Yes, 2 = unknown) (Note 2)
6:7	V.42 *bis* compression support (0 = No, 1 = Yes, 2 = unknown)
8:9	MNP5 compression support (0 = No, 1 = Yes, 2 = unknown)
10:15	Reserved for ITU-T; these bits are set to zero by the transmitter and not interpreted by the receiver.
16:23	**XID/LR field Octet 2**: Indicates V.42 *bis* data compression request (P0)
24:39	**XID/LR field Octet 3 and 4**: V.42 *bis* number of codewords (P1)
40:47	**XID/LR field Octet 5**: V.42 *bis* maximum string length (P2)
48:55	**XID/LR field Octet 6**: V.44 capability (C0)
56:63	**XID/LR field Octet 7**: V.44 data compression request (P0)
64:79	**XID/LR field Octet 8 and 9**: V.44 number of codewords in transmit direction (P1T)
80:95	**XID/LR field Octet 10 and 11**: V.44 number of codewords in receive direction (P1R)
96:103	**XID/LR field Octet 12**: V.44 maximum string length in transmit direction (P2T)
104:111	**XID/LR field Octet 13**: V.44 maximum string length in receive direction (P2R)
112:127	**XID/LR field Octet 14 and 15**: V.44 length of history in transmit direction (P3T)
128:143	**XID/LR field Octet 16 and 17**: V.44 length of history in receive direction (P3R)

NOTE 1 – If protocol support = Yes, the modem will attempt/accept the use of this protocol during V.42 detection phase (which includes attempting/accepting the "alternate" – Annex A protocol). If protocol support = No, the modem explicitly does not ever attempt/accept this protocol, and will be unable to operate in this mode. If protocol support = unknown, the gateway has insufficient knowledge to make a prediction.

NOTE 2 – If compression support = Yes, an originating modem will propose this compression method during XID/LR negotiation. A terminating modem will choose one such proposed method during negotiation, if the terminating modem itself supports that method. If several acceptable compressions were proposed, a terminating modem must choose only one, and will "prefer" V.44 to V.42 *bis* (LAPM), or prefer V.42 *bis* to MNP5 – if none of the proposed compressions are acceptable, No Compression is chosen.

If all fields in b0 to b15 are "unknown", the PROF_XCHG message need not be sent, since the non-reception of this message indicates the gateway's lack of knowledge.

If compression support = No, the modem will not propose this method (if originating), and will never make this choice (if terminating). Note that LAPM supports only (V.44, V.42 *bis*) and Annex A/V.42 (1996) supports only (V.42 *bis*, MNP5), regardless of noted compression support; no compression is supported if there is no protocol.

If compression support = Unknown, the gateway has insufficient knowledge to make a prediction.

If all the protocols that support a compression method are not supported, then the compression method is not supported and should be notated as such. If all the protocols which support a compression have unknown support, then the compression's support is unknown and again should be notated as such. Finally, repeating from above, if all protocols and compressions have unknown support, the PROF_XCHG message need not be sent.

If a compression method is not supported or the support is unknown, the corresponding Compression parameter fields must be set to zero. However, it is permissible not to send trailing unsupported/unknown compression fields.

15.4.11 User data (INFO messages)

This clause defines the user-data message formats used for Modem Relay.

15.4.11.1 Octet without formatting (I_OCTET)

If the CONNECT message indicates not to use a DLCI then the format of this message is shown in Figure 14.

r	msgID	Octets

Figure 14/V.150.1 – Octet without formatting message format, without DLCI

If the CONNECT message indicates the use of a DLCI then the format of the message is shown in Figure 15.

r	msgID	DLCI	Octets

Figure 15/V.150.1 – Octet without formatting message format, with DLCI

In these messages:

r — is a 1-bit field reserved for future use by the ITU-T. Message senders shall set this field to 0. Message receivers shall ignore the value of this field.

MsgID — is a 7-bit field whose value identifies the message; unique values are assigned to the octet compressed raw data message and the bit compressed raw data message.

DLCI — is an eight- or sixteen-bit field containing the DLCI. The same formatting as defined in ITU-T Rec. V.42 is used to encode this field.

Octets — is a sequence of octet user-data.

If this data type is to be used for start-stop character data, then the mapping rules as defined in Annex B/V.42 shall be used. (See 15.4.11.5.1.)

15.4.11.2 Message common format for raw compressed data types

The format of these messages is given in Figure 16.

r	msgID	segment	...	segment

Figure 16/V.150.1 – Octet and bit compressed raw data message common format

In these messages:

r is a 1-bit field reserved for future use by the ITU-T. Message senders shall set this field to 0. Message receivers shall ignore the value of this field.

MsgID is a 7-bit field whose value identifies the message; unique values are assigned to the octet compressed raw data message and the bit compressed raw data message.

Segment is an octet or bit compressed data segment.

Each message consists of the reserved field, the message ID field and one or more segments. The reserved and message ID field together comprise one octet. Each segment contains an integral number of octets. The length of the message, exclusive of transport protocol headers, is the sum of the lengths of the segments plus one.

If the message is an octet compressed or bit compressed raw data message, each segment in the message shall be an octet compressed data segment or a bit compressed data segment, respectively.

The data represented by a raw data message shall consist of the concatenation of the data represented by the compressed data segments of the message, in the order that the segments appear in the message.

The data represented by an octet compressed raw data message is an integral number of octets. The data represented by a bit compressed raw data message may not be an integral number of octets.

15.4.11.3 Octet compressed data segment (I_RAW-OCTET)

The formats of an octet compressed data segment are given in Figure 17.

0	L	octet	...	octet

1	L	N	octet	...	octet

Figure 17/V.150.1 – Octet compressed data segment format

In these segments:

L is a 7-bit field whose value is one less than the number of octets in the segment.

N is a 1-octet field whose value is two less than the number of times the octets of the segment appear in the data represented by the message in which the segment appears.

Octet is an octet of octet compressed data.

The data represented by the first format of an octet compressed data segment is the data in the octets contained in the segment. The represented data is an integral number of octets in length.

The data represented by the second format of an octet compressed data segment is the data in the octets contained in the segment, repeated the indicated number of times. The represented data is an integral number of octets in length.

15.4.11.4 Bit compressed data segment (I_RAW-BIT)

The formats of a bit compressed data segment are given in Figure 18.

00	L	octet	...	octet

01	L	P	octet	...	octet

1	L	P	N	octet	...	octet

Figure 18/V.150.1 – Bit compressed data segment format

In these segments:

L — is a 6-, 3-, or 4-bit field whose value is one less than the number of octets in the segment.

P — is a 3-bit field whose value is the number of low order bits in the last octet that are NOT in the data represented by this segment.

N — is a 1-octet field whose value is two less than the number of times the octets of the segment appear in the data represented by the message in which the segment appears.

Octet — is an octet of bit compressed data.

The data represented by the first format of a bit compressed data segment is the data in the octets contained in the segment. The represented data is an integral number of octets in length.

The data represented by the second and third formats of a bit compressed data segment is the data in the octets contained in the segment, less the indicated number of low order bits in the last octet, repeated the indicated number of times. The represented data is not necessarily an integral number of octets in length.

15.4.11.5 Character with Static format (I_CHAR-STAT)

The format of this data type message is given in Figure 19.

r	msgID	r	D	P	S	Data in octet format

Figure 19/V.150.1 – Character with static format

In this message:

r — is a 1-bit field reserved for future use by the ITU-T. Message senders shall set this field to 0. Message receivers shall ignore the value of this field.

MsgID — is a 7-bit field whose value identifies the message; a unique value is assigned to the I_CHAR data type.

D — is a 2-bit field indicating the number of data bits:

0-5 bits; 1-6 bits; 2-7 bits; 3-8 bits.

P — is a 3-bit field indicating the parity type:

0 – unknown; 1 – none; 2 – even parity; 3 – odd parity; 4 – space parity; 5 – mark parity; 6 – reserved; 7 – reserved.

S is a 2-bit field indicating the number of stop bits:

0-1 stop bit; 1-2 stop bits; 2 – reserved; 3 – reserved.

Octets is an integer number of octets representing start-stop characters.

The format of the Start-Stop characters is the same as defined in Annex B/V.42. For convenience, it is reproduced here.

15.4.11.5.1 Mapping format from Annex B/V.42

This is the mapping for converting between character formats used on the DTE/DCE interface and those used on the control function/error control function interface. Only support of the 10-bit DTE-to-DCE format is mandatory; support of the other formats shown here is optional. Character formats other than those listed below are not supported.

Table 30/V.150.1 – Character format

DTE/DCE: Total bits per character	Specific formats of octets supported	Control function to error control function formatting of octets
11	Start/8 data/2 stop Start/8 data/parity/stop	8 data (parity or second stop bit is independently generated on each DTE/DCE interface)
10	Start/8 data/stop	8 data
	Start/7 data/2 stop	7 data plus 0-bit pad in high-order bit
	Start/7 data/parity/stop	7 data plus parity as high-order bit
9	Start/7 data/stop	7 data plus 0-bit pad in high-order bit
	Start/6 data/2 stop	6 data plus two 0-bit pads in two highest-order bits
	Start/6 data/parity/stop	6 data plus parity in next-to-high-order bit plus 0-bit pad in high-order bit
8	Start/6 data/stop	6 data plus 0-bit pad in two highest-order bits
	Start/5 data/2 stop	5 data plus three 0-bit pads in three highest-order bits
	Start/5 data/parity/stop	5 data plus parity in third highest-order bit plus two 0-bit pads in two highest-order bits

This data type does not allow a change of format. The character format is static for the remainder of the MR session.

15.4.11.6 Character with dynamic format (I_CHAR-DYN)

The format of this data type message is the same as for Character with Static format.

This data type allows a change of format. The character format may be changed dynamically during a MR session.

15.4.11.7 Framed data format (I_FRAME)

The format of this data type message is given in Figure 20.

r	msgID	res	Fr	octets

Figure 20/V.150.1 – Framed data type message format

In this message:

r	is a 1-bit field reserved for future use by the ITU-T. Message senders shall set this field to 0. Message receivers shall ignore the value of this field.
MsgID	is a 7-bit field whose value identifies the message; a unique value is assigned to the Framed data type.
res	is a 6-bit field reserved for future use by the ITU-T. Message senders shall set this field to 0. Message receivers shall ignore the value of this field.
Fr	is 2-bit field indicating data frame state: 0 – data frame without termination; 1 – data frame with termination; 2 – data frame with abort termination.
Octets	is an integer number of octets representing the body of framed data frame.

Each message consists of the r field, the message ID field, res field, Fr field, and Octets field. The r field and message ID field together comprise one octet. The res field and Fr field comprise one octet. Octets field contains an integer number of octets.

15.4.12 Vendor-specific messages (VENDOR)

Vendor-specific IP-TLP messages shall begin with an eight-bit vendor-Tag as defined in clause 8/V.150.0. This is followed by the user-defined data. The length of this message is variable.

16 Start-up mode of operation

MoIP gateways will need to co-exist with other "over-IP" mechanisms. (For example Voice-over-IP and Facsimile-over-IP). Depending upon the set of supported modes a MoIP gateway shall start up as indicated in Table 31.

Table 31/V.150.1 – MoIP initial modes

Additional supported modes by MoIP gateway		Start as
Facsimile-over-IP (T.38)	Voice-over-IP	
No	No	Modem-over-IP
No	Yes	Voice-over-IP
Yes	No	Modem-over-IP
Yes	Yes	Voice-over-IP

17 Facsimile interworking requirements

The support and interworking of FoIP gateways with MoIP gateways is for further study.

18 Text telephony interworking requirements

The support and interworking of ToIP gateways with MoIP gateways is for further study.

19 Call set up procedures

The call set up procedures are defined in Annexes E and F and Annex P/H.323.

20 Call discrimination procedures

The following clause defines the procedures to be used by a MoIP gateway during the call discrimination phase of the connection set up.

20.1 V.25 calling tone, V.8 CI processing and bell-type modem answer tone

V.25 Calling tone, V.8 CI and the Answer Tone for Bell type modems (see Appendix VI) are not explicitly supported by this Recommendation. Future support is for further study.

20.2 V.8 *bis* processing

Gateways shall monitor and detect V.8 *bis* dual tone on the PSTN link and prevent further V.8 *bis* signals from being transmitted into the IP network.

20.3 Call discrimination procedure/answer tone treatment selection

During the gateway capability and call set up phase, the gateways indicate to each other their preference for the call discrimination procedure/answer tone treatment in the CDSCselect signalling parameter. The selection of the call discrimination procedure/answer tone treatment that will be used for the MoIP session is determined by Table 32.

Table 32/V.150.1 – Gateways call discrimination/answer tone treatment selection

Local *CDSCselect* preference	Remote *CDSCselect* preference	Selected call discrimination/ Answer tone treatment
Audio	–	Audio
–	Audio	Audio
Mixed	VBD-select or Mixed	Mixed
VBD-select or Mixed	Mixed	Mixed
VBD-select	VBD-select	VBD-select

20.4 Answer tone processing

The proper handling of DCE Answer Tone signals is necessary to support the call discrimination process. The Answer Tone shall be transported according to the procedures specified in this clause. The transport of Answer Tones encoded with codecs that are not optimal for modem signals is not supported by this Recommendation. Gateways shall preserve the characteristics of an Answer Tone from an answering end-point DCE while transporting the signal across an IP network. Gateways, when regenerating Answer Tone, shall preserve its type. Phase reversal characteristic regeneration is optional but should be maintained. MoIP gateways shall support this IP transport mechanism in two ways. The first is RFC 2833 encoded and the second is VBD encoded.

It shall be mandatory for MoIP gateways to indicate through external signalling the capability to support both RFC 2833 and VBD Answer Tone processing. RFC 2833 Answer Tone Events processing is required and may be used at any point during a call. VBD encoded Answer Tone processing does require a successful negotiation as defined in Table 32 which requires both gateways consent.

A gateway that prefers RFC 2833 encoding of Answer Tone will generate the RFC 2833 messages upon detection of a valid Answer Tone. Gateways receiving this message shall regenerate the appropriate Answer Tone signal. In VBD mode, VBD can be used to transport the Answer Tone signals using a codec identified for use in VBD. RFC 2833 encoding of Answer Tone in VBD mode is also allowed.

Upon the Off-ramp gateways transition to VBD mode and prior to reception of any confirmation by the *On-ramp* gateway that it has also transitioned to VBD, VBD encoded Answer Tone signals shall be immediately transmitted across the packet network (if not RFC 2833 encoded). A gateway shall discard any VBD packets it receives while in Audio Mode.

20.4.1 Answer tone treatment selection

The type of answer tone treatment (Audio, VBD-Select, or Mixed) is negotiated between gateways using the CDSCselect signalling parameter as described in Table 32.

20.4.1.1 Audio (RFC 2833) answer tone treatment

In this mode of operation, the gateways shall use RFC 2833 encoding of Answer Tone as described in 20.4.3.

20.4.1.2 VBD-select answer tone treatment

When VBD-Select procedure has been negotiated, a VBD-Select gateway will usually transition to VBD mode upon answer tone detection. The gateways may use RFC 2833 if transitioning into VBD mode. The gateway will use RFC 2833 if not transitioning into VBD mode.

20.4.1.3 Mixed answer tone treatment

In this mode, the Off-ramp gateway shall use RFC 2833 encoding of Answer Tone if it detects ANSam or if both of the gateways are U-MR types. If either of the gateways is a V-MR type, and V.25 ANS is detected, then G2 shall initiate using VBD encoding.

20.4.2 Initial answer tone processing

Upon initial detection of an answer tone, the detecting gateway shall prevent the transport of this signal encoded in audio across the network within 50 ms. The gateway may suppress the Answer Tone signal or switch to VBD. The amount of answer tone sent in the audio encoding shall not exceed 50 ms for the entire duration of the signal.

20.4.3 RFC 2833 encoded answer tone

While suppressing the Answer Tone, a gateway shall verify the type of answer tone (ANS or ANSam). The maximum time from the detection of Answer tone until the type has been verified is 350 ms (or 400 ms after initiation of the Answer Tone). When the gateway determines the Answer Tone type it shall transmit the appropriate RFC 2833 event to the remote gateway. Answer Tone shall continue to be blocked during this period, per RFC 2833.

NOTE – The method of blocking the 2100 Hz tone in audio packets is beyond the scope of this Recommendation, but the gateway should continue to generate audio packets to prevent any interpolation of 2100 Hz tone by the other gateway, due to apparent loss of packets.

When regenerating the answer tone, a gateway shall preserve the same answer tone type.

During the regeneration of the Answer Tone, the gateway receiving the answer tone from the remote modem shall monitor for phase reversal events. These events are sent to the remote gateway again using the appropriate RFC 2833 event. The RFC 2833 event may be either of definite (i.e., normal codec frame duration) or indefinite (i.e., maximum) duration. The gateway shall generate another RFC 2833 event at the end of the specified duration. The number of duplicate RFC 2833 events generated by the gateway before the specified duration (e.g., for packet transmission reliability) is beyond the scope of this Recommendation.

Upon detection of phase reversals in the Answer Tone, the gateway shall switch to generating /ANSam or /ANS, respectively, RFC 2833 events, as specified above. The reception of a RFC 2833 /ANS and /ANSam event means that the On-Ramp gateway shall send a phase reversal immediately and every 450 ms thereafter (per the phase reversal requirements of V.25 and V.8) until a different message is received. Only the first instance of a message shall be used. At the completion of the detection of Answer Tone, the gateway shall generate one or more short duration RFC 2833 events with the end-of-event marker set.

20.4.4 VBD encoded answer tone

In this method, both gateways have transitioned to VBD mode and the channel is configured into VBD as defined in clause 8, wherein the Answer Tone may be transported and regenerated using RTP packets across the network. RFC 2833 encoding of Answer Tone signals in VBD mode shall be optional at the transmitter, but mandatory for the receiver.

If already started prior to the transition to VBD encoding of Answer Tone from audio, RFC 2833 Answer Tone encoding shall continue until the end of the Answer Tone, or until switching to modem relay mode.

20.4.5 Call discrimination modes

The following clauses describe the procedures and modes to be used for call discrimination as they relate to the negotiation of call discrimination procedure/answer tone treatment as described in Table 32.

20.4.6 Audio call discrimination mode

In this mode of operation, the gateways shall remain in audio mode upon answer tone detection.

20.4.7 VBD-select call discrimination mode

For this call discrimination mode, the Off-ramp gateway (G2) decides whether to remain in Audio mode, or to transition to VBD upon answer tone detection. The gateways shall use RFC 2833 encoding of answer tone if remaining in audio mode, the gateways may use RFC 2833 if transitioning into VBD mode.

When VBD-Select procedure has been negotiated and the VBD-Select gateway does not transition to VBD mode upon answer tone detection (i.e., uses RFC 2833 encoding in Audio mode), running the call discrimination time-out timer is not required on the peer gateway.

When a VBD-Select gateway has negotiated audio or mixed procedures, running the call discrimination time-out timer is required on the peer gateway when not in VBD mode.

20.4.8 Mixed call discrimination mode

In this mode, the Off-ramp gateway shall remain in audio mode if it detects ANSam, or if both of the gateways are U-MR types. If either of the gateways is a V-MR type and V.25 ANS is detected, then G2 shall transition to VBD mode. The gateways shall use RFC 2833 encoding of answer tone if remaining in audio mode; they may use RFC 2833 if transitioning into VBD mode.

20.5 Modem-over-IP mode selection procedures

For the case where the end-point modems M1 and M2 use V.8 procedures to determine the modulation to be used, the set of modulations supported by each entity in the link (M1, G1, G2 and M2) introduces a dependency upon which Modem-over-IP mode should be used, be it Modem Relay or VBD.

For example, gateways may support the entire set of modulations available to the end-point modems, in which case Modem Relay can be used. If, however, the gateways do not support the entire set, and the modulation that would be selected by M1 and M2 is one of the non-supported sets, then VBD may be used.

The following are a set of procedures which determine which MoIP mode should be used depending upon the modulation capabilities of M1, G1, G2 and M2, as negotiated by their V.8 procedures.

Let G be the set of common modulations between gateways G1 and G2. These are exchanged by the gateways to each other during the call set-up messaging phase (see 15.2.2.1). M_{G1} and M_{G2} are the set of modulations supported by gateways G1 and G2, respectively. M_{CM1} and M_{JM2} are the set of modulations as indicated in the V.8 CM sequence of M1 and the JM sequence of M2.

$$\text{i.e., } G = \left(M_{G1} \cap M_{G2}\right)$$

For these procedures there are three cases to consider.

20.5.1 Disjoint case

For all $x \in M_{CM1}, x \notin G$

I.e., there is no common modulation set between M1 and the gateways.

For this disjoint case the gateways shall use VBD mode.

20.5.2 Subset case

For the case where $M_{CM1} \subseteq G$

I.e., the modulation set being indicated by M1 is a subset of the gateways capability.

For the subset, the gateways shall use Modem Relay mode.

20.5.3 Intersection case

There are two situations to be considered for the intersection case. The first is if there is a common set of modulation capabilities between the gateways and M1, and the second is if M1 is not a subset of the combined gateways modulation capabilities.

Condition A: $M_{CM1} \cap G \neq 0$

Condition B: $M_{CM1} \not\subset G$

Upon reception of CM from M1 for both these conditions, G1 may choose to send either an SSE:V (if not already in VBD) or a SSE:M. If G1 sends a SSE:M, then it shall not generate a JM to M1 until it receives a JM_INFO IP-TLP message or a V.8 timeout occurs at G1.

Upon reception of a SSE:M for both these conditions, G2 may respond immediately with a SSE of its choice, or optionally invoke the CM-JM procedures as described in 20.6.

20.6 CM-JM procedures for determining MoIP mode

The following are the procedures to be used when using the M2 JM sequence in the determination of MoIP mode.

20.6.1 The contents of the CM received by G1 from M1 shall not be modified by G1.

20.6.2 G1 may request a switch to Modem Relay to initiate the CM-JM procedures. G1 shall continue to generate Answer Tone, until it is no longer in Modem Relay mode. For example if G2 makes a request to switch to VBD or after receiving a JM content message.

20.6.3 G2 shall generate a CM sequence (M_{CM2}) to be transmitted to M2 that meets the following condition.

$$M_{CM1} \subseteq M_{CM2} \subseteq \left(M_{CM1} \cup M_{G2}\right)$$

NOTE 1 – This allows the gateways to connect at the highest common modulation between M1, G1, G2 and M2. It also facilitates the ability to insert a modulation not necessarily supported by M1 and G1 (e.g., V.91).

20.6.4 On receiving the JM sequence from M2 the following rules are applied by G2:

i) If $\left(M_{JM2} \subseteq M_{G2}\right)$ and $\left(M_{JM2} \not\subset M_{CM1}\right)$ or

$\left(M_{JM2} \subseteq M_{CM1}\right)$ and $\left(M_{JM2} \subseteq M_{G2}\right)$

then gateways shall select Modem Relay mode.

NOTE 2 – The first condition allows for asymmetric modulation selection and the second condition allows for a symmetric modulation.

ii) If $\left(M_{JM2} \subseteq M_{CM1}\right)$ and $\left(M_{JM2} \not\subset M_{G2}\right)$

then gateways shall select VBD mode.

NOTE 3 – This condition allows for the case where there is no common modulation between G2 and M2 even though there is a common selection between M1, G1 and G2.

20.7 JM delay procedure

The procedure defined in this clause is applicable to gateways when connecting in Modem Relay connection scenario MR1. It allows for the optional delay of transmission of JM by the On-Ramp gateway when using V.8 starting session procedures. This procedure can be used in conjunction with the N-TCX to N-TCX to support end-to-end XID exchanges as illustrated in the example shown in Figure 21.

The ability to support this procedure is indicated by both gateways during the Call Set-up capability exchange. If both gateways indicate the ability to support the JM-delay procedure, the following is specified:

20.7.1 Procedure for on-ramp gateway (G1)

G1 should wait to receive IP-TLP message *START_JM* from the off-ramp gateway G2 before proceeding to generate JM.

G1 has the ability to abort the delayed JM procedure due to V.8 time-outs or error events. In this case, G1 shall transmit JM and proceed with the V.8 procedure.

20.7.2 Procedure for off-ramp gateway (G2)

G2 shall send an IP-TLP START_JM message to G1. This may happen immediately, or after a period after the physical layer start-up of G2 has begun.

If a gateway is operating in a connection scenario MR1, the *START_JM* message may be generated. If a gateway is operating in a non-connection scenario MR1, received *START_JM* messages shall be ignored.

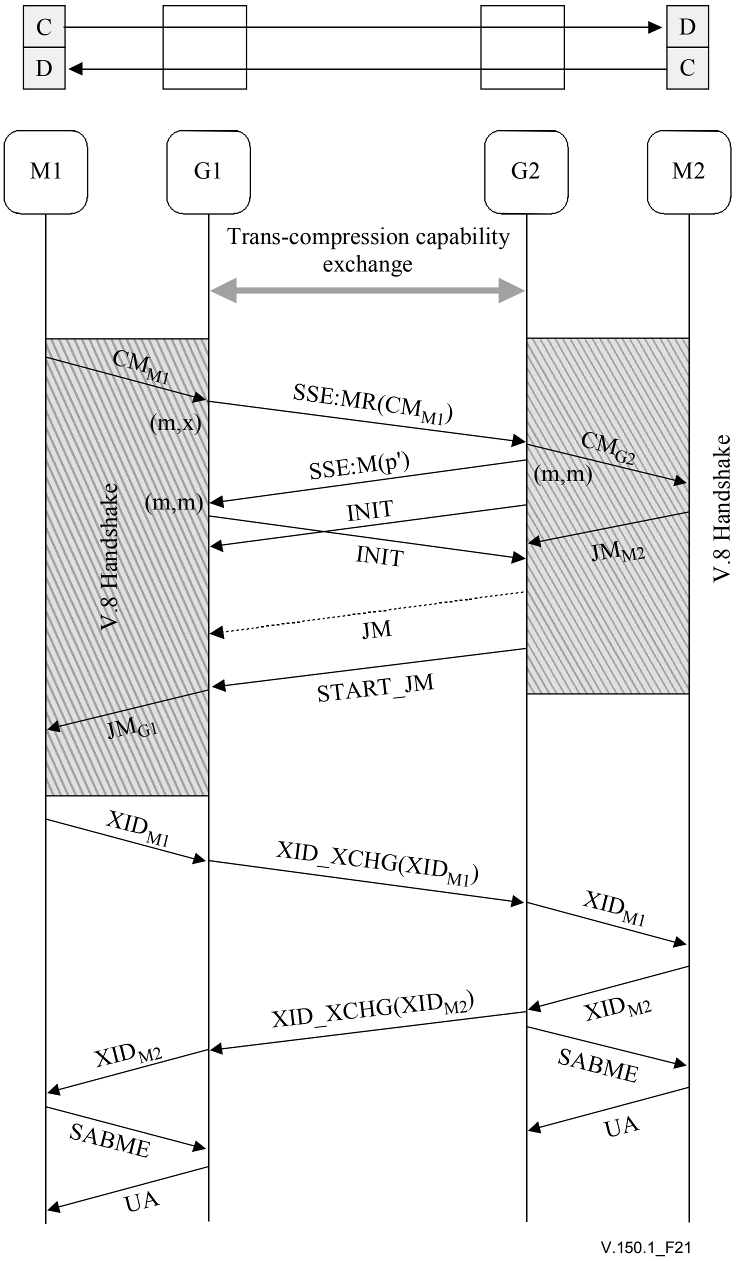


Figure 21/V.150.1 – Example delayed JM procedure for end-to-end XID exchange

20.8 Call discrimination SDL diagrams

This clause provides the SDL processing description for the procedures defined by the call discrimination call flows as described in Appendix II. Figure 22 gives the list of symbols used in the SDL definition process.

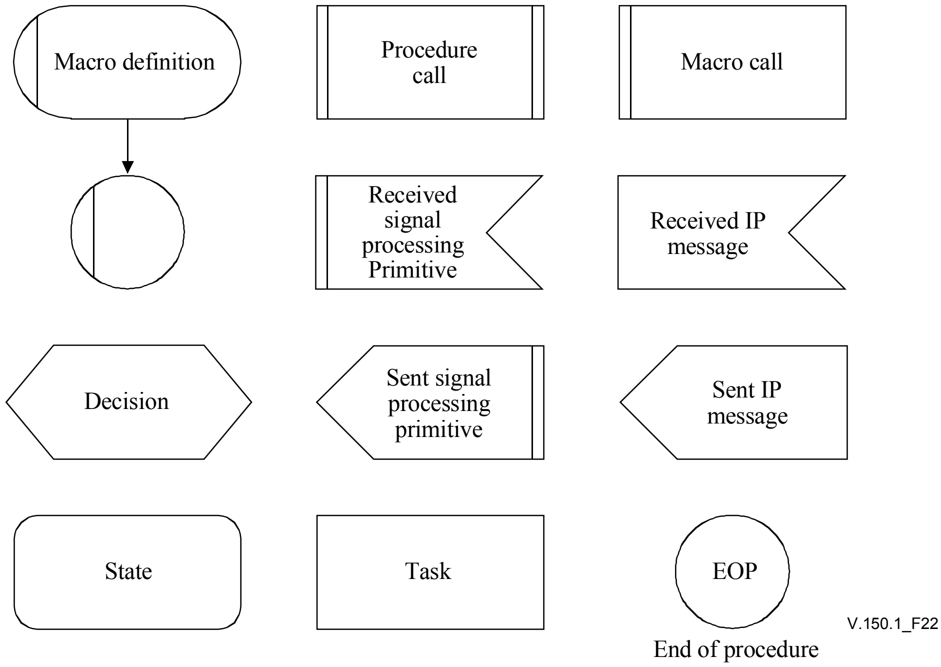


Figure 22/V.150.1 – SDL symbol definitions

20.8.1 System reference model

Figure 23 is the reference model that is used in conjunction with the SDL definitions. The signals to be considered are shown in Figure 24.

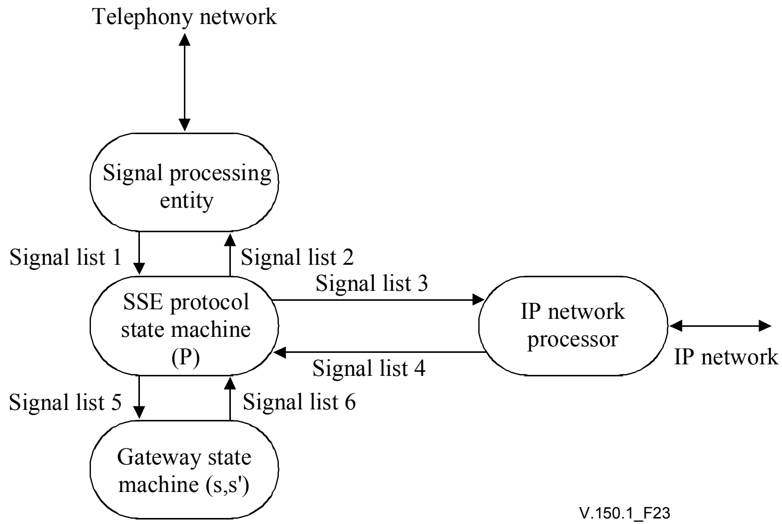


Figure 23/V.150.1 – SDL reference model

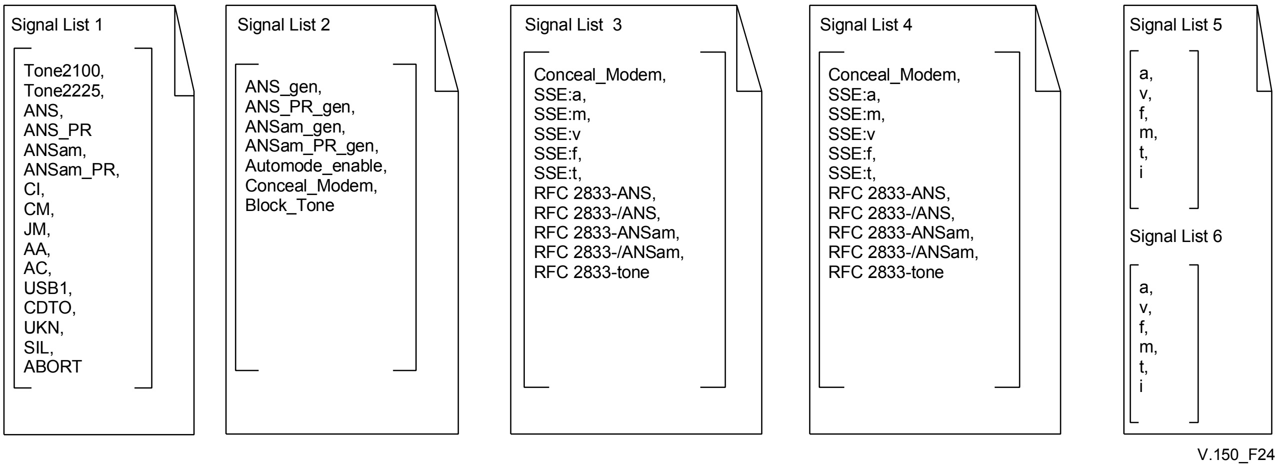


Figure 24/V.150.1 – Signal list definitions

20.8.2 Signal descriptions

The following tables provide the descriptions for each signal in the signal lists described in Figure 24 above.

Table 33/V.150.1 – Signal List 1: Primitive signals received from Signal Processing Entity (SPE)

Signal	Definition
Tone2100	SPE has detected 2100 Hz tone for a duration less than 50 ms
Tone2225	SPE has detected 2225 Hz tone for a duration less than 50 ms
ANS	SPE has verified presence of V.25 ANS type Answer Tone
ANS_PR	SPE has detected a 180-degree phase reversal in a verified ANS type Answer Tone
ANSam	SPE has verified presence of V.8 ANSam type Answer Tone
ANSam_PR	SPE has detected a 180-degree phase reversal in a verified ANSam type Answer Tone
CM	SPE has detected a V.8 CM signal
JM	SPE has detected a V.8 JM signal
AA	SPE has detected a V.32/V.32 *bis* AA signal
AC	SPE has detected a V.32/V.32 *bis* AC signal
USB1	SPE has detected a V.22 *bis* unscrambled binary one's signal
SB1	SPE has detected a V.22 *bis* scrambled binary one's signal
S1	SPE has detected a V.22 *bis* S1 signal
V.21 (low)	SPE has detected a V.21 low channel signal
V.21 (high)	SPE has detected a V.21 high channel signal
V.23 (low)	SPE has detected a V.23 low channel signal
V.23 (high)	SPE has detected a V.23 high channel signal
SIL	SPE has detected silence
CDTO (Note)	Call Discrimination Time-out
UKN (Note)	SPE has detected an unknown or unsupported signal
ABR	SPE has initiated an Abort Request
NOTE – These primitives initiate a transition to VBD when operating in (a,a) state. These primitives are required for gateways advertising MIXED and AUDIO procedures and are optional for gateways advertising VBD-preferred procedure.	

Table 34/V.150.1 – Signal List 2: Primitive signals sent to Signal Processing Entity (SPE)

Signal	Definition
ANS_gen	SPE requested to generate a V.25 ANS type Answer Tone signal
ANS_PR_gen	SPE requested to generate a V.25 ANS type Answer Tone signal with 180-degree phase reversals every 450 ms
ANSam_gen	SPE requested to generate a V.8 ANSam type Answer Tone signal
ANSam_PR_gen	SPE requested to generate a V.8 ANSam type Answer Tone signal with 180-degree phase reversals every 450 ms
ANS2225_gen	SPE requested to generate a 2225 Hz Tone
Automode_en	SPE requested to enable automode function
Block_Tone	SPE requested to block 2100 Hz tone
Conceal_Modem	SPE requested to prevent any modem signal to be output to the telephony side of the gateway

Table 35/V.150.1 – Signal List 3: Primitive signals sent to IP network controller

Signal	Definition
Conceal_Modem	Request to prevent any modem signal to be output to the telephony side of the gateway
SSE:a(RC)	Send SSE audio request with reason code RC
SSE:v(RC)	Send SSE VBD request with reason code RC
SSE:m(RC)	Send SSE modem relay request with reason code RC
RFC 2833-ANS	Send RFC 2833 ANS event
RFC 2833-/ANS	Send RFC 2833 ANS with phase reversals event
RFC 2833-ANSam	Send RFC 2833 ANSam event
RFC 2833-/ANSam	Send RFC 2833 ANSam with phase reversals event

Table 36/V.150.1 – Signal List 4: Primitive signals received from IP network controller

Signal	Definition
Conceal_Modem	Request to prevent any modem signal to be output to the telephony side of the gateway
SSE:a(RC)	A SSE audio request detected with reason code RC
SSE:v(RC)	A SSE VBD request detected with reason code RC
SSE:m(RC)	A SSE MR request detected with reason code RC
RFC 2833-ANS	A RFC 2833 ANS event detected
RFC 2833-/ANS	A RFC 2833 ANS with phase reversals event detected
RFC 2833-ANSam	A RFC 2833 ANSam event detected
RFC 2833-/ANSam	A RFC 2833 ANSam with phase reversals event detected

Table 37/V.150.1 – Signal List 5 and 6: Gateway state primitives

Signal	Definition
a	Audio State
v	VBD State
m	Modem Relay State
f	Facsimile Relay State
t	Text Relay State
i	Indeterminate State

20.8.3 SDL macro definitions

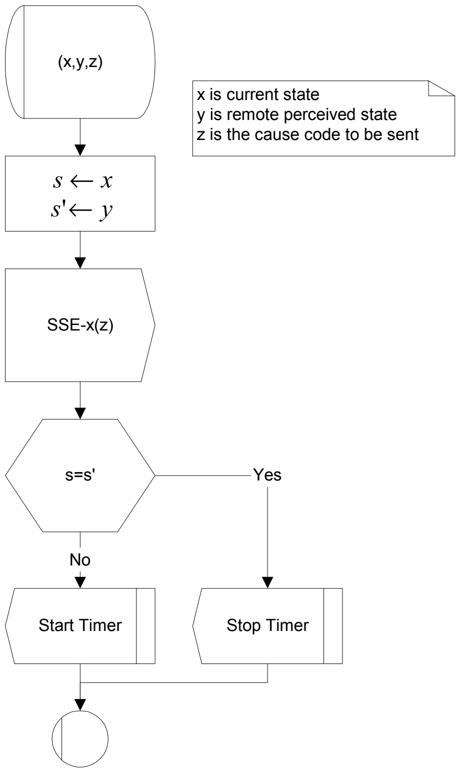


Figure 25/V.150.1 – Generic macro

20.8.4 SDL diagrams for state (a,a)

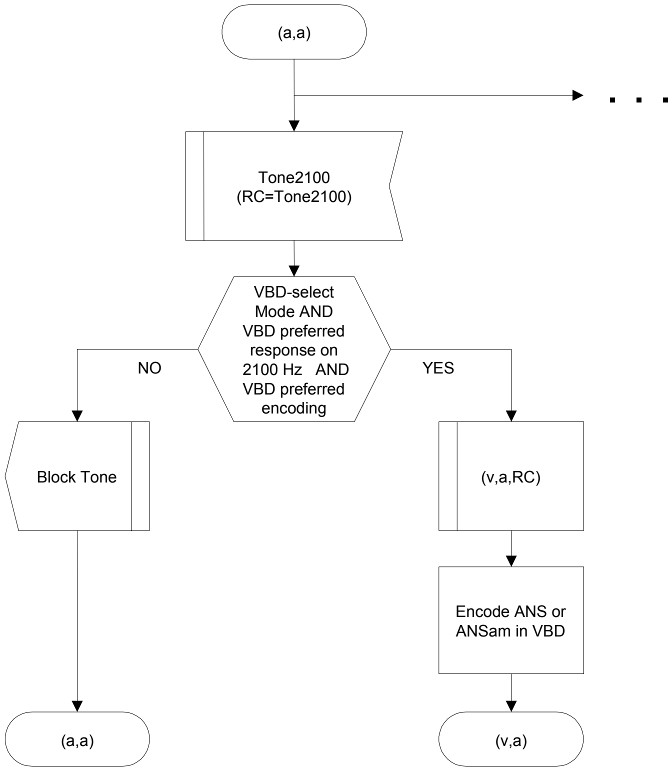


Figure 26/V.150.1 – SDL for state (a,a) part 1

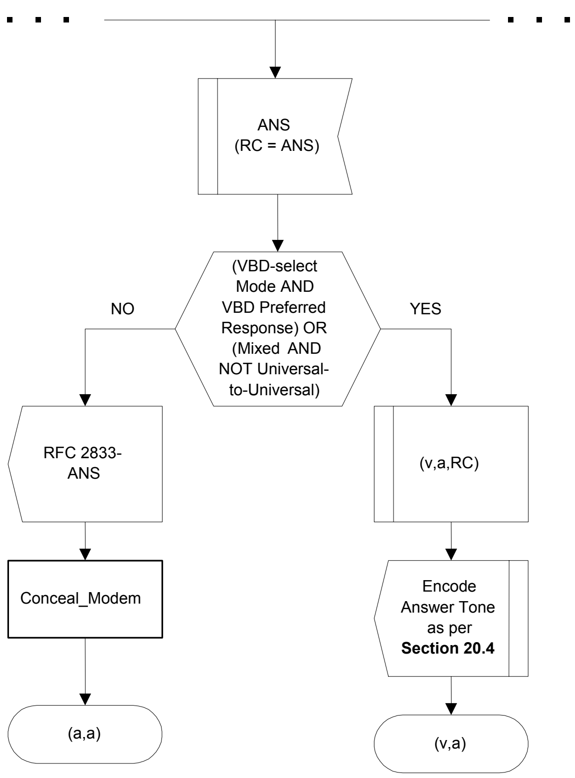


Figure 27/V.150.1 – SDL for state (a,a) part 2

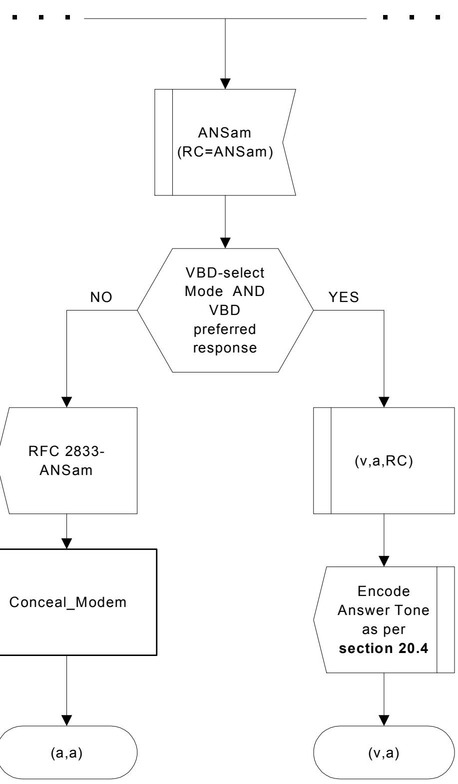


Figure 28/V.150.1 – SDL for state (a,a) part 3

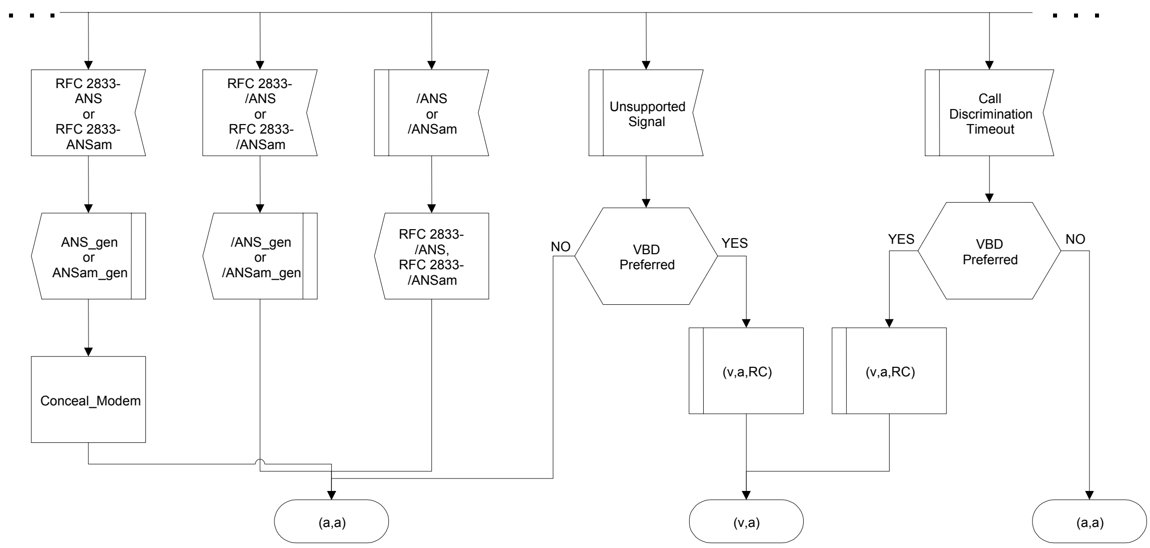


Figure 29/V.150.1 – SDL for state (a,a) part 4

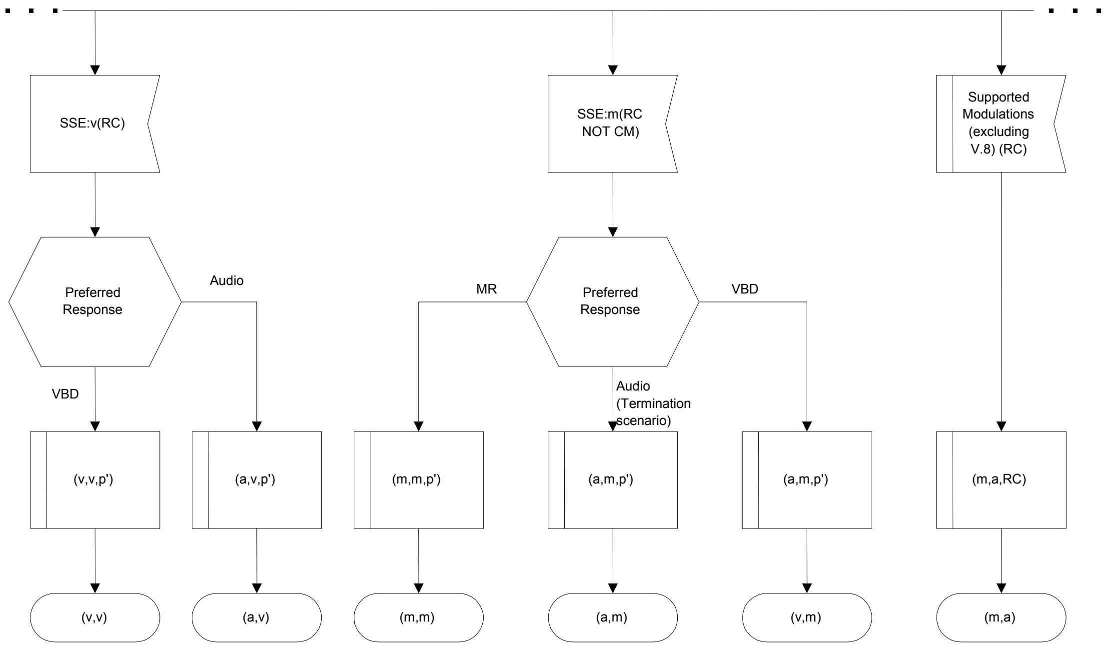


Figure 30/V.150.1 – SDL for state (a,a) part 5

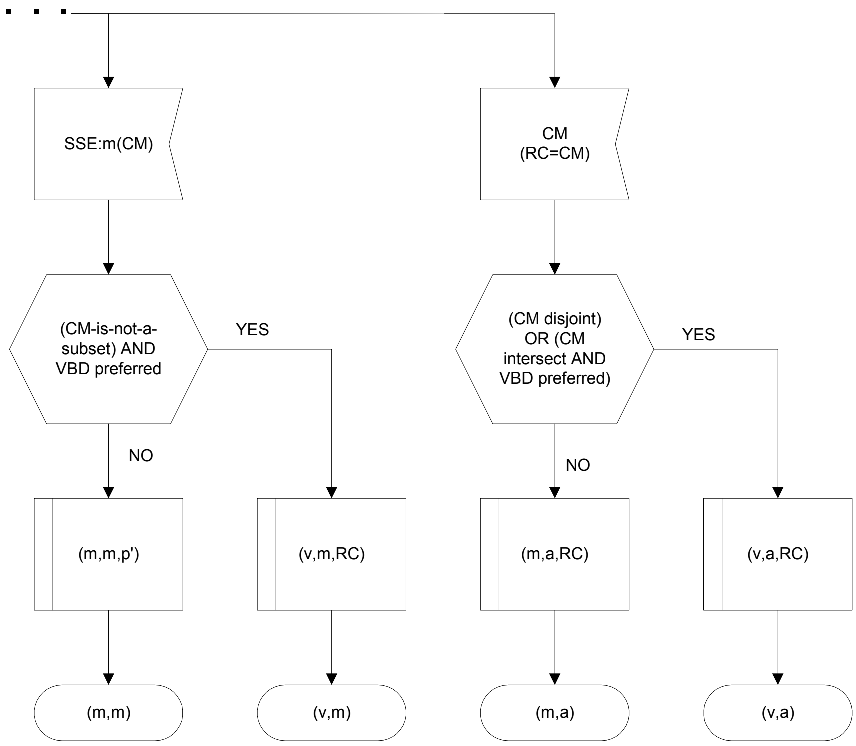


Figure 31/V.150.1 – SDL for state (a,a) part 6

20.8.5 SDL diagrams for state (a,m)

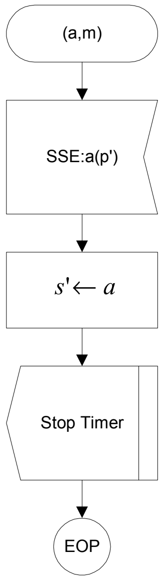


Figure 32/V.150.1 – SDL for state (a,m)

20.8.6 SDL diagrams for state (a,v)

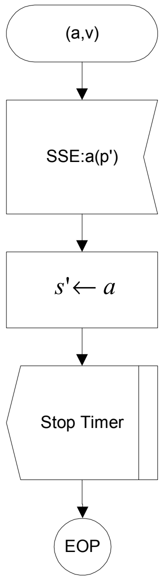


Figure 33/V.150.1 – SDL for state (a,v)

20.8.7 SDL diagrams for state (m,a)

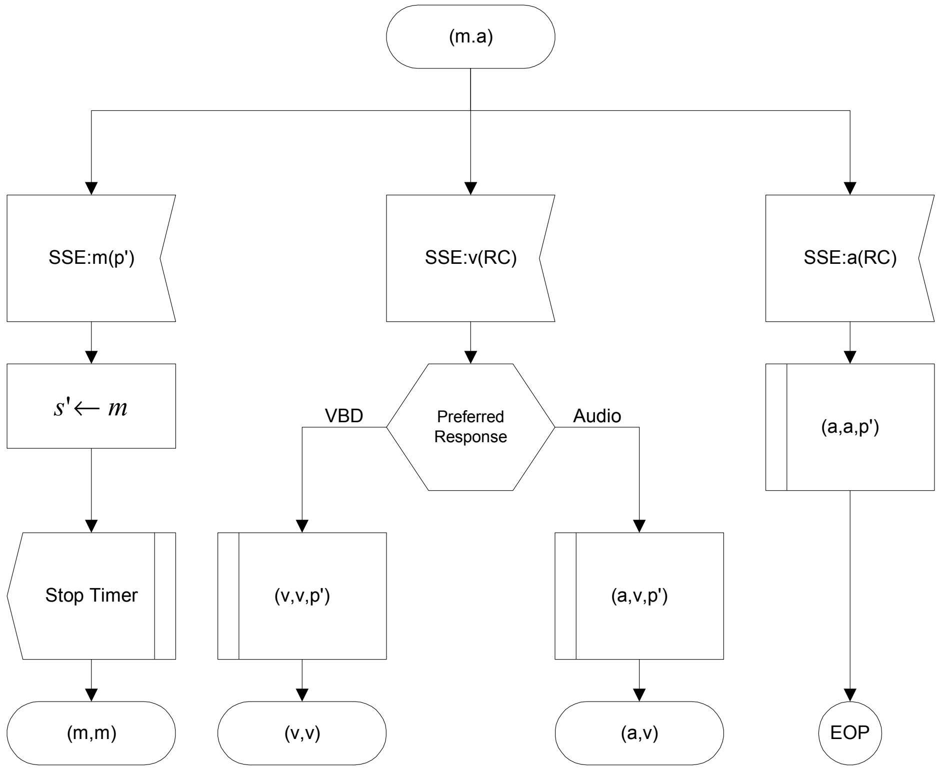


Figure 34/V.150.1 – SDL for state (m,a)

20.8.8 SDL diagrams for state (m,m)

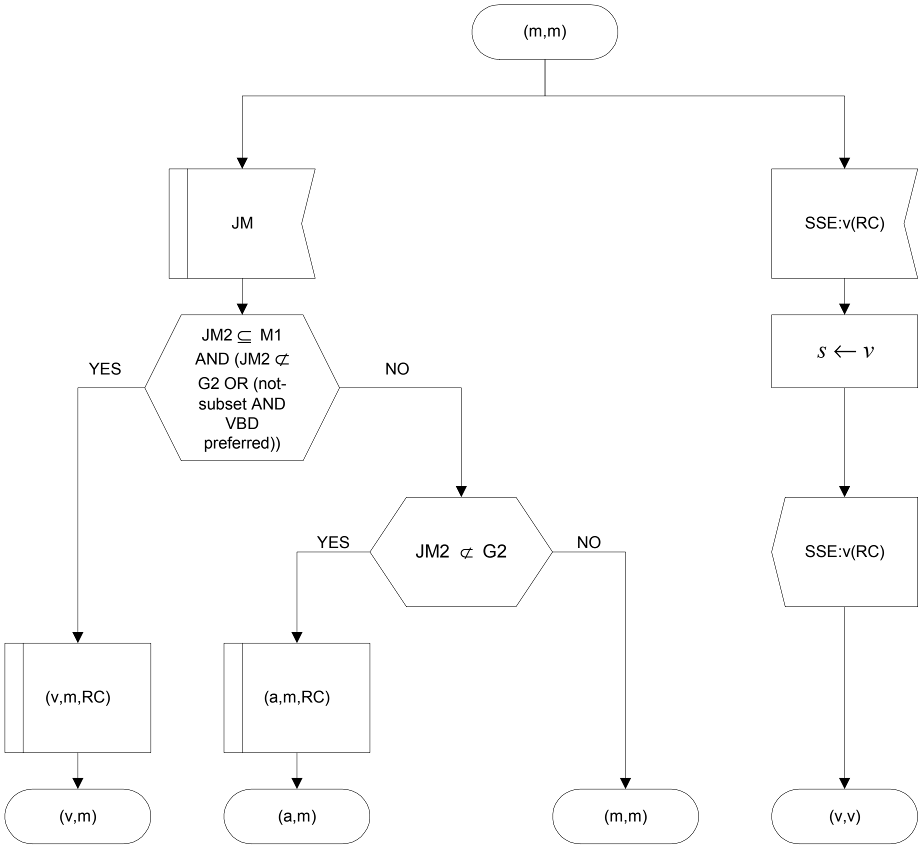


Figure 35/V.150.1 – SDL for state (m,m)

20.8.9 SDL diagrams for state (m,v)

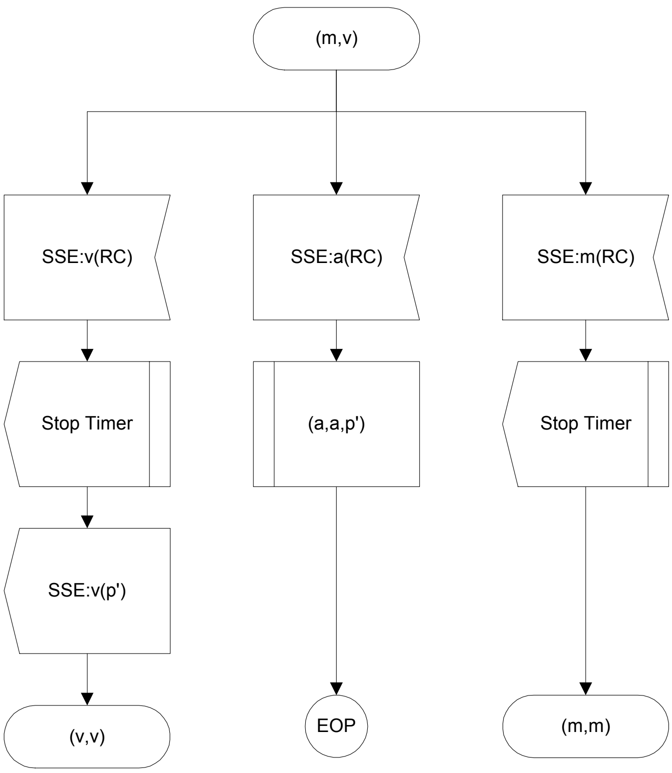


Figure 36/V.150.1 – SDL for state (m,v)

20.8.10 SDL diagrams for state (v,a)

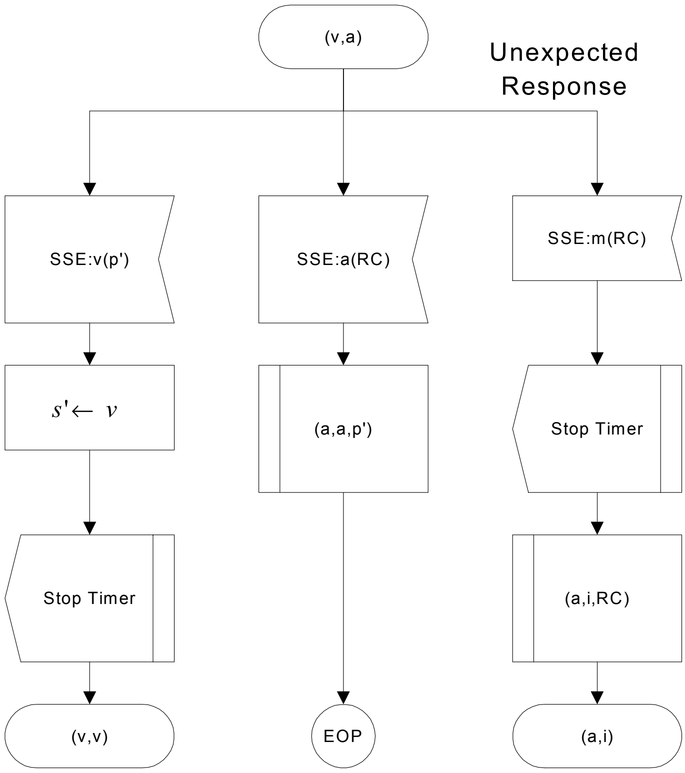


Figure 37/V.150.1 – SDL for state (v,a)

20.8.11 SDL diagrams for state (v,m)

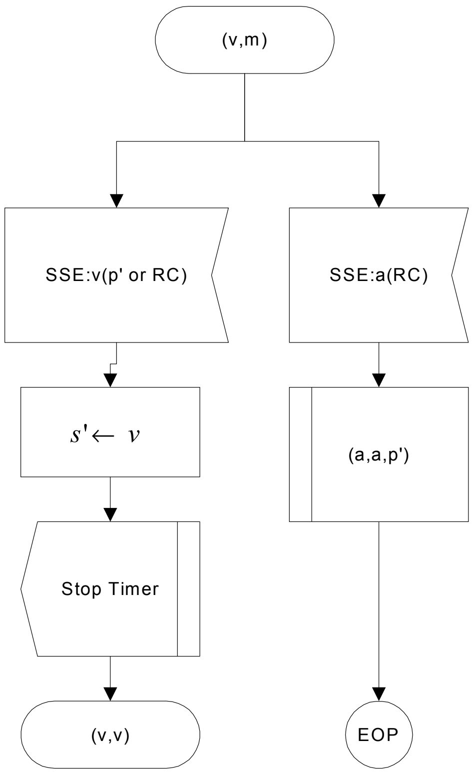


Figure 38/V.150.1 – SDL for state (v,m)

20.8.12 SDL diagrams for state (v,v)

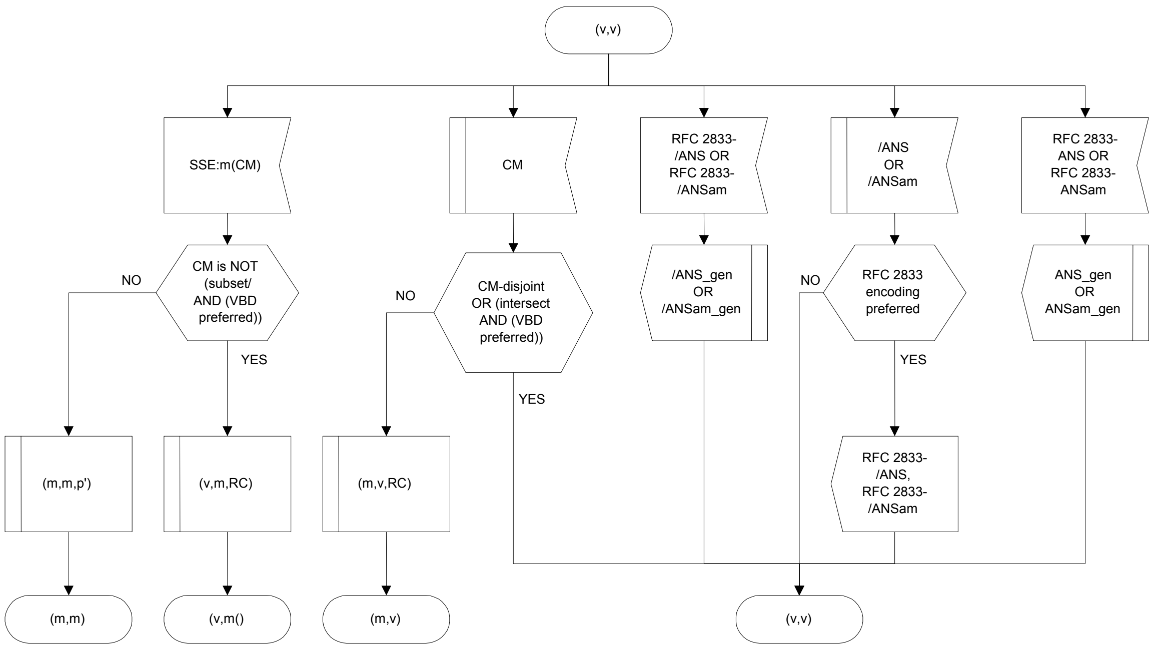


Figure 39/V.150.1 – SDL for state (v,v)

20.8.13 SDL diagrams for state (*,*)

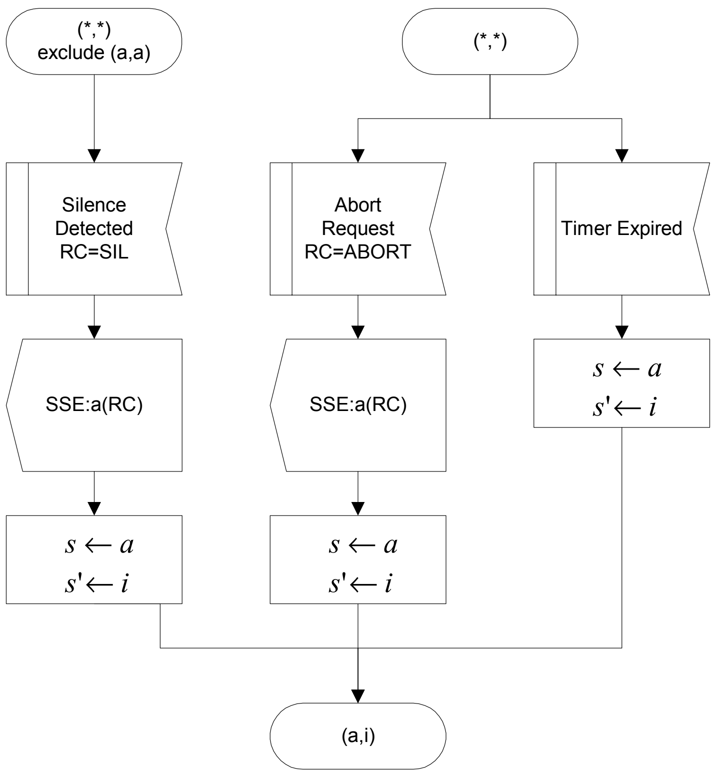


Figure 40/V.150.1 – SDL for state (*,*)

21 Procedures for audio to MoIP transport switching

This Recommendation requires that SSE and RTP encodings shall use the same UDP port. The gateway shall be capable of supporting all MoIP communications over a single pair of UDP ports, including RTP and non-RTP (e.g., Multiple Payload Streams as defined in ITU-T Rec. H.245). The gateway may choose to use additional UDP ports for the non-RTP communications.

For all RTP encodings that share the same UDP port, they shall also use the same SSRC (e.g., same sequence number and time source). This requirement does not apply for FEC packets as per RFC 2733. The use of RFC 2733 shall not violate the timing requirements described above.

It is necessary to ensure the timely indication of various events that will cause a change of media type on these UDP ports. Examples include transitions to or from Audio to VBD or Modem Relay as appropriate. All the call flow procedures considered by this Recommendation use a protocol designed specifically for this task. This protocol is called the State Signalling Event Protocol (SSE) and is defined in Annex C.

22 Procedures for modem relay operation

This clause provides the procedures necessary to support Modem Relay operation of MoIP.

22.1 Procedures used for gateway to DCE error control

There are no specific procedures for Modem Relay to DCE error control. The only requirement is that the gateway has to inform its peer of the error control parameters it has negotiated with its DCE. This information is sent using the CONNECT message.

22.1.1 Break signal procedures

The following clauses describe the procedures to be used by a MoIP gateway in order to handle the reception, transport and generation of break and break acknowledgement signals.

22.1.1.1 Break signal detection procedures

These are the procedures used when gateways detect a break signal from a connected DCE. The clause divides the procedures for error corrected and non-error corrected scenarios.

Every unique (non-repeated) break signal generated by a local end-point DCE shall be relayed to the remote end-point DCE. Repeated break signals shall not be relayed.

22.1.1.1.1 Error-corrected scenarios

Expedited and destructive break messages shall be transmitted on the IP-TLP expedited channel. The non-destructive, non-expedited break message shall use the same IP-TLP channel that is being used for the transport of user data.

If the MoIP break control entity detects a destructive break signal it shall indicate to the IP-TLP that it initiate the selective destruction of data.

For the cases where the error correction protocols are not the same on each side of the link, the mapping defined in Table 38 shall be used.

Table 38/V.150.1 – Break mapping table

<table>
<tr><td colspan="2"></td><td colspan="3">To</td></tr>
<tr><td colspan="2"></td><td>LAPM</td><td>Annex A/V.42</td><td>V.14</td></tr>
<tr><td rowspan="3">From</td><td>LAPM</td><td>Same</td><td>Break length is lost</td><td>Break type is lost; if no break length provided, use default duration. (Note)</td></tr>
<tr><td>Annex A/ V.42</td><td>Same (no length sent in LAPM message)</td><td>Same</td><td>Break type is lost; break length uses default duration. (Note)</td></tr>
<tr><td>V.14</td><td>If received from an expedited channel, generate either a DE or NDE break. If received from a non-expedited channel, generate an NDNE break.
Break length is passed</td><td>If received from an expedited channel, generate either a DE or NDE. If received from a non-expedited channel, generate an NDNE break.
Break length is discarded</td><td>Same</td></tr>
<tr><td colspan="5">NOTE – The default break length duration is 1.5 second.</td></tr>
</table>

Table 39 defines the behaviour of the MoIP gateway related to the actions it shall take with respect to the data and break messages.

Table 39/V.150.1 – Break procedure behaviour

Break handling option	With respect to data and break messages			
	Going to remote gateway	**Going to local modem**	**Coming from remote gateway**	**Coming from local modem**
Destructive break signal from PSTN	– Send "selective destructive" primitive to IP-TLP – Discard data not yet delivered to the IP-TLP – Transmit BREAK message on expedited channel	– Discard data not yet delivered	– Discard data until receive BREAKACK	– Hold and flow off data until receive BREAKACK
Destructive break message from IP	– Send "selective destructive" primitive to IP-TLP – Discard data not yet delivered to the IP-TLP	– Complete data packet transmission in progress, then transmit destructive break – Discard data not yet transmitted	– Discard data until receive break ack signal.	– Discard data until receive break acknowledge-ment signal

Table 39/V.150.1 – Break procedure behaviour

Break handling option	With respect to data and break messages			
	Going to remote gateway	Going to local modem	Coming from remote gateway	Coming from local modem
Destructive break acknowledgement signal from PSTN	– Reset trans-compression engine(s) – Send BREAKACK followed by data	– Reset trans-compression engine(s) and resume data transmission	– Reset trans-compression engine(s) and resume data reception	– Reset trans-compression engine(s) and resume data reception
BREAKACK from IP link	– Reset trans-compression engine(s) and resume data transmission	– Reset trans-compression engine(s) and resume data transmission	– Reset trans-compression engine(s) and resume data reception	– Reset trans-compression engine(s) and resume data reception

22.1.1.1.2 Non-error corrected scenarios

In the symmetric non-error corrected scenario, the transport of the break signal is dependent upon the data type being used in the MR session. If a Raw data type is being used then no procedures are necessary. The break signal is conveyed within the data stream. If another data type is being used, then the break signal shall be extracted from the data stream and a V.14 BREAK IP-TLP message is transmitted for each unique break signal detected on the V.14 DCE connection. The gateway may send the BREAK indication on the IP-TLP expedited channel, or on the same channel that is being used for user data.

The response time in sending the BREAK message is implementation-specific and is not defined in this Recommendation. There is no requirement to send a break acknowledgement signal (BREAKACK) for the symmetrical non-error corrected case.

If a gateway operating in an error corrected configuration receives a V.14 BREAK indication from an IP-TLP expedited channel, it shall generate either a NDE or a DE break on the telephony link. If the received V.14 BREAK indication is off the IP-TLP non-expedited channel, it shall generate a NDNE break on the telephony link.

For the asymmetric non-error corrected scenario, BREAK messages being conveyed in the error corrected to non-error corrected direction shall use the same BREAK message as used in the symmetrical error corrected case.

22.1.1.2 Break acknowledge procedures

A gateway shall generate a break acknowledgement signal to the local DCE in response to detecting a break signal from the local DCE within either a timeout or the reception of a BREAKACK message from the peer gateway, whichever occurs first. The value of this timeout is manufacturer-specific and may be zero to a very large value. Implementations should maintain compatibility with V.42.

A gateway shall transmit a BREAKACK IP-TLP message to its peer gateway in response to a break acknowledgement signal from a local DCE.

The BREAKACK IP-TLP message shall use the same IP-TLP channel as being used for user data.

For MoIP sessions that are asymmetrical non-error corrected, the break Acknowledgement signal sent to the local DCE may be generated immediately upon detecting the local break.

22.2 Compression negotiation procedures

The connection scenario or Trans-compression configuration is determined statically as defined in 13.5. For all the possible connection scenarios considered by this Recommendation there are only two sets of procedures used to establish the appropriate compression modes. The first is for the N-TCX to N-TCX case and the second is for the remaining possibilities which include various combinations of D-TCX, S-TCX and N-TCX as described in Table 1. The rules that guide these procedures in terms of how gateways negotiate their TCX capabilities and parameters are given below.

The capability for renegotiating V.44 parameters after link establishment for connection scenarios MR2, MR3 and MR4 is for further study.

22.2.1 General definitions for TCX selection rules

The following are the definitions as used in the selection rules for MoIP TCX procedures described in this clause.

$G1_P$: Represents the negotiation posture of G1 when it performs XID negotiation with M1. These are the "maximum" compression options that can be specified by the gateway in XID frames for negotiation with M1.

$G2_P$: Represents negotiation posture of G2 when it performs XID negotiation with M2. These are the "maximum" compression options that can be specified by the gateway in XID frames for negotiation with M2.

GIP: Is the compression posture on the IP link between gateways for Double-to-Double TCX connection scenario.

G_d: Is the capability vector of the Double trans-compression gateway.

The negotiation posture is represented in vector form as follows:

$$Gx_P = GxPV44TX + GxPV44RX + GxPV42BIS + GxPMNP5$$

where:

$$\begin{aligned} x &\ \ \text{is 1 or 2} \\ GxPV44TX &= \{GxV44TxDictionary, GxV44TxStringLength, GxV44TxHistory\} \\ GxPV44RX &= \{GxV44RxDictionary, GxV44RxStringLength, GxV44RxHistory\} \\ GxPV42BIS &= \{GxV42bisDictionary, GxV42bisStringLength\} \\ GxPVMNP5 &= \{GxMNP5Support\} \end{aligned}$$

Vectors $G1_T$ and $G2_T$ represent the trans-compression capability of a gateway as follows:

$$Gn_T = GnV44TX + GnV44RX + GnV42BIS + GnMNP5$$

where:

$$\begin{aligned} \mathrm{n} &= \text{1 or 2 and:} \\ GnV44TX &= \{GnV44TxDictionary, GnV44TxStringLength, GnV44TxHistory\} \\ GnV44RX &= \{GnV44RxDictionary, GnV44RxStringLength, GnV44RxHistory\} \\ GnV42BIS &= \{GnV42bisDictionary, GnV42bisStringLength\} \\ GnVMNP5 &= \{GnMNP5Support\} \end{aligned}$$

If a particular compression function is not supported, its corresponding parameters shall be set to zero in above vector. For non-zero values, trans-compression may occur from any supported compression type (up to the supported values) to any other supported compression type. Furthermore, Trans-compression capabilities are symmetric meaning that, when a trans-compression capability is indicated, the gateway may trans-compress from a supported compression type to another supported compression type regardless of whether the conversion is for data flow from telephony to IP side or in the reverse direction.

The vector operation **MIN** is defined as:

$$\boldsymbol{V} = (v1, v2, v3, \ldots.)$$

$$\boldsymbol{U} = (u1, u2, u3, \ldots.)$$

$$\boldsymbol{MIN}(\boldsymbol{V}, \boldsymbol{U}) = (min(v1, u1), min(v2, u2), min(v3, u3), \ldots.)$$

22.2.2 Procedures for none-TCX configurations

The following procedures are valid for both the On-Ramp and Off-Ramp gateways having selected to connect in the No-TCX to No-TCX configuration (connection scenario MR1).

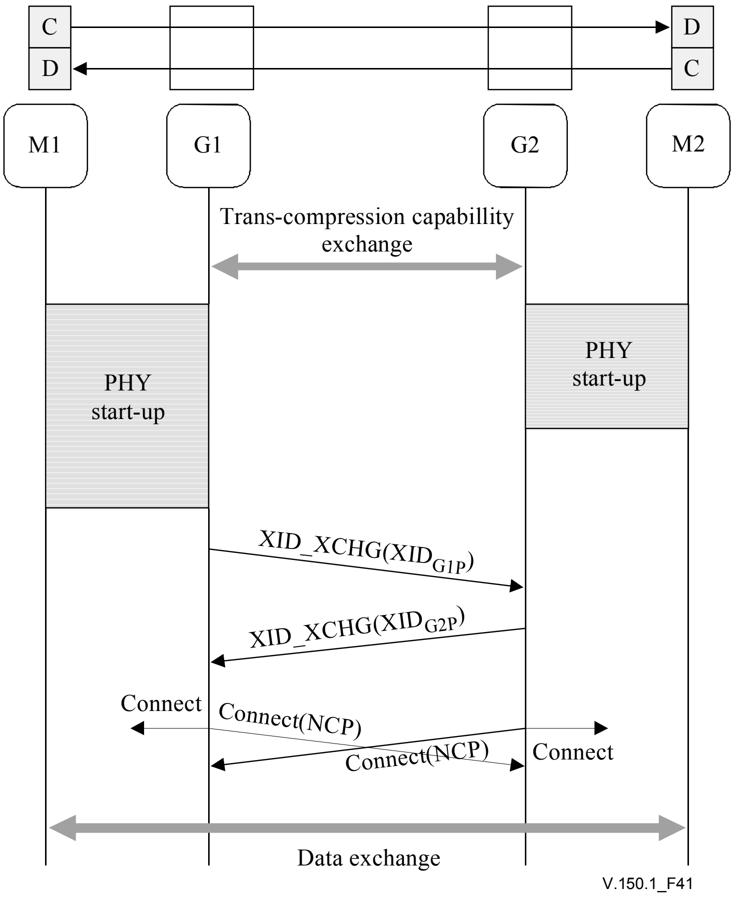


Figure 41/V.150.1 – XID negotiation for N-TCX to N-TCX

22.2.2.1 Trans-compression configuration for none-TCX scenarios

The default set of compression parameters for N-TCX negotiation shall be V.42 *bis* bidirectional with dictionary size of 1 k and string length of 32.

22.2.2.2 On-ramp gateway procedures for N-TCX negotiations

22.2.2.2.1 At the beginning of the Compression Negotiation Phase, G1 shall send to G2 an XID_XCHG message containing the maximum compression parameters (G1P) that G1 prefers for the On-Ramp connection and continue with 22.2.2.2.2.

The gateway may use either end-to-end parameter exchange as defined by these procedures, or the default parameter set for its G1P.

22.2.2.2.2 To support an end-to-end XID exchange between M1 and M2, G1 should wait to receive XID_{M1} from its local modem and use this information as XID_{G1P} in a XID_XCHG(XID_{G1P}) message that it transmits to G2. This does not preclude G1 from sourcing XID_{G1P} on its own using the default parameter set at an earlier time if it does not wish to use end-to-end XID exchange.

22.2.2.2.3 G1 shall wait for an XID_XCHG(XID_{G2P}) message from G2. G1 shall ensure that XID_{G2P} is identical to the compression negotiated on the On-Ramp connection; if it is not, it shall initiate a disconnect.

22.2.2.3 Off-ramp gateway procedures for N-TCX negotiation

22.2.2.3.1 After G2 activates its IP-TLP it shall wait for a XID_XCHG(XID_{G1P}) message from gateway G1. Gateways may opt not to use end-to-end XID exchange procedures. In this situation, the default compression and parameter set may be used to establish compression for the connection.

22.2.2.3.2 Upon receipt of an XID_XCHG message from G1, if it has not already done so, G2 shall transmit a XID_XCHG(XID_{G2P}). G2P is the compression negotiation outcome (or desired outcome in the case where XID_{def} is used) of the Off-Ramp connection.

22.2.2.3.3 G2 shall not negotiate anything above XID_{G1P} on the Off-Ramp connection.

22.2.2.3.4 If the Local XID exchange at G2 occurs before it receives a XID_XCHG message from G1 (see Figure 42B) and the result of this local XID negotiation (i.e., XID_{M2}) is known, G2 shall send XID_{M2} in its XID_XCHG response to G1. However, if the result is not known due to the local XID exchange result occurring after the reception of the XID_XCHG message from G1, then G2 may either wait for the local XID negotiation to complete and use the result of the exchange (see Figure 42D) or use the default parameter set in its XID_XCHG response to G1 (see Figure 42C).

22.2.2.4 Examples

Figure 42 shows several examples of N-TCX XID negotiation. Example A is a typical end-to-end XID negotiation. Examples B, C and D illustrate scenarios whereby the Off-Ramp gateway prefers to use the default parameter set. Example E is the case of the On-Ramp gateway preferring using the default parameter set.

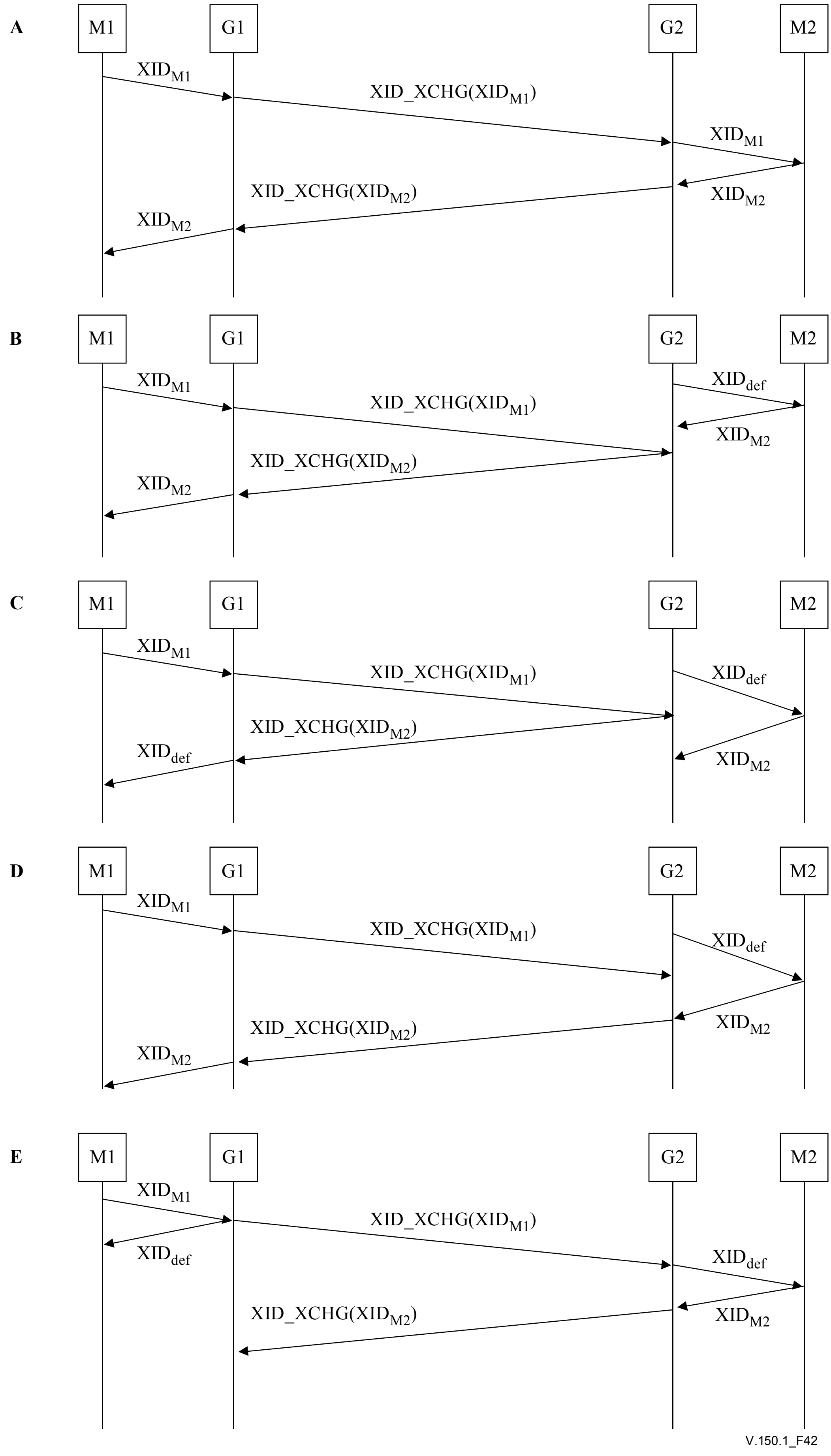


Figure 42/V.150.1 – Examples of N-TCX XID negotiation

22.2.2.5 XID deferral

To assist end-to-end XID negotiation, the JM delay procedure defined in 20.7 may be used (as shown in Figure 21). The use and definition of other XID deferral procedures is for further study.

22.2.3 Procedures and rules for single and double-TCX configurations

The following procedures are valid for both the On-Ramp and Off-Ramp gateways operating in either S-TCX or D-TCX configurations. For both of these configurations there are two local XID negotiations. Each XID negotiation is independent of the other and can occur in any order of time.

These procedures utilize two messages. The first of these messages is exchanged during the Call Set-up phase of the connection and it indicates the Trans-compression capabilities of the gateway. The second message is exchanged between peer gateways following their own local modem compression negotiations. This message contains the negotiated compression parameters as consented to by the local modem and gateway. Figure 43 illustrates such an example.

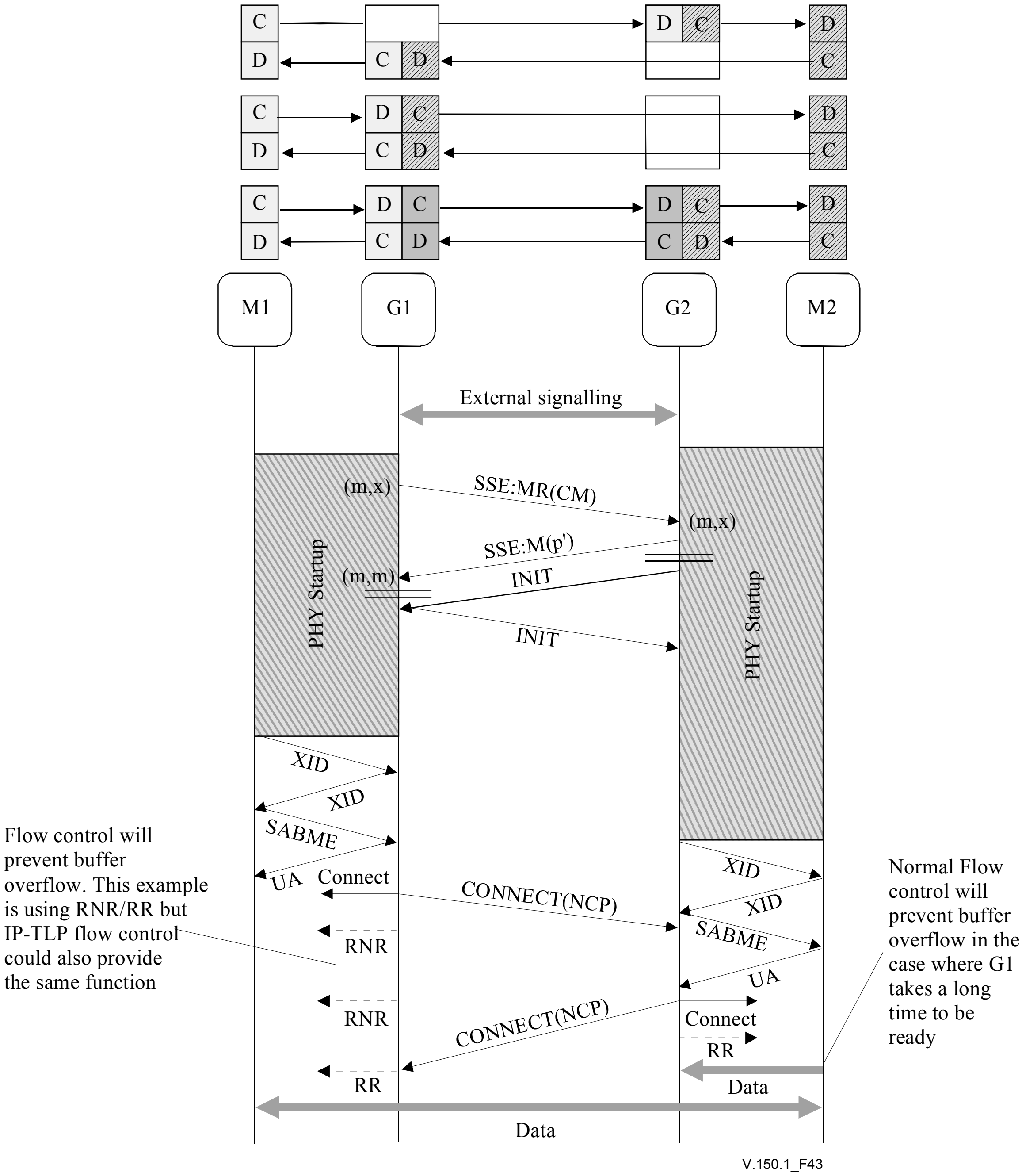


Figure 43/V.150.1 – Example signal flow of S-TCX or D-TCX procedures

22.2.3.1 S-TCX to S-TCX (connection scenario MR3) rules

The following rules apply for gateways when negotiating from an initial Trans-compression configuration of S-TCX to S-TCX.

$$G1V44TX \geq G1PV44TX \geq MIN(G1V44TX, G2V44RX)$$

$$G2V44RX \geq G1PV44RX \geq MIN(G1V44TX, G2V44RX)$$

$$G2V44TX \geq G2PV44TX \geq MIN(G2V44TX, G1V44RX)$$

$$G1V44RX \geq G2PV44RX \geq MIN(G2V44TX, G1V44RX)$$

$$G1PV42BIS = MIN\ (G1V42BIS, G2V42BIS)$$

$$G2PMNP5 = MIN\ (G1MNP5, G2MNP5)$$

22.2.3.2 D-TCX to N-TCX (connection scenario MR4) rules

The following rules apply for gateways when negotiating from an initial Trans-compression configuration of D-TCX to N-TCX (or vice versa).

$$G1_P = G_d$$

$$G2_P = G_d$$

22.2.3.3 Procedures for both gateways

22.2.3.3.1 On entering the (m,m) state, the gateway shall enable its IP-TLP, transmit its INIT message and shall be able to receive messages.

22.2.3.3.2 On completion of physical layer start-up between the gateway and local modem, they begin their ODP/ADP and XID negotiations. Due to possible differences in the physical layer start up times, these negotiations need not occur at the same time for each gateway-modem pair.

22.2.3.3.3 Having completed their XID/SABME/UA exchange, at the point in which an indication of Connect is sent to the host, an IP-TLP CONNECT message containing the Negotiated Compression Parameters is transmitted to the peer gateway. This message contains the compression parameters to be used for the local dial-up link. If the gateway does not receive a similar message from the remote gateway and is not yet ready to transmit or receive user data, it shall prevent any data exchange to the peer gateway by means of data flow control procedures (e.g., V.42 or SPRT).

22.2.3.3.4 On reception of an IP-TLP Negotiated Compression Parameters message from the Remote gateway, the local gateway shall determine if those parameters allow the TCX function to be switched to another TCX configuration. If this is the case, the TCX function reconfigures to this mode of operation and Receive Data is enabled.

22.2.4 Trans-compression configuration for double-TCX scenarios

For the Double-to-Double Trans-compression case, the gateways can decide during call set-up whether to use S-TCX to S-TCX or D-TCX to D-TCX configurations.

In the case of D-TCX to D-TCX, the final configuration of the Trans-compression elements is based upon each of the compression elements negotiating independent of each other. M1 and G1, G1 and G2 and G2 and M2 each determine their own compression mechanisms.

22.2.4.1 D-TCX to D-TCX (connection scenario MR2) rules

The following rules apply for gateways when negotiating from an initial Trans-compression configuration of D-TCX to D-TCX.

$$G1_P = G1_T$$

$$G2_P = G2_T$$

$$GIPV44(G1_T{\rightarrow}G2_T) = MIN(G1V44TX, G2V44RX)$$

$$GIPV44(G2_T{\rightarrow}G1_T) = MIN(G1V44RX, G2V44TX)$$

$$GIPV42BIS = MIN(G1V42BIS, G2V42BIS)$$

Compression coding on the IP portion of the connection shall be symmetric (i.e., same method used by either gateway) and based on the GIP element values and would use the set of GIP elements corresponding to V.44 if available, else V.42 *bis*.

22.2.5 PROF_XCHG procedures

Clause VII.4 contains some implementation guidelines and suggestions for performing these procedures.

22.2.5.1 Indication of support

A gateway may indicate its capability to receive the optional XID/LR profile exchange message, by using the appropriate code-point in the IP-TLP INIT message. Only if the peer receiver indicates this capability shall a transmitter send the PROF_XCHG message.

22.2.5.2 Sending PROF_XCHG

If a gateway knows a particular local modem's protocol/compression capabilities (M1 for G1, M2 for G2), and if the peer gateway's INIT indicates that PROF_XCHG may be sent, then the gateway may send these capabilities to the peer gateway in a PROF_XCHG message. These capabilities are known as the particular modem's "profile". If a gateway has incomplete knowledge for a particular modem, then the PROF_XCHG is partially populated (some of the protocols and/or compressions are noted as "unknown" and the corresponding value fields are zero, per 15.4.10). If the gateway has no knowledge for a particular modem, then no PROF_XCHG should be sent.

For example, if the modem is known to propose V.44 compression during call origination, or to accept this compression when answering, then the PROF_XCHG V.44 capability field (see 15.4.10 (bits 4:5) above, PROF_XCHG, octets 1-2, bits 4-5) would be set to "yes". Again, if the modem is known to propose 2048 receive codewords during call origination, or to negotiate with this value when answering, then the PROF_XCHG V.44 receive codewords field (same clause, octets 11-12) would be set to 2048.

If the gateway knows that a particular protocol or compression is definitely not supported by a particular modem, then the PROF_XCHG is sent with that particular capability field set to "no". Note that a "no" is different from an "unknown", and the receiving gateway should distinguish the two.

22.2.5.3 Receiving PROF_XCHG

A received PROF_XCHG is valid if the gateway has agreed to such reception in the IP-TLP INIT message it sent.

If a gateway sends a PROF_XCHG to its peer gateway (which has agreed to such reception), and receives a valid PROF_XCHG from that peer, then the gateway should use these two profiles to compute the optimal compression parameters. These parameters are sent (via XID or LR) when the gateway's telephony link reaches protocol negotiation phase, without the need to synchronize with the peer gateway's start-up.

This computation is simply an application of the standard protocol negotiation rules for LAPM, Annex A/V.42 (1996), V.44, V.42 *bis* and MNP5, as appropriate. See VII.4.3 "Operation when both gateways know their modems' profiles" and its references for guidelines on this negotiation.

Optimization is still possible when the gateways have incomplete knowledge: see clauses VII.4.4 and VII.4.5 for suggestions. With no knowledge, PROF_XCHG is neither sent nor received, and the gateways must fall back to end-to-end procedures, or the use of a "default XID".

22.3 Data transfer phase

This clause describes the procedures that apply to the transfer of user-data between gateways.

22.3.1 Data type capability indication

Upon IP-TLP bring up gateways shall transmit an INIT message. This message indicates the optional data type capability of the transmitter's receiver and whether the gateway is capable of supporting asymmetric data types. gateways may use asymmetric data types only if both gateways mutually consent to do so, otherwise the gateways shall operate in the default symmetric data type mode.

22.3.2 Data type selection

Upon completion of the DCE link layer protocol negotiation, (successful or unsuccessful) a gateway shall transmit an IP-TLP CONNECT Message. The gateway will indicate in the CONNECT message the optional data types it has available to be selected by the peer gateway transmitter. For symmetric data type mode, the gateway shall only indicate a maximum of one optional data type in the CONNECT message. If the optional type indicated by a peer gateway in its CONNECT message matches the local gateways indication, then that data type is used otherwise the default data type for the error correction scenario (also indicated in the CONNECT message) shall be used.

If both gateways have negotiated the support asymmetric data types, then the transmitter may select from any of the available optional data types indicated in the received CONNECT message or the default data type.

22.3.3 Enabling transmission

The gateway shall wait for a CONNECT message from the remote gateway before it starts sending user data packets. The reception of the CONNECT message will allow the gateway to decide which IP-TLP channel and data types to use based upon the link layer parameters.

Data types shall not change during a MR session.

23 Modem delay procedures

This clause describes procedures related to events that have impact on the PSTN physical layer. These events include indication of the completion of modem start-up, retrains and rate-renegotiations.

23.1 Initial start up

In modem relay, the time it takes for the modem physical link to be established on each side of the IP connection may differ. It is also possible that modulation and data signalling rates may not be the same. To facilitate the gateway's ability to support these differences, a gateway, once it has completed its initial start-up and handshake (if appropriate), shall transmit an indication to the peer gateway using the IP-TLP MR_EVENT(PHYSUP) message. The point at which a modem is considered to have completed its start-up is the point in its defined start-up sequence at which ITU-T Circuit 106 or 109 would be activated.

23.2 Retrain and rate renegotiations

Both these event types are treated in the same way. On either initiating or responding to a retrain or rate renegotiation on the PSTN link, a gateway shall inform its peer gateway of the change of state by immediately sending a MR_EVENT(RETRN) or MR_EVENT(RRNEG) IP-TLP message (see 15.4.8).

A corresponding MR-EVENT(PHYSUP) message is sent within a time of 100 ms (Tp) of a return to the data mode state. Figure 44 illustrates two examples of how the MR_EVENT message may be used. Figure 44A shows a single retrain or rate renegotiation event and Figure 44B is an example of how to utilize the Tp timeout to control when to send a MR_EVENT(PHYSUP).

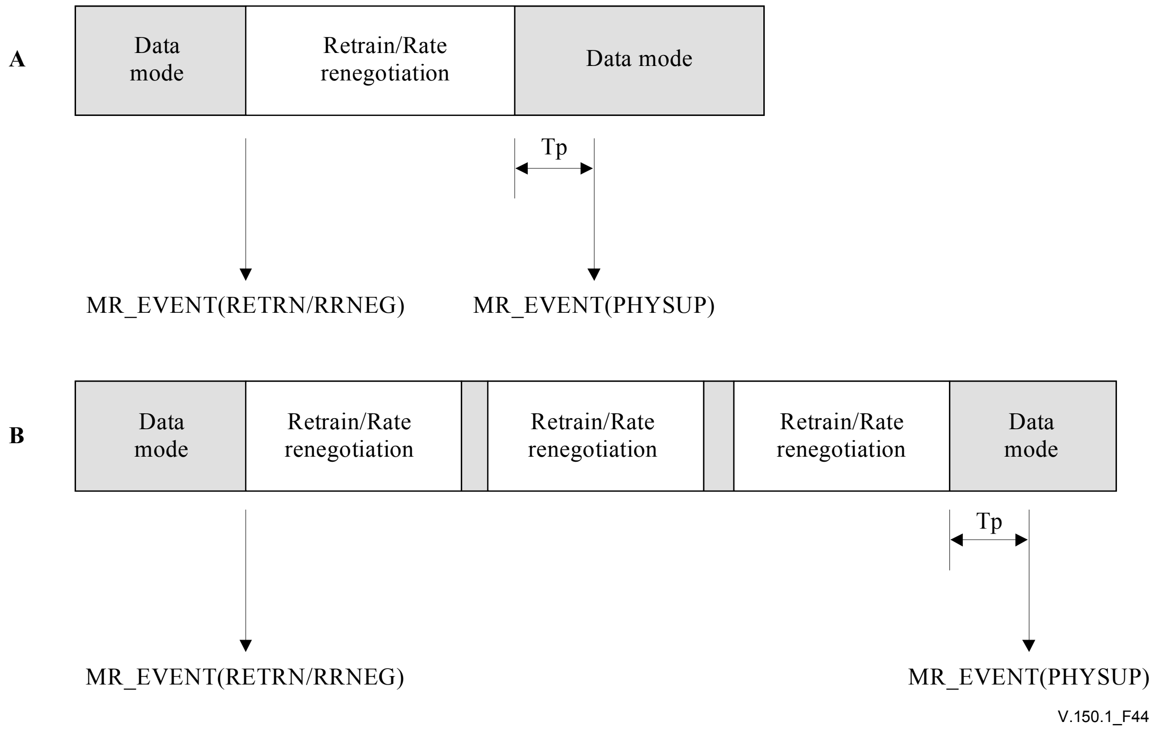


Figure 44/V.150.1 – Example uses of MR_EVENT messages during retrain or rate renegotiations

24 Cleardown procedures

A gateway shall return to its initial state after either a MR or VBD session has completed. When a transition to the initial state from MR is caused by the termination of a modem session, the SSE message should contain the reason code field with the disconnect reason. This will allow for the graceful transition to audio for modem calls. The disconnect message in IP-TLP may also be used to provide this information.

25 IP transport

Due to its unique real time requirements, the IP transport protocol for Modem-over-IP applications shall have the following attributes:

- Be reliable;
- Support point-to-point and duplex transport;
- Be packet preserving;
- Be uniquely identifiable and allow for the seamless transition to and from RTP;
- Be error detecting and correcting, non-corruptive, non-erasing and non-duplicating;
- Support expedited and sequenced delivery of packets;
- Have low latency, be bandwidth efficient;
- Support windowed flow control;
- A lightweight implementation.

This Recommendation assumes that the IP protocol conforms to the following standards:

IETF RFC 791, RFC 950, RFC 919 and RFC 920. This Recommendation does not impact any IP network topology, IP packet distribution and routing protocols, which are independent of this Recommendation.

For this version of the Recommendation, the default IP Transport Layer Protocol shall be Simple Packet Transport Protocol (SPRT) as defined in Annex B. The use of other reliable IP Transport Protocols is for further study.

25.1 SPRT packet structure for MoIP

This clause describes the Payload Profile to be used by SPRT for Modem-over-IP applications.

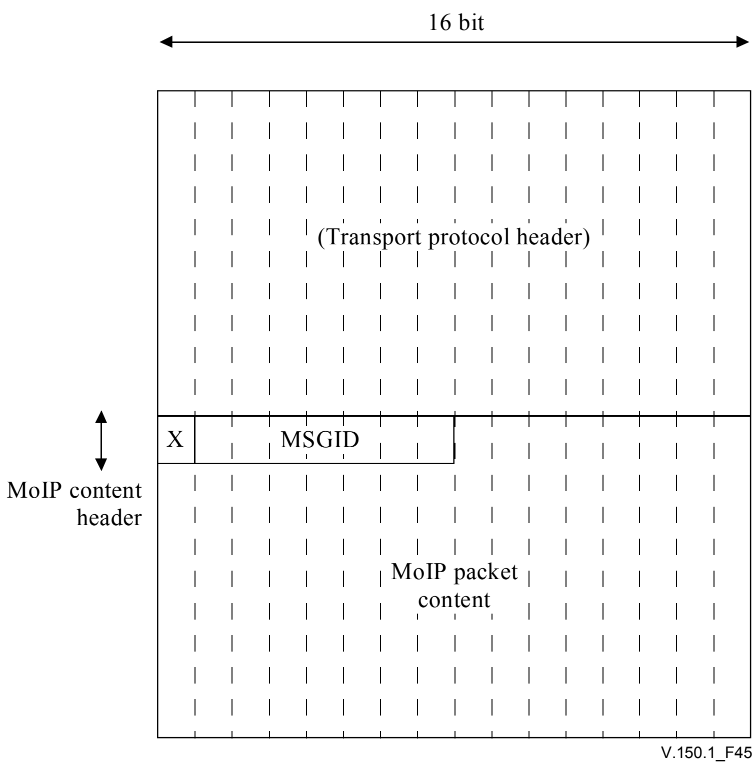


Figure 45/V.150.1 – Reference diagram for SPRT profile for V.150.1

25.2 MoIP content header

X: *Content Header Extension Bit*. Must be set to zero (0) in this release.

MSGID: *Message ID*. MoIP message identifier.

Annex A

ASN.1 notation

The provision of ASN.1 for the definition of the IP-TLP messages is for further study. All IP-TLP messages are fully defined in the main body of the Recommendation.

Annex B

Simple Packet Relay Transport (SPRT) protocol

This annex describes Simple Packet Relay Transport (SPRT) protocol, which is a reliable transport protocol that is encapsulated in UDP/IP (RFC 768) and is suitable for the reliable transport between gateways of data from facsimile, data-modem, and other such telephony applications across Voice-over-IP networks. This protocol is intended for gateway-to-gateway transport of bearer channel and control channel data.

B.1 Overview

This annex defines a reliable transport protocol that is encapsulated in UDP/IP. The protocol is suitable for real-time media applications that require reliable transport. Examples of these applications include voiceband modem, facsimile, and bearer data transport.

B.1.1 Transport layer (simple packet relay transport)

SPRT is a simple packet-based protocol layered upon UDP/IP, which provides reliable in-sequence delivery of data across an IP network. As a lightweight protocol, SPRT provides:

- No provision for the opening and closing of SPRT transport channels is defined, as it is outside the scope of this Recommendation.

  NOTE – This will eliminate the need to maintain various associated states. It is assumed that on transition into the SPRT transport protocol, the requested channels will have been opened external to the protocol and closure is by the users on session termination. Provisioning requirements for opening of new UDP ports are beyond the scope of this Recommendation.

- No provision to negotiate parameters associated with SPRT protocol is provided in this Recommendation.

Peer SPRT users are required to have compatible parameters for max message size (bytes) and window size (i.e., number of packets transmitted without any acknowledgements).

There is no provision within the SPRT protocol to negotiate these parameters, which may optionally be negotiated out-of-band (e.g., H.245, H.248 and RFC 2327). SPRT parameter negotiation is beyond the scope of this Recommendation. Clause B.2.2.1 defines the mandatory set of default parameters that shall be used in the absence of any out-of-band negotiation.

B.2 SPRT transport protocol specification

B.2.1 SPRT protocol reference model

Figure B.1 provides the reference model for the SPRT protocol. The reference model is provided to aid in the protocol description provided in this annex. The reference model is a conceptual tool and it is not intended to specify or constrain implementations.

The *SPRT user* is the application that uses the SPRT protocol. It interfaces to the protocol using primitives in both directions.

The *SPRT entity* is the protocol. It includes a *transmitter part* and a *receiver part*. The *SPRT entity* interfaces to the *SPRT User* through primitives in both directions. The *SPRT entity* interfaces to the network through the transmission and reception of packets. The *SPRT entity transmit part* includes the transmitter interface to the packet network (i.e., packets from the *SPRT entity* to the network). The *SPRT entity receiver part* includes the receiver interface to the packet network (i.e., packets from the network to the *SPRT entity*).

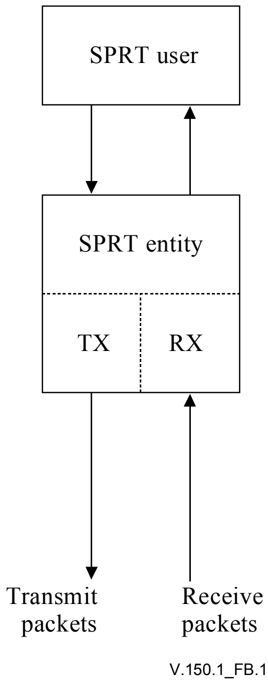


Figure B.1/V.150.1 – Reference model for SPRT

Figure B.1 illustrates the reference model used in the description of the SPRT protocol. SPRT protocol consists of three parts which are: SPRT Entity, Transmitter (TX), and Receiver (RX).

B.2.2 SPRT packet structure

B.2.2.1 SPRT header

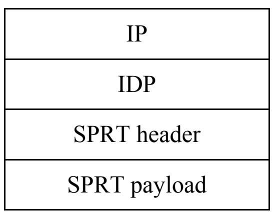


Figure B.2/V.150.1 – UDP encapsulation

The SPRT protocol is encapsulated in UDP as described in Figure B.2.

<table>
<tr><td>0</td><td>1</td><td>2</td><td>3</td><td>4</td><td>5</td><td>6</td><td>7</td><td>8</td><td>9</td><td>10</td><td>11</td><td>12</td><td>13</td><td>14</td><td>15</td><td>16</td><td>17</td><td>18</td><td>19</td><td>20</td><td>21</td><td>22</td><td>23</td><td>24</td><td>25</td><td>26</td><td>27</td><td>28</td><td>29</td><td>30</td><td>31</td></tr>
<tr><td>X</td><td colspan="7">SSID</td><td>R</td><td colspan="7">PT</td><td colspan="2">TC</td><td colspan="14">SEQUENCE NUMBER or Zero (0) (Note)</td></tr>
<tr><td colspan="2">NOA</td><td colspan="14">BASE SEQUENCE NUMBER</td><td colspan="2">TCN</td><td colspan="14">SQN</td></tr>
<tr><td colspan="2">TCN</td><td colspan="14">SQN</td><td colspan="2">TCN</td><td colspan="14">SQN</td></tr>
</table>

NOTE – The SEQUENCE NUMBER field contains the SEQUENCE NUMBER for packets which contain a payload. Packets without a payload (i.e., acknowledgment only packets) will have this field set to zero.

Figure B.3/V.150.1 – SPRT header

Note that the definition of the bit ordering shown in Figure B.3 is consistent with the definition described in 15.1.2.

There can be zero to three SQN fields included in the SPRT header. This means SPRT header size ranges from six to twelve bytes, depending on the number of acknowledgments contained in the header.

X: Header Extension Bit. This bit is set to zero and is reserved for use by the ITU-T.

SSID: SubSession ID. This parameter identifies a SPRT entity transmitter subsession. The first SSID used by the SPRT entity shall be zero (0). There is one SSID used for all the Transport Channels by the SPRT entity transmitter. On receipt of a destructive primitive command from the User, the SPRT entity transmitter shall increment SSID by one (1).

All subsequent packets on all Transport Channels are sent with this new SSID.

R: Reserved. This bit shall be set to zero.

PT: Payload Type. This field shall be set to the value assigned by external signalling when the call between the gateways is first set up.

The position of the R and PT fields in the SPRT header is consistent with the position of similarly named fields in the RTP header, so that the value of these fields may be used to distinguish between packets using the RTP protocol and packets using the SPRT protocol when these protocols use the same IP address/UDP port. SPRT and RTP are used in the same IP address/UDP port, the value of the PT field shall be different from any payload type used in this RTP session

TC: Transport Channel ID. Transport channel identifier.

Table B.1/V.150.1 – Transport channel ID definitions

TC No.	Title	Description
0	Unreliable, unsequenced transport	Used for acknowledgements only
1	Reliable, sequenced transport	Transport channel used for data
2	Expedited, reliable, sequenced transport	Transport channel used for control/signalling messages Data transported in this channel is expedited to the peer User relative to data transported in TC=1 (Reliable Transport)
3	Unreliable, sequenced transport	Transport channel for sequenced data that does not require reliable delivery

SEQUENCE NUMBER: This is an identifier used by the SPRT entity transmitter for packet sequencing when required. The first transmitted packet of each transmit channel will use the SEQUENCE NUMBER of zero. The SEQUENCE NUMBER is incremented by one for each subsequent packet of the sequenced transmit channels (TC of 1, 2 and 3). For retransmitted packets, the same SEQUENCE NUMBER that was used for the previous transmission of the same packet is used.

The packet Sequence Number for each transmit channel is independent of any other channel. The SEQUENCE NUMBER shall always be zero for the unsequenced channel (TC of 0).

NOA: Number of Acknowledgements. This field identifies the number of ACK fields included in the SPRT header. The valid values for this are zero, one, two and three.

BASE SEQUENCE NUMBER: The BASE SEQUENCE NUMBER identifies the sequence number of the packet that the SPRT user will receive next from its SPRT entity receiver for the TC indicated in this packet. The BASE SEQUENCE NUMBER is sent by the local SPRT entity transmitter and indicates to the remote SPRT entity the current state of the local SPRT entity receiver for this TC. This field is only applicable for reliable, sequenced channels (TC values of 1 and 2). For TC values of 0 and 3, this field is set to zero.

ACK fields: The ACK indication consists of a pair of fields TCN and SQN. Up to three ACK indications at any one time can be inserted into a SPRT header. The number of ACK fields is indicated in the NOA field. Packets received by the SPRT entity receiver with TC values of 1 and 2 shall be acknowledged by the transmission of a packet with an ACK field set to the TC/SEQUENCE NUMBER fields of the received packet. TCN is independent of the TC field, i.e., acknowledgment for any Transmit Channel receive packets can be placed in any Transmit Channel transmit packets.

B.2.2.2 SPRT payload

The SPRT payload contains a variable number of bytes of payload. A packet may be transmitted with no payload bytes if it is an acknowledgment only packet. An acknowledgment only packet is used to send ACK fields with no SPRT payload (see B.2.3.2).

B.2.3 SPRT operation

A SPRT Transport Channel operates independently of any other Transport Channels. The User indicates which Transport channel the data is to be transmitted on when providing the payloads to the SPRT entity. The Transport Channel of the received data is indicated to the User when the payload is delivered to the User. The Transport Channel characteristics are described in B.2.2.

B.2.3.1 SPRT transport channel buffer management

For the peer SPRT entities to be interoperable, it is required for them to have identical window size for each SPRT reliable Transport Channel as well as maximum SPRT payload size for each Transport Channel.

Table B.2/V.150.1 – SPRT buffer management parameters

Parameter	Description	Values
SPRT_TC1_PAYLOAD_BYTES	Maximum payload size for SPRT Transport Channel 1	132-256 bytes (default 132)
SPRT_TC1_WINDOWS_SIZE	Window size for SPRT Transport Channel 1	32-96 packets (default 32)
SPRT_TC2_PAYLOAD_BYTES	Maximum payload size for SPRT Transport Channel 2	132-256 bytes (default 132)

Table B.2/V.150.1 – SPRT buffer management parameters

Parameter	Description	Values
SPRT_TC2_WINDOWS_SIZE	Window size for SPRT Transport Channel 2	8-32 bytes (default 8)
SPRT_TC0_PAYLOAD_BYTES	Maximum payload size for SPRT Transport Channel 0	140-256 bytes (default 140)
SPRT_TC3_PAYLOAD_BYTES	Maximum payload size for SPRT Transport Channel 3	140-256 bytes (default 140)

B.2.3.2 SPRT reliable transport channel acknowledgments

For the sequenced transport channels (1, 2 and 3), the SPRT entity transmitter will number the payloads in order. On receiving a payload from a reliable transport channel (1 or 2), the SPRT entity receiver transmits an acknowledgement.

The transmitter using the ACK fields of the SPRT Header sends these acknowledgements. No payload is necessary in order to transmit an Acknowledgement packet (see B.2.2.1 and B.2.2.2).

The following are the conditions used by the SPRT transmitter for the generation of ACK and BASE SEQUENCE NUMBER updates:

a) If there is an SPRT packet to be transmitted by the SPRT transmit entity. Up to three ACK FIELDS may be added to this packet for any previously received packets that are to be acknowledged.

b) If there are a maximum of three received packets pending acknowledgment with no payload to be transmitted, the packet generated will have a NULL payload field, i.e., it is an acknowledgment only packet.

c) If one or two received packets that require acknowledgement after a timeout period given by TA01.

d) If there are no received packets pending acknowledgment and a timeout period TA02 expires since any packet was sent by the SPRT entity transmitter for reliable Transport Channel (1 and 2). A packet is generated to guarantee the remote SPRT entity receiver BASE SEQUENCE NUMBER for this Transport Channel is updated in a timely manner.

B.2.3.3 SPRT retransmits

For reliable transport channels, packets are retransmitted, if not acknowledged, after a timer period given by TR03 since the last transmit.

B.2.3.4 SPRT bandwidth management

SPRT operates in the VoIP continuous media environment and shall use the same bandwidth management and reservation mechanisms as specified for VoIP.

B.2.3.5 SPRT flow control

For reliable/sequenced channels (TC 1 and 2), SPRT protocol provides the ability for SPRT users to control their sources of information flowing into the packet network based on the ability for the remote SPRT user to consume the packet flow. The combination of the BASE SEQUENCE NUMBER returned from the remote SPRT Entity and the local SPRT entity's current SEQUENCE NUMBER describe the state of consumption of the packet flow by the remote SPRT user for this Transmit Channel.

B.2.3.6 SPRT timers TA01, TA02 and TR03

The setting and control of the timer values used in SPRT is implementation specific. These timers may be optionally modified dynamically during the session. Each Transport Channel can have its own unique timer values.

The setting and control of timers TA01, TA02 and TR03 depend upon application. The values of these timers is based upon consideration of IP network characteristics (e.g., Round Trip delay, jitter, packet loss).

The following are suggested values for these SPRT timers for applications that utilize static timers.

Table B.3/V.150.1 – Suggested values for SPRT timers

Reliable transport channel	Suggested timer value		
	TA01	TA02	TR03
1	90 ms	130 ms	500 ms
2	90 ms	500 ms	500 ms

Annex C

State signalling events protocol

This annex defines a mechanism to signal media states using RTP packets called State Signalling Events (SSEs). The MIME type for this RTP packet format is "audio/v150fw". SSE messages signal a media state that has no specified duration. The event referred to is the sending or receiving of an SSE message indicating a media state, and is not necessarily accompanied by a local media state change. Other possible triggers for sending SSE messages are a perceived change in the remote media state, and an attempt at SSE protocol recovery.

C.1 Introduction

This State Signalling Event (SSE) mechanism addresses a need for fast media state synchronization of media gateways and endpoints. State Signalling Events are RTP-encoded event messages that coordinate switches between different media states as defined in clause C.2

By definition, an SSE media stream governs all media streams in a session. These media streams may span several ports, and may be sent to the same or different connection addresses (e.g., IP addresses). Media streams that are not reflected in the set of SSE protocol states as defined in clause C.2, are unaffected by SSEs.

Associated with the SSE media state are sets of Reason Identifier Codes (RIC) (see C.3.2). The definitions of these RIC are specific to the media state application. An application may define its own unique set of RICs. The exception is for the VBD and Modem Relay states that share the same RIC set as defined in this Recommendation. Any additions or changes to either the VBD or MR RIC set shall be documented within this Recommendation.

C.2 Definition of media states

For the purposes of this annex, a **media state** is defined in terms of the ultimate use of the media. The definition of the "media state" is similar, though not identical, to that of the "media type" parameter in RFC 2327, which follows the definition of MIME types. Media states are represented numerically (Table C.1). The State Signalling Event (SSE) protocol defined in this annex is used, by compliant implementations, to synchronize shifts between these media states.

Apart from the high-level definitions in this clause, this annex does not detail the range of media properties that may be defaulted, provisioned or negotiated at the time of session establishment (Annex E for SDP and Annex F for H.245).

This annex defines the following values of the media state parameter:

C.2.1 Initial audio state

This is the initial state of any SSE-driven media state machine. By definition, the initial audio state excludes modulated voiceband data (see C.2.2). Note the subtle difference from the MIME media type "audio," which includes modulated data. While in the initial audio state, an audio codec that conforms to the Real-time Transfer Protocol (RTP) and has been negotiated by both gateways must be used.

C.2.2 Voice Band Data (VBD)

This refers to data modulated as a voiceband signal. This data could be modem or facsimile data. This stream has the properties as defined for VBD in clause 8. Note that the VBD media state is included in the MIME media type "audio". While in the voiceband data state, an audio codec conforming to the RTP/AVP profile must be used.

C.2.3 Modem Relay (MR)

This refers to the encapsulation of a baseband (unmodulated) data signal within an appropriate IP-TLP (e.g., SPRT). Modem relay media streams are defined in clause 9. If a UDP port is switched between a modem relay media stream and an RTP media stream, SPRT shall not use payload types assigned to an RTP encoding. The Modem Relay media state is included in the MIME media type "audio".

C.2.4 Fax Relay (FR)

This refers to the encapsulation of baseband (unmodulated) facsimile signals in the packet format defined in ITU-T Rec. T.38. The Fax Relay media state is included in the MIME media type "image".

C.2.5 Text Relay (TR)

This media stream is a simple sequence of text characters. This is primarily used in TDD (Telecommunications Device for the Disabled) applications.

C.3 RTP packet format for state signalling events

In conformance with the Internet Protocol, all fields are carried in network byte order, that is, most significant byte (octet) first. Within a byte, the most significant bit is transmitted first. This byte order is commonly known as big endian. In this Recommendation, bytes and bits shown on the left are more significant.

C.3.1 Use of RTP header fields

SSRC: The use of the SSRC field for State Signalling Events is in accordance with the RTP protocol. Regardless of the use of the SSRC field, an SSE stream qualifies all media streams associated with the session it is embedded within.

Timestamp: The RTP timestamp reflects when a local decision is made to issue the SSE message. An SSE message can have the same timestamp as another RTP packet such as an audio packet or an RFC 2833 event packet.

Marker bit: Since a SSE has no duration, the marker field is a "do not care" condition. A SSE, receiver shall ignore the RTP marker bit. Transmitters should be set to zero.

C.3.2 RTP payload format

The payload format for the State Signalling Events is shown in Figure C.1. The dashed-line boxes represent fields added to the payload only if the Extension bit (X) is set.

<table>
<tr><td>0</td><td>1</td><td>2</td><td>3</td><td>4</td><td>5</td><td>6</td><td>7</td><td>8</td><td>9</td><td>10</td><td>11</td><td>12</td><td>13</td><td>14</td><td>15</td><td>16</td><td>17</td><td>18</td><td>19</td><td>20</td><td>21</td><td>22</td><td>23</td><td>24</td><td>25</td><td>26</td><td>27</td><td>28</td><td>29</td><td>30</td><td>31</td></tr>
<tr><td colspan="6">Event (6 bits)</td><td>F</td><td>X</td><td colspan="8">Reason identifier code (8 bits)</td><td colspan="16">Reason identifier code information (16 bits)</td></tr>
<tr><td colspan="5">Reserved (5 bits)</td><td colspan="11">Optional extension field length (11 bits)</td><td colspan="16"></td></tr>
<tr><td colspan="32">Optional extension field (0-2047 bytes)</td></tr>
</table>

Figure C.1/V.150.1 – Payload format for state signalling events

Event:	The event field is used to code local media states as shown in C.5.2. A six-bit field is used. The "event" referred to is SSE message exchange, which may or may not be accompanied by a local media state change. The range 32-63, inclusive, is reserved for vendor-defined SSE events.
F:	"Force Response" bit. This bit is meaningful only if an explicit acknowledgement feature is negotiated at call establishment (Annex E for SDP and Annex F for H.323). A value of binary-one forces the other end to send an SSE response containing its local media state. If the explicit acknowledgement feature has not been negotiated, then this bit should be set to binary-zero by the transmitter, and must be ignored by the receiver.
X:	"Extension" bit. If this field is set to a value of binary-zero, then the SSE payload does not have a payload extension at the end. Otherwise, there is a payload extension. The payload extension length field and the five reserved bits preceding it, are present only if there is a payload extension. The Extension bit must be set to binary-one if a vendor-defined local media state is indicated, or if an explicit acknowledgement feature is negotiated at call establishment (Annex E for SDP and Annex F for H.323).
Reason Identifier Code (RIC)	This eight-bit code indicates the rationale for sending the SSE message. The rationale may be local event detection, a received SSE message, or a combination of local events and received SSE messages. A value of all zeros is a null RIC code, indicating non-communication of the rationale for sending an SSE message. When the RIC is 0, a receiver may assume a default value. Distinct events may share RICs, or may have RICs that are unique to the event. However, note that the RIC space for each event is distinct.
Reason identifier code information	This sixteen-bit field is used to provide additional information associated with the RIC. For instance, if the RIC refers to the modem CM (call menu) signal, this field may be used to indicate the CM information. If the RIC is null, then this field is always null regardless of its value. A value of all zeros indicates a null RIC information field.

Reserved:	Five bits reserved for use by ITU-T. Each of these shall be set to binary-zero. This field exists only if the extension (X) bit is set to binary-one.
Extension length:	This optional eleven-bit field is used to indicate the number of extension bytes after it. A value of all zeros in the extension length field, although permissible, is not useful. This field exists only if the extension (X) bit is set to binary-one.
Extension field:	This field consists of a variable number of bytes (0-2047), indicated by the extension length field.

When a vendor-defined event (32-63) is indicated in an SSE message, the first byte of the extension field must indicate a vendor-specific data tag. The extension field for this case is shown in Figure C.2. If the value of this field is in the range 1-255, it is dynamically mapped, at call establishment, to a vendor identification parameter value (Annex E for SDP and Annex F for H.323). A vendor-specific data tag value of 0 is a null value. In this case, the identity of the vendor must be known by other means such as a configurable or fixed default.

<table>
<tr><td>0</td><td>1</td><td>2</td><td>3</td><td>4</td><td>5</td><td>6</td><td>7</td><td>8</td><td>9</td><td>10</td><td>11</td><td>12</td><td>13</td><td>14</td><td>15</td><td>16</td><td>17</td><td>18</td><td>19</td><td>20</td><td>21</td><td>22</td><td>23</td><td>24</td><td>25</td><td>26</td><td>27</td><td>28</td><td>29</td><td>30</td><td>31</td></tr>
<tr><td colspan="8">Vendor Index (8 bits)</td><td colspan="24" rowspan="2">Remainder of the optional extension field (0-2046 bytes)</td></tr>
<tr><td colspan="8"></td></tr>
</table>

Figure C.2/V.150.1 – SSE Extension field for vendor-defined media states

If an explicit acknowledgement feature is negotiated at call establishment (Annex E for SDP and Annex F for H.323), the last six bits of the first octet of the extension field indicate the endpoint's or media gateway's perception of the remote end's media state. The first two bits are padded to binary-zero, as indicated by the Pad (P) bits in Figure C.3. The values used for Remote Media State are listed in Table C.1.

<table>
<tr><td>0</td><td>1</td><td>2</td><td>3</td><td>4</td><td>5</td><td>6</td><td>7</td><td>8</td><td>9</td><td>10</td><td>11</td><td>12</td><td>13</td><td>14</td><td>15</td><td>16</td><td>17</td><td>18</td><td>19</td><td>20</td><td>21</td><td>22</td><td>23</td><td>24</td><td>25</td><td>26</td><td>27</td><td>28</td><td>29</td><td>30</td><td>31</td></tr>
<tr><td>P</td><td>P</td><td colspan="6">Remote Media State
(6 bits)</td><td colspan="24" rowspan="2">Remainder of the optional extension field (0-2046 bytes)</td></tr>
<tr><td colspan="8"></td></tr>
</table>

Figure C.3/V.150.1 – SSE Extension field with explicit acknowledgement feature

When a vendor-defined event (32-63) is indicated in an SSE message and an explicit acknowledgement feature is negotiated at call establishment (Annex E for SDP and Annex F for H.323), then the first and second bytes of the extension field indicate a vendor-specific data tag and an appropriately padded, remote media state respectively. This is depicted in Figure C.4.

<table>
<tr><td>0</td><td>1</td><td>2</td><td>3</td><td>4</td><td>5</td><td>6</td><td>7</td><td>8</td><td>9</td><td>10</td><td>11</td><td>12</td><td>13</td><td>14</td><td>15</td><td>16</td><td>17</td><td>18</td><td>19</td><td>20</td><td>21</td><td>22</td><td>23</td><td>24</td><td>25</td><td>26</td><td>27</td><td>28</td><td>29</td><td>30</td><td>31</td></tr>
<tr><td colspan="9">Vendor Index (8 bits)</td><td>P</td><td>P</td><td colspan="5">Remote Media State (6 bits)</td><td colspan="16"></td></tr>
<tr><td colspan="32">Remainder of the optional extension field (0-2045 bytes)</td></tr>
</table>

Figure C.4/V.150.1 – SSE Extension field with explicit acknowledgement feature

C.4 Reliability

Three options exist for ensuring SSE reliability. These are:

1) Simple SSE repetition as defined in C.4.1. This option is not declared at call establishment time. As the default option, it is used if one of the remaining two options is not declared. Note that it is permissible to set the number of transmissions to one (no redundancy).
2) Use of RFC 2198-based redundancy for SSEs (see C.4.2). This must be explicitly declared at call establishment.
3) Explicit acknowledgement of SSEs (see C.4.3). This scheme is based on the inclusion, in an SSE message, of the value of the endpoint's or gateway's rmt_mode variable, which indicates its view of the remote media state. Additionally, a gateway or endpoint may force the other end to respond with an SSE by setting the Forced Response (F) bit. To be used, this option must be explicitly declared by both ends at call establishment time.

C.4.1 Use of packet repetition

Redundant transmission of State Signalling Event messages is the default means for ensuring SSE reliability. The number of redundant transmissions and the inter-transmission interval may be provisionable parameters, with three transmissions and 20 ms as the default. Since these parameters can be selected independently by each end, they need not be negotiated. When the number of transmissions is set to one (no redundancy), the inter-transmission interval does not apply.

A receiver acts upon the first instance of the message it receives. Except for incrementing the sequence number count, it shall ignore the remaining redundant SSE messages. Although they may have different sequence numbers, redundant SSE messages shall have identical timestamps. RTP packets with another format (e.g., PCMU) may be placed in the interval between redundant SSE messages. If this is done, then the timestamps pertaining to the composite media stream will not increase monotonically.

C.4.2 Use of RFC 2198-based redundancy

It is possible to combine the SSE payload with other RTP payloads, including itself, within an RFC 2198 payload. If this combination is used, the association between the different constituents of an RFC 2198 payload shall be defined at session establishment using H.245 or other protocol such as SDP (RFC 2327).

Simple repetitive redundancy and RFC 2198 redundancy shall not be simultaneously used for SSE.

An implementation need not support RFC 2198 encapsulation of SSEs to be compliant with this annex.

C.4.3 Explicit acknowledgement of SSEs

Explicit acknowledgement is an optional procedure that is used on a call-by-call basis only if both endpoints of a call support it.

During call set up, each endpoint indicates whether it supports explicit acknowledgement. If both endpoints of a call indicate they support the procedure, it may be used for the call.

C.4.3.1 Explicit acknowledgement variables

If an endpoint supports the explicit acknowledgement procedure, it must implement three variables, two timers, and a counter, as follows:

- variable *lcl_mode* indicates the current media state of the local endpoint (i.e., the value that will be sent to the remote gateway or endpoint in the Event field of an SSE message);
- variable *rmt_mode* indicates the last known media state of the remote endpoint, as known by the local endpoint (i.e., the value that will be sent to the remote gateway or endpoint in the Remote Media State Field of an SSE Extension field with explicit acknowledgement);
- variable *rmt_ack* indicates the last known mode of the local endpoint known by the remote endpoint, as known by the local endpoint (i.e., the value that was received from the remote gateway or endpoint in the Remote Media State field of an SSE Extension field with explicit acknowledgement);
- timer *t0* is used to control sending mode change messages to the remote endpoint;
- timer *t1* is used to recover from lost acknowledgements sent by the remote gateway; and
- counter *n0* is used to control sending mode change messages to the remote endpoint.

The timers, when non-zero, decrement toward zero with real time, stopping when they reach zero. Timer *t0* when started is set to the value *t0interval*, similarly with *t1* and *t1interval.* Counter *n0* is set to the value *n0count* when it is initialized.

The values *t0interval*, *t1interval*, and *n0count* may be either static in an endpoint implementation, or determined dynamically during the call based on network statistics. If static, values of 10 milliseconds, 300 milliseconds, and 3 are recommended for *t0interval*, *t1interval*, and *n0count* respectively.

If determined dynamically, then the following values are recommended.

Let:	**Be:**
P	the probability that a packet sent by one MoIP endpoint through the packet network will be successfully received by the other endpoint
T	the latency that can be tolerated in the delivery of mode updates
Q	the reliability required in the delivery of mode updates within the given latency
RTD	the round trip delay through the packet network between the two endpoints
OWD ≈ RTD/2	the one way delay through the packet network from one endpoint to the other
Then the value of:	**is:**
n0count	floor(log(1 – q)/log(1 – p))
T0interval	max(0, (OWD – t)/(N0count – 1))
T1interval	1.5 × RTD

C.4.3.2 Explicit acknowledgement procedures

If the explicit acknowledgement procedure is being used for a call, the endpoints shall execute the following procedures.

When an endpoint's MoIP application goes to a new mode, it:

- sends an SSE message to the other endpoint containing the current value of the variables lcl_mode and rmt_mode to the other endpoint, with the must respond flag set to FALSE;
- sets counter n0 to the value n0count;
- sets timer t0 to t0interval (even if it was non-zero); and
- sets timer t1 to t1interval (even if it was non-zero).

When:

- timer t0 decrements to 0;
- counter n0 is not equal to 0; and
- the value of lcl_mode is not equal to the value of rmt_ack.

The endpoint sends an SSE message to the other endpoint exactly as above except:

- counter n0 is decremented rather than set to n0count;
- timer t1 is not set; and
- the must respond flag is set to TRUE if the value of timer t1 is zero.

NOTE – If timer t0 decrements to 0 and counter n0 is equal to zero, no action is taken until timer t1 decrements to 0.

When:

- timer t1 decrements to 0;
- counter n0 is equal to 0; and
- the value of lcl_mode is not equal to the value of rmt_ack.

The endpoint sends an SSE message to the other endpoint exactly as first given above except:

- counter n0 is not decremented, it is left equal to zero;
- timer t0 is not set (It too is left equal to 0.); and
- the must respond flag is set to TRUE.

Upon receipt of an SSE message from the other endpoint:

- if the message is duplicated or out of sequence (determined using the RTP header sequence number), the endpoint ignores the received message.

If the message is not ignored, the endpoint:

- sets the values of rmt_mode and rmt_ack to the values in the message; and
- if:
  - the message contained a new value for the remote endpoint's mode; or
  - the message's must respond flag is set to TRUE,

  the endpoint sends an SSE message to the other endpoint exactly as first given above except counter n0 and timers t0 and t1 are not (re)set.

C.5 State signalling event definitions

C.5.1 Protocol extension mechanism

Apart from vendor-defined events (range 32-63), the base SSE protocol definition in this annex may not be extended without revising this Recommendation. Further, it shall be possible for a media gateway or endpoint to ignore vendor-defined events and related RICs, extensions etc., without impacting the operation of a MoIP gateway.

C.5.2 List of state signalling events

The encoding of events indicating the media states defined in clause C.2 is shown in Table C.1. The encoding is the same regardless of whether an endpoint or media gateway is indicating its local media state, or its perception of the remote media state.

A gateway that supports the SSE protocol for MoIP must be able to understand events 1-3. The ability to understand the remaining events shall not be presumed unless they are explicitly declared at call establishment.

Table C.1/V.150.1 – Coding of the media states

Event encoding (Decimal)	Indicated media state
0	Reserved for future use by ITU-T
1	Initial Audio
2	Voice Band Data (VBD)
3	Modem Relay
4	Fax Relay
5	Text Relay
6-31	Reserved for future use by ITU-T
32-63	Vendor-defined

C.5.3 SSE protocol operation

For a media stream (port) or set of media streams (ports or "flow") governed by an SSE payload type value, the "local" media state represents the view of the gateway or end-point regarding the media state. The "remote" media state represents the corresponding view of the remote endpoint, communicated via the SSE. The "SSE protocol state" for the port, or ports, is a pair consisting of the local and remote media states.

The local state (represented by S) can take on the following values (based on clause C.2):

a: Initial Audio;

v: Voice Band Data (VBD);

m: Modem Relay (MR);

f: Fax Relay (FR);

t: Text Relay (TR).

In addition to all of these values, the remote state (represented by S') may take on the following value:

i: indeterminate.

For the port(s) governed by an SSE payload type value, an SSE protocol state, P, which is a composite of the local and remote media states is defined and expressed as a pair, P = (S, S').

Upon initialization, the SSE protocol state, (S, S'), is set to (a, a). Thus, the initial audio state is the "base" state of any SSE-driven media state machine.

C.5.3.1 SSE generation rules

Consider a change in the SSE protocol state from P1 = (S1, S1') to P2 = (S2, S2'), where one or both of the following propositions is true:

S1 is not the same as S2;

S1' is not the same as S2'.

On any change in the SSE protocol state from P1 = (S1, S1') to P2 = (S2, S2'), an SSE indicating media state S2 shall be sent to the remote gateway or end-point.

An exception to this is the case where the following propositions are all true: S1' is not the same as S2', S1 is the same as S2 and S2 is the same as S2' are all true. In this case, the last SSE message received is a response to a prior SSE sent to the remote end, indicating that the remote end has switched to a media state S2' that is identical to the local media state S2, which is itself unchanged. In this case, an SSE indicating media state S2 shall not be sent to the remote end.

In the context of protocol error recovery, the gateway or end-point is permitted to resend the initial audio SSE on timeout expiration (see C.5.3.3), even though there has been no change in protocol state.

It is not the intention of this annex to delineate all the internal triggers that can cause the local media state, S1, to change. One such trigger is the receipt of an SSE (see C.5.3.2). Other local triggers are specific to the MoIP, FoIP, and ToIP applications.

C.5.3.2 Media state transition rules

Upon receipt of a new SSE message, the remote media state, S', is set to the media state (a, v, f, m, t) indicated in the SSE.

The setting of the local media state, S, on receipt of a new SSE message depends upon factors such as:

- Permitted media states. See the list below.
- Current resource availability.
- Support of the media state by the design.

The permitted local media states on receipt of a new SSE are:

Table C.2/V.150.1 – Permitted media states

Rule	Condition	Description
1	If (S' is a), then S = a	When an SSE indicating the initial audio media state is received, a compliant implementation shall change the local media state (for the port(s) in question) to initial audio. There is no choice, since the initial audio media state is the base state of any SSE-driven media state machine.
2	If (S' is v), then S = a or v	This rule allows the possibility of not changing to a VBD media state when the other side indicates that it has changed media state to VBD. Such a choice is application-specific.
3	If (S' is m), then S = a or v or m	Rules 3, 4 and 5 allow the possibility of not setting the local media state to match the remote media state, and instead, selecting initial audio (a) or VBD (v) as the local state. Such a choice is application-specific.
4	If (S' is f), then S = a or v or f	
5	If (S' is t), then S = a or v or t	

These rules limit the freedom of a compliant endpoint or gateway with respect to local media state change on receipt of an SSE message. Any contravention of these rules by the remote gateway or endpoint shall be handled via the recovery procedures in C.5.4. Implementations may further limit the range of values for the local media state in response to an SSE.

C.5.3.3 The use of the P' RIC

This Recommendation defines a RIC called P' which is used by the SSE protocol to indicate that a gateway sending the P' RIC is following the rule defined in this annex, whereby a gateway sends such an indication upon recognizing that the remote gateway's SSE protocol state has changed.

C.5.4 Protocol error recovery

C.5.4.1 No explicit SSE acknowledgement

The protocol error recovery mechanism in this clause is applicable only if the explicit SSE acknowledgment procedures (see C.4.3) are not being used.

This recovery mechanism aims at setting both sides to the initial audio media state which is the "base", "ground" or "reset" state for the SSE protocol state machine. This mechanism may be used in lieu of terminating the session (clearing the call) when the following conditions occur:

1) Inability to comply with the rules of C.5.3.2. This covers the receipt of out-of-context SSE, and an inability to make one of the permitted local media state transitions for reasons such as resource unavailability.
2) If S is not the same as S' (local and remote media state not the same) for more than T2 seconds (defined below).
3) If the received payload type and/or packet format is inconsistent with the local media state, S1, for more than T2 seconds (defined below). Note that this allows asymmetrical payload types, but not asymmetrical media states. Sessions with media states, as defined in this annex, that are directionally asymmetrical are out of the scope of this Recommendation.

Note that the list of conditions, above, that trigger protocol recovery is not necessarily exhaustive. In lieu of protocol recovery, media gateways may be designed or provisioned to terminate the session (clear the call) when one or more of these conditions arise. No interoperability problems arise when the two ends are designed or provisioned differently, since clearing the call, by definition, pre-empts an attempt to reset both sides to the initial audio media state.

Protocol recovery, as defined here, consists of the following sequence of actions:

1) Set S to a and S' to i (local media state set to audio, remote media state set to indeterminate).
2) Send an SSE indicating the initial audio state. This is to be repeated every T1 seconds (defined below) until S' is a. If S' is not the same as a after N tries, then the session shall be terminated.

The following is a list of timeouts associated with this recovery procedure:

1) **T1**. Repeat interval for initial audio SSEs used for resetting the SSE protocol. This may be provisionable. The recommended default is 1 second. Packet repetition during normal operation (see C.4.1). The SSE redundancy options described in clauses C.4.1 and C.4.2 may be used in addition to SSE repetitions that are meant to reset the SSE protocol. The default value of N (number of retries) shall be five.

2) **T2**. Transience interval for media states. This is the time interval for which an inconsistency in the local and remote media states is permitted. Theoretically, this can be made equal to the Round Trip Delay for SSE, plus a margin for processing delays, delay fluctuations etc. In practice, it is not always possible to set this parameter separately for each possible session. A value that is large enough for all connections should be chosen; a recommended default is 1 second.

The timeouts, T1 and T2, and the retry count, N, may be provisionable.

C.5.5 SSE reason identifier codes

The SSE Reason Identifier Codes (RIC) values and formats for MoIP gateways are described in 15.3.1.

Annex D

Procedures for voiceband data only mode of operation

This annex is reserved and the procedures for Voice Band Data only mode of operation are for further study. Equipment that implements VBD-only mode of operation that will be defined by this annex will be compatible but not compliant with this Recommendation.

Annex E

SDP description of sessions supporting SPRT-based modem relay

This annex describes how to use SDP, as defined in IETF RFC 2327, to describe sessions that support SPRT-based modem relay. SDP descriptors that address modem relay functionality shall comply with RFC 2327.

As defined in RFC 2327, SDP keywords are case-significant. This includes SDP parameter values unless a field defines it otherwise. Following MIME conventions, names and values associated with MIME definitions are case-insignificant. This inconsistency is inherited from prior standardization.

E.0 Abbreviations

This annex uses the following abbreviations:

AVP Audio/Video Profile

FEC Forward Error Correction

FID Flow Identification

JM Joint Mode

MIME Multipurpose Internet Mail Extensions

PCMA Pulse Code Modulation, A-law

PCMU Pulse Code Modulation, Mu-law

SDP Session Description Protocol

XID Exchange Identification

E.1 Introduction

This clause describes the representation of information without which an SPRT-based modem relay capability cannot be described.

The following subclauses of E.1 define the SDP objects that are required in the context of a SPRT-based modem relay. SDP objects that are normally needed to describe RTP media are not redefined here. However, they are included, for completeness, in the examples.

For SPRT-based modem relay sessions that conform to this Recommendation, it is assumed that the session is initialized in a non-Modem Relay, non-VBD mode. Contingent upon the detection of certain events, SSE-based coordination is used to transition the session to VBD or Modem Relay. Items that must be included in the description of such sessions are:

1) A declaration of the UDP-based SPRT transport protocol, along with the UDP port and payload type associated with it (see E.1.1 and E.1.2).
2) A VBD media description (see E.1.4). The call-originating end must include either PCMA or PCMU (or both) in the list of VBD codecs, though other VBD codecs may be additionally specified. The call-terminating end must indicate support for at least one VBD codec, which need not be PCM-based.
3) Declaration of support for the following RFC 2833 events: ANS (32), /ANS (33), ANSam (34) and /ANSam (35).
4) Declaration of support of SSE signalling (Annex C), along with the UDP port and payload type associated with it. To meet the timing constraints imposed by modem interaction, SSE signalling is used for media state changes by implementations that comply with this Recommendation.

E.1.1 Description of SPRT media on a dedicated UDP port

This clause addresses the case where a UDP port is dedicated for use by the MR media. Note that, by including media-level connection lines, it is also possible to separate the MR and non-MR (RTP) streams in a session on distinct IP addresses.

A media information ('m' line) is described in RFC 2327 as:

```
m=<media> <port> <transport> <fmt list>
```

For V.150.1 sessions, the <media> parameter is set to 'audio'.

The <transport> parameter for V.150.1 media is assigned a value of 'udpsprt'. This refers to Simple Packet Relay Transport over UDP. Per RFC 2327 convention, the <transport> parameter is case-sensitive. Transmitters must build a lower-case value; receivers may be case-tolerant.

The <fmt list> consists of one dynamically assigned payload type in the range 96-127.

The 'sprtmap' attribute is defined below to map this payload type into the modem relay payload format, v150mr:

```
a=sprtmap:< payload type> v150mr/<clock rate>
```

In this context, the clock rate refers to the rate at which the voiceband modem signal is sampled prior to its conversion into the modem relay format.

An example of the use of the 'm' line to describe SPRT-based modem relay media is:

```
m=audio 49232 udpsprt 98
a=sprtmap:98 v150mr/8000
```

E.1.2 Switching a UDP port between MR and non-MR media

This clause addresses the case in which a single UDP port assigned to non-MR (e.g., RTP) media may be later switched to MR. In this case, MR is described as a latent capability embedded in a capability set (RFC 3407). This is illustrated by the following SDP lines that allow destination UDP port # 49230 to be switched between the RTP/AVP and modem relay payload formats:

```
m=audio 49230 RTP/AVP 0 2 8
a=sqn:0
a=cdsc:1 audio udpsprt 100
a=cpar:a=sprtmap:100 v150mr/8000
```

In this example, an SPRT payload type of 100 is used if port # 49230 is switched to MR. As in clause E.1.1, the 'sprtmap' attribute is used to map this payload type into the modem relay payload format, v150mr. Capability set usage conforms to the syntax rules of RFC 3407. This capability set is assigned a sequence number of 0 ('sqn'), and the embedded MR capability is numbered 1.

E.1.3 Description of state signalling event protocol support

The SSE protocol shall be declared as a dynamic RTP/AVP payload type, as shown below. The RPT/AVP encoding name 'v150fw' indicates the SSE protocol as defined in this Recommendation.

```
m=audio 3456 RTP/AVP 0 15 96
a=rtpmap:96 v150fw/8000
```

Declaration of SSE support may optionally be accompanied by a format-specific 'fmtp' attribute line that lists all supported events that are not vendor-specific:

```
a=fmtp:<v150fw payload type> <list of supported events>
```

The supported events are listed as comma-separated elements. Each element can either be a single integer, or two integers separated by a hyphen. In the latter case, a range of events is indicated. No white space is allowed. Lists with one element are permitted. The V.150.1 events 1 through 3 need not be explicitly declared in the SDP.

For example,

```
m=audio 3456 RTP/AVP 0 15 96
a=rtpmap:96 v150fw/8000
a=fmtp:96 4,5
```

indicates support for the V.150.1 events 1 through 3, and events 4 and 5. When this optional 'fmtp' attribute is omitted, support of events 1 through 3 as defined in Table C.1 is implied by default.

Vendor-specific events may be declared as vendor-specific parameters (see E.2.2.2).

The scope of the SSE messages should extend to all media streams declared in the SDP session descriptor that are covered by the set of SSE protocol states as defined in this Recommendation. SSE messages are used to coordinate the assignment of resources to these media streams. These media streams may span several ports, or may share a single port. If they span several ports, they may be sent to the same or different connection addresses (e.g., IP addresses).

Three options exist for ensuring SSE reliability. These are:

1) Simple SSE repetition as defined in Annex C. This is not declared at call establishment time. This is the default option to be used if any of the remaining two options are not declared.
2) Use of RFC 2198 for SSE. This shall be explicitly declared at call establishment.
3) Inclusion, in the SSE message, of the endpoint's or gateway's view of the far-end media state (S'). To be used, this option must be explicitly declared by both ends using an optional, format-specific Boolean parameter, '*expack*' ('Explicit Acknowledgement') as

defined below. The possible values of this parameter are 'yes' and 'no'. The default, when this parameter is omitted, is 'no'. If only one side enables this option, then this method is not used. It is if, and only if this parameter is enabled that the "Must Respond Bit" defined in C.3.2 may be used. Otherwise, the value of this bit is a do not care.

```
a=fmtp:<sse payload type> expack=yes
```

The following example:

```
m=audio 3456 RTP/AVP 0 15 96
a=rtpmap:96 v150fw/8000
a=fmtp:96 expack=yes
```

indicates a gateway's consent to the reliability procedure in line item (3), above.

E.1.4 Description of VBD media

The 'gpmd' (general-purpose media descriptor) attribute is used to associate payload types in a media information ('m') line with VBD. The general form of this attribute line is:

```
a=gpmd:<format><parameter list>
```

In the context of VBD declaration, the <format> must be an RTP payload type. The <parameter list> is a semicolon-separated list of "parameter = value" pairs. For RTP formats, these pairs address parameters that are not part of their standard MIME definition. For sessions supporting this Recommendation, the parameter of interest is the Boolean 'vbd' that may have the value of 'yes' or 'no'.

The payload type marked for Voice Band Data (VBD) treatment may be a static payload type or a dynamic payload type. It is possible that a codec, such as PCMU, be declared with both static and dynamic payload types, with only one of the two marked use with for Voice Band Data.

```
m=audio 3456 RTP/AVP 0 15 98 99
a=rtpmap:98 PCMU/8000
a=gpmd:98 vbd=yes
a=rtpmap:99 G726-32/8000
a=gpmd:99 vbd=yes
```

In the example directly above, static payload type '0' and dynamic payload type '98' each represent the encoding format 'PCMU'. The payload type '0' is not associated with VBD. The payload types '98' (PCMU) and '99' (32 kbit/s ADPCM) are however associated with VBD.

```
m=audio 3456 RTP/AVP 0 18 98
a=gpmd:0 vbd=yes
a=rtpmap:98 G726-32/8000
a=gpmd:98 vbd=yes
```

In this example, the static payload type of 0 (PCMU) is marked for VBD treatment, along with the dynamic payload type '98' (mapped to 32 kbit/s ADPCM).

E.1.5 Description of mandatory V.150.1 attributes

Although all attributes are optional at the SDP parser-level, some might be made mandatory at the application level. For applications that comply with this Recommendation, the following parameters are mandatory:

1) Modem relay type, 'mr'. Allowed values are 0 (V-MR) and 1 (U-MR).

2) Media gateway type, 'mg'. Allowed values are: 0 ('No Trans-compression'), 1 ('Single Trans-compression') and 2 ('Double Trans-compression').

3) Call discrimination mode select, 'CDSCselect'. This indicates preference for one of three call discrimination modes (see 20.3). Allowed values are: 1 ('audio (RFC 2833)'), 2 ('VBD-preferred'), 3 (Mixed).

4) List of V-series modulations supported in Modem Relay mode by the gateway, 'mrmods'. These modulations are listed as one or more comma-separated elements, where each element is either a single integer or two integers separated by a hyphen. No white space is allowed. The integers, which designate modulation types, are defined in Table E.1.

Table E.1/V.150.1 – Coding of modulation types in the 'mrmods' list

Modulation type	Integer representation
V.34 duplex	1
V.34 Half-duplex	2
V.32 *bis*/V.32	3
V.22 *bis*/V.22	4
V.17	5
V.29 half-duplex	6
V.27 *ter*	7
V.26 *ter*	8
V.26 *bis*	9
V.23 duplex	10
V.23 half-duplex	11
V.21	12
V.90 analogue	13
V.90 digital	14
V.91	15
V.92 analogue	16
V.92 digital	17

5) Boolean parameter, 'jmdelay'. This parameter indicates the ability of a gateway to support the JM delay procedure as defined in clause 20.7. Values are "yes" and "no".

These parameters are included in the MIME definition, audio/v150mr. On this basis, they shall be declared as format-specific parameters, using the 'fmtp' attribute:

```
a=fmtp:<ITU V.150.1 payload type> <parameter list>
```

where <parameter list> is a list of <parameter>=<value> pairs delimited by semicolons and optional white space(s). For example, "mr=1" and "mg=1" are shown in the third line of the following example:

```
m=audio 49232 udpsprt 98
a=sprtmap:98 v150mr/8000
a=fmtp:98 mr=1; mg=1;CDSCselect=3;mrmods=1-4,10-12,14,17;jmdelay=no
```

It is also permissible to place each <parameter>=<value> pair in a separate 'fmtp' attribute line, as depicted below:

```
m=audio 49232 udpsprt 98
a=sprtmap:98 v150mr/8000
a=fmtp:98 mr=1
a=fmtp:98 mg=1
```

```
a=fmtp:98 CDSCselect=3
a=fmtp:98 mrmods=1-4,10-12,14,17
a=fmtp:98 jmdelay=no
```

E.2 Optional information

This clause describes the SDP representation of information that may be optionally declared at session establishment time. In the absence of their declaration, media gateways may determine this information, where applicable, using fixed defaults or by parameters configured via a management interface.

E.2.1 Description of transmission fault tolerance

As with any RTP payload format, codecs marked for VBD treatment may be subject to:

1) RFC 2198 packet redundancy.

2) RFC 2733 Forward Error Correction with a separate FEC stream.

3) RFC 2733 Forward Error Correction combined with RFC 2198 packet redundancy.

Per RFC 2198 and RFC 2733, use of these fault tolerance schemes shall be contingent upon their declaration at session establishment time. For SDP, their declaration must be in strict conformance with the SDP rules outlined in the applicable IETF RFC (RFC 2198 and/or RFC 2733). Although these rules are not repeated here, declaring RFC 2198 support for a VBD codec is illustrated with an example:

```
m=audio 3456 RTP/AVP 0 15 102
a=gpmd:0 vbd=yes
a=rtpmap:102 red/8000
a=fmtp:102 0/0
```

Examples of the declaration of FEC support are found in RFC 2733. RFC 2733 describes the use of an 'fmtp' line to associate a separate FEC stream with an IP address and port. Here, 'separate' means that RFC 2198 redundancy is not used to combine the FEC information with the media stream it qualifies. When a separate FEC stream is sent to the same IP address and port (albeit a different SSRC), as the media stream it qualifies, then there is no need for the 'fmtp' line to associate the 'parityfec' payload type with an IP address and port. Thus, in the following SDP segment:

```
c=IN IP4 224.2.17.12
t=0 0
m=audio 49170 RTP/AVP 0 15 78
a=gpmd:0 vbd=yes
a=rtpmap:78 parityfec/8000
a=fmtp:78 49170 IN IP4 224.2.17.12
```

The last line is superfluous and may be omitted. Likewise, the absence of a line associating an IP address and port with a FEC stream shall be construed to mean that the FEC stream is sent to the same IP address and port as the media stream it qualifies.

E.2.2 Description of the optional attributes of sessions capable of modem relay

These are classified into two groups: attributes associated with the SPRT protocol, and attributes associated with the V.150.1 modem relay media format. Recall that SPRT is a general-purpose transport protocol represented in the media ('m=') line by a transport parameter value of 'udpsprt'. On the other hand, 'v150mr' is one of the possible media formats that can use this transport protocol.

E.2.2.1 Optional SPRT protocol parameters

The optional parameters associated with the SPRT transport protocols are declared via the SDP attribute 'sprtparm'; this line has the format:

```
a=sprtparm:<maxPayload0> <maxPayload1> <maxPayload2> <maxPayload3>
<maxWindow1> <maxWindow2>
```

The parameters maxPayload0, maxPayload1, maxPayload2, maxPayload3, maxWindow1 and maxWindow2 represent integer values as shown in Table E.2 below. Any of these parameters can be omitted by setting it to '$'. When this is done, the default values shown in Table E.2 shall be used. Default values shall also be used when the 'sprtparm' attribute line is omitted.

Table E.2/V.150.1 – Definition and values of 'sprtparm' parameters

Parameter	Definition	Value Range	Default
maxPayload0	Maximum payload size of SPRT channel 0 in bytes	Integer 140-256	140
maxPayload1	Maximum payload size of SPRT channel 1 in bytes	Integer 132-256	132
maxPayload2	Maximum payload size of SPRT channel 2 in bytes	Integer 132-256	132
maxPayload3	Maximum payload size of SPRT channel 3 in bytes	Integer 140-256	140
maxWindow1	Maximum window size of SPRT channel 1 in bytes	Integer 32-96	32
maxWindow2	Maximum window size of SPRT channel 2 in bytes	Integer 8-32	8

Examples of the use of this optional attribute are:

```
a=sprtparm:160 200 220 200 40 25

a=sprtparm:180 100 $ 240 40 25

a=sprtparm:220 200 $ $ $ $
```

If an originating gateway proposes a value for one of the 'sprtparm' parameters, any corresponding value proposed by the terminating gateway must then be equal to, or less than, the value proposed by the originating gateway. Note that either gateway may set this value to '$'.

It is permissible to omit trailing '$' tokens at the end of the 'sprtparm' line. For instance,

```
a=sprtparm:200 $ $ $ $ $
```

is equivalent to

```
a=sprtparm:200
```

E.2.2.2 Optional vendor-specific parameters

The 'vndpar' (vendor parameters) attribute may be used to declare vendor codes for coordinating enhanced operation over and above the V.150.1 modem. It shall be possible to safely ignore vendor-specific parameters and still maintain interoperability with equipment conforming to this Recommendation. Hence, proprietary enhancements cannot be a substitute for the basic features required for compliance with this Recommendation.

The format of the 'vndpar' attribute line is as follows:

```
a=vndpar:<vendorIDformat> <vendorID> <vendorSpecificDataTag>
         [<vendorSpecificData>]
```

The <vendorIDformat>, a decimal, indicates the format of the following <vendorID> field. The following values are defined:

Integer representation	Vendor ID format
1	ITU-T Rec. T.35
2	IANA private enterprise number

The <vendorID> may be represented in hex or in decimal format. If represented in hex, it has a '0x' prefix. Generally, if the vendor ID format is T.35, the hexadecimal format is preferred. If it is the IANA private enterprise number (http://www.iana.org/assignments/enterprise-numbers), the decimal format is preferred.

When the vendor ID format is T.35, the vendor ID consists of a country code followed by a vendor code. The country code consists of four octets and the vendor ID consists of two octets. If the representation of the vendor ID is hexadecimal, leading zeros in the country code may be omitted, while leading zeros in the vendor code may not be omitted.

When the <vendorID> is the vendor's private enterprise number, leading zeros may be omitted.

The <vendorSpecificDataTag> is a decimal integer between 0-255. If used, values in the range 1-255 are uniquely mapped, via the 'vndpar' attribute, to the combination of the vendor specified in the <vendorID> and the proprietary capabilities indicated by <vendorSpecificData>. This mapping, which exists for the duration of a session, does not persist across sessions. Further, each side may choose this integer independently of the other end. Due to the compactness of this index, a gateway or endpoint may use it in a number of places such as in SSE messages (Annex C). A value of 0 is a null value. When present, it is equivalent to omitting the <vendorSpecificDataTag>. A null value of the <vendorSpecificDataTag> is not associated with any vendor ID.

It shall be possible for an endpoint or gateway to declare multiple (1-255) 'vndpar' attribute lines in an SDP session description. Each of these lines may indicate a different vendor. In addition, multiple 'vndpar' lines may indicate the same vendor. When multiple 'vndpar' lines are declared in an SDP session descriptor, each value of <vendorSpecificDataTag> must either be unique within all 'vndpar' lines in the session descriptor or null (0). If non-null, the <vendorSpecificDataTag> may serve as a dynamically assigned, feature identifier for the vendor.

Inclusion of the parameter `<vendorSpecificData>` is optional. When included, this is a vendor-defined octet string consisting of one or more octets. Since it consists of an integer number of octets, it is represented by an even number of hex characters. No '0x' prefix is needed. No size limitation is specified since SDP parsers can ignore another vendor's string without checking its length. A vendor is permitted to add additional structure to the `<vendorSpecificData>` field such that features are identified by their position in this field. A vendor may also elect to add explicit feature identification within the `<vendorSpecificData>` field. When present, these supplement the <vendorSpecificDataTag>.

Note that the vendor is not precluded from using the `<vendorSpecificData>` field to communicate parameters that are not related to modem relay.

E.2.2.3 Optional media format parameters

The optional attributes associated with the V.150.1 modem relay format can be declared as parameters specific to the 'v150mr' payload format using the 'fmtp' attribute. This is along the same lines as clause E.1.5.

1) **Name of Optional Attribute**: versn

   Definition: Complete representation of the ITU-T V.150.x family of Recommendations (as defined in clause 7/V.150.0). This complete representation is of the form x.y, where the first integer 'x' is the number trailing the dot in the Recommendation number e.g., '1' in ITU-T Rec. V.150.1, '2' in ITU-T Rec. V.150.2 etc. The second integer 'y' refers to the version of this Recommendation. Thus, the complete version of ITU-T Rec. V.150.1 version 2 is 1.2. Declaration of a version number, x.y, shall imply backward compatibility with earlier versions represented by smaller values of 'y'. The complete representation of the version of this Recommendation is 1.1.

Value: Dotted representation, x.y.

Default: Fixed or configured.

2) **Name of Optional Attribute**: txalgs

Definition: Supported optional Trans-Compression algorithms

Value: Comma-separated integer values and hyphenated integer ranges e.g., "1-2" or "1,2". The call-originating end may declare multiple values as alternative Trans-Compression schemes. If a value of 1 (V.44 Trans-Compression) is included in this list, then the following attributes must be provided in the session description: v44NumTxCodewords, v44NumRxCodewords, v44MaxTxStringLength, v44MaxRxStringLength, V44LenTxHistory and V44LenRxHistory. These are defined below.

Default: None of the values of the txalgs attribute. Since the support of ITU-T Rec. V.42 *bis* is mandatory for this Recommendation, it is not declared at call establishment. Hence, ITU-T Rec. V.42 *bis* is not included in Table E.3.

Table E.3/V.150.1 – Integer representation of trans-compression algorithms

Trans-compression algorithm	Integer representation
V44	1
MNP5	2

3) **Name of Optional Attribute**: V42bNumCodewords

Definition: Proposed number of codewords.

Value: 512-65535

Default: 1024.

4) **Name of Optional Attribute**: v42bMaxStringLength

Definition: Maximum V.42 string size.

Value: 6-250

Default: 32.

5) **Name of Optional Attribute**: v44NumTxCodewords

Definition: Proposed number of codewords in the transmitter.

Value: 256-65535

Default: 1024.

6) **Name of Optional Attribute**: v44NumRxCodewords

Definition: Proposed number of codewords in the receiver.

Value: 256-65535

Default: 1024.

7) **Name of Optional Attribute**: v44MaxTxStringLength

Definition: Maximum string length in the transmitter.

Value: 32-255

Default: 64.

8) **Name of Optional Attribute**: v44MaxRxStringLength

Definition: Maximum string length in the receiver.

Value: 32-255

Default: 64.

9) **Name of Optional Attribute**: V44LenTxHistory

Definition: Proposed size of the transmitter history.

Value: 512-65535

Default: 3072.

10) **Name of Optional Attribute**: V44LenRxHistory

Definition: Proposed size of the receiver history.

Value: 512-65535

Default: 3072.

11) **Name of Optional Attribute**: TCXpreference

Definition: When two double Trans-Compression (D-TCX) gateways are connected, this parameter indicates preference for the initial Trans-Compression mode.

Values: 1 ('single'), 2 ('double').

Default: 1.

The following example (fourth line onwards) illustrates the declaration of some of the optional, V.150.1 media format parameters described above:

```
m=audio 49232 udpsprt 98
a=sprtmap:98 v150mr/8000
a=fmtp:98 mr=1; mg=1;CDSCselect=3;mrmods=1-4,10-12,14,17;jmdelay=yes
a=fmtp:98 versn=1.1; txalgs=2
```

Note that is also possible to combine, as shown below, the mandatory and optional V.150.1 media format parameters on the same 'fmtp' attribute line:

```
m=audio 49232 udpsprt 98
a=sprtmap:98 v150mr/8000
a=fmtp:98 mr=1;mg=1;CDCselect=3;mrmods=1-4,10-
12,14,17;jmdelay=yes;versn=1.1;txalgs=2
```

E.2.3 Version negotiation

Endpoints shall signal the version that it supports in the **versn** attribute in its offer. The recipient of the offer shall accept that version or modify the version attribute to be equal or a lower version when transmitting an answer to the initial offer. The recipient of an offer shall not respond with an answer containing a higher version than that which was offered.

E.3 Examples of complete SDP descriptors

The examples in this clause show the minimum number of lines needed to construct an SDP-compliant session descriptor that includes all attributes that are mandatory (see E.1) for the representation of SPRT modem relay.

```
v=0
o=- 25678 753849 IN IP4 128.96.41.1
s=
c=IN IP4 128.96.41.1
t=0 0
m=audio 49230 RTP/AVP 0 2 8 18 97 98
a=gpmd:0 vbd=yes
a=gpmd:8 vbd=yes
a=rtpmap:97 telephone-event/8000
a=fmtp:97 0-15,32,33,34,35,66,70
a=rtpmap:98 v150fw/8000
m=audio 49232 udpsprt 100
a=sprtmap:100 v150mr/8000
a=fmtp:100 mr=0; mg=1; CDSCselect=3;mrmods=1,2;jmdelay=no;versn=1.1
```

In this example, ports 49230 and 49232 are used for the RTP/AVP and SPRT media streams respectively. Within the RTP/AVP media stream, the static payload types of 0 (PCMU) and 8 (PCMA) are marked for VBD treatment via the 'gpmd' attribute.

The telephone-event format is dynamically mapped into the payload type of 97. The 'fmtp' attribute is used to declare support of individual RFC 2833 events. In compliance with the requirements of this Recommendation, this includes events 32-35 (ANS etc.).

Support for the SSEs defined in Annex C is indicated by associating the token 'v150fw' with the dynamic payload type 98. By default, the scope of the SSE with payload type 98 extends to all media ports (49230 and 49232) declared in this session description. Since the list of supported events that are not vendor-specific is not explicitly listed via an 'fmtp' attribute, this is defaulted to the set of mandatory SSE events (Table C.1).

SPRT-based, modem relay media are associated with a port number of 49232. The payload type associated with the format 'v150mr' is 100. The 'fmtp' attribute is used to indicate that Version 1.1 of this Recommendation is being used, that the type of modem relay is 'V.8' and not 'universal' and that single Trans-Compression is being used.

Another variant of this descriptor, shown below, depicts 49230 as being shared between an RTP/AVP and an SPRT media stream. In this example, the type of modem relay is U-MR (mr=1) rather than V-MR (mr=0), and the modulations supported for modem relay are different.

```
v=0
o=- 25678 753849 IN IP4 128.96.41.1
s=
c=IN IP4 128.96.41.1
t=0 0
m=audio 49230 RTP/AVP 0 2 8 18 97 98
a=gpmd:0 vbd=yes
a=gpmd:8 vbd=yes
a=rtpmap:97 telephone-event/8000
a=fmtp:97 0-15,32,33,34,35,66,70
a=rtpmap:98 v150fw/8000
a=sqn:0
a=cdsc:1 audio udpsprt 100
a=cpar:a=sprtmap:100 v150mr/8000
a=cpar:a=fmtp:100 mr=1; mg=1;CDSCselect=3;
mrmods=1-4,10-12,14,17;jmdelay=no;versn=1.1
```

Annex F

Definition of capabilities for use within H.245-based systems

F.1 Scope

This annex defines the capabilities that need to be exchanged between H.245-based systems for the transmission of modem signals over packet-based networks.

F.2 Introduction

ITU-T Rec. H.245 defines a "Generic Capability" mechanism for adding new capabilities to H.245-based signalling systems. This mechanism allows for the addition of new capabilities without requiring the introduction of new ASN.1 to the base H.245 specification. Newer capabilities added to ITU-T Rec. H.245 are typically defined as generic capabilities and this annex serves to hold those capability definitions for modem signalling over packet-based networks.

F.3 Modem over IP (MoIP) capability identification and exchange

Like fax signalling, modem signalling over packet-based networks is considered a data application. As such, the MoIP capability defined in this annex shall be signalled as a **DataApplicationCapability** within H.245. Table F.1 defines the coding of the V.150.1 generic capability for H.245.

Table F.1/V.150.1 – Capability identifier for V.150.1

Capability name	V150MoIP
Capability class	Data application capability
Capability identifier type	Standard
Capability identifier value	{itu-t (0) recommendation (0) v (22) 150 moip (0) major-version-one(1) minor-version-one(1)}
maxBitRate	The maxBitRate field shall not be included and shall be ignored if received
collapsing	This field shall not be included and shall be ignored if received
nonCollapsing	This field shall not be included and shall be ignored if received
nonCollapsingRaw	This field shall be present and shall contain a value encoded using the ALIGNED variant of BASIC-PER for the ASN.1 type defined in clause F.4
transport	This field shall not be included and shall be ignored if received

The version number may increase in subsequent publications of this Recommendation. Refer to clause 1.1 for more information.

H.245-based systems may advertise more than one mode of MoIP operation by advertising multiple capabilities within the **TerminalCapabilitySet** message. For example, if a gateway has the wherewithal to act as a v8 gateway with one set of modulation capabilities and a universal gateway with a different set of modulation capabilities, the gateway may advertise each of those capabilities separately, resulting in the advertisement of two capabilities.

F.4 MoIP capability definition syntax

```
V150MOIP-CAPABILITY DEFINITIONS AUTOMATIC TAGS ::= BEGIN

IMPORTS
    NonStandardParameter FROM MULTIMEDIA-SYSTEM-CONTROL;

V150MoIPCapability ::= SEQUENCE
{
    nonStandard   SEQUENCE OF NonStandardParameter OPTIONAL,
    modemRelayType CHOICE
    {
      v-mr        NULL,
      u-mr        NULL,
      ...
    },
    gatewayType CHOICE
    {
      ntcx        NULL,          -- No Transcompression
      stcx        NULL,          -- Single Transcompression
      dtcx CHOICE                -- Double Transcompression
      {
         single   NULL,          -- Preferred mode between two gateways
         double   NULL,          -- with double transcompression ability
         ...
      },
      ...
    },
    callDiscriminationMode CHOICE
    {
      audio       NULL,
      g2-choice   NULL,
      combination NULL,
      ...
    },
    sprtParameters SEQUENCE
    {
      maxPayloadSizeChannel0    INTEGER(140..256) OPTIONAL,      -- Default 140
      maxPayloadSizeChannel1    INTEGER(132..256) OPTIONAL,      -- Default 132
      maxWindowSizeChannel1     INTEGER(32..96) OPTIONAL,        -- Default 32
      maxPayloadSizeChannel2    INTEGER(132..256) OPTIONAL,      -- Default 132
      maxWindowSizeChannel2     INTEGER(8..32) OPTIONAL,         -- Default 8,
      maxPayloadSizeChannel3    INTEGER(140..256) OPTIONAL,      -- Default 140
      ...
    } OPTIONAL,
    modulationSupport SEQUENCE
    {
      v34FullDuplex         NULL OPTIONAL,
      v34HalfDuplex         NULL OPTIONAL,
      v32bis-v32            NULL OPTIONAL,
      v22bis-v22            NULL OPTIONAL,
      v17                   NULL OPTIONAL,
      v29HalfDuplex         NULL OPTIONAL,
      v27ter                NULL OPTIONAL,
      v26ter                NULL OPTIONAL,
      v26bis                NULL OPTIONAL,
      v23FullDuplex         NULL OPTIONAL,
      v23HalfDuplex         NULL OPTIONAL,
      v21                   NULL OPTIONAL,
      v90Analog             NULL OPTIONAL,
```

```
      v90Digital            NULL OPTIONAL,
      v92Analog             NULL OPTIONAL,
      v92Digital            NULL OPTIONAL,
      v91                   NULL OPTIONAL,
      ...
    },
    compressionMode SEQUENCE
    {
      -- Including a SEQUENCE for a particular compression mode, but not
      -- including any of the optional parameters within the SEQUENCE,
      -- indicates support for the specific compression mode, but assumes that
      -- all parameter values are set to their default values
      mnp5                  NULL OPTIONAL,
      v44 SEQUENCE
      {
         numTxCodewords     INTEGER(256..65535),
         numRxCodewords     INTEGER(256..65535),
         maxTxStringLength  INTEGER(32..255),
         maxRxStringLength  INTEGER(32..255),
         lenTxHistory       INTEGER(512..65535),
         lenRxHistory       INTEGER(512..65535),
         ...
      } OPTIONAL,
      v42bis SEQUENCE
      {
         numCodewords       INTEGER(512..65535) OPTIONAL,
         maxStringLength    INTEGER(6..250) OPTIONAL,
         ...
      } OPTIONAL,
      ...
    } OPTIONAL,
    delayedJMEnabled   BOOLEAN,
    ...
}

END -- End of ASN.1 definition
```

F.5 Explanation of V150MoIPCapability elements

Non-standard parameter

Equipment manufacturers may use this field to signal any non-standard information that is specific to their Modem over IP implementations. The first octet of the **data** field within the non-standard parameter shall be the vendor-specific data tag, as specified in clause 8/V.150.0. If explicitly providing a vendor-specific tag is not necessary for a particular non-standard data parameter, the value of the first octet shall be 0 (zero). This data tag may be used in other related messages, such as SSE messages.

Modem relay type

Specifies the type of MoIP-capable gateway.

Gateway type

This field specifies the type Trans-Compression supported by the gateway. The choices are no Trans-Compression (ntcx), single Trans-Compression (stcx), or double Trans-Compression (dtcx). When selecting double Trans-Compression, the gateway must select the preferred mode (mr2 or mr3).

Call discrimination mode

These parameters specify the preferred mode for discriminating the call as defined in 15.2.11.

SPRT parameters

This field allows the gateway to advertise the maximum payload and window size values as defined by SPRT. These parameters are optional. Default values are defined in Table B.2.

Modulation support

This parameter allows the gateway to specify the modulations that are supported. Refer to 15.2.4.

Compression

This parameter allows the gateway to advertise the type of compression supported.

Delayed-JM

Indicates that the gateway supports and would like to use the Delayed-JM procedures. Both gateways must indicate "true" in this field for delayed-JM procedures to be used.

F.6 SSE capability identification and exchange

The SSE capability defined in this annex shall be signalled as a **DataApplicationCapability** within H.245. Tables F.2, F.3 and F.4 define the coding of the SSE capability within H.245.

Table F.2/V.150.1 – Capability identifier for SSEs

Capability name	V150SSE
Capability class	Data application capability
Capability identifier type	Standard
Capability identifier value	{itu-t (0) recommendation (0) v (22) 150 sse (1)}
MaxBitRate	The maxBitRate field shall not be included and shall be ignored if received
Collapsing	This field shall not be included and shall be ignored if received
NonCollapsing	This field shall contain the parameters as defined below
NonCollapsingRaw	This field shall not be included and shall be ignored if received
Transport	This field shall not be included and shall be ignored if received

Table F.3/V.150.1 – Signals and events parameter

Parameter name	SignalsAndEvents
Parameter description	This is a nonCollapsing GenericParameter **signalsAndEvents** indicates the SSE values that are supported
Parameter identifier value	Standard: 0
Parameter status	Required for capability exchange, but may be absent from logical channel signalling if capability has been previously exchanged
Parameter type	octetString. A comma-separated ASCII string of supported events identical in format to the "<list of supported events>" defined in E.1.3
Supersedes	Shall not be present and shall be ignored if received

Table F.4/V.150.1 – SSE Explicit acknowledgement

Parameter name	Explicit Ack
Parameter description	This is a nonCollapsing GenericParameter **signalsAndEvents** indicates the SSE values that are supported
Parameter identifier value	Standard: 1
Parameter status	Required for capability exchange, but may be absent from logical channel signalling if capability has been previously exchanged
Parameter type	Logical. Presence of this parameter indicates that the endpoint supports sending acknowledgements. Endpoints may set the "Must Respond Bit" only if both endpoints include this parameter in their respective open logical channel messages
Supersedes	Shall not be present and shall be ignored if received

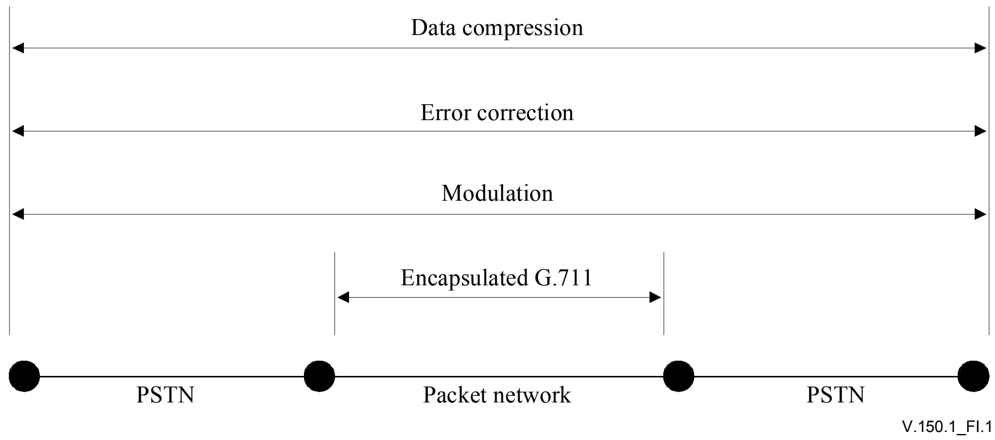
Appendix I

Connection scenarios

This appendix provides examples of the connection scenarios that are considered in the normative section of this Recommendation. This material is for reference only and is not intended to be part of the Recommendation.

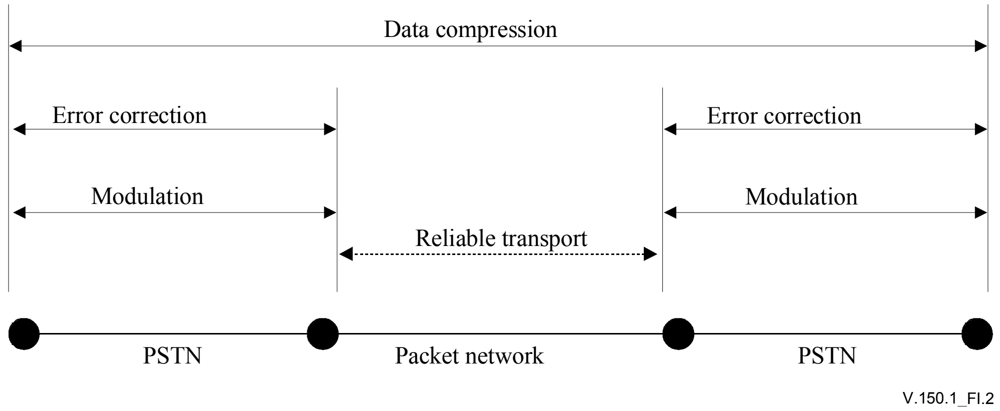
The examples indicate the termination points for the physical layer, Error Correction and Data Compression layers.

I.1 Voice Band data (VBD) mode

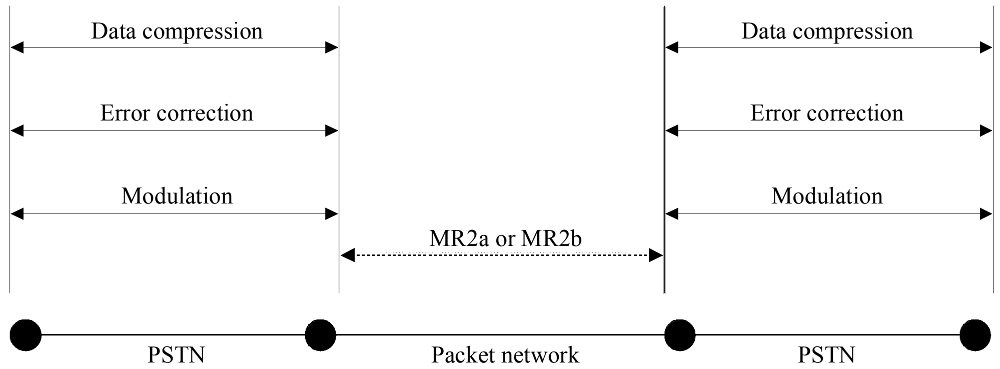


NOTE – Although support of G.711 is the minimum requirement, other means of coding if appropriate to the application are permitted.

I.2 Modem relay connection scenario MR1



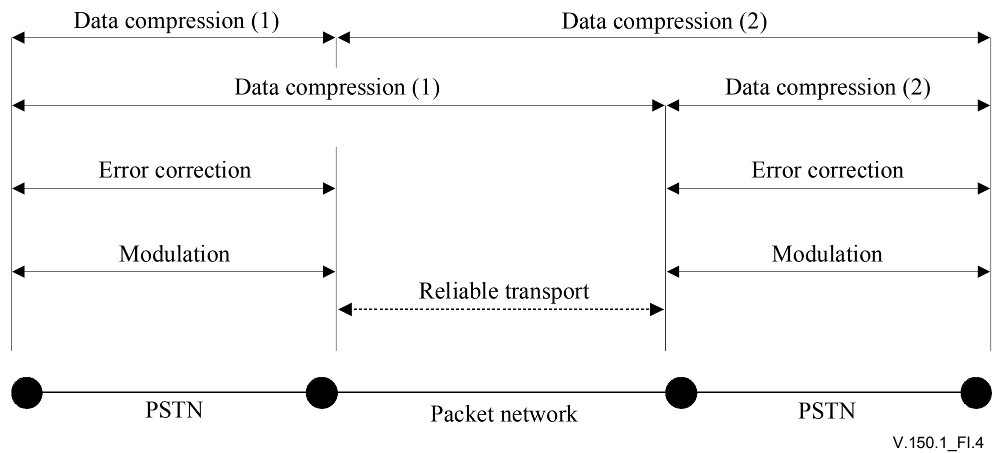
I.3 Modem relay connection scenario MR2



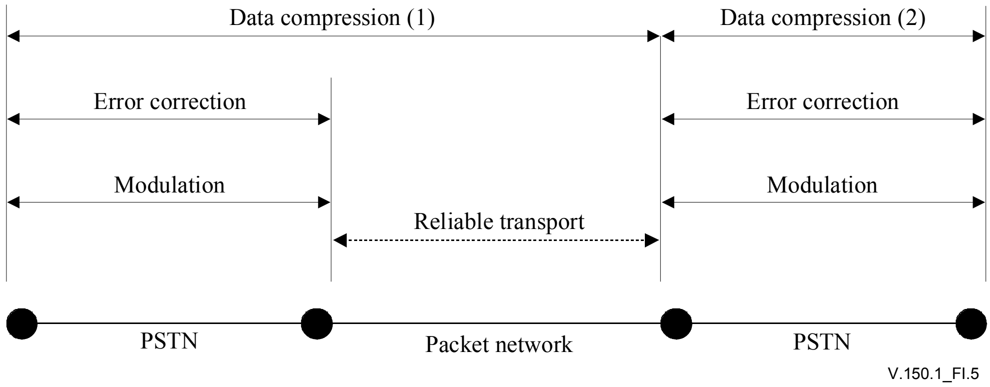
MR2a Reliable transport without data compression.
MR2b Reliable transport with data compression.

V.150.1_FI.3

I.4 Modem relay connection scenario MR3



I.5 Modem relay connection scenario MR4



Appendix II

Call discrimination call flows

II.1 Scope

This appendix contains a set of example call flow diagrams. It does not represent a complete set. If there is any conflict between these diagrams and the SDL contained in the main body of the Recommendation, the SDL will govern.

The following diagrams illustrate MoIP call flows. In the diagrams:

- The white vertical rectangles under the MoIP endpoints (G1 and G2) give the state of the respective endpoint.
- The shaded vertical rectangles under the modems (M1 and M2) and the MoIP endpoints (G1 and G2) give modem signals that are being transmitted by the respective modem or endpoint.
- That while in audio and VBD mode, the MoIP endpoints are continually transmitting and receiving audio CODEC packets as defined elsewhere in this Recommendation. For clarity, these packets are only explicitly shown when special circumstances surround them.

II.2 Answer tone treatment

II.2.1 Answer tone treatment using VBD

Figure II.1 illustrates VBD mode answer tone treatment. For this and Figure II.2 AT indicates Answer Tone and may be V.25 ANS or V.8 ANSam.

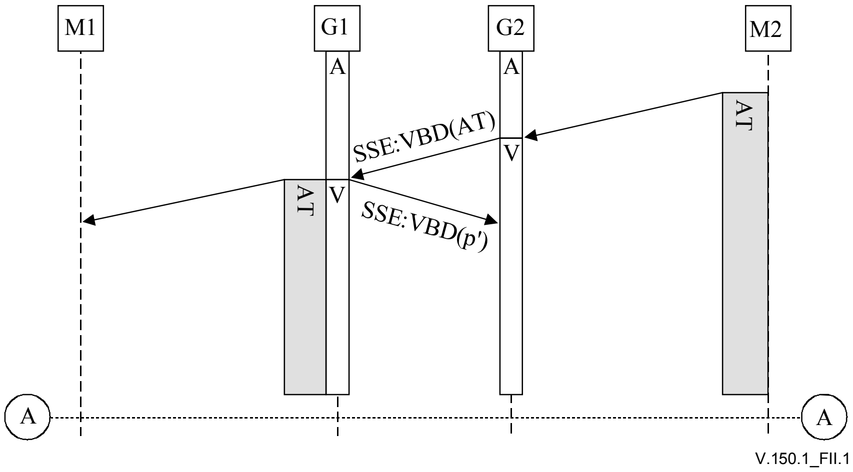
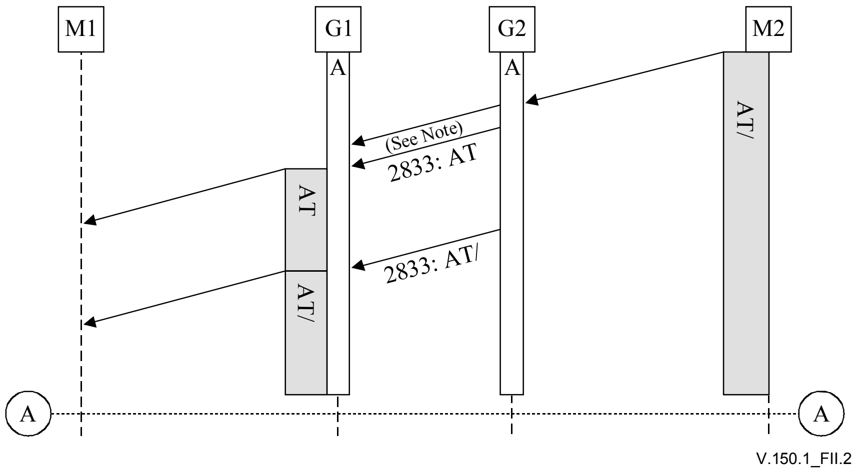


Figure II.1/V.150.1 – VBD mode answer tone treatment

II.2.2 Answer tone treatment using RFC 2833

Figure II.2 illustrates audio mode answer tone treatment.



NOTE – The 2100 Hz tone is suppressed in audio CODEC packets sent at this time.

Figure II.2/V.150.1 – Audio mode answer tone treatment

II.3 Call discrimination

Figures II.3 to II.5 illustrate MoIP call discrimination. They are continuations of the answer tone call flow diagrams in II.2. These diagrams assume the answer tone treatment has been RFC 2833 mode, as illustrated in Figure II.2. If the answer tone treatment has been VBD mode, as illustrated in Figure II.1, then transitions to VBD mode in the following diagrams are redundant and should be omitted in practice.

II.3.1 V.8 procedures

The following diagrams illustrate cases of a V.8 compliant modem, M1, calling a V.8 compliant modem, M2.

II.3.1.1 G1 transitions to VBD mode

Figure II.3 illustrates the call sequence when G1 transitions to VBD mode after receipt of CM from M1. This sequence is mandatory in the disjoint case, and is optional by G1 in the intersect and subset cases.

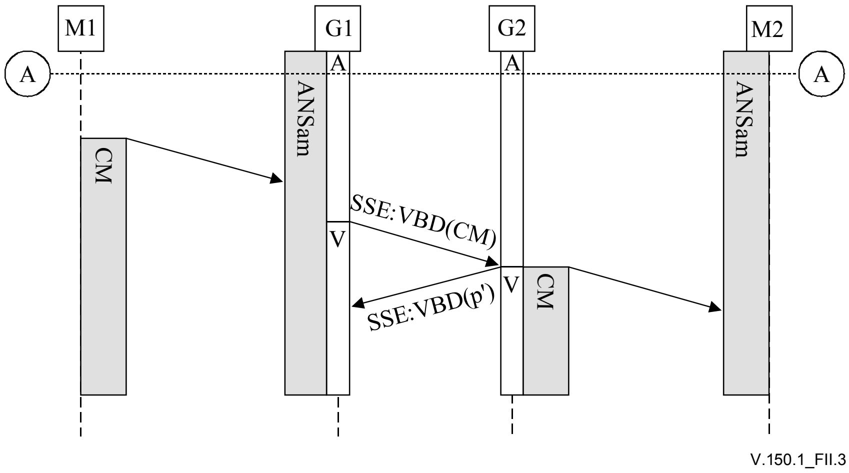


Figure II.3/V.150.1 – G1 transitions to VBD mode

II.3.1.2 G1 mandates modem relay mode

Figure II.4 illustrates the call sequence when G1 mandates modem relay mode after receipt of CM from M1. This sequence describes a subset case.

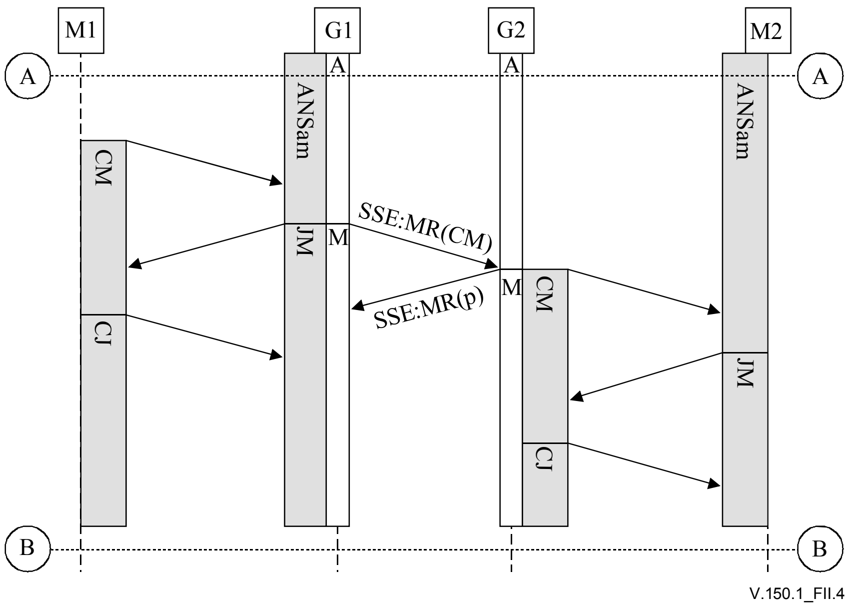
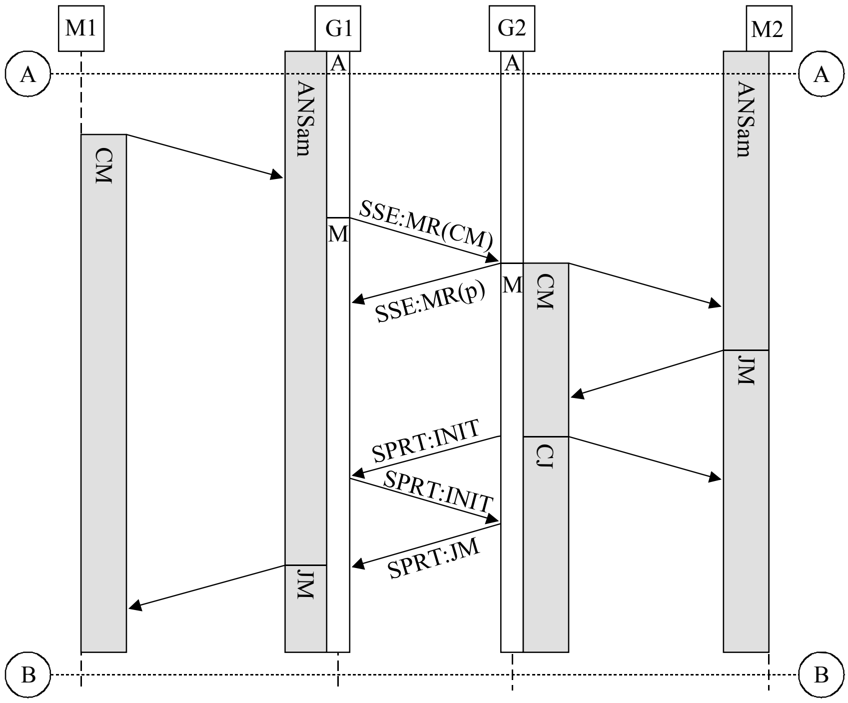


Figure II.4/V.150.1 – G1 mandates modem relay mode

II.3.1.3 G2 proceeds with modem relay mode in the intersect case

Figure II.5 illustrates the call sequence when G2 proceeds with modem relay mode after receipt of CM from M1 and after receipt of JM from M2. This sequence may be used by G2 in the intersect case if the JM signal accepts a modulation that is supported by G2 in modem relay mode. The transmission of the IP-TLP INIT message need not wait for the reception of JM from M2 nor an INIT from G2 as shown in Figure II.5.



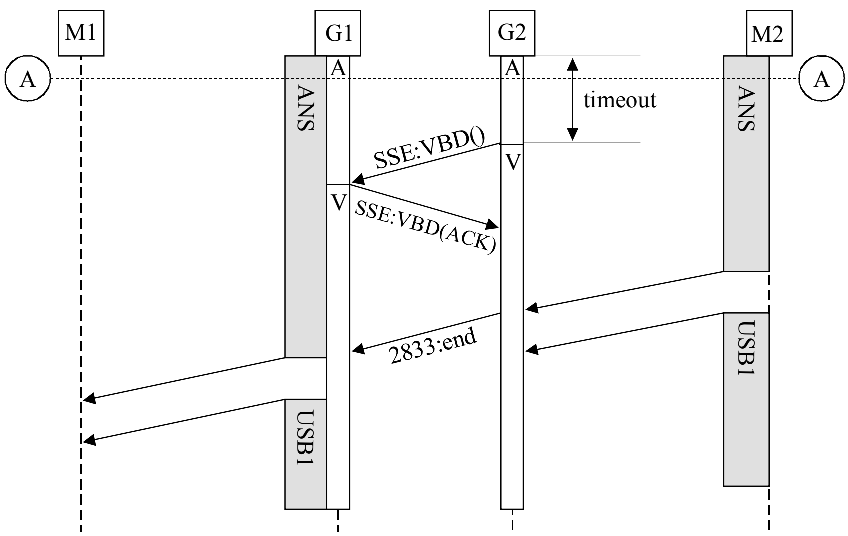
V.150.1_FII.5

Figure II.5/V.150.1 – G2 proceeds with modem relay mode

II.3.2 Non-V.8 procedures

II.3.2.1 G2 has requested audio mode answer tone treatment; M1 does not respond to answer tone

Figure II.6 illustrates the call sequence when G2 has requested audio mode answer tone treatment, G1 has requested VBD-select Answer Tone treatment, M1 does not respond to the answer tone, and M2 then continues with an automode sequence for a modulation that is not supported in modem relay mode by one of the gateways (in this case, USB1 for V.22).



V.150.1_FII.6

Figure II.6/V.150.1 – G2 has requested audio mode answer tone treatment; transition to VBD mode

Appendix III

Call discrimination call flows suitable for use with facsimile-over-IP

The call flow diagrams included in this appendix describe procedures that may be suitable for use with facsimile over IP applications. They are not official procedures, but are only for information only.

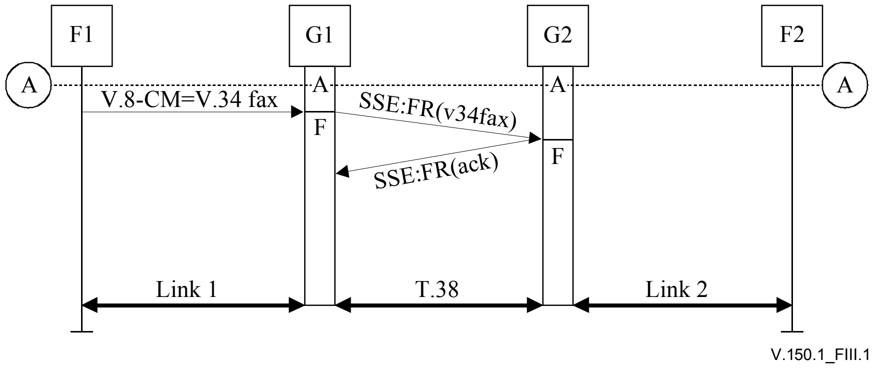


Figure III.1/V.150.1 – T.38 FoIP (V.34 fax)

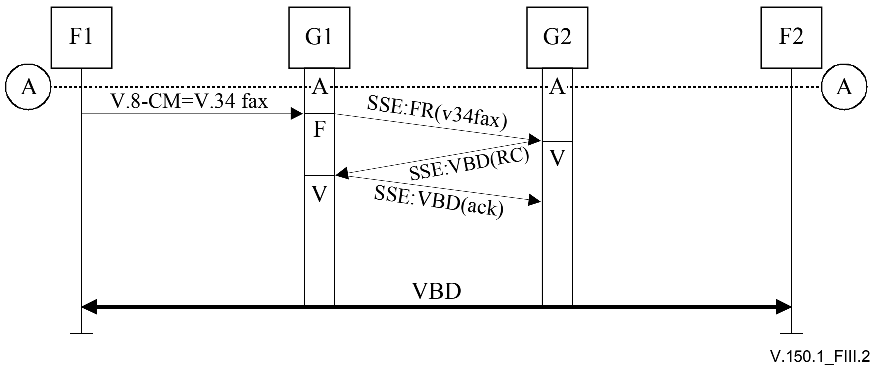


Figure III.2/V.150.1 – VBD (V.34 fax)

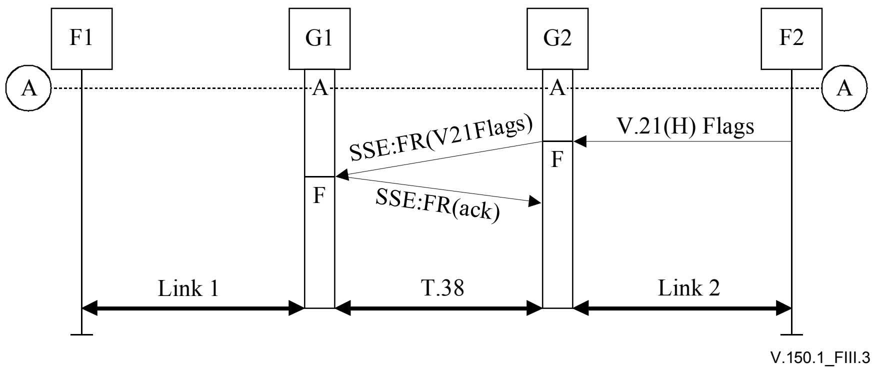


Figure III.3/V.150.1 – T.38 FoIP (T.30 fax)

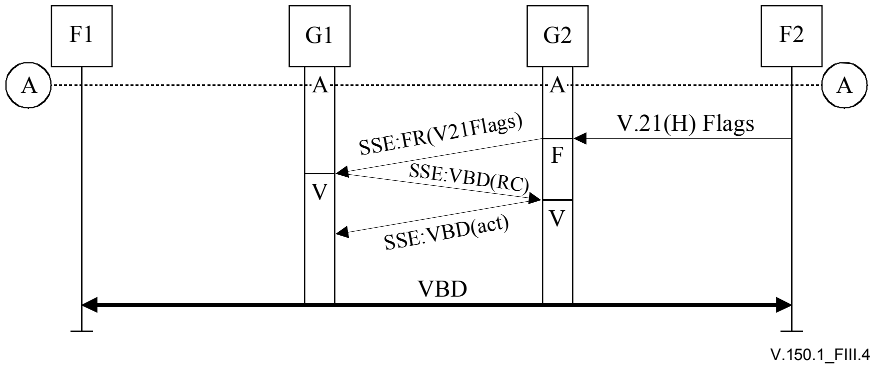


Figure III.4/V.150.1 – VBD (T.30 fax)

Appendix IV

Call discrimination call flows suitable for use for text-over-IP

The call flow diagrams included in this appendix describe procedures that may be suitable for use with text over IP applications. They are not official procedures, but are for information only.

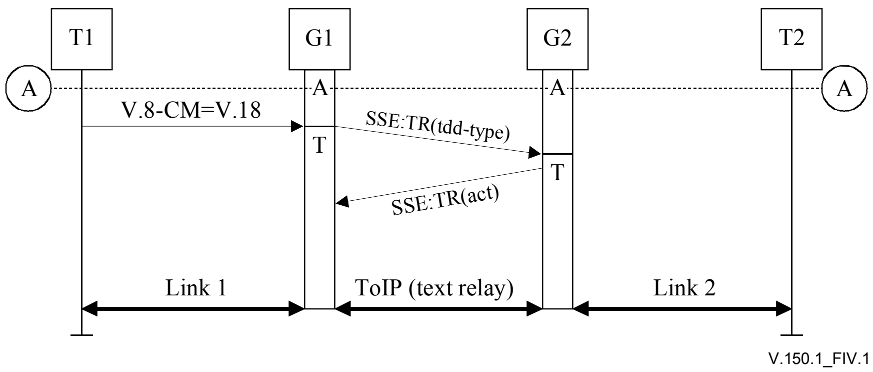


Figure IV.1/V.150.1 – V.18 to V.18 into text relay

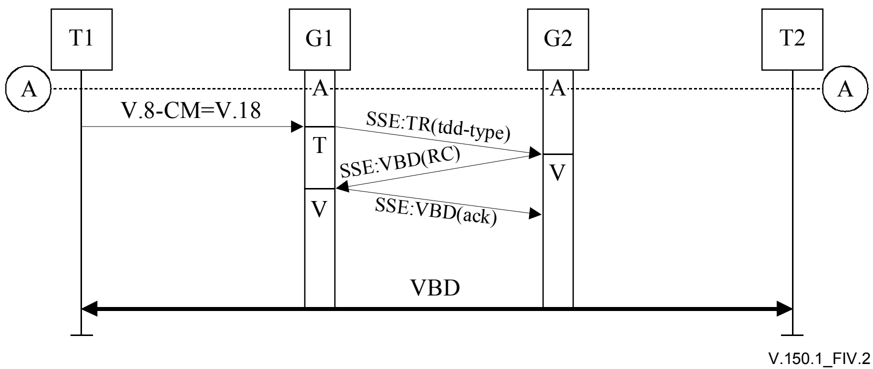


Figure IV.2/V.150.1 – V.18 toV.18 into VBD

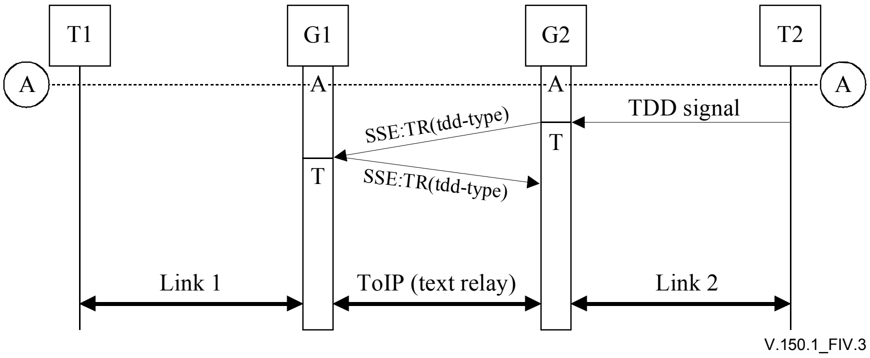


Figure IV.3/V.150.1 – Non-V.18 to non-V.18 (T2 first) into text relay

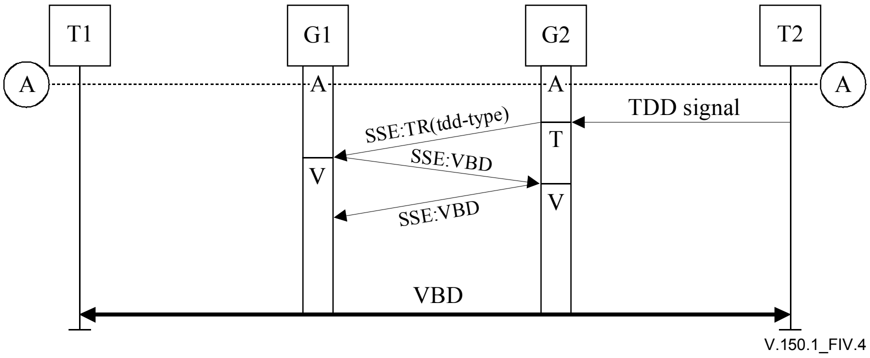


Figure IV.4/V.150.1 – Non-V.18 to non-V.18 (T2 first) into VBD

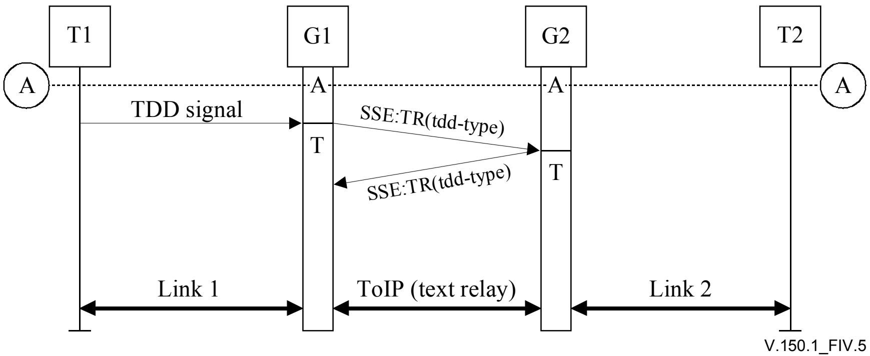


Figure IV.5/V.150.1 – Non-V.18 to non-V.18 (T1 first) into text relay

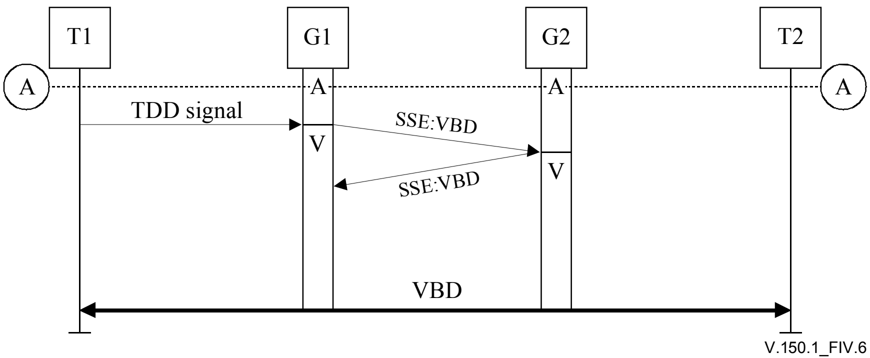


Figure IV.6/V.150.1 – Non-V.18 to non-V.18 (T1 first) into VBD

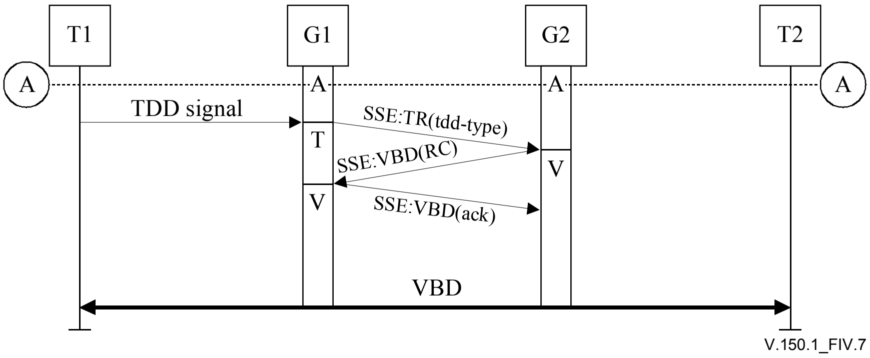


Figure IV.7/V.150.1 – TDD fallback with incompatible gateway types

Appendix V

Summary of DCE signals used during call discrimination

This appendix contains informational characterizations of DCE signals that may be considered during the call discrimination process.

V.1 Definition of answer DCE generated signals to be considered for discrimination

The following clauses describe the various signals that are generated by answering data and facsimile modems.

V.1.1 Answer tone (ANS, $\overline{\text{ANS}}$, CED, ANSam, $\overline{\text{ANSam}}$, ANSpcm, ANSpn)

The answering DCE requires the transmission of an Answer Tone that is compliant to ITU-T Recs V.25, V.8, V.92 and T.30.

V.1.2 ANS or CED

ANS (and CED) is a 2100 Hz tone transmitted by the answering DCE at its nominal power level. The frequency tolerance is specified by ITU-T Rec. G.164 as ±15 Hz. If the signal ANS is being used in its V.25 mode then maximum duration is 3.3 ± 0.7 seconds. For facsimile, ITU-T Rec. T.30 specifies that CED shall be transmitted for not less than 2.6 seconds and not more than 4.0 seconds. CED does not include phase reversals.

V.1.3 ANSam

ANSam is the Answer Tone generated by a V.8 capable answering DCE. This modified answering tone has a frequency tolerance of only ±1 Hz, and is also Amplitude Modulated by a sine wave at 15 ± 0.1 Hz. The depth of modulation is 0.2 ± 0.01 of the average amplitude. The duration of ANSam is application specific. For facsimile it is governed by ITU-T Rec. T.30 and is specified the same as for CED. For Data modems it is 5 ± 1 second.

V.1.4 $\overline{\text{ANS}}$ or $\overline{\text{ANSam}}$

These signals represent a phase reversal to the ANS or ANSam signals. These phase reversals are 180 ± 10 degrees in 1 ms occurring at a period of 450 ± 25 ms. Note, no phase reversal is specified for CED by ITU-T Rec. T.30.

V.1.5 ANSpcm

ANSpcm is a special form of ANS and $\overline{\text{ANS}}$. This signal is specially adapted for use with V.92 DCEs. As such, it complies with V.25 in its physical attributes.

V.1.6 ANSpn or ANSam-pn

ANSpn is a proprietary form of Answer tone. It complies in most aspects with ANS and ANSam, except it has a modulated signal added to it at a low power level so as not to interfere with the detection of ANS and ANSam. Unless the DCE has the capability to use this signal, it should interpret it as either ANS or ANSam accordingly.

V.1.7 Unscrambled binary-one (USB1)

USB1 is used both as an Answer Tone and as an indicator of an Answering V.22, V.22 *bis* or Bell-type DCE. For a Bell-type DCE the frequency is 2225 Hz. For a V.22 or V.22 *bis* DCE, the tone generated is that of a 2250 Hz signal. Due to the similarity of these two signals, some DCEs use 2250 Hz for both cases or alternate between the two frequencies.

V.1.8 V.32 *bis* AC

AC is a phase alternating modulated signal used by V.32 and V.32 *bis* DCEs. The tone is generated by modulating a Carrier Frequency of 1800 ± 1 Hz with alternating points from a four-point QAM constellation at the rate of 2400 ± 0.01% symbols/sec. The result is a combination signal consisting of two tonal frequencies, 600 and 3000 Hz.

V.1.9 V.21 mark frequency

This signal is generated by a V.21 enabled DCE. It is the tone generated by a DCE when transmitting constant binary ones in the channel No. 2 as defined by ITU-T Rec. V.21. The frequency characteristic of this signal is 1650 ± 6 Hz.

V.1.10 V.23 mark frequency

There are two signals to be considered. The first Mark Frequency signal is generated by the forward channel of V.23 mode 1 and 2 enabled DCE. It is the tone generated by a DCE when transmitting constant binary ones as defined by ITU-T Rec. V.23. The frequency characteristic of this signal is 1300 ± 10 Hz. The second Mark Frequency is for the V.23 backward channel and is 390 ± 3 Hz. (This tolerance is specified in ITU-T Rec. R.35.)

V.1.11 V.8 *bis* initiating signals

V.8 *bis* can select one of seven possible message types as an initiating signal. The selection is dependent upon which of the V.8 *bis* transactions are to be used. The signals are Mre, MRd, Cre, CRd, MS, CL and ESi. Each of these signals consists of two segments. The first consists of the simultaneous transmission of a pair of tones (1375 and 2002 Hz) for 400 ms (although in certain situations the shorter time of 285 ms is permissible). The tolerance on the frequency is ±250 ppm and for the duration, it is ±2%. The second segment is of a single frequency and is the signal type indicator.

NOTE – The end-to-end transport of V.8 *bis* signals is not supported in Version 1 of this Recommendation.

V.1.12 V.21 Flags

Detection of V.21 Channel 2 HDLC encoded FLAGS is a special signal indicating facsimile operation.

V.2 Definition of calling DCE generated signals to be considered for discrimination

The following subclauses describe the various signals that are generated by Calling data and facsimile modems.

V.2.1 T.30 CNG tone

CNG is a tone defined in ITU-T Rec. T.30. It is used to indicate that a calling DCE is a facsimile terminal. This 1100 Hz Tone is transmitted with a repeated cadence of 0.5 seconds on 3 seconds off until CED (or ANSam) is detected. The tolerance on the frequency is ±38 Hz and for the timing, it is ±15%. Note that this signal alone is not enough to indicate that a call is Facsimile and the originating terminal in all cases may not transmit it.

V.2.2 1300 Hz Calling Tone (CT)

Some DCEs use a 1300 Hz tone to indicate that the calling device is a non-speech variety. The cadence of this signal is 0.5 to 0.7 s ON and 1.5 to 2.0 s OFF as specified in ITU-T Rec. V.25. Per ITU-T Rec. V.18, there may be 3 s of 400 ms and 800 ms ON periods interleaved with periods of modulated carrier. The frequency tolerance is ±10 Hz as specified in ITU-T Rec. V.23.

V.2.3 1500 Hz proprietary calling tone

Some cellular terminals use a proprietary calling tone. The frequency of this tone is 1500 ± 15 Hz. The cadence of the signal is 0.7 s ON and 1.5 s OFF. Note that this signal alone is not enough to indicate that the call is a Cellular Data call.

V.2.4 V.8 Function Indicator (CI)

A V.8 capable calling DCE may optionally transmit the V.8 Function Indicator (CI) as an alternative to CED and CT. This signal is not tonal but is a coded modulated signal using V.21 Channel 1 (low band) at 300 bit/s. CI is transmitted with a regular cadence. The On duration is not less than 3 periods of the CI signal, and not greater than 2 seconds. The Off duration is less than 0.4 seconds and not greater than 2 seconds.

V.2.5 V.8 Call Menu signal (CM)

The V.8 Call Menu signal (CM) is transmitted in response to the detection of ANSam. CM is an encoded V.21 Channel 1 modulated signal at 300 bit/s. The signal is protected from being falsely detected as an HDLC FLAG.

V.2.6 V.8 *bis* responding signals

V.8 *bis* can select one of three possible responding signals: MRd, CRd and ESr. Each of these signals consists of two segments. The first consists of the simultaneous transmission of a pair of tones (1529 and 2225 Hz) for 400 ms (although in certain situations the shorter time of 285 ms is permissible). The tolerance on the frequency is ±250 ppm and for the duration, it is ±2%. The second segment is of a single frequency (1900 Hz) and is the signal type indicator.

V.2.7 V.92 QCA1a and QCA1d

QCA1a and QCA1d are signals consisting of a sequence of bits transmitted using V.21 (H) modulation. The sequences consist of 10-bit frames using V.8-type formatting as described in clauses 8.2.3 and 8.3.4/V.92 for QCA1a and QCA1d, respectively.

V.2.8 V.92 QCA2a or QCA2d

If V.92 QCA2a or QCA2d is detected and the gateway knows the other gateway supports MoIP, the gateway shall clamp off the data being sent via VoIP to the other gateway, switch to MoIP mode, and send one or more RTP packets telling the other gateway to switch to MoIP mode as well.

V.2.9 V.32 *bis* signal AA

AA is a single point modulated signal used by V.32 and V.32 *bis* DCEs. The result is the generation of a single 1800 Hz tone, with a tolerance of 1 Hz.

V.2.10 V.22 *bis* signal S1

S1 is unscrambled repetitive double dibit 00 and 11 at 1200 bit/s for 100 ± 3 ms. Using V.22 *bis* low channel (1200 ± 0.5 Hz Carrier).

NOTE – A guard tone 1800 ± 20 or 550 ± 20 Hz may also be transmitted simultaneously with this signal.

V.2.11 V.22 Scrambled Binary-1 (SB1)

V.22 Originating DCEs transmit this modulated signal. The signal consists of scrambled binary-ones on the V.22 low channel (1200 ± 0.5 Hz Carrier) at the data rate of 1200 bit/s.

V.2.12 Bell 103-type 1270 Hz

An initial signal of 1270 ± TBD Hz is transmitted by Bell 103 type originating DCEs.

V.2.13 V.21 channel 1 mark tone

V.21 DCEs will respond by transmitting continuous binary ones. This results in a tone of 980 ± 6 Hz.

V.2.14 V.23 mark tone

If a V.23 DCE is configured in its half-duplex mode, then it will transmit a Mark tone in the same way as described above. If asymmetric duplex is to be used, the DCE will transmit a 390 ± 3 Hz as a Mark signal.

Appendix VI

Descriptions of non-V-series modes of operation

This annex provides characterization and descriptions of non-V-series-standard modes of operation that are commonly used in deployed DCEs.

VI.1 Bell 103 mode of operation

The communication circuit for data transmission is a duplex circuit whereby data transmission in both directions simultaneously is possible at 300 bits/s or less. The frequency of the Answer Tone used by this DCE is 2225 Hz.

NOTE – This mode of operation is also described in Annex D/V.18.

VI.1.1 Modulation

The modulation is a binary modulation obtained by frequency shift, resulting in a modulation rate being equal to the data signalling rate.

For channel No. 1, the nominal mean frequency is 1170 Hz; for channel No. 2, it is 2125 Hz.

The frequency deviation is ±100 Hz. In each channel, the higher characteristic frequency (FA) corresponds to a binary 1 (i.e., channel No. 1 (FA = 1270 Hz and Fz = 1070 Hz); channel No. 2 (FA = 2225 Hz and Fz = 2025 Hz)).

VI.2 Bell 212A mode of operation

The communication circuit for data transmission is a duplex circuit provided by frequency-division multiplexing. Two modes of operation are provided, one is Bell 103 mode and the second is a 1200 bit/s duplex data channel. The frequency of the Answer Tone used by this DCE is 2225 Hz.

VI.2.1 Modulation

The modulation is four-phase Phase Shift Keying with a symbol rate of one-half of the data signalling rate (600 symbols/s).

For channel No. 1, the nominal carrier frequency is 1200 Hz, for channel No. 2 it is 2400 Hz.

For each two bits of binary data input, a dibit symbol is mapped to a phase differential from the previous transmitted symbol.

Both synchronous and asynchronous data format are supported.

VI.3 TIA/EIA-825 mode of operation

This Recommendation specifies a Frequency Shift Keyed (FSK) modem operating at a data signalling rate of 50 and 45.45 symbols/sec. The deaf and hard of hearing generally use this modem for real time 2-way text-based communication over the public switched telephone network.

NOTE – This mode of operation is also described in Annex A/V.18.

VI.3.1 Modulation

The transmission mode is character oriented Frequency Shift Keying (FSK) using two tones to represent the asynchronous serial data. A binary ONE is represented by 1400 Hz ±1% tone and a binary ZERO is represented by 1800 Hz ±1% tone.

VI.4 MNP5 mode of operation

MNP5 is a data compression procedure that uses adaptive frequency encoding and run length encoding to achieve up to a 2:1 compression ratio. Adaptive frequency encoding operates on each 8-bit character. Codewords, that are 4 to 10 bits long, are dynamically assigned to characters, with the most frequently occurring characters represented by the shorter codewords. However, when the same character is encountered in a sequence of length greater than 3, the protocol switches to run length encoding. In this mode, the characters in the sequence are counted and the sequence is represented as a repetition count.

Appendix VII

Gateway implementation guide

VII.1 Scope

The scope of this appendix is to provide additional information that may be useful to the implementer of a MoIP gateway. The material in this appendix is non-normative and provides only possible methods and considerations. Implementations compliant to this Recommendation are not required to use the procedures described in this appendix.

VII.2 VBD

Any delay introduced by a VBD compatible speech codec should be kept to a minimum (see Annex A/G.114).

VII.3 Rate control for configurations using SPRT channel 1

If the reliable SPRT channel 1 is used to transmit data onto the IP network, then flow control can be achieved using the implicit SPRT flow control mechanism as in the LAPM case (i.e., using base sequence numbers).

VII.3.1 Rate control procedures for symmetrical case

For the Symmetric case, the following Simple Rate Control (SRC) procedure is suggested:

1) Gateways inform each other of their initial physical layer speed (rx and tx) on their respective telephony legs (rx and tx are defined with respect to the view of the gateway modem, i.e., rx refers to direction of flow towards gateway, and tx refers to direction of flow towards client modem). If the "rx" speed on a gateway is larger than the "tx" speed on peer gateway, the gateway reduces its "rx" speed through rate negotiation so that it would be equal or lower than the "tx" speed on peer gateway. This assumes asymmetrical rates can be used at the physical layer.

   In case that neither gateway supports asymmetrical rates, the objective of this step can be achieved by setting all "rx" and "tx" rates the same across the two gateways (i.e., rx gw1 = rx gw 2 = tx gw 1 = tx gw 2).

   In case that one gateway supports asymmetrical physical layer rates (rx, tx), but the other gateway supports symmetrical physical layer rate (R), the following relationship should be enforced through rate negotiation:

   $$rx \leq R \leq tx.$$

2) Step 1 above establishes appropriate physical layer rates initially. Subsequently, in case of any further rate renegotiations, the gateways may need to perform rate adjustments again according to relationships described in Step 1. This requires each gateway to let its peer know of any rate changes on its telephony interface.

3) Depending on the mode (Raw or Character mode), the gateway packetizes the input and forwards it to its peer. The packet may be sent to the peer after a certain amount of bits or characters have accumulated, or after a certain amount of time has elapsed.

VII.3.2 Rate control procedures for hybrid case

For the Hybrid case, after switching over to modem relay state, the following procedure is applied:

1) Gateways inform each other of their initial physical layer speed (rx and tx) on their respective telephony legs (rx and tx are with respect to the view of the gateway modem). The rate procedures in this case cannot simply rely on physical layer rates, and instead must rely on "Effective Data Rate" (i.e., rate of actual information/data bits transferred). This is the case since the overhead on V.42 link is substantially lower than V.14 link, since V.14 link uses 2 extra bits, 1 start and 1 stop bits per character. This is true when there is limited, or no, impairments on telephony links. However, in case of significant telephony impairments, there would be error recovery by V.42 protocol which may substantially reduce the throughput of the V.42 side, and create the opposite situation.

In general, the following Effective Rate Control (ERC) procedure should allow satisfactory operation:

2) The physical layer "receive" rate on gateway running V.14 on its DCE interface should be set such that the received "Effective Data Rate" for the gateway is equal to or lower than the V.42 side transmit "Effective Data Rate". In circumstances where the V.42 leg may be expected to experience significant impairment, it would make sense to introduce an "Effective Data Rate Differential" by introducing a larger speed differential (i.e., by setting a lower maximum received "Effective Data Rate" for the gateway running V.14 on its DCE) than would have been considered otherwise.

3) In the reverse direction (i.e., data flow from the gateway running V.42 towards the gateway running V.14), no physical layer rate constraints need to be applied. This is the case, since the "Effective Data Rate" can be controlled by the gateway, since V.42 link offers flow control (e.g., gateway can send its client V.42 RNR to flow control data when needed). However, the suggested procedure for the gateway with the V.42 link would be to execute a

"Leaky Bucket" algorithm to decide when to generate RNR. The single Leaky Bucket is set up with a "leak rate" equal to the Effective Data Rate on the "tx" direction on the peer gateway. The bucket is filled at the Effective Data Rate on the "rx" direction of the gateway running V.42 link. The size of the Leaky Bucket must be smaller than the "data/information" storage capacity of the dejitter buffer used in the system between the two gateways. When the amount of data in the bucket exceeds a certain preset threshold, the gateway would be required to generate V.42 RNR to shutdown its client modem. Subsequently, when the amount of data remaining in the bucket goes lower than an appropriate threshold, the gateway would indicate V.42 RR to its client to resume the data flow. The leaky bucket procedure in this clause is referred to as Leaky Rate Control (LRC) procedure.

4) Steps 1 through 3 above establish appropriate physical/data rates initially. Subsequently, in case of any further rate renegotiations, the gateways may need to perform rate adjustments or generate RR and RNR according to relationships described above in Steps 1 through 3. This requires each gateway to let its peer know of any rate changes on its telephony interface. During the data transfer phase, the gateway packetizes the input Start/Stop or 8 bit characters and sends it to its peer. The packet is sent to the peer after certain amount of characters are accumulated, or after a certain amount of time has elapsed.

5) In case of telephony impairments on the V.42 link, the throughput of the link may be degraded such that it cannot cope with the incoming data flow from the peer gateway. Because of telephony impairments, the V.42 state machine may go into timer recovery state. It is permissible to lose some data in this case, but in order for the V.42 gateway not to fall significantly behind, it would be required to drop all incoming characters received from the remote peer during such periods.

VII.3.3 Retrain/speed shift

During retrain or speed shifts, one link may become temporarily unavailable which could likely result in loss of some data. If the link that has gone into retrain/speed shift is the V.42 link, then during retrain/speed shift all data arriving from peer gateway should be discarded.

VII.4 XID/LR profiles

A particular modem's (M1's or M2's) profile is:

- That modem's internal configuration and capabilities, with respect to protocol and compression.
- The prediction data kept by that modem's local gateway.
- The prediction data received in a PROF_XCHG message.

VII.4.1 Predicting the XIDc/LRc M1 will send

Note that M1's profile converts simply to the XIDc or LRc that M1 will send:

– For LAPM (build the XIDc that M1 would send):
  - If LAPM is not supported in M1's profile, then no XID is sent.
  - If V.42 *bis* is supported in M1's profile, insert a V.42 *bis* group into the XID and copy the profile's P0 (directions), P1 (codewords) and P2 (string) values into the appropriate fields. Additionally,
  - If V.44 is supported in M1's profile, insert a V.44 group into the XID and copy the profile's C0 (capability), P0 (directions), P1T (Tx dictionary size), P1R (Rx dictionary size), P2T (Tx string size), P2R (Rx dictionary size), P3T (Tx history size), and P3R (Rx history size) into the appropriate fields.

– For Annex A/V.42 (1996) (build the LRc that M1 would send):

- If Annex A/V.42 (1996) is not supported in M1's profile, then no LR is sent.
- If MNP5 is supported in M1's profile, insert a parameter 9 (compression) into the LR, with the MNP5 bit (LS bit) set. Additionally,
- If V.42*bis* is supported in M1's profile, insert a parameter 14 into the LR, and copy the profile's P0 (directions), P1 (codewords) and P2 (string) values into the appropriate subfields.

VII.4.2 Predicting the XIDr/LRr M2 will reply for a particular XIDc/LRc

Note that M2's profile is not simply copied to the XIDr/LRr that M2 will send, since XIDr and LRr are the result of negotiation. When M2 is sent to a particular XIDc or LRc, the predicted XIDr or LRr is calculated from M2's profile as follows:

– For LAPM (build the XIDr that M2 would send, on receiving a particular XIDc):
  - If LAPM is not supported in M2's profile, then no XID is sent.
  - If V.44 is supported in M2's profile and if the V.44 group exists in the XIDc sent to M2:
    - – Negotiate C0 (capability): XIDr's C0 = profile's C0.inband & XIDc's C0.inband.
    - – Negotiate P0 (directions): XIDr's P0.Rx = profile's P0.Rx & XIDc's P0.Tx; XIDr's P0.Tx = profile's P0.Tx & XIDc's P0.Rx. If P0 is missing from the XIDc, then use the default value (both Tx and Rx).
    - – Negotiate P1T (Tx codewords): XIDr's P1T = min(profile's P1T, XIDc's P1R). If P1R is missing from the XIDc, use the default P1R value.
    - – Similar negotiation for P1R, P2T, P2R, P3T, P3R. Note that the defaults for P3T and P3R depend on the value of P1T and P1R.
    - – Insert a V.44 group into the XIDr, and copy the negotiated C0, P0, P1T, P1R, P2T, P2R, P3T and P3R into the appropriate fields.
  - Otherwise, If V.42 *bis* is supported in M2's profile and if the V.42 *bis* group exists in the XIDc sent to M2:
    - – Negotiate P0 using V.42 *bis* semantics. The bits are "originator to responder" and "responder to originator" versus V.44's Rx and Tx.
    - – Negotiate P1 (codewords): XIDr's P1 = min(profile's P1, XIDc's P1), using the default if P1 does not appear in the XIDc.
    - – Negotiate P2 (string size) similarly.
    - – Insert a V.42 *bis* group into the XIDr, and copy the negotiated P0, P1 and P2 into the appropriate fields.
  - Otherwise, neither compression group will appear in M2's XIDr.

– For Annex A/V.42 (1996) (build the LRr that M2 would send, on receiving a particular LRc):
  - If Annex A/V.42 (1996) is not supported in M2's profile, then no LR is sent.
  - If V.42 *bis* is supported in M2's profile and if V.42 *bis* is requested in either parameter 9 or parameter 14 of the LRc sent to M2:
    - – Negotiate P0 using V.42 *bis* semantics, as above; there are no defaults.
    - – Negotiate P1 (codewords): XIDr's P1 = min(profile's P1, XIDc's P1), as above; there are no defaults.
    - – Negotiate P2 (string size) similarly.
    - – Insert parameter 14 (V.42 *bis* compression) into the LR, and copy the negotiated P0, P1 and P2 into the appropriate fields.

- Otherwise, if MNP5 is supported in M2's profile and if the MNP5 bit is set in parameter 9 of the LRc sent to M2:
  - Insert parameter 9 (compression) into the LR, and set its MNP5 bit.
- Otherwise, neither compression parameter will appear in M2's LRr.

If M2's profile specifies no-protocol operation, then G2's PROF_XCHG is sent with octets 1-2 (Protocol and Compression Support) set to zero, indicating no support of protocols or compressions. G1 should send an XIDr/LRr to M1 indicating no compression.

VII.4.3 Operation when both gateways know their modems' profiles

If a gateway receives a PROF_XCHG message and also has knowledge of its own local modem (it has sent its own PROF_XCHG), the two gateways are capable of completely predicting XID or LR exchanges in a distributed fashion.

Both gateways will compute the XID/LR value described in VII.4.2 "Predicting the XIDr/LRr M2 will reply for a particular XIDc/LRc" above, using the profile data sent by G1 and G2. This computed value is the same for both gateways, since each knows what it sent and received. Note that this value corresponds to the compression portion of a complete XID or LR.

This compression value is sent from G1 to M1 in an XIDr or LRr, in reply to M1's XIDc/LRc. It is also sent from G2 to M2 in an XIDc or LRc, giving M2 one compression choice, at most. (If the prediction is to be unique and correct, M2 should not be given any decision leeway.) Note that the following relations will hold if the gateways' profiles for M1 and M2 were valid:

$$\text{XIDr/LRr[G1]} \subseteq \text{XIDc/LRc[M1]}$$

$$\text{XIDr/LRr[G1]} = \text{XIDc/LRc[G2]} = \text{XIDr/LRr[M2]}$$

VII.4.4 Operation when only G1 knows its modem's profile

Even if G2 does not "know" a profile to send via PROF_XCHG, G1 can still send its own PROF_XCHG if G2 is willing. If the G2 leg trains up earlier than the G1 leg, G2 can then send M2 an XIDc/LRc immediately, without having to wait for the M1-G1 trainup. This frame will be formed from G1's PROF_XCHG by using the calculation method in VII.4.1 "Predicting the XIDc/LRc M1 will send" above. For end-to-end negotiation to complete, M2's XIDr/LRr information must be sent to G1 via XCHG_XID to be relayed to M1 when it trains up.

If G2 is not willing to receive the PROF_XCHG message, G1 may send the calculated XIDc/LRc in an XCHG_XID to achieve similar results.

VII.4.5 Operation when only G2 knows its modem's profile

Even if G1 does not "know" a profile to send via PROF_XCHG, G2 can still send its own PROF_XCHG, if G1 is willing. If the G1 leg trains up earlier than the G2 leg, G1 can then send M1 an XIDr/LRr immediately on receiving M1's XIDc/LRc, without having to wait for the M2-G2 trainup. This frame will be formed from G2's PROF_XCHG and M1's XIDc/LRc by using the calculation method in VII.4.2 "Predicting the XIDr/LRr M2 will reply for a particular XIDc/LRc" above. For end-to-end negotiation to complete, M1's XIDc/LRc information must be sent to G2 via XCHG_XID, to be relayed to M2 when it trains up.

If G1 is not willing to receive the PROF_XCHG message, G2 may send the calculated XIDr/LRr in an XCHG_XID message after receiving G1's XCHG_XID, even if the M2-G2 trainup is not yet complete.

VII.4.6 Operation when neither gateway knows its modem's profile

In this case, neither GW will send a PROF_XCHG message. Both gateways will default to "default XID" techniques or a full, synchronized end-to-end exchange.

Appendix VIII

Bibliography

The following is a bibliography of informative references used in this Recommendation.

– ITU-T Recommendation G.114 (2003), *One-way transmission time*.

– IETF RFC 791 (1981), *Internet protocol*.

– IETF RFC 919 (1984), *Broadcasting Internet datagrams*.

– IETF RFC 920 (1984), *Domain requirements*.

– IETF RFC 950 (1985), *Internet standard subnetting procedure*.

SERIES OF ITU-T RECOMMENDATIONS

Series A	Organization of the work of ITU-T
Series B	Means of expression: definitions, symbols, classification
Series C	General telecommunication statistics
Series D	General tariff principles
Series E	Overall network operation, telephone service, service operation and human factors
Series F	Non-telephone telecommunication services
Series G	Transmission systems and media, digital systems and networks
Series H	Audiovisual and multimedia systems
Series I	Integrated services digital network
Series J	Cable networks and transmission of television, sound programme and other multimedia signals
Series K	Protection against interference
Series L	Construction, installation and protection of cables and other elements of outside plant
Series M	TMN and network maintenance: international transmission systems, telephone circuits, telegraphy, facsimile and leased circuits
Series N	Maintenance: international sound programme and television transmission circuits
Series O	Specifications of measuring equipment
Series P	Telephone transmission quality, telephone installations, local line networks
Series Q	Switching and signalling
Series R	Telegraph transmission
Series S	Telegraph services terminal equipment
Series T	Terminals for telematic services
Series U	Telegraph switching
Series V	**Data communication over the telephone network**
Series X	Data networks and open system communications
Series Y	Global information infrastructure and Internet protocol aspects
Series Z	Languages and general software aspects for telecommunication systems

Printed in Switzerland
Geneva, 2004