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SERIES V: DATA COMMUNICATION OVER THE TELEPHONE NETWORK

Interworking with other networks

Procedures for the end-to-end connection of analogue PSTN text telephones over an IP network utilizing text relay

ITU-T Recommendation V.151

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ITU-T Recommendation V.151

Procedures for the end-to-end connection of analogue PSTN text telephones over an IP network utilizing text relay

Summary

This Recommendation defines the inter-operation of two PSTN to IP Network gateways that facilitate the end-to-end connection of analogue text telephone terminals over an IP network utilizing text relay.

Amendment 1 introduces Annex E, which defines the RTP payload format for the transport of text as T.140 character data over IP networks, removing the need to make a reference to an historic RFC published by the IETF. It also includes minor editorial changes throughout the document to make references to the newly introduced Annex, removing references to the RFC.

Source

ITU-T Recommendation V.151 was approved on 29 May 2006 by ITU-T Study Group 16 (2005-2008) under the ITU-T Recommendation A.8 procedure.

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Keywords

PSTN textphone, text over IP, text relay, text telephony, ToIP.

FOREWORD

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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.

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

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Procedures for the end-to-end connection of analogue PSTN text telephones over an IP network utilizing text relay

1 Scope

This Recommendation defines the interoperation of two PSTN to IP Network gateways that facilitate the end-to-end connection of analogue text telephone terminals over an IP network.

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 F.700] ITU-T Recommendation F.700 (2000), *Framework Recommendation for multimedia services*.

[ITU-T F.703] ITU-T Recommendation F.703 (2000), *Multimedia conversational services*.

[ITU-T G.177] ITU-T Recommendation G.177 (1999), *Transmission planning for voiceband services over hybrid Internet/PSTN connections*.

[ITU-T G.711] ITU-T Recommendation G.711 (1998), *Pulse code modulation (PCM) of voice frequencies*.

[ITU-T H.245] ITU-T Recommendation H.245 (2006), *Control protocol for multimedia communication*.

[ITU-T H.248.1] ITU-T Recommendation H.248.1 (2005), *Gateway control protocol: Version 3*.

[ITU-T H.323 Ann.P] ITU-T Recommendation H.323 (2003), *Packet-based multimedia communications systems, Annex P: Transfer of modem signals over H.323*.

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

[ITU-T V.8 *bis*] 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 general switched telephone network and on leased point-to-point telephone-type circuits*.

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

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

[ITU-T V.150.0] ITU-T Recommendation V.150.0 (2003), *Modem-over-IP networks: Foundation*.

[ITU-T V.150.1]	ITU-T Recommendation V.150.1 (2003), *Modem-over-IP networks: Procedures for the end-to-end connection of V-series DCEs*.
[ITU-T V.152]	ITU-T Recommendation V.152 (2005), *Procedures for supporting voice-band data over IP networks*.
[IETF RFC 2198]	IETF RFC 2198 (1997), *RTP Payload for Redundant Audio Data*.
[IETF RFC 2327]	IETF RFC 2327 (1998), *SDP: Session Description Protocol*.
[IETF RFC 2733]	IETF RFC 2733 (1999), *An RTP Payload Format for Generic Forward Error Correction*.
[IETF RFC 2833]	IETF RFC 2833 (2000), *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*.
[IETF RFC 3407]	IETF RFC 3407 (2002), *Session Description Protocol (SDP) Simple Capability Declaration*.

3 Definitions

This Recommendation defines the following terms:

3.1 Baudot: A five-bit code formerly used in Teletype machines including those text telephones that operate per TIA-825-A standard. The use of the term "Baudot" in this Recommendation is expanded to also include the modulation used by text telephones complying with TIA-825-A.

3.2 constant carrier PTP: A PSTN Textphone (PTP) modulation that uses a carrier signal whose transmission is maintained in-between text spurts. Examples of constant carrier PTP modulations schemes include PTPs that utilize V.21 and V.23 modulations. Constant carrier modulations are also referred to as full duplex (FDX) modulations in this Recommendation.

3.3 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).

3.4 IP text device: Native IP device that supports text communications and provides text communications interoperability with V.151-compliant gateways. Examples include IP-based IVR systems, voicemail systems, and IP phones.

3.5 non-constant carrier PTP: A PTP modulation that uses a carrier that is only transmitted during a text spurt (and possibly for a period before the text spurt as a preamble). An example of non-constant carrier PTP modulations schemes is that used by EDT-based PTP devices. Non-constant carrier modulations are also referred to as half-duplex (HDX) modulations in this Recommendation.

3.6 PSTN textphone: Term used in this Recommendation to represent all classes of text telephones, including TDD devices.

3.7 protocol conversion: The interworking of dissimilar PTP terminals performed by the ToIP gateways.

3.8 telecommunications terminal for the deaf: A term used in the USA for equipment that provides text telephony. It is synonymous to text telephone.

3.9 text over IP: The dependable transport of analogue PSTN text telephones signals over IP networks using the methods and procedures as defined in this Recommendation.

3.10 text spurt: A sequence of one or more characters transmitted in a text telephony signal without loss of carrier.

3.11 text telephone mode: The operational mode when two devices are interconnected to provide text telephone communications.

3.12 text telephone: A device incorporating text telephony functions.

3.13 text telephony: A telecommunications capability that supports real-time text conversation on communication networks.

3.14 voice-band data: The transport of modem signals over a voice channel of a packet network with the encoding appropriate for modem signals.

3.15 Conventions

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

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

4 Abbreviations

This Recommendation uses the following abbreviations:

ANS	V.25 Answer Tone
ASNam	V.8 Answer Tone
ASN.1	Abstract Syntax Notation One
CI	V.8 Call Indicator
DS0	Digital Signal level 0
DCE	Data Circuit-terminating Equipment (Modem)
DTE	Data Terminal Equipment
FEC	Forward Error Correction
FDX	Full Duplex
FoIP	Fax over Internet Protocol
G1	On-Ramp Gateway
G2	Off-Ramp Gateway
HDX	Half Duplex
I1	Originating end-point IP Text telephone
I2	Answering end-point IP Text telephone
IP	Internet Protocol
IP-TLP	IP Transport Layer Protocol
ITD	IP Text Device
IVR	Interactive Voice Response
MoIP	Modem over Internet Protocol
PCM	Pulse Code Modulation
PHY	Physical transport layer of modem connection
PSTN	Public Switched Telephone Network

PTP	PSTN Textphone
QoS	Quality of Service
RIC	Reason Identifier Code
RTP	Real Time Protocol
SPRT	Simple Packet Relay Transport (Annex B of [ITU-T V.150.1])
SSE	State Signalling Events
SSRC	Synchronization Source
T1	Originating end-point analogue PSTN Text Telephone
T2	Answering end-point analogue PSTN Text Telephone
TDD	Telecommunications Devices for the Deaf
TDM	Time division multiplex(ing)
ToIP	Text Telephony over IP
UDP	User Datagram Protocol
VBD	Voice Band Data Mode
VoIP	Voice over Internet Protocol
XCI	Signal used in V.18

5 Introduction

5.1 Overview of Text over IP

Text Telephony over IP (ToIP) is the dependable transport of analogue modulated signals generated by PSTN text telephone over an IP Network. The types of such text telephones and the characteristics of their line signals are described in [ITU-T V.18]. There are three basic models that can be considered for support of this application. These models are dependent upon the transmission characteristics and Quality of Service of the IP network in which the text telephone conversation is being established. The models can be summarized as:

a) Support of text telephony over VoIP connections (i.e., utilizing audio mode transport). This model is outside the scope of this Recommendation.

b) Support of text telephony utilizing Voice Band Data (VBD) transport as defined in [ITU-T V.152].

c) Support of text telephony utilizing Text Relay transport with VBD fallback according to this Recommendation.

This Recommendation includes support for PSTN Textphone (PTP) devices that utilize modulations covered in [ITU-T V.18], including its Annexes A through G. Gateways compliant to this Recommendation are required to support full re-modulation/demodulation of one or more of these PTP modulations in order to transport the data using text relay. The remaining modulations shall be supported using VBD transport. The text relay procedures described in this Recommendation cover the entire set of PTP modulations and can be scaled down for specific implementations that support a subset of the modulations described in [ITU-T V.18].

5.2 ToIP system architecture

Architecturally, this Recommendation primarily considers the support of a PSTN to IP to PSTN structure otherwise called a hybrid Internet/PSTN connection as defined in [ITU-T G.177]. Figure 5-1 illustrates a typical Text Telephony over IP application.

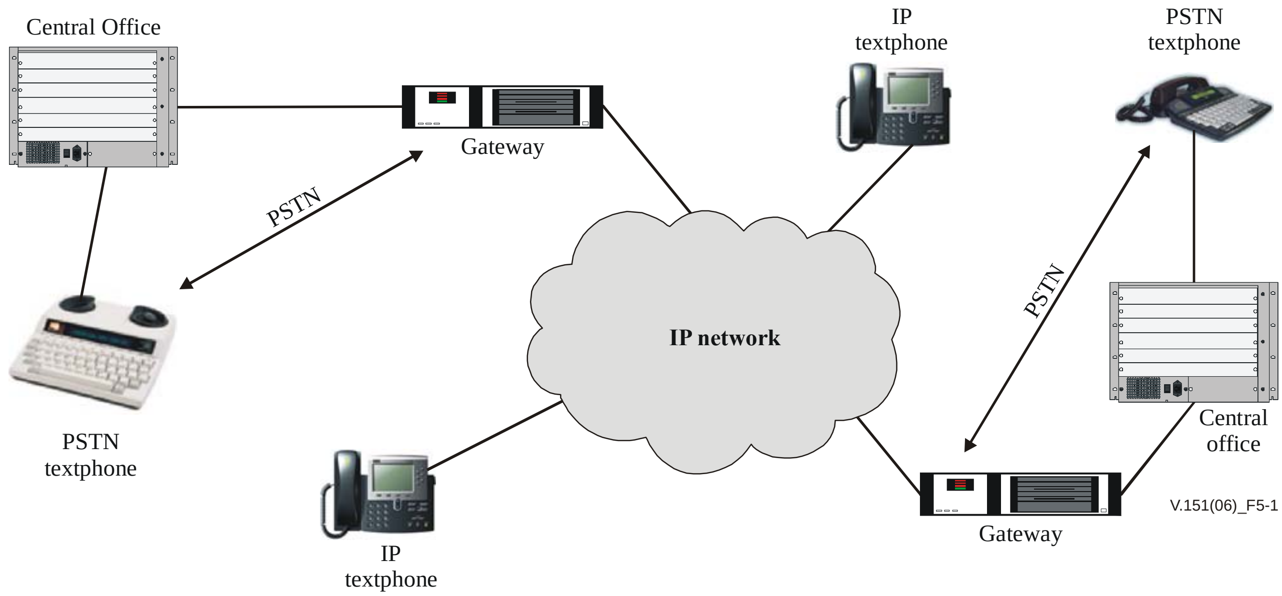


Figure 5-1 – Typical ToIP applications

The possible connection combinations are:

a) PSTN Text Telephone to PSTN Text Telephone;

b) PSTN Text Telephone to IP Text Telephone.

This Recommendation deals with PSTN Text Telephone to PSTN Text Telephone operation. PSTN Text Telephone to IP Text Devices that support the protocol defined in Annex E is described in Annex D.

5.3 Compliance requirements

This Recommendation does not require behaviour that is inconsistent with other Recommendations, or with national regulatory requirements, and shall be interpreted accordingly. Neither does it preclude the use of proprietary or non-standard text telephones; however, it does caution that if such devices are used, then care should be taken so 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.

5.4 Relationship with other V.15x Recommendations

The V.15x series of Recommendations (currently including [ITU-T V.150.1] and [ITU-T V.152]) define the support of data modems (using VBD or relay) and facsimile modems (using VBD) over IP networks. Modem over IP (MoIP) support is either through modem-relay mode or voiceband data mode. Since PTP endpoints utilize data modems at the physical layer, there may be interaction between the procedures defined by the various V.15x-series Recommendations for the case when the gateway has implemented and successfully negotiated the support with the remote gateway. For each V.15x Recommendation, there is a requirement that the support for it be successfully negotiated with the remote gateway during the signalling phase of the call for the procedures defined in the Recommendation to be used by the gateway.

Gateways compliant with this Recommendation may also concurrently implement data MoIP support via [ITU-T V.150.1] and/or VBD support via [ITU-T V.152].

6 Text telephone-over-IP functions

Figure 6-1 provides a conceptual reference model for a ToIP gateway. The model shows two stacks conjoined by the ToIP application. The left-hand stack is that of a typical text telephone which has a signal converter (modulation). The right-hand stack represents the IP networking functions of a ToIP Gateway. The ToIP application as indicated in Figure 6-1 is defined by the normative contents of this Recommendation.

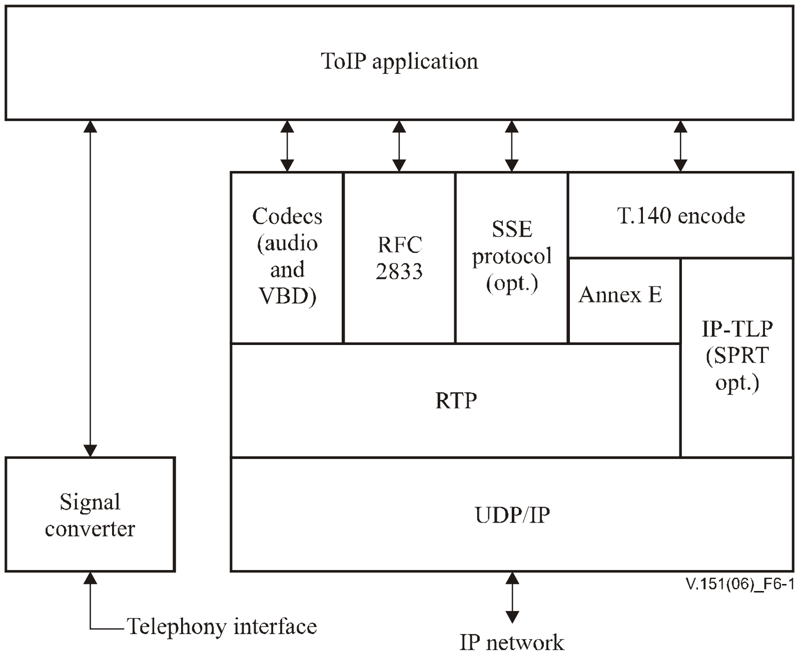


Figure 6-1 – ToIP gateway reference model

7 Connection and network scenarios

Figure 7-1 below provides a reference diagram which matches that of Figure 5-1. It adds the concepts that a Text Telephone may be connected via a PSTN telephone or an IP telephone as well through acoustic couplers.

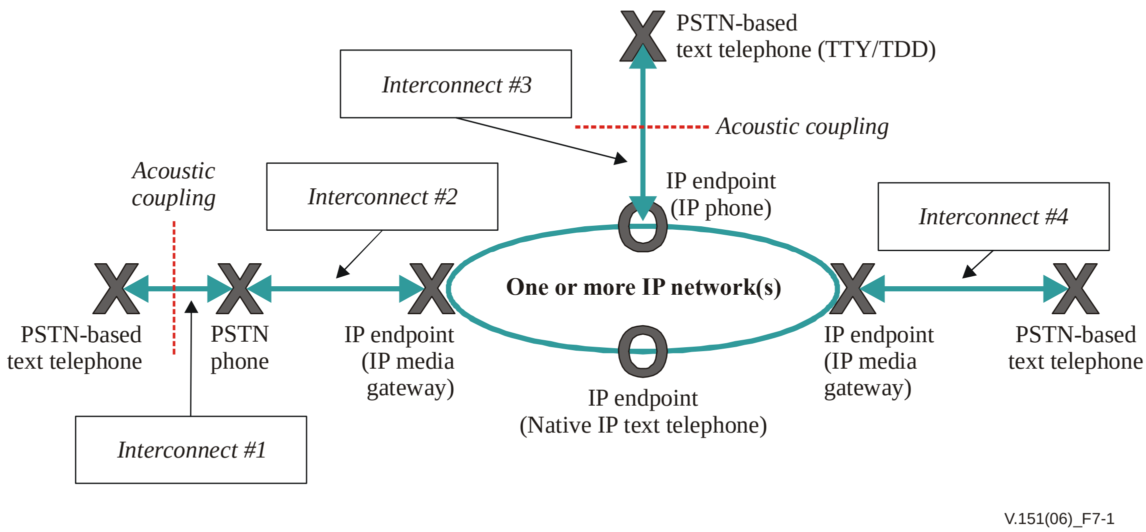


Figure 7-1 – Reference diagram for a Text-over-IP network architecture

Interconnect #1 is an acoustic coupling between a PTP device to an analogue PSTN phone. Interconnect #2 is the connection between the PSTN phone that is acoustically coupled to the PTP device. The IP gateway connected at the interconnect #2 point shall support the transmission of text signals from the end PTP device using this Recommendation.

Interconnect #3 is an acoustic coupling of a PTP device to an IP endpoint (e.g., an IP phone). The support of the transport of the PTP signals over IP in this interconnect is described in Annex D.

Interconnect #4 is the direct interconnection of a PTP device to a gateway via the PSTN or other time division multiplexing (TDM) network (e.g., an enterprise TDM network). The IP gateway connect at the interconnect #4 point shall support the transmission of text signals from the end PTP device using this Recommendation.

7.1 Service and performance requirements for ToIP

7.1.1 Applicability of [ITU-T F.703] to ToIP

[ITU-T F.703] describes the end-user performance requirements for PTP devices operating in the PSTN. Gateways compliant to this Recommendation, when inserted between two end PTP devices, should be designed to maintain the user experience as defined in [ITU-T F.703].

7.1.2 Gateway performance

In order to maintain the end-user performance to that specified in [ITU-T F.703], gateways should consider the appropriate design and use of:

- modulator/demodulators;
- jitter buffer operation and jitter buffer sizes;
- fault-tolerant schemes to compensate for IP network impairments when appropriate. Fault-tolerant schemes that should be used include [IETF RFC 2198] redundancy, [IETF RFC 2733] FEC, or SPRT;
- minimizing delay through gateway while maintaining performance related to character loss.

The performance of ToIP can be dependent upon the performance of the IP network connecting the two gateways. Gateways compliant to this Recommendation should utilize supplementary fault-tolerance schemes in order to meet the performance requirements given the network conditions that they are operating under. In environments with network QoS that does not achieve the end-to-end performance requirements, there is a need to supplement text telephone connections with fault-tolerance schemes such as [IETF RFC 2198] redundancy, [IETF RFC 2733] FEC, or SPRT.

7.1.3 Support of simultaneous voice and text

[ITU-T V.18] describes the application of [ITU-T V.61] for the simultaneous transmission of voice and text. The support of this mode of operation is for further study.

7.1.4 Interoperability of various text telephone types

PTP devices can use one of a number of modulations and character encodings as described in [ITU-T V.18]. This Recommendation makes allowances for the interconnection of PTP devices that utilize different modulations and/or character encodings through the use of text relay. As described further in clause 20, if the gateways support the modulation of their local PTP devices, the connection may result in use of Text Relay mode achieving protocol conversion scenario. If the connection results in use of VBD mode for PTP signal transport, the PTP devices must have a common capability or else the connection will not be established between the PTP devices.

7.2 Audio and text functionality

Many PTP schemes on the PSTN support alternating between voice and text during the call. For the hearing impaired, a common use scenario of PTP is for the hearing-impaired user to speak to the remote user but receive responses using text telephony. Another common use scenario of PTP for persons with a speech disability is to use text telephony to transmit while receiving voice responses. Other scenarios include interactive voice response (IVR) systems where network voice announcements may be preceded by a short burst of text, with the voice announcement followed by a longer burst of text telephony. For all PTP types, alternation between text and voice is bidirectional; i.e., there will never be text in one direction while speech is being sent in the other direction. Systems that are constant carrier-based will drop carrier during portions of the call that speech is being sent. Systems that are not based on constant carrier for text will alternate between text spurts and speech spurts during the call.

This Recommendation provides support for alternating between voice and text spurts as long as both end-point PTPs are operating in a mode that supports voice and text.

7.2.1 Interactive voice response system support

In interactive voice response (IVR) applications that support text telephony, the IVR system will generate voice and text announcements to the end user. The end user may generate DTMF signals in response to these voice and/or text announcements. Gateways that are compliant to [ITU-T V.151] shall support IVR applications which include text signalling as described in this clause.

For support of IVR, a gateway shall enable its DTMF receiver processing concurrently with the ToIP support. If a DTMF digit is received while the gateway is regenerating a text message, the gateway shall signal this DTMF digit to the remote gateway using the pre-negotiated method for DTMF signalling across the IP network (e.g., voice encoded or [IETF RFC 2833]) while continuing to play out all the remaining text of the message out the PSTN link.

Since DTMF ToIP is supported using normal DTMF signalling methods across the IP network, the gateway shall be able to support DTMF text-based IVR applications without any additional procedures.

8 Transport methods

This clause provides an overview of the transport modes with which a ToIP gateway could be operating. Gateways that are compliant to this Recommendation shall support ToIP utilizing either text relay or VBD mode.

8.1 Audio mode

In this mode, the channel processes speech signals. This mode may include the use of compression algorithms and other processing functions that are not suitable for the transport of text telephony signals. This mode of operation may be considered suitable for the purposes of transporting text telephony signals if the speech processing is benign to those signals. Depending upon the QoS of the network, the use of fault-tolerant mechanisms such as FEC and redundancy may be appropriate.

Gateways operating per this Recommendation shall not use Audio mode for the transport of text telephony signals.

8.2 VBD mode

Voice Band Data mode (VBD) of operation is defined in [ITU-T V.152]. VBD mode is suitable for the transport of text telephony signals. Depending upon the QoS of the network the use of fault-tolerant mechanisms such as FEC and redundancy may be appropriate.

Since VBD mode is also suitable for the transport of non-text signals (e.g., speech), gateways may choose to stay in VBD mode for the duration of the call after the transition to ToIP, including possibly periods of speech interspersed with the text telephony signals. Gateways negotiate the capability to leave VBD mode during call set-up procedures as defined in clause 15.1.2. If the negotiation of this operation fails, gateways shall not transition out of VBD mode when PTP signals are no longer present (see clause 8.4.4).

A V.151-compliant gateway may use VBD mode for the transport of PTP signals for the following conditions:

1) The PTP signal modulation detected is one that is not supported for text relay by both gateways.

2) Probing sequence timeouts as specified in clause 20.

Since VBD mode provides a transport of the PTP signals without the demodulation of these signals, interworking of dissimilar PTP types (i.e., protocol conversion) is not supported when using VBD mode.

8.3 Text relay mode

Text relay mode of operation is characterized by the termination of the text telephone physical layer function at the gateway and the transport of the text characters between gateways.

Text relay gateways demodulate the text telephony signals and present the user data to the gateway's ToIP application. The format of this data can be variable. This user data is then encoded as defined by [ITU-T T.140]. Once the characters are encoded, they may be relayed over the IP network using a suitable transport protocol (i.e., an IP Transport Layer Protocol, IP-TLP). The default IP-TLP for this Recommendation is defined in Annex E. Depending upon the QoS of the network the use of fault-tolerant mechanisms (redundancy and/or FEC) may be appropriate. The optional use of SPRT as defined in [ITU-T V.150.1] as the IP-TLP is also supported by this Recommendation.

8.3.1 Supported modes in text relay

FDX text telephony modes are those modes that have a constant carrier present. For V.18 modes, FDX modes supported by this Recommendation include:

a) ITU V.18 text telephony utilizing V.21 modulation.

b) Bell 103 based text telephony modems.

c) ITU V.23 Videotex terminals ("Minitel").

d) ITU V.21 encoded per ITU T.50.

HDX-based text telephony modes do not have a carrier present while not transmitting text. For V.18 modes, HDX modes supported by this Recommendation include:

- EDT (European Deaf Telephone) utilizing V.21 frequencies at 110 bit/s;
- 5-bit FSK Baudot terminals at 45.45 bit/s or at 50 bit/s.

Gateways shall support one or more of the modes specified above using text relay. The modes supported by the gateway shall be advertised in the call signalling protocols as described in their appropriate annexes. If the gateway only supports a single mode for text relay, this mode should be a mode that is commonly used in the region in which the gateway is deployed.

DTMF is used for text telephony in some PTP devices in addition to other purposes for DTMF (e.g., address signalling, control of voice mail). DTMF is not supported as a text relay method by gateways following this Recommendation. DTMF is supported by using normal DTMF transport over IP schemes as described in clause 12. DTMF support using text relay and interoperability of DTMF PTP devices with PTP devices that are currently supported using text relay is for further study.

8.4 Switching between transport modes

This clause describes the requirements for switching between the transport modes described above.

For alternating text and voice, text shall be given priority. The gateway shall use text mode as long as there is text to be transmitted and played out.

When the gateway is in connection with an HDX text telephone, voice can be conveyed between bursts of text without further consideration.

When the gateway is in connection with an FDX-based text telephone, detection of a drop in carrier from the text telephone shall cause the gateway to also drop the carrier and go to voice mode. Detection of carrier again, or text arriving from the IP side shall cause the gateway to start sending carrier again and entering text mode. Reception of a command between the gateways for changing mode to voice may cause the gateway to change mode to voice.

8.4.1 Payload type indication method

The default method for switching between transport modes is using RTP payload types to indicate a transition from one mode to another.

For SPRT, the payload type field in the SPRT header can be used for indication of a media switch to SPRT.

A gateway can initiate a switch to a new transport mode (initiator gateway) by generating packets encoded in the new transport mode and using the payload type indicating the transport mode being switched to. The initiator gateway will then discard incoming packets that arrive that do not use the new payload type encoding. These packets may be received until the remote gateway receives the new payload type encoding and switches to the new transport mode. The remote gateway (responder gateway), upon receiving packets with the new payload type, will immediately transition to the new transport mode and generate packets (if required) using the new encoding and proper payload type.

8.4.2 SSE method

State signalling event (SSE) protocol may optionally be used to control transitions between transport methods. Use of SSE protocol for transport mode transitions is determined at call set-up through the gateways indicating the capability of SSEs. If both gateways indicate the SSE capability, SSEs shall be used to control the transition of transport methods. If either gateway does not indicate SSE capability, the default method of payload type indication shall be used.

The optional SSE protocol is given in Annex C of [ITU-T V.150.1]. For the purposes of ToIP, the accepted media states used are audio, voice band data (VBD), text relay probing and text relay.

When the SSE protocol is being used, the optional Reason Identifier Code (RIC) field in the SSE payload shall be sent and filled in with the appropriate code.

8.4.3 Transitioning to ToIP transport

This clause provides the requirements at the gateway when transitioning to a non-audio transport mode that will be used to carry the text telephony signal over the IP network.

The ToIP gateway should minimize as much as possible the leakage (audio encoding) of the PTP signals into the IP network. If more than 50% of a bit of a character is leaked into the IP network, an extraneous character may be generated to the remote PTP device.

To minimize the possibility of an erroneous character due to signal leakage in audio mode, the regenerating gateway should wait one character time from the onset of the possible leakage to regenerate the character. Silence may be generated when the gateway introduces this delay.

The first character received on the PSTN interface of the gateway shall be transmitted only once to the IP network, i.e., it shall not be lost nor erroneously repeated by the gateway.

8.4.4 Transitioning from ToIP transport

Upon detection of loss of carrier or other signal event indicating that the local PTP device is no longer in the connected state, the gateways shall initiate a transition out of text relay mode and into audio mode. For payload type switching protocol, the gateways shall encode incoming signals using the RTP audio encoding. For the SSE protocol, the gateway shall initiate a switch to Audio mode by generating an SSE(AUDIO) to the remote gateway.

When a gateway detects that the remote gateway has initiated a switch out of text relay mode and into audio mode, it shall drop carrier with its local PTP device (if carrier is still being generated by the gateway) and transition to audio mode transport.

8.4.5 Modulation support scenarios for text relay

To be compliant with this Recommendation, a gateway shall support full physical layer modulation/demodulation for at least one of the modes used by PTP devices as described in clause 8.3.1. The gateway shall also support the detection of signals used for all the PTP modulations listed in clause 8.3.1. This detection capability is used by the call discrimination procedures for the initiation of transitioning to VBD mode so that VBD can be used between the PTP devices even if the modulation is not fully supported by the gateway.

The transition to text relay not only depends on the modulation being supported by the local gateway and PTP device, but also on the modulation supported by the remote gateway and the modulation used by the remote PTP device; in other words, even though the local gateway and the local PTP support the same modulation, text relay might not be the mode that is used. VBD could be the mode decided on instead. The call discrimination procedures defined in clause 20 use the knowledge of the modulations supported by both gateways and the signals observed from the PTP devices in determining whether text relay can be used. The overall goal of the call discrimination procedures is to ensure connectivity of the PTP devices with the use of text relay as the secondary goal.

In the case where the PTP devices do not support a common modulation, the call discrimination procedures may still result in the use of text relay mode if the gateways do support their local PTP modulations. This scenario is what is referred to as protocol conversion throughout this Recommendation.

Since the optional SSE protocol includes the mandatory indication of the signals that are being detected which caused the SSE to be generated, call discrimination procedures that use SSE will have additional flexibility resulting in more cases of using text relay in protocol conversion scenarios.

9 ToIP operational modes

PTP signals shall either be carried using VBD mode or Text Relay mode. The call discrimination procedures defined in clause 20 define how the gateways shall transition from audio mode to either VBD or text relay mode when PTP signals are present. In the case where text relay mode is used, there still might be a stage where VBD is used during the call discrimination phase.

The call discrimination procedures will guarantee that PTP devices will communicate if they are able to communicate over a PSTN connection, either through the preferred mode of text relay or using the fallback VBD mode. Additionally, the call discrimination procedures may result in the use of text relay mode to allow for connectivity of PTP devices that would not normally connect in a PSTN connection through the use of protocol conversion.

10 Text Relay PHY layer operation

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

The goal of ToIP is to ensure connectivity of analogue PSTN text telephony terminals on IP networks. This Recommendation does not require nor preclude the use of non-standard mechanisms. The ToIP 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 text telephony terminals. The establishment of the text telephone physical layer consists of two stages. The first stage involves call discrimination including the detection and discrimination of the text telephony signals on the PSTN interface of the gateway. The second stage involves the establishment of the physical layer connection between the gateway and the text telephony terminal. The procedures that define this process are described in clause 20.

The overall connection physical layer may be selectively transported in either VBD or text relay. The call discrimination and mode selection procedures define the rules for this choice.

Text relay supports the ability for each gateway to connect with text telephones using different modulation modes. This allows for the possible interoperability of dissimilar text telephony devices across the IP network, although there are some limitations to this mode of interoperation. 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 their supported modulation capability set. This exchange happens during call set-up using signalling mechanisms.

The call discrimination procedures include mechanisms that attempt to match the modulation used on each PSTN call leg. This is not guaranteed, though, so differing modulations and differing signalling rates may result. In the case that the signalling rates differ on each PSTN call leg in text relay mode, buffer management and flow control may be required; Appendix IV provides relevant information on text buffering and transmission.

11 IP transport for text relay

This Recommendation assumes that the IP protocol conforms to the following standards: [IETF RFC 791], [IETF RFC 950], [IETF RFC 919] and [IETF RFC 920]. This Recommendation does not impact any IP network topology, IP packet distribution and routing protocols, which are independent of this Recommendation.

The default IP Transport Layer Protocol for this Recommendation shall be the protocol defined in Annex E. Optionally, the simple packet transport protocol (SPRT) as defined in Annex B of [ITU-T V.150.1] may be used if mutually negotiated. The use of other IP transport protocols is for further study.

Since Annex E specifies transport of T.140-encoded data on RTP, it inherits the same reliability characteristics of any RTP media stream. Improved transport reliability may be achieved by utilizing [IETF RFC 2198] (Redundancy) and or [IETF RFC 2733] (Forward Error Correction). The use of these is optional but encouraged. Appendix III provides guidance for the use of [IETF RFC 2198] and [IETF RFC 2733].

If using RTP for text relay transport, audio/t140c media type shall be supported by the gateway. Use of text/t140 media type for interoperability with non-V.151 devices that also use text/t140 is for further study.

[IETF RFC 2198] (RTP redundancy) shall be implemented by gateways compliant to this Recommendation. [IETF RFC 2733] (RTP FEC) and SPRT may also be implemented by the gateway. Use of one or more of these mechanisms during a ToIP session is required if network conditions are such that acceptable performance cannot be achieved without them. Appendix III provides guidance on the use of [IETF RFC 2198].

11.1 Single and dual port operation of text relay

Text relay packets shall be transmitted in the IP network on the same UDP port used for non text-relay packet (e.g., voice). Transmission of text relay packets on a separate UDP port is for further study.

11.2 Text relay throughput

Gateways compliant to this Recommendation shall support PTPs operating at their full character speed.

Characters received by the gateway from the PTP shall be transmitted to the IP network character by character. Characters should be sent into the IP network by the receiving gateway without delay. In the case that flow control is required (e.g., to honour the IP network character per second (CPS) value provided by the remote gateway through external signalling), some buffering of the character data before transmission into the IP or PSTN network may be required.

Appendix IV provides information on how a gateway should handle buffering for the case when different signalling rates are used on the different PSTN call legs.

12 Support of DTMF in ToIP

PTP devices that utilize DTMF tones to transmit characters as per Annex B of [ITU-T V.18] are supported by this Recommendation. DTMF tones should be carried in the IP network using [IETF RFC 2833], but may be supported using *UserInputIndication* H.245 messages for systems based on [ITU-T H.245]. Since DTMF PTP is only supported through the transmission of DTMF digits through the IP network without the discrimination of PTP DTMF digits from DTMF digits used for other purposes during the call (e.g., voice mail control, IVR), both end PTP devices must operate in DTMF mode for the end-to-end PTP connection to work.

13 Transition out of text relay

A gateway receiving PTP signals shall initiate a switch to audio mode from text relay mode upon detection of loss of carrier. A gateway may implement a silence guard time between detection of loss of carrier and the initiation of a switch to audio mode to avoid multiple mode transitions in the case of two closely-timed text spurts. This silence guard time is recommended to be between 700 and 1000 ms. If non-carrier signals are detected in this guard time, the gateway should immediately initiate a switch to audio mode.

For payload type switching, the gateway transitions to audio by encoding signals using the audio encoding and audio payload type.

For the SSE protocol, the gateway will generate an SSE(AUDIO) to the remote gateway to initiate the switch to audio mode (see clause 20).

The remote gateway, upon detecting the switch to audio mode initiated by the local gateway, shall stop generating carrier and transition to audio encoding, generating acknowledgement SSE(AUDIO) if SSE protocol is being used.

14 T.140 encoding of text relay

Annex E requires the encoding of all text data using [ITU-T T.140]. For consistency and ensuring compatibility, all transport of text, whether it uses RTP, SPRT or some other IP-TLP, shall use [ITU-T T.140].

14.1 Character set support

Gateways compliant with this Recommendation shall support 7-bit T.50 characters and 5-bit characters in Annex A of [ITU-T V.18] if the associated PTP mode is supported by the gateway for text relay. If the character set is supported, the gateway shall have the capability of mapping the T.50 or V.18 Annex A character set to T.140 upon demodulation of the PTP signal. The gateway shall also support the mapping of T.140 to T.50 or V.18 Annex A when remodulating the signal to the local PTP device. The mapping of the 5-bit character codes to 7-bit T.50 characters is described in Annex A of [ITU-T V.18].

For PTP modes that utilizes a character set other than T.50 or V.18 Annex A, issues may arise between mapping of certain national characters to a character set other than the PTP's character set. This may lead to the inability to map characters correctly for the case when using protocol conversion between two different national PTP modes. Mapping between national character sets is not addressed in [ITU-T V.18]. Support of character sets other than T.50 and 5-bit characters in Annex A of [ITU-T V.18] and for mapping between national character sets is for further study.

14.2 TIA-825A and T.50 encoding requirements

These requirements are required for gateways that support TIA-825A and T.50 Text telephones.

For the case where a gateway is receiving text data from a peer gateway which in turn is connected to a TIA-825A or T.50 Text telephone and if for some reason that gateway detects the loss of a character in the packet network (e.g., due to congestion or packet loss) then it shall use the apostrophe character to replace the missed character if no other mechanisms for determining the lost character value are used. Also, if the T.140 lost character (0xFFFD hexadecimal) is received, then it too shall be translated to an apostrophe when playing out the character to a TIA-825A or T.50 receiver.

Note that [ITU-T T.140] specifies that the 'new line' character is encoded as 0x2028 hexadecimal. If received by a gateway from a peer gateway and it is to be encoded to TIA-825A or T.50, the character shall be encoded as the two characters 'CR' 'LF' (Carriage Return and Line Feed).

15 Gateway-to-gateway protocol definitions and procedures

15.1 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 G of [ITU-T H.323], [ITU-T H.245] and SIP/SDP. The values shown in this set of messages represent what should be functionally indicated by the signalling protocol.

15.1.1 Text telephone modulation support

A list of the modulations supported in text relay mode by the gateway is shown in Table 15-1. Gateways compliant to this Recommendation shall advertise the support for at least one of these modulations.

Table 15-1 – PTP modulations

Modulation	Description
V18	V.18 "native mode" V.21 modulation
BELL103	Bell 103 based PTP modems (Annex D of [ITU-T V.18])
V23	Videotext terminals ("Minitel"; see Annex E of [ITU-T V.18])
V21	V.21 encoded per [ITU-T T.50] (Annex F of [ITU-T V.18])
EDT	European Deaf Telephone utilizing V.21 frequencies at 110 bit/s (Annex C of [ITU-T V.18])
TIA825	Baudot terminals at 45.45 bit/s or at 50 bit/s (Annex A of [ITU-T V.18])

15.1.2 Remain in VBD preference

This capability indicates the preference of the gateway to switch out of VBD mode when text signals are no longer present when using VBD for text transport. Both gateways must indicate this preference for it to be used in the call. Table 15-2 shows the valid values for VBD mode preference.

Table 15-2 – VBD mode preference

VBD mode preference	Description
False (default)	Stay in VBD mode for the duration of the call.
True	Switch back to AUDIO mode when text signals are no longer present.

15.1.3 Characters per second (CPS)

This optional parameter is an indication of the maximum number of characters per second that may be received in a session. If this parameter is not received by a gateway during the signalling of the call, a value of 30 shall be used. Gateways shall utilize this parameter to implement flow control with their local PTP device so as not to exceed transmission of characters into the IP network at a rate higher than indicated.

15.1.4 VBD parameters

The specification of the VBD parameters is found in [ITU-T V.152].

15.1.5 SSE parameters

If the optional SSE protocol is to be used, parameters are signalled as per clause E.1.3 of [ITU-T V.150.1] for SDP and clause F.6 of [ITU-T V.150.1] for H.245. To use the SSE protocol instead of the default payload type switching protocol for mode control, both gateways must signal the support for all of the SSE event encodings defined in Table 15-3.

Table 15-3 – Required SSE event encodings

Event encoding (Decimal)	Indicated media state
1	Initial Audio
2	Voice Band Data (VBD)
5	Text Relay
6	Text Relay Probing

15.1.6 SPRT parameters

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

Maximum payload size of SPRT channels 0 to 2, and maximum window size for SPRT channels 1 and 2. SPRT channel 3 shall not be used, so no configuration of it is required.

15.2 Gateway call discrimination messages

15.2.1 SSE reason identifier codes

Reason Identifier Codes to be used with the optional SSE protocol are defined in Table 12 of [ITU-T V.150.1]. The appropriate SSE RIC shall be sent with the SSE messages when the SSE protocol has been successfully negotiated between the gateways.

16 Start-up mode of operation

ToIP gateways will need to co-exist with other "over-IP" mechanisms. (For example Voice over IP, Facsimile over IP, and Modem over IP.) The entry into Text over IP mode corresponding to the beginning of the call discrimination procedures defined in clause 20 happens when certain signals are detected while the gateway is operating in Voice over IP mode. Text over IP mode may also be entered from the Modem over IP (V.150.1) call discrimination procedures as described in clause 16.1.

16.1 Inter-relationship with V.150.1

If V.150.1 support is implemented and successfully negotiated by the gateway (in addition to V.151 support), text relay mode as described in this Recommendation shall be entered into using the SSE protocol procedures as described in clause 18 of [ITU-T V.150.1] for supported PTP types that utilize answer tones. For supported PTP types that do not use answer tones, the procedures defined in this Recommendation shall be used.

16.2 Inter-relation with V.152

If V.152 support is implemented and successfully negotiated by the gateway (in addition to V.151 support) and V.150.1 support is not successfully negotiated or implemented by the gateway, the procedures described in this Recommendation shall take precedence over those defined in [ITU-T V.152] for signal stimuli that represents V.151 text relay mode supported PTP types. For non-PTP signal stimuli that would initiate use of VBD operation, procedures defined in [ITU-T V.152] shall be used.

17 Voiceband data modem interworking requirements

Certain PTP modes are indistinguishable from voiceband data applications. Examples of this include PTP modes that utilize V.21 and V.23 modulations. Call discrimination procedures defined in clause 20 have been designed so that if both end devices are voiceband data modems that use such modulations, the connection will result in use of text relay between the two end modems if V.151 text relay using these modulations is supported by both gateways. If either gateway does not support the modulation for text relay, the connection will result in VBD mode between the two gateways.

In the case that the connection results in text relay, the modem data will be encoded utilizing T.140 encoding. Data integrity between the end modems will be ensured as long as the data modems are using 8-bit character sizes (including parity). For other character sizes, the data will be corrupted due to the assumption that the data is encoded using T.140.

So as not to corrupt the data in the case that text relay mode is used for voiceband data modems, the CPS (character per second) value advertised by the gateway in the call signalling parameters should be greater than or equal to the highest signalling rate for all modulations supported by the gateway.

18 Facsimile interworking requirements

Facsimile uses similar signals to those used by certain PTP devices, including V.8 start-up sequence and answer tone. If ToIP is supported for modulations that use signals also used for facsimile, the procedures defined in this Recommendation may be exited and facsimile processing procedures entered (e.g., relay or VBD) upon detection of the signal being generated by a facsimile instead of a PTP. Examples of these detection events include recognition of the CM call function as indicating facsimile, or the existence of a V.21 flag sequence.

19 Call set-up procedures

The call set-up procedures are defined in Annexes B and C of this Recommendation and in Annex G of [ITU-T H.323].

20 Call discrimination procedures

The following clause defines the procedures to be used by a ToIP gateway during the call discrimination phase of the connection. The call discrimination phase starts when text telephony signals are first present on the PSTN interface of either gateway. The call discrimination phase ends when end-to-end connectivity using V.151 is established between the two terminal PTP devices.

20.1 V.8 *bis* processing

In all operating modes (e.g., audio, VBD), ToIP 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, thereby disabling V.8 *bis* protocol between the PTP devices. The support of V.8 *bis* by the PTP devices is for further study.

20.2 V.8 CI/XCI processing

V.8 CI/XCI signals can be carried in audio mode or, optionally, cause initiation of a transition to VBD mode. If CI/XCI is carried in voice mode, there is a probability that the CI/XCI will not successfully be detected by the remote PTP. This misdetection is due to the possible corruption of the signal by the voice codec and other signal processing distortions implicit in voice mode.

[ITU-T V.8] does not treat CI/XCI as reliable signals (i.e., V.8 allows for the loss or non-transmission of CI/XCI in the protocol definition), so carriage of these signals using voice mode is acceptable. If CI/XCI is not detected properly by the remote PTP device, there can be up to an additional 3 seconds of delay for the ANSam tone generation relative to the case where CI/XCI is detected.

20.3 Call discrimination overview

Gateways compliant to this Recommendation shall transport PTP signals that they are capable of detecting and discriminating using either text relay transport or VBD transport. Text relay is the preferred mode of transport for the support PTP signals with VBD the fallback mode of transport. The fallback to VBD may occur because the end-to-end establishment of a PTP connection was not possible due to the lack of support of the local PTP modulation type at one or both of the gateways, or the inability to negotiate to text relay mode based on the procedures defined in this clause.

Appendix II provides flow diagrams of example call discrimination scenarios that are useful in understanding the call discrimination procedures described in this clause. These call flows are examples only. The text and SDL diagrams provided in this clause provide the normative description of the call discrimination for ToIP.

Depending on the type of modulation supported by the PTPs and the gateways, this Recommendation supports several call discrimination procedures.

A gateway that supports a subset of the V.18 PTP modulations may scale down the procedures defined in this clause. The scaled-down procedures will fully support text relay for the modulations supported by the gateway, with VBD support for those modulations not supported by the gateway.

20.4 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.

The SDL diagrams define the procedures at the gateway for call discrimination processing. These SDL diagrams are derived from the call discrimination call flows given in Appendix II. The symbols used in the SDL diagrams are defined in Figure 20-1 below.

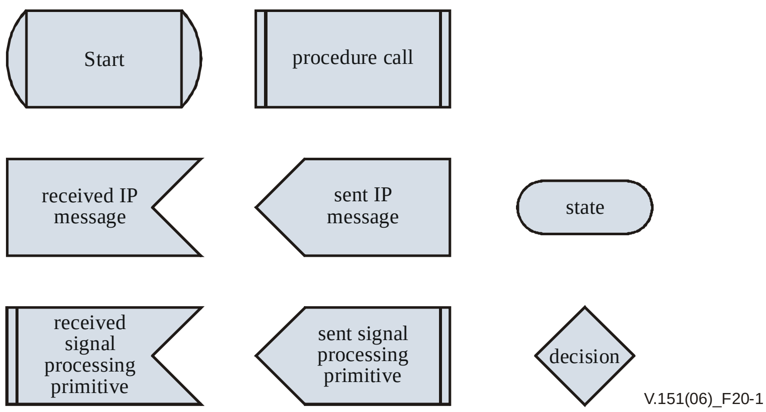


Figure 20-1 – SDL diagram definition

20.4.1 Detection of PTP signals

Table 20-1 describes the signals that shall be detected by the gateway for the purpose of transitioning to either VBD or text relay mode per the SDLs.

Table 20-1 – Signal detection criteria

Signal detected	Detected TTY mode	SSE RIC
390-Hz tone for 1 second	V.23 Annex E of [ITU-T V.18] call mode	V.23 Low channel
1300-Hz tone for 1 second	V.23 Annex E of [ITU-T V.18] answer mode	V.23 High channel
980-Hz tone for 1 second	V.21 Annex F of [ITU-T V.18] call mode	V.21 Ch1
1650-Hz tone for 1 second	V.21 Annex F of [ITU-T V.18] answer mode	V.21 Ch2
V.21 low-band modulated character, 110 bit/s	EDT Annex C of [ITU-T V.18]	V.21 Ch1
1270-Hz tone for 1 second	Bell 103 Annex D of [ITU-T V.18] call mode	Bell 103 modem
1400-Hz mark, 1800-Hz space FSK signals	Baudot Annex A of [ITU-T V.18]	TIA-825A (50 bit/s) TIA-825A (45.45 bit/s)

A PTP conforming to Annex C of [ITU-T V.18] can send a 980-Hz tone for 1 second after the last character transmitted in a burst of characters. Since, in this case, the 980-Hz tone follows a valid character sequence that indicates Annex C of [ITU-T V.18], the gateway should not use this signal to incorrectly designate the PTP as an Annex F of [ITU-T V.18] PTP.

The Bell 103 high-band mark tone (2225 Hz) is treated the same as ANS in the call discrimination procedures, not as a PTP mark tone. This is because a Bell 212 data modem connection will start out with a 2225-Hz tone as well, but a 1270-Hz tone will never be received, so the connection should stay in VBD.

The 1300-Hz V.23 mark should not be confused with a calling tone using the same frequency (duration of 0.5-0.7 s on). Implementations shall take care not to initiate transitions to VBD/text relay based on a calling tone.

Further details on the requirements for the detection of PTP characters for the purpose of mode transitions are given in clause 8.4.3.

20.4.2 SDL state variables

The SDLs make use of the *tty_mode* state variable to track the PTP modulation type detected by the gateway. This state variable is local to each gateway and is initialized to "unknown" at the start of the procedures.

20.4.3 Payload type switching SDL diagrams

The following SDL diagrams define the procedures at the gateway for call discrimination processing when using payload type switching. These SDL diagrams are derived from the call discrimination call flows given in Appendix II. The symbols used in the SDL diagrams are defined in Figure 20-1.

NOTE 1 – In the following diagrams (Figure 20-2) for payload type switching SDLs, the paths are only used when both gateways have indicated support for native V.18 text relay.

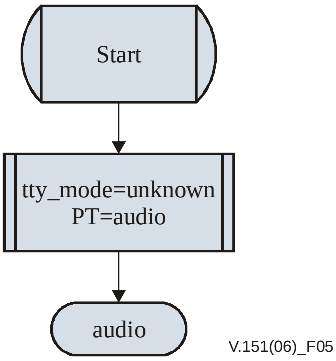


Figure 20-2 – SDL for payload type switching (sheet 1 of 6)

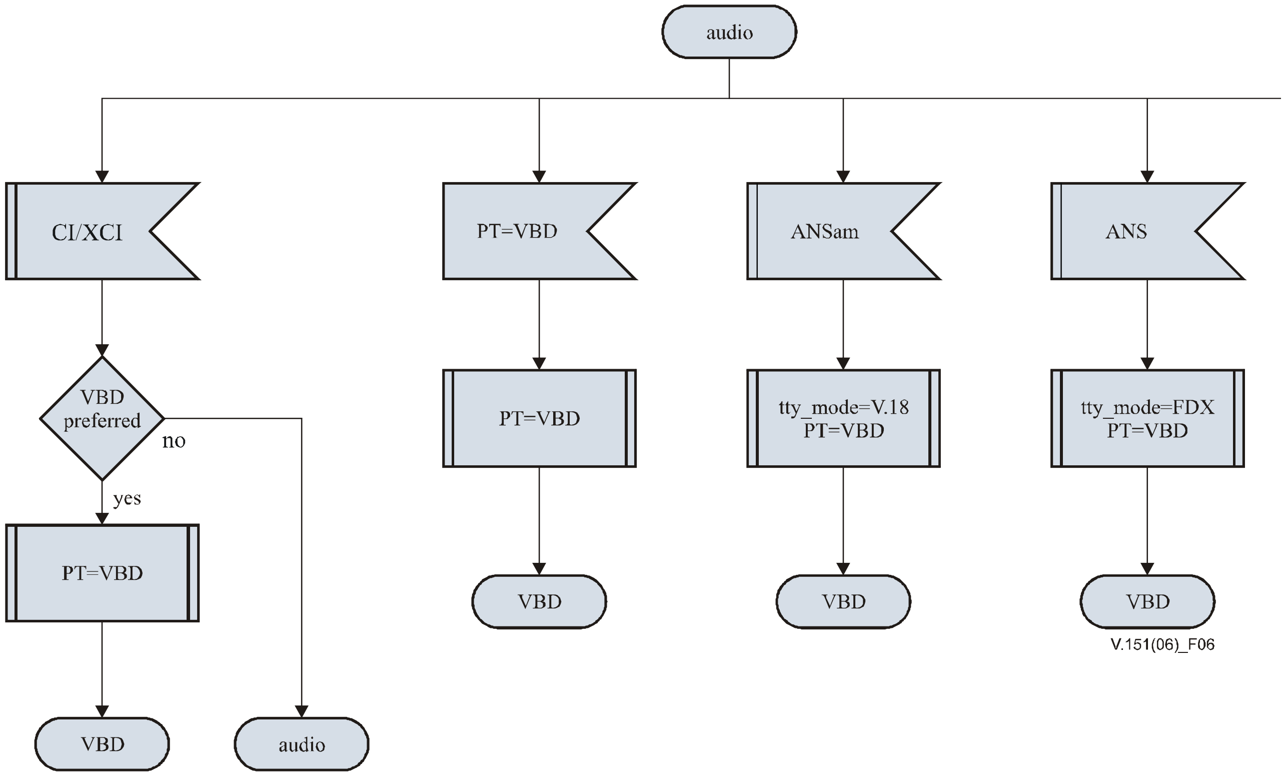


Figure 20-2 – SDL for payload type switching (sheet 2 of 6)

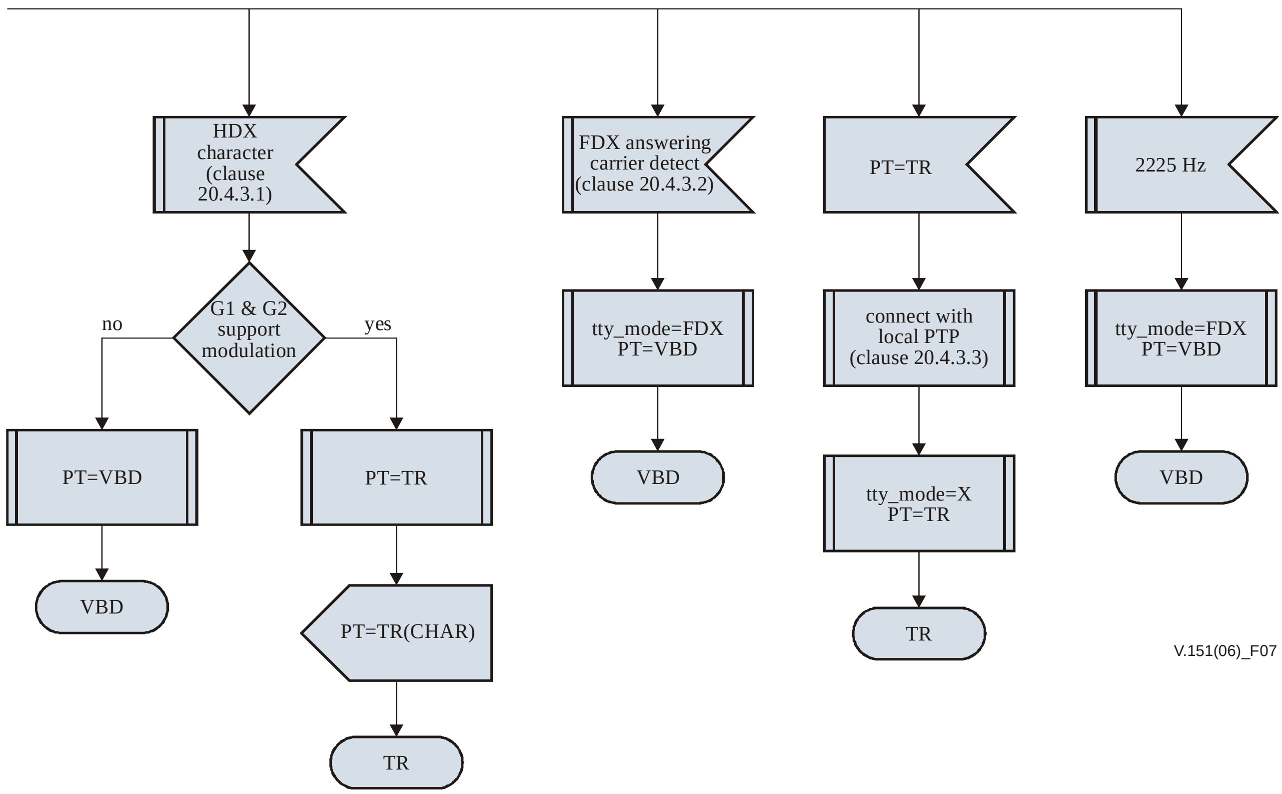


Figure 20-2 – SDL for payload type switching (sheet 3 of 6)

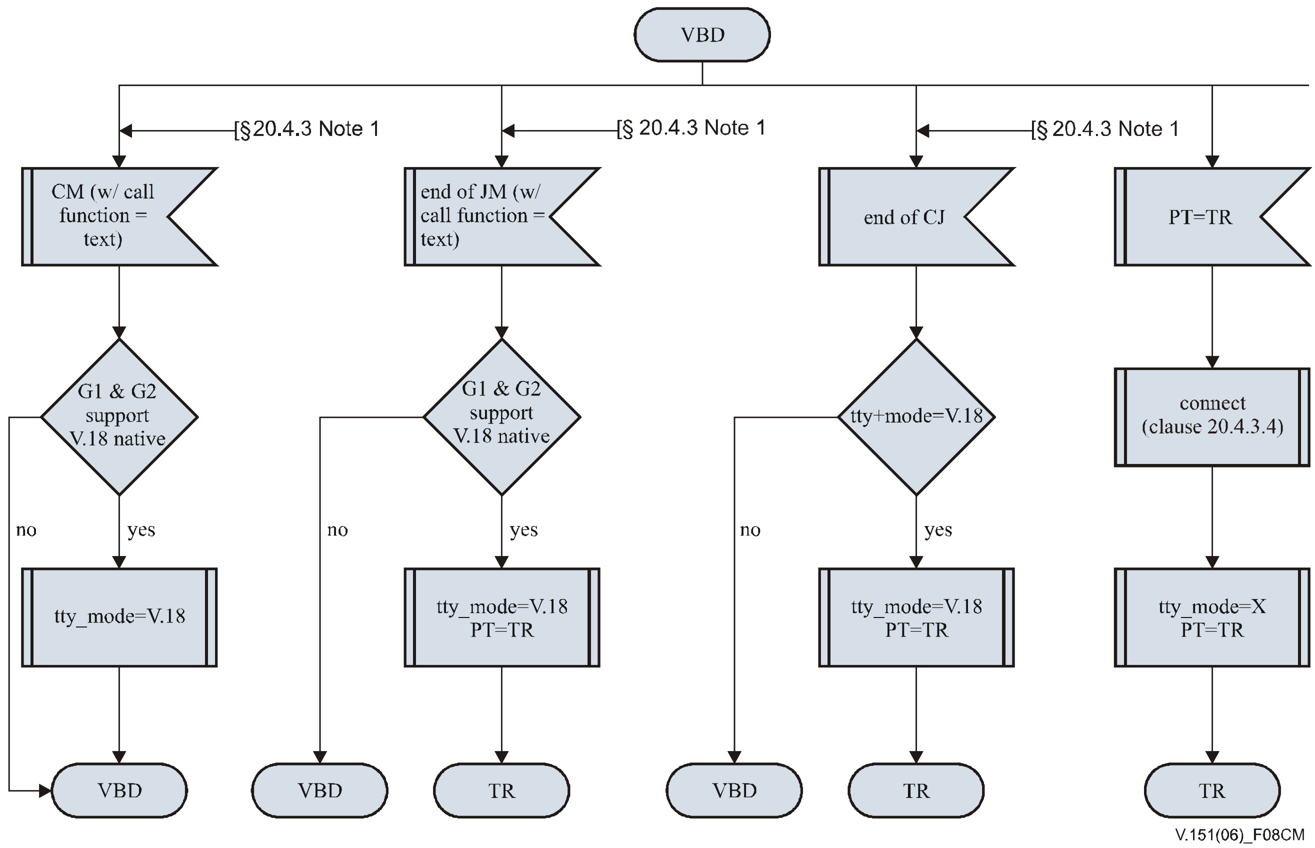


Figure 20-2 – SDL for payload type switching (sheet 4 of 6)

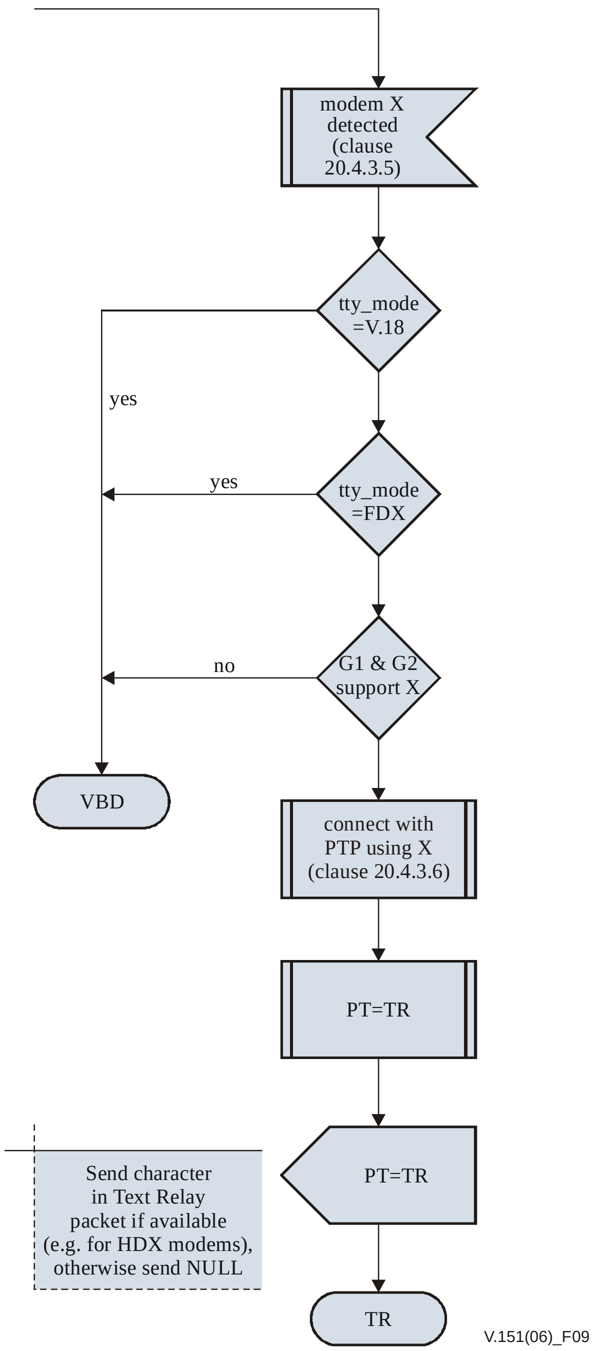


Figure 20-2 – SDL for payload type switching (sheet 5 of 6)

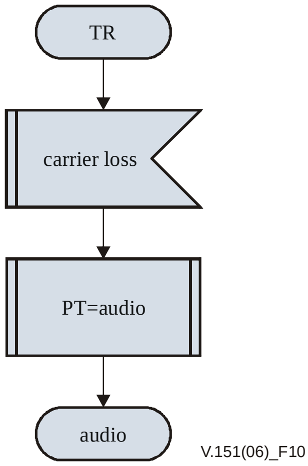


Figure 20-2 – SDL for payload type switching (sheet 6 of 6)

20.4.3.1 HDX character

This event corresponds to the detection of an HDX PTP signal as per clause 20.4.1.

20.4.3.2 FDX answering carrier detect

This event corresponds to the detection of a carrier generated by an FDX PTP device as described in clause 20.4.1.

20.4.3.3 Connect with local PTP

This procedure box is entered when a TR payload type is received while the connection is in audio mode. This is the case when an HDX modulation has been detected by the remote gateway and both gateways support this HDX modulation. The gateway entering this procedure box shall attempt to connect with its local PTP device using an HDX modulation.

20.4.3.4 Connect

In this procedure box, the gateway shall attempt to connect with its local PTP device using modulations that it supports. The gateway should be able to connect to its local PTP device since the remote gateway would not have sent the TR payload unless the PTP modulation that it detected was also supported for text relay by the local gateway. It is assumed for this case that the PTP devices are compatible.

The gateway shall perform the automode probing as defined in Figure 2b of [ITU-T V.18] and its corresponding text. The gateway shall not generate the ANSam signal at the beginning of these procedures, i.e., it shall skip over the generate ANSam block in Figure 2b of [ITU-T V.18] and go straight to the probing flows. The gateway shall only probe for modulations that it supports for text relay.

In order to allow for data modem connections over text relay, the gateway shall probe for supported FDX modulations before probing for supported HDX modulations.

20.4.3.5 Modem X detected

This event corresponds to the detection of a PTP modulation as per clause 20.4.1.

20.4.3.6 Connect with PTP using X

The gateway shall connect with its local PTP device using the modulation detected and transition the channel to text relay mode through the generation of TR encoded RTP packets to the remote gateway.

20.4.4 SDL for SSE protocol

The following SDL diagrams define the procedures at the gateway for call discrimination processing when using SSE protocol switching. These SDL diagrams are derived from the call discrimination call flows given in Appendix II. The symbols used in the SDL diagrams are defined in Figure 20-1.

NOTE 1 – In the following diagrams (Figure 20-3) for SSE SDLs, the paths are only used when both gateways have indicated support for native V.18 text relay.

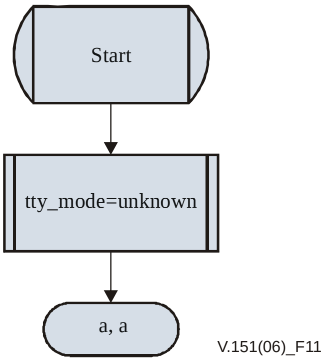


Figure 20-3 – SDL for SSE protocol (sheet 1 of 10)

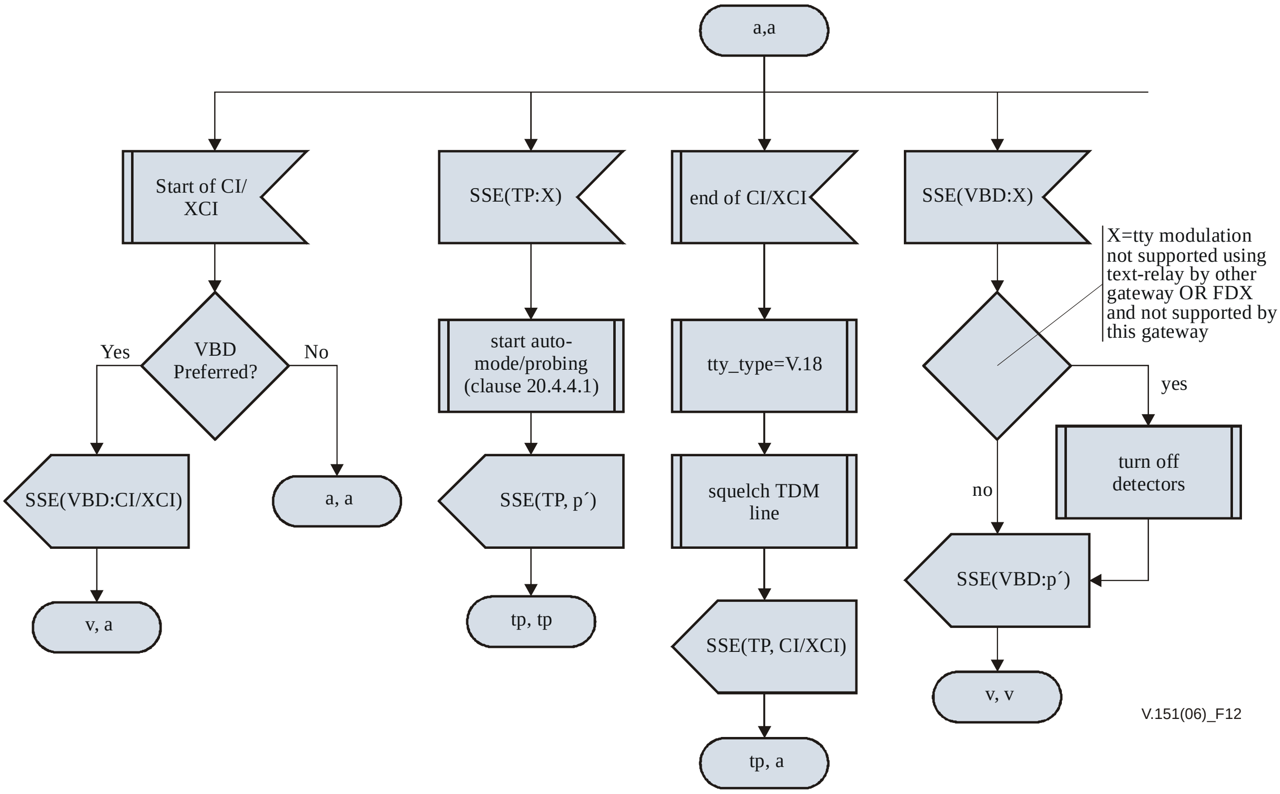


Figure 20-3 – SDL for SSE protocol state (a,a) Part 1 (sheet 2 of 10)

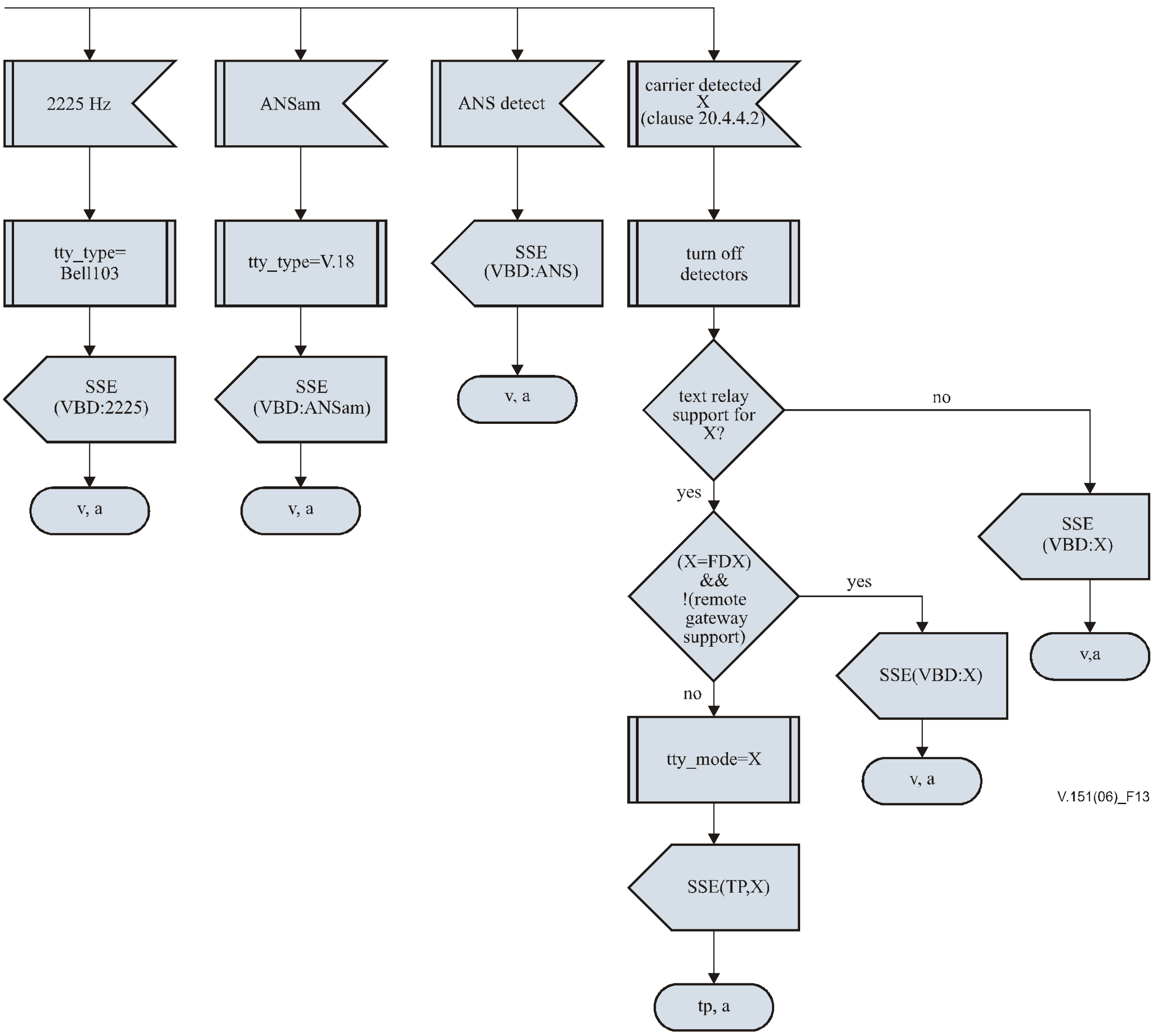


Figure 20-3 – SDL for SSE protocol state (a,a) Part 2 (sheet 3 of 10)

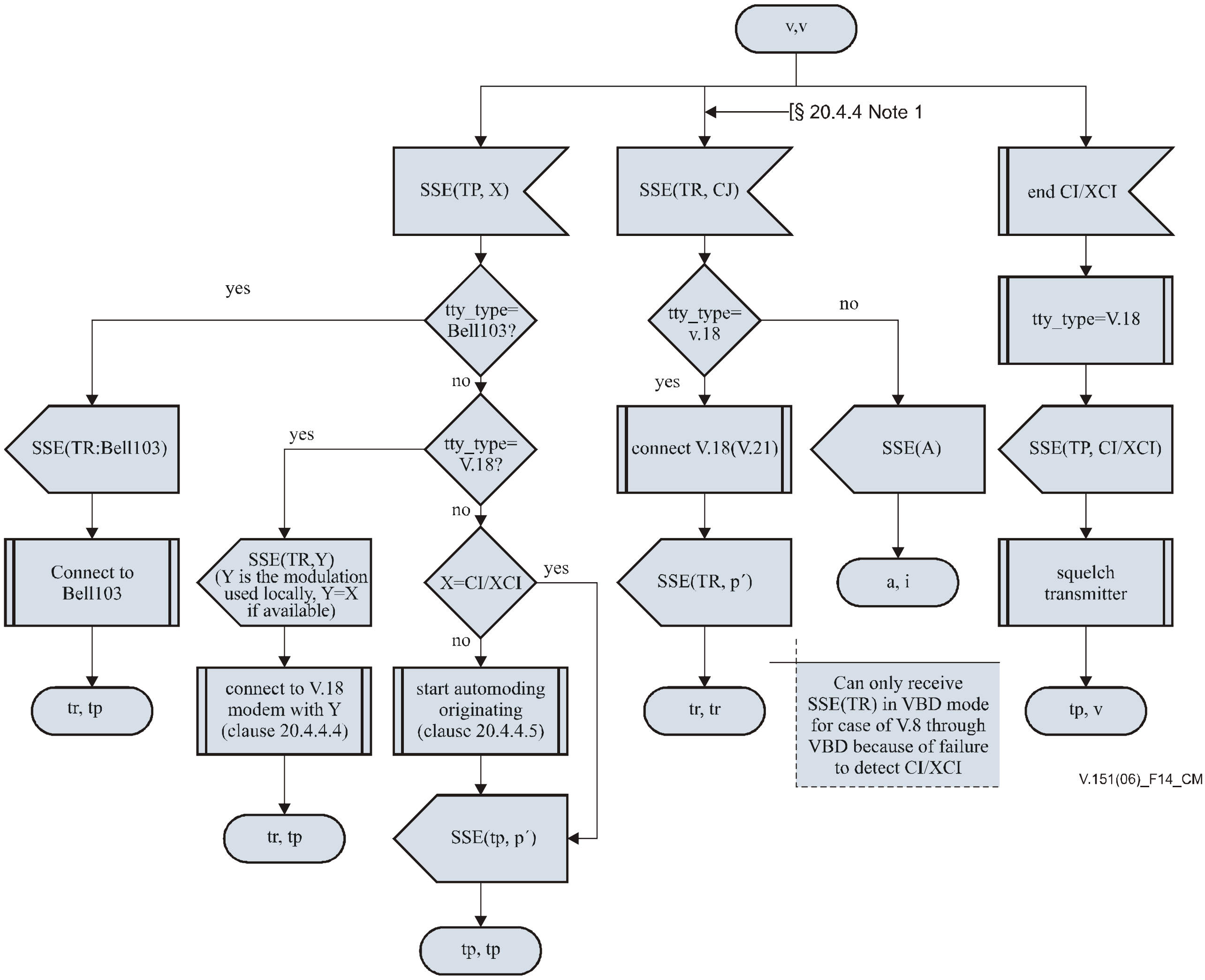


Figure 20-3 – SDL for SSE protocol state (v,v) Part 1 (sheet 4 of 10)

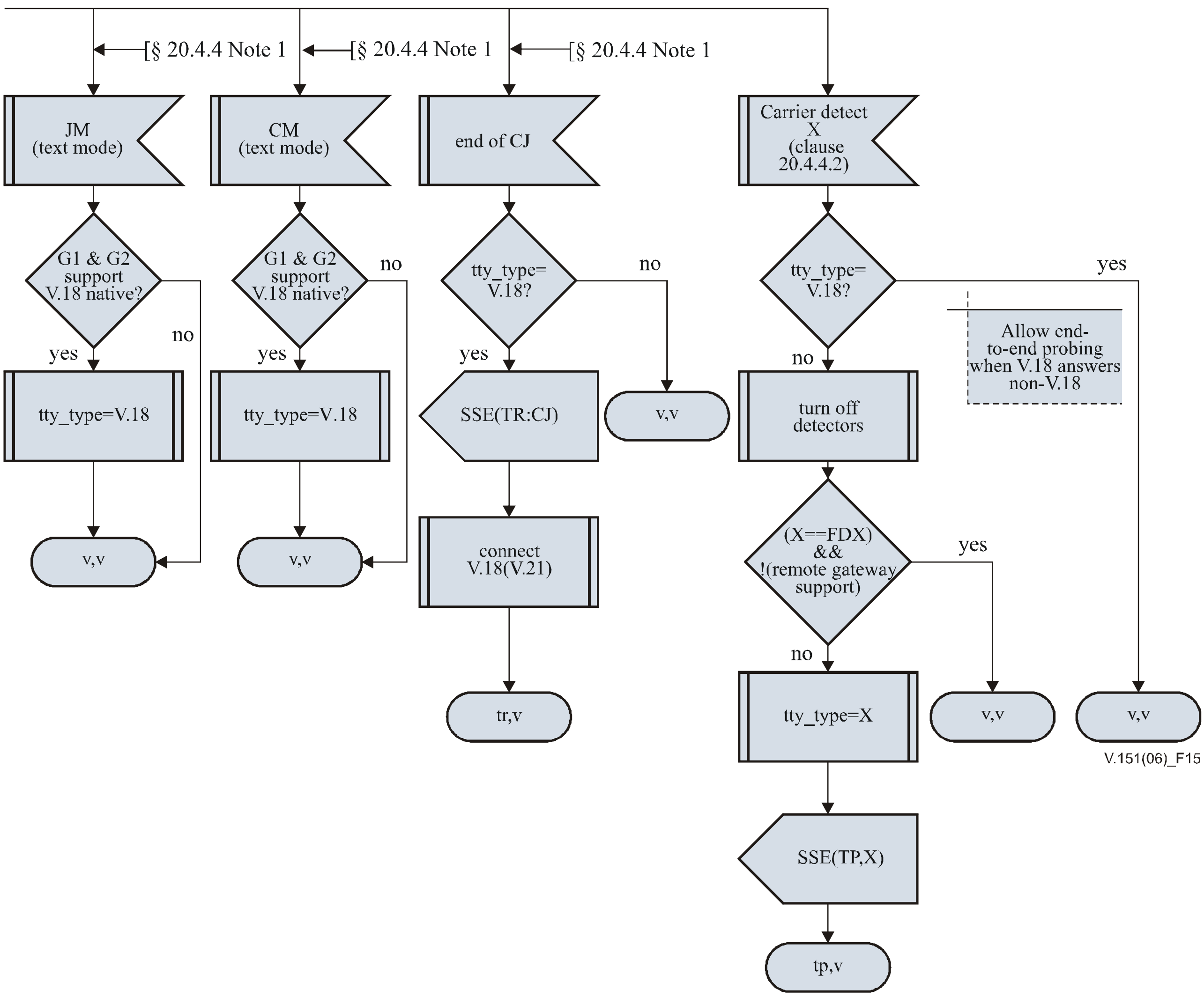


Figure 20-3 – SDL for SSE protocol state (v,v) Part 2 (sheet 5 of 10)

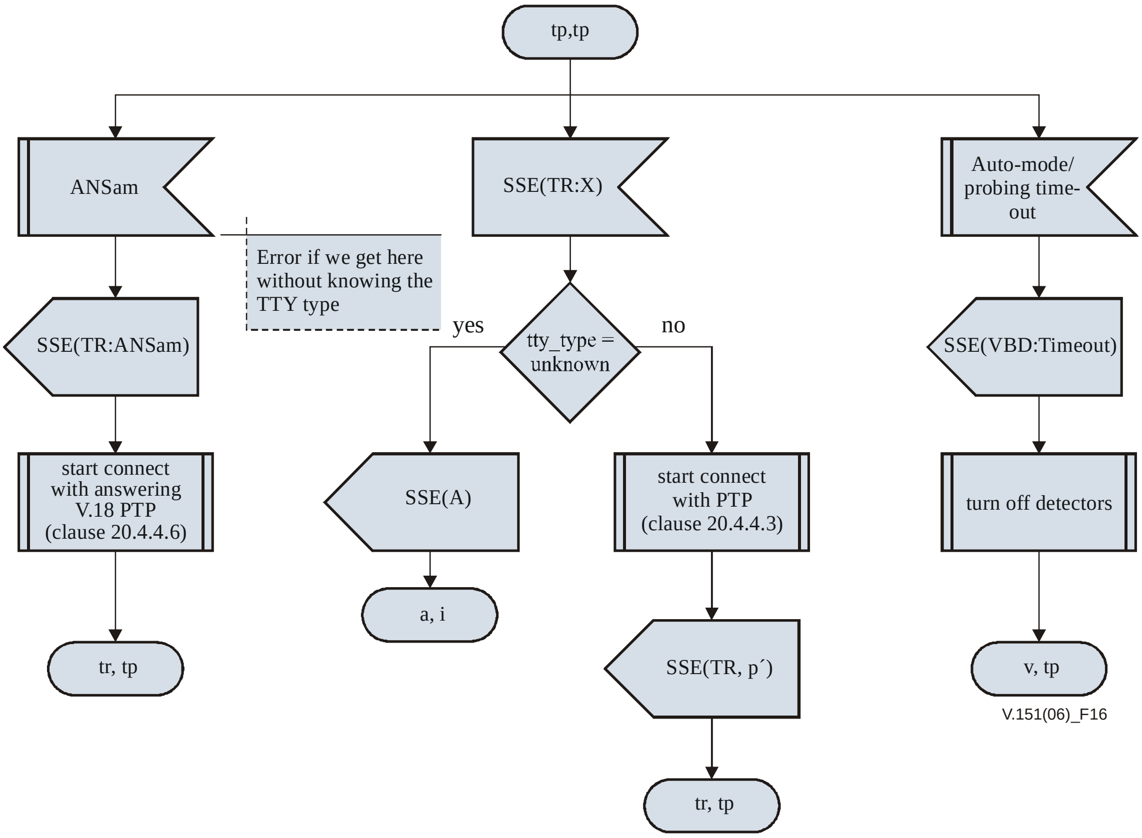


Figure 20-3 – SDL for SSE protocol state (tp,tp) Part 1 (sheet 6 of 10)

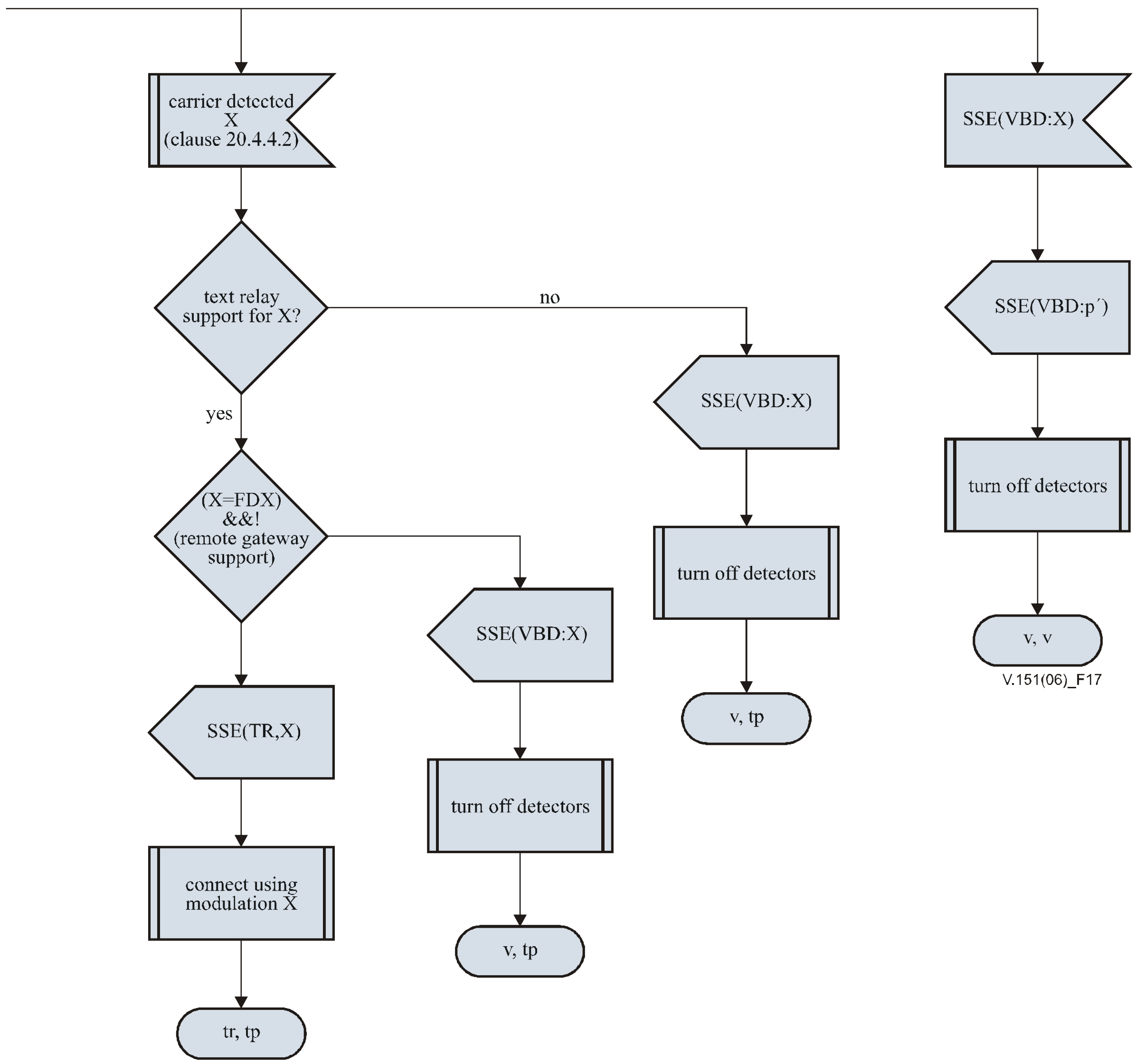


Figure 20-3 – SDL for SSE protocol state (tp,tp) Part 2 (sheet 7 of 10)

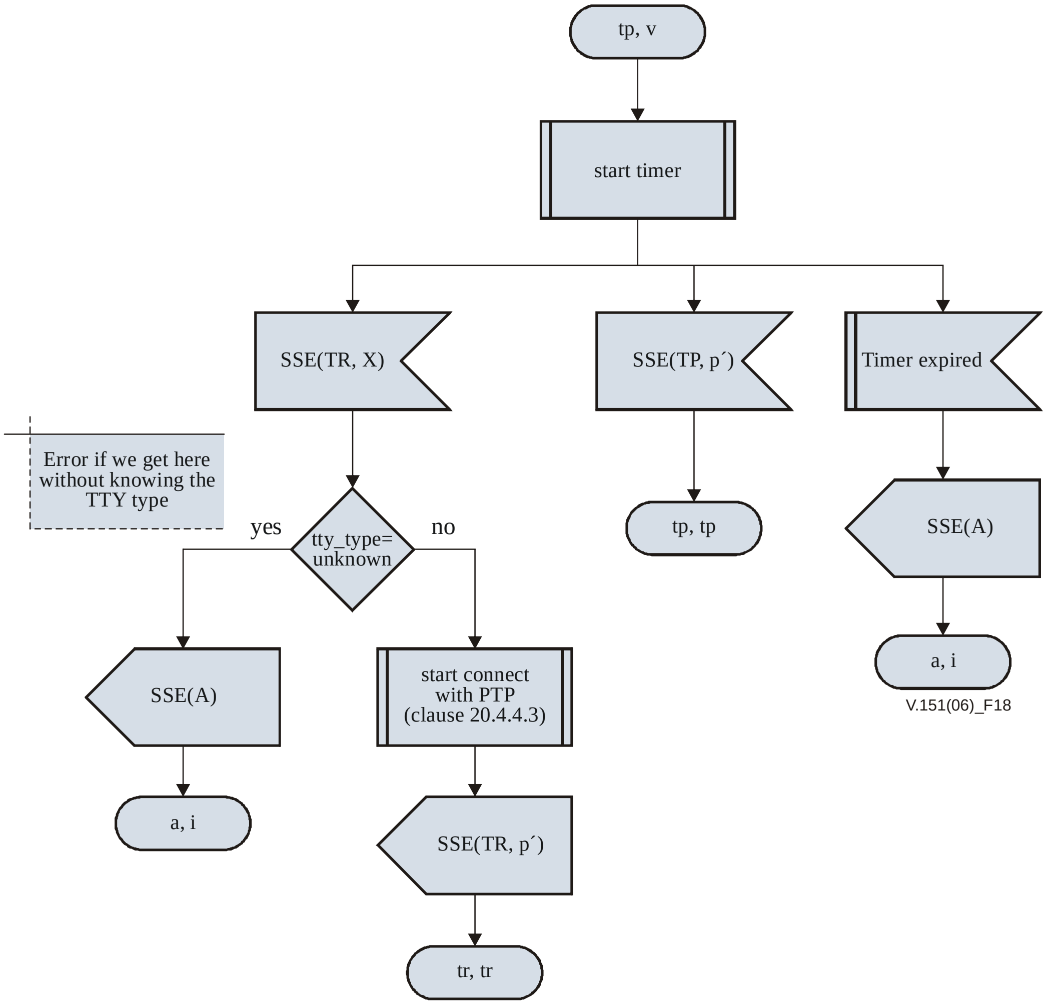


Figure 20-3 – SDL for SSE protocol (sheet 8 of 10)

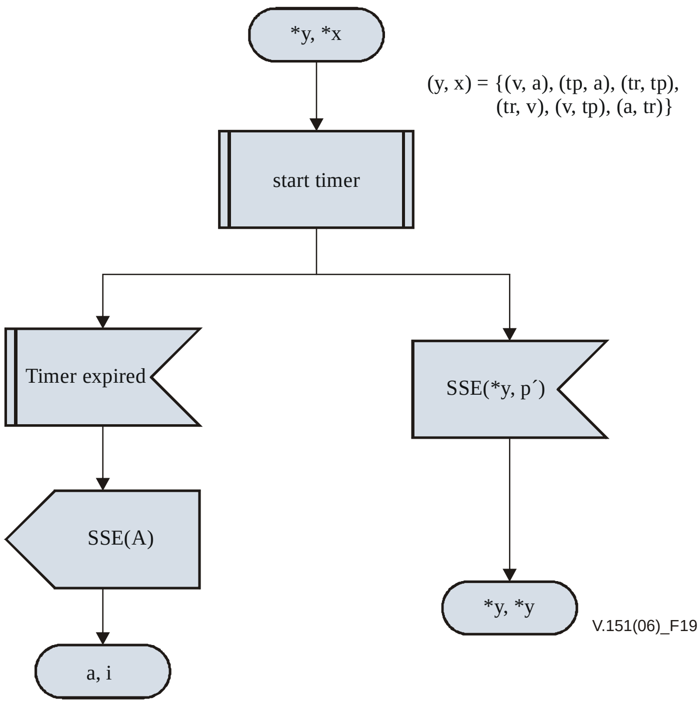


Figure 20-3 – SDL for SSE protocol (sheet 9 of 10)

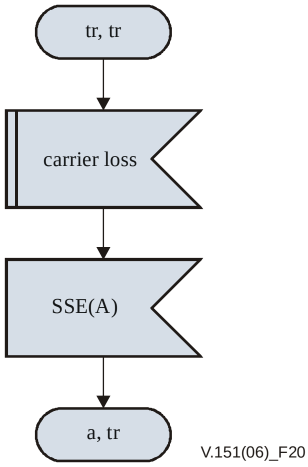


Figure 20-3 – SDL for SSE protocol (sheet 10 of 10)

20.4.4.1 Auto-mode/probing

In this procedure box, the gateway starts the auto-mode/probing processing described in this clause. The purpose of the auto-mode/probing process is to determine what modulation its local PTP device supports. This procedure starts when the gateway receives an SSE(TP) from the remote gateway. If the RIC in the SSE(TP) indicates CI/XCI was detected, the gateway shall perform the "automoding originating" procedure as defined in clause 5.1 of [ITU-T V.18] with the modification of only looking for modulations that it supports for text relay.

If the RIC in the SSE(TP) indicates a signal detected other than CI/XCI, the gateway shall perform automode probing as defined in Figure 2b of [ITU-T V.18] and its corresponding text. The gateway shall not generate the ANSam signal at the beginning of these procedures, i.e., it shall skip over the generate ANSam block in Figure 2b of [ITU-T V.18] and go straight to the probing flows. The gateway shall only probe for modulations that it supports for text relay.

The gateway shall continue the auto-mode/probing procedures until one of the following events occur:

– A PTP modulation that is supported for text relay is detected.

– A PTP modulation that is not supported for text relay is detected.

– Auto-mode/probing time-out occurs.

Auto-mode/probing time-out occurs when the gateway has proceeded through at least two cycles of probing as defined in [ITU-T V.18] for the modulations that it supports for text relay without detection of a response from the local PTP device.

In the scenario where an HDX PTP is originating the call and that PTP generates characters before the answering PTP generates any signal, the gateway connected to the answering PTP would begin the probing sequence as described in this clause. Since the PTP device that the gateway is probing is the answering PTP, there is likelihood that the probing would fail in the case the answering PTP is HDX. The fall back to VBD would not result in a successful connection because the two PTP devices are of different types; i.e., the originating is HDX and the answering is FDX. If the answering PTP is FDX, the probing should still be successful and the PTP modems connect using text relay.

If a gateway supports a subset of the modulations described in [ITU-T V.18] for text relay, the gateway only probes for those modulations it supports. If the gateway supports the modulation indicated by the SSE(TP) RIC, the gateway shall start probing with this modulation if the modulation is an FDX type, else the gateway should start probing with this modulation (i.e., for HDX type).

20.4.4.2 Carrier detect X

In this event box, the gateway has detected a signal that indicates the presence of a local PTP device. Clause 20.4.1 describes the signals that shall be detected for determining the presence of a local PTP device.

20.4.4.3 Start connect with PTP

In this procedure box, the gateway shall start the connect sequence with its local PTP.

If the gateway's *tty_mode* state variable is V.18, the gateway shall use the modulation indicated by the RIC of the received SSE(TR) to connect with the PTP if that modulation is supported by the gateway. If the gateway does not support the modulation indicated by the RIC in the received SSE(TR), the gateway shall use a supported modulation that is closest to the signalling rate of the modulation indicated in the RIC.

If the gateway's *tty_mode* state variable is not V.18, the gateway shall use the modulation indicated by *tty_mode* to connect with the local PTP device.

20.4.4.4 Connect to V.18 modem with Y

A gateway will enter this procedure box only if it is connected with a V.18-compliant PTP device.

The gateway shall use the modulation indicated by the RIC of the received SSE(TR) to connect with the V.18 PTP if that modulation is supported by the gateway. If the gateway does not support the modulation indicated by the RIC in the received SSE(TR), the gateway shall use a supported modulation that is closest to the signalling rate of the modulation indicated in the RIC.

20.4.4.5 Start "automoding originating"

The gateway shall perform the "automoding originating" procedure as defined in 5.1 of [ITU-T V.18]. The gateway shall continue the "automoding originating" processing until one of the following events occurs:

– A PTP modulation that is supported for text relay is detected.

– A PTP modulation that is not supported for text relay is detected.

– Auto-mode/probing time-out occurs.

Auto-mode/probing time-out occurs when the gateway has proceeded through at least two cycles of probing as defined in [ITU-T V.18] for the modulations that it supports for text relay without the detection of a response from the local PTP device.

20.4.4.6 Start connect with answering V.18 PTP

When this procedure box is entered, both PTP devices are V.18-compliant. A gateway that enters this procedure box is connected to the answering V.18 PTP device. The gateway shall use V.21 as defined in V.18 if this modulation is supported. If V.21 as defined in [ITU-T V.18] is not supported, the gateway should use a supported modulation that is closest to the V.21 signalling rate.

20.5 Visual flow control

During normal call discrimination procedure there may be situations where the local PTP device is connected for a time to the local gateway without an end-to-end connection established between the endpoint PTP devices. An example would be a local PTP connected to its local gateway while the remote gateway is performing a probing sequence with its local PTP device in order to establish a connection with it. In these situations, the gateway could optionally provide a message to its local PTP device indicating to the user to wait before sending text. The gateway could then follow up with a message indicating to the user that it is clear to send once the remote gateway establishes a connection with its PTP device. The messages to be used are beyond the scope of this Recommendation.

Annex A

Procedures for the optional support of SPRT protocol

(This annex forms an integral part of this Recommendation)

A.1 Overview

To use SPRT protocol, the SSE protocol must be negotiated in addition to SPRT being negotiated. If the SSE protocol is not negotiated, RTP shall be used for transmitting T.140 encoded data.

Parameters for SPRT are defined in Annex B of [ITU-T V.150.1]. Transport channel 3 shall not be used for transporting T.140 encoded data.

SPRT defines several formats for carrying data. If SPRT is used then the default format for transporting T.140 encoded data shall be I_OCTET_INFO as described in clause 15.4.11.1 of [ITU-T V.150.1].

A.2 SDP negotiation

The SPRT capability is advertised in SDP as a latent capability following the syntax specified in [IETF RFC 3407]. The media type is "audio" and the transport name assigned "udpsprt". The media format is "v150tr". The following example illustrates the advertisement of an RTP media stream that uses G.711 for audio and SPRT for the transport of real-time text:

```
m=audio 49230 RTP/AVP 0
a=sqn:0
a=cdsc:1 audio udpsprt 100
a=cpar:a=sprtmap:100 v150tr/8000
```

In the above example, the payload type associated with SPRT is 100.

The optional parameters associated with the SPRT transport protocols are declared via the SDP attribute "sprtparm" and vendor-specific parameters are declared via the SDP attribute "vndpar" as illustrated in the syntax below:

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

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

Refer to Annex E of [ITU-T V.150.1] for a full description of the above parameters.

A.3 H.245 negotiation

The SPRT capability is advertised in [ITU-T H.245] as a generic audio capability. The capability is specified as follows:

Table A.1 – Capability identifier for V.150.1

Capability name	SPRT Text Relay Capability
Capability class	Audio capability
Capability identifier type	Standard
Capability identifier value	{itu-t (0) recommendation (0) v (22) 151 toip (0) }
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 below.
Transport	This field shall not be included and shall be ignored if received.

The ASN.1 syntax associated with SPRT is:

```
V151SPRT-CAPABILITY DEFINITIONS AUTOMATIC TAGS ::= BEGIN

IMPORTS
      NonStandardParameter FROM MULTIMEDIA-SYSTEM-CONTROL;

V150MoIPCapability ::= SEQUENCE
{
      nonStandard       SEQUENCE OF NonStandardParameter OPTIONAL,
      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,

      ...
}
END
```

Equipment manufacturers may use the **nonStandard** field to signal any non-standard information that is specific to their ToIP implementations as it relates to SPRT. The first octet of the data field within the non-standard parameter shall be the vendor-specific "vendor-tag", as specified in clause 8 of [ITU-T V.150.0]. A value of 0 in the first octet means that the parameters is not bound to the specified vendor ID (i.e., it is not bound to the value of **nonStandard.object** or **nonStandard.h221NonStandard** values). This "vendor-tag" may be used in other related messages, such as SSE messages.

Refer to Table B.2 of [ITU-T V.150.1] for description and default values for the **sprtParameters**.

Annex B

Definition of capabilities for use within H.245-based systems

(This annex forms an integral part of this Recommendation)

This annex defines the capabilities that need to be exchanged between H.245-based systems for the transmission of analogue PSTN text telephony signals over packet-based networks.

The payload format used for the transmission of text characters is described in Annex E.

H.323 systems have a means of advertising the ability to transport text over RTP using the "t140" capability defined in [ITU-T H.245]. However, this capability does not provide a means to advertise supported modulations, such as Baudot. Annex G of [ITU-T H.323] also does not define those modulations, but it does have a means to allow another specification, such as [b-TIA-1001] or [ITU-T V.151], to do that.

Gateways compliant to this Recommendation must be able to support the transmission of text interleaved with audio. The respective capability definition for H.245-based systems is in Table B.1, as taken from Annex G of [ITU-T H.323].

Table B.1 – H.245 capability definition for text transport (T140Audio)

Capability name:	T140Audio
Capability class:	Audio Capability
Capability identifier type:	Standard.
Capability identifier value:	itu-t (0) recommendation (0) h (8) 323 annex(1) g (7) audio(0)
maxBitRate	The maxBitRate field shall be included and indicate the maximum bits per second. When using the Flow Control Command or other signals in relation to this capability, any maxBitRate field shall be interpreted to be in units of bits/s, as opposed to the typical 100 bits/s used in H.245. This is due to the low bit rate nature of real-time text communication, including the low bit rates used by many PSTN textphone protocols.
nonCollapsing	This field shall not be included and shall be ignored if received.
nonCollapsingRaw	This field shall not be included and shall be ignored if received.
Transport	This field shall not be included.

Devices may also advertise, either in the terminal capability set, the Open Logical Channel, or both, the capability to receive a specified number of characters per second. Devices may advertise the maximum characters per second via the below optional parameter.

Table B.2 – Characters per second parameter

Parameter name:	cps
Parameter description:	This is a collapsing capability. Indicates the maximum number of characters per second that may be received on a session. When carried inside an Open Logical Channel (OLC), it indicates the maximum transmission rate that the other endpoint may use if it opens a corresponding text session.
Parameter identifier value:	standard: 0
Parameter status:	Optional
Parameter type:	unsignedMin
Supersedes:	–

The Textphone Modulations parameter is an optional parameter that indicates the set of supported modulations for text relay. Absence of this parameter may be interpreted to mean that all modulations are supporter or that the device is a native IP device.

Table B.3 – Textphone Modulations Parameter

<table>
<tr><td>Parameter name:</td><td>Textphone Modulations</td></tr>
<tr><td>Parameter description:</td><td>This is a non-collapsing capability.
Indicates the gateway's supported textphone modulations.</td></tr>
<tr><td>Parameter identifier value:</td><td>standard: 100</td></tr>
<tr><td>Parameter status:</td><td>Optional</td></tr>
<tr><td>Parameter type:</td><td>booleanArray, populated as follows (bit 0 = lsb)
<table>
<tr><th>Bit</th><th>Meaning</th></tr>
<tr><td>0</td><td>TIA825</td></tr>
<tr><td>1</td><td>EDT</td></tr>
<tr><td>2</td><td>BELL103</td></tr>
<tr><td>3</td><td>V23</td></tr>
<tr><td>4</td><td>V18</td></tr>
<tr><td>5</td><td>V21</td></tr>
<tr><td>6-7</td><td>Reserved</td></tr>
</table></td></tr>
<tr><td>Supersedes:</td><td>–</td></tr>
</table>

Devices may advertise their preference for switching out of VBD mode between text spurts when VBD mode is used for text.

Table B.4 – Remain In VBD Not Preferred Parameter

Parameter name:	RemainInVBDNotPreferred
Parameter description:	This is a collapsing capability. Indicates that the gateway would prefer to switch out of VBD mode between text spurts. Both gateways must indicate this preference for the gateway to switch out of VBD mode after a text spurt, else the gateway shall remain in VBD mode for the duration of the call.
Parameter identifier value:	standard: 101
Parameter status:	Optional
Parameter type:	Logical
Supersedes:	–

These capabilities are advertised as part of the Terminal Capability Set message exchanged between two devices. These capabilities are also used in Open Logical Channel (OLC) messages in Fast Connect and normal H.245 logical channel signalling.

Below is an example of an OLC message that has G.729 audio, text, and redundancy protecting the text packets, and RFC 2833 for DTMF transport.

```
{
        forwardLogicalChannelNumber 1,
        forwardLogicalChannelParameters {
           dataType : multiplePayloadStream {
            element {
              dataType : audioData : g729 2
            },
            element {
              dataType : redundancyEncoding {
               primary {
                 dataType : audioData : genericAudioCapability {
                   capabilityIdentifier : standard {
                     itu-t (0) recommendation (0) h (8)
                     323 annex(1) g (7) audio(0)
                   },
                   nonCollapsing {
                   {
                     parameterIdentifier : standard : 100,
                     parameterValue : booleanArray : 00000011b
                   },
                   {
                     parameterIdentifier : standard : 101,
                     parameterValue : logical : NULL
                   },
                 },
                 payloadType 97 -- The PT for the redundant encoding
               },
               secondary {
                 {
                   dataType : audioData : genericAudioCapability {
                     capabilityIdentifier : standard {
                         itu-t (0) recommendation (0) h (8)
                         323 annex(1) g (7) audio(0)
                     },
                     nonCollapsing {
                     {
                         parameterIdentifier : standard : 100,
                         parameterValue : booleanArray : 00000001b
                   },
                   {
                     parameterIdentifier : standard : 101,
                     parameterValue : logical : NULL
                   },
                   payloadType 97 -- The PT for the redundant encoding
```

```
          }
        }
      },
      payloadType 101     -- The PT for the RFC 2198 packet
    },
    element {
      dataType : audioData : audioTelephonyEvent {
         audioTelephoneEvent : "0-15"
      },
      payloadType 102
    }
  }
},
  multiplexParameters : h2250LogicalChannelParameters {
  sessionID 1
}
}
```

Note that the gateway only indicated support for [b-ANSI/TIA-825] (Baudot) and EDT PSTN PTP devices in the above example. The assumption is that the gateway will either use VBD (not shown in this example) for other PSTN PTP types, or will transition to text relay in order to provide interworking between dissimilar PSTN PTP types.

Annex C

SDP description of sessions supporting V.151

(This annex forms an integral part of this Recommendation)

This annex defines the syntax that needs to be exchanged between SDP-based systems for the transmission of analogue PSTN Text Telephony signals over packet-based networks. The payload format used for the transmission of text characters is described in Annex E.

The text relay capability and associated procedures are represented in SDP using the MIME type "audio/t140c". A very simple example of the use of the audio/t140c within SDP is:

```
m=audio 7200 RTP/AVP 18 100
a=rtpmap:98 t140c/8000
```

This specifies the use of G.729 as payload type 18 along with audio/t140 as payload type 100. Note that the clock rate specified is 8000 Hz. The clock rate value should have the same value as for any audio codec packets interleaved in the same RTP stream.

In some cases, it is necessary to limit the rate at which characters are transmitted. For example, when a PSTN gateway is interworking between an IP device and a PSTN textphone, it may be necessary to limit the character rate from the IP device in order to avoid throwing away characters in case of buffer overflow at the PSTN gateway.

To control the character transmission rate, the MIME parameter "cps" in the "fmtp" attribute is defined. It is used in SDP with the following syntax:

```
a=fmtp:<format> cps=<integer>
```

The <format> field is populated with the payload type that is used for text. The <integer> field contains an integer representing the maximum number of characters that may be received per second. The value shall be used as a mean value over any 10-second interval.

Devices in receipt of this parameter shall adhere to the request by transmitting characters at a rate at or below the specified <integer> value. In absence of this parameter, devices shall not transmit more than 30 characters per second.

The 'gpmd' (general purpose media descriptor) syntax attribute is used for signaling the set of supported textphone modulations. The general format for this attribute line is:

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

The value of <format> shall be set to the value of the RTP payload type for text. The <parameter list> shall be a comma-separated list of modulations following the following syntax:

```
modulations   =    "tpmods=" modulation *("," modulation)
modulation         =    "tia825" | "edt" | "bell103" | "v23" |
"v18" | "v21"
```

Absence of the textphone modulations attribute may be interpreted to mean all modulations are supported or that the device is a native IP device. A gateway compliant to this Recommendation shall send the textphone modulations attribute.

The VBD mode preference attribute is used to indicate the preference a gateway has for switching out of VBD mode and into AUDIO mode between text spurts when VBD is used for transmission of text. Both gateways must indicate the preference to switch out of VBD mode after a text spurt else the gateway shall remain in VBD mode for the duration of the call. The value of <format> shall be set to the value of the RTP payload type for text. The <parameter list> shall indicate the gateway's preference. A 'remain-in-vbd=yes' indicates preference to stay in VBD mode, while 'remain-in-vbd=no' indicates preference to switch between VBD and AUDIO modes between text spurts. This attribute is optional. If it is not sent, the gateway is indicating that its preference is to not switch out of VBD mode after text spurt.

Redundancy of audio media as defined in [IETF RFC 2198] may be used to provide better reliability in networks where packet loss exists. The example below shows how SDP may be used to signal an offer to send G.729 audio interleaved with text protected with two levels of redundancy:

```
m=audio 7200 RTP/AVP 18 98 100
a=rtpmap:98 t140c/8000
a=fmtp:98 cps=20
a=gpmd:98 tpmods=baudot,edt
a=gpmd:98 remain-in-vbd=no
a=rtpmap:100 red/8000
a=fmtp:100 98/98/98
```

Note that the gateway only indicated support for Baudot PSTN and EDT PTP devices in the above example. The assumption is that the gateway will either use VBD (not shown in this example) for other PSTN PTP types or will use relay in order to provide interworking between dissimilar PSTN PTP types.

Annex D

Interworking IP text devices with V.151 gateways

(This annex forms an integral part of this Recommendation)

D.1 Introduction

Some non-gateway IP devices in the network that do not have means of modulating or demodulating PSTN textphone signals, such as IVR systems, voicemail systems, IP phones, or other devices, may support the transmission of real-time text over IP networks and, ideally, interwork with V.151-compliant gateways in order to provide a means through which users of PSTN textphone devices can communicate with those IP devices. This class of devices is referred to herein as an "IP text device" (ITD). This annex defines the procedures that may be used in order to allow for such interworking between V.151 gateways and ITDs.

D.2 Exchanging capabilities and opening media streams

When establishing a call, an ITD shall advertise support for ToIP in accordance with Annex B or Annex C, with one notable important exception: the ITD shall not include a list of supported modulations. Absence of the list of modulations shall be used as an indicator to the V.151 gateway that the remote side is an ITD.

Further, ITD devices shall not utilize SSEs in order to control state transition to ToIP. Rather, once a media flow is established to transport T.140 characters, either the gateway or the ITD may transmit text immediately and without further negotiation. Payload type switching is used to transition between audio and text relay modes.

In general, ITDs transmit text characters when communicating with other ITDs using [b-IETF RFC 4103], which specifies the establishment of a separate RTP stream specifically for transmitting text characters. However, since PSTN gateway devices interleave audio and text data, ITDs need to support both [b-IETF RFC 4103] to interwork with other ITDs and V.151 for communicating with PSTN gateways. As such, ITDs shall indicate support for both transport methods when exchanging capability information.

Further, when H.323-based ITDs utilize the Fast Connect procedures to offer media streams, those systems should offer proposals for both [b-IETF RFC 4103] and V.151 (refer to Annexes C and G of [ITU-T H.323]). Those proposals would be separate proposals within the fastStart SEQUENCE transmitted to the receiver. As an abbreviated example, an H.323 device might send the following three forward logical channel proposals in fastStart (the proposed media format(s) within a single OLC are shown within braces):

```
{ G.711, V.151 }, { G.711, RFC 4103 }
```

If the called device is an ITD, it may accept the G.711 proposal to establish an audio stream and the RFC 4103 for text. The fastStart response would contain:

```
{ G.711 }, { RFC 4103 }
```

If the called device is a gateway, it would accept the proposal for G.711 and V.151/Annex E data interleaved within the same media stream. The fastStart response would contain:

```
{ G.711, V.151 }
```

The foregoing does not preclude the use of H.245 logical channel signalling to open compatible media streams and merely serves as an illustration.

Similar logic follows for SDP-based systems. Consider the following abbreviated example SDP that might be found in an offer sent by an ITD:

```
m=audio 7200 RTP/AVP 0 98
a=rtpmap:98 t140c/8000
a=fmtp:98 cps=20
m=text 7202 RTP/AVP 99
a=rtpmap:99 t140/1000
a=fmtp:99 cps=20
```

If the answering device is another ITD, it would accept the RFC 4103 stream and remove the V.151 (t140c) proposal, as shown in this response:

```
m=audio 7200 RTP/AVP 0
m=text 7202 RTP/AVP 99
a=rtpmap:99 t140/1000
a=fmtp:99 cps=20
```

If the answering device is a gateway, it would accept the V.151 stream and indicate that does not wish to utilize RFC 4103 by setting the port to zero (0), as illustrated below:

```
m=audio 7200 RTP/AVP 0 98
a=rtpmap:98 t140c/8000
a=fmtp:98 cps=20
m=text 0 RTP/AVP 99
```

Note that while the use of RFC 2198 redundancy or other fault tolerance scheme is not shown in these abbreviated examples, appropriate mechanisms should be employed to protect the transmission of the text stream in accordance with this Recommendation.

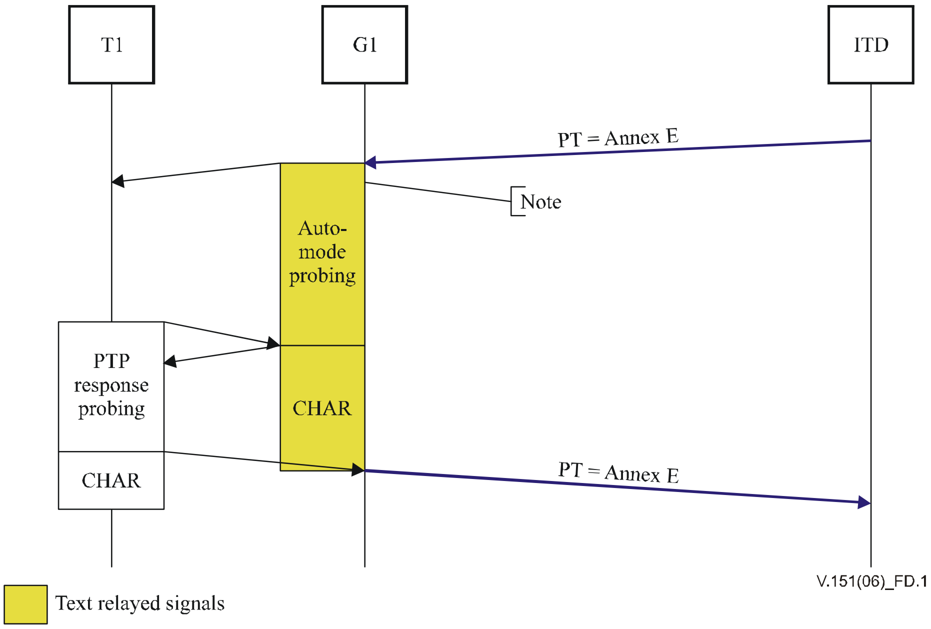
D.3 State transition and text handling

The following description assumes that a V.151-compliant gateway is in communication with an ITD and media flows have been established to transport text over IP in accordance with Annex E.

When the gateway compliant to this annex detects a PSTN textphone that it supports using text relay, it shall autonomously connect to the textphone. After connection, the gateway shall decode the characters received and transmit those to the ITD, giving proper respect to the maximum character-per-second (CPS) parameter advertised by the ITD. Determining the type of PSTN textphone device in use is the responsibility of the gateway and the ITD need not concern itself with what kind of PSTN textphone device is connected to the gateway.

Likewise, when the ITD sends characters to the gateway using TR payload type, the gateway shall perform a probing, as necessary, to determine the type of PSTN device connected, if any. While probing, the gateway shall buffer any characters received and transmit those when probing completes and the gateway is connected to the text phone. The size of the character buffer is a matter of implementation, but should support the reception and buffering of characters for at least 60 seconds at the specified maximum character-per-second (CPS) value signalled to the ITD.

If the gateway does not support the modulation used by the PSTN textphone device, the gateway may transmit the received textphone signals via VBD or the audio stream, depending on the capabilities of the ITD. Further, the gateway may simply discard characters received from the ITD or transmit them to the PSTN textphone device using a pre-provisioned modulation.



NOTE – Automode probing as defined in Figure 2b of [ITU-T V.18]. The character(s) sent in the Annex E payload by ITD should be used in the auto-mode probing sequence. Further characters received from ITD shall be buffered by G1 until connection is established with T1. Upon connection with T1, G1 shall transmit all buffered characters.

Figure D.1 – Example text relay call flow for ITD/gateway text relay interoperability

Annex E

Payload format and signaling syntax for real-time text transported within an audio stream

(This annex forms an integral part of this Recommendation)

E.1 Overview

This annex defines the payload format specification for transporting real-time text over IP networks within an audio stream. The payload format for these packets utilizes ITU-T Rec. T.140 for the character encoding. Characters are transmitting through an audio stream, either switched with voice packets or in parallel to voice packets.

E.2 Payload format

When transporting real-time text within an audio steam, text packets are differentiated from audio packets, SSE messages, DTMF signals, or other packets by the payload type value negotiated during session establishment or when the session is later modified. Text packets shall be assigned a payload type value that allows the endpoint to unambiguously recognize text packets from other packets transmitted within the audio stream.

The real-time text packets shall utilize T.140 for character encoding, transported using the real-time transport protocol (RTP), and shall be encoded as shown in Figure E.1.

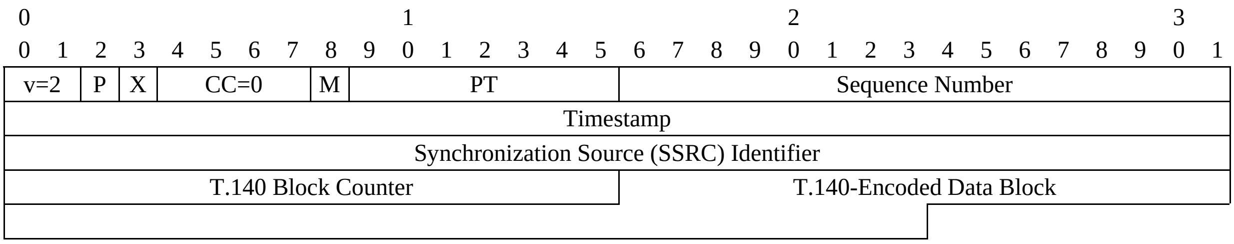


Figure E.1 – Payload format

The definition of the first 3 octets is found in Section 5.1 of IETF RFC 3550. Implementations shall adhere to the following usage of these fields:

This M-bit shall be set to '1' for the initial packet transmitted for any given SSRC value presented on the wire and shall be set to '1' following a period of 'silence' where no RTP packets are transmitted. In all other cases where packets are transmitted successively, this bit shall be set to '0'. A period of 'silence' is defined to be more than 300 ms without transmitting a packet.

The payload type (PT) field shall contain a dynamic payload type value negotiated by the two endpoints.

The sequence number and timestamp fields shall increase in accordance with IETF RFC 3550.

The SSRC used for text should be the same SSRC value as used for other audio, allowing for synchronized interleaving of audio and text. This means that the clock rate used for audio transmission should be the same rate as used for text packets.

There may be cases, however, where it is desired to transmit audio and text within the same stream in parallel and having overlapping timestamps. For example, if a user speaks while typing at the same time using devices that enable simultaneous voice and text, both audio and text may be presented to the network. In such cases, a separate SSRC may be used in order to separate the voice source from the text source and allow for overlapping timestamps and different sequence number spaces. Such usage of multiple payloads with an RTP stream is discussed in IETF RFC 3550.

The actual payload of the RTP packet is comprised of a T.140 block counter and a chunk of T.140-encoded data. Those two components of the payload are described in the following clauses.

E.2.1 T.140 block counter

The T.140 block counter is similar in purpose to the sequence number and is necessary since text packets and audio packets may share the same sequence number space. Without this counter, it would not otherwise be possible to detect lost text packets.

The T.140 block counter shall be initialized to zero for the first text packet transmitted and, once passing the value 0xFFFF, shall be reset to zero.

Devices that receive a text packet containing a T.140 block counter that has incremented higher than expected shall assume that the difference between the recently received counter value and the expected counter value indicates the number of lost text packets. Note, however, that there may multiple characters within a text packet, so this does not serve to indicate the number of lost characters.

E.2.2 T.140-Encoded Data Block

The T.140-encoded data block ("T.140 block") contains text information as described in ITU-T Rec. T.140. The contents of this field shall be UTF-8 encoded and shall include no extra framing.

It should be noted that, in most cases, this field is comprised of one or more text characters. However, it is permissible to have zero characters for the purpose of enabling transmission of redundant data packets as discussed in clause E.3. It should also be noted that, while most elements within this field constitute single characters, some elements are multiple-character sequences. Any composite character sequence (CSS) elements should be placed in a single RTP packet.

E.3 Use of redundancy

Use of redundancy, such as the redundancy mechanism described in [IETF RFC 2198], provides a high degree of resiliency in the face of packet loss. When transmitting a constant stream of packets, [IETF RFC 2198] is quite clear on the contents of the data packets, including the primary and redundant parts of the RFC 2198 packet.

However, in cases where there is "silence" for a period of time, [IETF RFC 2198] is not explicitly clear about how a device should deliver redundant data. More specifically, [IETF RFC 2198] requires that all packets contain a primary encoding, though during a period of "silence" there would not be information available to transmit as primary data. For this reason, when transmitting text as redundant information and without any new text to transmit, devices shall transmit RTP packets that contain no "primary" data: both the "T.140 block counter" and "T.140 block" shall be absent from the RTP payload. This absent primary T.140 data is referred to as an "empty T.140 block".

Figure C.2 shows what an RFC 2198 redundancy packet might look like when the primary encoding is absent (i.e., there is an empty T.140 block) and a single redundant encoding of a previously transmitted T.140 block ("R") is present. It is important to note that the sequence number for the RFC 2198 packet continues to increase, as specified in [IETF RFC 2198] and IETF RFC 3550, so that it is possible for the system and network to accurately measure overall packet loss. However, character loss is detected through missing T.140 block counters, not gaps in the RTP packet sequence numbers. In this way, if a packet is lost, then the character may be recovered through these redundant transmissions without any apparent lost observed by the user.

A device should not re-transmit an empty T.140 block as a redundant encoding, as doing so merely consumes bandwidth unnecessarily and does not improve the robustness of the system.

<table>
<tr><td colspan="10">0</td><td colspan="10">1</td><td colspan="10">2</td><td colspan="2">3</td></tr>
<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>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>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>0</td><td>1</td></tr>
<tr><td colspan="2">v=2</td><td>P</td><td>X</td><td colspan="4">CC=0</td><td>M</td><td colspan="7">"RED" PT</td><td colspan="16">Sequence Number of Primary</td></tr>
<tr><td colspan="32">Timestamp of Primary Encoding</td></tr>
<tr><td colspan="32">Synchronization Source (SSRC) Identifier</td></tr>
<tr><td>1</td><td colspan="7">"t140c" PT</td><td colspan="14">Timestamp offset of "R"</td><td colspan="10">"R" Block Length</td></tr>
<tr><td>0</td><td colspan="7">"t140c" PT</td><td colspan="16">T.140 Block Counter for "R"</td><td colspan="8"></td></tr>
<tr><td colspan="32">T.140 Block "R"</td></tr>
</table>

Figure E.2 – RFC 2198 redundancy packet with empty T.140 block

Appendix I

Background to PSTN text telephony

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

Text telephone systems were implemented mainly for distant conversation with deaf, hard-of-hearing, speech-impaired and deaf-blind users. The text telephone systems offer a real-time, character-by-character, conversation in text, optionally combinable with voice. The text telephone service is described generally in [ITU-T F.700] and [ITU-T F.703] and user needs are described in [b-ETSI ETR 333], *Human Factors, Text Telephony; Basic user requirements and recommendations*.

With the PSTN, there are seven different, openly specified signalling methods used in text telephony. These signalling methods are country-specific. The signalling methods are TIA-825A (Baudot), DTMF, EDT, V.21, Bell103, Minitel and V.18. Each uses a different modulation and character encoding in the transmission of text. The methods are described in the annexes of [ITU-T V.18].

[ITU-T V.18] is an auto-mode mechanism that attempts to enable communication with all the legacy mode of operation at the physical layer level. If both end-point text telephones are V.18, they encode their data using the character set, defined in ITU-T Rec. T.140. [ITU-T V.21] is the common or native modulation for V.18, while ITU-T Rec. V.61 is specified for use of simultaneous voice and text.

There also exist several proprietary flavours of the existing standardized modulations, which are not part of [ITU-T V.18].

Appendix II

ToIP call discrimination call flows

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

II.1 Scope

This appendix contains a set of example call flow diagrams for this Recommendation. 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 ToIP call flows. In the diagrams:

- the white vertical rectangles under the ToIP endpoints (G1 and G2) give the state of the respective endpoint;
- the shaded vertical rectangles under the text telephony terminals (T1 and T2) and the ToIP endpoints (G1 and G2) give the signals that are being transmitted by the respective Terminal or endpoint;
- while in audio and VBD mode, the ToIP 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 Scenarios for call discrimination flows

Table II.1 lists all possible scenarios of PTP devices and gateway types. Each of these scenarios is considered in the following clauses.

Table II.1 – Possible PSTN text phone connection scenarios

T1 Type (calling)	T2 Type (called)	$T1 \cap G1 \neq 0$	$T2 \cap G2 \neq 0$	Connect mode
V.18	V.18	Y	Y	TR
V.18	V.18	N	Y	SSEs: TR Else: VBD
V.18	V.18	Y	N	SSEs: TR Else: VBD
V.18	V.18	N	N	Same as scenario 3
FDX	V.18	Y	Y	TR unless T1 is Bell 103 and G2 does not support Bell 103
FDX	V.18	N	Y	VBD
FDX	V.18	Y	N	Same as scenario 5
FDX	V.18	N	N	VBD
V.18	FDX	Y	Y	TR unless no SSEs and G1 and G2 do not both support T2's mode
V.18	FDX	N	Y	Same as scenario 9

Table II.1 – Possible PSTN text phone connection scenarios

T1 Type (calling)	T2 Type (called)	T1 ∩ G1 ≠ 0	T2 ∩ G2 ≠ 0	Connect mode
V.18	FDX	Y	N	VBD
V.18	FDX	N	N	VBD
FDX	FDX	Y	Y	TR VBD if G1 does not support T2 and T1 = T2.
FDX	FDX	N	Y	VBD No connect if T1 != T2
FDX	FDX	Y	N	Same as scenario 14
FDX	FDX	N	N	Same as scenario 14
HDX	V.18	Y	Y	TR
HDX	V.18	N	Y	VBD
HDX	V.18	Y	N	VBD
HDX	V.18	N	N	VBD
V.18	HDX	Y	Y	TR
V.18	HDX	N	Y	VBD
V.18	HDX	Y	N	VBD
V.18	HDX	N	N	VBD
HDX	HDX	Y	Y	TR
HDX	HDX	N	Y	VBD No connect if T1 != T2
HDX	HDX	Y	N	Same as scenario 26
HDX	HDX	N	N	Same as scenario 26
FDX	HDX	Y	Y	TR
FDX	HDX	N	Y	No connect
FDX	HDX	Y	N	No connect
FDX	HDX	N	N	No connect
HDX	FDX	Y	Y	TR
HDX	FDX	N	Y	No connect
HDX	FDX	Y	N	No connect
HDX	FDX	N	N	No connect

II.3 Scenarios with SSE protocol being used

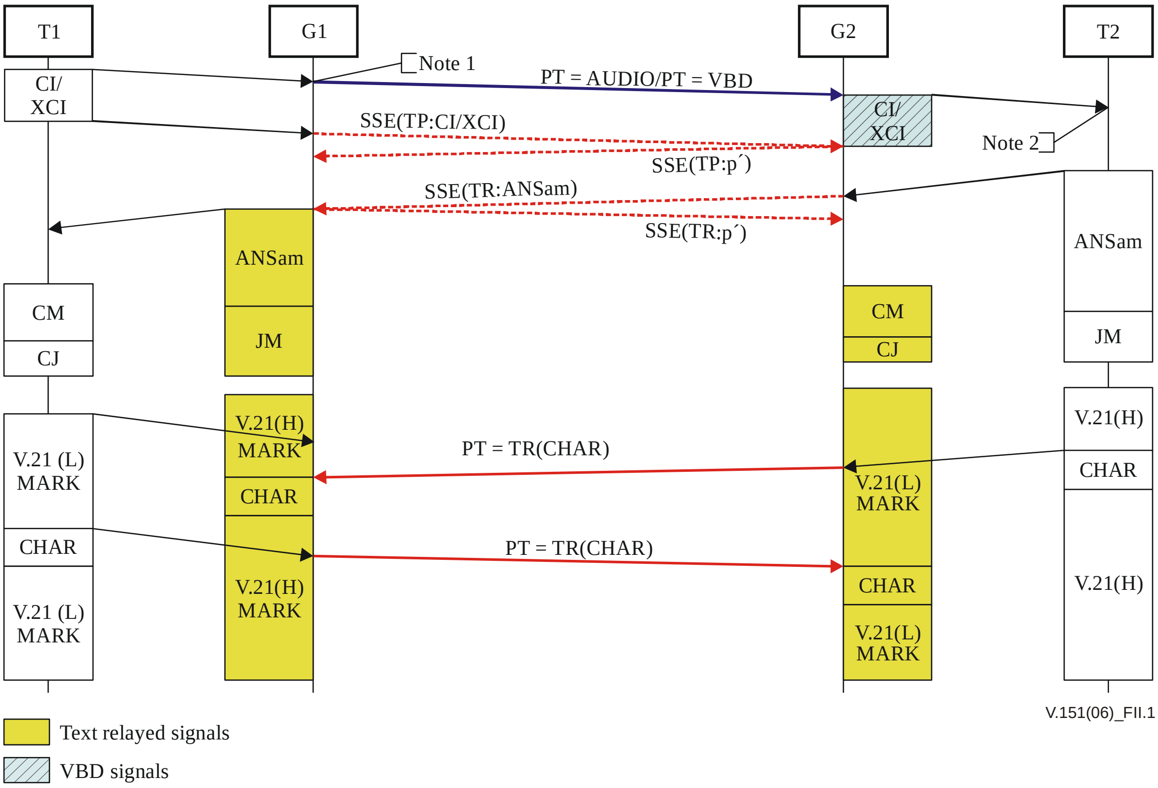
II.3.1 Scenario #1

T1 = V.18 PTP

T2 = V.18 PTP

G1 = supports V.18 native mode

G2 = supports V.18 native mode



NOTE 1 – VBD may optionally be used for CI/XCI signal. If VBD is used, there is an SSE(VBD:CI/XCI) handshake here.
NOTE 2 – CI/XCI may be detected by T2. If not detected, there will be up to 3 seconds of additional delay before ANSam is generated.

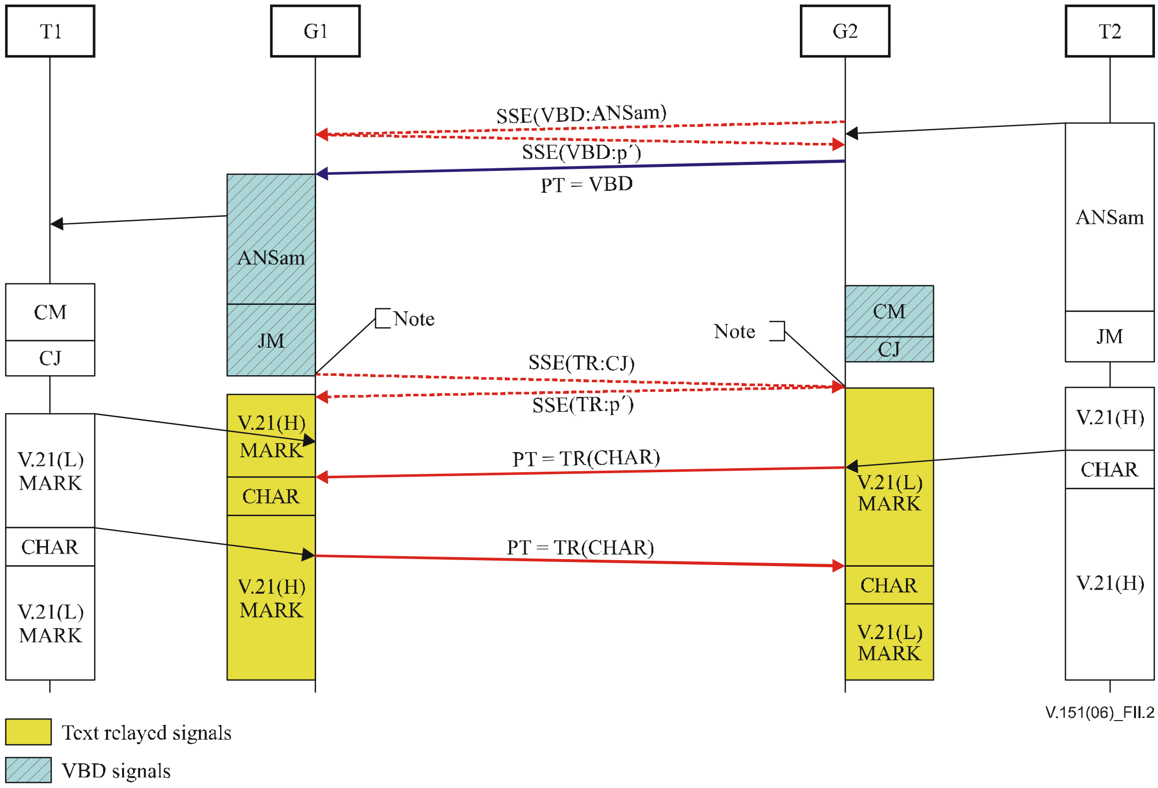
Figure II.1 – Scenario #1 call flow diagram (SSE) (sheet 1 of 2)

Description

Since T1 is the V.18 calling PTP, it will generate a CI/XCI sequence. Since G1 is a V.151-compliant gateway that supports SSE protocol, it will detect CI/XCI indicating to G1 that it is connected to a V.18 PTP terminal. Upon detection of CI/XCI, G1 may continue to transmit the CI/XCI sequence using audio mode to G2, or it may optionally initiate a transition to VBD mode by sending an SSE(VBD:CI/XCI) to G2. The diagram does not show the optional VBD SSE negotiation at this point. After CI/XCI sequence is completely transmitted to G2 (either through audio or VBD encoding), G1 shall send an SSE(TP:CI/XCI) to G2 and terminate regeneration of signals to T1. G1 shall stay in the mode of silence generation to T1 until it exits the TP state via an SSE from G2.

Upon reception of the SSE(TP:CI/XCI) from G1, G2 will enter an answering auto-mode probing state. The answering auto-mode probing state would consist of G2 listening for PTP signals from T2. Since T2 is a V.18 PTP answering terminal, it will generate ANSam signal either immediately after the CI/XCI sequence if it is detector, or within 3 seconds if the CI/XCI is not detected. Upon detection of the ANSam signal from T2 by G2, G2 will send an SSE(TR:V.18) indicating that it has detected ANSam signal from T2 and that T2 is a V.18 modem and that G1 should start its text relay connect sequence. G2 will then start the V.8 connect sequence with T2 (since G2 supports native V.18).

Upon reception of the SSE(TR:V.18) from G2, G1 will start a V.8 connect sequence with T1 and successfully connect with it. At this time, G1 and G2 are free to transmit receive characters via RTP. Any characters that are received by the gateway from the remote gateway before the gateway has completed the start-up (or probing) sequence with its local PTP will be buffered and sent after establishing connection with the local PTP.



NOTE – As per V.8 specification, the silence period between CJ/JM signal and the generation of V.21 carrier must be 75 ± 5 ms.

Figure II.2 – Scenario #1 call flow diagram (SSE) (sheet 2 of 2)

Description

This call flow takes into consideration the case were CI/XCI is not detected by G1, possibly because the ANSam tone was generated first due to the manner in which the call was established. G2, upon detecting the ANSam, will initiate a transition to VBD mode. While in VBD mode, both gateways will monitor the V.8 signals at their PCM interfaces, G1 seeing the CM signal indicating a PTP device and G2 seeing the JM signal indicating a PTP device. At the end of CJ reception, G1 will initiate a transition to text relay mode through the generation of an SSE(TR) to G2. The transition from VBD to text relay mode needs to be handled so that the V.21 carrier generated by the gateways meets the 75 ± 5 ms as required by [ITU-T V.8].

For the case where the end of CI/XCI corresponds to the beginning of ANSam such that there is a cross-over of TP and VBD SSEs in the IP network, per the SSE protocol definition the gateways will end up in VBD mode. As this diagram demonstrates, in this case the gateway will eventually go to text relay mode at the end of V.8.

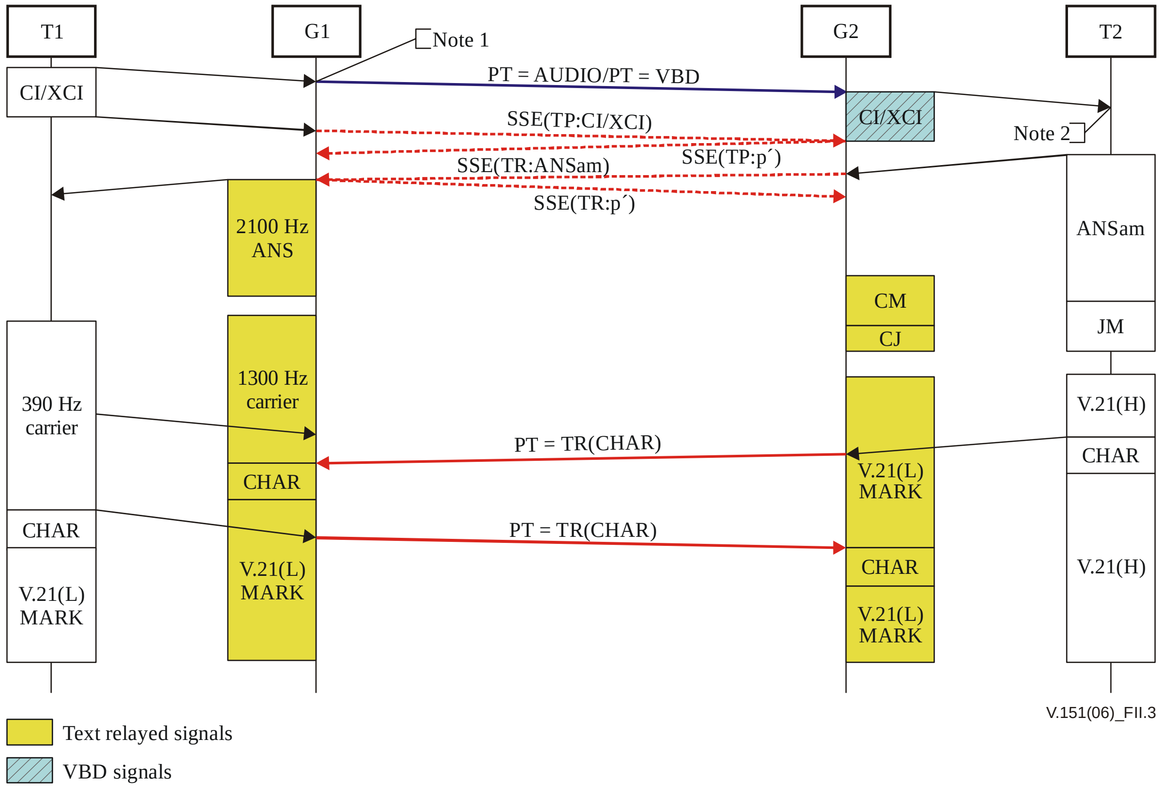
II.3.2 Scenario #2

T1 = V.18 PTP

T2 = V.18 PTP

G1 = supports FDX modulation (V.23 in this example)

G2 = support V.18 native mode



NOTE 1 – VBD may optionally be used for CI/XCI signal. If VBD is used, there is an SSE(VBD:CI/XCI) handshake here.
NOTE 2 – CI/XCI may be detected by T2. If not detected, there will be up to 3 seconds of additional delay before ANSam is generated.

Figure II.3 – Scenario #2 call flow diagram (SSE)

Description

Since T1 is the V.18 calling PTP, it will generate a CI/XCI sequence. Since G1 is a V.151-compliant gateway that supports SSE protocol, it will detect CI/XCI indicating to G1 that it is connected to a V.18 PTP terminal. Upon detection of CI/XCI, G1 may continue to transmit the CI/XCI sequence using audio mode to G2, or it may optionally initiate a transition to VBD mode by sending an SSE(VBD:CI/XCI) to G2. The diagram does not show the optional VBD SSE negotiation at this point. After CI/XCI sequence is completely transmitted to G2 (either through audio or VBD encoding), G1 shall send an SSE(TP:CI/XCI) to G2 and terminate regeneration of signals to T1. G1 shall stay in the mode of silence generation to T1 until it receives an SSE response from G2.

Upon reception of the SSE(TP:CI/XCI) from G1, G2 will enter an answering auto-mode probing state. Since T2 is a V.18 PTP answering terminal, it will generate ANSam signal either immediately after the CI/XCI sequence if it is detector, or within 3 seconds if the CI/XCI is not detected. Upon detection of the ANSam signal from T2 by G2, G2 will send an SSE(TR:V.18) indicating that it has detected ANSam signal from T2 and that T2 is a V.18 modem and that G1 should start its text relay connect sequence. G2 will then start the V.8 connect sequence with T2 (since G2 supports native V.18).

Upon reception of the SSE(TR:V.18) from G2, G1 will start a connect sequence with T1. Since it knows that T1 is a V.18 PTP terminal, G1 will successfully be able to connect with it using the FDX modulation scheme that G1 supports. G1 starts the answering sequence for its supported modulation. At this time, G1 and G2 are free to transmit receive characters via RTP. Any characters that are received by the gateway from the remote gateway before the gateway has completed the start-up (or probing) sequence with its local PTP will be buffered and sent after establishing connection with the local PTP.

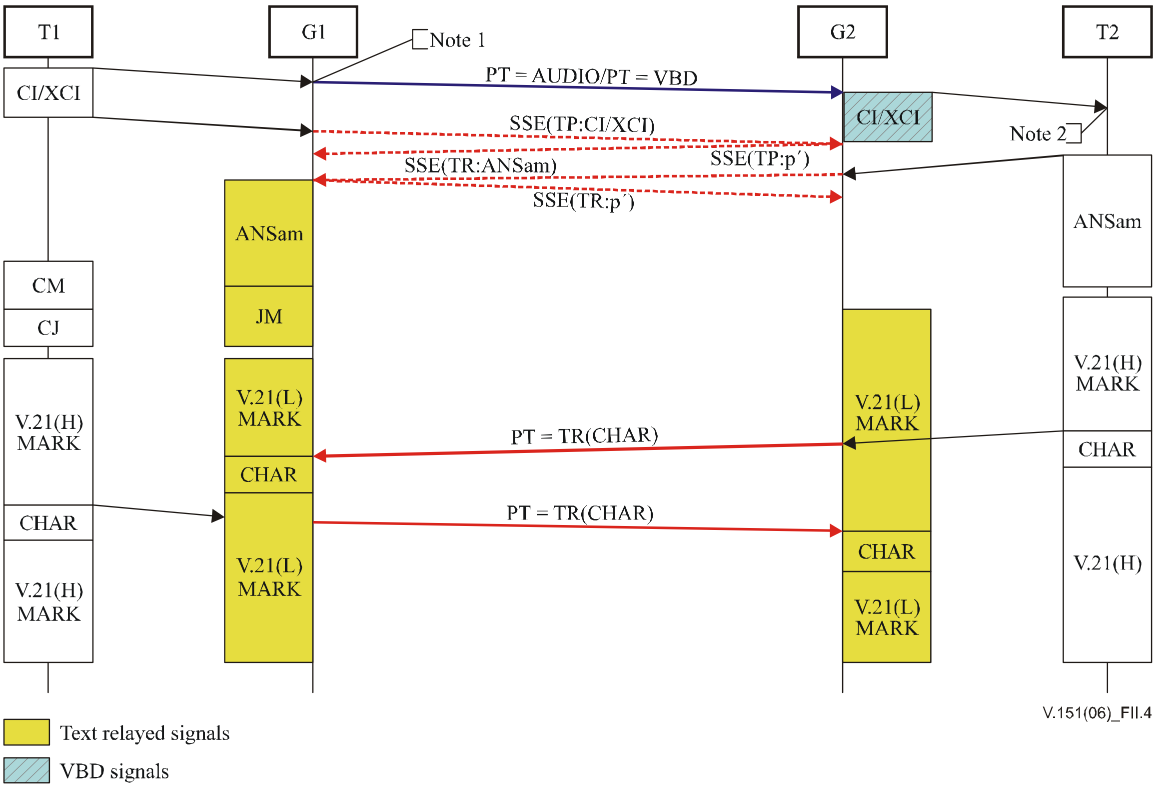
II.3.3 Scenario #3

T1 = V.18 PTP

T2 = V.18 PTP

G1 = support V.18 native mode

G2 = supports FDX mode (V.21 in this example)



NOTE 1 – VBD may optionally be used for CI/XCI signal. If VBD is used, there is an SSE(VBD:CI/XCI) handshake here.
NOTE 2 – CI/XCI may be detected by T2. If not detected, there will be up to 3 seconds of additional delay before ANSam is generated.

Figure II.4 – Scenario #3 call flow diagram (SSE)

Description

Since T1 is the V.18 calling PTP, it will generate a CI/XCI sequence. Since G1 is a V.151-compliant gateway that supports SSE protocol, it will detect CI/XCI indicating to G1 that it is connected to a V.18 PTP terminal. Upon detection of CI/XCI, G1 may continue to transmit the CI/XCI sequence using audio mode to G2, or it may optionally initiate a transition to VBD mode by sending an SSE(VBD:CI/XCI) to G2. The diagram does not show the optional VBD SSE negotiation at this point. After CI/XCI sequence is completely transmitted to G2 (either through audio or VBD encoding), G1 shall send an SSE(TP:CI/XCI) to G2 and terminate regeneration of signals to T1. G1 shall stay in the mode of silence generation to T1 until it receives an SSE indicating a new mode from G2.

Upon reception of the SSE(TP:CI/XCI) from G1, G2 will enter an answering auto-mode probing state. Since T2 is a V.18 PTP answering terminal, it will generate ANSam signal either immediately after the CI/XCI sequence if it is detector, or within 3 seconds if the CI/XCI is not detected. Upon detection of the ANSam signal from T2 by G2, G2 will send an SSE(TR:V.18) indicating that it has detected ANSam signal from T2 and that T2 is a V.18 modem and that G1 should start its text relay connect sequence. G2 will then continue with its answering auto-mode probing procedures, waiting for V.21 mark to be generated by T2 to respond with V.21 mark.

Upon reception of the SSE(TR:V.18) from G2, G1 will reply with an SSE(TR:p') completing the switch to text relay mode and it will start a V.18 answering modem sequence to connect with T1. Any characters that are received by the gateway from the remote gateway before the gateway has completed the start-up (or probing) sequence with its local PTP will be buffered and sent after establishing connection with the local PTP.

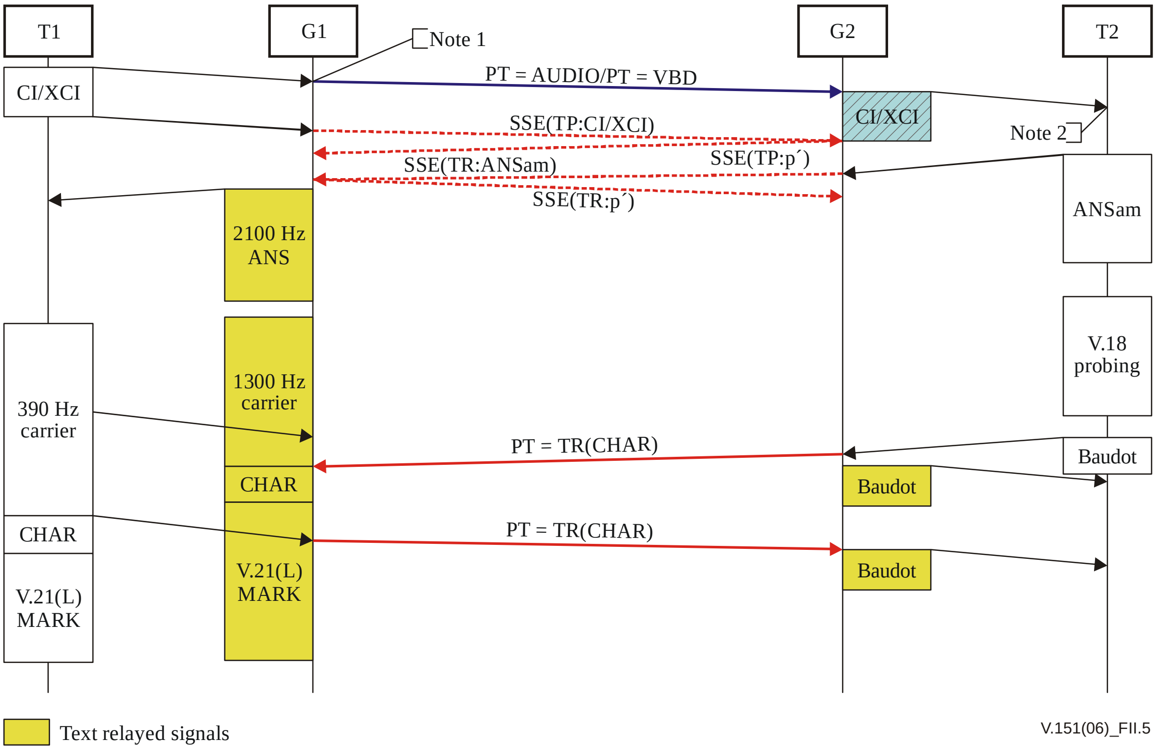
II.3.4 Scenario #4

T1 = V.18 PTP

T2 = V.18 PTP

G1 = supports FDX modulation (V.23 in this example)

G2 = support HDX modulation (Baudot in this example)



NOTE 1 – VBD may optionally be used for CI/XCI signal. If VBD is used, there is an SSE(VBD:CI/XCI) handshake here.
NOTE 2 – CI/XCI may be detected by T2. If not detected, there will be up to 3 seconds of additional delay before ANSam is generated.

Figure II.5 – Scenario #4 call flow diagram (SSE)

Description

Since T1 is the V.18 calling PTP, it will generate a CI/XCI sequence. Since G1 is a V.151-compliant gateway that supports SSE protocol, it will detect CI/XCI indicating to G1 that it is connected to a V.18 PTP terminal. Upon detection of CI/XCI, G1 may continue to transmit the CI/XCI sequence using audio mode to G2, or it may optionally initiate a transition to VBD mode by sending an SSE(VBD:CI/XCI) to G2. The diagram does not show the optional VBD SSE negotiation at this point. After CI/XCI sequence is completely transmitted to G2 (either through audio or VBD encoding), G1 shall send an SSE(TP:CI/XCI) to G2 and terminate regeneration of signals to T1. G1 shall stay in the mode of silence generation to T1 until it receives an SSE response from G2 indicating a new mode (either VBD or text relay).

Upon reception of the SSE(TP:CI/XCI) from G1, G2 will enter an answering auto-mode probing state. Since T2 is a V.18 PTP answering terminal, it will generate ANSam signal either immediately after the CI/XCI sequence if it is detected, or within 3 seconds if the CI/XCI is not detected. Upon detection of the ANSam signal from T2 by G2, G2 will send an SSE(TR:V.18) indicating that it has detected ANSam signal from T2 and that T2 is a V.18 modem and that G1 should start its text relay auto-mode probing sequence. Since G2 does not support native V.18 mode, it will not respond to the ANSam generated by T2, but will wait until it detects a PTP modulation that it supports (Baudot in this example). After detecting the Baudot probing signal from T2, G2 will immediately respond with Baudot, triggering T2 to go into Baudot mode. The Baudot probing character that was detected by G2 is sent to G1 using text relay mode.

Upon reception of the SSE(TR:V.18) from G2, G1 will start a connect sequence with T1. Since it knows that T1 is a V.18 PTP terminal, G1 will successfully be able to connect with it using the FDX modulation scheme that G1 supports. At this time, G1 and G2 are free to transmit receive characters via RTP. Any characters that are received by the gateway from the remote gateway before the gateway has completed the start-up (or probing) sequence with its local PTP will be buffered and sent after establishing connection with the local PTP.

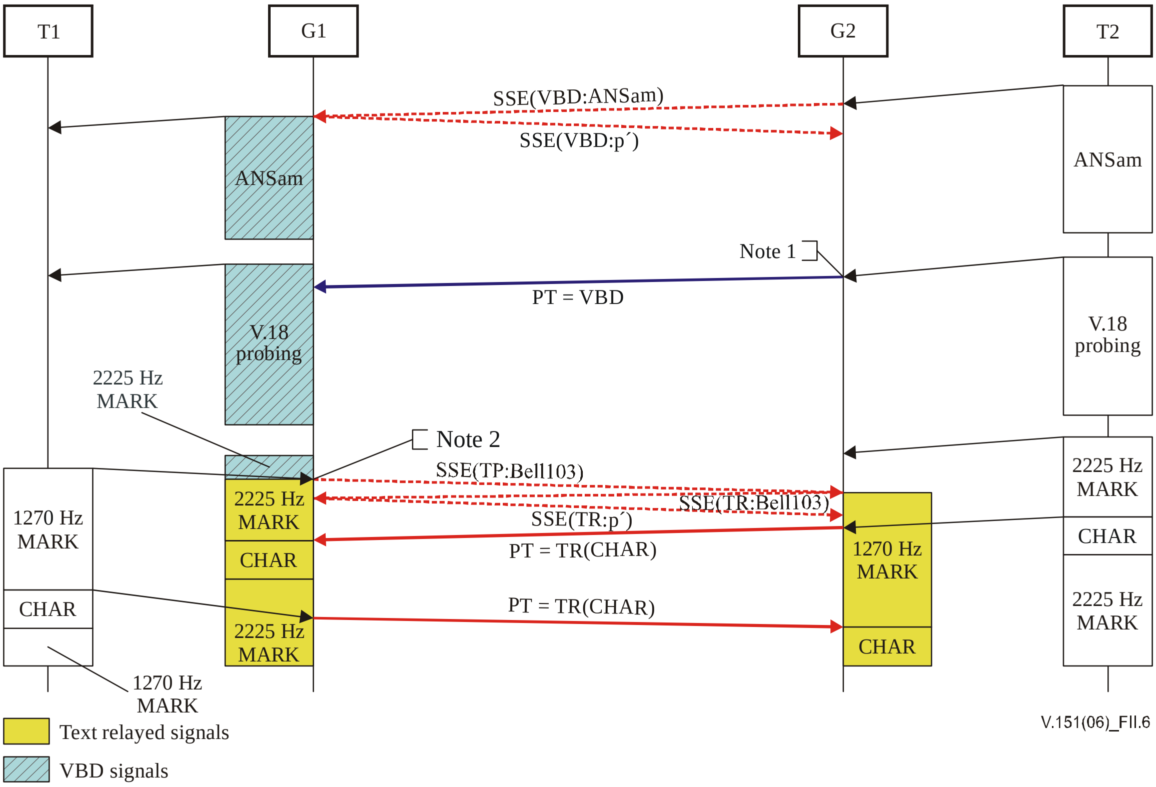
II.3.5 Scenario #5

T1 = FDX modulation PTP (Bell 103 in this example)

T2 = V.18 PTP

G1 = supports FDX modulation (Bell 103 in this example)

G2 = supports FDX modulation (Bell 103 in this example)



NOTE 1 – Allow end-to-end V.18 probing until T1 responds. G2 does not respond to any probing sequences that it supports during this probing.

NOTE 2 – SSE(TR) is only sent by G1 at this point if G2 supports T1's modulation; otherwise, stay in VBD mode.

Figure II.6 – Scenario #5 call flow diagram (SSE)

Description

Since T2 is a V.18 answering PTP, it will generate ANSam after 3 seconds (because CI/XCI was not received). Since G2 is a V.151-compliant gateway that supports SSE protocol, it will detect ANSam and initiate a switch to VBD mode. T2 is allowed to probe T1 using VBD mode. G2 does not respond to any of the T2 probing sequences, even those that represent modulations that G2 supports since it has not been established that an end-to-end text relay connection can be successfully made between all gateways and PTP devices. Upon detection by G1 of a PTP FDX modulation probing response from T1 that it supports, G1 will send an SSE(TP:X) to G2 (X is the modulation that was detected, Bell 103 in this example) only if G2 also supports this modulation. If G2 does not support this modulation, the gateways stay in VBD mode and the PTP devices connect using VBD for the entire session. In the case where T1 and T2 are not PTP, but data modems, requiring that both PSTN legs of the call be of the same modulation type will result in successful connection for data modems as well as PTP devices using proper character treatments.

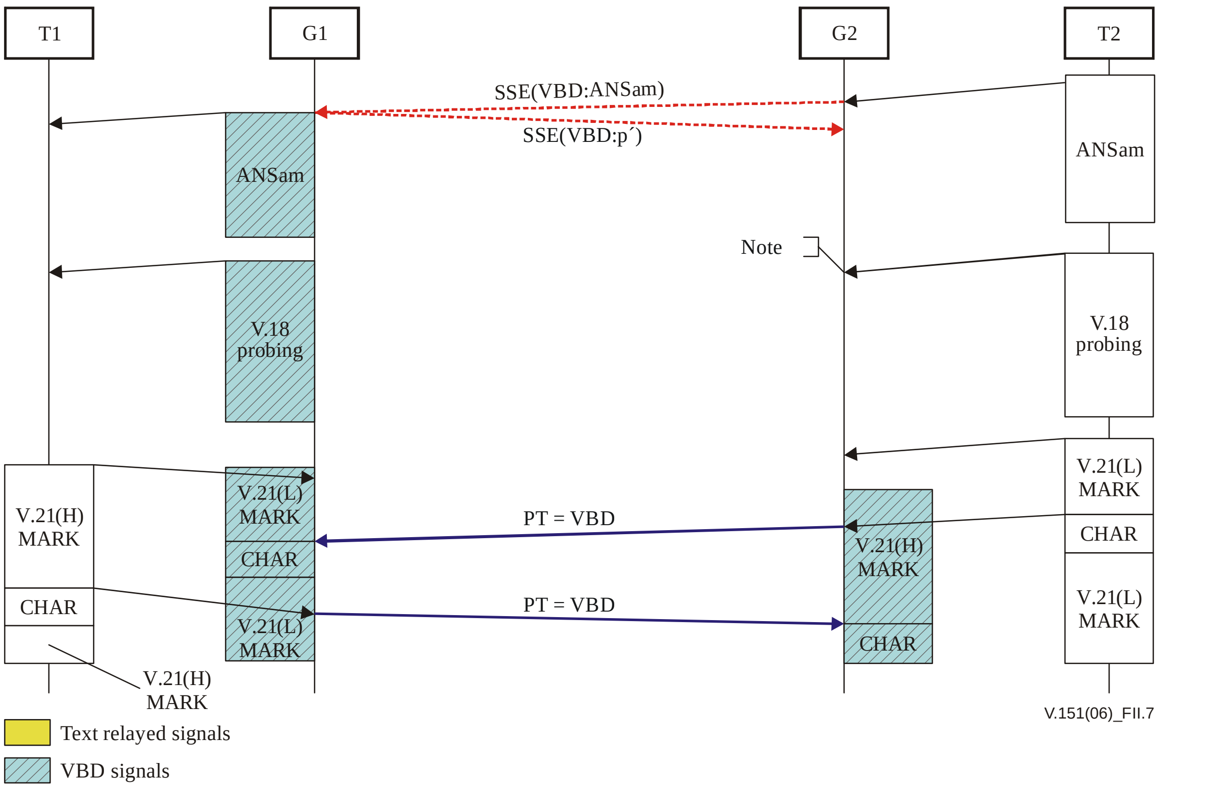
II.3.6 Scenario #6

T1 = FDX modulation PTP (V.21 in this example)

T2 = V.18 PTP

G1 = does not support modulation used by T1

G2 = don't care



NOTE – Allow end-to-end V.18 probing until T1 responds. G2 does not respond to any probing sequences that it supports during this probing.

Figure II.7 – Scenario #6 call flow diagram (SSE)

Description

Since T2 is a V.18 answering PTP, it will generate ANSam after 3 seconds (because CI/XCI was not received). Since G2 is a V.151-compliant gateway that supports SSE protocol, it will detect ANSam and initiate a switch to VBD mode. T2 is allowed to probe T1 using VBD mode. G2 does not respond to any of the T2 probing sequences, even those that represent modulations that G2 supports since it has not been established that a end-to-end text relay connection can be successfully made between all gateways and PTP devices. Since the modulation that is responded to by T1 is not supported by G1, G1 does not initiate a switch to text relay mode and the connection remains in VBD mode for the entire call.

II.3.7 Scenario #7

T1 = FDX modulation PTP (Bell 103 in this example)

T2 = V.18 PTP

G1 = supports FDX modulation (Bell 103 in this example)

G2 = supports FDX modulation (Bell 103 in this example) (Does not support native V.18)

Description

This scenario plays out identical to scenario #5. The only difference between these scenarios is that G2 does not support native V.18 mode, but the call flow is identical.

II.3.8 Scenario #8

T1 = FDX modulation PTP (V.21 in this example)

T2 = V.18 PTP

G1 = does not support modulation used by T1

G2 = does not support native V.18 mode

Description

This scenario plays out identical to scenario #6. The only difference between these scenarios is that G2 does not support native V.18 mode, but the call flow is identical.

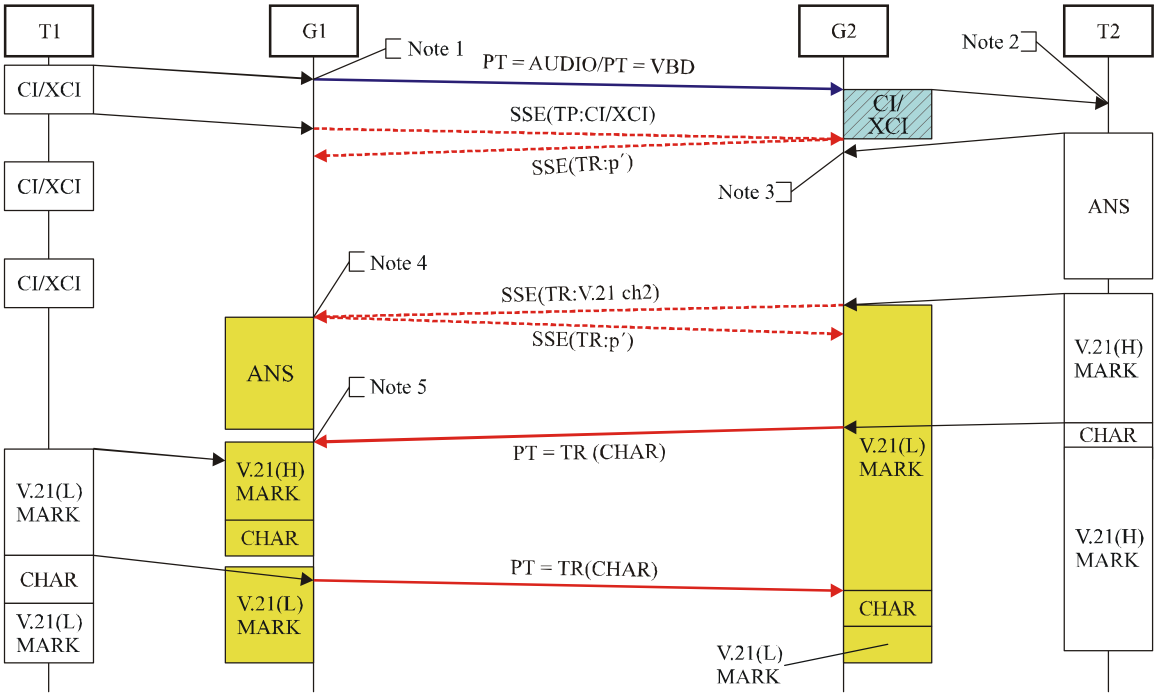
II.3.9 Scenarios #9 and #10

T1 = V.18 PTP

T2 = FDX modulation PTP (V.21 in this example)

G1 = supports any modulation

G2 = supports modulation of T2 (V.21 in this example)



Text relayed signals

VBD signals

NOTE 1 – VBD may optionally be used for CI/XCI signal. If VBD is used, there is an SSE(VBD:CI/XCI) handshake here.
NOTE 2 – CI/XCI may be detected by T2.
NOTE 3 – ANS tone is optional.
NOTE 4 – Start originate, mode answer probing upon reception of SSE(TR) with same modulation as supported by T2 if available at G1.
NOTE 5 – G1 buffers characters received from G2 until it has completed connection with T1.

Figure II.8 – Scenarios #9 and #10 call flow diagram (SSE)

Description

Since T1 is the V.18 calling PTP, it will generate a CI/XCI sequence. Since G1 is a V.151-compliant gateway that supports SSE protocol, it will detect CI/XCI indicating to G1 that it is connected to a V.18 PTP terminal. Upon detection of CI/XCI, G1 may continue to transmit the CI/XCI sequence using audio mode to G2, or it may optionally initiate a transition to VBD mode by sending an SSE(VBD:CI/XCI) to G2. The diagram does not show the optional VBD SSE negotiation at this point. After CI/XCI sequence is completely transmitted to G2 (either through audio or VBD encoding), G1 shall send an SSE(TP:CI/XCI) to G2 and terminate regeneration of signals to T1. G1 shall stay in the mode of silence generation to T1 until it receives an SSE response from G2 indicating a new mode (either VBD or text relay).

Upon reception of the SSE(TP:CI/XCI) from G1, G2 will enter an answering auto-mode probing state. Since T2 is a V.21 PTP answering terminal, it can either generate ANS or go straight to V.21 MARK generation (ANS signal is optional). Upon detection of the V.21 MARK from T2 by G2, G2 will send an SSE(TR:V.18) indicating that it has detected a valid PTP signal from T2 and that G1 should start its text relay auto-mode probing sequence. G2 will then connect with T2 and start to transmit receive characters using text relay mode.

Upon reception of the SSE(TR:V.18) from G2, G1 will start a connect sequence with T1. Since it knows that T1 is a V.18 PTP terminal, G1 will successfully be able to connect with it using the first modulation scheme used in the probing sequence. G1 should use the modulation scheme being used by T2 and G2 if it is supported by G1 (i.e., attempt to match modulations at both call legs). Any characters that are received by the gateway from the remote gateway before the gateway has completed the start-up (or probing) sequence with its local PTP will be buffered and sent after establishing connection with the local PTP.

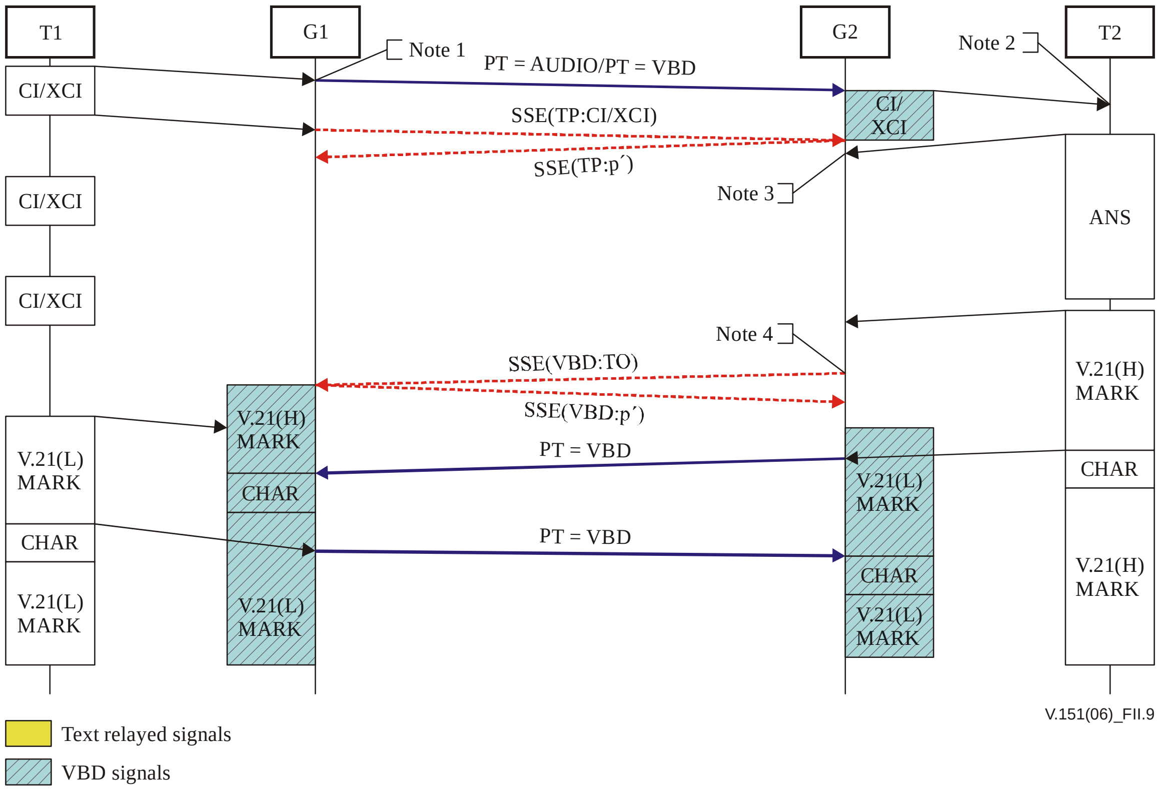
II.3.10 Scenarios #11 and #12

T1 = V.18 PTP

T2 = FDX modulation PTP (V.21 in this example)

G1 = supports any modulation

G2 = does not support modulation of T2



NOTE 1 – VBD may optionally be used for CI/XCI signal. If VBD is used, there is an SSE(VBD:CI/XCI) handshake here.
NOTE 2 – CI/XCI may be detected by T2.
NOTE 3 – ANS tone is optional.
NOTE 4 – G2 times out looking for valid PTP signal that it supports.

Figure II.9 – Scenarios #11 and #12 call flow diagram (SSE)

Description

Since T1 is the V.18 calling PTP, it will generate a CI/XCI sequence. Since G1 is a V.151-compliant gateway that supports SSE protocol, it will detect CI/XCI indicating to G1 that it is connected to a V.18 PTP terminal. Upon detection of CI/XCI, G1 may continue to transmit the CI/XCI sequence using audio mode to G2, or it may optionally initiate a transition to VBD mode by sending an SSE(VBD:CI/XCI) to G2. The diagram does not show the optional VBD SSE negotiation at this point. After CI/XCI sequence is completely transmitted to G2 (either through audio or VBD encoding), G1 shall send an SSE(TP:CI/XCI) to G2 and terminate regeneration of signals to T1. G1 shall stay in the mode of silence generation to T1 until it receives an SSE response from G2 indicating a new mode (either VBD or text relay).

Upon reception of the SSE(TP:CI/XCI) from G1, G2 enters into an answering auto-mode probing state, searching for a signal that it detects as a valid PTP signal. Since G2 does not support the modulation supported by T2, it will detect a non-text relay support signal in the answering auto-mode probing state and generate an SSE(VBD:TO) to G1. This will transition the call to VBD mode and the connection with stay in this mode for the remainder of the PTP session.

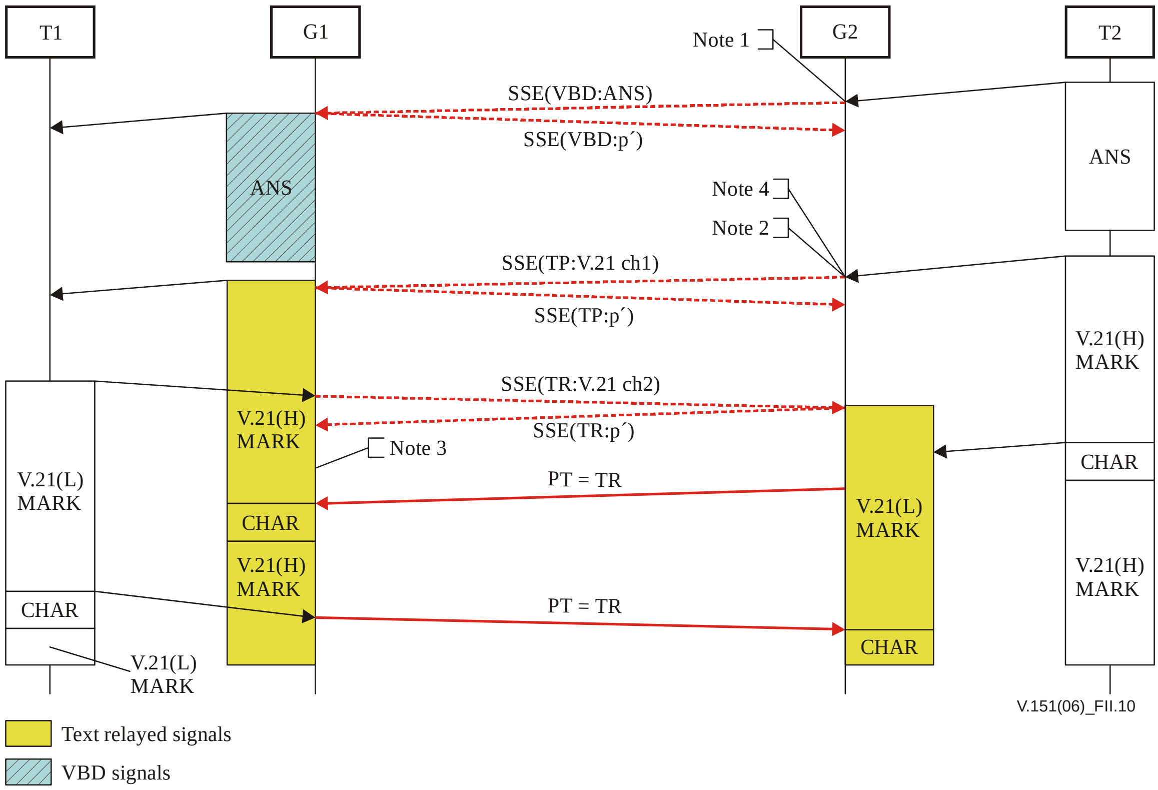
II.3.11 Scenarios #13 and #14

T1 = FDX modulation PTP (V.21 in this example)

T2 = FDX modulation PTP (same as T1)

G1 = supports modulation of T1

G2 = supports modulation of T2



NOTE 1 – Generation of ANS signal by T2 is optional.
NOTE 2 – If ANS was not generated by T2, G2 shall initiate transition to VBD upon V.21 mark detection.
NOTE 3 – G1 responds with SSE(TR) only if it detects T1 to be same modulation as T2. If not, G1 responds with SSE(VBD).
NOTE 4 – Only send SSE(TR) if G1 supports modulation of T2; otherwise, stay in VBD mode.

Figure II.10 – Scenarios #13 and #14 call flow diagram (SSE)

Description

T2 being the answering PTP device will initiate the generation of a FDX signal, optionally proceeded by the ANS signal. Since G2 is a V.151-compliant gateway that supports SSE protocol, upon detection of the ANS tone, G2 will initiate a switch to VBD mode. When G2 detects the carrier for a FDX modulation that it supports for text, it shall generate a SSE(TP) if G1 has also indicated support for this modulation. If G1 does not support the modulation used by T2, the connection will remain in VBD for the duration of the call.

Upon reception of the SSE(TP), G1 shall enter an originate mode auto-probing sequence, starting with the modulation indicated in the SSE(TP) from G2. Since T1 is the same modulation as T2, G1 will connect with T1 using this modulation. Upon detection of the response signal from T1, G1 will generate an SSE(TR) to G2 indicating that the connection has transitioned to text relay mode.

Upon reception of the SSE(TR) from G1, G2 will start the connect sequence with T2.

In this scenario, T1 and T2 modulations must be the same for the PTP connection to be established. There is no support for protocol conversion in this scenario. Ensuring that T1 and T2 modulations are the same allows non-PTP data modems that use the same physical layer modulation to also be supported using relay.

T1 = FDX modulation PTP (Bell 103 in this example)

T2 = FDX modulation PTP (same as T1)

G1 = supports modulation of T1

G2 = supports modulation of T2

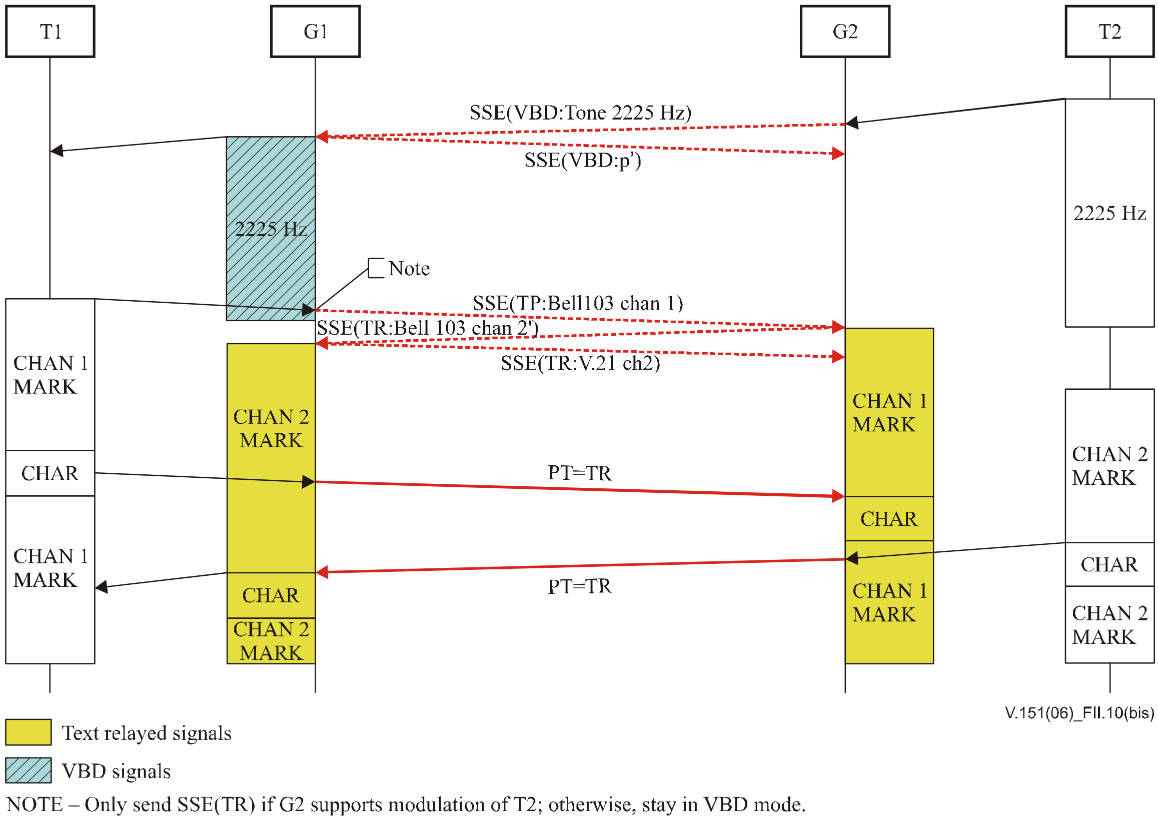


Figure II.10-1 – Scenarios #13 and #14 (Bell 103) call flow diagram (SSE)

Description

Bell 103 is a special case of the FDX to FDX scenario since, unlike other FDX modems such as V.21, the originating modem is first to generate carrier upon detection of the 2225-Hz answer tone.

T2 being the answering Bell 103 PTP device will initiate the generation of 2225-Hz answer tone. Since G2 is a V.151-compliant gateway that supports SSE protocol, upon detection of the 2225-Hz tone, G2 will initiate a switch to VBD mode. When G1 detects MARK for a Bell 103 originating modem that it supports for text, it shall generate an SSE(TP) if G2 has also indicated support for this modulation. If G2 does not support the modulation used by T2, the connection will remain in VBD for the duration of the call.

Upon reception of the SSE(TP), G2 shall respond with an SSE(TR) to transition the connection into text relay mode and start a connection with T2.

Upon reception of the SSE(TR) from G2, G1 will start the connect sequence with T1.

In this scenario, T1 and T2 modulations must be the same for the PTP connection to be established. There is no support for protocol conversion in this scenario. Ensuring that T1 and T2 modulations are the same allows non-PTP data modems that use the same physical layer modulation to also be supported using relay.

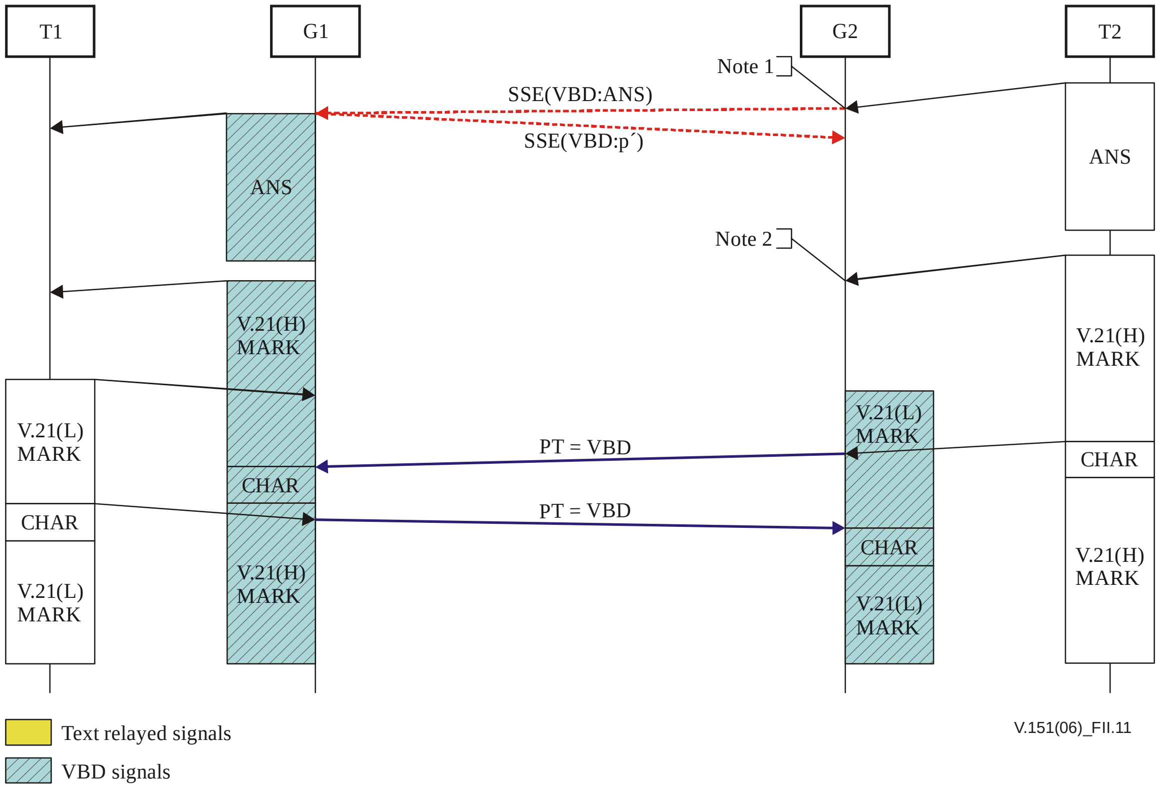
II.3.12 Scenarios #15 and #16

T1 = FDX modulation PTP (V.21 in this example)

T2 = FDX modulation PTP (same as T1)

G1 = don't care

G2 = does not support modulation of T2



NOTE 1 – Generation of ANS signal by T2 is optional.
NOTE 2 – If ANS was not generated by T2, G2 shall initiate transition to VBD upon V.21 mark detection.

Figure II.11 – Scenarios #15 and #16 call flow diagram (SSE)

Description

T2 being the answering PTP device will initiate the generation of an FDX signal, optionally preceded by the ANS signal. Since G2 is a V.151-compliant gateway that supports SSE protocol, upon detection of the FDX signal or the ANS tone, G2 will initiate a switch to VBD mode. Since G2 does not support the FDX modulation being used by T2, it will not initiate a transition to text relay mode through generation of SSE(TR). Upon detection of the modulation signal, if the channel is not already in VBD mode, G2 will initiate a transition to VBD mode. When T1 detects the carrier from T1 (since it may support the T1/T2 modulation), it will not initiate a transition to text relay since G2 had indicated that it does not support this modulation.

In this scenario, T1 and T2 modulations must be the same for the PTP connection to be established. There is no support for protocol conversion in this scenario.

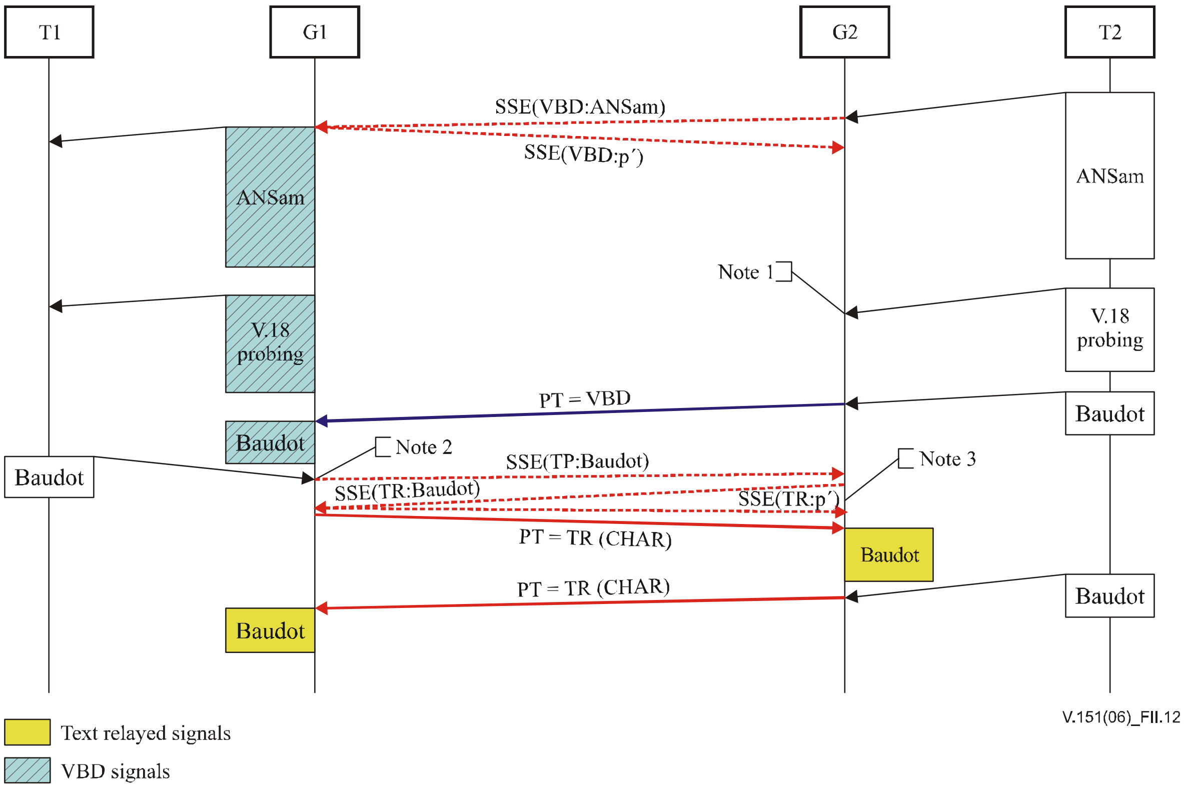
II.3.13 Scenarios #17 and #19

T1 = HDX modulation PTP (Baudot in this example)

T2 = V.18 PTP

G1 = supports T1's modulation

G2 = don't care



NOTE 1 – Allow end-to-end V.18 probing until T1 responds. G2 does not respond to any probing sequences that it supports during this probing.
NOTE 2 – SSE(TP) is only sent by G1 at this point if G2 is not V.18 native only; otherwise, stay in VBD mode.
NOTE 3 – G2 starts auto-mode probing after receiving SSE(TR) from G1. Probing should start with T1 modulation if supported by G2.

Figure II.12 – Scenarios #17 and #19 call flow diagram (SSE)

Description

T2 will generate a V.8 ANSam signal at the start of the call. Upon detection of the ANSam, G2 will initiate a transition to VBD mode through generation of a SSE(VBD:ANSam). The V.18 probing is allowed to be performed end-to-end through the VBD channel. G2 will not respond to any probing signals even in cases where it supports the modulation in the probing signals.

Upon detection of a PTP modulation signal supported by G1 (a Baudot character in this example), G1 shall immediately send an SSE(TP) to G2. In cases where G2 only support V.18 native mode, G1 shall not generate the SSE(TR) and the connection shall remain in VBD mode for the duration of the call.

Upon reception of the SSE(TR) from G1, G2 shall start its originating auto-mode probing state, waiting for T2 to generate an appropriate signal to respond to. If G2 supports T1's modulation, it shall use that modulation.

For the case of EDT, discrimination between data modems and PTP devices is determined by the half-duplexity of the signal. For data modems, the signal will be full duplex (i.e., there will be energy in both directions). For PTP, this will be half duplex. If EDT was being used by T1, G1 will only initiate a switch to text relay mode if it did not see energy in the opposite direction when it detected the EDT character.

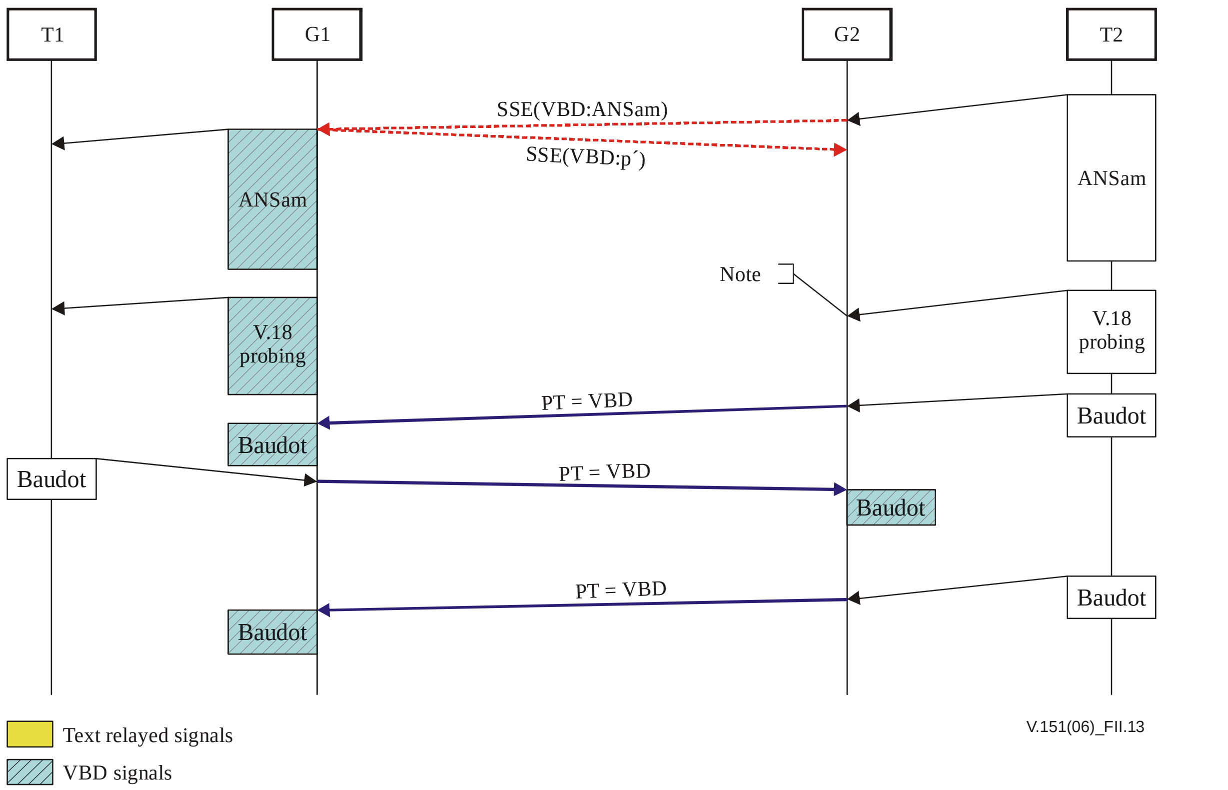
II.3.14 Scenarios #18 and #20

T1 = HDX modulation PTP (Baudot in this example)

T2 = V.18 PTP

G1 = does not support T1's modulation

G2 = don't care



NOTE – Allow end-to-end V.18 probing until T1 responds. G2 does not respond to any probing sequences that it supports during this probing.

Figure II.13 – Scenarios #18 and #20 call flow diagram (SSE)

Description

T2 will generate a V.8 ANSam signal at the start of the call. Upon detection of the ANSam, G2 will initiate a transition to VBD mode through generation of a SSE(VBD:ANSam). The V.18 probing is allowed to be performed end to end through the VBD channel. G2 will not respond to any probing signals even in cases where it supports the modulation in the probing signals.

Since G1 does not support the modulation used by T1, it will not detect a valid PTP signal and no transition to text relay mode will be initiated. T1 will connect with T2 using VBD mode for the duration of the call.

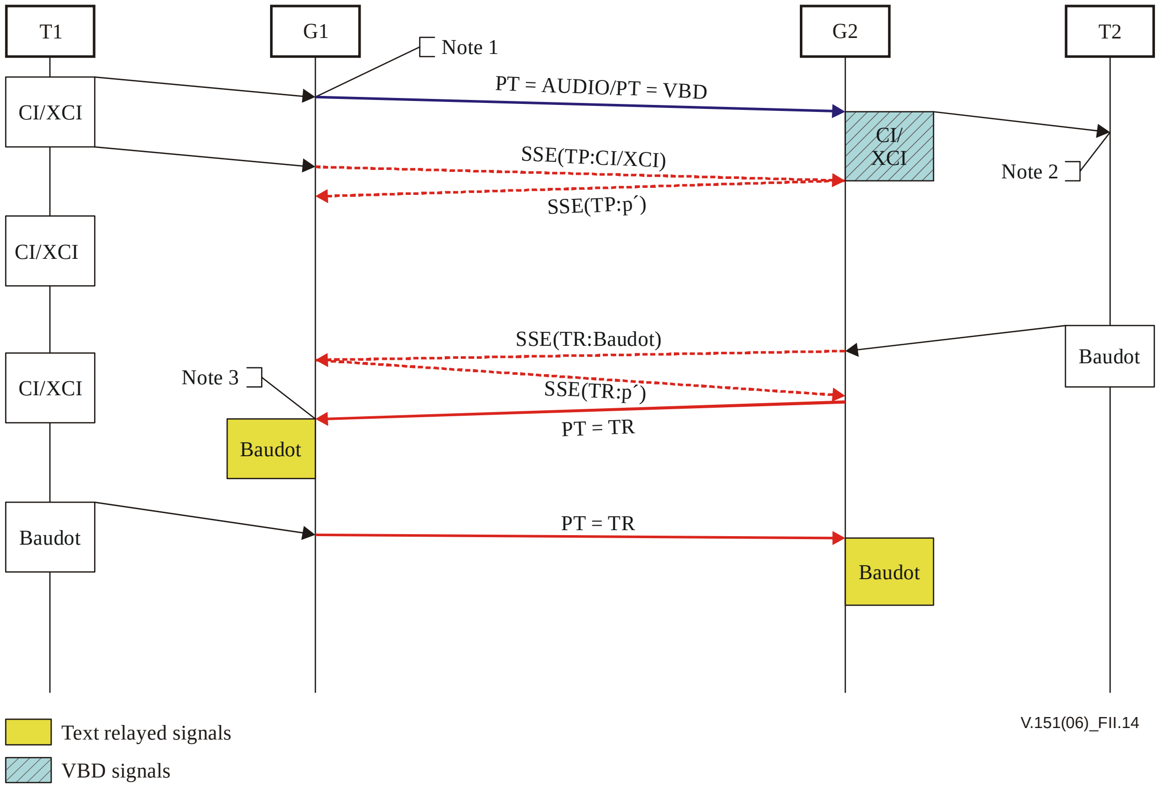
II.3.15 Scenarios #21 and #22

T1 = V.18 PTP

T2 = HDX modulation PTP (Baudot in this example)

G1 = supports any modulation

G2 = supports modulation of T2



NOTE 1 – VBD may optionally be used for CI/XCI signal. If VBD is used, there is an SSE(VBD:CI/XCI) handshake here.
NOTE 2 – CI/XCI may be detected by T2.
NOTE 3 – G1 will use modulation of T2 if it is available; otherwise, use first modulation on normal probing sequence.
.

Figure II.14 – Scenarios #21 and #22 call flow diagram (SSE)

Description

Since T1 is the V.18 calling PTP, it will generate a CI/XCI sequence. Since G1 is a V.151-compliant gateway that supports SSE protocol, it will detect CI/XCI indicating to G1 that it is connected to a V.18 PTP terminal. Upon detection of CI/XCI, G1 may continue to transmit the CI/XCI sequence using audio mode to G2, or it may optionally initiate a transition to VBD mode by sending an SSE(VBD:CI/XCI) to G2. The diagram does not show the optional VBD SSE negotiation at this point. After CI/XCI sequence is completely transmitted to G2 (either through audio or VBD encoding), G1 shall send an SSE(TP:CI/XCI) to G2 and terminate regeneration of signals to T1. G1 shall stay in the mode of silence generation to T1 until it receives an SSE response from G2 indicating a new mode (either VBD or text relay).

Upon reception of the SSE(TR:CI/XCI) from G1, G2 will enter an answering auto-mode probing state, searching for a signal that it detects as a valid PTP signal. Once a valid PTP signal is detected by G2, G2 will send an SSE(TR) to G1. After the SSE(TR:p´) response is received from G1, G2 will start to transmit received characters using text relay.

Upon receiving a SSE(TR) from G2, G1 will start a connect sequence with T1 (which it knows is a V.18 PTP device). G1 shall use the modulation being used by T2 if it is supported, or else will use the first modulation in the normal auto-mode probing sequence. G1 shall buffer the character received from G2 until it has established connection with T1.

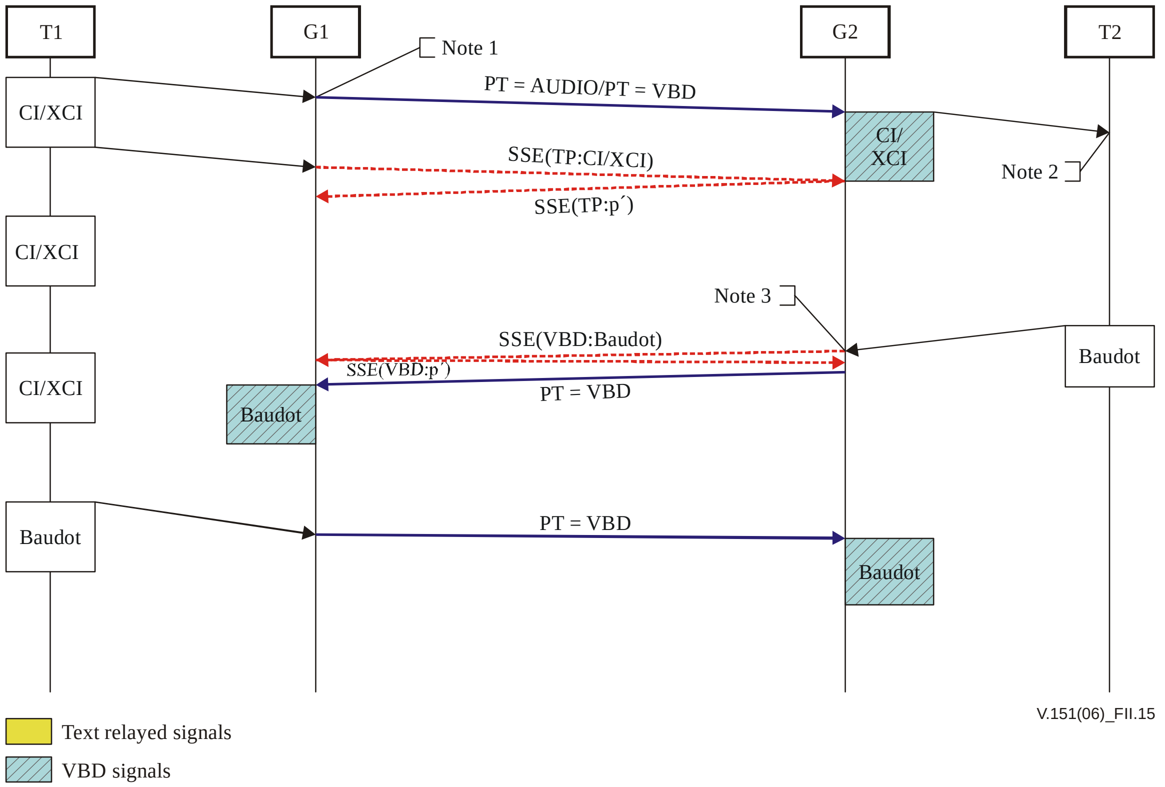
II.3.16 Scenarios #23 and #24

T1 = V.18 PTP

T2 = HDX modulation PTP (Baudot in this example)

G1 = supports any modulation

G2 = does not support modulation of T2



NOTE 1 – VBD may optionally be used for CI/XCI signal. If VBD is used, there is an SSE(VBD:CI/XCI) handshake here.
NOTE 2 – CI/XCI may be detected by T2.
NOTE 3 – Since G2 is a V.151-compliant gateway, it is required to detect all varieties of V.18 modulations for the purposes of initiating transition to VBD mode.

Figure II.15 – Scenario #23 and #24 call flow diagram (SSE)

Description

Since T1 is the V.18 calling PTP, it will generate a CI/XCI sequence. Since G1 is a V.151-compliant gateway that supports SSE protocol, it will detect CI/XCI indicating to G1 that it is connected to a V.18 PTP terminal. Upon detection of CI/XCI, G1 may continue to transmit the CI/XCI sequence using audio mode to G2, or it may optionally initiate a transition to VBD mode by sending an SSE(VBD:CI/XCI) to G2. The diagram does not show the optional VBD SSE negotiation at this point. After CI/XCI sequence is completely transmitted to G2 (either through audio or VBD encoding), G1 shall send an SSE(TP:CI/XCI) to G2 and terminate regeneration of signals to T1. G1 shall stay in the mode of silence generation to T1 until it receives an SSE response from G2 indicating a new mode (either VBD or text relay).

Upon reception of the SSE(TP:CI/XCI) from G1, G2 will enter an answering auto-mode probing state, searching for a signal that it detects as a valid PTP signal. When G2 detects a signal that is not a PTP modulation that it supports for modem relay, G2 will transmit to G1 an SSE(VBD). G1 will respond with an SSE(VBD:p´) completing the transition to VBD mode. The connection will stay in VBD mode for the duration of the session.

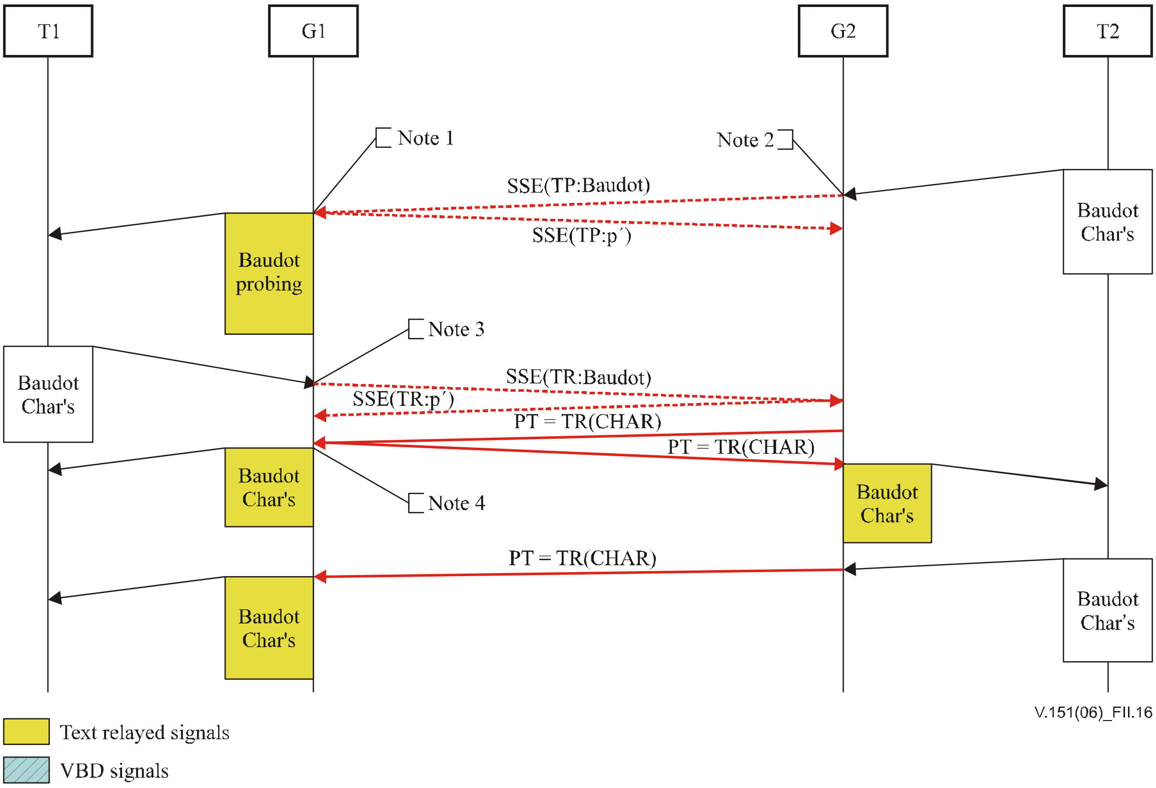
II.3.17 Scenario #25

T1 = HDX protocol (Baudot in this example)

T2 = HDX protocol (Baudot in this example)

G1 = supports T1's modulation

G2 = supports T2's modulation



NOTE 1 – Probing sequence start with Baudot modulation if available (since SSE indicated Baudot being used by T2).
NOTE 2 – Received characters are buffered by G2 until mode transition to TR is complete.
NOTE 3 – Received characters are buffered by G1 until mode transition to TR is complete.
NOTE 4 – Since Baudot is half duplex, G1 shall wait for loss of carrier from T1 before transmitting received characters from G2.

Figure II.16 – Scenario #25 call flow diagram (SSE)

Description

T2, being the called PTP terminal, will, according to PTP etiquette, initiate the first text transmission. Since G2 supports the HDX modulation used by T2, it will detect the modulation per V.151 requirements and send an SSE(TP:Baudot) to G1 indicating this G2 shall not respond to T1 until transition to text relay mode is complete and it receives characters from G1. G2 may send the received characters to G1 after receiving the SSE(TP:p´) from G1. These characters may be used by G1 when generating probing sequences.

Upon reception of the SSE(TP:Baudot) from G2, G1 shall enter originating auto-mode probing state. If G1 supports Baudot, it shall start the probing sequence with Baudot. Upon receiving a valid response from T1, G1 shall send an SSE(TR:Baudot) response back to G2. G2 will then immediately respond with an SSE(TR:p´) completing the transition to text relay mode.

Once in text relay mode, G1 and G2 will send their received characters via RTP. Note that since Baudot (and all other NCC modulations) are half duplex, the gateways might have to buffer characters received from the IP network until their local terminals have stopped transmitting.

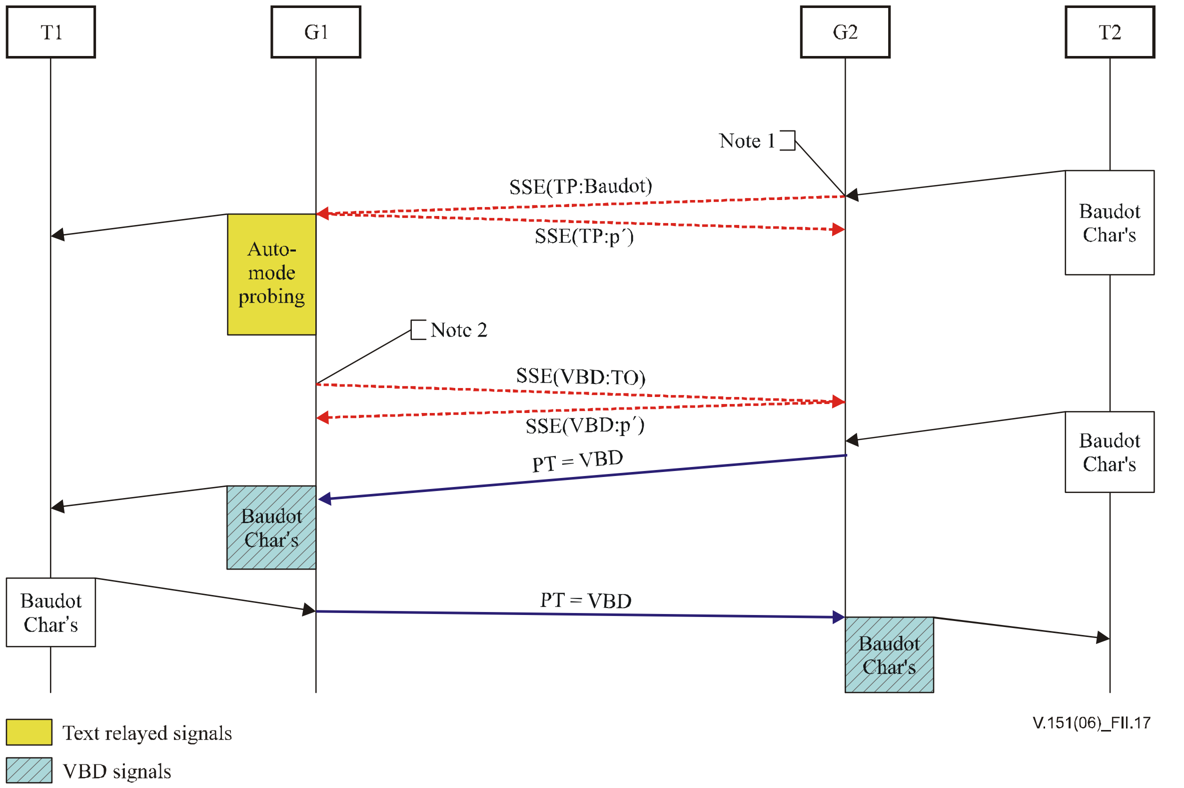
II.3.18 Scenario #26

T1 = HDX protocol (Baudot in this example)

T2 = HDX protocol (Baudot in this example)

G1 = does not support T1's modulation

G2 = supports T2's modulation



NOTE 1 – Received characters are buffered by G2 until mode transition to TR is complete. These characters will be discarded in this case since the channel does not complete transition to text relay.

NOTE 2 – G1 times out during auto-mode probing.

Figure II.17 – Scenario #26 call flow diagram (SSE)

Description

T2, being the called PTP terminal, will, according to PTP etiquette, initiate the first text transmission. Since G2 supports the HDX modulation used by T2, it will detect the modulation per V.151 requirements and send an SSE(TP:Baudot) to G1 indicating a request to transition to text relay mode. G2 shall buffer all received characters until a transition to text relay mode is complete. G2 shall not respond to T1 until transition to text relay mode is complete and it receives characters from G1.

Upon reception of the SSE(TP:Baudot) from G2, G1 shall enter originating auto-mode probing state. Since G1 does not support the T1 modulation in this scenario, G1 will eventually time out and send a SSE(VBD) to G2, transitioning the session to VBD mode. G2 will discard the buffered characters and complete the transition to VBD mode by responding to G1 with an SSE(VBD:p´). The channel shall remain in VBD mode for the duration of the session.

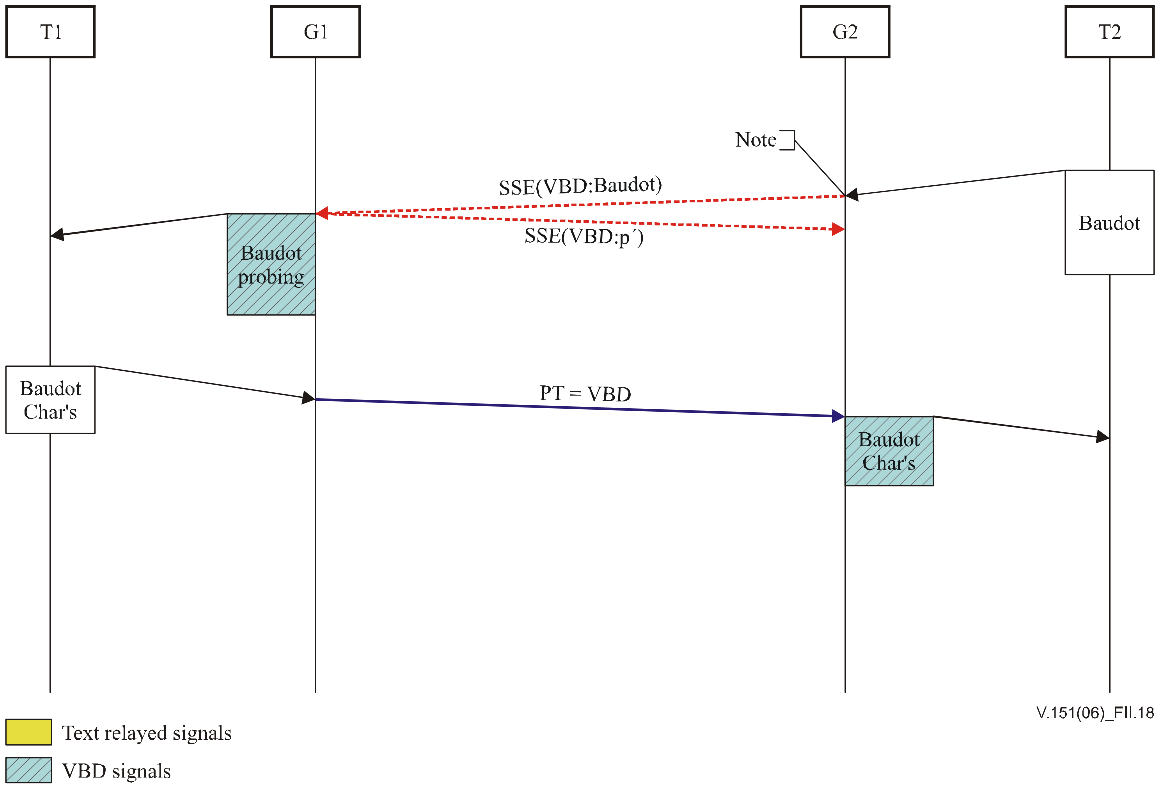
II.3.19 Scenario #27

T1 = HDX protocol (Baudot in this example)

T2 = HDX protocol (Baudot in this example)

G1 = supports T1's modulation

G2 = does not support T2's modulation



NOTE – V.151-compliant gateways are required to detect PTP modulations for purposes of initiating transition to VBD mode.

Figure II.18 – Scenario #27 call flow diagram (SSE)

Description

T2, being the called PTP terminal, will, according to PTP etiquette, initiate the first text transmission. G2 will detect the Baudot signal and initiate a transition to VBD mode since it is not a modulation that it supports for text relay. The session will remain in VBD mode.

If the connection did not follow PTP etiquette, and T1 transmitted first, G1 would initiate a transition to text relay mode through transmission of an SSE(TR) to G2. G2 would time out (or detect unsupported PTP signal) and respond with SSE(VBD) as per scenario #26.

II.3.20 Scenario #28

T1 = HDX protocol (Baudot in this example)

T2 = HDX protocol (Baudot in this example)

G1 = does not support T1's modulation

G2 = does not support T2's modulation

Description

In this scenario, neither gateway supports the modulation used by their local PTP devices. Since the gateways are V.151-compliant, they will detect a PTP signal that is not supported via text relay and will initiate a transition to VBD mode. The gateway receiving the SSE(VBD) will respond with SSE(VBD) and the transition to VBD will be completed. The gateways will stay in VBD mode for the duration of the call.

II.3.21 Scenario #29

T1 = FDX PTP device (V.21 in this example)

T2 = HDX PTP device (Baudot in this example)

G1 = supports T1's modulation

G2 = supports T2's modulation

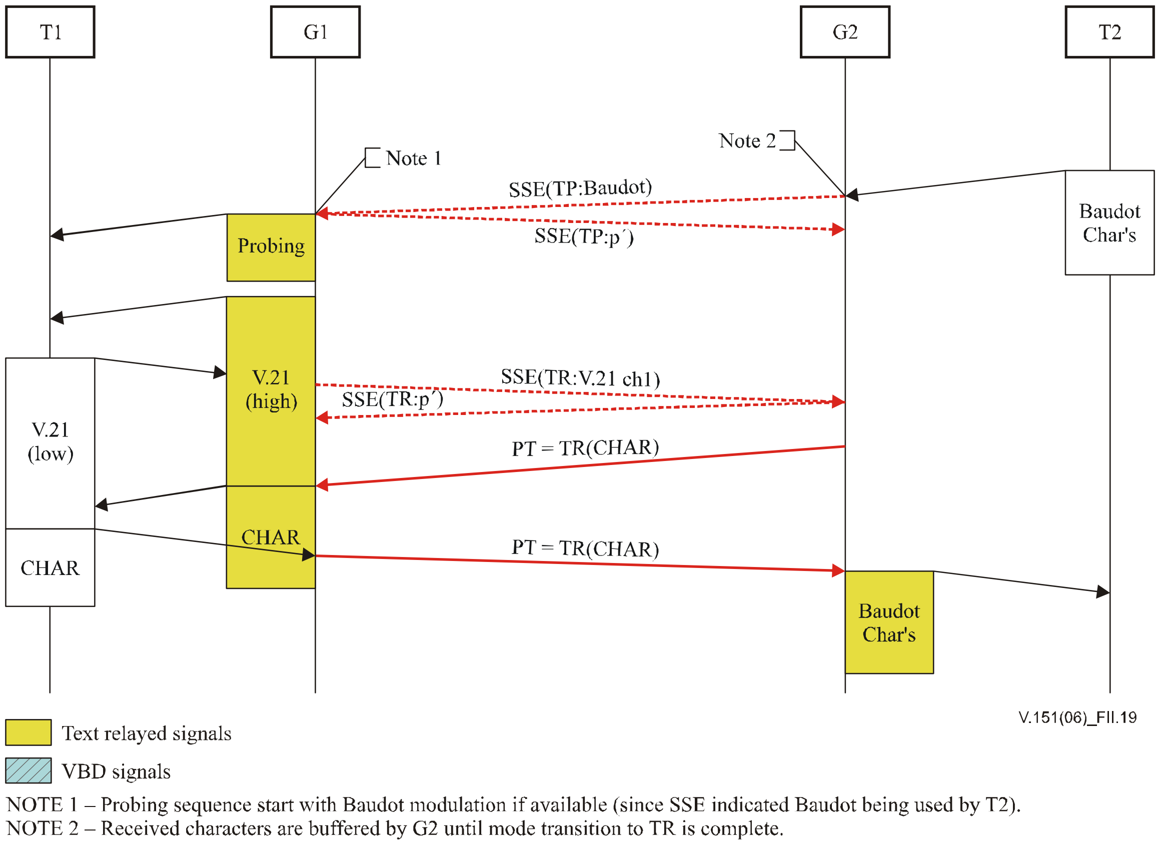


Figure II.19 – Scenario #29 call flow diagram (SSE)

Description

As per PTP etiquette, T2 will initially transmit a character which will be detected by G2 as a valid PTP modulation supported for text relay. G2 will then generate an SSE(TP) to G1. G2 may send characters received to G1 so that G1 could use these in its probing signals.

Upon reception of the SSE(TP) from G2, G1 will start an originating auto-mode probing sequence with T1. G1 should start the auto-mode probing sequence with T2's modulation if it is supported. Since G1 supports the FDX modulation in this scenario, it will eventually connect with T1, at which time it shall send an SSE(TR) to G2 indicating it has successfully connected with a PTP device in text relay mode.

When G2 receives the SSE(TR) from G1, it will respond with SSE(TR:p´) completing the transition to text relay mode and send any buffer characters to G1.

II.3.22 Scenarios #30 to #32

In these scenarios, either or both of the gateways does not support the modulation of its local PTP device. Both PTP devices are not V.18 PTP devices, and one is FDX based while the other is HDX based. No successful connection will be achieved, although the channel may be transitioned to VBD mode since the gateway should detect either of the PTP signals.

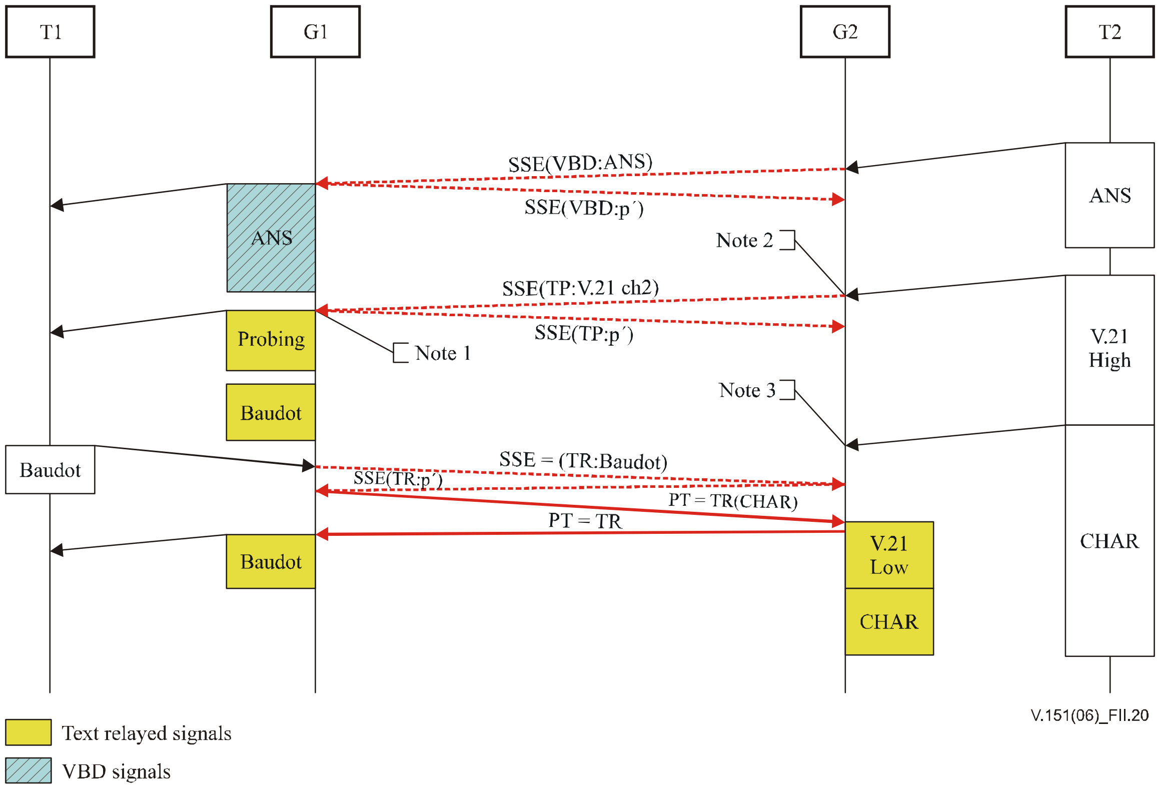
II.3.23 Scenario #33

T1 = HDX PTP device (Baudot in this example)

T2 = FDX PTP device (V.21 in this example)

G1 = supports T1's modulation

G2 = supports T2's modulation



NOTE 1 – Probing sequence starts with V.21 modulation if available (since SSE indicated V.21 being used by T2).
NOTE 2 – Send SSE(TR:V.21) if G1 supports T2 and reason codes are used; otherwise, stay in VBD mode.
NOTE 3 – Characters are buffered by G2 until transition to text relay mode complete.

Figure II.20 – Scenario #33 call flow diagram (SSE)

Description

Since T2 is an FDX answering PTP, it will initiate the call by either sending the optional ANS signal or carrier. Upon detection of the ANS signal, G2 will transition to VBD mode by sending an SSE(VBD). Upon detection of a PTP character, G2 will send an SSE(TP) to G1 to not respond to T2 (until it gets an SSE(TR) from T1.

Upon receiving the SSE(TR) from G2, G1 will start an auto-mode probing sequence, starting with the modulation used by T2 if available. If G1 receives a response from T1 that indicates that T1 is a HDX PTP device that uses a modulation that is supported by G1, G1 will respond with a SSE(TR) indicating this to G2. If G1 gets a response from T1 for an FDX modulation that is not the same as that being used by T2, G1 will send an SSE(VBD) as per scenarios #13 and #14.

When G2 receives the SSE(TR) response from G1, it will respond with SSE(TR:p´) completing the transition to text relay mode. G2 will then start the connect sequence with T2.

II.3.24 Scenarios #34 to #36

In these scenarios, either or both of the gateways does not support the modulation of its local PTP device. Both PTP devices are not V.18 PTP devices, and one is FDX based while the other is HDX based. No successful connection will be achieved, although the channel may be transitioned to VBD mode since the gateway should detect either of the PTP signals.

II.4 Scenarios with payload type switching being used

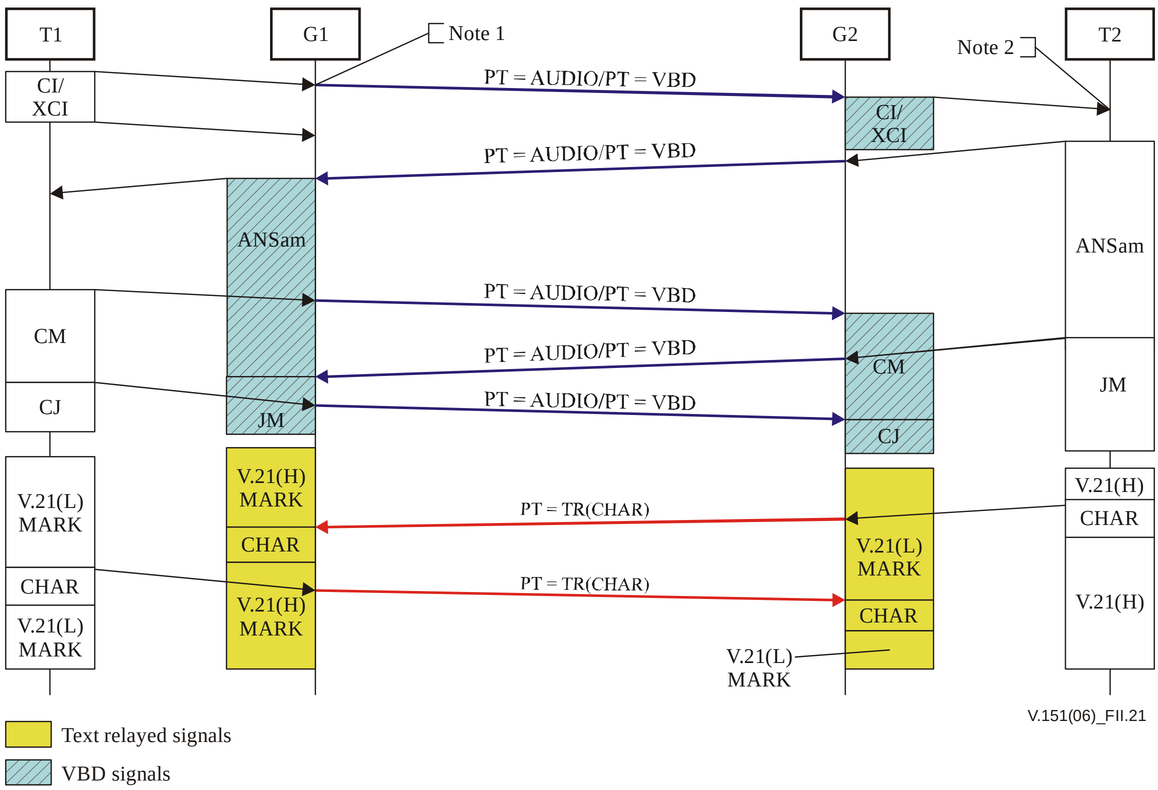
II.4.1 Scenario #1

T1 = V.18 PTP

T2 = V.18 PTP

G1 = supports V.18 native mode

G2 = supports V.18 native mode



NOTE 1 – VBD may optionally be used for CI/XCI signal.
NOTE 2 – CI/XCI may be detected by T2. If not detected, there will be up to 3 seconds of additional delay before ANSam is generated.

Figure II.21 – Scenario #1 call flow diagram (Payload type)

Description

Since T1 is the V.18 calling PTP, it will generate a CI/XCI sequence. G1 may optionally detect CI/XCI and transmit it in VBD mode. Upon detection of ANSam by G2, if G1 has not already switched to VBD encoding based on CI/XCI detection, G2 will start VBD encoding. The V.8 handshake is allowed to proceed using VBD mode. G1 and G2 monitor the local CM or

CJ sequences to determine that the endpoint devices are PTP devices. After completion of V.8 sequence between the two PTP devices using VBD mode, the gateways shall (within the 75-ms requirement) transition to text relay mode, locally generating the appropriate V.21 carrier and transmitting received characters via text relay encoding.

II.4.2 Scenarios #2, #3 and #4

In these scenarios, both of the PTP devices are V.18 devices. As in scenario #1, the connection will enter VBD mode either through detection of the CI/XCI or through detection of ANSam. Since one or both of the gateways in these scenarios does not support V.18 native mode, the connection shall stay in VBD mode. The gateways shall monitor the CM/JM sequence, detecting the V.18 PTP devices but maintaining a VBD connection.

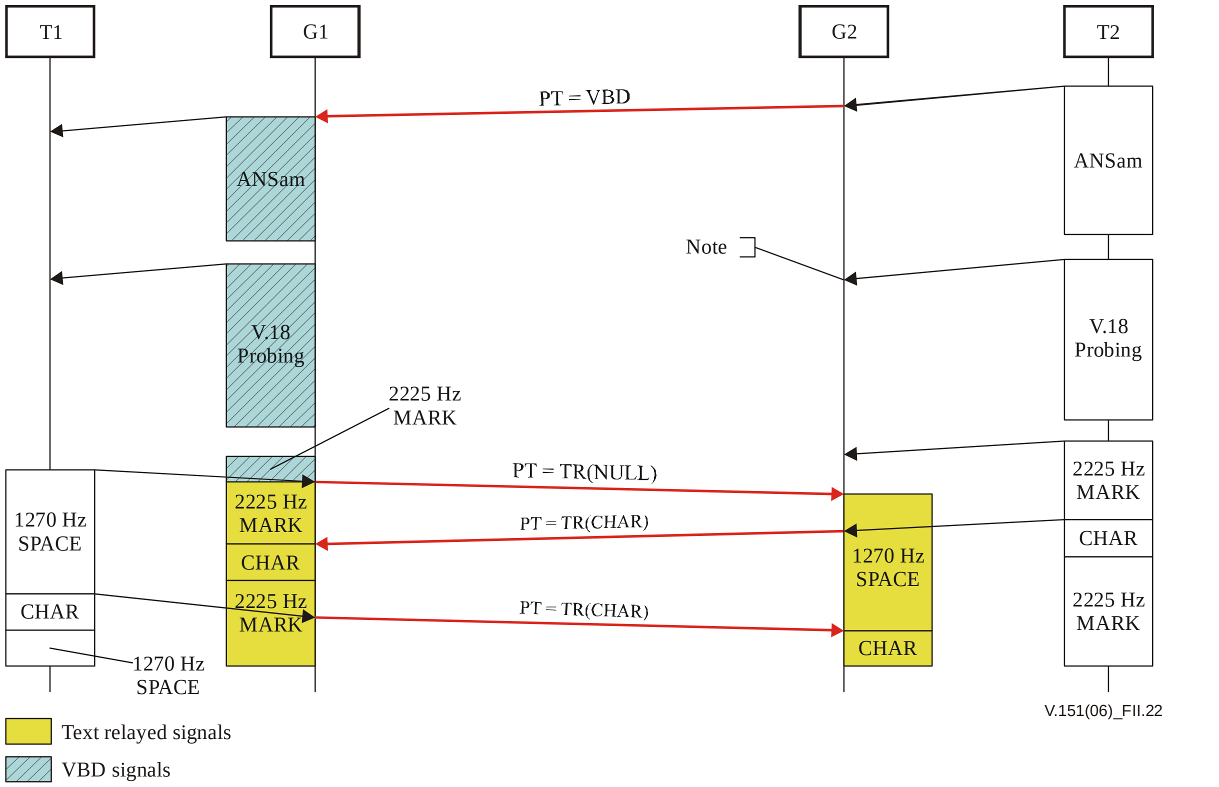
II.4.3 Scenario #5

T1 = FDX modulation PTP (Bell 103 in this example)

T2 = V.18 PTP

G1 = supports FDX modulation (Bell 103 in this example)

G2 = supports FDX modulation (Bell 103 in this example)



NOTE – Allow end-to-end V.18 probing until T1 responds. G2 does not respond to any probing sequences that it supports during this probing.

Figure II.22 – Scenario #5 call flow diagram (Payload type)

Description

Since T2 is a V.18 answering PTP, it will generate ANSam after 3 seconds (because CI/XCI was not received). Since G2 is a V.151-compliant gateway, it will detect ANSam and initiate a switch to VBD mode. T2 is allowed to probe T1 using VBD mode. G2 does not respond to any of the T2 probing sequences, even those that represent modulations that G2 supports since it has not been established that an end-to-end text relay connection can be successfully made between all gateways and PTP devices. Upon detection by G1 of a PTP FDX modulation probing response from T1 that it supports, G1 will initiate a switch to text relay. In the case where T1 and T2 are not PTP, but data modems, we should require that both PSTN legs of the call be of the same modulation type so as to result in successful connection for data modems as well as PTP devices using proper character treatments. There are no mechanisms in payload type switching for ensuring this.

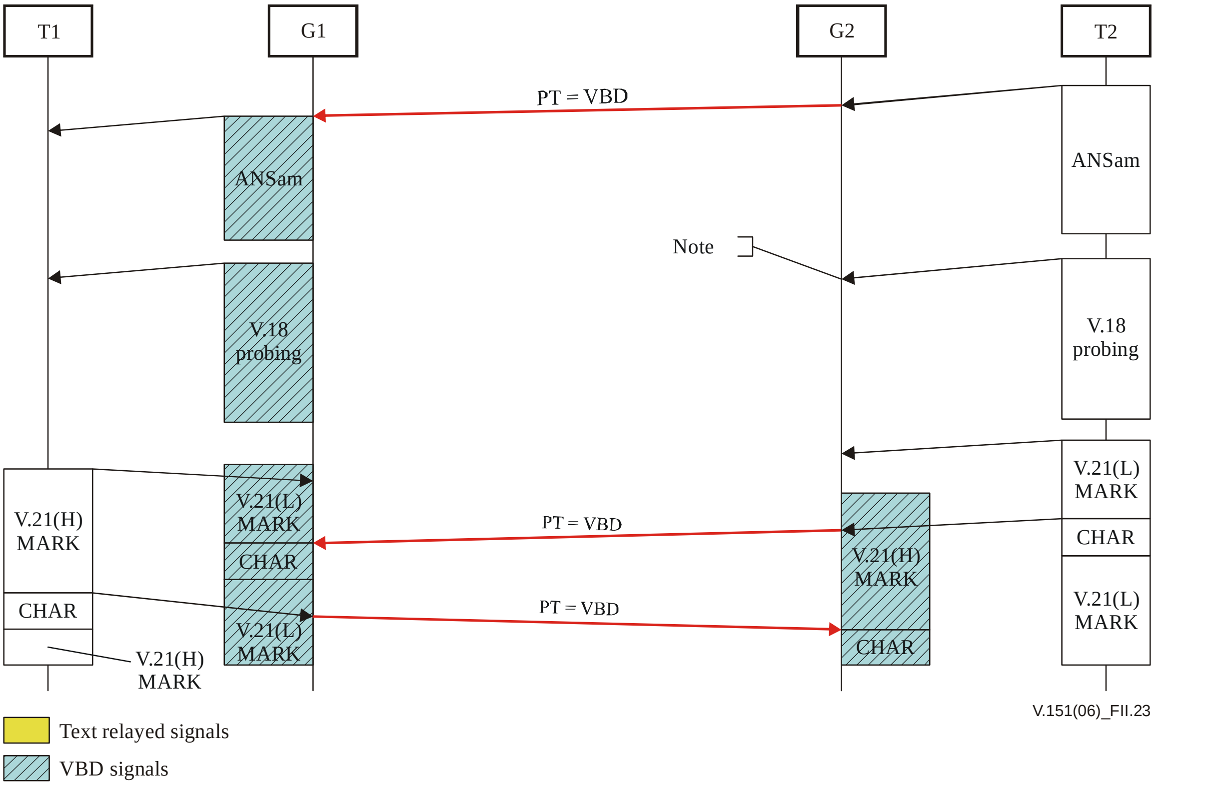
II.4.4 Scenario #6

T1 = FDX modulation PTP (V.21 in this example)

T2 = V.18 PTP

G1 = does not support modulation used by T1

G2 = don't care



NOTE – Allow end-to-end V.18 probing until T1 responds. G2 does not respond to any probing sequences that it supports during this probing.

Figure II.23 – Scenario #6 call flow diagram (Payload type)

Description

Since T2 is a V.18 answering PTP, it will generate ANSam after 3 seconds (because CI/XCI was not received). Since G2 is a V.151-compliant gateway, it will detect ANSam and initiate a switch to VBD mode. T2 is allowed to probe T1 using VBD mode. G2 does not respond to any of the T2 probing sequences, even those that represent modulations that G2 supports since it has not been established that an end-to-end text relay connection can be successfully made between all gateways and PTP devices. Since the modulation that is responded to by T1 is not supported by G1, G1 does not initiate a switch to text relay mode and the connection remains in VBD mode for the entire call.

II.4.5 Scenario #7

T1 = FDX modulation PTP (Bell 103 in this example)

T2 = V.18 PTP

G1 = supports FDX modulation (Bell 103 in this example)

G2 = supports FDX modulation (Bell 103 in this example) (Does not support native V.18)

Description

This scenario plays out identical to scenario #5. The only difference between these scenarios is that G2 does not support native V.18 mode, but the call flow is identical.

II.4.6 Scenario #8

T1 = FDX modulation PTP (V.21 in this example)

T2 = V.18 PTP

G1 = does not support modulation used by T1

G2 = does not support native V.18 mode

Description

This scenario plays out identical to scenario #5. The only difference between these scenarios is that G2 does not support native V.18 mode, but the call flow is identical.

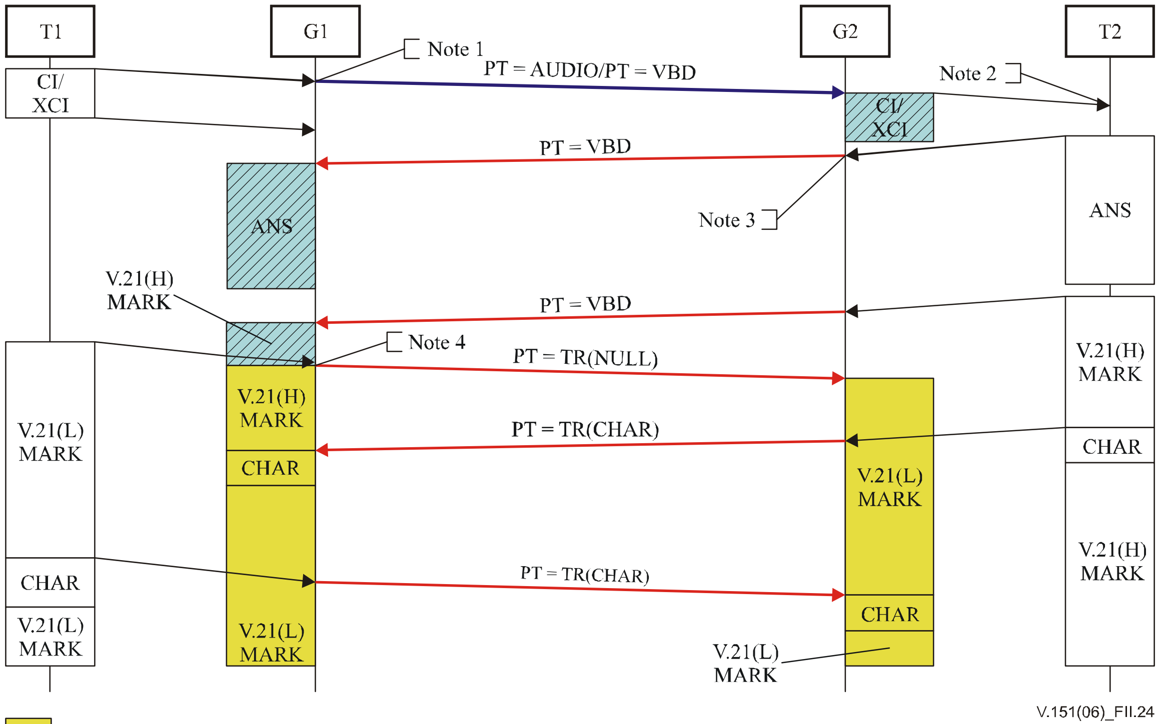
II.4.7 Scenarios #9 and #10

T1 = V.18 PTP

T2 = FDX modulation PTP (V.21 in this example)

G1 = supports any modulation

G2 = supports modulation of T2 (V.21 in this example)



Text relayed signals

VBD signals

NOTE 1 – VBD may optionally be used for CI/XCI signal. If VBD is used, there is an SSE(VBD:CI/XCI) handshake here.
NOTE 2 – CI/XCI may be detected by T2.
NOTE 3 – ANS tone is optional.
NOTE 4 – Transition to text relay mode only if both G1 and G2 support the modulation used by T1 and T2.

Figure II.24 – Scenarios #9 and #10 call flow diagram (Payload type)

Description

Since T1 is the V.18 calling PTP, it will generate a CI/XCI sequence. G1 may optionally detect CI/XCI and transmit it in VBD mode. Upon detection of ANS by G2, if G1 has not already switched to VBD encoding based on CI/XCI detection, G2 will start VBD encoding. Text relay mode is entered by G1 detecting a valid PTP modulation from T1 in response to signals generated by T2. G1 only switches to text relay encoding if both G1 and G2 support the modulation that is being used by T1 and T2. There is no protocol conversion supported in this scenario, since it is not known at the time of switchover to text relay if the terminal devices are PTP or data modem devices. By guaranteeing that both PSTN call legs are the same modulation, data modems can transparently be supported.

In the case where ANS was not generated, G2 will detect the carrier signal from T2 and initiate switch to VBD mode. Note that for V.21, initial FDX answering signals are distinct from signals used by EDT modems. This is important because the call flow from non-FDX answer devices is quite different.

II.4.8 Scenarios #11 and #12

In these scenarios, G2 does not support the modulation used by T1. As in scenarios #9 and #10, the connection will enter VBD mode either through the detection of CI/XCI by G1 or ANS by G2. Since G2 does not support the modulation used by T2, G1 will not transition the connection to text relay encoding even if it detects the PTP modulation used. The connection will stay in VBD mode.

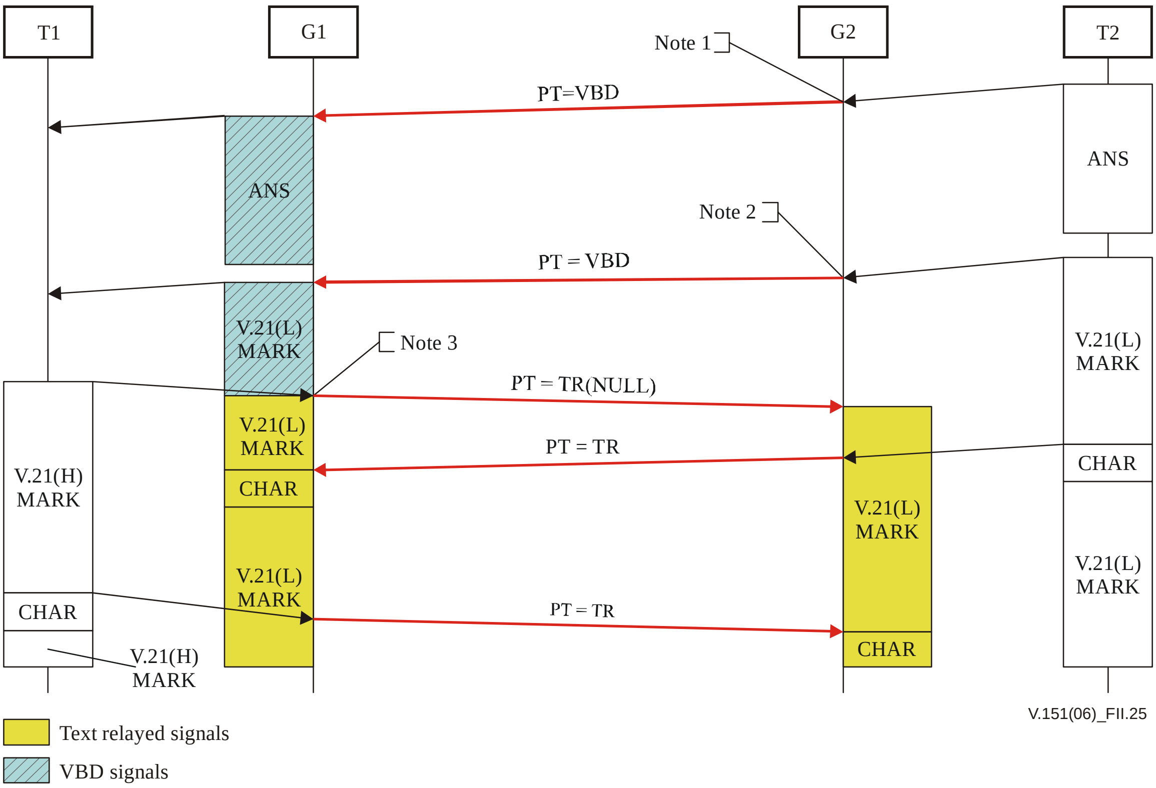
II.4.9 Scenarios #13 and #14

T1 = FDX modulation PTP (V.21 in this example)

T2 = FDX modulation PTP (same as T1)

G1 = supports modulation of T1

G2 = supports modulation of T2



NOTE 1 – Generation of ANS signal by T2 is optional.
NOTE 2 – If ANS was not generated by T2, G2 shall initiate transition to VBD upon V.21 mark detection.
NOTE 3 – Only use text relay encoding if G2 supports modulation of T1; otherwise, stay in VBD mode.

Figure II.25 – Scenarios #13 and #14 call flow diagram (Payload type)

Description

T2 being the answering PTP device will initiate the generation of a FDX signal, optionally proceeded by the ANS signal. Since G2 is a V.151-compliant gateway, upon detection of the ANS tone, G2 will initiate a switch to VBD mode. When G2 detects the carrier for a FDX modulation that it supports for text, it shall initiate a transition to text relay mode through the generation of a packet containing the NULL character encoded using TR payload type if G1 has also indicated support for this modulation. If G1 does not support the modulation used by T2, the connection will remain in VBD for the duration of the call.

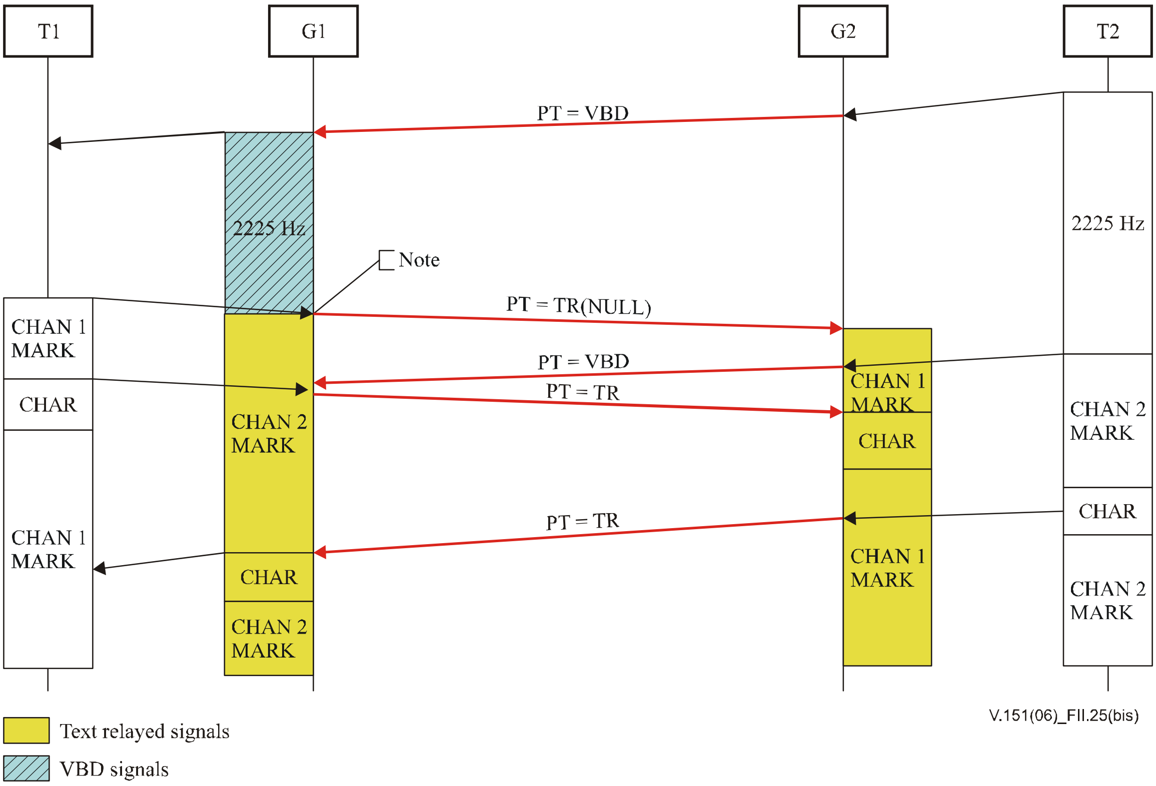
Upon reception of TR payload type from G2, G1 shall enter an originate mode auto-probing sequence. Since T1 is the same modulation as T2, G1 will connect with T1 using this modulation. Upon detection of the response signal from T1, G1 will encode the received characters using text relay payload type.

T1 = FDX modulation PTP (Bell 103 in this example)

T2 = FDX modulation PTP (same as T1)

G1 = supports modulation of T1

G2 = supports modulation of T2



NOTE – Only use text relay encoding if G2 supports modulation of T1; otherwise, stay in VBD mode.

Figure II.25-1 – Scenarios #13 and #14 (Bell 103) call flow diagram (Payload type)

Description

Bell 103 is a special case of the FDX to FDX scenario since, unlike other FDX modems such as V.21, the originating modem is first to generate carrier upon detection of the 2225-Hz answer tone.

T2 being the answering Bell 103 PTP device will initiate the generation of 2225-Hz answer tone. Since G2 is a V.151-compliant gateway, upon detection of the 2225-Hz tone, G2 will initiate a switch to VBD mode. When G1 detects MARK for a Bell 103 originating modem that it supports for text, it shall initiate a transition to text relay mode through the generation of a packet containing the NULL character encoded using TR payload type if G2 has also indicated support for this modulation. If G2 does not support the modulation used by T2, the connection will remain in VBD for the duration of the call.

Upon reception of the TR payload type, G2 will start connect sequence with T2.

In this scenario, T1 and T2 modulations must be the same for the PTP connection to be established. There is no support for protocol conversion in this scenario. Ensuring that T1 and T2 modulations are the same allows non-PTP data modems that use the same physical layer modulation to also be supported using relay.

II.4.10 Scenarios #15 and #16

In these scenarios, G2 does not support the modulation used by T2. As in scenario #13, the connection will enter VBD mode by the detection of ANS or constant carrier by either gateway.

T1 will not initiate a transition to text relay encoding since G2 does not support the detected modulation.

II.4.11 Scenarios #17, #18, #19 and #20

These scenarios are identical to the call flow described in the SSE case, with the exception of using payload type switching instead of SSE protocol.

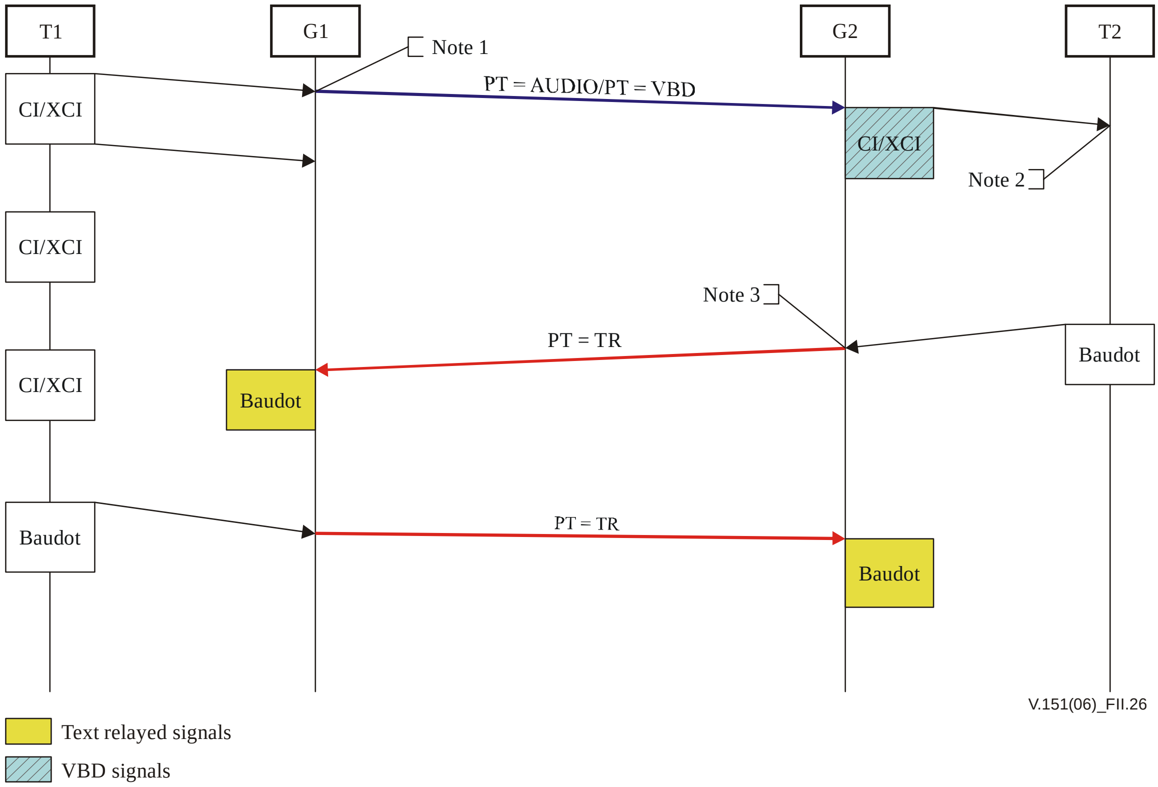
II.4.12 Scenarios #21 and #22

T1 = V.18 PTP

T2 = HDX modulation PTP (Baudot in this example)

G1 = supports any modulation

G2 = supports modulation of T2



NOTE 1 – VBD may optionally be used for CI/XCI signal. If VBD is used, there is an SSE(VBD:CI/XCI) handshake here.
NOTE 2 – CI/XCI may be detected by T2.
NOTE 3 – Only use text relay encoding if G1 supports the modulation being used by T2.

Figure II.26 – Scenarios #21 and #22 call flow diagram (Payload type)

Description

In this scenario, G1 may switch to VBD encoding on the detection of CI/XCI. When the signal from T2 is detected by G2, G2 will start text relay encoding only if G1 also supports this modulation. In scenario #21, this will result in text relay being used for the session.

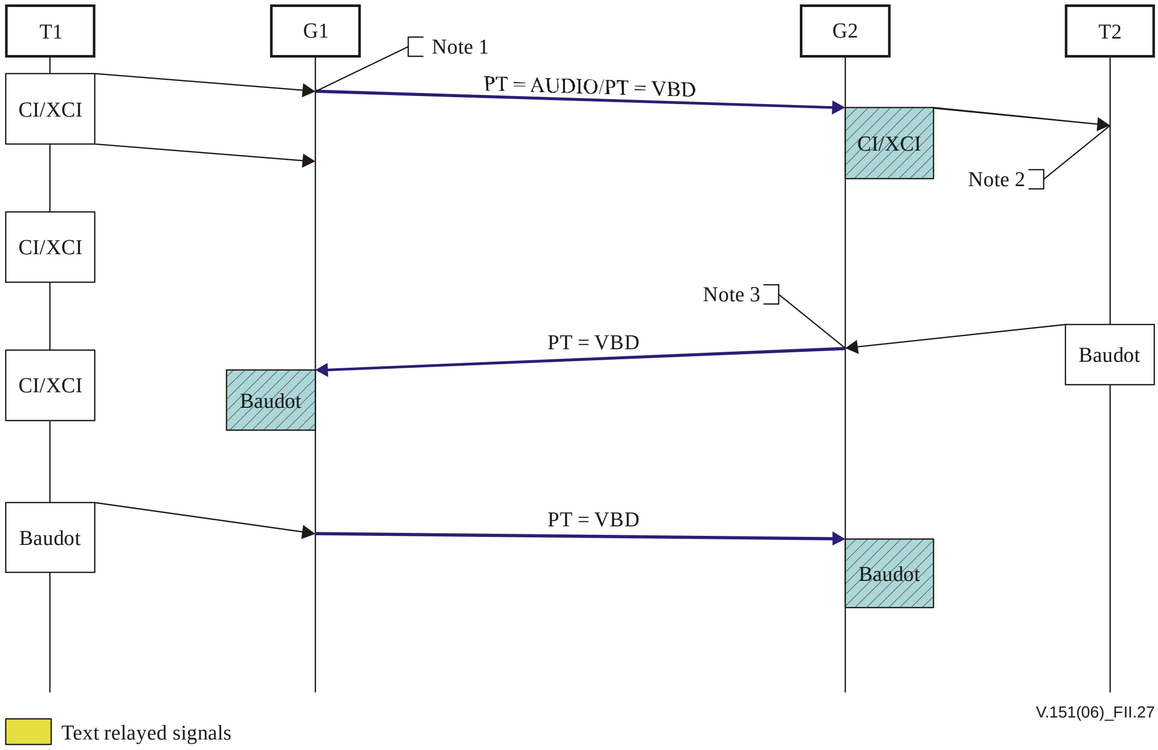
II.4.13 Scenarios #23 and #24

T1 = V.18 PTP

T2 = HDX modulation PTP (Baudot in this example)

G1 = don't care

G2 = does not support modulation of T2



NOTE 1 – VBD may optionally be used for CI/XCI signal. If VBD is used, there is an SSE(VBD:CI/XCI) handshake here.
NOTE 2 – CI/XCI may be detected by T2.
NOTE 3 – G2 is required to be able to detect modulation for the purpose of transitioning to VBD encoding.

Figure II.27 – Scenarios #23 and #24 call flow diagram (Payload type)

Description

In this scenario, G1 may switch to VBD encoding on the detection of CI/XCI. If G1 did not switch to VBD on CI/XCI, G2 will initiate a switch to VBD encoding upon detection of a PTP signal that it does not support in text relay mode. The session will stay in VBD mode. Note G1 will not initiate a transition to text relay mode upon detection of the PTP signal from T1 since it knows that G2 does not support this modulation.

II.4.14 Scenarios #25 and #26

The call flow for these scenarios is the same as scenarios #21 and #22. The difference in the scenarios is that T1 is a V.18 modem, but this does not affect the operation of the gateways. If T1 and T2 are not the same modulation, the connection will fail since there is no support for conversion in these scenarios.

II.4.15 Scenario #27

The connection stays in VBD for this scenario. Since G2 does not support text relay for the HDX modulation used by T2, it will transition to VBD mode upon detection of the signal. G1 does support the modulation used by T1, but it will not transition to text relay mode since G2 does not support this modulation. If G1 detects the first signal, it will instead transition to VBD mode. If T1 and T2 use different HDX modulations, the connection will fail since there is no support for conversion in this scenario.

II.4.16 Scenario #28

Neither gateway supports text relay for the modulations used by T1 and T2. The gateways will transition to VBD encoding upon detection of the signal from there local PTP devices and the connection will remain in VBD mode.

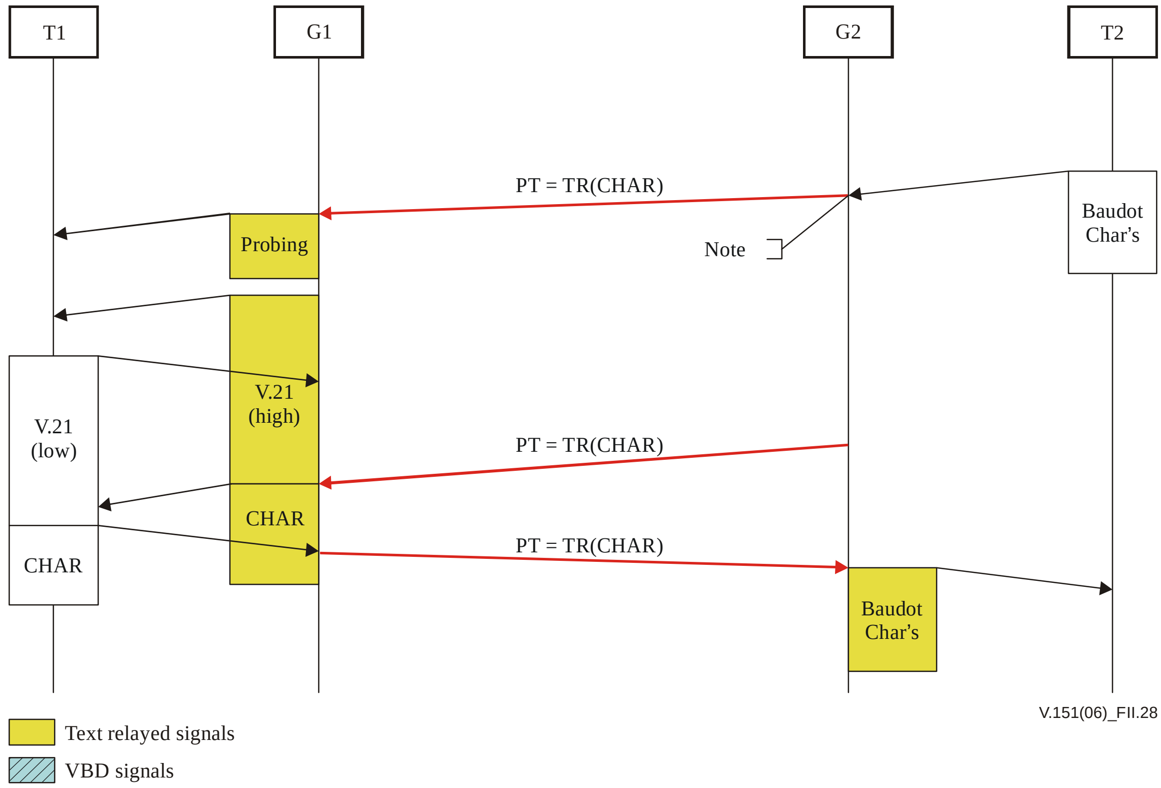
II.4.17 Scenario #29

T1 = FDX PTP device (V.21 in this example)

T2 = HDX PTP device (Baudot in this example)

G1 = supports T1's modulation

G2 = supports T2's modulation



NOTE – Text relay is only used if T2 is included in the modulations supported by G1. Use VBD if T2 is not included in G1's modulations.

Figure II.28 – Scenario #29 call flow diagram (Payload type)

Description

Strange restriction since not using SSEs: If T1 is supported by G1, T1 is not T2, and G1 does not support T2, then this will not connect (since G2 will try to go to VBD). It will work if SSEs are used. Restriction since we are leaning towards T1=T2 and want to go to VBD for that mode (no signalling mechanisms available to put us back into VBD).

Upon detection of the non-constant carrier signal from T2, G2 will start text relay encoding if G1 supports this modulation, otherwise G2 will initiate switch to VBD. Upon receiving text relay encoded packet, G1 will start its auto-mode probing sequence with T1.

II.4.18 Scenario #30

T1 = FDX PTP device (V.21 in this example)

T2 = HDX PTP device (Baudot in this example)

G1 = does not support T1's modulation

G2 = supports T2's modulation

Description

In this scenario, G1 does not have a modulation that is used by T1. G2, after detecting the PTP modulation from T2 that it supports and that G1 supports, will start sending text relay packets to G1. G1 will start its auto-mode probing sequence, but will eventually time out since there is no common modulation between T1 and G1. The connection will fail as it would if there was no IP network involved since T1 and T2 are not compatible devices.

II.4.19 Scenarios #31 and 32

In these scenarios, either or both of the gateways does not support the modulation of its local PTP device. Both PTP devices are not V.18 PTP devices, and one is FDX based while the other is HDX based. No successful connection will be achieved, although the channel may be transitioned to VBD mode since the gateway should detect either of the PTP signals.

II.4.20 Scenarios #33 and #36

Description

Will not connect. G2 will go into VBD mode on ANS, but never get to a character. If G1 does send a character (originator initiates text), then G1 will send PT=TR, then G2 will do FDX handshake. Implies that G2 is monitoring incoming signals, even though it has not yet sent PT=TR.

Appendix III

The use of [IETF RFC 2198] in [ITU-T V.151]

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

Redundancy is implemented using [IETF RFC 2198]. Annex E provides further clarification of how RFC 2198 redundancy can be used for the ToIP application. If using redundancy to improve performance over lossy networks, gateways should implement redundancy as described in these RFCs. The exception is that the buffer time for sending packets with redundant block and no primary blocks at the end of a text spurt should be 300 ms.

Appendix IV

Text buffering and transmission

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

When using the procedures in this Recommendation, text transmission may be negotiated with different speeds for different endpoints. There may also be cases when one leg of the call has successfully negotiated and connected and starts to handle text, while the other has not completed text negotiation.

Both these situations require a buffer for text to be maintained in the gateways. There is no flow control available for PSTN text telephony, so human habits will be the limiting factor for the gateway buffer.

The transport methods have an out-of-band signalling parameter for character rate that can control where the buffer needs to be located.

A general rule is that a user will not create continuous output for more than one minute before expecting a response or making a pause. However, if input is by means of voice-to-text technologies at the maximum character rate of 30 characters per second, that means there is an input of 1800 characters. If the output is to the slowest possible text telephone, the US Baudot method, performing 6 characters per second, the buffer will build up 1800 – 360 = 1440 characters.

This calculation leads to a general indication that a gateway buffer of 2 kbytes should be sufficient.

Once connected through a gateway, characters shall be transmitted between terminals involved in the call using negotiated protocols and buffering strategies.

Appendix V

Probing sequence

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

Appendix I of [ITU-T V.18] provides useful information on the ordering of automode probing sequences. These orderings are based upon the transmitting PTP supporting all of the V.18 modulations and with the *a priori* knowledge that the PTP is trying to call another PTP. These assumptions are not necessarily the case for media gateways. Once being given some indication that the gateway is to establish a connection for the purposes of connecting two PTPs, a gateway has to discover whether the endpoint on its PSTN connection is a PTP by the generation of probing sequences. It is the response to these probes that assist in the discrimination. A gateway does not have the geographical advantage either, since a gateway may be physically located in a different country or area than the PTP to which it is connected. Consequently, depending upon circumstances, it is possible that either one or two complete automode probing cycles may occur before being able to discriminate a PTP type. The amount of time to perform this discovery if a full V.18 implementation is used may take longer than one minute. This is not desirable from a user satisfaction perspective, and could lead to larger instances of call failure for the IP network than that of the legacy PSTN. If connect times at the remote gateway/PTP connection take a significant amount of time, the calling side user may start typing instead of waiting for the answering user to start typing, possibly causing the calling PTP device to start carrier before receiving any answering signals.

An alternative approach that can be utilized by gateways is to use an adaptive automode probing sequence ordering. This approach can be used by gateways to order their automode probing sequences based upon what modulations they support and what information they are provided by a peer gateway. This information could be provided by the optional use of SSEs.

For typical usage scenarios, the majority of PTP connections will be between like PTP devices. That is expected because the call will be established between two endpoints that have already been connected on previous occasions or the number of the PTP being called is a known PTP endpoint. Examples of known PTP endpoints include government services, relay services, and published numbers. This type of information can be used to determine the best ordering of the automode sequence.

There are a variety of methods and criteria that can be used when defining and using the automode probing sequences. These guidelines recommend that such criteria be determined by typical usage and application of PTP than an inflexible fixed sequence order.

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