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Procedures for real-time Group 3 facsimile communication over IP networks

ITU-T Recommendation T.38

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Summary

This Recommendation defines the procedures to be applied to allow Group 3 facsimile transmission between terminals where in addition to the PSTN or ISDN a portion of the transmission path used between terminals includes an IP network, e.g. the Internet.

Source

ITU-T Recommendation T.38 was prepared by ITU-T Study Group 16 (2001-2004) and approved under the WTSA Resolution 1 procedure on 29 March 2002.

FOREWORD

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

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

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

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

NOTE

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

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As of the date of approval of this Recommendation, ITU had not 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 real-time Group 3 facsimile communication over IP networks

1 Scope

This Recommendation defines the procedures to be applied to allow Group 3 facsimile transmission between terminals where in addition to the PSTN or ISDN a portion of the transmission path used between terminals includes an IP network, e.g. the Internet.

2 Normative references

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

- ITU-T Recommendation F.185 (1998), *Internet facsimile: Guidelines for the support of the communication of facsimile documents.*
- ITU-T Recommendation H.225.0 (1998), *Call signalling protocols and media stream packetization for packet-based multimedia communication systems*.
- ITU-T Recommendation H.248.1 (2002), *Gateway control protocol*, *Version 1*.
- ITU-T Recommendation H.248.2 (2000), *Gateway control protocol*: *Facsimile, text conversation and call discrimination packages.*
- ITU-T Recommendation H.323 (2000), *Packet-based multimedia communications systems.*
- ITU-T Recommendation Q.850 (1993), *Usage of cause and location in the Digital Subscriber Signalling System No. 1 and the Signalling System No. 7 ISDN User Part*.
- ITU-T Recommendation T.4 (1996), *Standardization of Group 3 facsimile terminals for document transmission.*
- ITU-T Recommendation T.6 (1988), *Facsimile coding schemes and coding control functions for Group 4 facsimile apparatus.*
- ITU-T Recommendation T.30 (1996), *Procedures for document facsimile transmission in the general switched telephone network.*
- ITU-T Recommendation X.680 (1997), *Information technology – Abstract Syntax Notation One (ASN.1): Specification of basic notation.*
- ITU-T Recommendation X.691 (1997), *Information technology – ASN.1 encoding rules – Specification of Packet Encoding Rules (PER).*
- IETF RFC 2327 (1998), *SDP: Session Description Protocol*.
- IETF RFC 2543 (1999), *SIP: Session Initiation Protocol*.
- IETF RFC 768 (1980), *User Datagram Protocol.*
- IETF RFC 791 (1981), *Internet Protocol – DARPA Internet Program – Protocol Specification.*
- IETF RFC 793 (1981), *Transmission Control Protocol – DARPA Internet Program – Protocol Specification.*

– IETF RFC 1006 (1987), *ISO transport services on top of the TCP*.

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

3 Definitions

Unless otherwise noted, the definitions in ITU-T Rec. F.185 shall apply. This Recommendation defines the following terms:

3.1 emitting gateway: The IFP peer which initiates IFT service for a calling G3FE. It initiates a TCP or UDP connection to a receiving gateway to begin an IFT session.

3.2 receiving gateway: The IFP peer which accepts a TCP or UDP connection from an emitting gateway, providing IFT service to a called G3FE.

3.3 G3 facsimile equipment (G3FE): In this Recommendation, G3FE refers to any entity which presents a communications interface conforming to ITU-T Recs T.30, T.4, and, optionally, T.6. A G3FE may be a traditional G3 facsimile machine, an application with a T.30 protocol engine, or any of the other possibilities mentioned in the network model for IP Facsimile.

4 Abbreviations

This Recommendation uses the following abbreviations:

ECM	Error Correction Mode
IAF	Internet Aware Fax device
IFP	Internet Facsimile Protocol
IFT	Internet Facsimile Transfer
IP	Internet Protocol
LSB	Least Significant Bit
MSB	Most Significant Bit
TCF	Training check
TCP	Transmission Control Protocol
TPKT	Transport Protocol Data Unit Packet
UDP	User Datagram Protocol
UDPTL	Facsimile UDP Transport Layer protocol
SUB	Sub-address

5 Introduction

The availability of IP networks such as the Internet for international communication provides the potential for utilizing this transmission medium in the transfer of Group 3 facsimile messages between terminals. Since the characteristics of IP networks differ from those provided by the PSTN or ISDN, some additional provisions need to be standardized to maintain successful facsimile operation.

The protocol defined in this Recommendation specifies the messages and data exchanged between facsimile gateways and/or IAFs connected via an IP network. The reference model for this Recommendation is shown in Figure 1.

This model shows a traditional Group 3 facsimile terminal connected to a gateway emitting a facsimile through an IP network to a receiving gateway which makes a PSTN call to the called Group 3 facsimile equipment. Once the PSTN calls are established on both ends, the two Group 3 terminals are virtually linked. All standard T.30 session establishment and capabilities negotiation is carried out between the terminals. TCF is either generated locally or transferred between the terminals, depending on the mode of operation to synchronize modulation rates between the gateways and G3FEs.

An alternate scenario would be a connection to a facsimile-enabled device (for example, a PC) which is directly connected to an IP network. In this case, there is a virtual receiving gateway as part of the device's facsimile-enabling software and/or hardware. In other environments, the roles could be reversed, or there might be two facsimile-enabled network devices. The protocol defined by this Recommendation operates directly between the emitting and receiving gateways. Communication between the gateways and facsimile terminals and/or other devices is outside the scope of this Recommendation.

The protocol defined in this Recommendation was chosen on the basis of efficiency and economy. For optimum performance, the IP transmission paths should have reasonably low delays to meet the F.185 requirements. Good image quality is provided by error control in the network in addition to the means provided by the T.30 protocol.

Reliable data transport is provided in two ways: by using TCP over IP networks, or by using UDP over IP networks with optional means for error control. H.323 systems may utilize either method as described in Annex D/H.323. The H.323 environment is being used to support voice transmission over IP as an alternative to the PSTN. Since facsimile generally uses the same facilities as voice communications, it may be desirable to utilize the H.323 environment when implementing facsimile over IP.

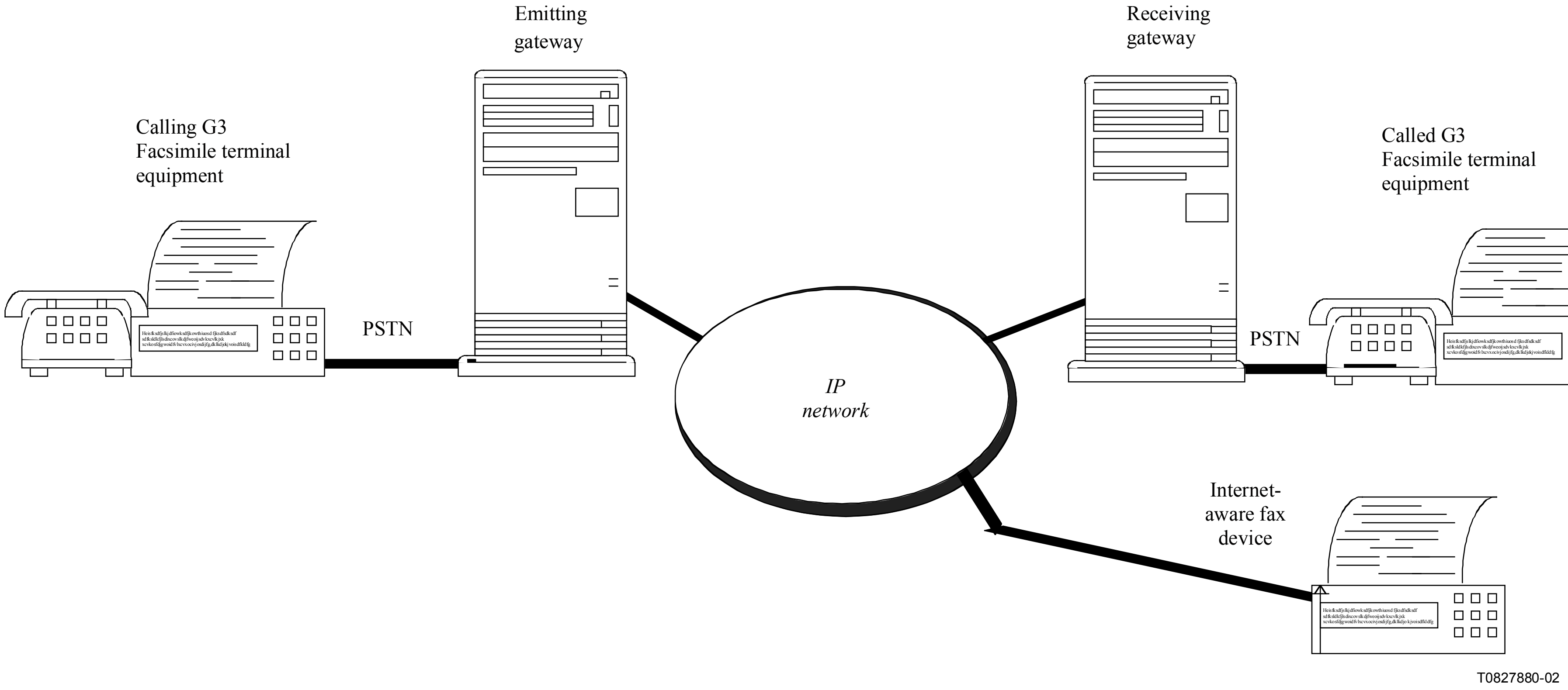


Figure 1/T.38 – Model for facsimile transmission over IP Networks

Under some circumstances it may be necessary to make some adjustments to the procedures between the gateway and the Group 3 terminal. Any such adjustments should not go beyond those available in the T.30 protocol. These adjustments are implementation-dependent.

The protocol defined in this Recommendation focuses on the interval where a network connection has been established between two peers (gateway or IAF) implementing the Real-Time Facsimile document transfer over Internet Protocol.

Management issues, such as directory services (converting PSTN numbers to IP addresses when required), network hunting, user authentication and CDR (Call Detail Record) collection and network management (SNMP or others) are important but are not addressed in this Recommendation. Standardization of these issues will allow the implementation of a network based on third party management devices, including sharing such devices with other Internet gateways such as Internet telephony and video, remote access and e-mail.

In addition, user interface aspects, such as the way that the facsimile operator selects the PSTN number of the destination or identifies himself to the system (for security purposes) are also not in the scope of this Recommendation. However, it is reasonable to assume that the facsimile operator uses the Group 3 terminal equipment keypad (using DTMF signals) or the IAF keyboard to provide the gateway with the required information.

Some of these issues mentioned here are being addressed in other ITU-T Recommendations. Specifically, ITU-T Recs H.323/H.225.0 and the Gatekeeper Recommendations address some of the above-mentioned dependencies.

It is intended that all procedures in this Recommendation conform to the requirements of ITU-T Rec. F.185.

The main body of this Recommendation describes the protocol and communication procedures between the emitting gateway and the receiving gateway. Communication between the gateways and the calling and called G3FEs as well as call control procedures are described in Annex B.

Additional methods for call establishment over IP networks are described in Annexes D and E.

T.38 version numbers

Version number	Content summary
0	Annex B
0	Annex D, E, Appendix II
1	TPKT, IAF Support, Amendment to Annex D, E, Appendix III
2	TCP Startup, Mandatory T.30 Indicator

6 Communication between gateways

6.1 Internet protocol – TCP or UDP

The public Internet service provides two principal modes of data transmission:

- TCP (Transmission Control Protocol) – A session-based, confirmed delivery service;
- UDP (User Datagram Protocol) – Datagram service, non-confirmed delivery.

This Recommendation allows the use of either TCP or UDP depending on the service environment. It defines a layered protocol such that the T.38 messages exchanged for TCP and UDP implementations are identical.

6.2 Gateway facsimile data transfer functions

The emitting gateway shall demodulate the T.30 transmission received from the calling terminal. The T.30 facsimile control and image data shall be transferred in an octet stream structure using the IFP packets, over a transport protocol (TCP or UDP). The following signals are not transferred between gateways but are generated or handled locally between the gateway and the G3FE: CNG, CED, and in one mode, TCF. The gateways may indicate the detection of the tonal signals CNG and CED so that the other gateway can generate them.

The receiving gateway shall decode the transferred information and establish communication with the called facsimile terminal using normal T.30 procedures. The receiving gateway shall forward all relevant responses from the called terminal to the emitting gateway.

The facsimile data transfer structure is described in 7.1.3. The flow between gateways is described in clause 8.

6.2.1 Treatment of non-standard facilities requests

The emitting gateway may optionally ignore NSF, NCS and NSS, take appropriate action or pass the information to the receiving gateway. The receiving gateway may optionally ignore NSF, NCS and NSS or take appropriate action including passing the information to the receiving G3FE. Information in other frames related directly to these frames may be altered by the gateway.

7 IFT protocol definition and procedures

7.1 General

This clause contains the textual description of the IFT protocol. The IFT protocol is specified by the ASN.1 description in Annex A. In the case of a conflict between the ASN.1 and the text, the ASN.1 governs. The ASN.1 encoding in Annex A should employ the BASIC-ALIGNED version of Packed Encoding Rules (PER) according to ITU-T Rec. X.691.

7.1.1 Bit and octet transmission order

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

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

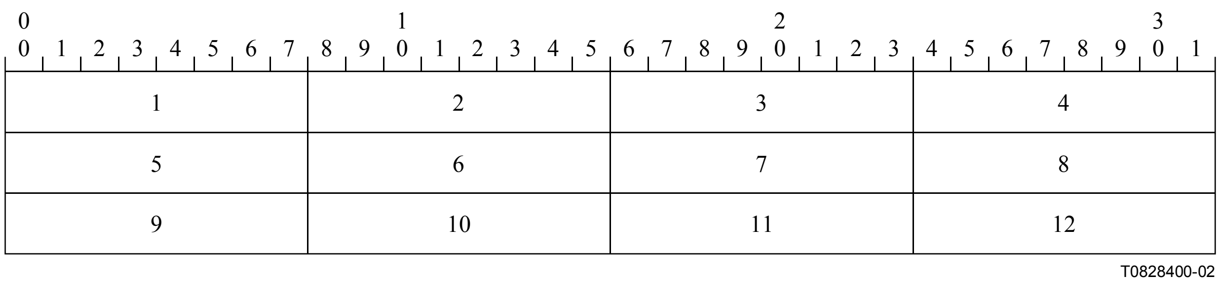


Figure 2/T.38 – Transmission order of octets (based on RFC 791, Figure 10)

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

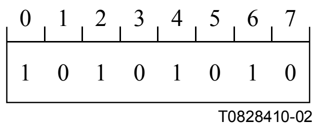


Figure 3/T.38 – Significance of bit (based on RFC 791, Figure 11)

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

7.1.2 Mapping of the T.30 bit stream

The T.30 bit stream is mapped so that *bit* order is maintained between the PSTN and IP networks. This means that the first bit transmitted is stored in the MSB of the first octet, where the MSB is defined as in 7.1.1.

7.1.3 IFP packet layers for TCP/IP and UDP/IP

The IFP packets described in 7.2 are combined with the appropriate headers for TCP/IP and UDP/IP as shown in Figures 4 and 5. In Figure 4, the UDPTL header represents the additional header information required for error control over UDP. To provide interoperability in H.323 environments, the TPKT header defined in RFC 1006 shall precede the IFP Packet in TCP implementations as shown in Figure 4. Implementations using TPKT shall set the version to 1 or higher.

NOTE – Implementations of T.38 with TCP/IP transport prior to version 1 were not required to support the TPKT facility.

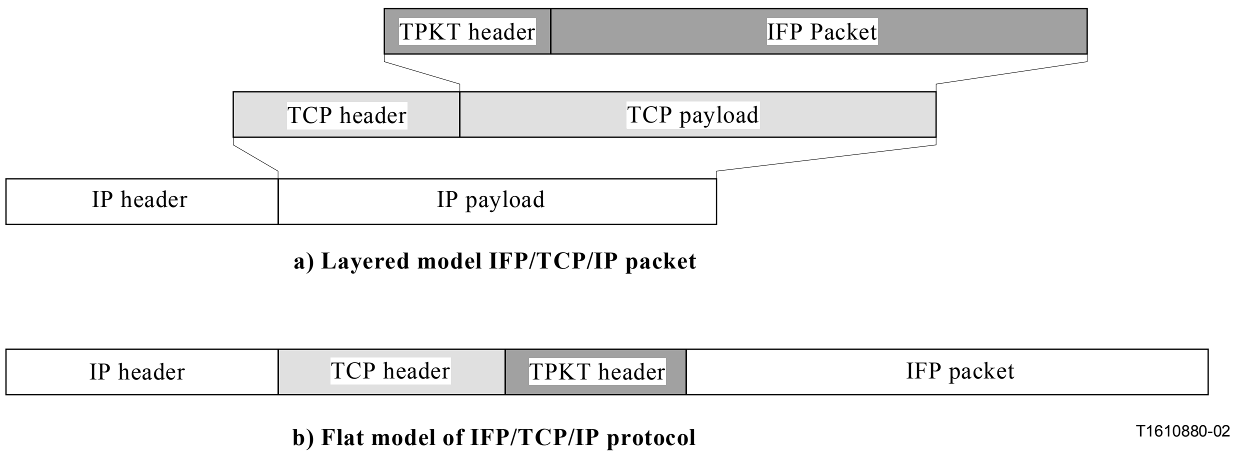


Figure 4/T.38 – High-level TCP/TPKT/IP packet structure

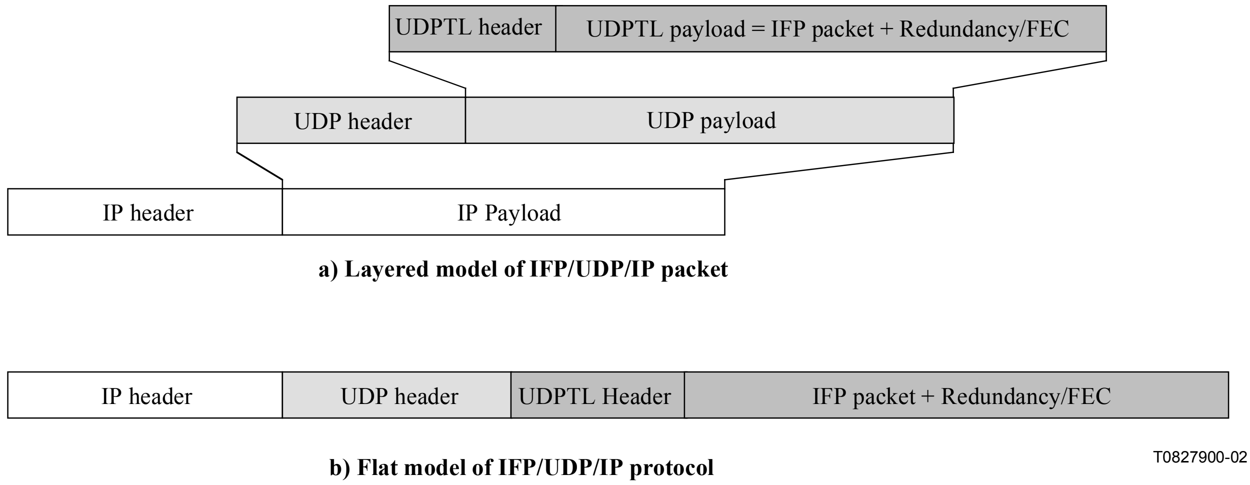


Figure 5/T.38 – High-level UDPTL/IP packet

7.2 IFP packet format

In the following discussion, a message is the protocol or data information transferred in one direction from a G3FE to or from a gateway during a single period. It may include, for example, one or more HDLC frames, or a "page" of Phase C data. Messages may be sent across the IP network in multiple packets. The packets may, for example, contain partial or full, singular or multiple HDLC frames. Support for multiple packets is provided in this protocol. The DATA element uses Fields to support partial and full HDLC frames.

IFP operates (listens) over TCP/IP or UDP/IP using a port determined during call setup. All communication between IFP Peers is done using packets, identified as IFPPackets.

Table 1 summarizes the IFPPackets (for full explanation, refer to the following subclauses).

Table 1/T.38 – IFP packet elements

Field	Description
TYPE	Type of message
DATA	Dependent on TYPE

7.2.1 T.38 packet

The T.38 packet element provides an alert for the start of a message. It is used by the IFP peer to verify message alignment. It is identified by an ASN.1 Application tag. When data is read by the peer from its TCP/IP or UDP/IP stack, and the expected tag is not present, the session should be immediately aborted by the receiver.

7.2.2 TYPE

The TYPE element describes the function of, and optionally, the data of the packet. The legitimate TYPEs are given in Table 2. Each TYPE is separately explained in the following subclauses. The table also indicates whether the TYPEs are Mandatory or Optional for implementations using TCP and UDP.

If the TYPE element is not recognized, it and the related data element shall be ignored.

Table 2/T.38 – IFP packet TYPE field

Type	DATA Type	Mandatory	Description
T30_INDICATOR	Regular	Yes	Carries indication about the presence of a facsimile signal (CED/CNG), preamble flags or modulation indications
T30_ DATA	Field	Yes	T.30 HDLC Control and Phase C data (e.g. T.4/ T.6 image segment.)
NOTE – If both G3FE devices are identified via DIS/DCS exchange as Internet Aware Fax (IAF) devices, the use of T30_INDICATOR is optional.			

7.2.3 DATA-Field

The DATA-Field element contains the T.30 HDLC control data and the Phase C image (or BFT) data. The structure of the DATA-Field is defined in 7.4. The structure carries the modulation data as well as indicators for the end of an HDLC frame, the status of the Frame Check Sequence (FCS) for an HDLC frame, and whether the data represents the end of a message.

7.3 TYPE definitions

The following subclauses describe the message TYPEs.

7.3.1 T30_INDICATOR

The T30_INDICATOR TYPE is used by the gateways to indicate the detection of signals such as CED, HDLC preamble flags, and modem modulation training. It is sent by the receiving gateway to the emitting gateway, and by the emitting gateway to the receiving gateway. The use of this message is mandatory, except in the case where both G3FE devices are identified via DIS/DCS exchange as Internet Aware Fax Devices. A peer may send this message in order to notify its peer about upcoming messages. The T30_INDICATOR TYPE has one of the following values (see Table 3):

Table 3/T.38 – Listing of T30_INDICATOR values

Signal/Indication
No signal
CNG (1100 Hz)
CED (2100 Hz)
V.21 Preamble Flags
V.27 2400 modulation training
V.27 4800 modulation training
V.29 7200 modulation training
V.29 9600 modulation training
V.17 7200 modulation short training
V.17 7200 modulation long training
V.17 9600 modulation short training
V.17 9600 modulation long training
V.17 12 000 modulation short training
V.17 12 000 modulation long training
V.17 14 400 modulation short training
V.17 14 400 modulation long training

NOTE – It is the responsibility of the gateway receiving the indicator to properly generate the appropriate analogue signal, including, for example, the ON-OFF cadence, and to terminate it appropriately.

7.3.2 T30_DATA TYPE

The T30_DATA TYPE is used to indicate that the packet contains data in the DATA element and what modulation was used to carry the data. The T30_DATA TYPE is used to indicate both HDLC control data as well as any Phase C data (T.4/T.6 or other). It has the following values (see Table 4):

Table 4/T.38 – Listing of T30_DATA values

Modulation
V.21 Channel 2
V.27 *ter* 2400
V.27 *ter* 4800
V.29 7200
V.29 9600
V.17 7200
V.17 9600
V.17 12 000
V.17 14 400

NOTE – If both G3FE devices are identified via DIS/DCS exchange as IAF devices, T30_DATA values shall be ignored.

7.4 The IFP DATA element

The DATA Element of the IFP packets contains the data from the PSTN connections and some indication of the data format. The DATA element is a structure containing one or more Fields. Each Field has two parts: the first part indicates the Field-Type, the second part contains the Field-Data. The meanings for the Field-Type are shown in Table 5.

Table 5/T.38 – Field-type and field-data description

Field-type	Field-type description
HDLC data	Data transmitted over the PSTN connection as HDLC. This includes the T.30 control messages as well as Phase C data sent using ECM.
	The Field-Data which follows contains some, or all, of a single HDLC data frame starting with the address frame of the HDLC frame, up to but not including FCS. Bit stuffing is removed from all data. The end of a frame is indicated by the FCS Indicator field. The gateway is responsible for bit stuffing, FCS generation, and separating frames with one or more flags (0x7E) when sending the HDLC data to a G3FE. The FCS-xx-Sig-End Fields indicate the end of the final frame.
HDLC-Sig-End	Indicates that the HDLC power level has dropped below the turnoff threshold. There is no Field-Data with this Field-Type.
HDLC-FCS-OK	Indicates the end of an HDLC frame and that the proper FCS was received. It also indicates that this frame is not the final frame. There is no Field-Data with this Field-Type.
HDLC-FCS-Bad	Indicates the end of an HDLC frame and that the proper FCS was not received. It also indicates that this frame is not the final frame. There is no Field-Data with this Field-Type.

Table 5/T.38 – Field-type and field-data description

Field-type	Field-type description
HDLC-FCS-OK-Sig-End	Indicates the end of an HDLC frame and that the proper FCS was received. It also indicates that this frame is the final frame. There is no Field-Data with this Field-Type.
HDLC-FCS-BAD-Sig-End	Indicates the end of an HDLC frame and that the proper FCS was not received. It also indicates that this frame is the final frame. There is no Field-Data with this Field-Type.
T.4-Non-ECM	T.4 Phase C data that is not sent using ECM or TCF data in the case of Method 2 of Rate Adaptation. It also indicates that this is not the end of the Phase C data.
	The Field-Data which follows is the demodulated Phase C data, including fill bits and RTC.
T.4-Non-ECM-Sig-End	T.4 phase C data that is not sent using ECM or TCF data in the case of Method 2 of Rate Adaptation. It also indicates that this is the end of the Phase C data.
	The Field-Data which follows is the demodulated Phase C data, including fill bits and RTC.

Multiple fields can appear in a single IFP DATA Element. The example below shows two HDLC frames arranged in a single DATA Element.

Field-Type	HDLC-Data	FCS-OK	HDLC-Data	FCS-OK-Sig-End
Field part description	First HDLC frame. The HDLC octets with zero stuffing and FCS removed in the Field-Data.	Indicates end of HDLC frame and more data to follow	Second HDLC frame	Indicates end of HDLC frame and end of HDLC data

NOTE – When the field-type DATA element is received, the receiver should analyze it by examining each field separately. If the receiver does not recognize a certain FIELD-TYPE of the field it is examining, the entire field shall be skipped, and the receiver shall continue with the next field.

The IFP peer may elect to send the message data in several packets. Although relatively large packets may be sent, smaller data packets are recommended. It is entirely up to the emitting gateway to decide on the size of packets being sent. The xx-Sig-End Field-Types indicate the end of the message data. Note that for each packet sent, the whole header is repeated.

A message with zero length data field may be sent to indicate, as early as possible, that T30_DATA messages are coming. Alternately, the appropriate T30_INDICATOR signal for High Speed could be sent. Implementations shall support both methods.

Partial HDLC frames are also supported. The next example shows how two HDLC frames would be transmitted using three consecutive IFP packets. (Data transport headers are not shown.)

TYPE element	DATA element								
V.21 Data	Field-Type: HDLC Data	HDLC Address (0xff)	HDLC Control	HDLC Octet 1	HDLC Octet 2	HDLC Octet 3	HDLC Octet 4	HDLC Octet 5	HDLC Octet 6
V.21 Data	Field-Type: HDLC Data	HDLC Octet 7	HDLC Octet 8	HDLC Octet 9	Field-Type FCS-OK				
V.21 Data	Field-Type: HDLC Data	HDLC Address (0xff)	HDLC Control	HDLC Octet 1	Field-Type FCS-OK-Sig-End				

8 IFP message flow for facsimile rates up To V.17

The gateways follow the T.30 message flow and use the packet format in clause 7 to transmit these messages. This means, for example, that error correction in ECM mode is done between the sending G3FE and the receiving G3FE. The PPS, PPR, etc. signals are sent between the end G3FE devices. In another example, negotiation of security keys, etc. as proscribed in Annex H/T.30 is done between the end G3FE devices. Examples of typical message flows are shown in Appendix I.

There are two methods of handling the TCF signal for determining the high speed data rate. Either of these methods ensures that both PSTN facsimile sessions be conducted at the same speed.

8.1 Data rate management method 1

Method 1 of data rate management requires that the TCF training signal be generated locally by the receiving gateway. Data rate management is performed by the emitting gateway based on training results from both PSTN connections.

Method 1 is used for TCP implementations and is optional for UDP implementations.

When a CFR (Confirmation to receive) or an FTT (failure to train) is received from a G3FE at the receiving gateway, a T.30 HDLC packet (indicating CFR or FTT respectively) should be forwarded to the emitting gateway.

Based on the results of a TCF received from a G3FE and the T.30 HDLC packet (CFR or FTT) forwarded from a receiving gateway, an emitting gateway shall transmit FTT or CFR according to Table 6.

Table 6/T.38 – Decision table of signalling rate of an emitting gateway

T.30 signal message forwarded from receiving gateway	TCF signal received from a G3FE at emitting gateway	Signal to be transmitted to G3FE (emitter)
CFR	Success	CFR
FTT	Success	FTT
CFR	Failure	FTT
FTT	Failure	FTT

In the case where the Emitting Device is an Internet Aware Fax (IAF) device and there is no Emitting Gateway, the IAF device shall respond to FTTs from the Receiving Gateway with appropriate DCS responses, including possibly modulation changes.

In the case where the Receiving Device is an IAF device and there is no Receiving Gateway, the IAF device shall respond to DCS from the Emitting Gateway with CFR, but shall be prepared for a DCS in case the Emitting Gateway generates an FTT.

In the case where the Emitting Device and the Receiving Device are IAF devices, the Emitting Device shall send DCS with the modulation bits set to 0, and the Receiving Device shall respond with CFR. The data rate over the IP network is established during Call Setup.

8.2 Data rate management method 2

Data rate management method 2 requires that the TCF be transferred from the sending G3FE to the receiving G3FE rather than having the receiving gateway generate it locally. Speed selection is done by the G3FEs in the same way as they would on a regular PSTN connection.

In the case where the Emitting Device is an Internet Aware Fax (IAF) device and there is no Emitting Gateway, the IAF device shall respond to FTTs from the Receiving Gateway with appropriate DCS + TCF responses, including possibly modulation changes.

In the case where the Receiving Device is an IAF device and there is no Receiving Gateway, the IAF device shall respond to DCS from the Emitting Gateway with either CFR or FTT, depending upon the received TCF signal.

In the case where the Emitting Device and the Receiving Device are IAF devices, the Emitting device shall send DCS with the modulation bits set to 0, and the Receiving Device shall respond with CFR. The data rate over the IP network is established during Call Setup. Data Rate Management Method 2 is mandatory for use with UDP. Method 2 is not recommended either for use with TCP, or for the case where both G3FE devices are identified via DIS/DCS exchange as IAF devices.

9 IFT over UDP transport: IFT/UDP

9.1 Overview of UDPTL protocol

In the following discussion, a packet is regarded as a block of information which has the overall structure of that presented in 7.1.3.

The layered model in Figure 5 a) may be visualised more simply [Figure 5 b)] in a flat space which allows packets to be regarded as a composite of headers plus the IFP payload. It is the IFP payload which is used to convey facsimile related information between gateways; all other information should be regarded as overhead necessary for the safe transportation and interpretation of IFP messages as described in clause 7. This clause describes the UDPTL payload. Descriptions of the IP and UDP headers and payloads are found in RFCs 791 and 768 respectively.

UDPTL packets comprise a Sequence Number and a variable length, octet aligned, payload.

UDPTL packets are based upon the principle of framing. Each packet may contain one or more IFP packets in its payload section. The first packet in any payload is always formatted in accordance with the specifications of clause 7 and must correspond to the sequence number supplied in the header (for instance, the first field in a payload with sequence number 15 must have been generated 5 payloads later than the first field in the payload with sequence number 10). The IFP packet in a UDPTL payload is referred to as the "primary". Additional fields may be included in a payload after the primary. These fields are referred to as "secondaries" and may or may not be formatted as per clause 7 specifications depending on their form.

9.2 UDPTL header section format

The UDPTL Sequence Number is used to identify the sequencing in a payload.

9.2.1 UDPTL sequence number element

Each packet, and therefore primary field, has its own corresponding unique sequence number which specifies an ordering at the receiving gateway should packets arrive out of sequence. To enable gateways to be synchronized upon receipt of any packet, the first primary field transmitted shall have sequence number zero. Successive primaries shall have linearly increasing (integer adjacent) sequence numbers.

9.3 UDPTL payload section format

During H.323 capabilities exchange, a gateway shall indicate its support of the available error protection schemes, parity FEC, or redundancy. Based on these capabilities, a choice may be made on which scheme is used for error protection. If a capability is indicated to receive both parity error correction frames and redundant frames, then either scheme may be used. If, however, a gateway indicates a capability to receive only redundant error protection frames, then the transmitting gateway may not send parity FEC frames. The support of parity FEC is optional; a gateway providing parity FEC receive services should, however, also be capable of receiving redundant messages.

The IFP payload section comprises one or more fields. The basic format of an UDPTL payload is as shown in Figure 6.

Figure 6 specifies the order in which different messages are to be assembled into the UDPTL payload. It is invalid to transmit both redundant and FEC fields within the same packet.

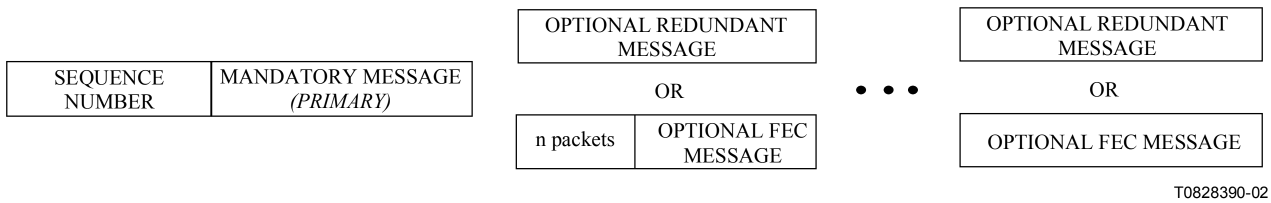


Figure 6/T.38 – Basic format of UDPTL Payload Section (UDP header not shown)

9.3.1 UDPTL FEC message format

An FEC contains a parity encoded representation of a number of primaries. The number of primary IPF packets represented by an FEC field is given by the fec-n-packets element of the UDPTLPacket.

9.4 IFP/UDP facsimile data transfer functions

9.4.1 Use of redundancy messages

Each primary contains an IFP Packet. As packets, and therefore primaries, are assigned unique and linearly increasing sequence numbers, receiving gateways can detect packet loss and re-sequencing requirements. By imposing a simple structure it is possible to provide error recovery by means of transmitting redundant information in the form of prior primary packets within each payload. The strategy used is to assemble an additional *n* prior packets after the primary with monotonically decreasing sequence numbers. Thus, should each payload contain a primary and two or more secondary fields, a loss of two consecutive UDPTL packets will be protected against. In order to provide a redundancy service in the UDPTL, it is necessary to maintain a buffer of "old" primaries for assembly into new packets. An illustration of such a buffer is provided in Figure 7 to demonstrate the principles of redundancy transfer by example.

Note that the UDPTL scheme is capable only of transmitting a block of redundant IFP packets whose sequence numbers are contiguous. Thus, if the current IFP packet has sequence number *C* and it is desired to redundantly transmit the IFP packet from UDPTL packet sequence number *C*-2, then the UDPTL packet must contain all the IFP packets from *C*, *C*-1, *C*-2 in the order given.

Gateways need not be capable of transmitting redundant packets. Receiving gateways may ignore them.

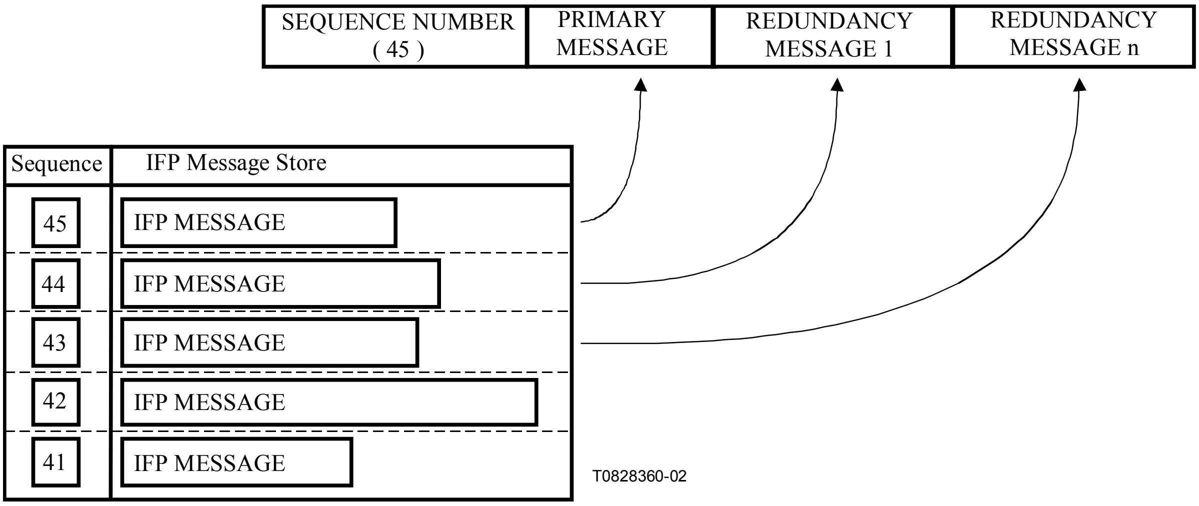


Figure 7/T.38 – Including prior (secondary) IFP packets (fields) into an UDPTL packet

Annex A

ASN.1 notation

A.1 ASN.1 notation

```
T38 DEFINITIONS AUTOMATIC TAGS ::=
BEGIN
IFPPacket ::= SEQUENCE
{
     type-of-msg   Type-of-msg,
     data-field         Data-Field OPTIONAL
}
Type-of-msg ::= CHOICE
{
     t30-indicator ENUMERATED
     {
       no-signal,
       cng,
       ced,
       v21-preamble,
       v27-2400-training,
       v27-4800-training,
       v29-7200-training,
       v29-9600-training,
       v17-7200-short-training,
```

```
        v17-7200-long-training,
        v17-9600-short-training,
        v17-9600-long-training,
        v17-12000-short-training,
        v17-12000-long-training,
        v17-14400-short-training,
        v17-14400-long-training,
        …
      },
      data ENUMERATED
      {
        v21,
        v27-2400,
        v27-4800,
        v29-7200,
        v29-9600,
        v17-7200,
        v17-9600,
        v17-12000,
        v17-14400,
        …
      }
}
Data-Field ::= SEQUENCE OF SEQUENCE
{
      field-type  ENUMERATED
      {
        hdlc-data,
        hdlc-sig-end,
        hdlc-fcs-OK,
        hdlc-fcs-BAD,
        hdlc-fcs-OK-sig-end,
        hdlc-fcs-BAD-sig-end,
        t4-non-ecm-data,
        t4-non-ecm-sig-end,
        …
      },
      field-data  OCTET STRING (SIZE(1..65535)) OPTIONAL
}
UDPTLPacket ::= SEQUENCE
{
      seq-number          INTEGER (0..65535),
      primary-ifp-packet TYPE-IDENTIFIER.&Type(IFPPacket),
      error-recovery CHOICE
      {
      secondary-ifp-packets SEQUENCE OF TYPE-IDENTIFIER.&Type(IFPPacket),
        fec-info          SEQUENCE
        {
           fec-npackets  INTEGER,
           fec-data  SEQUENCE OF OCTET STRING
        }
      }
}
END
```

Annex B

H.323 call establishment procedures

B.1 Introduction

This annex describes system level requirements and procedures for Internet-aware facsimile implementations and Internet-aware facsimile gateways conforming to ITU-T Rec. T.38 to establish calls with other ITU-T T.38 implementations including those using the procedures defined in this annex as well as Annex D/H.323.

B.2 Communication between facsimile terminal and gateway

Communication between a sending Group 3 facsimile terminal and the incoming gateway is generally effected using dial-up procedures over the PSTN. Basic and optional T.30 procedures are supported. The support for V.34 is for further study.

The gateway may receive the facsimile transmission from the calling terminal as a modem signal on the PSTN if the gateway supports a direct dial-in procedure. Where the gateway is located within the network, it may receive the transmission in the form of a PCM encoded digital channel. Internet-aware facsimile (IAF) implementations are connected directly to the IP network and act as a gateway for call establishment.

B.2.1 Transfer of addressing information

The conveyance of the E.164 address of the called terminal from the calling terminal to the emitting gateway may be by manual procedures using prompts; by means of double dialling; or by any other suitable means. In addition, there are some applications which may benefit from placing the destination E.164 address in the IRA (Internet Routing Address)/ISP (Internet Selective Polling) signals, as described in ITU-T Rec. T.30.

B.3 Communication between gateways

B.3.1 Overview

B.3.1.1 Call setup

Call setup for T.38 Annex B compliant implementations is based on the Fast Connect Procedure defined in ITU-T Rec. H.323. T.38 implementations may operate in two distinct H.323 compatible environments.

1) A facsimile-only over IP environment. In this environment, no voice support is provided. The procedures and requirements of this annex shall apply to implementations operating in this environment unless they are superseded by an H.323 Annex D implementation.

2) A facsimile and voice over IP environment. Implementations in this environment shall use the methods described in Annex D/H.323.

T.38 Annex B implementations use only the Fast Connect Procedure for call setup and do not support H.245 negotiation. H.323 Annex D implementations, on the other hand, support both the Fast Connect Procedure and the normal H.323 procedure for call setup. Most H.323 implementations also support H.245 negotiation.

B.3.1.2 Media channels

ITU-T Rec. H.225.0 requires that T.38 facsimile packets are sent on a separate TCP/UDP port from H.225.0 call signalling (TCP). All required ports are established during the initial **fastStart** exchange. A minimal T.38 Annex B implementation requires a TCP port for call signalling and either a UDP port or a TCP port for T.38 facsimile information.

B.3.1.3 Usage of ITU-T Rec. H.245

Endpoints conforming to this annex are not required to support ITU-T Rec. H.245, except as required in this annex to support **fastStart** signalling. As described in B.3.9 below, an H.323 endpoint can use the *Facility* message to determine that the T.38 Annex B endpoint does not support ITU-T Rec. H.245.

B.3.2 Basic call setup

H.323 implementations have a multi-phase call setup procedure, which includes:

– RAS (Registration, Admissions and Status) signalling using UDP between the endpoint and the gatekeeper.

– Q.931-based call signalling either directly between endpoints, or between endpoints and gatekeeper depending on the call model in use, using TCP/IP.

– H.245 capability negotiation and logical channel management using TCP/IP.

Although support for RAS is mandatory, it is not mandatory to use RAS. Thus, an Annex B implementation could be used with or without a gatekeeper. It could obtain its IP addresses in any fashion desired, such as LDAP or a personal directory. However, if placed in a gatekeeper environment, it would register and operate as per ITU-T Rec. H.323.

Implementations conforming to this annex shall conform to H.323 RAS signalling. RAS signalling allows a T.38 implementation to initiate a call, using the H.323 well-known TCP port, and provides dynamic assignment of the port to use for the T.38 messages.

Implementations conforming to this annex utilize H.323 call setup messages as described in 8.1.1/H.323: "Basic Call Setup – Neither Endpoint Registered" assuming this to be the case. The initial text of 8.1/H.323: "Phase A – Call Setup" is also relevant to T.38 implementations. The rest of 8.1/H.323 applies if one or both endpoints are registered with a gatekeeper.

Implementations conforming to this annex shall initially start calls by opening a TCP/IP session and sending an H.225.0 SETUP message with the fast connect fields filled in as described in 8.1.7/H.323.

The receiving terminal replies with an H.225.0 ALERTING, CALL PROCEEDING, PROGRESS, or CONNECT message as per the procedures of ITU-T Rec. H.323 "fast connect". The Annex B implementation shall not include any video, voice, or data OLC elements in the "fastStart" structure. Instead it includes OLC elements pertinent to facsimile as described in the next clause.

B.3.3 Capabilities negotiation

There are several options that need to be negotiated to determine which options the gateways support and use. See Table B.1.

Table B.1/T.38 – Gateway option capability support indications

Option	Description
Data rate management method	Method 1, local generation of TCF is required for use with TCP. Method 2, transfer of TCF is required for use with UDP. Method 2 is not recommended for use with TCP.
Data transport protocol	The emitting gateway may indicate a preference for either UDP or TCP for transport of T.38 IFP-Packets. The receiving device selects the transport protocol.
Fill bit removal	Indicates the capability to remove and insert fill bits in Phase C, non-ECM data to reduce bandwidth in the packet network. Optional. See Note.
MMR transcoding	Indicates the ability to convert to/from MMR from/to the line format for increasing the compression of the data and reducing the bandwidth in the packet network. Optional. See Note.
JBIG transcoding	Indicates the ability to convert to/from JBIG to reduce bandwidth. Optional. See Note.
Maximum buffer size	For UDP mode, this option indicates the maximum number of octets that can be stored on the remote device before an overflow condition occurs. It is the responsibility of the transmitting application to limit the transfer rate to prevent an overflow. The negotiated data rate should be used to determine the rate at which data is being removed from the buffer.
Maximum datagram size	This option indicates the maximum size of a UDPTL packet that can be accepted by the remote device.
Version	This is the version number of ITU-T Rec. T.38. New versions shall be compatible with previous versions.
NOTE – Bandwidth reduction shall only be done on suitable Phase C data, i.e. MH, MR and – in the case of transcoding to JBIG – MMR. MMR and JBIG require reliable data transport such as that provided by TCP. When transcoding is selected, it shall be applied to every suitable page in a call.	

These capabilities are negotiated using the OLC elements as defined in the T38faxProfile of H.245 V7.

Two unidirectional, reliable or unreliable, logical channels (sender to receiver channel and receiver to sender channel) as shown in Figure B.1 or, optionally, one bidirectional reliable channel as shown in Figure B.2 shall be opened for the transfer of T.38 packets. T.38 packets can be transferred using either TCP or UDP. In general, the usage of TCP is more effective when the bandwidth for facsimile communication is limited, or for IAF to IAF transfers since TCP provides flow control. On the other hand, the usage of UDP may be more effective when the bandwidth for facsimile communication is sufficient.

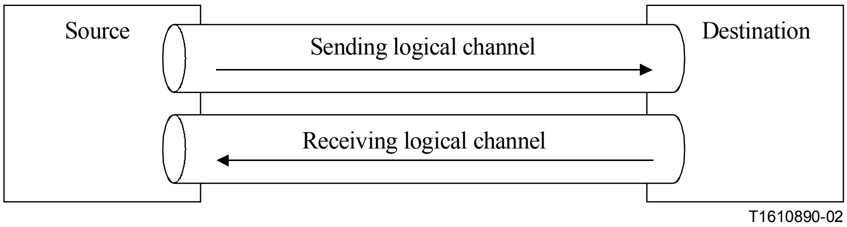


Figure B.1/T.38 – A pair of unidirectional channels

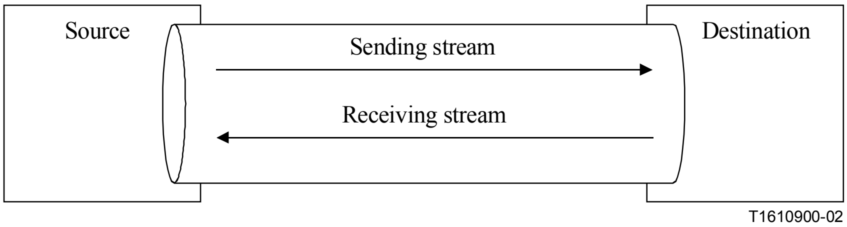


Figure B.2/T.38 – A single of bidirectional channels

The sender terminal specifies a TCP/UDP port in the **OpenLogicalChannel** in the **fastStart** element of *Setup*. The receiver terminal shall provide its TCP (or UDP) port in the **OpenLogicalChannel** of the **fastStart** element as specified by the procedures in 8.1.7/H.323: "Fast connect".

The receiver should open the TCP/UDP port based on the preference of the sender. If the sender terminal has a preference for UDP or TCP, then it shall provide its preference in the **OpenLogicalChannel** with the appropriate port in the **fastStart** sequence. The receiving terminal can select the transport, TCP or UDP, by specifying one of the two in **OpenLogicalChannel** structures in the **fastStart** element of *Connect*.

All T.38 Annex B implementations shall include a T38facsimile OLC with **udp** and **transferredTCF** set in the **fastStart** structure. Note that all H.323 Annex D devices also are required to include this structure. In addition, T.38 Annex B devices shall include an OLC with **tcp** and **localTCF** set. As described in 8.1.7/H.323, the order in which OLCs are included in the **fastStart** element indicates preference on the part of the sender. The receiver only includes the OLCs that it wishes to use in the **fastStart** element of the *Connect*.

NOTE – In the first version of Annex B, it was not possible to use a single bidirectional reliable channel. In order to retain backward compatibility, the endpoint may specify support for bidirectional reliable channels by including the **t38FaxTcpOptions** SEQUENCE and setting the **t38TCPBidirectionalMode** field to TRUE. If the other endpoint does not include the **t38FaxTcpOptions** SEQUENCE, the endpoint shall assume that a single bidirectional reliable channel for T.38 is not supported and shall use either two unidirectional reliable or unreliable channels.

B.3.4 Examples of call set-up OLCs

The examples in this clause illustrate the OLC elements that are sent in various cases. The rules of 8.1.7/H.323 are followed using OLC definitions in ITU-T Rec. H.245. Refer to ITU-T Rec. H.245 for the relevant ASN.1.

B.3.4.1 TCP, UDP support

The default case requires support for both TCP and UDP. In this case, the sender shall send OLCs for **T38/TCP&localTCF** and **T38/UDP&transferredTCF**. If the receiver wishes to use UDP, an OLC for **T38/UDP&transferredTCF** is returned; otherwise, the OLC for **T38/TCP&localTCF** is returned.

B.3.4.2 UDP with data rate management method 1 support

For the case where the sender wishes to use data rate management method 1 and UDP for data transport, it shall send OLCs for **T38/UDP&transferredTCF**, **T38/UDP&localTCF**, **T38/TCP&localTCF**. If the receiver agrees to use **UDP&localTCF**, an OLC for **T38/UDP&localTCF** is returned.

B.3.5 Mandatory call setup messages

The Annex B implementation shall support the following clauses of H.225.0 for call setup:

– Mandatory elements in Table 4/H.225.0, i.e. ALERTING, CONNECT, CALL PROCEEDING, SETUP, RELEASE COMPLETE, etc. shall be supported by T.38 endpoints conforming to Annex B. Note that there is no requirement to send ALERTING if CONNECT, CALL PROCEEDING, or RELEASE COMPLETE is sent within 4 seconds of the receipt of SETUP, as described in ITU-T Rec. H.323. Note also that gateways shall send CALL PROCEEDING.

– The Information elements of FACILITY as described in 7.4.1/H.225.0.

– The Information elements of ALERTING as described in 7.3.1/H.225.0.

– The Information elements of CALL PROCEEDING as described in 7.3.2/H.225.0.

– The Information elements of CONNECT as described in 7.3.3/H.225.0

– The information elements of PROGRESS as described in 7.3.7/H.225.0.

– The Information elements of RELEASE COMPLETE as described in 7.3.9/H.225.0.

– The Information elements of SETUP as described in 7.3.10/H.225.0.

– The ASN.1 of H.225.0 as described in ITU-T Rec. H.225.0.

NOTE – H.225.0 ASN.1 supports a large number of optional features. T.38 Annex B implementations may implement the full range of optional H.225.0 features, including authentication features that are potentially available. They may also implement H.450.x supplementary services. H.225.0 options are outside (i.e. prior) to the OLC negotiations. If a real time fax endpoint (H.323 Annex D or T.38 Annex B) makes use of H.450.x supplementary services, it must take into account that the remote endpoint may or may not support them. In the worst case, the supplementary service is ignored by the receiver. Thus the requesting endpoint must handle this condition, with, for example, a timeout mechanism.

B.3.6 Mapping of call progress signals

For call setup and call progress, the return signals can be simplified to the set shown in Table B.2. These are all returned prior to or instead of a connect message.

The CONNECT message is returned when the gateway, *by some means*, determines that a connection to the terminal G3FE has been established. If CED or FSK flags are detected, the appropriate T.38 messages can be sent. This level of call setup and progress works in both H.323 as well as non-H.323 environments.

B.3.7 Usage of the maxBitRate in messages

When TCP is used for T.38 fax transmission, **maxBitRate** in the ARQ/BRQ does not include the fax data rate. When UDP is used for T.38 fax transmission, **maxBitRate** in the ARQ/BRQ does include the bit rate needed for the fax session. The endpoint (terminal, gateway) shall send BRQs to the gatekeeper as bandwidth needs change during the call. It is noted that the **maxBitRate** in the OpenLogicalChannel element in the *Setup* during fast start is different from the **maxBitRate** in ARQ/BRQ, and does refer to the peak bit rate that the fax call will use.

B.3.8 DTMF transmission

For further study. Note that UserInputIndication as described in Annex D/H.323 is an H.245 signal. H.245 is not required for T.38 Annex B devices.

Table B.2/T.38 – Call Progress Mapping

Meaning	Mapping/Comments
Busy1. Subscriber busy tone as defined in ITU-T Rec. E.180/Q.35.	Q.850 cause value 17.
Busy2. Sometimes referred to as "Distinctive Busy" on some PABX models.	Q.850 cause value 17.
Congestion busy as defined in ITU-T Rec. E.180/Q.35.	Q.850 cause value 34.
Ring1. Ringing tone as defined in ITU-T Rec. E.180/Q.35. This is an intermediate call progress indicator. It can be used to generate a ringback signal to the originating G3FE as if it were an end-to-end PSTN connection.	ALERTING
Ring2. Ringing tone similar to Ring1 where two short rings are generated instead of one long ring. This is an intermediate call progress result.	ALERTING
SIT Intercept. Special Information Tones are defined in ITU-T Rec. E.180/Q.35. Intercept Tone is one combination of tones – frequency and duration.	Q.850 cause value 4. NOTE – SIT tones are not distinguished because they generally indicate a problem with the number to dial.
SIT Vacant. Special Information Tones are defined in ITU-T Rec. E.180/Q.35. Circuit Vacant Tone is one combination of tones – frequency and duration.	Q.850 cause value 4.
SIT Reorder. Special Information Tones are defined in ITU-T Rec. E.180/Q.35. Reorder Tone is one combination of tones – frequency and duration.	Q.850 cause value 4.
SIT No Circuit. Special Information Tones are defined in ITU-T Rec. E.180/Q.35. No Circuit Tone is one combination of tones – frequency and duration.	Q.850 cause value 4.

B.3.9 Interoperability

Both H.323 direct call model and T.38 Annex B require a well-known port to initiate call signalling. As described in ITU-T Rec. H.323, the H.323 well-known port is 1720. T.38 Annex B endpoints shall use the H.323 well-known port. In order for a single implementation (such as a gateway) to support multiple endpoints, dynamic ports must be used. A facsimile gateway conforming to this annex shall support H.323 RAS. Also, note that when the gatekeeper-routed call model is used, a well-known port is not needed.

An H.323 Annex D implementation becomes aware that it is communicating with a T.38 Annex B implementation due to the following sequence of events:

– The T.38 Annex B implementation does not supply an H.245 port in the *connect* or *setup*.

– The H.323 Annex D transmits a **FACILITY** message with a **FacilityReason** of **startH245** and provides its H.245 address in the **h245Address** element, as described in 8.2.3/H.323. The T.38 Annex B implementation receiving a **FACILITY** message with a **FacilityReason** of **startH245** shall respond with a **FACILITY** message having a **FacilityReason** of **noH245.** At this point, the H.323 Annex D implementation should cease all attempts to open the H.245 channel.

If the Annex B implementation connects with an H.323 non-facsimile capable implementation, it shall disconnect after noting the lack of facsimile OLCs in the **fastStart** elements in the responding messaging such as ALERTING, CALL PROCEEDING, PROGRESS, or CONNECT. If it notes the facsimile start procedure initiation in the responding message, it proceeds according to the fast connect procedures, with the exception that as an Annex B implementation, it need not support any

H.323 video, voice, or data feature or H.245 messaging. Thus, the T.38 Annex B implementation will disconnect from any H.323 (1996) implementation as it will not find the fast connect OLCs in the messages from those implementations. The T.38 implementation may also disconnect on seeing the H.323 version number 1.

Annex C

The optional forward error correction scheme for UDP

C.1 Overview of the optional forward error correction mechanism

The parity FEC scheme is symmetrical in that it is identical in both encode and decode modes, and may be computed for an arbitrary number of arbitrarily sized IFP messages. A transmitting gateway generates FEC messages by passing in a number of primary IFP packets; these FEC messages may then be assembled into a packet in accordance with Figure 5.

Receiving gateways which detect the loss of a primary IFP packet that is covered by an FEC message may be able to reconstruct it by passing in the remaining (received) primary IFP packets and the FEC message itself to the parity encode/decode algorithm. Certain conditions apply in order for a lost primary IFP packet to be recovered using the parity encoder/decoder; these shall be discussed in the following clauses.

C.2 Parity encode/decode scheme operation and characteristics

The parity scheme accepts a number of arbitrarily sized IFP messages. It aligns them vertically and zero pads the shorter length messages to produce a 2D matrix as shown in Figure C.1 a). A one-bit piece-wise sum is then performed on a column by column basis (equivalent to exclusive OR logic function) across the width of the matrix, each summation resulting in a binary digit. This process is illustrated in Figure C.1 b). The output from the parity scheme is the row of resulting binary data.

The basic error recovery scheme works by assuming that 1 loss in n packets may occur. If the $(n + 1)^{\text{th}}$ packet contains an FEC message generated from the primary IFP packets of the n preceding packets, then provided no more than one of the first n packets is lost, any missing IFP message can be reconstructed. The generation and reconstruction of primary IFP packets using the parity scheme outlined above is described in the following subclauses.

C.2.1 Generating and transmitting FEC messages

By utilizing a buffer similar to that shown in Figure C.2, it is possible to pass multiple prior primary IFP packets into the parity FEC algorithm for processing. The FEC scheme returns with a frame of encoded data that may then be assembled into a packet after the current primary IFP packet. The transmitting gateway must decide in advance the number of prior IFP messages it shall use to generate the FEC information. The n prior primary IFP packets are sent to the parity encoding scheme which results in a single message of FEC data of length l octets where l is the largest message length value encountered in the list of primary IFP packets plus 2 octets. Finally, the newly generated FEC message is assembled as in Figure C.2 and inserted into the packet after the primary IFP packet.

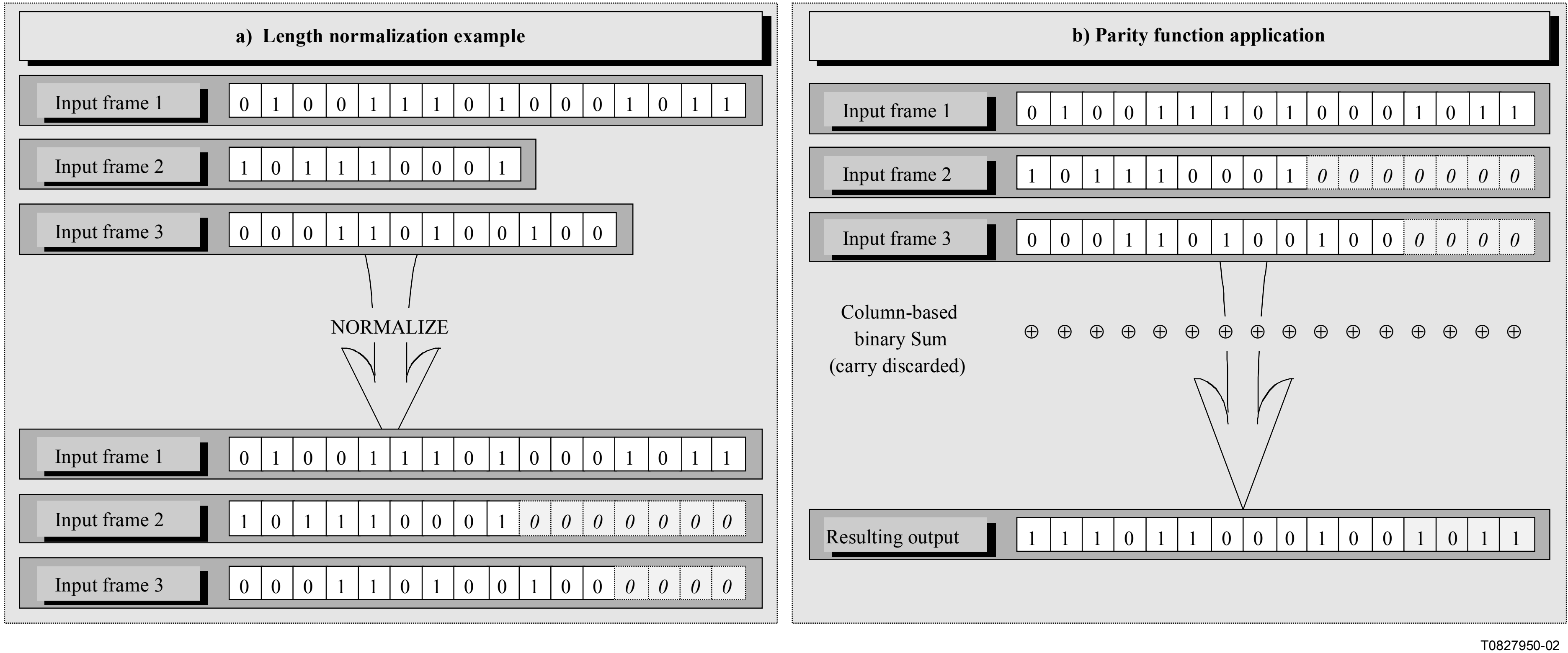


Figure C.1/T.38 – Illustration of Length normalization and Parity function operation

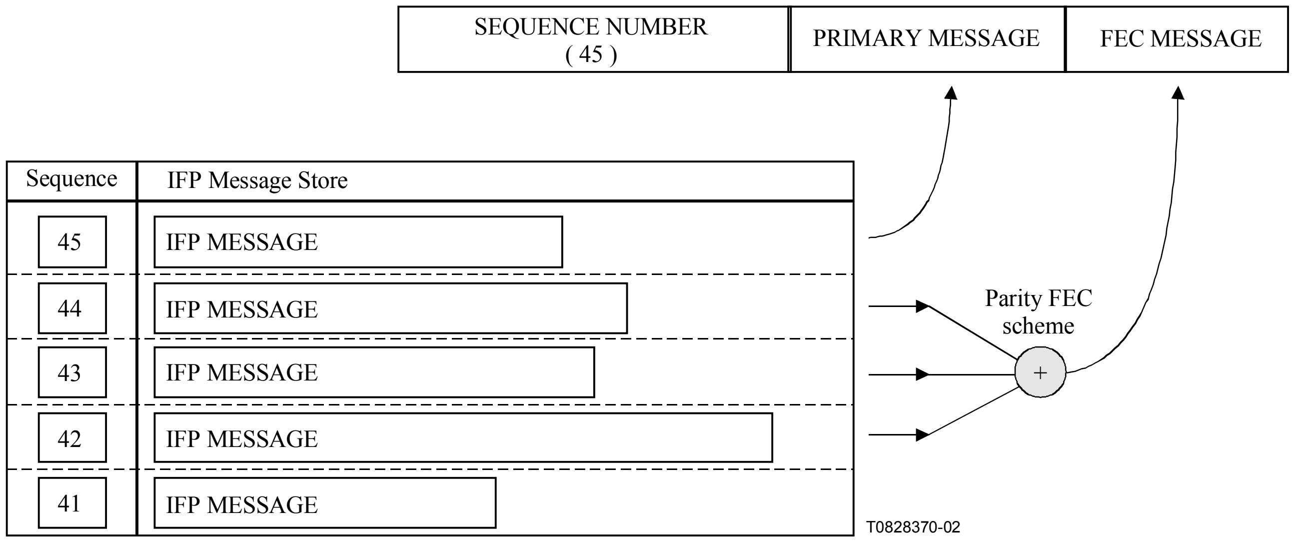


Figure C.2/T.38 – Generation and packetization of a single parity FEC frame

Multiple FEC messages may be sent in a single packet, each generated from *fec_npackets* (i.e. the number of) prior primary IFP packets. Unlike the instance where only one FEC message is present, when multiple FEC messages are transmitted in a single packet, the contributing primary IFP packets for each FEC message are not consecutive, but are interleaved. This is illustrated in Figure C.3 which shows an example providing protection against a burst of three consecutive lost packets.

C.2.2 Receiving FEC messages and primary IFP packet reconstruction

A gateway in receipt of FEC messages in a packet must determine from the UDPTL packet:

- the number of FEC messages present in the packet;
- the sequence numbers of the primary IFP packets contained in each FEC message;
- the sequence numbers of any packets which have been "lost" over the network.

In order to determine the sequence numbers of the primary IFP packets encoded in a given FEC message, the receiving gateway must extract the number of primary IFP packets covered by that frame. For a packet containing a single FEC message, the sequence numbers covered by that message are simply those from $[Seq - 1]$ to $[Seq - (n + 1)]$ where n is the value in the fec_npackets element and Seq is the value in the seq-number element. For a UDPTL packet containing m FEC messages with sequence number Seq and a message control field setting of n, the sequence number ranges for FEC message I (for $1 \le I \le m$) are trivially extracted from the following equations:

$$\mathrm{StartSeq} = Seq - I$$

$$\mathrm{EndSeq} = Seq - I - (m - 1)n$$

Intermediate sequence numbers between these ranges are linearly spaced with gap m. Once the sequence numbers of the primary IFP packets encoded in an FEC message have been determined, the receiving gateway may check to determine whether any of the primary IFP packets listed has failed to arrive. If one, and only one, of these primary IFP packets has failed to arrive, then the FEC message and the remaining (delivered) primary IFP packets may be sent to the parity algorithm to recover the missing sequence.

The number of FEC messages, m, is the number of octet strings contained in the **fec-data** element (as encoded in the SEQUENCE OF construct).

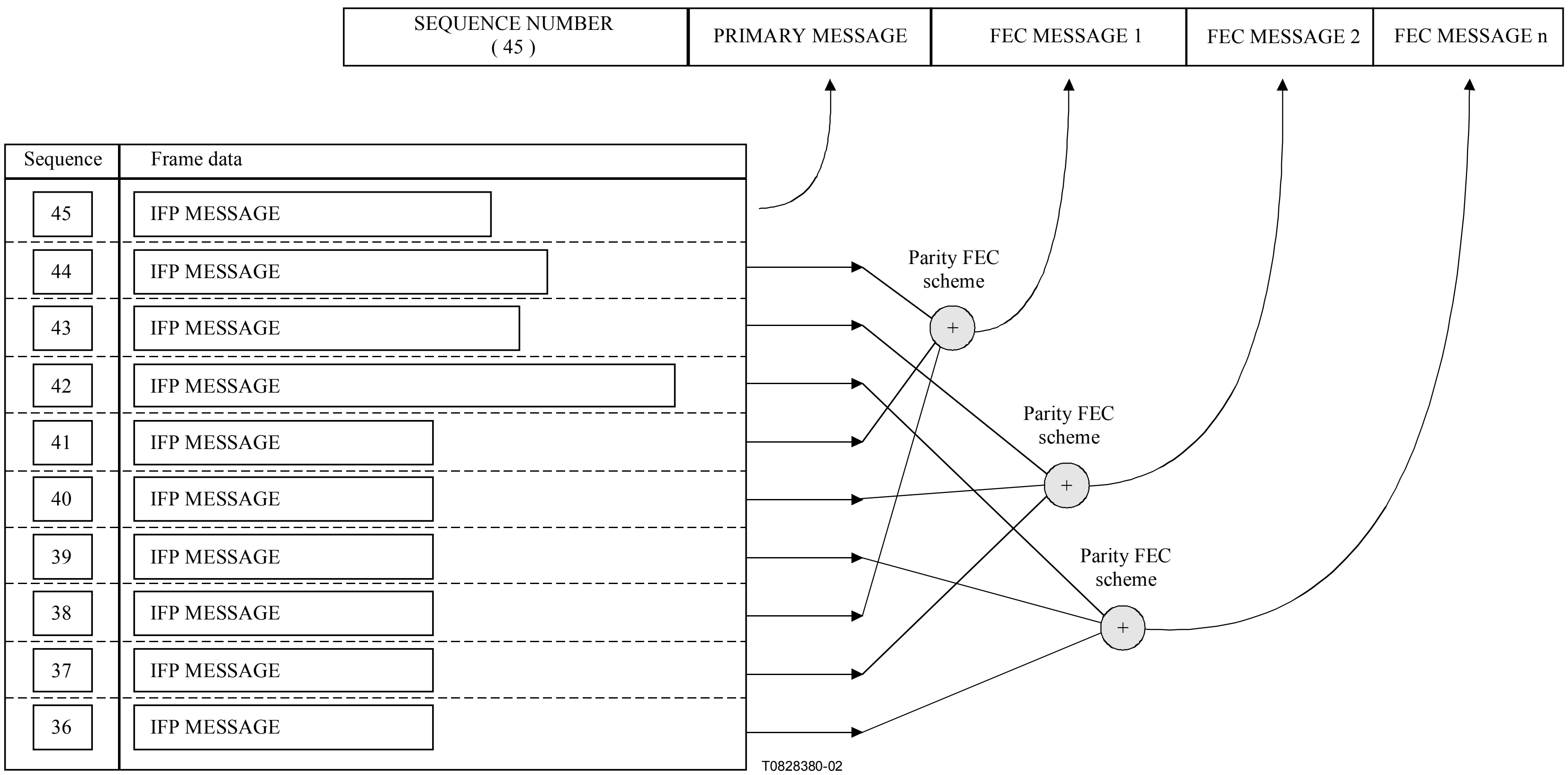


Figure C.3/T.38 – Generating multiple FEC messages to protect against burst errors

Annex D

SIP/SDP call establishment procedures

D.1 Introduction

This annex describes system level requirements and procedures for internet aware facsimile implementations and internet aware facsimile gateways conforming to ITU-T Rec. T.38 to establish calls with other ITU-T T.38 implementations using the procedures defined in RFC 2543 (SIP) and RFC 2327 (SDP).

D.2 Communication between gateways

D.2.1 Overview

D.2.1.1 Call setup

Call setup for Annex D/T.38 compliant implementations is based on SIP (Session Initiation Protocol) defined in RFC 2543. As in Annex B/T.38, implementations may operate in two distinct compatible environments:

1) A facsimile-only over IP environment – In this environment, no voice support is provided. The procedures and requirements of D.2.2.3 shall apply to implementations operating in this environment.

2) A facsimile and voice over IP environment – The procedures and requirements of this annex shall apply to implementations operating in this environment.

D.2.1.2 Media channels

ITU-T T.38 facsimile packets are sent on a separate TCP/UDP port from SIP call signalling. A minimal Annex D/T.38 implementation requires a TCP/UDP port (default is 5060) for call signalling and either a UDP port or a TCP port for ITU-T T.38 facsimile information.

D.2.1.3 Usage of SDP

Endpoints conforming to this annex are required to support SDP, including the extensions described below.

D.2.2 Basic call setup

D.2.2.1 Choosing call setup mechanism

Annex B/T.38 indicates that H.323 FastCall Setup is the basic mechanism for establishing a T.38 call. The method described in this annex is intended for use in conjunction with this mechanism in a decomposed gateway model. In addition, this annex may also be used if the emitting gateway is aware that the destination gateway supports the call establishment mechanism of this annex.

D.2.2.2 SIP call setup

According to RFC 2543 section 1, SIP supports a five-phase process for establishing and terminating a call:

User location	determination of the end system to be used for communication
User capabilities	determination of the media and media parameters to be used
User availability	determination of the willingness of the called party to engage in communications
Call setup	"ringing", establishment of call parameters at both called and calling party
Call handling	including transfer and termination of calls

SIP can also be used in conjunction with other call setup and signalling protocols. In that mode, an end system uses SIP exchanges to determine the appropriate end system address and protocol from a given address that is protocol-independent. For example, SIP could be used to determine that the party can be reached via H.323, obtain the H.245 gateway and user address and then use H.225.0 to establish the call.

SIP can invite users to sessions with and without resource reservation. SIP does not reserve resources, but can convey to the invited system the information necessary to do this.

D.2.2.3 Facsimile-only connection

The emitting gateway sends a SIP INVITE request (with the appropriate options set) for a T.38 facsimile connection with the receiving SIP server. The receiving server will likely be the receiving gateway; however, it may also proxy or redirect the SIP connection to the actual gateway through SIP or other means. In any case, a response will be sent to the emitting gateway indicating acceptance, redirection or failure of the request.

If accepted (or a redirected INVITE is accepted), the T.38 facsimile call proceeds.

Once the call is completed, the call may be disconnected with a SIP BYE command.

D.2.2.4 Voice and facsimile connection

A SIP INVITE is made to the called party requesting a voice connection per the requirements of RFC 2543. A voice connection is then established.

Upon detection of facsimile by the receiving gateway, a SIP INVITE request is sent to the emitting gateway (with the same Call-ID as the existing voice connection) for a T.38 facsimile connection. Upon completion of the facsimile call establishment (noted in D.2.2.3), the T.38 facsimile call proceeds with a T.38 V.21 flags indicator packet.

Note that during this switchover and the facsimile call, it may be useful to mute the voice channel. The voice channel may be used again once the end of facsimile transmission is detected. Alternately, some implementations may choose to replace the voice channel with a facsimile channel.

Once the call is completed, the call may be disconnected with a SIP BYE command.

D.2.3 Capabilities negotiation

There are several capabilities that need to be negotiated to determine which options the gateways support and use. These are described in Table B.1.

The RFC 2327 Session Description Protocol (SDP) provides mechanisms for describing sessions for SIP. However, new attributes (section 6 of SDP) are required to support ITU-T Rec. T.38. Specifically, the following options are registered with IANA as valid att-field and att-value values per the procedure noted in Appendix B of SDP (RFC 2327). Note that options without values are boolean – their presence indicates that they are valid for the session. These capabilities are negotiated using the following ABNF elements defined for use with ITU-T Rec. T.38:

```
Version
     Att-field=T38FaxVersion
     Att-value = 1*(DIGIT)
     ;Version 0, the default, refers to T.38 (1998)
Maximum Bit Rate
     Att-field=T38MaxBitRate
     Att-value = 1*(DIGIT)
Fill Bit Removal
     Att-field=T38FaxFillBitRemoval
MMR Transcoding
     Att-field=T38FaxTranscodingMMR
JBIG Transcoding
     Att-field=T38FaxTranscodingJBIG
Data Rate Management Method
     Att-field=T38FaxRateManagement
     Att-value = localTCF | transferredTCF
UDP Options
Maximum Buffer Size
     Att-field=T38FaxMaxBuffer
     Att-value = 1*(DIGIT)
     ;optional
Maximum Datagram Size
     Att-field=T38FaxMaxDatagram
     Att-value = 1*(DIGIT)
     ;optional
Error Correction
     Att-field=T38FaxUdpEC
     Att-value = t38UDPFEC | t38UDPRedundancy
```

D.2.3.1 Declaration of T.38 in SDP

The image/t38 MIME content type in SDP indicates ITU-T Rec. T.38.

This choice is consistent with image/tiff used in ITU-T Rec. T.37 and image/g3fax used for ITU-T Rec. X.420.

D.2.3.2 Use of either TCP or UDP

Two logical channels (sender to receiver channel and receiver to sender channel) shall be opened for the transfer of T.38 packets. T.38 packets can be transferred using either TCP or UDP. In general, the usage of TCP is more effective when the bandwidth for facsimile communication is limited, or for IAF to IAF transfers since TCP provides flow control. On the other hand, the usage of UDP may be more effective when the bandwidth for facsimile communication is sufficient.

Note that during the SIP call setup, the calling party suggests the transport (TCP or UDP) by listing its preferred first in the SDP of a SIP INVITE. The receiver should open the TCP/UDP port based on the preference of the sender, but the receiver decides.

In support of T.38 choice of UDP or TCP transport, SDP extensions:

– indicate UDPTL (facsimile user datagram protocol transport layer) as a valid transport value (third field).
– indicate TCP (transmission control protocol) as a valid transport value (third field).
– include t38 as a valid format type value (fourth field).

NOTE – As this is not an RTP-defined value, it has to be a MIME sub-type of the media type. As a result, this is awaiting the publication of an IETF RFC to define the registration of image/t38 with IANA as a valid MIME content-type per the procedure noted in Appendix B of SDP (RFC 2327).

D.2.4 Examples of call setup

D.2.4.1 Facsimile only invite

The default case requires support for both TCP and UDP. In this case, two 'm=' lines are listed with the preferred first in the invite. The rejected media connection will be indicated with a port number set to zero in the response.

For a two-party facsimile-only call between T.38 gateways:

```
C->S:  INVITE sip:+1-212-555-1234@bell-tel.com SIP/2.0
       Via: SIP/2.0/UDP kton.bell-tel.com
       From: A. Bell <sip:+1-519-555-1234@bell-tel.com>
       To: T. Watson <sip:+1-212-555-1234@bell-tel.com>
       Call-ID: 3298420296@kton.bell-tel.com
       CSeq: 1 INVITE
       Subject: Mr. Watson, here is a fax
       Content-Type: application/sdp
       Content-Length: ...
       v=0
       o=faxgw1 2890844526 2890842807 IN IP4 128.59.19.68
       e=+1-212-555-1234@bell-tel.com
       t=2873397496 0
       c=IN IP4 128.59.19.68
       m=image 49170 udptl t38
       a=T38FaxRateManagement:transferredTCF
       a=T38FaxUdpEC:t38UDPFEC
       m=image 49172 tcp t38
       a=T38FaxRateManagement:localTCF

S->C: SIP/2.0 200 OK
       Via: SIP/2.0/UDP kton.bell-tel.com
       From: A. Bell <sip:+1-519-555-1234@bell-tel.com>
       To: T. Watson <sip:+1-212-555-1234@bell-tel.com>
       Call-ID: 3298420296@kton.bell-tel.com
       CSeq: 1 INVITE
       Contact: sip:watson@boston.bell-tel.com
       Content-Type: application/sdp
       Content-Length: ...
       v=0
       o=faxwatson 4858949 4858949 IN IP4 192.1.2.3
       c=IN IP4 boston.bell-tel.com
       m=image 5002 udptl t38
       a=T38FaxRateManagement:transferredTCF
       a=T38FaxUdpEC:t38UDPFEC
       m=image 0 tcp t38
```

D.2.5 Minimum call setup messages

The implementation of this annex shall support the minimum requirements for a SIP client and server as defined in RFC 2543 sections A.1 and A.2:

All clients MUST be able to generate the INVITE and ACK requests. Clients MUST generate and parse the Call-ID, Content-Length, Content-Type, CSeq, From and To headers. Clients MUST also parse the Require header. A minimal implementation MUST understand SDP (RFC 2327). It MUST be able to recognize the status code classes 1 through 6 and act accordingly.

A minimally compliant server implementation MUST understand the INVITE, ACK, OPTIONS and BYE requests. A proxy server MUST also understand CANCEL. It MUST parse and generate, as appropriate, the Call-ID, Content-Length, Content-Type, CSeq, Expires, From, Max-Forwards, Require, To and Via headers. It MUST echo the CSeq and Timestamp headers in the response. It SHOULD include the Server header in its responses.

D.2.6 Mapping of call progress signals

For call setup and call progress, the return signals can be simplified to the following set. These are all returned prior to or instead of a 200 OK response to the INVITE request.

Table D.1/T.38 – Call Progress Mapping

Meaning	SIP Response Mapping
Busy1. Subscriber busy tone as defined in ITU-T Rec. E.180/Q.35.	486 Busy here
Busy2. Sometimes referred to as Distinctive Busy on some PABX models.	486 Busy here
Congestion busy as defined in ITU-T Rec. E. 180/Q.35.	600 Busy everywhere
Ring1. Ringing tone as defined in ITU-T Rec. E.180/Q.35. This is an intermediate call progress indicator. It can be used to generate a ringback signal to the originating G3FE as if it there were an end-to-end PSTN connection.	180 Ringing
Ring2. Ringing tone similar to Ring1 where two short rings are generated instead of one long ring. This is an intermediate call progress result.	180 Ringing
SIT Intercept. Special Information Tones are defined in ITU-T Rec. E.180/Q.35. Intercept Tone is one combination of tones – frequency and duration.	503 Service Unavailable
SIT vacant. Special Information Tones are defined in ITU-T Rec. E.180/Q.35. Circuit Vacant Tone is one combination of tones – frequency and duration.	503 Service Unavailable
SIT Reorder. Special Information Tones are defined in ITU-T Rec. E.180/Q.35. Reorder Tone is one combination of tones – frequency and duration.	503 Service Unavailable
SIT No Circuit. Special Information Tones are defined in ITU-T Rec. E.180/Q.35. No Circuit Tone is one combination of tones – frequency and duration.	503 Service Unavailable
NOTE – SIT tones are not distinguished because it generally indicates a problem with the number dialled.	

The 200 OK message in response to an INVITE request is returned when the gateway, *by some means*, determines that a connection to the terminal G3FE has been established. If CED or FSK flags are detected, the appropriate ITU-T T.38 messages can be sent.

D.2.7 Usage of the T38maxBitRate in messages

T38maxBitRate refers to the maximum fax data rate that is supported by an endpoint. When TCP is used for T.38 facsimile transmission, **T38maxBitRate** does not apply. When UDP is used for T.38 facsimile transmission, **T38maxBitRate** should be specified to aid in bandwidth allocation.

D.2.8 DTMF transmission

SIP can transfer collected DTMF dialling digits as a SIP URL as defined in RFC 2543 section 2:

sip:+1-212-555-1212@gateway.com;user=phone

DTMF transmission during an established voice and facsimile connection may be completed using the RTP tone payload described in RFC 2833.

D.2.9 Interoperability

Both SIP and Annex B/T.38 require a well-known port to initiate call signalling. As described in SIP, its well-known port is 5060. Endpoints in this annex shall use the SIP well-known port by default.

Annex E

H.248.1 call establishment procedures

E.1 Introduction

This annex describes system level requirements and procedures for internet aware facsimile implementations and internet aware facsimile gateways conforming to ITU-T Rec. T.38 to establish calls with other ITU-T T.38 implementations using the procedures defined by ITU-T Rec. H.248.1.

E.2 Communication between gateways

E.2.1 Overview

E.2.1.1 Gateway architecture

The method described in this annex is intended for use in conjunction with other methods in a decomposed gateway model as shown in Figure E.1. In this model, the media gateway controller (MGC) has knowledge of all the endpoints within a domain and has control over connections being created and terminated at its media gateways (MG).

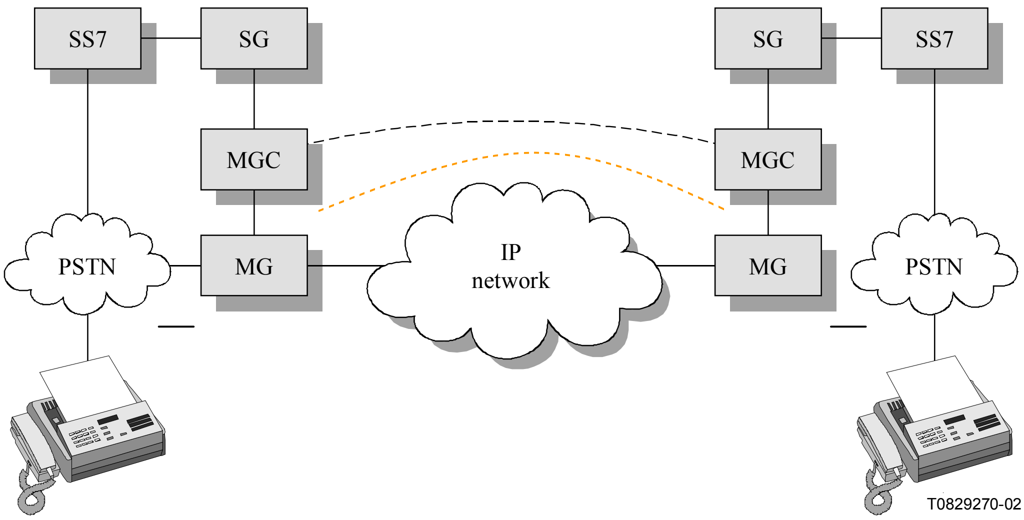


Figure E.1/T.38 – Typical decomposed model

The mechanism in this annex complements the mechanism of Annex D/H.323 (which describes a simple case without a decomposed gateway). In the situation where more than one MGC is involved in a call, the mechanism in this annex (other methods are for further study) is used to signal between them.

E.2.1.2 Call setup

Call setup for implementations compliant to this annex is based on ITU-T Rec. H.248.1. As in the basic Annex B/T.38, implementations may operate in two distinct compatible environments:

1) A facsimile-only over IP environment – In this environment, no voice support is provided. The procedures and requirements of D.2.2.1 shall apply to implementations operating in this environment.

2) A facsimile and voice over IP environment – The procedures and requirements of E.2.2.2 shall apply to implementations operating in this environment.

E.2.1.3 Media channels

ITU-T T.38 facsimile packets are sent on a separate TCP/UDP port from H.248 message transport. A minimal implementation of this annex requires a TCP port for call signalling and either a UDP port or a TCP port for ITU-T T.38 facsimile information.

E.2.2 Basic call setup

According to 8.2.1/H.248.1:

– the connection model for the protocol describes the logical entities, or objects, within the Media Gateway that can be controlled by the Media Gateway Controller. The main abstractions used in the connection model are Terminations and Contexts;

– a *termination* is an object that sources and/or sinks media streams;

– a *context* represents a collection of *terminations* in a single conference.

Terminations recognize events that invoke a response by the MGC to create another event (e.g. recognizing off-hook invokes play dial tone). This interaction proceeds throughout a typical call setup process initiated at the MG (e.g. H.323 FastCall Setup).

Note that two cases exist for the use of this mechanism:

1) If the Call Agent (MGC & Gatekeeper) controls both MGs, then ITU-T Rec. H.248.1 is used to modify the existing connection between the two MGs.

2) If different Call Agents are involved (e.g. when two different service providers are involved in completing a call), then MGC-MGC communication is required (i.e. using the mechanism of Annex D/T.38). On confirmation of a connection, the on-ramp call agent instructs its media gateway (via ITU-T Rec. H.248.1) to initiate a T.38 session with the off-ramp MG.

E.2.2.1 Facsimile-only connection

Digits are collected by the media gateway (MG) and sent to the calling agent to invite the called party on a facsimile call.

Once connected, the call proceeds as in Annex B/T.38.

E.2.2.2 Voice and facsimile connection

Digits are collected by the media gateway (MG) and sent to the calling agent to invite the called party to a voice connection as defined in ITU-T Rec. H.248.1. A voice connection is set up.

Upon detection of CNG by the emitting media gateway (MG), the calling agent is informed (via ITU-T Rec. H.248.1) of this event and instructs the destination MG to play CNG. If the destination MG then notifies the MGC of a CED (or V.21 flags) event and is capable of T.38, the MGC requests that each MG open a T.38 connection. Details for discrimination of the call as facsimile are described in clause 8/H.248.2. The MGC may also request that a new MG handle the facsimile connection. The T.38 protocol proceeds with a T.38 V.21 flags indicator packet.

Note that if T.38 is not supported by one of the MGs, the MGC may decide to attempt the facsimile call over G.711 (using G.711 in this case is beyond the scope of this annex). Full flexibility of switching between MGs (e.g. voice+facsimile, voice-only or facsimile-only) and deciding on options will not be possible if the MGC is not notified of the facsimile events (and the MG alone detects facsimile and switches blindly to T.38). Upon completion of the facsimile call (T.38 completion) by the off-ramp media gateway (MG), the calling agent is informed (via ITU-T Rec. H.248.1) of this event and may request that the connection be reverted to voice.

E.2.3 Event and signal indication

There are several events and signals that need to be transferred from the MG to the MGC and vice versa during the setup of a fax call. These events are defined in ITU-T H.248 packages. The base packages are in Annex E/H.248.1. Additional signals for fax are defined in ITU-T Rec. H.248.2.

E.2.4 Capabilities negotiation

There are several options that need to be negotiated to determine which options the gateways support and use. These are described in Table B.1 and are defined as SDP extensions in D.2.3. They are also defined as binary types in the IP Fax package of ITU-T Rec. H.248.2.

A T.38 Annex E implementation may use the SDP extensions to describe the fax media terminations in text mode of the protocol. A H.248.1 implementation shall use the IP Fax package as the preferred method to describe the fax media termination. These media descriptors indicate the capabilities of, or requested of, a media gateway (e.g. TCP or UDPTL transport).

In addition, as well as being able to identify that a call is using T.38 transport for facsimile, ITU-T Rec. H.248.1 may also indicate other transports.

E.2.5 Examples of call setup

Examples of this procedure are described in Appendix III.

E.2.6 Minimum call setup messages

The implementation of this annex shall support the minimum requirements for H.248.1 as noted in 8.2/H.248.1.

E.2.7 Mapping of call progress signals

For call setup and call progress, the return signals are identical to those in Annex B/T.38 (for H.323 FastCall setup) and Annex D/T.38 (for SIP).

E.2.8 DTMF transmission

H.248.1 supports collection of DTMF digits to make a call.

DTMF tones transmission during an established voice and facsimile call is handled within DTMF packages of E.5 and E.6 of H.248.1.

E.2.9 Interoperability

Both ITU-T Rec. H.248.1 and Annex B/T.38 require a well-known port to initiate call signalling. T.38 Annex E endpoints shall use the H.248.1 well-known port of 2944 for the text protocol and 2945 for the binary protocol.

Appendix I

Session examples

I.1 Session examples

This appendix contains a number of examples to show how the sending and receiving G3FEs communicate with the gateways and what packets the gateways exchange. All examples show a TCP implementation using Method 1 Rate Adaptation.

Time proceeds downward. Information flows on the solid lines in the direction of the arrows. The box superimposed on each line indicates what information is being transmitted. All information between the G3FE and a gateway is T.30/T.4/T.6-conforming information. Information transmitted between the gateways is in the form of packets as described in this Recommendation. The contents of the labelling box on a packet transmission indicates the packet type, followed by any additional information which is carried in the packet's payload.

Dashed lines are used to clarify the point in time at which a piece of information begins transmission (for example, T30_INDICATOR: Flags packets are sent when flags are noticed, not necessarily when the flags begin or end transmission). Dashed lines do not indicate any type of information flow.

Packet labels indicate the type of packet as well as any field information, for field-type packets. For example, a label such as "V.21:HDLC:TSI/FCS" indicates a V.21 HDLC (T.30 Control) packet with a field containing TSI information and a field indicating FCS. Due to space constraints, the FCS is generalized to include FCS and FCS-Sig-End.

I.1.1 Two traditional facsimile devices communicating using ECM

Figure I.1 shows two traditional Group 3 facsimile devices that use the PSTN for communicating with facsimile gateways. ECM is used for image transfer. The example begins after the transport connection/session has been established and the Receiving G3FE has answered a call from the Receiving Gateway and is about to generate CED.

I.1.2 Traditional facsimile device and Internet-aware facsimile device

Figure I.2 shows a traditional Group 3 facsimile device sending to an Internet-aware facsimile device without using ECM. The example begins after the transport connection/session has been established and the Receiving is about to generate CED.

I.1.3 Two traditional facsimile devices using frequent frames

Figure I.3 shows two traditional Group 3 facsimile devices that use the PSTN for communicating with facsimile gateways. It is similar to the scenario described in I.1.1 except that the image transfer does not use ECM and the receiving gateway does not wait for complete HDLC BCS sequences before beginning to send frames.

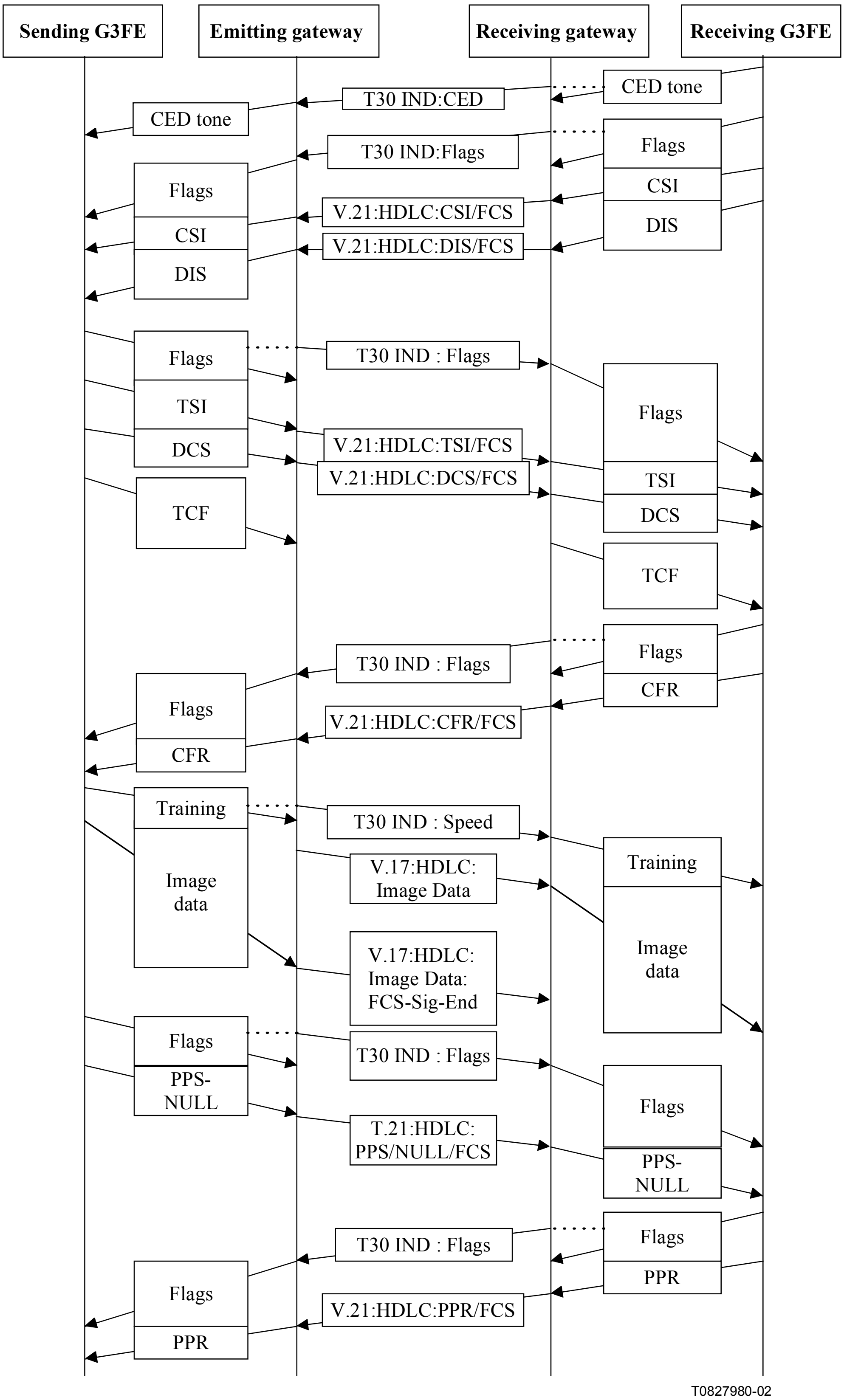


Figure I.1/T.38 – Two Group 3 facsimile devices communicating through gateways (*sheet 1 of 2*)

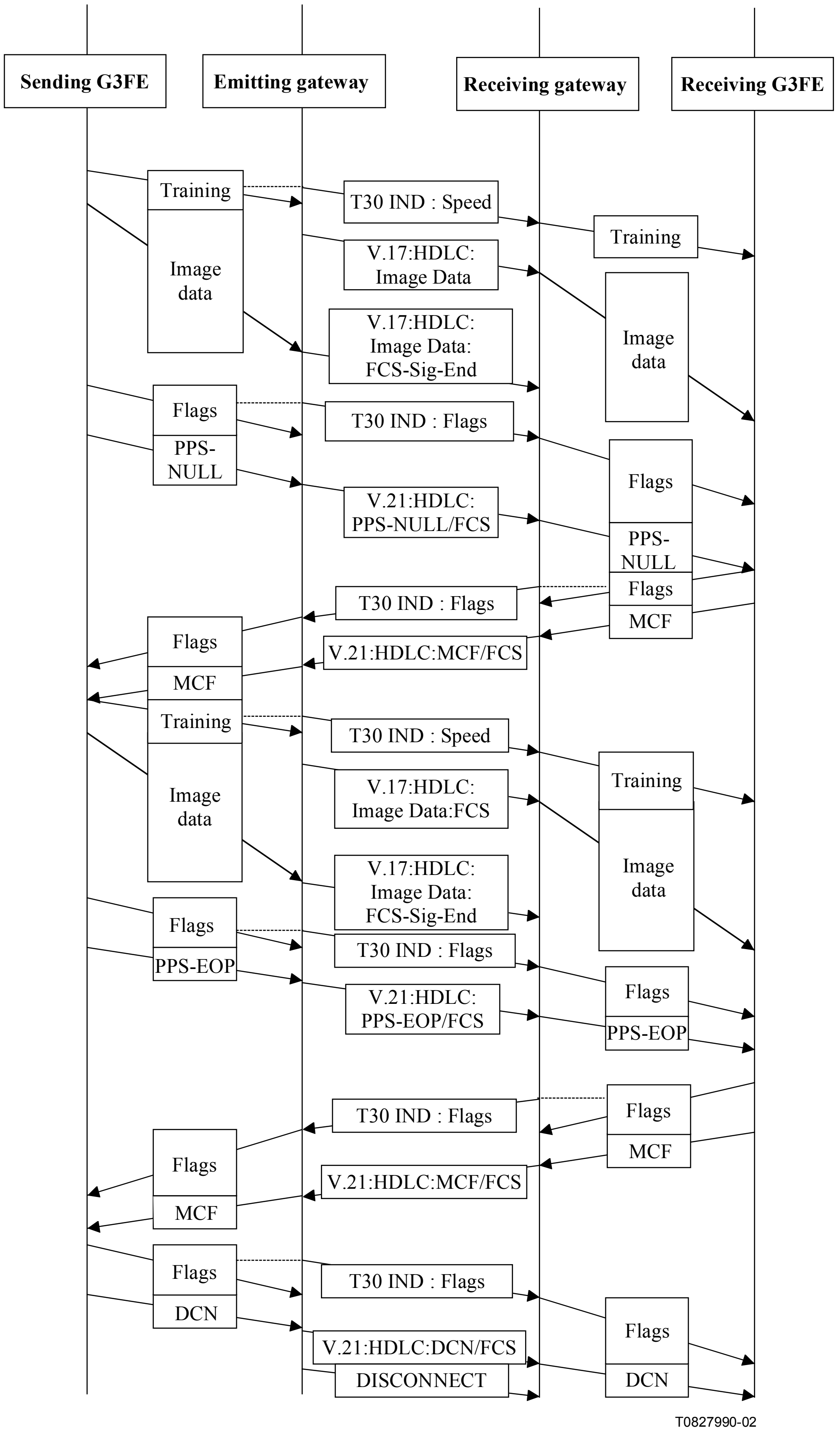


Figure I.1/T.38 – Two Group 3 facsimile devices communicating through gateways (*sheet 2 of 2*)

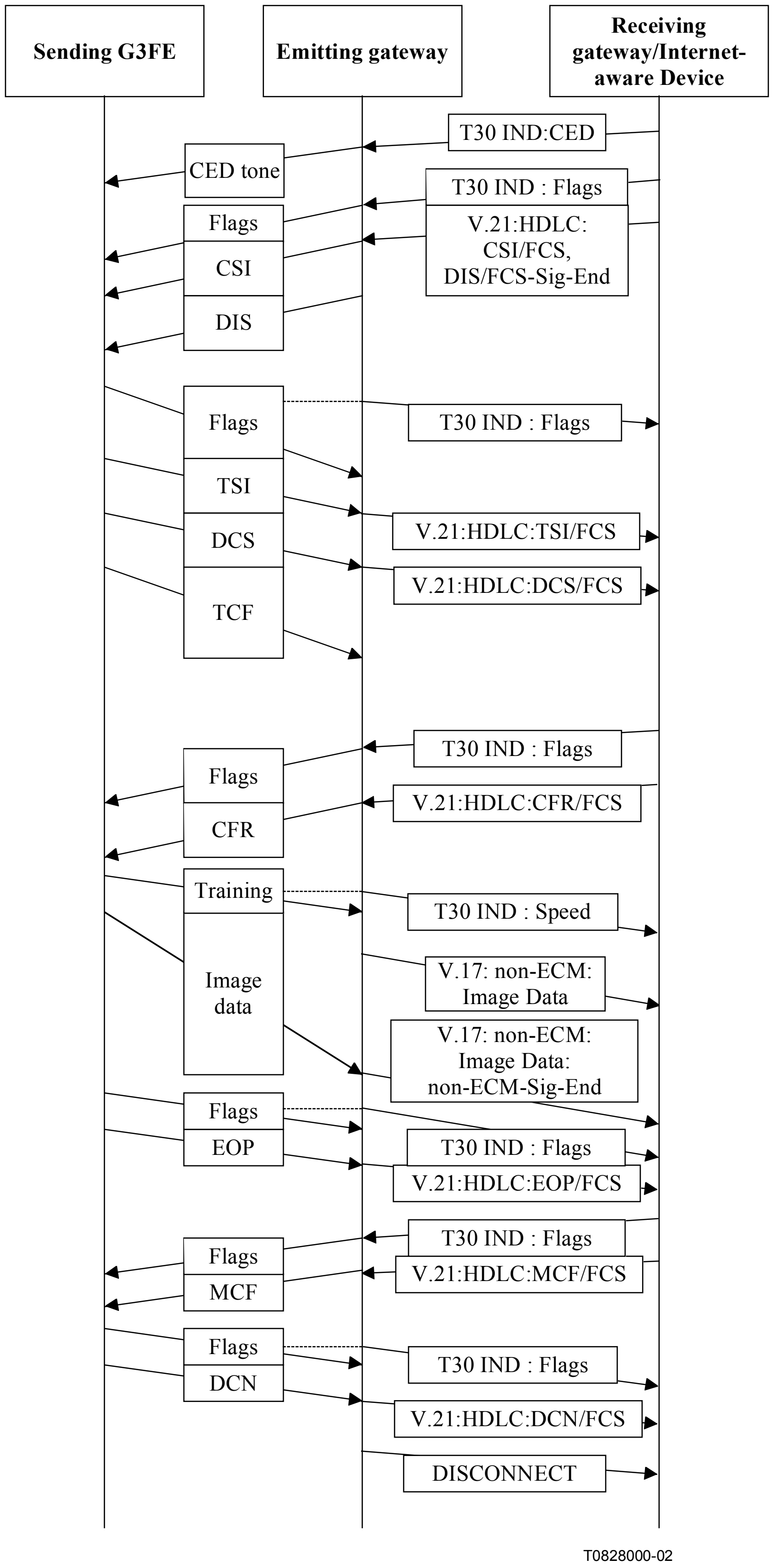


Figure I.2/T.38 – Traditional facsimile device and Internet-aware device

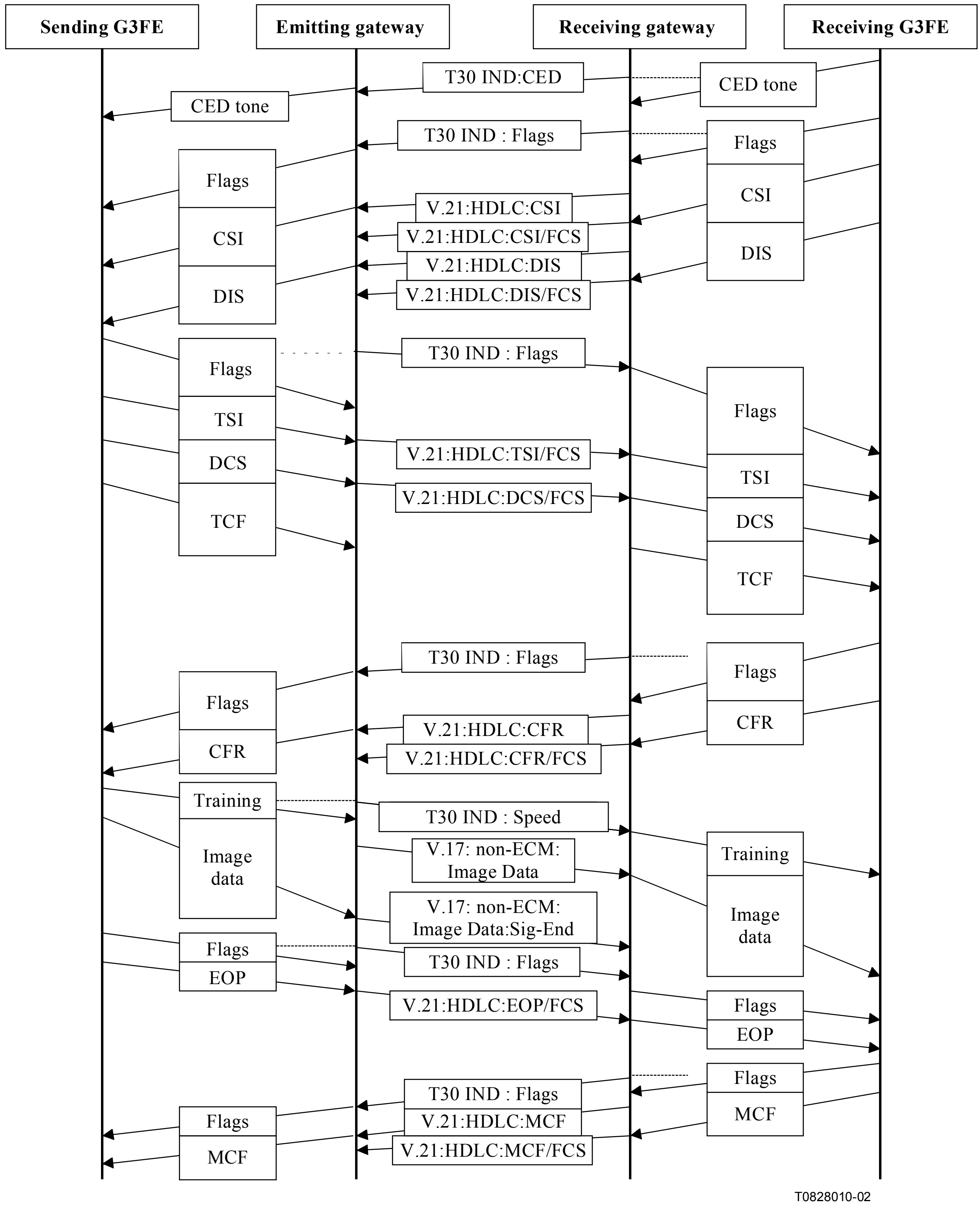


Figure I.3/T.38 – Use of multiple frames per BCS sequence

I.2 IAF device

This clause addresses the sequences that are considered in IAF device communication.

I.2.1 Sender is an IAF device, receiver is G3fax

CFR signal receive timing at the IAF device

It is recommended that IAF devices wait to receive the CFR signal, taking into account the period during which the Gateway sends TCF to G3fax. As Figure I.4 shows, this prevents the DCS signal of the IAF device from colliding at the Gateway with the CFR signal of the G3fax.

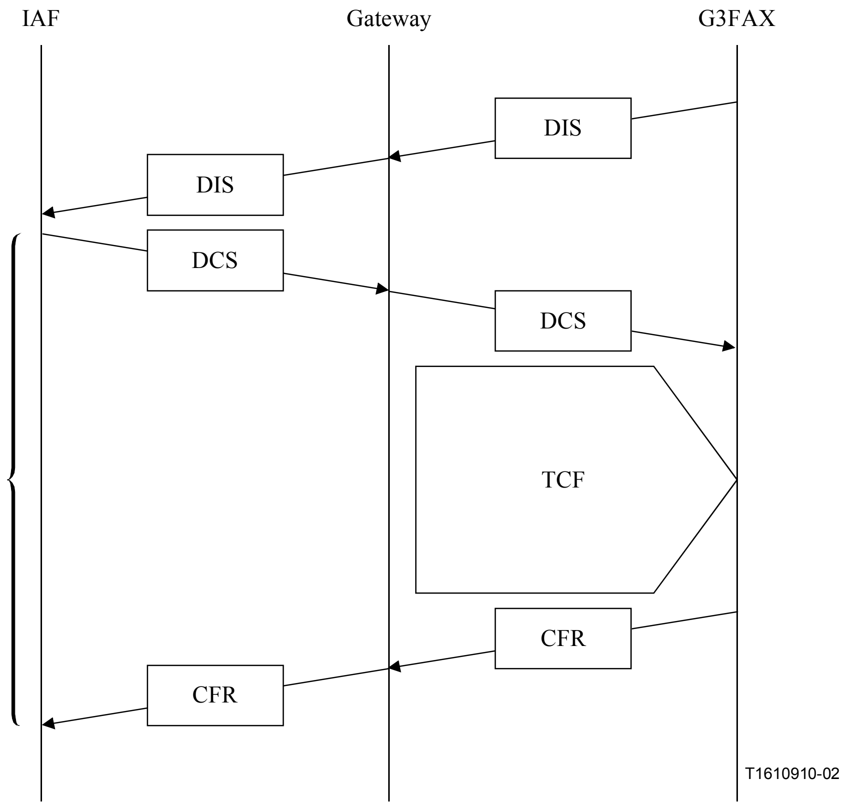


Figure I.4/T.38 – IAF transmit timing from DCS to CFR

I.2.2 Receiver is an IAF device, sender is G3fax

CFR signal send timing at the IAF device

It is recommended that IAF devices send the CFR signal, after taking into account the period during which the Gateway receives TCF from the G3fax device. As Figure I.5 shows, this prevents the TCF from colliding with the CFR signal from the IAF device.

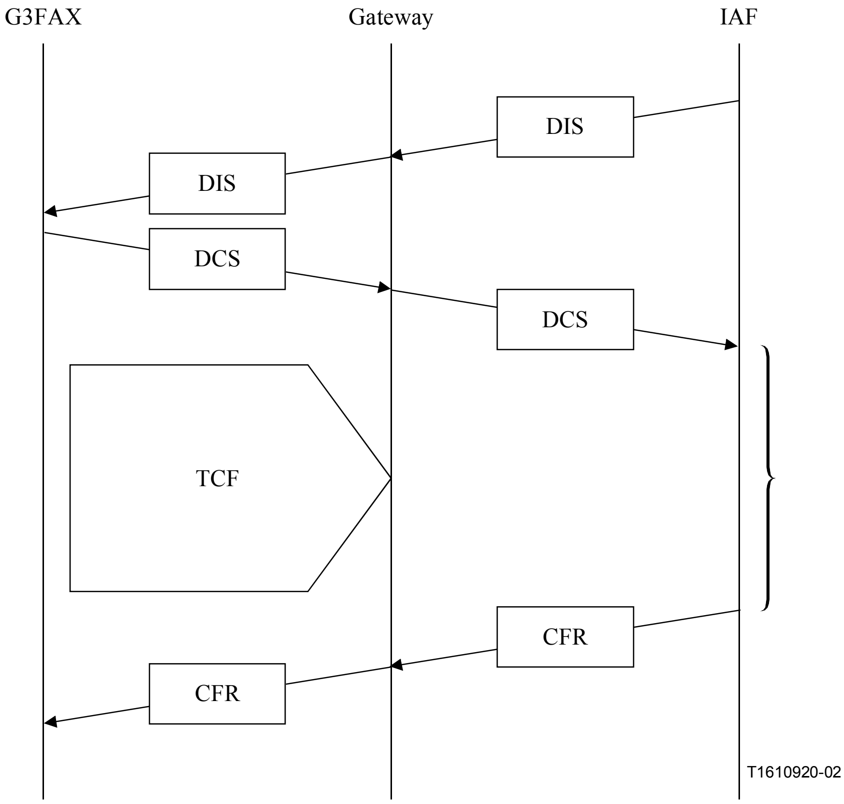


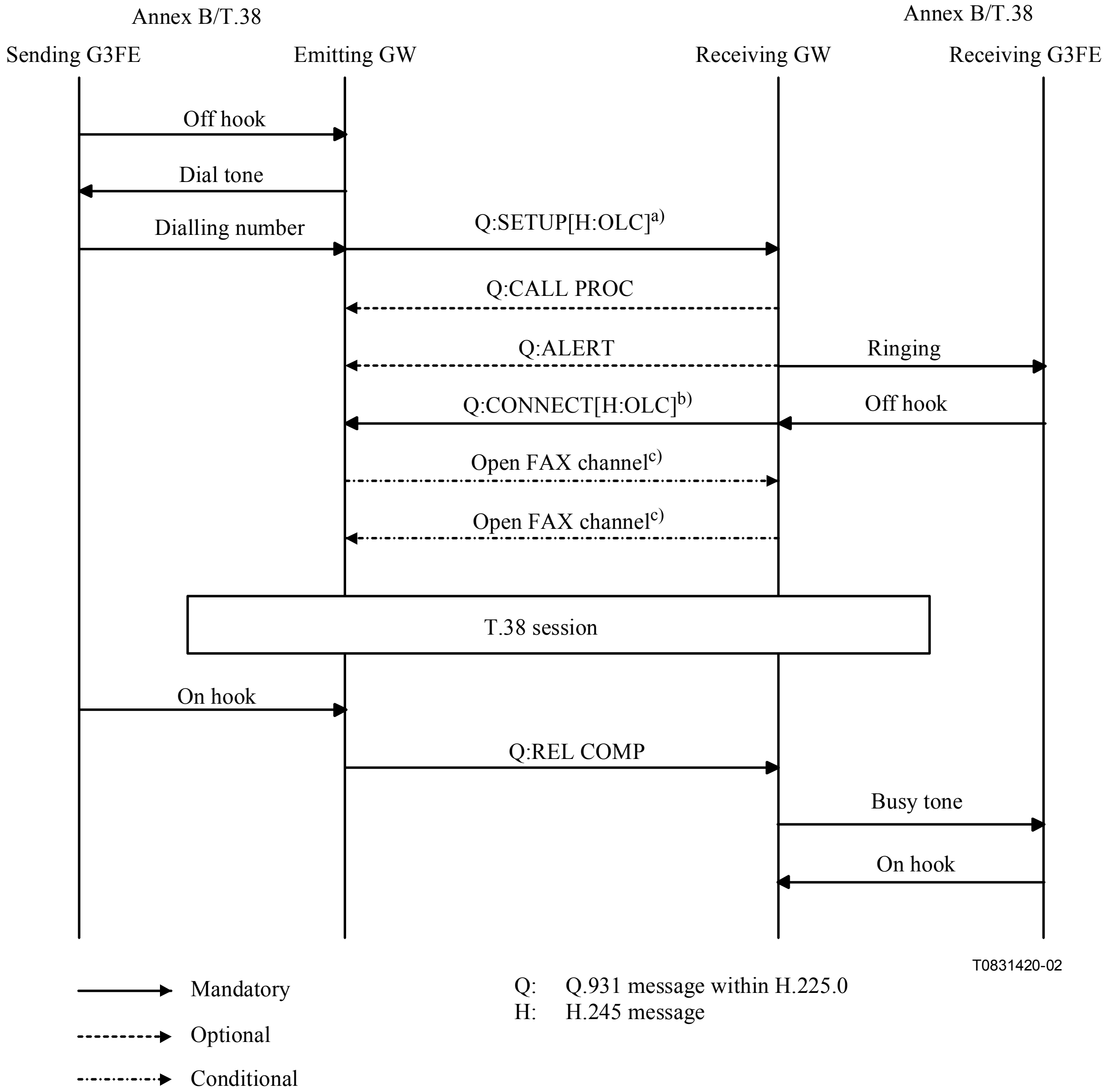
Figure I.5/T.38 – IAF receive timing from DCS to CFR

Appendix II

Examples of call establishment procedures described in Annex B/T.38

II.1 Sequence examples of call establishment procedures

II.1.1 Between Annex B/T.38 gateways



[a] SETUP contains Setup-UUIE, which includes fastStart element that is linked to OpenLogicalChannel (OLC) of ITU-T Rec. H.245.

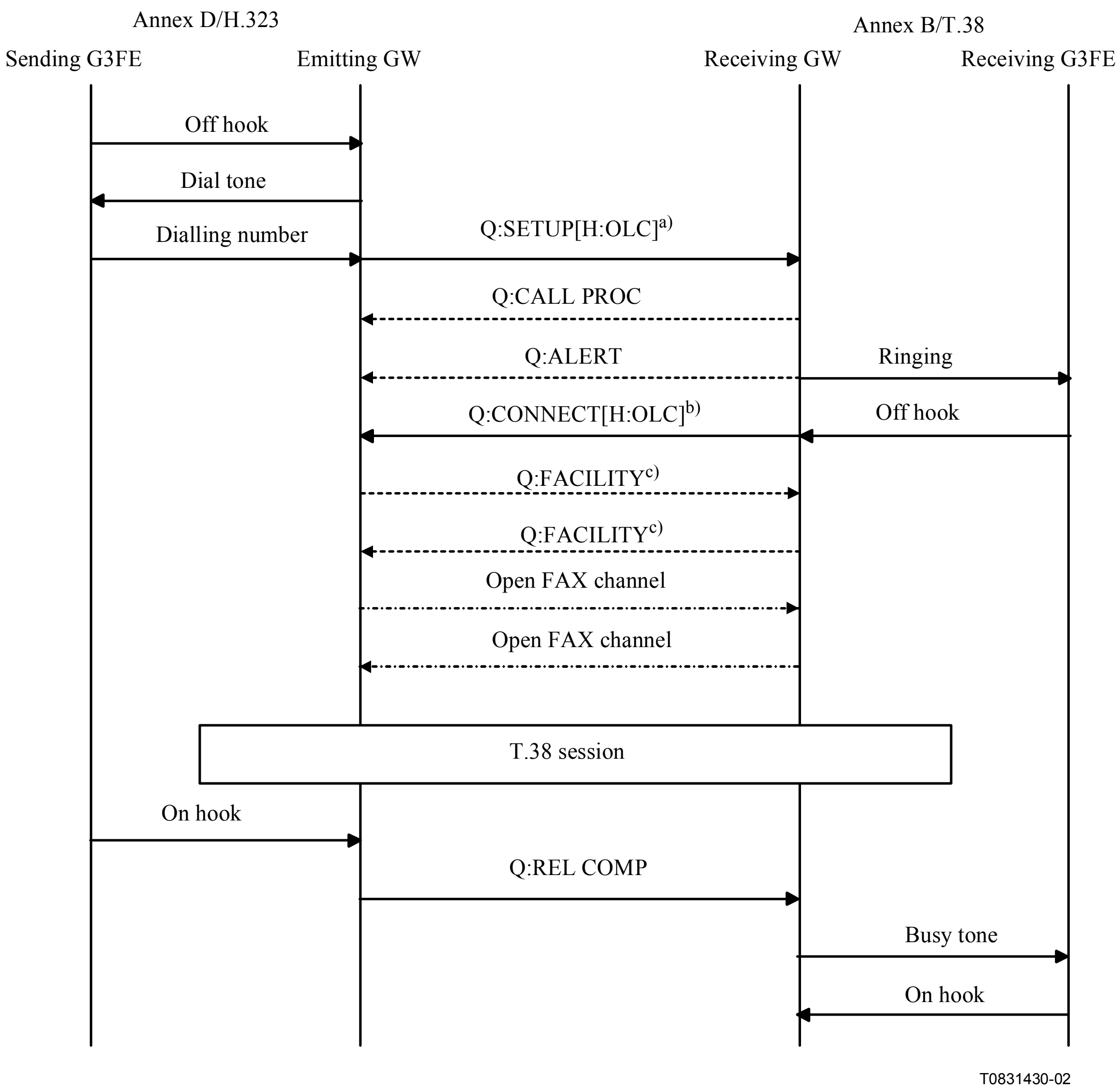
[b] CONNECT contains Connect-UUIE which includes fastStart element which is linked to OpenLogicalChannel (OLC) of ITU-T Rec. H.245.

[c] FAX channel is opened using either TCP or UDP. This phase specifically describes the operation of TCP connection between Annex B/T.38 devices. When UDP applies, this phase does not appear because UDP is connection-less transport.

NOTE – Basically, the same between-gateway sequences will apply to the Internet aware fax devices which do not function as a gateway to G3FE.

II.1.2 Between Annex B/T.38 and Annex D/H.323 gateways

II.1.2.1 Normal connection and disconnection sequence (Annex B/T.38 supporting only FAX)

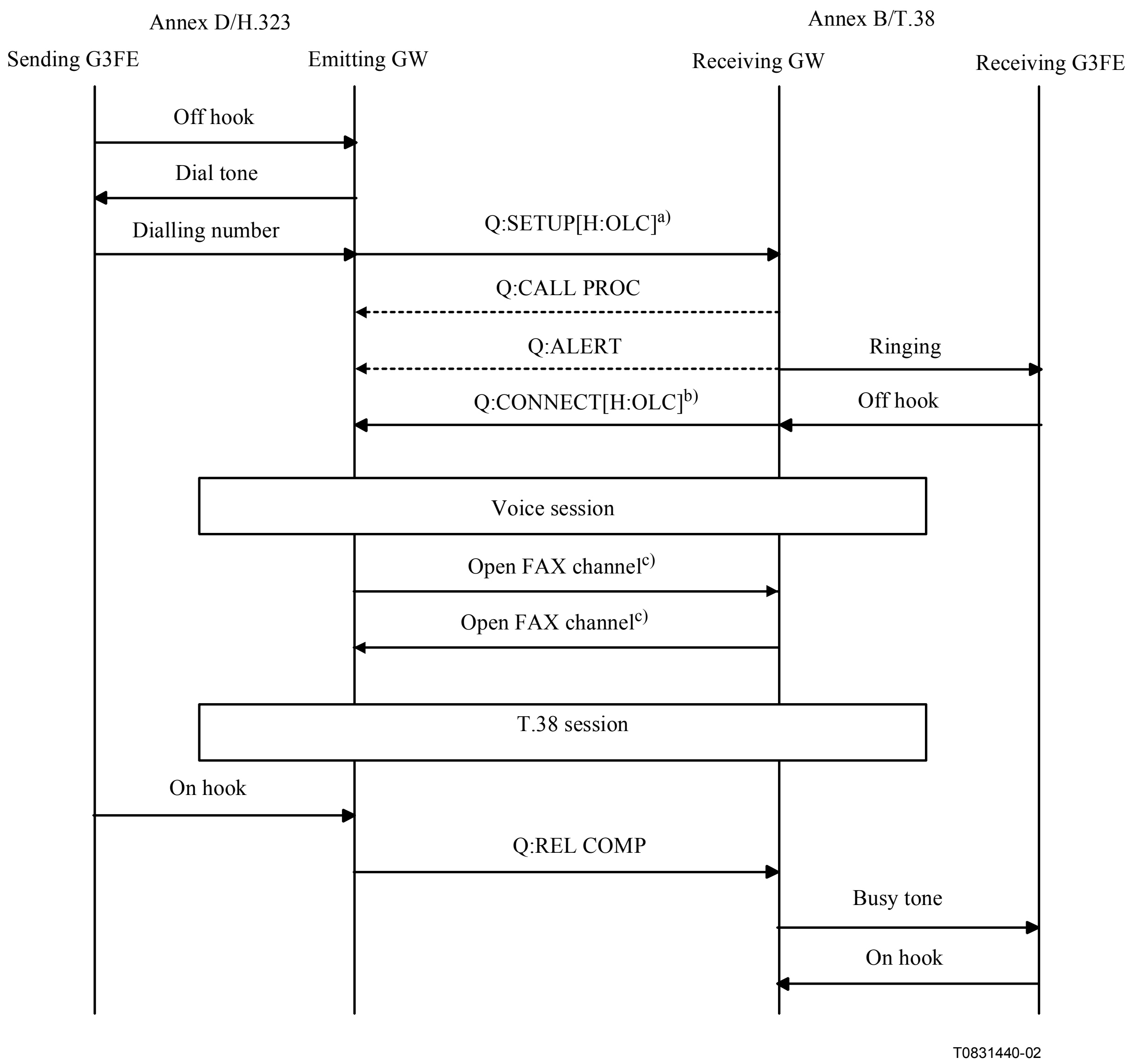


[a] Annex D/H.323 implementation uses fastStart element to send OLCs which include voice and facsimile capabilities.

[b] Annex B/T.38 implementation returns OLC which includes only facsimile capability in response to SETUP from Annex D/H.323 implementation. Note that Annex B/T.38 implementation does not return the value of H.245 port.

[c] Annex D/H.323 implementation needs to open H.245 Channel to exchange capabilities which have not been sent. Thus, it sends a Facility message with a FacilityReason of startH245 to facilitate opening H.245 channel with the peer. In response, Annex B/T.38 implementation returns a Facility message with a FacilityReason of noH245 to indicate that it does not support H.245 operation. This sequence allows FAX communication without opening H.245 Channel when Annex D/H.323 implementation does not need a voice channel.

II.1.2.2 Normal connection and disconnection sequence (Annex B/T.38 supporting FAX and voice)



[a] Annex D/H.323 implementation uses fastStart element to send OLC, which includes voice capability as minimum.

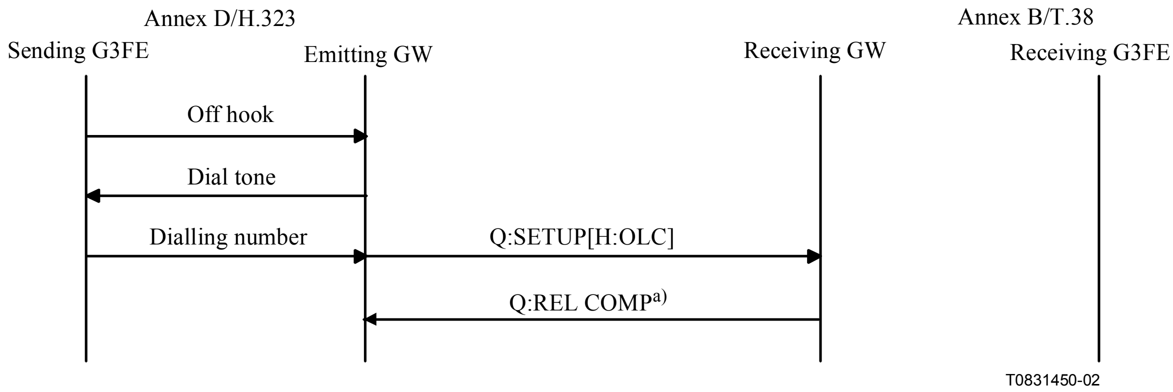
[b] Annex B/T.38 implementation returns OLCs which include both voice and facsimile capabilities in response to SETUP from Annex D/H.323 implementation. Note that Annex B/T.38 implementation supporting voice and FAX is capable of ITU-T H.245 procedures.

[c] This opens the FAX channel negotiated by exchanging OLCs in ITU-T H.245 procedures from both directions. Note that variables such as the voice conversation, CNG, CED and V.21 signals (which do not appear in the figure) will trigger the sequence. Annex D/H.323 and Annex B/T.38 implementations need to recognize T.30 signals (such as CNG, CED & V.21) sent from the peer terminal. These cannot be forwarded via T.38 until the FAX channel is opened.

NOTE 1 – Annex B/T.38 supporting FAX and optional voice shall use the methods in Annex D/H.323 as described in B.3.1.1. Thus, the above figure shows the sequences conforming to Annex D/H.323.

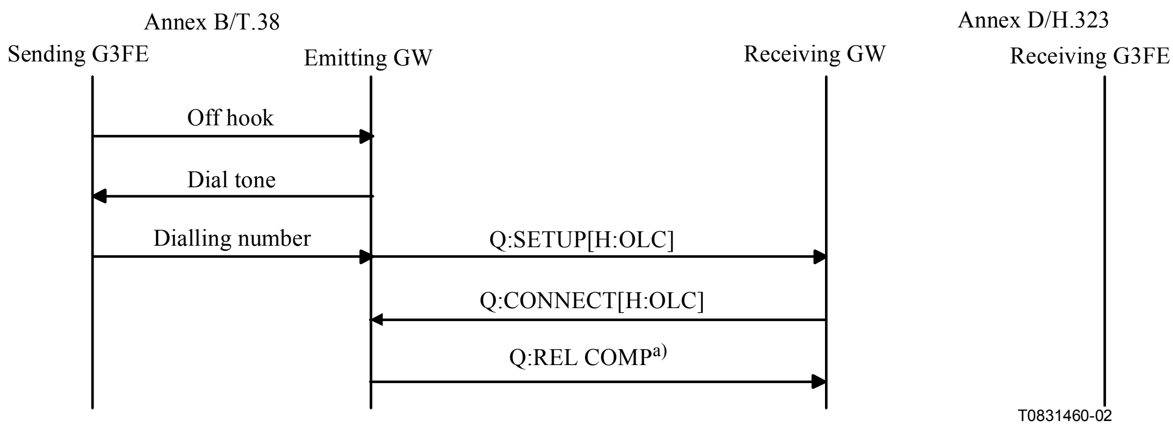
NOTE 2 – The switching mechanism should refer to section "D.5 replacing an existing audio stream with a T.38 fax stream" in Annex D/H.323.

II.1.2.3 Connection rejected sequence 1 (when the calling side, Annex D/H.323, does not support Fast Connect Procedures)



[a)] Annex B/T.38 implementation rejects connection by sending Q.931: RELEASE COMPLETE when it receives SETUP message without fastStart element.

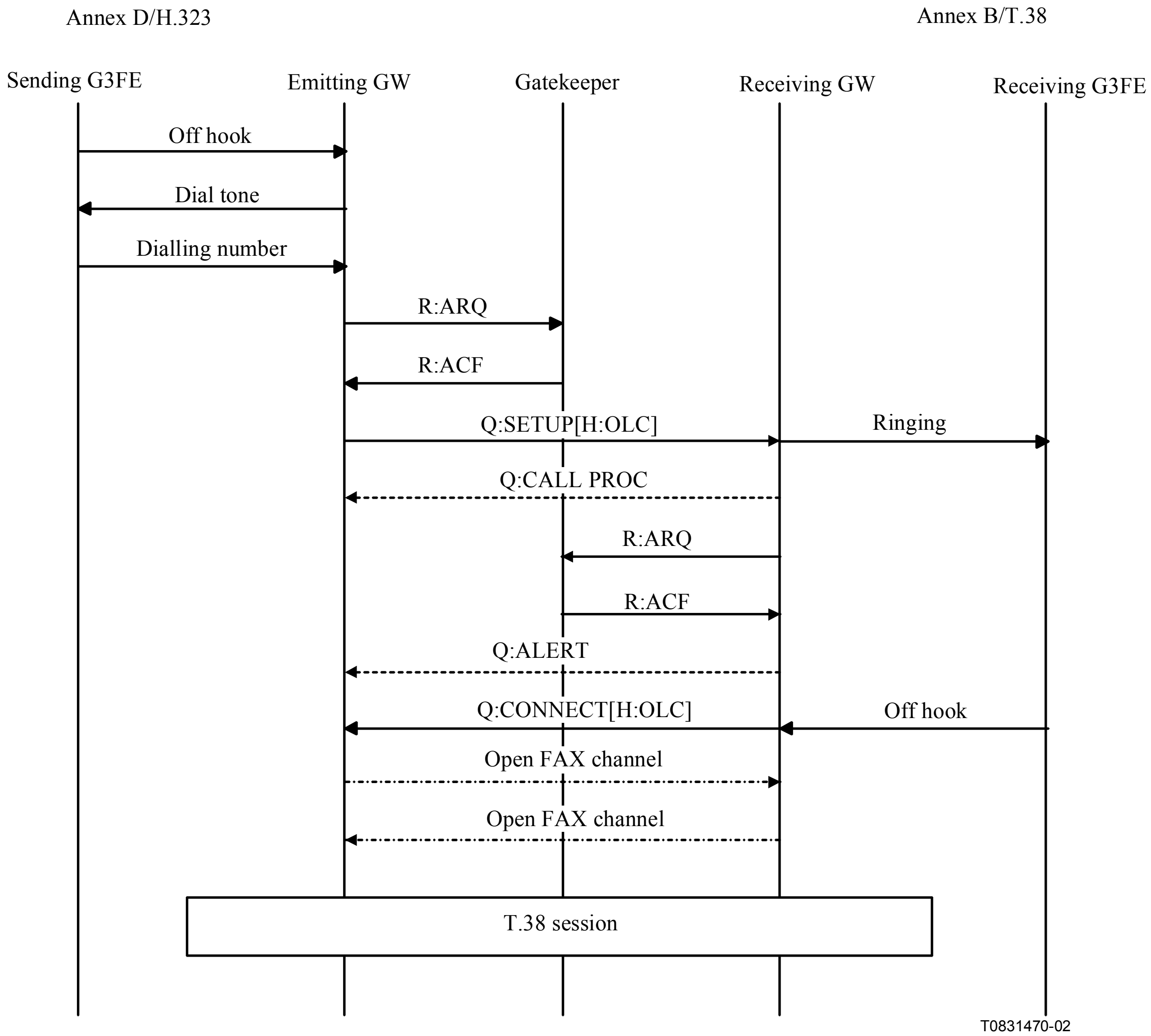
II.1.2.4 Connection rejected sequence 2 (when the called side, Annex D/H.323, does not support Fast Connect Procedures)



[a)] Annex B/T.38 implementation rejects connection by sending Q.931: RELEASE COMPLETE when it receives CONNECT message without fastStart element in response to its SETUP message with fastStart element.

II.1.3 Between Annex B/T.38 supporting FAX and Annex D/H.323 gateways that are both registered to the same Gatekeeper

II.1.3.1 Normal connection sequence (when Gatekeeper has chosen Direct Call Signalling)



R RAS (Registration, Admission and Status) messages

NOTE – Various call models are described in 8.1/H.323.

II.2 Protocol data used in call establishment procedures

II.2.1 General

Two Recommendations – ITU-T H.225.0 (as a subset of ITU-T Q.931) and ITU-T H.245 – define the protocol data used in Call establishment procedures of Annex B/T.38, while ITU-T Rec. H.323 gives the general protocol design of the whole system. For example SETUP message is defined in Table 13/H.225.0 and its User-user Information Element (UUIE) is defined as Setup-UUIE under H323-UU-PDU in ITU-T Rec. H.225.0. Then fastStart element which is defined as SEQUENCE OF OCTET STRING by ASN.1 definition of Setup-UUIE encapsulates OpenLogicalChannel which is defined under MultimediaSystemControlMessage in ITU-T Rec. H.245.

Additionally, RAS messages need to be understood to fully implement Annex B/T.38. RAS is also defined in ITU-T Rec. H.225.0 as RasMessage using ASN.1 and Table 18/H.225.0 gives its support requirements.

II.2.2 Examples of the protocol data

II.2.2.1 Supported H.225.0 (Q.931) message types

Tables II.1 to II.3 show the supported H.225.0 (Q.931) message types in three phases.

Table II.1/T.38 – Messages of call setup phase

Message type	transmit	receive
ALERT	CM[a)]	M
CALL PROC	CM[a)]	M
CONNECT	M	M
CONNECT ACK	F	F
PROGRESS	O	O
SETUP	M	M
SETUP ACK	O	O

M Mandatory

O Optional

F Forbidden

CM Conditional Mandatory

a) Note that gateways shall send ALERT and CALL PROC messages while IAF (Internet Aware Fax) may not send them. Note that an Annex D/H.323 GW may send ALERTING or CALL PROC to an IAF.

Table II.2/T.38 – Messages of call release phase

Message type	transmit	receive
DISCONNECT	F	F
RELEASE	F	F
RELEASE COMP	M	M

Table II.3/T.38 – Messages of other phase

Message type	transmit	receive
FACILITY	CM[a)]	M[a)]

a) Note that Annex B/T.38 implementation shall receive and send FACILITY when connecting to Annex D/H.323 implementation.

II.2.2.2 Information elements of SETUP

Tables II.4 to II.6 show information elements of SETUP message.

Table II.4/T.38 – Information elements of SETUP

Information element	Parameter	Status	Description
Protocol discriminator	Reference H.225.0	M	
Call reference	Reference H.225.0	M	
Message type	Reference H.225.0	M	
Bearer capability	Reference H.225.0	M	

Table II.4/T.38 – Information elements of SETUP

Information element	Parameter	Status	Description
Calling party number	Reference H.225.0	O	
Calling party subaddress	Reference H.225.0	CM	
Called party number	Reference H.225.0	O	
Called party subaddress	Reference H.225.0	CM	
User-user	protocolIdentifier	M	H.225.0 version number
	SourceInfo	M	EndpointType
	destinationAddress	M	Used by Gatekeeper
	destCallSignalAddress	M	TransportAddress (IP address + Port number)
	activeMC	M	FALSE
	conferenceID	M	NULL
	conferenceGoal	M	NULL
	callType	M	PointToPoint
	callIdentifier	M	GloballyUniqueID
	mediaWaitForConnect	M	TRUE
	canOverlapSend	M	if TRUE, support overlap sending
	fastStart	M	Reference Table II.5

Table II.5/T.38 – Parameters of fastStart(OpenLogicalChannel)

Parameters	Description
ForwardLogicalChannelNumber	
ForwardLogicalChannelParameters	
PortNumber	
DataType	Reference Table II.6 dateType is linked with DataApplicationCapability in Annex B/T.38 Note that DataApplicationCapability in Annex B/T.38 is only extraction among CHOICEs of application of H.245.
MultiplexParameters	sessionID, mediaChannel and mediaControlChannel in H2250LogicalChannelParameters
ReverseLogicalChannelParameters	
DataType	Reference Table II.6 dateType is linked with DataApplicationCapability in Annex B/T.38 Note that DataApplicationCapability in Annex B/T.38 is only extraction among CHOICEs of application of H.245.
MultiplexParameters	sessionID, mediaChannel and mediaControlChannel in H2250LogicalChannelParameters

Table II.6/T.38 – Parameters of dataType(DataApplicationCapability)

Parameter	Status	Description
Application	–	CHOICE index shall be encoded to indicate the use of t38fax.
t38fax	M	
t38FaxProtocol	M	CHOICE index of DataProtocolCapability shall be encoded to indicate the use of tcp or udp.
t38FaxProfile	M	
FilBitRemoval	M	
TranscodingJBIG	M	
TranscodingMMR	M	
Version	M	
t38FaxRateManagement	M	CHOICE index shall be encoded to indicate the use of localTCF or transferredTCF.
t38FaxUdpOptions	O	
t38FaxMaxBuffer	O	
t38FaxMaxDatagram	O	
t38FaxUdpEC	O	CHOICE index shall be encoded to indicate the use of t38UDPFEC or t38UDPRedundancy.
MaxBitRate	M	Units 100 bit/s

II.2.2.3 Information elements of ALERT

Table II.7 shows information elements of ALERT message.

Table II.7/T.38 – Information elements of ALERT

Information element	Parameter	Status	Description
Protocol discriminator	Reference H.225.0	M	
Call reference	Reference H.225.0	M	
Message type	Reference H.225.0	M	
User-user	Reference H.225.0	M	

II.2.2.4 Information elements of CALL PROC

Table II.8 shows information elements of CALL PROC message.

Table II.8/T.38 – Information elements of CALL PROC

Information element	Parameter	Status	Description
Protocol discriminator	Reference H.225.0	M	
Call reference	Reference H.225.0	M	
Message type	Reference H.225.0	M	
User-user	Reference H.225.0	M	

II.2.2.5 Information elements of CONNECT

Table II.9 shows information elements of CONNECT message.

Table II.9/T.38 – Information elements of CONNECT

Information element	Parameter	Status	Description
Protocol discriminator	Reference H.225.0	M	
Call reference	Reference H.225.0	M	
Message type	Reference H.225.0	M	
User-user	protocolIdentifier	M	H.225.0 version number
	destinationInfo	M	EndpointType
	conferenceID	M	NULL
	callIdentifier	M	GloballyUniqueID
	FastStart	M	Reference Table II.5

II.2.2.6 Information elements of RELEASE COMPLETE

Table II.10 shows information elements of RELEASE COMPLETE message.

Table II.10/T.38 – Information elements of RELEASE COMPLETE

Information element	Parameter	Status	Description
Protocol discriminator	Reference H.225.0	M	
Call reference	Reference H.225.0	M	
Message type	Reference H.225.0	M	
Cause	Reference H.225.0	CM	Either the Cause IE or ReleaseCompleteReason in User-user shall be present.
User-user	Reference H.225.0	M	

II.2.2.7 Information elements of FACILITY

Table II.11 shows information elements of FACILITY message.

Table II.11/T.38 – Information elements of FACILITY

Information element	Parameter	Status	Description
Protocol discriminator	Reference H.225.0	M	
Call reference	Reference H.225.0	M	
Message type	Reference H.225.0	M	
User-user	protocolIdentifier	M	H.225.0 version number
	reason	M	FacilityReason
	callIdentifier	M	GloballyUniqueID

Appendix III

H.248 call establishment procedure examples for facsimile capable media gateways

III.1 Introduction

This appendix describes an example of the procedures for internet-aware facsimile implementations and internet aware facsimile gateways conforming to ITU-T Rec. T.38 to establish calls with other T.38 implementations using the procedures defined by T.38 Annex E and ITU-T Recs H.248 sub-series.

III.2 Examples of call setup

III.2.1 Voice to fax call setup with H.248 endpoints

This call flow example describes a voice call that originates and terminates in the SCN and is transported through the packet network. The packet network signalling in this example is not specified but any signalling protocols such as H.323 or SIP can be used – the purpose of the example is to describe MG/MGC interactions including the detection of fax and switching from voice to fax. See Figure III.1.

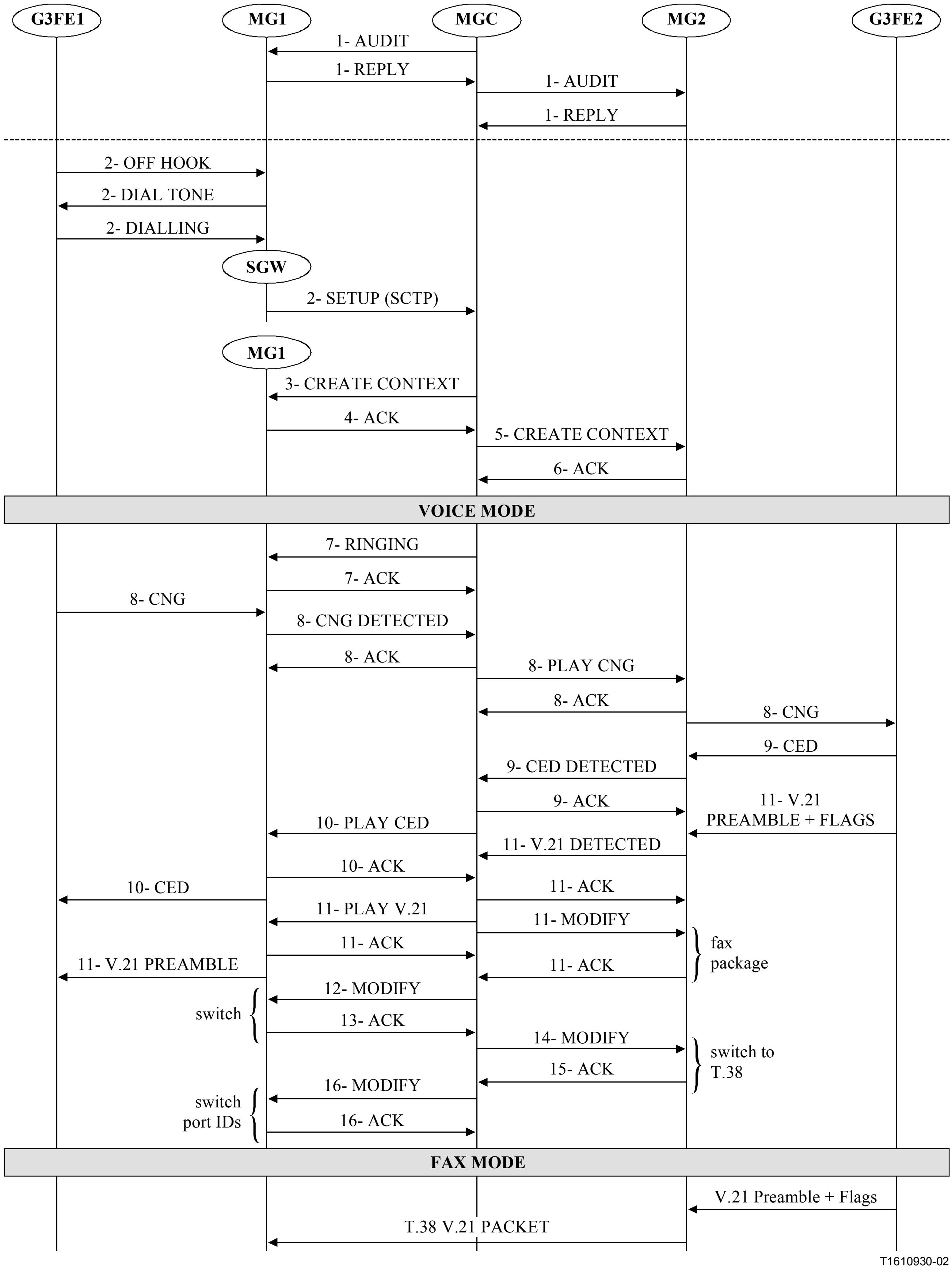


Figure III.1/T.38 – Voice to fax call setup with H.248 endpoints

The sequence of events is as follows:

1) At some point before a call, the Media Gateway Controller (MGC) will have issued an audit capabilities command to the Media Gateways under its control and will know what the voice and fax capabilities are for each gateway. In the scenarios below, if both Media Gateways support T.38, then this is the preferred mode for IP fax operations. In the event that one or both media gateways do not support T.38, then the fax call may proceed over the IP voice channel. However, since T.30 facsimile may fail over a compressed voice codec, it would be preferable to use a G.711 codec for communication between the media gateways. 'W-' is used to indicate that a wild card answer with a union of information on all terminations on the MG is requested, not an audit of each termination on the MG.

The MGC audits MG1.

MGC to MG1:

```
MEGACO/1.0 [123.123.123.4]:55555
Transaction = 10 {
  Context = - {W-AuditValue = * {Audit{Media, Packages}}}
}
```

MG1 replies. MG1 to MGC:

```
MEGACO/1.0 [125.125.125.111]:55555
Reply = 10 {
 Context = - {
  AuditValue = * {
   Media {
    Stream = 1 {
     Local  {
v=0
c=IN IP4 $
m=audio $ RTP/AVP 4
v=0
c=IN IP4 $
m=audio $ RTP/AVP 0
v=0
c=IN IP4 $
m=image $ udptl t38
      } ; RTP profile for G.711 is 0, G.723 is 4, t38 is T.38
     }
    },
   Packages {al, rtp, ipfax, fax, ctyp, cg}
   ; al = analog line pkg, rtp = rtp pkg, ipfax = T.38 fax pkg, fax = fax pkg
   ; ftmd = fax/textphone/modem tones detection pkg
   ; ctyp = Call Type Discrimination package)
   ; cg =call progress tones generator pkg
  }
 }
}
```

A similar exchange happens between the MGC and MG2.

2) The End User decides to send a fax from device F1 and enters the phone number. The fax device gets dial tone and then dials the phone number. As a result, the central office within the local SCN loop sends an SS7 message to the signalling gateway (SG). The SG sends a *Setup* message to the MGC after receiving this IAM from a SCN switch that conveys the called and calling phone numbers. Sigtran's SCTP carries the SS7 signalling from the SG to the MGC.

3) From the IAM message, the MGC may infer which circuit on which MG is involved and where to terminate the call. How the MGC does this is outside the scope of this appendix. The end points are found by the Media Gateway Controller (MGC) and it sets up the audio channel between the two media gateways and instructs the SS7 facility of the receiving CO to connect to the end phone destination, which results in the generation of ringing. So, to start, the controller determines that a connection needs to be made from MG1 to MG2. The MGC creates a Context for the call. Both the TDM termination DS0/1/1, and an RTP termination are added to a new context in MG1. Mode is ReceiveOnly since Remote descriptor values are not yet specified. Preferred codecs are in the MGC's preferred order of choice. The MGC sets to $ the fields in the sdp in Local that the MG will set itself.

MGC to MG1:

```
MEGACO/1.0 [123.123.123.4]:55555
Transaction = 11 {
  Context = $ {
    Add = DS0/1/1 {
      Events = 1 {al/on, ftmd/dtfmctyp/dtone, faxctyp/dtone{sdtt=cng},
faxctyp/dtone{dtst=cedans}, ctyp/dtone{dtt=v21flag}, al/of}
        },  ; SCN termination prepared to listen for tones
    Add = $ {
      Media {
        Stream = 1 {
          LocalControl { Mode = ReceiveOnly },
          Local {
v=0
c=IN IP4 $
m=audio $ RTP/AVP 4
v=0
c=IN IP4 $
m=audio $ RTP/AVP 0
           }; IP termination for audio
         }
       }
     }
  }
}
```

4) MG1 acknowledges the new Termination and fills in the Local IP address and UDP port. It also makes a choice for the codec based on the list of sdp blocks in Local. MG1 sets the RTP port to 2222. Note that MG1 could have sent back both codecs, to leave the final choice to MG2.

```
MEGACO/1.0 [124.124.124.222]:55555
Reply = 11 {
  Context = 2000 {
    Add = DS0/1/1,  ; SCN termination added
    Add = RTP/1 {
      Media {
        Stream = 1 {
          Local {
v=0
c=IN IP4 124.124.124.222
m=audio 2222 RTP/AVP 4
          } ; IP termination added
        }
      }
    }
  }
}
```

5) Assume that the MGC has control over the remote MG2 also. The MGC will now associate DS0/2/2 with a new Context on MG2, and establish an RTP Stream (i.e. RTP/2/2 will be assigned), SendReceive connection through to the originating user, User 1. MG1 is only offering G.723 (see Remote), so MGC only offers that to MG2.

MGC to MG2:

```
MEGACO/1.0 [123.123.123.4]:55555
Transaction = 30 {
  Context = $ {
    Add = DS0/2/2 {
      Media {
        Stream = 1 {
          LocalControl {Mode = SendReceive } } },
      Events = 10 {al/of, ftmd/dtfmctyp, faxctyp/dtone{sdtt=cng},
faxctyp/dtone{sdtt=ansced}, ctyp/dtone{dtt=v21flag}, al/on },
      Signals = {al/ri, ctyp/callsig, ctyp/ans}
      }
    },
    Add  = $ {
      Media {
        Stream = 1 {
          LocalControl {Mode = SendReceive },
          Local {
v=0
c=IN IP4 $
m=audio $ RTP/AVP 4
          },
          Remote {
v=0
c=IN IP4 124.124.124.222
m=audio 2222 RTP/AVP 4
          } ; RTP profile for G.723 is 4
        }
      }
    }
  }
}
```

6) This is acknowledged. The stream port number is different from the Megaco/H.248 control port number. In this case it is 1111 (in the SDP).

MG2 to MGC:

```
MEGACO/1.0 [125.125.125.111]:55555
Reply = 30 {
  Context = 5000 {
    Add = DS0/2/2,
    Add = RTP/2 {
      Media {
        Stream = 1 {
          Local {
v=0
c=IN IP4 125.125.125.111
m=audio 1111 RTP/AVP 4
          }
        }
      }
    }
  }
}
```

7) The above IPAddr and UDPport need to be given to MG1 now. Also apply ringback tone to the DS0/1/1 termination and change it to a SendReceive.

MGC to MG1:

```
MEGACO/1.0 [123.123.123.4]:55555
Transaction = 12 {
    Context = 2000 {
       Modify = DS0/1/1 {

Signals {cgal/rtri} }, ;apply ringing tone
       Modify = RTP/1 {
           Media {
             Stream = 1 {
             LocalControl {Mode = SendReceive }
                  Remote {
v=0
c=IN IP4 125.125.125.111
m=audio 1111 RTP/AVP 4
                  }
              }
           }
       }
    }
}
```

from MG1 to MGC:

```
MEGACO/1.0 [124.124.124.222]:55555
Reply = 12 {
   Context = 2000 {Modify = DS0/1/1, Modify = RTP/1}
}
```

8) The calling fax machine typically will start to generate CNG calling tones. It is envisioned that the CNG tone event will be detected by the first Media Gateway (MG1). This event will be reported to the Media Gateway Controller. The Media Gateway Controller then should issue a command to the second Media Gateway (MG2) to generate a CNG tone. At this point, the full duplex channel is still in a voice mode and uses the indicated audio codec such as G.723.1 and G.729A.

from MG1 to MGC:

```
MEGACO/1.0 [124.124.124.222]:55555
Transaction = 50 {
   Context = 2000 {
    Notify = DS0/1/1 {
     ObservedEvents = 1 {
       19991212T22110001:faxctyp/dtone{dtst=cng} }
    }
   }
}
```

from MGC to MG1:

```
MEGACO/1.0 [123.123.123.4]:55555
Reply = 50 {
   Context = 2000 {Notify = DS0/1/1}
}
```

MGC to MG2:

```
MEGACO/1.0 [123.123.123.4]:55555
Transaction = 31 {
  Context = 5000 {
    Modify = DS0/2/2 {
      Signals {faxctyp/callsig{callSigname=cng}}; issue CNG at remote end
    }
  }
}
```

MG2 to MGC:

```
MEGACO/1.0 [125.125.125.111]:55555
Reply = 31 {
  Context = 5000 {Modify = DS0/2/2}
}
```

9) In the previous step, the MG2 generated a CNG tone that the MGC asked it to in the Signals descriptor. In the typical case, if the final destination phone number is fax capable, this will result in the issuance of a CED tone by the recipient fax machine. This step is illustrated here. However, if there is not a fax receiver on the line, the typical response will be via voice.

from MG2 to MGC:

```
MEGACO/1.0 [123.123.123.4]:55555
Transaction = 70 {
   Context = 5000 {
    Notify = DS0/2/2 {
     ObservedEvents = 10 {
       19991212T22110031:faxctyp/dtone{dtst=ANSced}}; CED and ANS are
equivalent. Reported under the name ANS.
    }
   }
}
```

from MGC to MG2:

```
MEGACO/1.0 [125.125.125.111]:55555
Reply = 70 {
  Context = 5000 {Notify = DS0/2/2}
}
```

10) Assuming that a CED has been generated by the recipient fax device, the MG1 will then receive the CED and uses its tone detection algorithms to detect that it actually is a CED.

NOTE – Some research was done to check out modem answer tones, which are defined in ITU-T Recs V.25 and V.8. The modem answer tone without phase reversals is known as ANS in V.25 and with answer tones is ANS (with a bar on top denoting Phase reversals).

Some modems and DSPs may have a difficult time distinguishing among the CED, ANS and ANS(bar). However, the group felt that if a CED like tone was generated in response to a CNG, there is a very high likelihood that the tone is indeed a CED and not one of the ANS tones. Higher end modems are able to discriminate between ANSam and the other modem and fax tones. Since a CNG was reported by the calling side and a CED by the called side, the Media Gateway Controller will issue a command instructing MG1 to play the CED. Both media gateways shift into a fax mode (either T.38 if supported, or G.711). From this point, the V.21 fax data will be conveyed between the media gateways. Note that at this point, the MGC could decide that there is sufficient confidence to switch to fax, unless, for example, some other answer tone has been detected, such as ANSam (see step 18). For the purpose of this example, it does not believe it has sufficient confidence.

MGC to MG1:

```
MEGACO/1.0 [123.123.123.4]:55555
Transaction = 13 {
  Context = 2000 {
    Modify = DS0/1/1 {
      Signals {faxctyp/ans{anstype=ansced}}
    }
  }
}
```

MG1 to MGC:

```
MEGACO/1.0 [124.125.125.222]:55555
Reply = 13 {
  Context = 2000 {Modify = DS0/1/1}
}
```

11) When MG2 detects a V.21 carrier followed by flags, it will send a message to the MGC reporting this event. At this point, the MGC is certain that the call is a fax and will initiate a switch, first on the DS0 terminations. Note that V.21 flags are not signalled to MG1. The event causes the MGC to ask MG1 to play v21flags to its SCN termination.

MG2 notifying MGC of V.21 carrier event:

from MG2 to MGC:

```
MEGACO/1.0 [123.123.123.4]:55555
Transaction = 71 {
   Context = 5000 {
    Notify = DS0/2/2 {
     ObservedEvents = 10 {
       19991212T22110031:ctyp/dtone{dtst=v21flag}}
    }
   }
}
```

MGC responds.

from MGC to MG2:

```
MEGACO/1.0 [125.125.125.111]:55555
Reply = 71 {
  Context = 5000 {Notify = DS0/2/2}
}
```

MGC sends a command to MG1 to send the V.21 flags on its SCN termination and hand over the continuation of the session to the fax package.

MGC to MG1:

```
MEGACO/1.0 [123.123.123.4]:55555
Transaction =  5{
  Context = 2000 {
    Modify = DS0/1/1 {
      Signals {ctyp/ans{anstype=v21flags, SignalType=TimeOut}}
  Events = 2 { fax/faxconnchange}
  Media{
    Stream=1{
     LocalControl
      {fax/faxstate = Train;
      }
         }
     }
    }
```

```
    }
  }
```

MG1 to MGC:

```
MEGACO/1.0 [124.125.125.222]:55555
Reply = 5 {
  Context = 2000 {Modify = DS0/1/1}
```

The MG must generate the V.21 flags signal until the V.21 flags indication arrives in the T.38 media stream (see step 17) and then continue until the V.21 flags termination is indicated in the T.38 media stream.

12) At this point the SCN termination on MG2 and MG1 shall be put into fax mode (this stage is Negotiating). Only the example of MG2 is shown. Note that in the case of MG2, since the ctyp package is not mentioned in the Events Descriptor the MG is no longer required to perform call type discrimination event notification. Also, since CNG is not mentioned in the signal descriptor, this signal is terminated.

MGC to MG2:

```
MEGACO/1.0 [123.123.123.4]:55555
Transaction =  33{
  Context = 5000 {
    Modify = DS0/2/2 {
  Events = 12 { fax/faxconnchange}
  Signals{},
  Media{
    Stream=1{
     LocalControl
      {fax/faxstate = TrainNegotiating;
      }
         }
     }
    }
  }
}
```

And MG2 responds.

MG2 to MGC:

```
MEGACO/1.0 [125.125.125.111]:55555
Reply = 33 {
  Context = 5000 {Modify = DS0/2/2}
```

13) At this point in the call, the switch to fax continues with a request for each MG to switch to T.38 mode. Note that the MGC is aware that the MGs support T.38 as a result of a previous audit. If T.38 was not available, then the audio mode may be changed to G.711 (the details of this are outside the scope of this Recommendation). Selection among the voice, fax and data modes will have been achieved, unless some other answer tone has been detected, such as ANSam. In the event that ANSam is detected, the two MGs should be switched into a mode where they can conduct a V.8 session to further determine the type of call (e.g. V.34 fax, V.90 data, text telephone, etc.) The handling of V.34 fax calls in this environment is for further study.

MGC to MG1:

```
MEGACO/1.0 [123.123.123.4]:55555
Transaction = 15 {
  Context = 2000 {
    Modify = RTP/1 {
      Media {
        Stream = 1 {
          LocalControl
     {ipfax/faxstate = Negotiating;
     }
          Local {
v=0
c=IN IP4 124.124.124.222
m=image 2222 udptl t38
a=T38FaxRateManagement:transferredTCFlocalTCF
a=T38FaxUdpEC:t38UDPFEC
          } ; change to T.38 in the IP connection
        }
      }
    }
  }
}
```

14) Following is the response from MG1. MG1 changes one of the a = fields: The T.38 parameter transferredTCF is changed by MG1 to localTCF. MG1 may also change the port number if it does not want to switch the use of the existing voice channel to faxport. In this example, it changes the port from 2222 to 3333.

from MG1 to MGC:

```
MEGACO/1.0 [124.124.124.222]:55555
Reply = 15 {
  Context = 2000 {Modify = RTP/1 {
    Media {
      Stream = 1 {
        Local {
v=0
c=IN IP4 124.124.124.222
m=image 3333 udptl t38
a=T38FaxRateManagement:localTCF
a=T38FaxUdpEC:t38UDPFEC
        }; the IP connection brought into fax mode
      }
    }
  } }
}
```

15) The new media information must be passed on to MG2.

MGC to MG2:

```
MEGACO/1.0 [123.123.123.4]:55555
Transaction = 32 {
  Context = 5000 {
    Modify  = RTP/2 {
      Media {
        Stream = 1 {
          LocalControl
     {ipfax/faxstate = Negotiating;
     }
          Local {
v=0
c=IN IP4 125.125.125.111
```

```
m=image 1111 udptl t38
a=T38FaxRateManagement:localTCF
a=T38FaxUdpEC:t38UDPFEC
          },
          Remote {
v=0
c=IN IP4 124.124.124.222
m=image 3333 udptl t38
a=T38FaxRateManagement:localTCF
a=T38FaxUdpEC:t38UDPFEC
          }
        }
      }
    }
  }
}
```

16) This is acknowledged. MG2 elects NOT to change the port (it remains as 1111), and does not change any T.38 parameters.

MG2 to MGC:

```
MEGACO/1.0 [125.125.125.111]:55555
Reply = 32 {
  Context = 5000 {
    Modify = RTP/2 {
      Media {
        Stream = 1 {
          Local {
v=0
c=IN IP4 125.125.125.111
m=image 1111 udptl t38
a=T38FaxRateManagement:localTCF
a=T38FaxUdpEC:t38UDPFEC
           }
         }
       }
     }
  }
}
```

17) Now, MG1 needs to be given the new media information from MG2.

MGC to MG1:

```
MEGACO/1.0 [123.123.123.4]:55555
Transaction = 15 {
    Context = 2000 {
      Modify = RTP/1 {
           Media {
             Stream = 1 {
                  Remote {
v=0
c=IN IP4 125.125.125.111
m=image 1111 udptl t38
a=T38FaxRateManagement:localTCF
a=T38FaxUdpEC:t38UDPFEC
                  }
              }
           }
       }
    }
}
```

from MG1 to MGC:

```
MEGACO/1.0 [124.124.124.222]:55555
Reply = 15 {
   Context = 2000 { Modify = RTP/1}
}
```

The fax call will now proceed in T.38 mode between the MGs. The first from MG2 message will be a T.30 Indicator packet containing V.21 flags. This will be generated by the continued appearance of this signal on the DS0 since the MG has no memory of previous events.

Note that the event/faxconnchange is left on the event list of both MGs and so each state change will result in a notify to MCG; however, MCG need not explicitly set the fax/faxstate in response since faxstate should be set implicitly by each MG as it changes state. MCG may take no action for most state changes but will likely take appropriate action for state such as Disconnect.

18) Variant: In the event that a CED or similar tone is detected by the MG2, it will always report this to the MGC. For the case where the MGC had not previously received notification of CNG detection by MG1, the group felt that it is still not clear whether a fax or data mode is applicable. However, the compressed voice codecs are not adequate in either case, so the MGC should place both MGs into a data capable mode (i.e. G.711) or wait for additional tones to further discriminate the call.

19) if the MG2 has the facility to detect a V.21 carrier followed by flags, it will send a message to the MGC reporting this event. (It is assumed that MGs will generally not have memory of prior events, so that the notification of V.21 and flags will be sent even if the MGC has already placed the two MGs into a fax mode.) If the MGC had not previously placed the two MGs into a fax mode, it will do so now. If the MGs are already in a G.711 mode, the MGC shall have the choice of not requesting a mode change or, requesting that the two media gateways switch to a T.38 mode.

MG2 notifying MGC of V.21 carrier event:

from MG2 to MGC:

```
MEGACO/1.0 [123.123.123.4]:55555
Transaction = 4 {
   Context = 5000 {
    Notify = DS0/2/2 {
     ObservedEvents = 10 {
       19991212T22110031:fax/dtone{st=v21flag}}
     }
    }
}
```

20) Variant: At this point in the call, the selection among the voice, fax and data modes will have been achieved, unless some other answer tone has been detected, such as ANSam. In the event that ANSam is detected, the two MGs should be switched into a mode where they can conduct a V.8 session to further determine the type of call (e.g. V.34 fax, V.90 data, text telephone, etc.) The handling of V.34 fax calls in this environment is for further study.

MG notifying MG2 of an ANSam event:

from MG2 to MGC:

```
MEGACO/1.0 [123.123.123.4]:55555
Transaction = 4 {
   Context = 5000 {
    Notify = DS0/2/2 {
     ObservedEvents = 10 {
       19991212T22110031:faxctyp/dtone{dtst=ansam}}
     }
    }
}
```

III.2.2 Fax only call setup between H.248.1 and an H.323 endpoint

This facsimile only call flow example describes a facsimile call that originates in the SCN and is terminated in the packet network. The packet network signalling in this example is H.323 but other signalling protocols such as SIP can be used, the purpose of the example is to describe MG/MGC interactions.

The assumption is made that the signalling between the signalling gateway (SGW) and MGC is based on ITU-T Rec. Q.931. This does not indicate that no other signalling can be used on this interface. Capabilities described here are generic line package descriptions (but could also be SDP or H.245 messages).

The media gateway is configured for voice and fax; however, the H.323 endpoint is fax only taking the calls into fax only mode. (i.e. it is likely a T.38 Annex B endpoint). See Figure III.2.

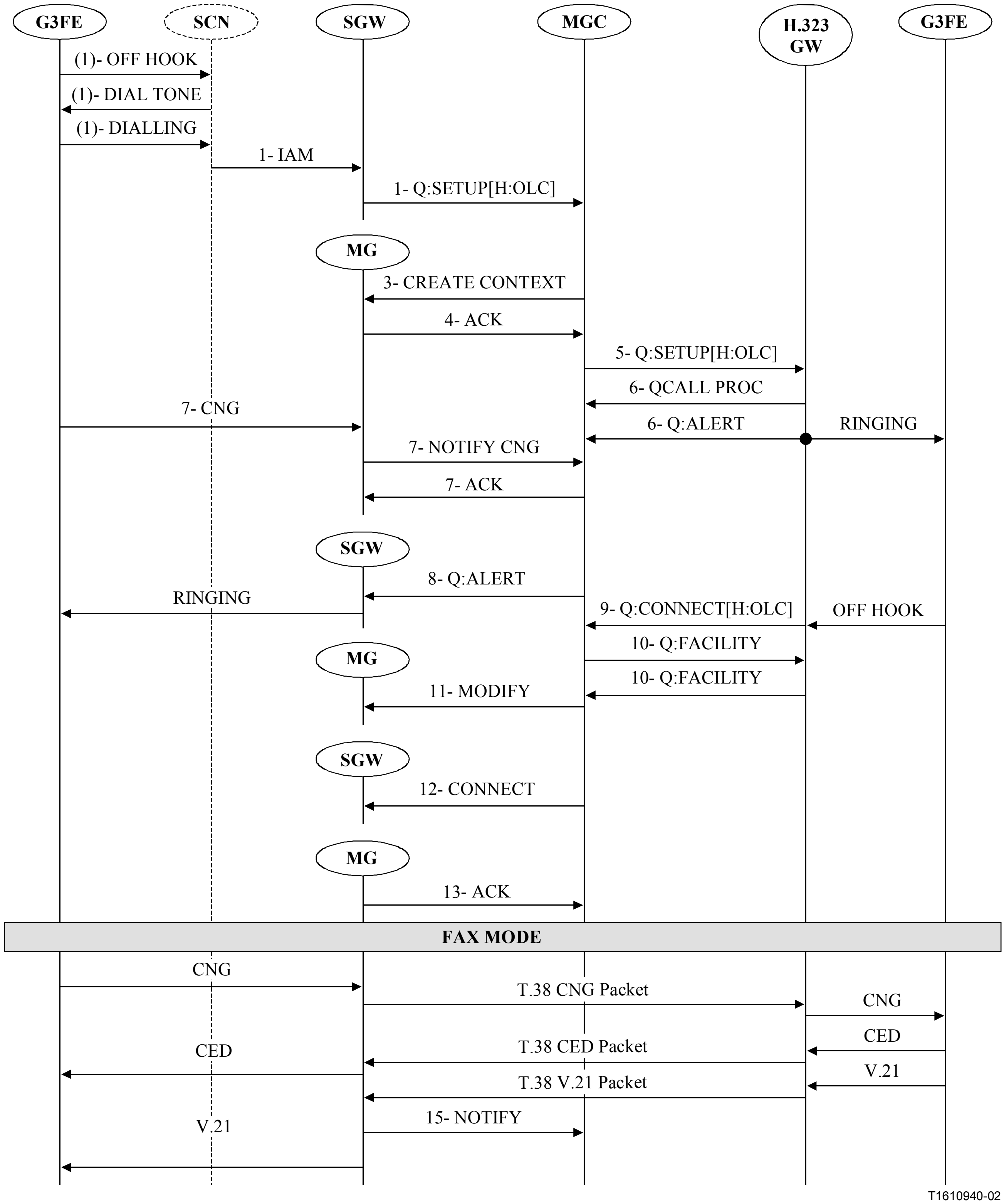


Figure III.2/T.38 – Fax only call setup between H.248 and an H.323 endpoint

1) The SGW sends a *Setup* message to the MGC after receiving an IAM from a SCN switch.

2) From the IAM message, the MGC may infer which circuit on which MG is involved and where to terminate the call. How the MGC does this is outside the scope of this appendix.

3) The MGC creates a Context for the call. The Context contains two terminations: one for the SCN side and one for the packet side:

MGC to MG:

```
MEGACO/1.0 [123.123.123.4]:55555
Transaction = 11 {
  Context = $ {
    Add = DS0/1/1 {
      Events = 1 {al/on, ftmd/dtfmctyp/dtone, faxctyp/dtone{dtst=cng},
faxctyp/dtone{dtst=cedans},ctyp/dtone, ctyp/dtone{dtt=cng},
ctyp/dtone{dtt=ans}, ctyp/dtone{dtt=v21flag}, fax/faxconnchange, al/of}
      }, ; the SCN side termination listening for call type indicating tones
    Add = $ {
      Media {
        Stream = 1 {
          LocalControl { Mode = ReceiveOnly },
          Local {
v=0
c=IN IP4 $
m=audio $ RTP/AVP 4
v=0
c=IN IP4 $
m=audio $ RTP/AVP 0
        } ; the IP side term. showing capability of RTP audio with PT 0 and 4.
         }
        }
      }
   }
}
```

4) The MG accepts the Context creation and fills in the unknown ($) parameters:

```
MEGACO/1.0 [124.124.124.222]:55555
Reply = 11 {
  Context = 2000 {
    Add = DS0/1/1,; the SCN termination is accepted
    Add = RTP/1 {
      Media {
        Stream = 1 {
          Local {
v=0
c=IN IP4 124.124.124.222
m=audio 2222 RTP/AVP 4
          } ; the IP RTP termination is accepted with audio payload type 4.
        }
      }
    }
  }
}
```

This shows how the MG reports to the MGC what parameters it filled in.

5) The MGC sends a *Setup* message to the destination endpoint, here assumed to be a H.323 endpoint (terminal, GW, etc.). It indicates in the fastStart elementthat either the capability to use UDP or TCP may be used for the T.38 facsimile stream.

6) The H.323 endpoint sends a *CallProceeding* message followed by an *Alerting* message back to the MGC, informing it of the mode to be used (assume UDP for both directions) and the transport address; followed by an Alerting message, indicating that it is ringing the G3FE.

   The MGC sends a Modify command to the MG to set the mode and remote termination description on the packet side:

MGC to MG:

```
MEGACO/1.0 [123.123.123.4]:55555
MEGACO/1.0 [124.124.124.222]:55555
Transaction = 1450 {
   Context = 2000 {
    Modify = RTP/1 {
      Media {
        Stream = 1 {
          Local {
v=0
c=IN IP4 124.124.124.222
m=image 2222 udptl t38
a=T38FaxRateManagement:transferredTCF
a=T38FaxUdpEC:t38UDPFEC
     } ; modify media stream 1 to use image media , udptl transport for T38
      LocalControl {
     fax/faxstate=Prepare;
     fax/trpt=T38UDPTL;
     Events=fax/faxconnchange;
              }

        }
      }
    }
  }
}
```

The MG accepts the Modify commands:

from MG to MGC:

```
MEGACO/1.0 [124.124.124.222]:55555
Reply = 14 {
  Context = 2000 {Modify = RTP/1 {
    Media {
      Stream = 1 {
        Local {
v=0
c=IN IP4 124.124.124.222
m=image 3333 udptl t38
a=T38FaxRateManagement:transferredTCF
a=T38FaxUdpEC:t38UDPFEC
      }; The fax udptl/t38 transport channel is accepted on the IP session
      }
    }
  } }
}
```

7) At about this time, the MG detects a CNG on the line and notifies the MGC. Since there is no way to intiate a playing of CNG on the H.323 endpoint, the MGC will wait until the connection is open. Note that the MGC may not receive a CNG before the H.323 *Connect*.

from MG to MGC:

```
MEGACO/1.0 [124.124.124.222]:55555
Notify = DS0/1/1 {
     ObservedEvents = 1 {
       19991212T22110001:ctyp/dtone{dtt=cng} }
    }
   }
}
```

from MGC to MG:

```
MEGACO/1.0 [123.123.123.4]:55555
Reply = 50 {
   Context = 2000 {Notify = DS0/1/1}
}
```

8) The MGC sends an *Alerting* message to the SGW.

9) The called endpoint at some instance sends a *Connect* message to the MGC once the G3FE goes offhook. Note that this message only contains facsimile capabilities and does not include a H.245 port

10) In response to the *Connect*, the MGC sends a *Facility* request to the endpoint in an attempt to open a H.245 channel to exchange capabilities. This is rejected by the endpoint (since it is not supported). As a result, the MGC has no choice to proceed in fax mode. This step could have been skipped since the MGC is aware of the CNG tone, but is included in this example for completeness.

11) A Modify command to the MG to change the mode of the SCN side termination to SendRecv and to fax mode. Moreover, the indication of fax capabilities to be set up on the T.38 is also included in this command (this information was included in the *Connect* from the H.323 endpoint):

MGC to MG:

```
MEGACO/1.0 [123.123.123.4]:55555
Transaction = 30 {
  Context = $ {
    Modify = DS0/1/1 {
      Media {
        Stream = 1 {
          LocalControl { fax/faxstate = Prepare } } },
      Events = 10 {al/of,ftmd/dtfmctyp, faxctyp/dtone{st=cng},
faxctyp/dtone{st=ced}, al/on, fax/faxconnchange },
Signals = {al/ri, ctyp/ans, ctyp/callsig}
} ; modify SCN termination to reflect that we are connected through
    Modify = RTP/1 {
      Media {
        Stream = 1 {
          Local {
         v=0
         c=IN IP4 124.124.124.222
         m=image 2222 udptl t38
         a=T38FaxRateManagement:transferredTCF
         a=T38FaxUdpEC:t38UDPFEC
        } ; modify media stream 1 to use image media, udptl transport for T38
       LocalControl { Mode = SendReceive,
     ipfax/faxstate=Prepare,
     ipfax/trpt=T38UDPTL

      }
      }
     }
     Events = 2 {ipfax/faxconnchng }
    }
   }
}
```

12) The MGC sends a *Connect* message to the SGW to indicate the call is connected.

The MG accepts the Modify command.

MG to MGC:

```
MEGACO/1.0 [125.125.125.111]:55555
Reply = 30 {
  Context = 5000 {
    ModifyAdd = DS0/1/1
  }/* we have a through connection. Fax signalling can start using package
ctyp up to the V.21 flags and then the fax package with property
Transport=T30 on the SCN side. On the IP side, T38 is activated by setting
the property fax/faxstate on the IP termination to Negotiating. After the
V.21 flags have passed, the session is handled by the two terminations both
using the fax package, and the SCN termination using T.30 transport
translates to and from T.38 transport form.
}
```

13) The MG accepts the Modify commands:

from MG to MGC:

```
MEGACO/1.0 [124.124.124.222]:55555
Reply = 14 {
  Context = 2000 {
   Modify = RTP/1 {
    Media {
      Stream = 1 {
        Local {
v=0
c=IN IP4 124.124.124.222
m=image 3333 udptl t38
a=T38FaxRateManagement:transferredTCF
a=T38FaxUdpEC:t38UDPFEC
      }; The fax udptl/t38 transport channel is accepted on the IP session
      }
     }
    },
    Modify = DS0/1/1
    }; The modify is accepted on the DS0 session
}
```

At this point the call proceeds in T.38 mode between the gateways. Likely the originating G3FE is still sending CNG so this will be sent first, followed by CED from the destination G3FE. Note that, since the MG has been asked to indicate when the fax connection state changes, the MG notifies the MGC of the reception of the V.21 flags packet after the occurrence of such event.

from MG to MGC:

```
MEGACO/1.0 [124.124.124.222]:55555
Transaction = 60 {
   Context = 2000 {
    Notify = RTP/1 {
     ObservedEvents = 1 {
       19991212T22110001:ipfax/faxconnchange{faxconnchng=Negotiating }
    }
   }
}
```

from MGC to MG:

```
MEGACO/1.0 [123.123.123.4]:55555
Reply = 60 {
   Context = 2000 {Notify = RTP/1}
}
```

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