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

Amendment 2

ITU-T Recommendation T.38 – Amendment 2

(Formerly CCITT Recommendation)

ITU-T Recommendation T.38

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

AMENDMENT 2

Summary

This amendment comprises new Annexes D and E. Annex D describes system level requirements and procedures for internet aware facsimile implementations and internet aware facsimile gateways conforming to ITU-T T.38 to establish calls with other ITU-T T.38 implementations using the procedures defined in RFC 2543 (SIP) and RFC 2327 (SDP).

Annex E describes system level requirements and procedures for internet aware facsimile implementations and internet aware facsimile gateways conforming to ITU-T T.38 to establish calls with other ITU-T T.38 implementations using the procedures defined by ITU-T H.248.

Source

Amendment 2 to ITU-T Recommendation T.38 was prepared by ITU-T Study Group 8 (1997-2000) and approved under the WTSC Resolution 1 procedure on 10 February 2000.

FOREWORD

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The World Telecommunication Standardization Conference (WTSC), 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 WTSC Resolution 1.

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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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ITU-T Recommendation T.38

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

AMENDMENT 2

1) Clause 2

Add the following references:

– ITU-T H.248 (2000), *Gateway Control Protocol*.

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

– IETF RFC 2543 (1999), *SIP: Session Initiation Protocol*.

2) Clause 5

Add the following sentence at the end of the final paragraph:

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

3) New Annexes D and E, and new Appendix II

Insert the following new Annexes D and E after the existing Annex C, and the following new Appendix II after the existing Appendix I.

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 T.38 to establish calls with other ITU-T T.38 implementations using the procedures defined in IETF RFC 2543 (SIP) and IETF 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 IETF 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 IETF 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 [7], obtain the H.245 [8] gateway and user address and then use H.225.0 [9] 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 IETF RFC 2543. A voice connection is then established.

Upon detection of facsimile by the emitting gateway, a SIP INVITE request is sent to the receiving 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.

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.

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/T.38.

The IETF 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 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 (IETF 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 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
```

NOTE – These values will require registration with IANA.

D.2.3.1 Declaration of T.38 in SDP

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

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

  NOTE 1 – This will also require the registration of UDPTL with IANA as a valid name for the prototype per the procedure noted in Appendix B of SDP (IETF RFC 2327).

– indicate TCP (transmission control protocol) as a valid transport value (third field).

  NOTE 2 – This will also require the registration of TCP with IANA as a valid name for the prototype per the procedure noted in Appendix B of SDP (IETF RFC 2327).

– include t38 as a valid format type value (fourth field).

  NOTE 3 – As this is not an RTP-defined value, it has to be a MIME sub-type of the media type. As a result, this will require the registration of image/t38 with IANA as a valid MIME content-type per the procedure noted in Appendix B of SDP (IETF 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.

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=T38FaxRateMgmnt:transferredTCF
        a=T38FaxUdpEC:t38UDPFEC
```

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 IETF 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 (IETF RFC 2327, [6]). 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.

Meaning	SIP Response Mapping
Busy1. Subscriber busy tone as defined in ITU-T 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 Q.35.	600 Busy everywhere
Ring1. Ringing tone as defined in ITU-T 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 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 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 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 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 IETF RFC 2543 section 2:

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

DTMF transmission during an established voice and facsimile connection is for further study.

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 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 T.38 to establish calls with other ITU-T T.38 implementations using the procedures defined by ITU-T H.248.

E.2 Communication between gateways

E.2.1 Overview

E.2.1.1 Gateway architecture

The method described in this annex is intended to be used 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).

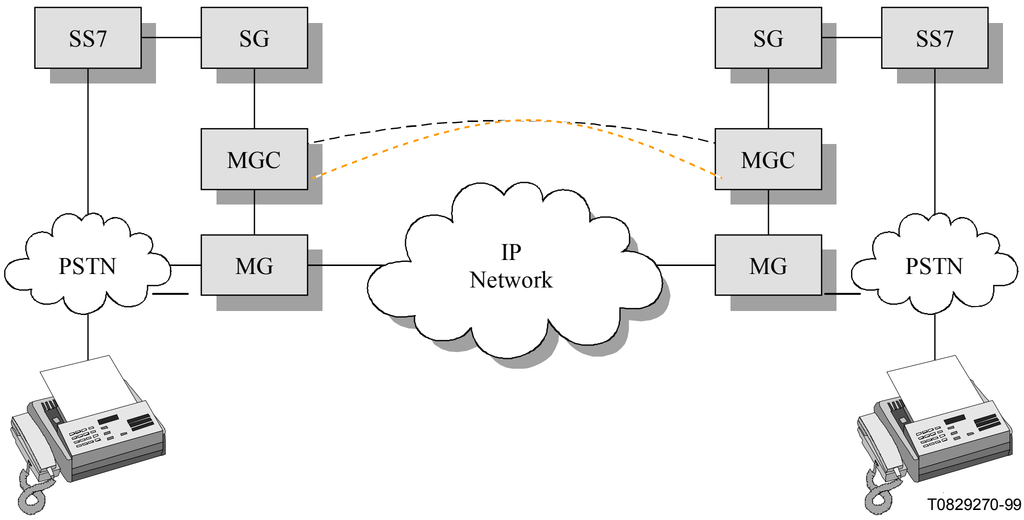


Figure E.1/T.38 – Typical decomposed model

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 H.248. 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 call signalling (TCP). 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:

– the connection model for the H.248 protocol describes the logical entities, or objects, within the Media Gateway that can be controlled by the Media Gateway Controller using the protocol. 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 the 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 H.248 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 H.248) 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 H.248. A voice connection is set up.

Upon detection of CNG by the emitting media gateway (MG), the calling agent is informed (via H.248) 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. 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 H.248) of this event and requests 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. Additional signals for fax are for further study.

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/T.38.

H.248 may use the SDP mechanisms as described in D.2.3/T.38 or the mechanisms described in ITU-T H.245.

In addition, the line packages of H.248 should include a mechanism to identify that a call is using T.38 transport for facsimile and also be able to identify capabilities (especially TCP/UDP).

E.2.5 Examples of call setup

Examples of this procedure are for further study.

E.2.6 Minimum call setup messages

The implementation of this annex shall support the minimum requirements for H.248 as noted in 8.2.

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 supports collection of DTMF digits to make a call.

DTMF transmission during an established voice and facsimile connection is for further study.

E.2.9 Interoperability

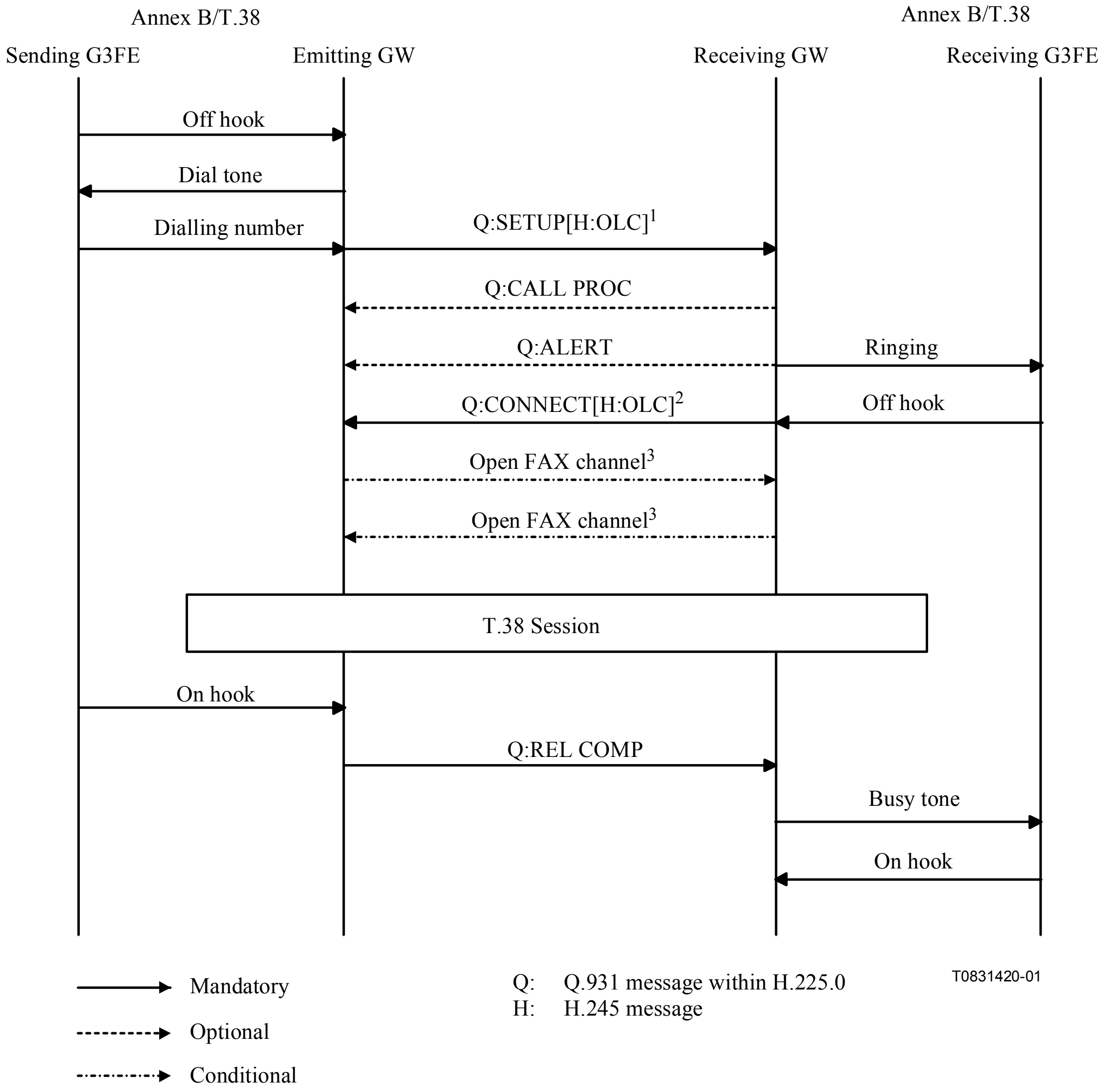
Both ITU-T H.248 and Annex B/T.38 require a well-known port to initiate call signalling. Endpoints of this annex shall use the H.248 well-known port.

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



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

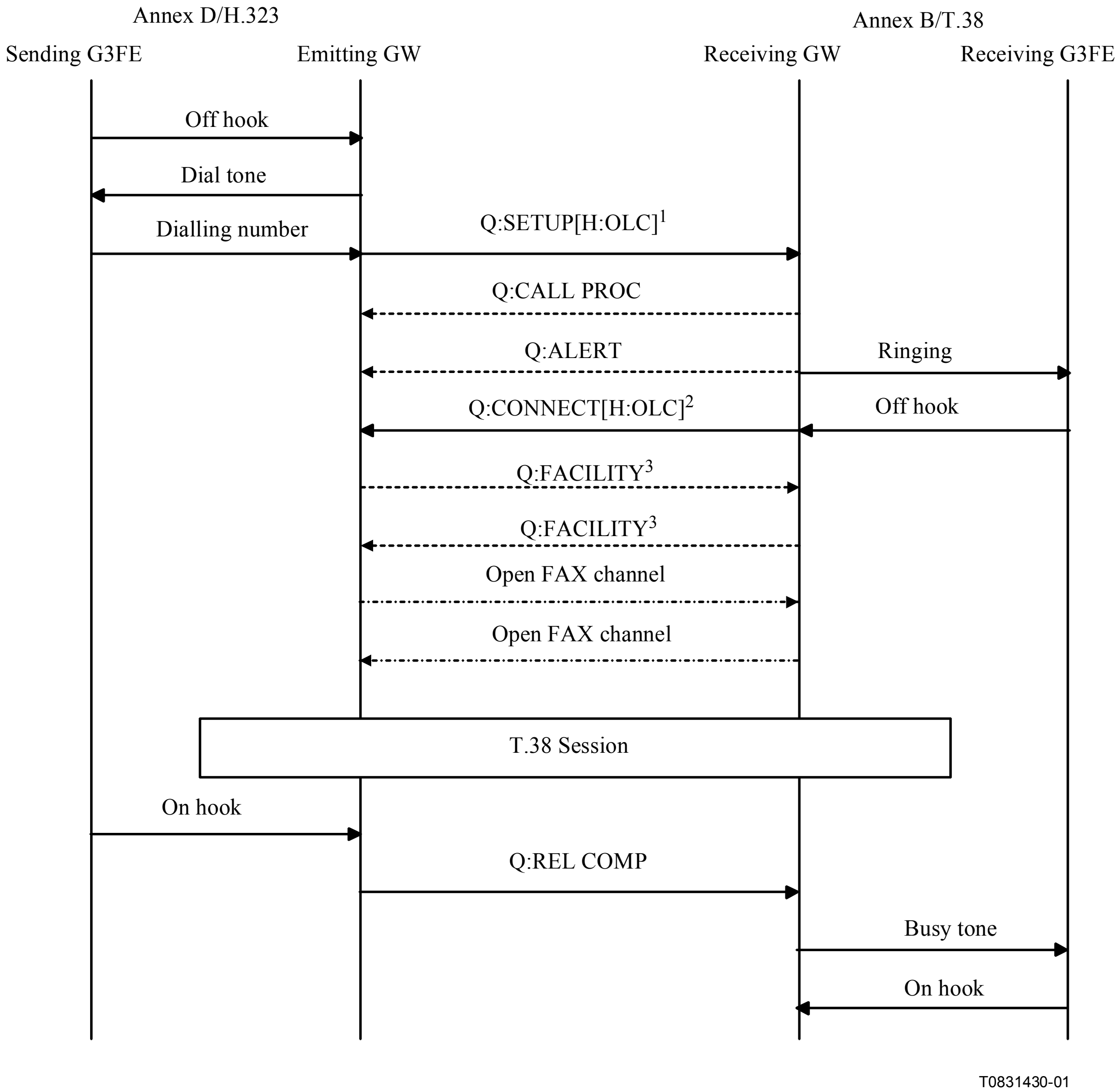
[2] CONNECT contains Connect-UUIE which includes fastStart element which is linked to OpenLogicalChannel (OLC) of ITU-T H.245.

[3] 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)

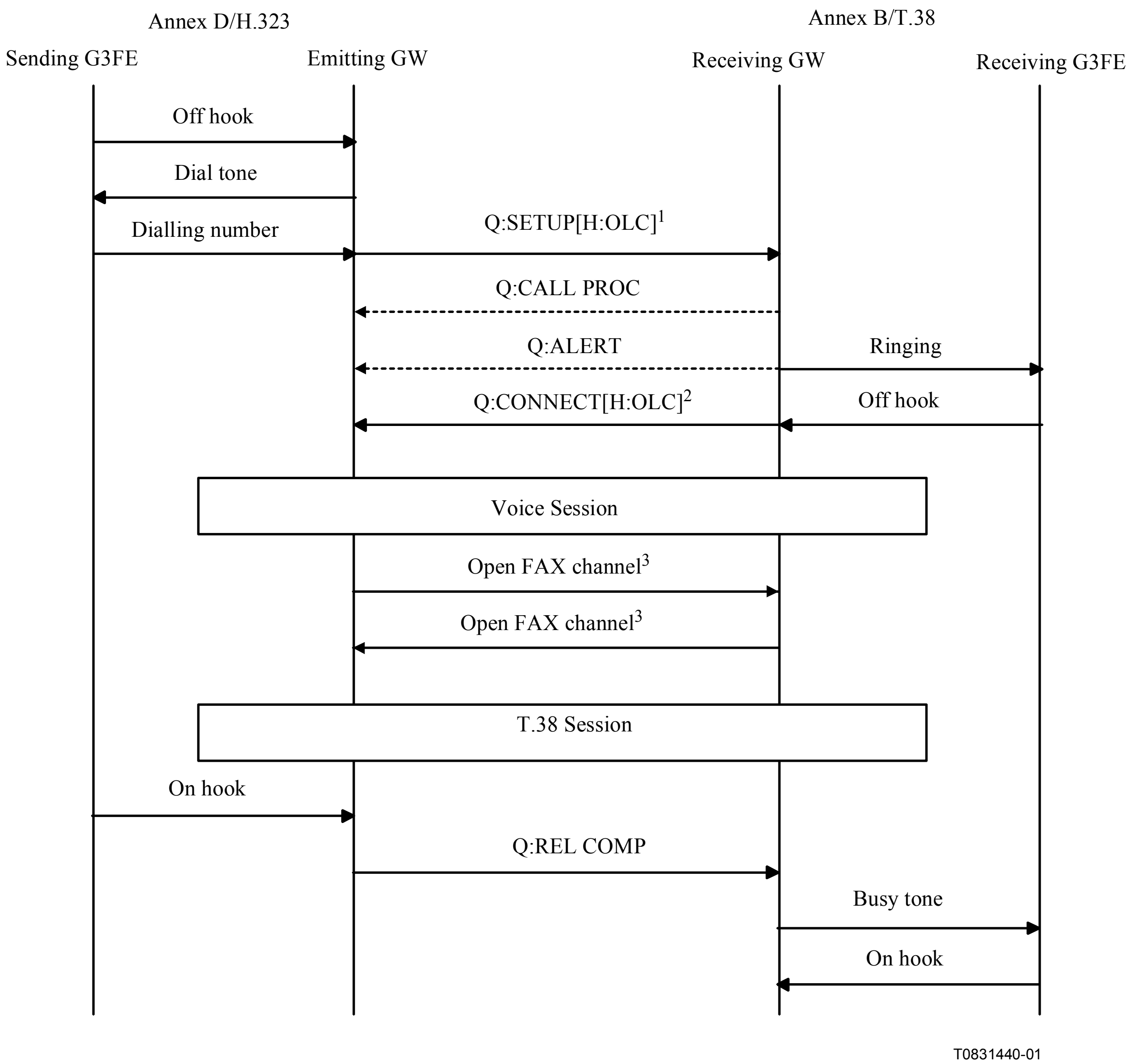


[1] Annex D/H.323 implementation uses fastStart element to send OLCs which include voice and facsimile capabilities.

[2] 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.

[3] 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)



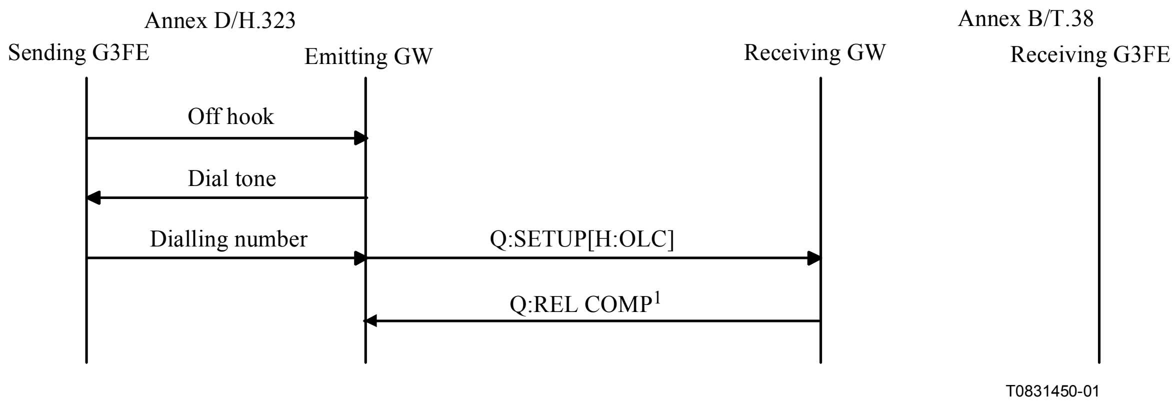
[1] Annex D/H.323 implementation uses fastStart element to send OLC, which includes voice capability as minimum.

[2] Annex B/T.38 implementation returns OLCs which include both voice and facsimile capabilities in responseto 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.

[3] 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 recognizeT.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.

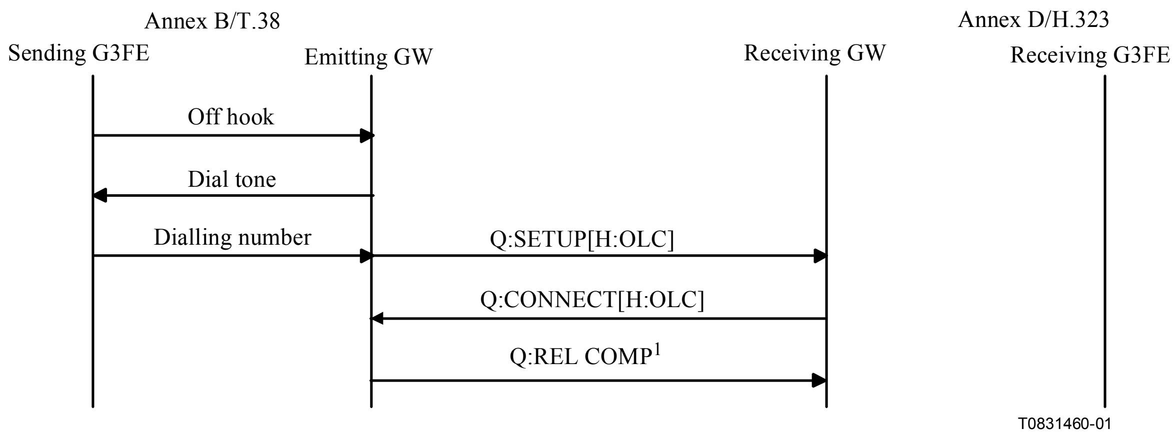
NOTE – Annex B/T.38 supporting FAX and optional voice shall use the methods in Annex D/H.323 as described in B.3.1.1/T.38. Thus, the above figure shows the sequences conforming to Annex D/H.323.

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



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

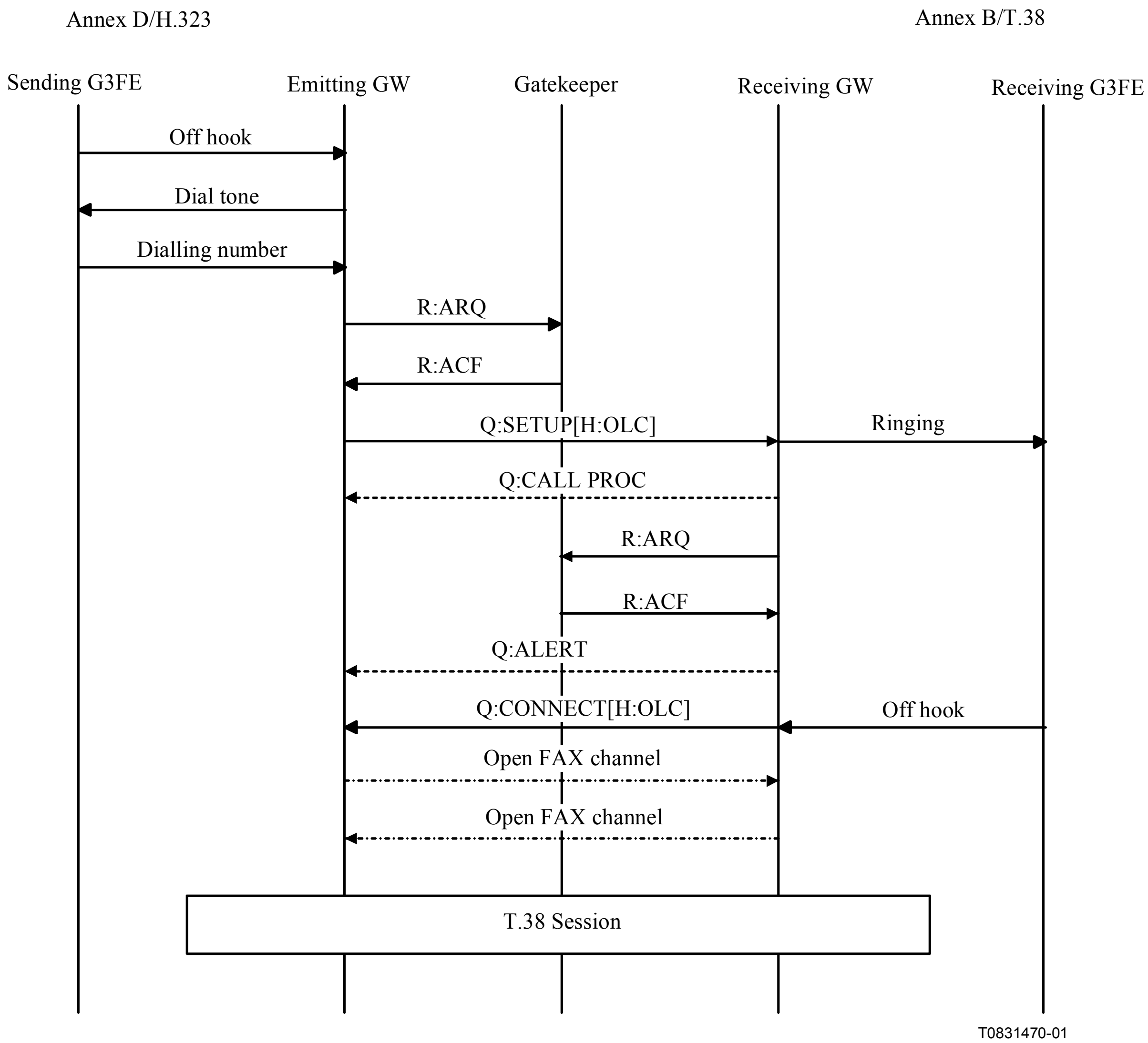
II.1.2.4 Connection rejected sequence 2 (when the called side, Annex D/H.323, does not support Fast Connect Procedures)



1 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 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 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 H.245.

Additionally, RAS messages need to be understood to fully implement Annex B/T.38. RAS is also defined in ITU-T 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	
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

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