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IMS VoIP

FAX over IP

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0. Document History

Every update of this document results in a complete new version with new version number and release date.

Version	Date	Main or important changes since previous version
1.0	December 20, 2010	First published version
1.1	August 19, 2011	Existing fax call scenarios clarified, some scenarios added
1.2	January 19, 2012	Scenario added
1.3	November 30, 2012	Scenarios using 2 m-lines (G.711 , T.38) changed
1.4	Januari 13, 2016	Changed "Belgacom" in "Proximus"
1.5	November 24, 2016	Update due to name change "Bizz IP telephony multi" into "Enterprise Voice Multi"
2.0	May 25, 2019	Document name changed from BGC to PXM

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1. Scope

This document defines the SIP signalling over the VoIP interface between the Proximus IP Multimedia Subsystem (IMS) and SIP enabled device for real-time fax over IP.

Two methods for the support of fax over IP have been identified:

- G.711 fax pass through
- T.38

The specifications listed in this document are not exhaustive but have to be interpreted as "minimal requirements" for fax support in the Proximus IMS network.

This document is part of a set of documents describing the UNI interface of the Proximus IMS Network. Other documents in this set are:

For Business Trunking:

> PXM IMS Corporate VoIP – UNI specification – General
>
> PXM IMS Corporate VoIP – UNI specification – SIP signalling – Business Trunking with IMS services
>
> PXM IMS Corporate VoIP – UNI specification – SIP signalling – Wireless Office extended
>
> PXM IMS Corporate VoIP – UNI specification – SIP signalling – Enterprise Voice Multi
>
> PXM IMS Corporate VoIP – UNI specification – Testing

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2. References

Whenever a date of edition is mentioned, the document with this date should be consulted. If no date is present, the latest version of this document should be consulted.

2.1. Normative references

1	Proximus	PXM IMS Corporate VoIP – UNI specification – SIP signalling – Business Trunking with IMS services
2	Proximus	PXM IMS Corporate VoIP – UNI specification – SIP signalling – Wireless Office Extended
3	Proximus	PXM IMS Corporate VoIP – UNI specification – SIP signalling – Enterprise Voice Multi
4	3GPP TS 24.229	IP Multimedia call control protocol based on SIP and SDP, Stage 3 Release 7
5	3GPP TS 23.167	IP Multimedia Subsystem (IMS) emergency sessions, Release 7
6	SIP forum	The SIP connect 1.1 technical recommendation (draft)
7	ETSI TS 182 025	Business Trunking; architecture and functional description v2.1.1
8	ITU-T E.164	The international telecommunication numbering plan
9	IETF RFC 3261	SIP: Session Initiation Protocol
10	IETF RFC 3262	Reliability of Provisional Responses in the Session Initiation Protocol (SIP)
11	IETF RFC 3264	An Offer/Answer Model with the Session Description Protocol (SDP)
12	IETF RFC 3265	Session Initiation Protocol (SIP)-Specific Event Notification
13	IETF RFC 4566	SDP: Session Description Protocol
14	IETF RFC 2976	The SIP INFO Method
15	IETF RFC 3311	The Session Initiation Protocol (SIP) UPDATE Method
16	IETF RFC 3323	A Privacy Mechanism for the Session Initiation Protocol (SIP)
17	IETF RFC 3325	Private Extensions to SIP for Asserted Identity within Trusted Networks
18	IETF RFC 3326	The Reason Header Field for the Session Initiation Protocol (SIP)
19	IETF RFC 3455	Private header extensions for SIP for 3GPP
20	IETF RFC 3515	The Session Initiation Protocol (SIP) REFER Method
21	IETF RFC 3891	The Session Initiation Protocol (SIP) "Replaces" Header
22	IETF RFC 3892	The Session Initiation Protocol (SIP) Referred-By Mechanism
23	IETF RFC 3960	Early Media and Ringing Tone Generation in SIP

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24	IETF RFC 4028	Session Timers in the Session Initiation Protocol (SIP)
25	IETF RFC 2617	HTTP Authentication: Basic and Digest Access Authentication
26	IETF RFC 1321	The MD5 Message-Digest Algorithm
27	IETF RFC 3550	RTP: A Transport Protocol for Real-Time Applications
28	IETF RFC 3551	RTP Profile for Audio and Video Conferences with Minimal Control
29	IETF RFC 4733	RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals
30	IETF RFC 4734	Definition of Events for Modem, FAX and Text Telephony signals
31	IETF RFC 5806	Diversion indication in SIP
32	IETF RFC 6044	Mapping and Interworking of Diversion Information between Diversion and History-Info Headers in the Session Initiation Protocol (SIP)
33	IETF RFC 4244	An Extension to SIP for Request History Information
34	IETF RFC 3966	The Tel URI for Telephone Numbers
35	IETF draft	Applying Loose Routing to Session Initiation Protocol (SIP) User Agents (UA)" (expired)
36	Proximus	PXM IMS Corporate VoIP – UNI specification – General
37	ITU-T T.38	Procedures for real-time Group 3 facsimile communication over IP networks
38	ITU-T V.15X series	Modem over IP series of recommendations

2.2. Informative references

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3.Symbols, Definitions and Abbreviations

3.1. Symbols

For the purpose of the present document, the following symbols apply:

None.

3.2. Definitions

For the purpose of the present document, the following definitions apply:

None.

3.3. Abbreviations

For the purpose of the present document, the following abbreviations apply:

3GPP	3th Generation Partnership Project
AGW	Access Gateway
ATA	Analogue Terminal Adapter
CED	Called station identification tone
CNG	Calling tone
CPE	Customer Premises Equipment
ETSI	European Telecommunications Standards Institute
FoIP	Fax over IP
HGW	Home Gateway
IAD	Integrated Access Device
IETF	Internet Engineering Task Force
IMS	IP Multimedia Subsystem
ITU	International Telecommunication Union
ITS	IP Telephony Services
MGW	Media Gateway

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RFC   Request For Comment

RTP   Real Time Transport Protocol

SDP   Session Description Protocol

SIP   Session Initiation Protocol

UNI   User Network Interface

VoIP   Voice over IP

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4. General

4.1. Structure of the document

The general structure of this document mainly covers the following aspects:

- General aspects
- G.711 pass-through
- T38
  - Originating scenarios
    - Fax detection at destination
      - Successful changeover: T.38 only proposed
      - Unsuccessful changeover, case 1: G.711 and T.38 proposed (200 OK)
      - Unsuccessful changeover, case 2: T.38 only proposed (re-INVITE)
      - Unsuccessful changeover, case 3: T.38 only proposed (ACK)
    - **Fax detection at origin**
      - Successful changeover: T.38 only proposed
  - Terminating scenarios
    - Fax detection at destination
      - Successful changeover: proposing T.38 only
      - Unsuccessful changeover, case 1: proposing G.711 and T.38 (200 OK)
      - Unsuccessful changeover, case 2: proposing T.38 only (re-INVITE)
      - Unsuccessful changeover, case 3: proposing T.38 only (ACK)
    - Fax detection at origin
      - Successful changeover: proposing T.38 only
  - SDP T38 attribute table

4.2. Types of Endpoints

Possible "Endpoints" are:

- the Homegateway (HGW) for I-Talk
- the Analogue Terminal Adapter (ATA) for ITS
- the Integrated access device (IAD) for ISDN on IMS
- the AGW/AGCF for PSTN replacement
- the MGW/MGCF for PSTN/ISDN gateway
- IP-PBXs for Business Trunking

4.3. Reference point

All characteristics described in this UNI-specification are applicable at the reference point indicated in the figure below.
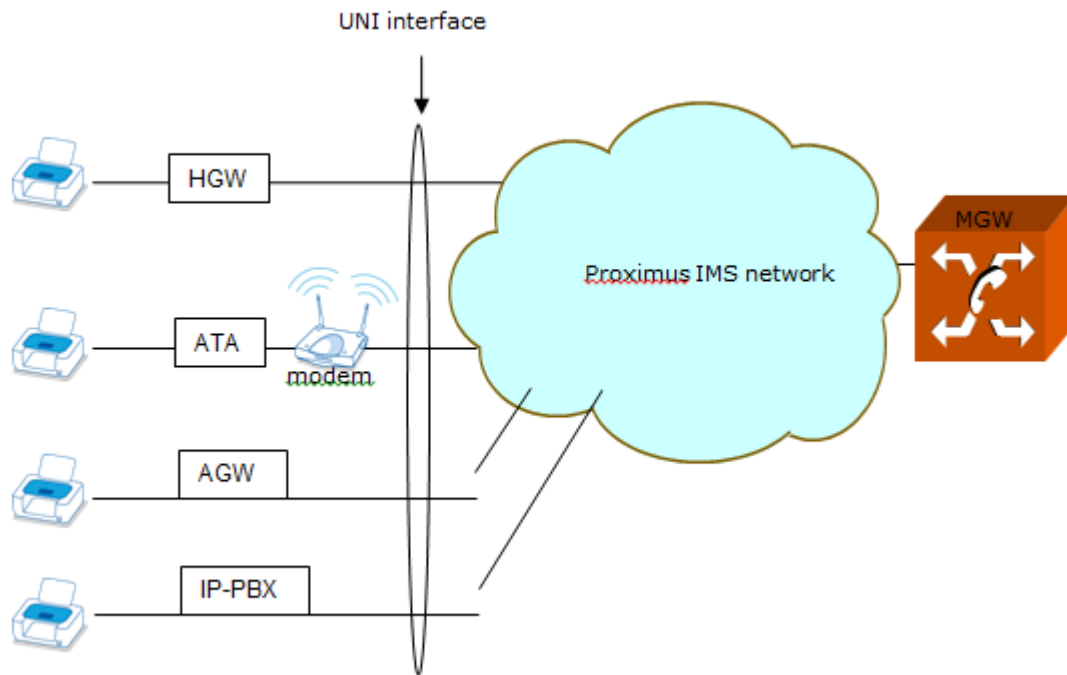
Figure 1 : UNI reference point

5. SIP Behaviour

5.1. Introduction

Besides the protocol support (see [1][2][3] also the expected behaviour is important. E.g. how does switching to FAX mode exactly take place.

5.2. General aspect

Two methods for the support of fax over IP have been identified:

- G.711 fax pass through
- T.38

Basically the support of T.38 fax relay in IMS depends highly on the capabilities of the "end devices". The core IMS network functions (e.g. P-CSCF, S-CSCF etc..) and the Application Servers do not interfere with T.38 media capabilities announced in the SDP of the session set-up.

Since we cannot guarantee that all possible "end devices" will support T.38, it will be impossible and it shall be avoided to have fax calls being set-up with T.38 as only proposed codec. This means that a fax call will always be set-up with a normal codec i.e. G.711 A-law or G.729 as proposed codec.

The "terminating" endpoint shall then upon detection of a fax call use a re-INVITE to switch the RTP stream from G.711 A-law or G.729 to T.38 (if supported)

Endpoints may detect various fax tones (CNG, CED) or flag sequence (like Preamble) and pass these in the audio RTP streams before they are detected. Once detected, Endpoints switch from audio to fax mode and initiate a T.38 fax packet transmission.

Two possible ways exist for detecting a fax communication between 2 Endpoints :

- On the emitting Endpoint (the one sending the facsimile document), the calling tone (CNG) is detected. The CNG tone is however an optional signal.
- On the receiving Endpoint, the V.21 Preamble flag sequence is detected. The Preamble is always present and follows the called station identification tone (CED) when CED is present.

Endpoints must support the detection of fax on Preamble flag sequence.

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Endpoints may support the detection of fax on CNG tone, but this is not recommended because this signal doesn't distinguish between the used standard (e.g. V.17 or V.34) and the simplicity of the tone might lead to false fax detection.

The SIP entities that support T.38, shall also support the G.711 fax pass-through mode.

If one of the SIP entities does not support T.38 real-time fax, a fall back to a fax pass-through mode shall occur to allow the facsimile transmission.

5.3. G.711 pass-through

This paragraph is applicable if T38 codec is not supported by the end device.

Incoming call to the SIP UA

Any incoming call may originate from a fax machine. Fax calls should use G.711 A-law as codec. In case an incoming fax call starts with another codec, the re-INVITE mechanism shall be used to change the media description parameters of the call:

- o Change codec to G.711 A-law

Outgoing call from the SIP UA

An outgoing call originated by a fax machine should use G.711 A-law as codec. In case an outgoing fax call starts with another codec, the re-INVITE mechanism shall be used to change the media description parameters of the call:

- o Change codec to G.711 A-law

5.4. T.38

This paragraph is applicable if T.38 codec is supported by the end device.

5.4.1. General aspects

The current version of this document deals primarily with one transport protocol for the media: T.38 over UDPTL. (packetization time: 20 ms)

5.4.2. T.38 versions

ITU-T recommendation T.38 defines the procedures for real-time Group 3 facsimile communication over IP networks.

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T.38 version numbers are 0, 1, 2, and 3.

- Version 0 : original version
- Version 1 : support for Internet Facsimile Protocol (IFP) over TCP.
- Version 2 : The abstract syntax notation 1 (ASN.1) notation is modified in version 2 with TCP support. The modified ASN.1 notation in version 2 and previous notations in version 0 or 1 cannot interoperate with each other.
- Version 3 : supports V.34 fax terminals

The IMS network only supports **T38 version 0**.

note: T.38 version 3 is not yet available on the network elements and most initial deployments.

Because T.38 version 0 does not support V.34, Super G3 fax transmissions (V.34) are transmitted in G.711 pass-trough mode.The implementation of T.38 version 3 is under study.

5.4.3.   Originating scenarios

5.4.3.1.   Fax detection at destination

5.4.3.1.1.   Successful changeover: T.38 only proposed

In this scenario

- The originating SIP UA supports T38
- The call is set-up with a speech codec (e.g. G.711 A-law or G.729)
- Fax is detected at called-party, initiating the re-INVITE proposing only T.38
- The originating SIP UA accepts T.38, answers with 200 OK (T.38)

Note: When proposing T.38 only, fallback to G.711 MUST be supported as well !

After the basic call set-up and fax detection at the called party, the originating SIP UA receives an *INVITE* to switch from audio to fax mode.

The *INVITE* contains SDP information regarding the proposed call parameters. Only Media type image T.38 is announced in the SDP information.

The originating SIP UA returns a *200 OK* response containing SDP information regarding the accepted call parameters. Media type image T.38 is accepted.

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Note: According to RFC 3264 for each "m=" line in the offer, there MUST be a corresponding "m=" line in the answer. The answer MUST contain exactly the same number of "m=" lines as the offer.

The originating SIP UA receives an *ACK* message in order to properly close the *INVITE* transaction.

Once the fax transmission is terminated, audio capabilities may be restored or normal call release procedures may apply.

Note: figure 2 does not show the restoration of audio capabilities.

Figure 2 : Successful changeover

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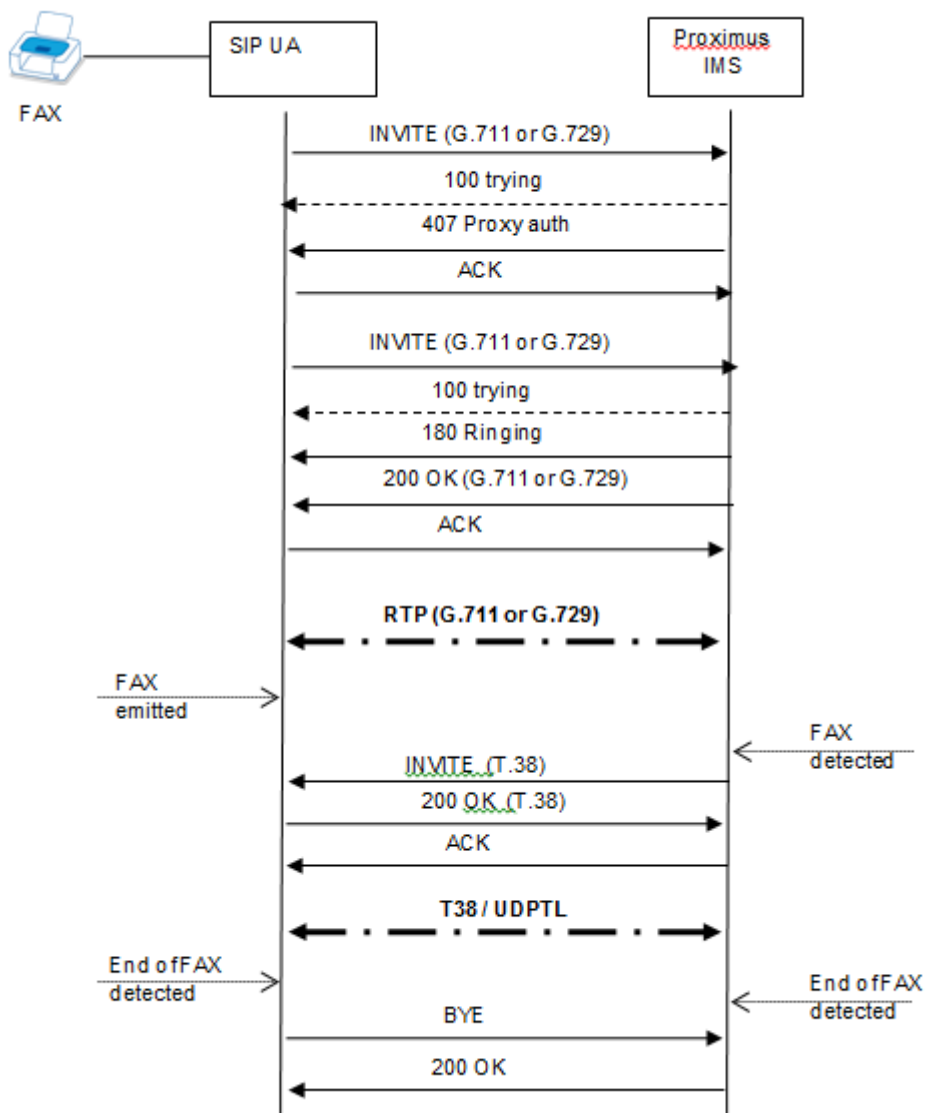
5.4.3.1.2. Unsuccessful changeover, case 1: G.711 and T.38 proposed (200 OK response)

In this scenario

- The originating SIP UA doesn't support T.38
- The call is set-up with a speech codec (e.g. G.711 A-law or G.729)
- Fax is detected at called-party, initiating the re-INVITE proposing G.711 A-law and T.38
- The originating SIP UA rejects the T.38 offer, answers with 200 OK (G.711 A-law),

After the basic call set-up and fax detection at the called party, the originating SIP UA receives an *INVITE* to switch from audio to fax mode.

Figure 3: Unsuccessful changeover, case 1

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The *INVITE* contains SDP information regarding the proposed call parameters. Media type audio G.711 A-law and media type image T.38 are announced both in the SDP information.

The originating SIP UA returns a *200 OK* response containing SDP information regarding the accepted call parameters. Media type audio G.711 A-law is accepted.

> *Note: According to RFC 3264 for each "m=" line in the offer, there MUST be a corresponding "m=" line in the answer. The answer MUST contain exactly the same number of "m=" lines as the offer. To reject an offered stream, the port number in the corresponding stream in the answer MUST be set to zero. Eg.:*

> o *(m):audio <port number> RTP/AVP 8 101*
> o *(m): image 0 udptl t38*

The originating SIP UA receives an *ACK* message in order to properly close the *INVITE* transaction.

Once the fax transmission is terminated, normal call release procedures apply.

5.4.3.1.3. Unsuccessful changeover, case 2: T.38 only proposed (488 or 415 response / re-INVITE)

In this scenario

- The originating SIP UA doesn't support T.38
- The call is set-up with a speech codec (e.g. G.711 A-law or G.729)
- Fax is detected at called-party, initiating the re-INVITE proposing only T.38
- The originating SIP UA answers with 488 (Not Acceptable here) or 415 (Unsupported Media Type)
- The called party sends a re-INVITE (G.711 A-law) in order to fallback to G.711 for fax transmission

After the basic call set-up and fax detection at the called party, the originating SIP UA receives an *INVITE* to switch from audio to fax mode.
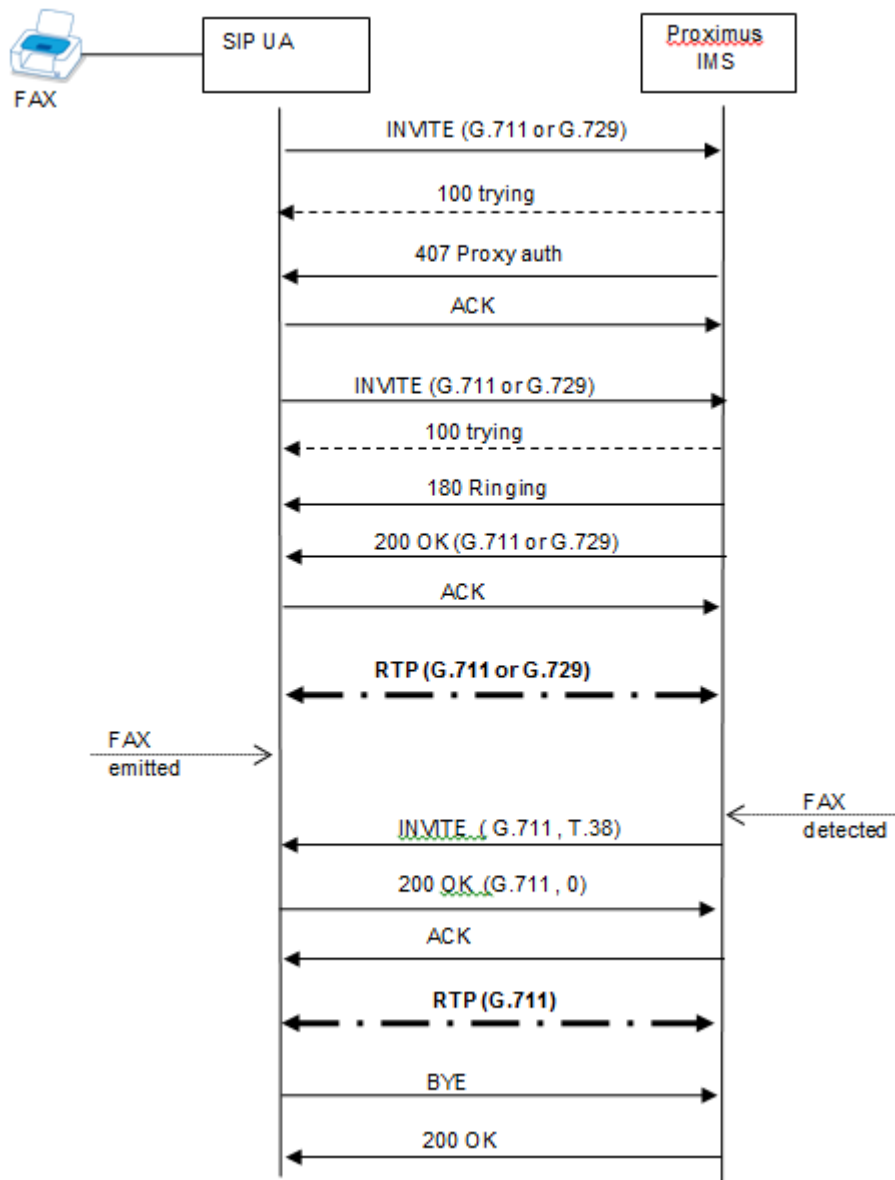
The *INVITE* contains SDP information regarding the proposed call parameters. Only the media type image T.38 is announced in the SDP information.

The originating SIP UA returns a 488 response (Not Acceptable here) or a 415 response (Unsupported Media Type).

- In case of a 488 Not acceptable Here response:
    - A Warning header may be present providing information about the reason why the offer was rejected. E.g. warning code 304 Media type not available may be used.
    - A message body containing a description of media capabilities MAY be present in the response
- In case of a 415 Unsupported Media type response:
    - An Accept header listing the acceptable types may be included.

The originating SIP UA receives an *ACK* message in order to properly close the INVITE transaction. However the failure of the re-INVITE should not cause the existing call to fail. In order to continue the

session using fallback a new re-INVITE with media type audio G.711 A-law and without the T.38 media type will be received by the originating SIP UA.

Once the fax transmission is terminated, normal call release procedures apply.
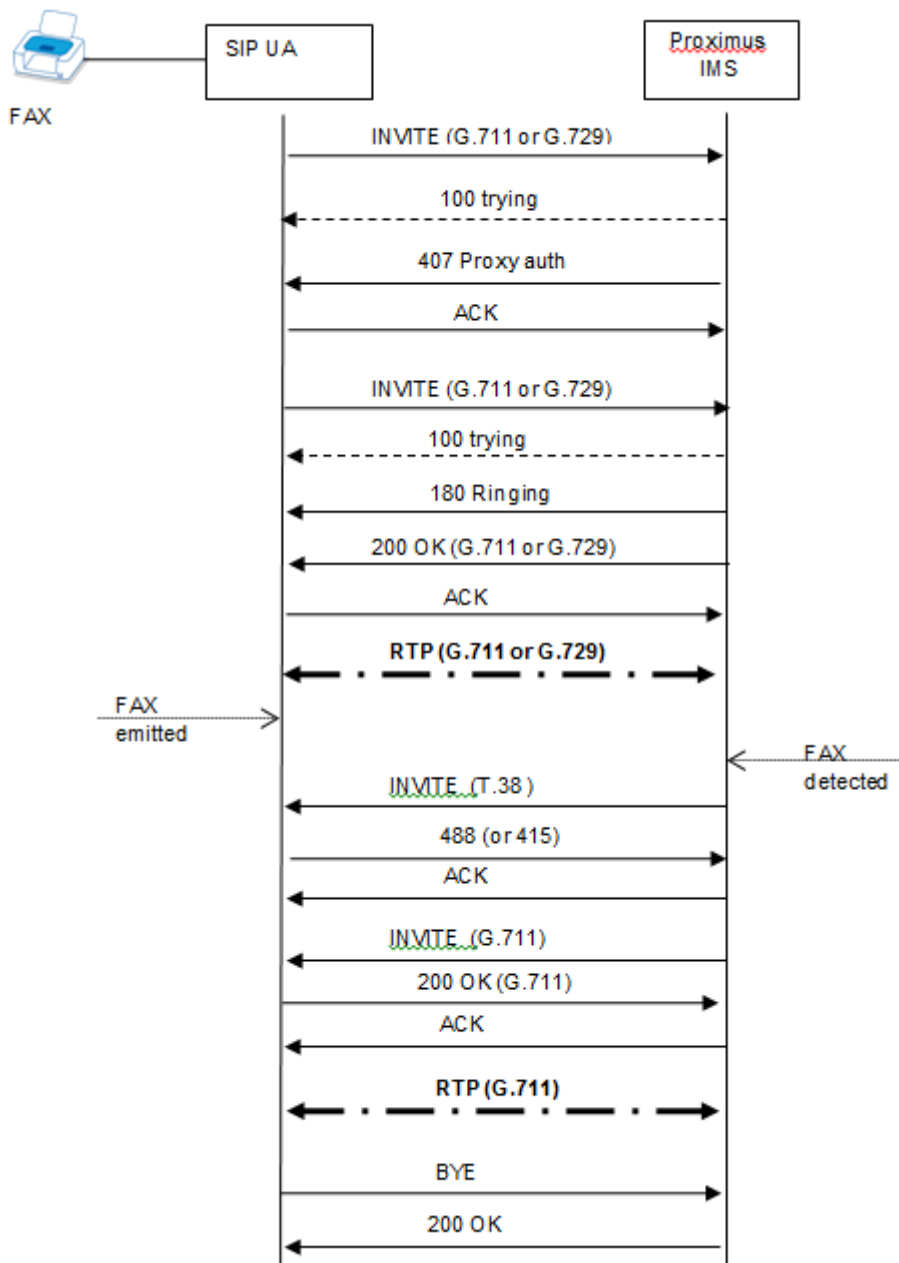
Figure 4: Unsuccessful changeover, case 2

5.4.3.1.4. Unsuccessful changeover, case 3: T.38 only proposed (488 or 415 response / ACK)

In this scenario

- The originating SIP UA doesn't support T.38

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- The call is set-up with a speech codec (e.g. G.711 A-law or G.729)
- Fax is detected at called-party, initiating the re-INVITE proposing only T.38
- The originating SIP UA answers with 488 (Not Acceptable here) or 415 (Unsupported Media Type)
- The called party sends a ACK in order to fallback to previously negotiated codec

After the basic call set-up and fax detection at the called party, the originating SIP UA receives an *INVITE* to switch from audio to fax mode.

The *INVITE* contains SDP information regarding the proposed call parameters. Only the media type image T.38 is announced in the SDP information.

The originating SIP UA returns a 488 response (Not Acceptable here) or a 415 response (Unsupported Media Type).
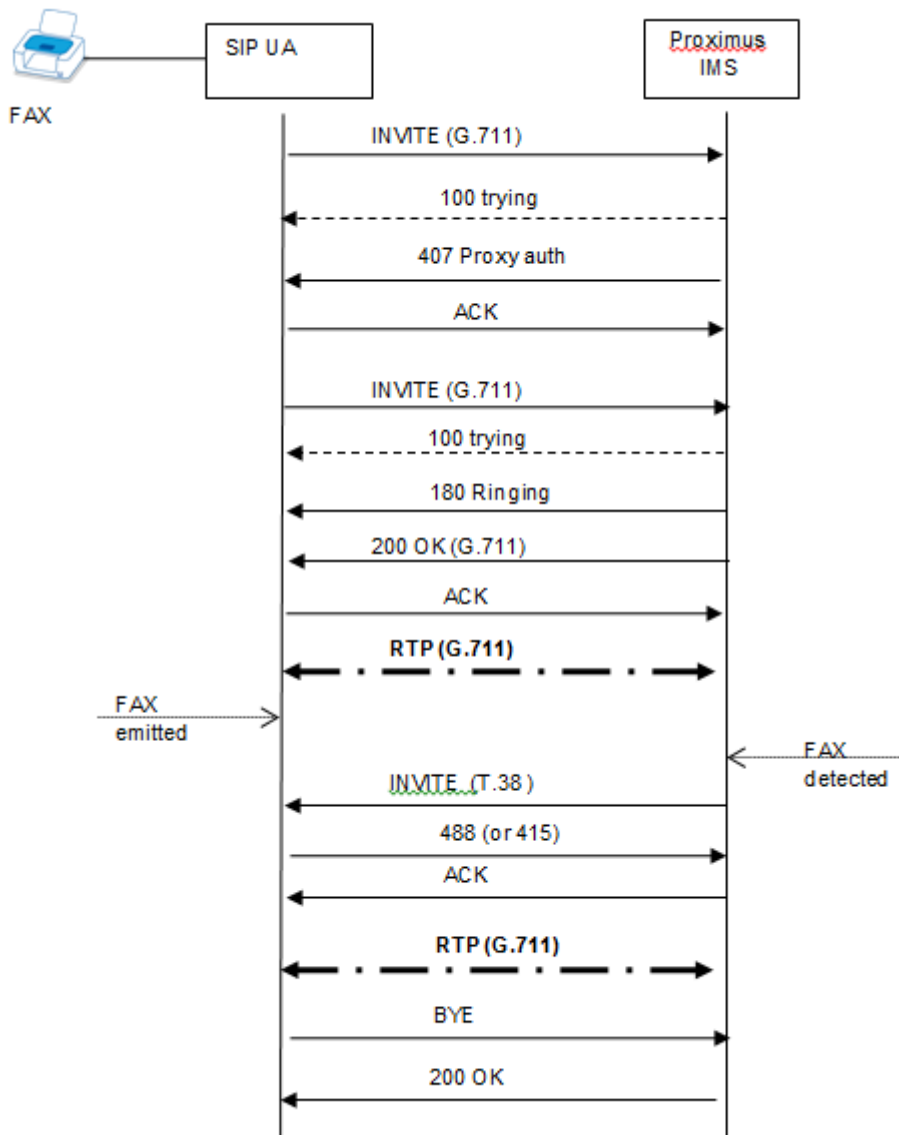

Figure 5: Unsuccessful changeover, case 3

The originating SIP UA receives an *ACK* message in order to properly close the INVITE transaction. However the failure of the re-INVITE should not cause the existing call to fail. The session continues using the previously negotiated characteristics.

> *Note: According to RFC 3261 section 3 :*
>
> *During the session, either Alice or Bob may decide to change the characteristics of the media session. This is accomplished by sending a re-INVITE containing a new media description. This re-INVITE references the existing dialog so that the other party knows*
>
> *that it is to modify an existing session instead of establishing a new session. The other party sends a 200 (OK) to accept the change.*
>
> *The requestor responds to the 200 (OK) with an ACK. If the other party does not accept the change, he sends an error response such as 488 (Not Acceptable Here), which also receives an ACK. However, the failure of the re-INVITE does not cause the existing call to fail - the session continues using the previously negotiated characteristics.*

Note: Fax calls should use G.711 A-law as codec (if T.38 not supported). In case an incoming fax call starts with another codec (e.g. G.729), the re-INVITE mechanism shall be used to change the media description parameters of the call: (see 5.3)

Once the fax transmission is terminated, normal call release procedures apply.

5.4.3.2. Fax detection at origin

5.4.3.2.1. Successful changeover: T.38 only proposed

In this scenario

- The originating SIP UA supports T.38
- The call is set-up with a speech coded (e.g. G.711 A-law or G.729)
- Fax is detected at calling-party side, initiating the re-INVITE proposing only T.38
- The called party accepts the switchover to T.38

Note: When proposing T.38 only, fallback to G.711 MUST be supported as well !

After the basic call set-up and fax detection at the calling party, the originating SIP UA initiates an *INVITE* to switch from audio to fax mode.

The *INVITE* contains SDP information regarding the proposed call parameters. Media type image T.38 and media type audio G.711 A-law are announced both in the SDP information.

The originating SIP UA receives a *200 OK* response containing SDP information regarding the accepted call parameters. The media type image T.38 is accepted.

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Note: According to RFC 3264 for each "m=" line in the offer, there MUST be a corresponding "m=" line in the answer. The answer MUST contain exactly the same number of "m=" lines as the offer.

The originating SIP UA sends an *ACK* message in order to properly close the *INVITE* transaction.

Once the fax transmission is terminated, normal call release procedures apply.

Figure 6: Successful changeover

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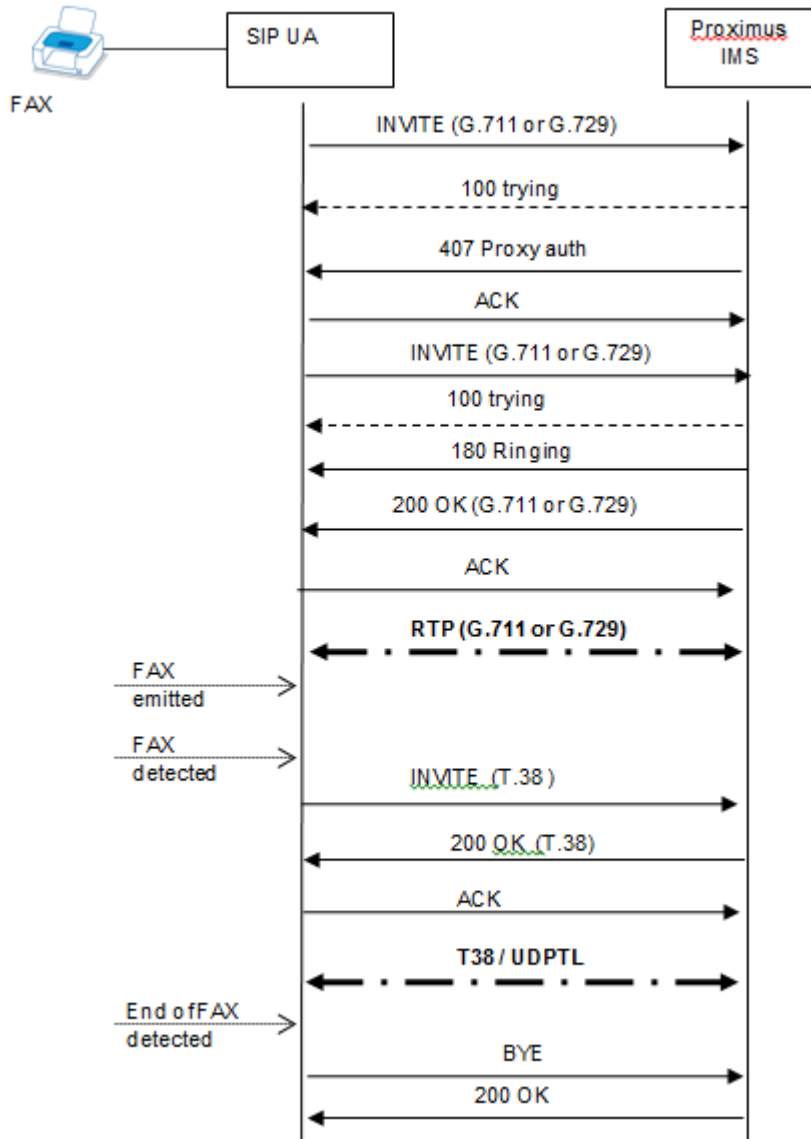#### 5.4.3.2.2. No changeover, dedicated SIP UA for fax

Particular configurations at customer side may include dedicated fax ports. In such case any call originating from that SIP UA instance will be a fax call and hence it could be attempted to set-up the call immediately with T.38 as only codec.

This scenario is not supported and shall not be used !

5.4.4. Terminating scenarios

5.4.4.1. Fax detection at destination

5.4.4.1.1. Successful changeover: proposing T.38 only

In this scenario

- The terminating SIP UA supports T.38
- The call is set-up with G.711 A-law (or G.729) codec
- Fax is detected at called-party, initiating the re-INVITE proposing T.38 only
- T.38 is accepted by the origin

Note: When proposing T.38 only, fallback to G.711 MUST be supported as well !

After the basic call set-up and fax detection at the called party, the terminating SIP UA initiates an *INVITE* to switch from audio to fax mode.

The *INVITE* contains SDP information regarding the proposed call parameters. Only media type image T.38 is announced in the SDP information.

The terminating SIP UA receives a *200 OK* response containing SDP information regarding the accepted call parameters. Media type image T.38 is accepted.

> *Note: According to RFC 3264 for each "m=" line in the offer, there MUST be a corresponding "m=" line in the answer. The answer MUST contain exactly the same number of "m=" lines as the offer.*
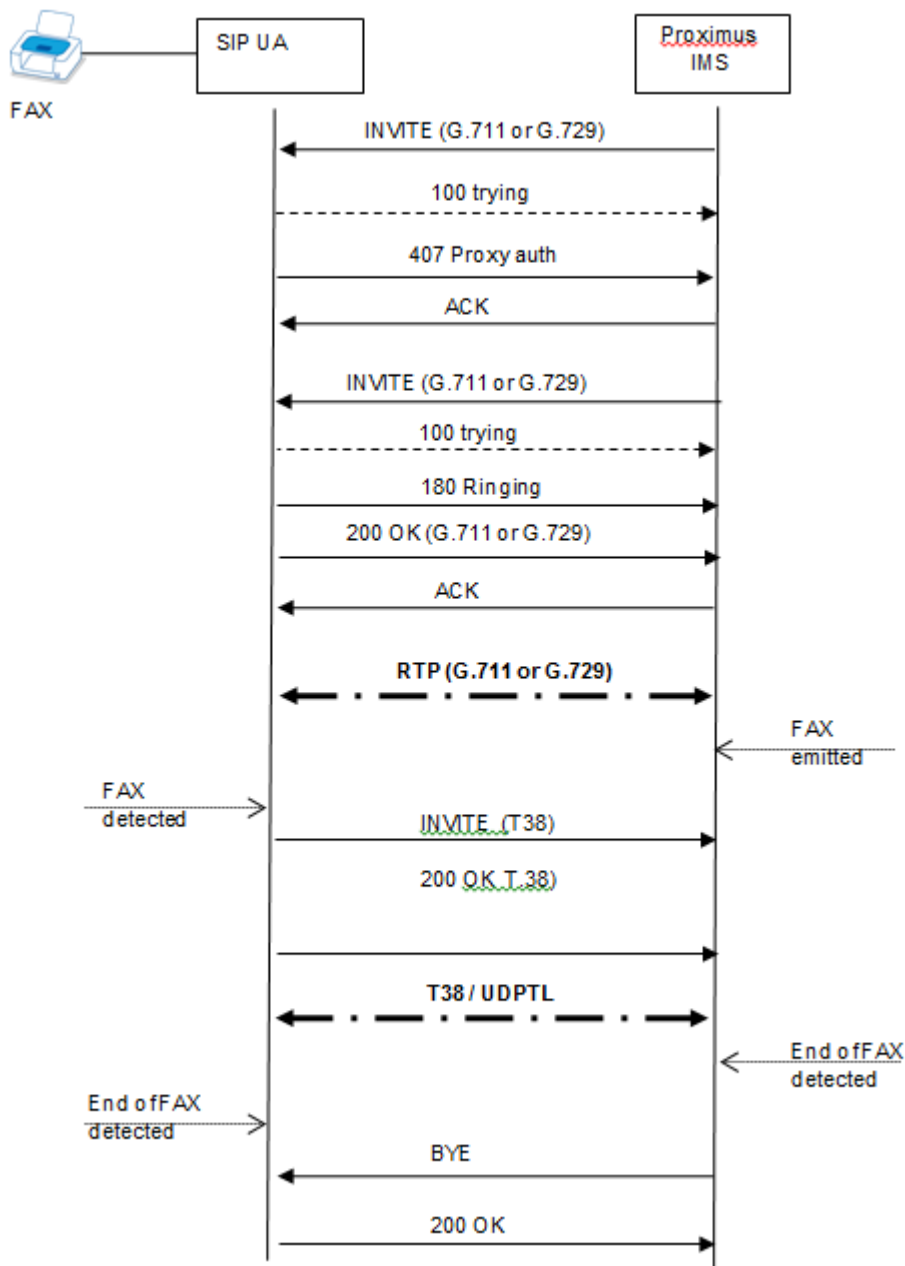
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Figure 7: Successful changeover

The terminating SIP UA sends an *ACK* message in order to properly close the *INVITE* transaction.

Once the fax transmission is terminated, audio capabilities may be restored or normal call release procedures may apply.

Note: figure 9 does not show the restoration of audio capabilities.

5.4.4.1.2. Unsuccessful changeover, case 1: proposing G.711 and T.38 (200 OK response)

In this scenario

- The terminating SIP UA supports T.38
- The call is set-up with G.711 A-law (or G.729) codec
- Fax is detected at called-party, initiating the re-INVITE proposing G.711 A-law and T.38
- The origin doesn't accept T.38
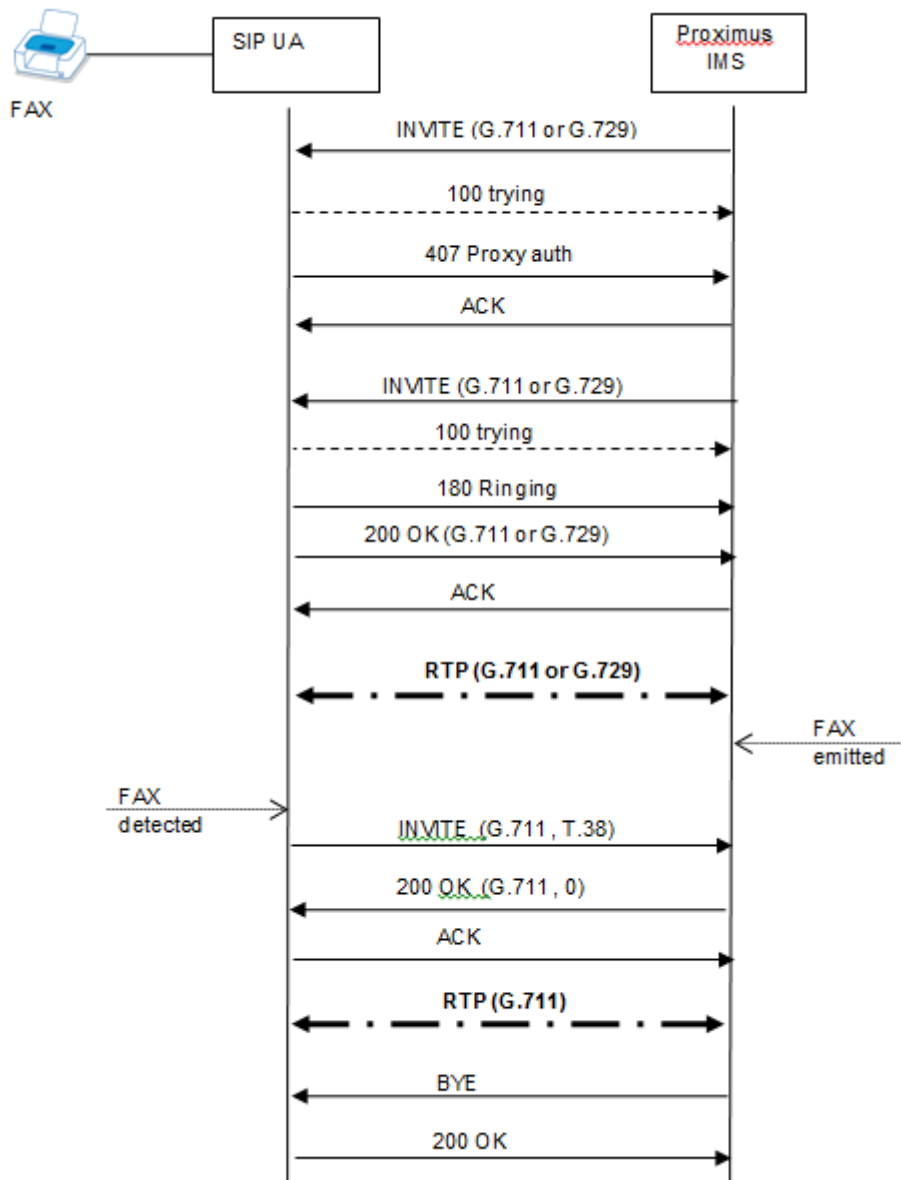
Figure 8: Unsuccessful changeover, case 1

After the basic call set-up and fax detection at the called party, the terminating SIP UA initiates an *INVITE* to switch from audio to fax mode.

The *INVITE* contains SDP information regarding the proposed call parameters.

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Media type audio G.711 A-law and media image T.38 A-law are announced both in the SDP information.

The termination SIP UA receives a *200 OK* response containing SDP information regarding the accepted call parameters. Media type image T.38 is not accepted..

> *Note: According to RFC 3264 for each "m=" line in the offer, there MUST be a corresponding "m=" line in the answer. The answer MUST contain exactly the same number of "m=" lines as the offer. To reject an offered stream, the port number in the corresponding stream in the answer MUST be set to zero. Eg.:*
>
> o *(m):audio <port number> RTP/AVP 8 101*
> o *(m): image 0 udptl t38*

The terminating SIP UA sends an *ACK* message in order to properly close the *INVITE* transaction.

Once the fax transmission is terminated, normal call release procedures apply.

5.4.4.1.3. Unsuccessful changeover, case 2:proposing T.38 only (488 or 415 response / re-INVITE)

In this scenario

- The terminating SIP UA supports T.38
- The call is set-up with G.711 A-law (or G.729) codec
- Fax is detected at called-party, initiating the re-INVITE proposing T.38 only
- The origin doesn't accept T.38 and answers with 488 (Not Acceptable here) or 415 (Unsupported Media Type)
- The terminating SIP UA sent an ACK and sends a re-INVITE in order to fallback to G.711 for fax transmission

After the basic call set-up and fax detection at the called party, the terminating SIP UA initiates an *INVITE* to switch from audio to fax mode.

The *INVITE* contains SDP information regarding the proposed call parameters. Only media type image T.38 is announced in the SDP information.

The terminating SIP UA receives a 488 response (Not Acceptable here) or a 415 response (Unsupported Media Type)

The terminating SIP UA sends an *ACK* message in order to properly close the INVITE transaction. However the failure of the re-INVITE does not cause the existing call to fail, the session continues using fallback.

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Once the fax transmission is terminated, normal call release procedures apply.
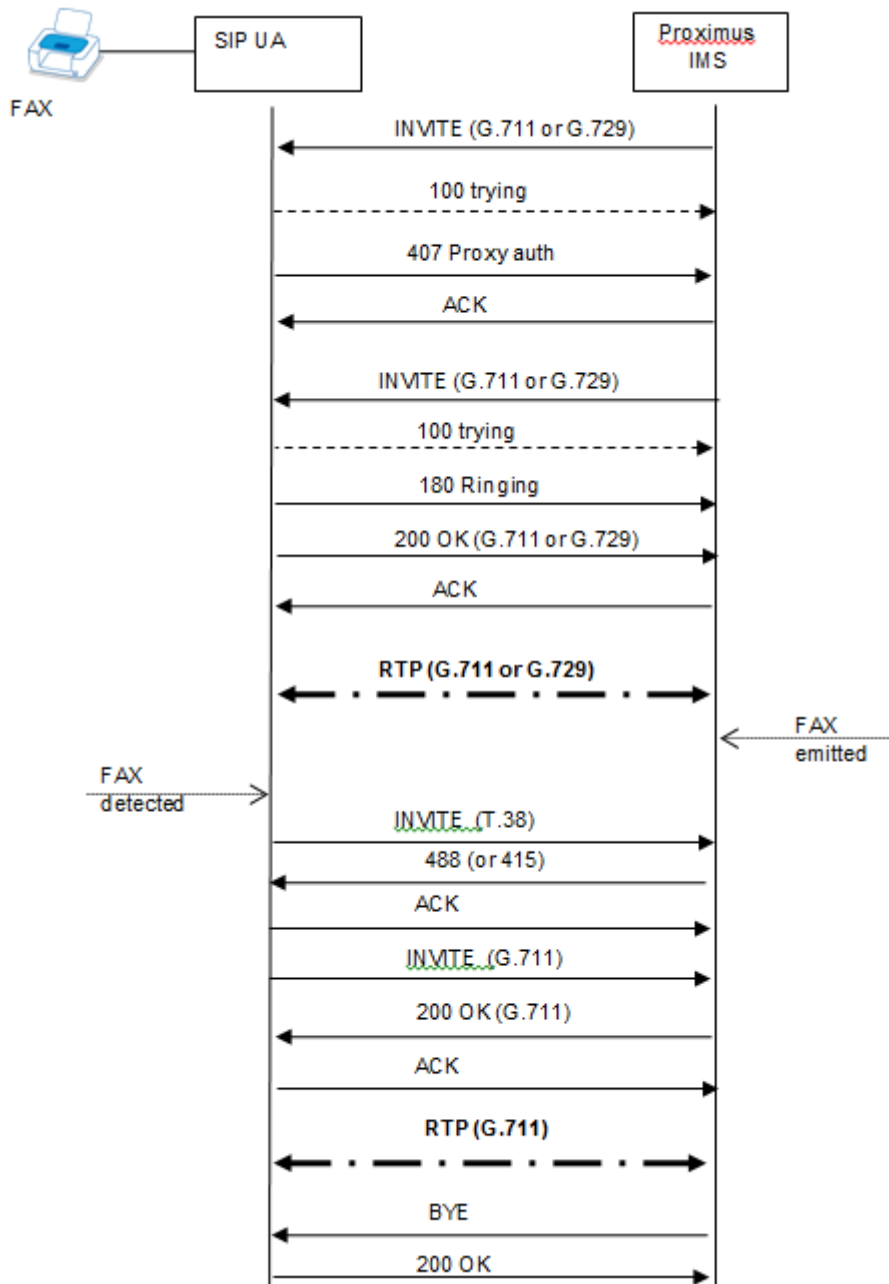

Figure 9: Unsuccessful changeover, case 2

5.4.4.1.4. Unsuccessful changeover, case 3: proposing T.38 only (488 or 415 response / ACK)

In this scenario

- The terminating SIP UA supports T.38
- The call is set-up with G.711 A-law (or G.729) codec
- Fax is detected at called-party, initiating the re-INVITE proposing T.38 only

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- The origin doesn't accept T.38 and answers with 488 (Not Acceptable here) or 415 (Unsupported Media Type)
- The terminating SIP UA sent an ACK order to fallback to previously negotiated codec

After the basic call set-up and fax detection at the called party, the terminating SIP UA initiates an *INVITE* to switch from audio to fax mode.

The *INVITE* contains SDP information regarding the proposed call parameters. Only media type image T.38 is announced in the SDP information.

The terminating SIP UA receives a 488 response (Not Acceptable here) or a 415 response (Unsupported Media Type)
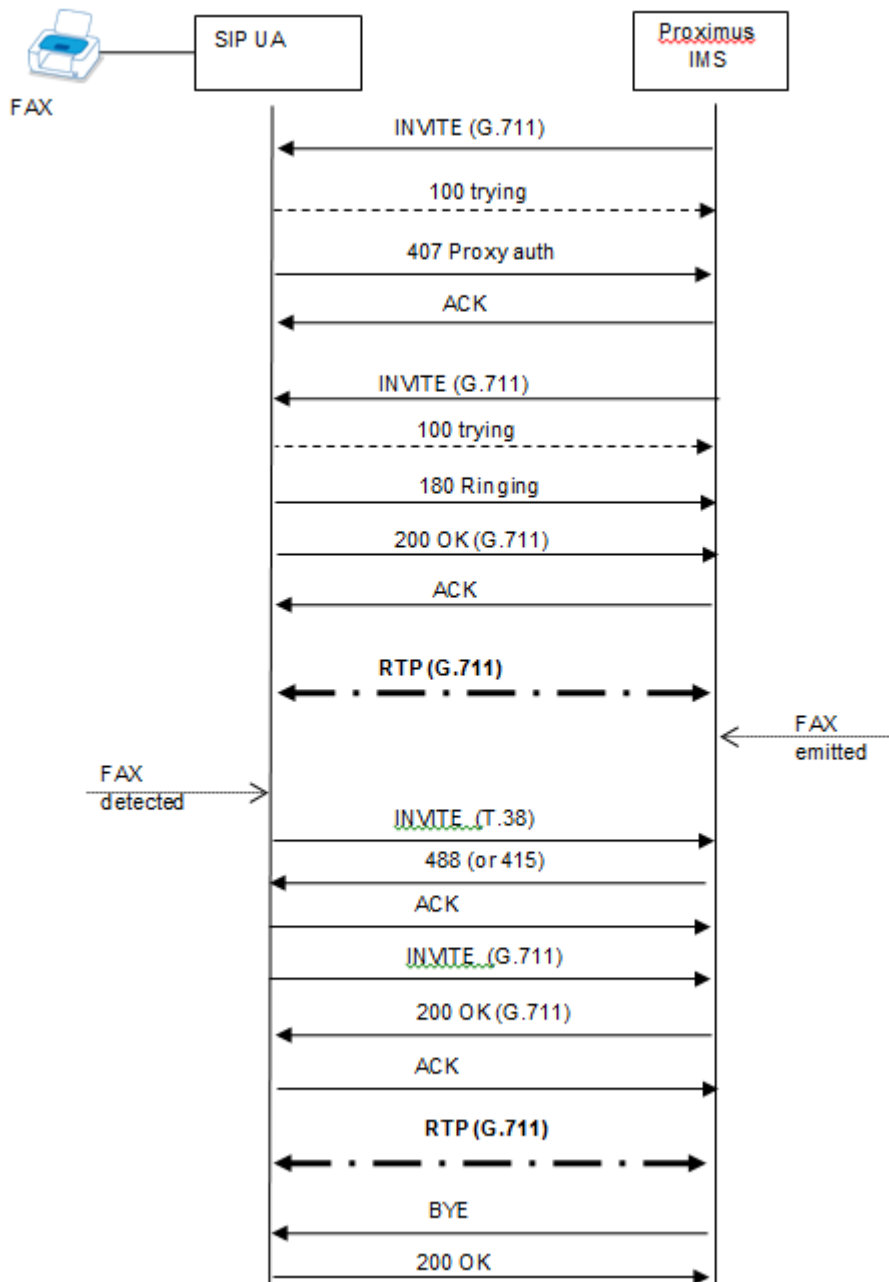
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Figure 10: Unsuccessful changeover, case 3

The terminating SIP UA receives an *ACK* message in order to properly close the INVITE transaction. However the failure of the re-INVITE should not cause the existing call to fail. The session continues using the previously negotiated characteristics.

> Note: According to RFC 3261 section 3 :
>
> During the session, either Alice or Bob may decide to change the characteristics of the media session. This is accomplished by sending a re-INVITE containing a new media description. This re-INVITE references the existing dialog so that the other party knows

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that it is to modify an existing session instead of establishing a new session. The other party sends a 200 (OK) to accept the change.

The requestor responds to the 200 (OK) with an ACK. If the other party does not accept the change, he sends an error response such as 488 (Not Acceptable Here), which also receives an ACK. However, the failure of the re-INVITE does not cause the existing call to fail - the session continues using the previously negotiated characteristics.

Note: Fax calls should use G.711 A-law as codec (if T.38 not supported). In case an incoming fax call starts with another codec (e.g. G.729), the re-INVITE mechanism shall be used to change the media description parameters of the call: (see 5.3)

Once the fax transmission is terminated, normal call release procedures apply.

5.4.4.2.   Fax detection at origin

5.4.4.2.1.   Successful changeover: T.38 only proposed

In this scenario

- The terminating SIP UA supports T.38
- The call is set-up with a speech coded (e.g. G.711 A-law or G.729)
- Fax is detected at calling-party side, initiating the re-INVITE proposing only T.38
- The called party accepts the switchover to T.38

Note: When proposing T.38 only, fallback to G.711 MUST be supported as well !

After the basic call set-up and fax detection at the originating party, the terminating SIP UA receives an *INVITE* to switch from audio to fax mode.

The *INVITE* contains SDP information regarding the proposed call parameters. Only media type image T.38 is announced in the SDP information.

The terminating SIP UA sends a *200 OK* response containing SDP information regarding the accepted call parameters. The media type image T.38 is accepted.

> *Note: According to RFC 3264 for each "m=" line in the offer, there MUST be a corresponding "m=" line in the answer. The answer MUST contain exactly the same number of "m=" lines as the offer.*

The terminating SIP UA receives an *ACK* message in order to properly close the *INVITE* transaction.

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Once the fax transmission is terminated, normal call release procedures apply.
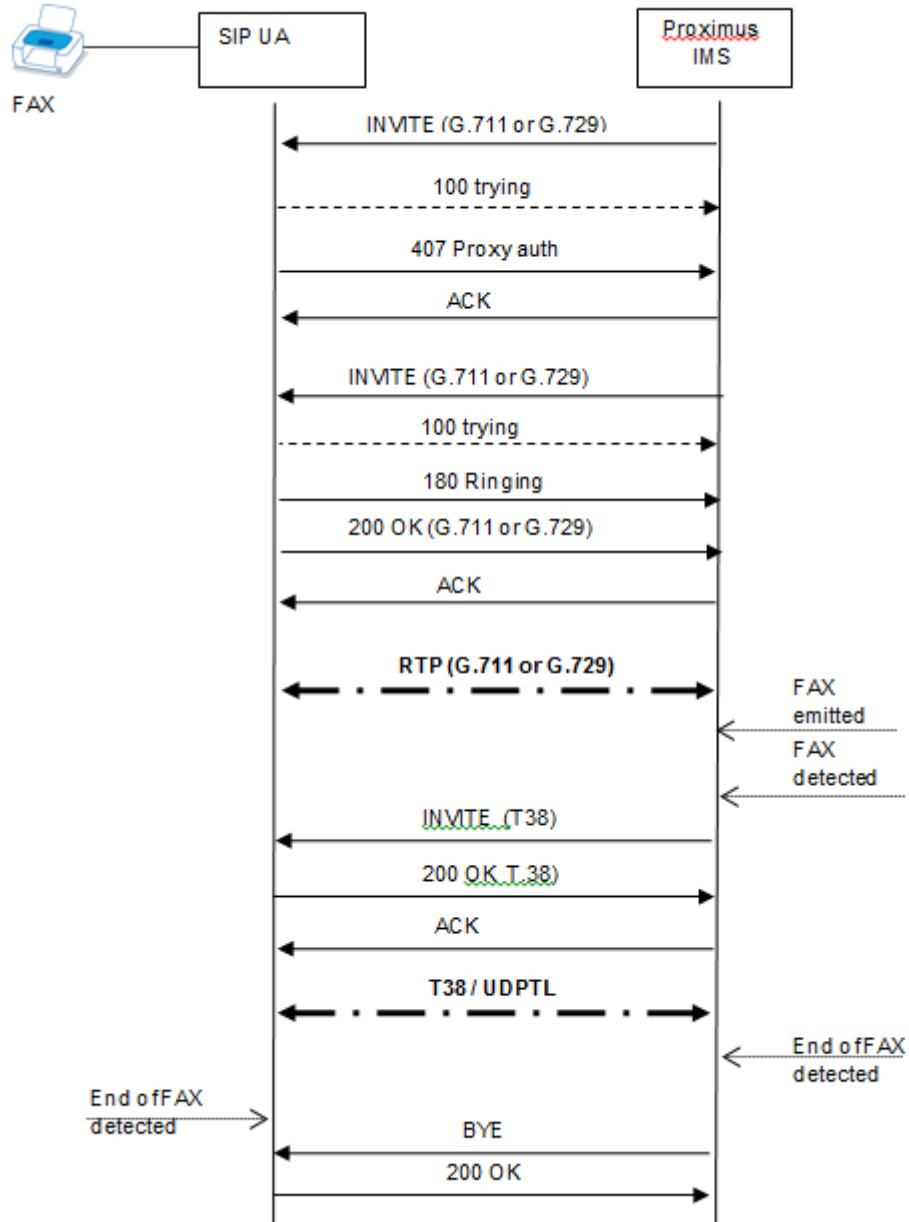

Figure 11: Successful changeover

5.4.5.   SDP T38 attribute table

5.4.5.1.   MGW

This is an overview of the implementation of the SDP T38 attribute parameters in MGW.

SDP attribute	MGW value
T38FaxRateManagement	Transferred TCF
T38FaxVersion	0
T38MaxBitRate	14400
T38FaxFillBitRemoval	Disable
T38FaxTranscodingMMR	Disable
T38FaxTranscodingJBIG	Disable
T38FaxMaxBuffer	1800
T38FaxMaxDatagram	260
T38FaxUdpEC	t38UDPRedundancy
T38VendorInfo	

5.5. V.152

The V.152 [38] procedures describe the support of voice band data over IP networks. As this standard is currently not widely implemented in endpoints it's use is for further study.