Fax over IP:
Testing results & conclusions

i3 forum Technical Stream
Paris, 25–26 September, 2012

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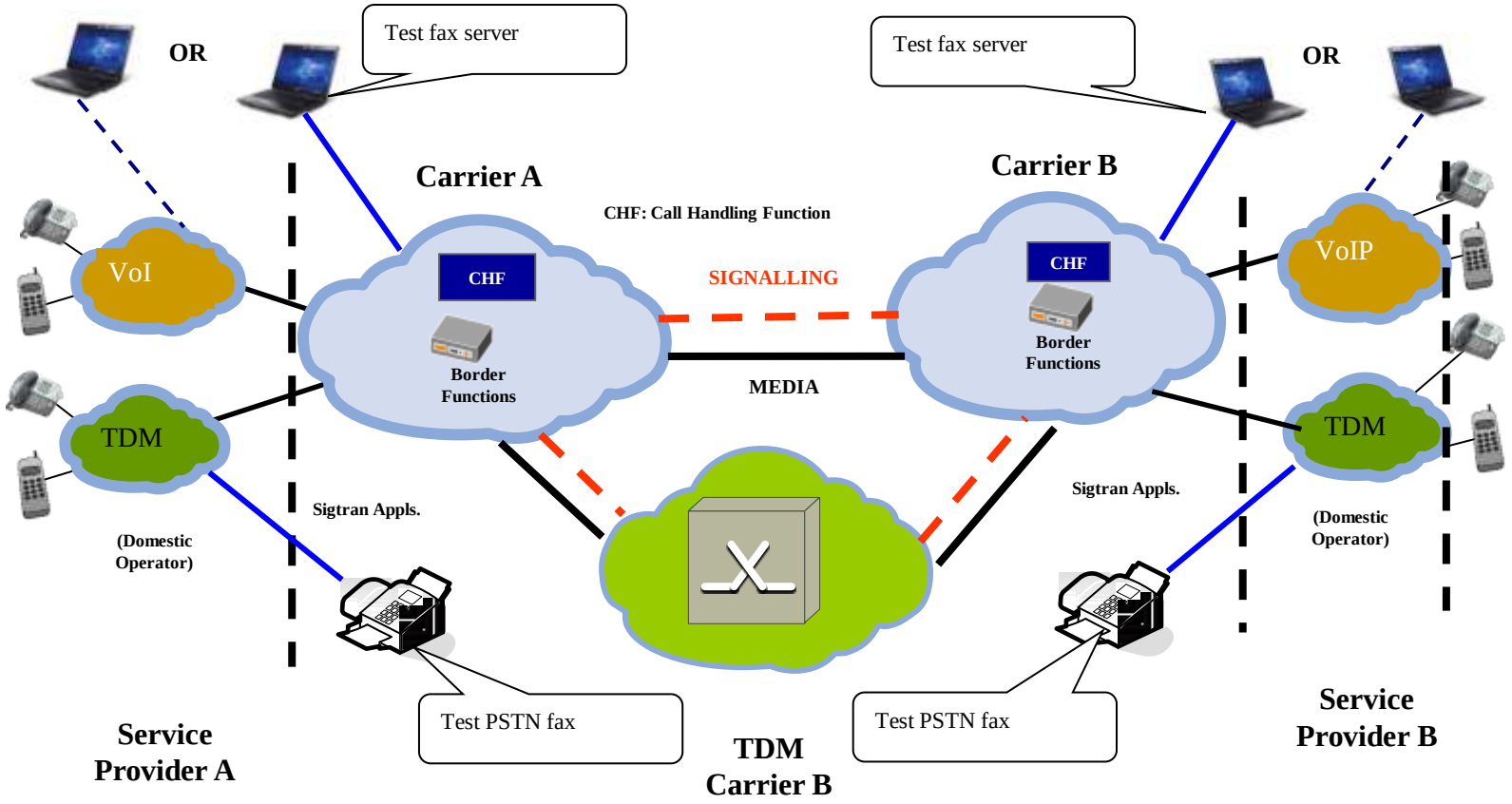
Testing Configuration



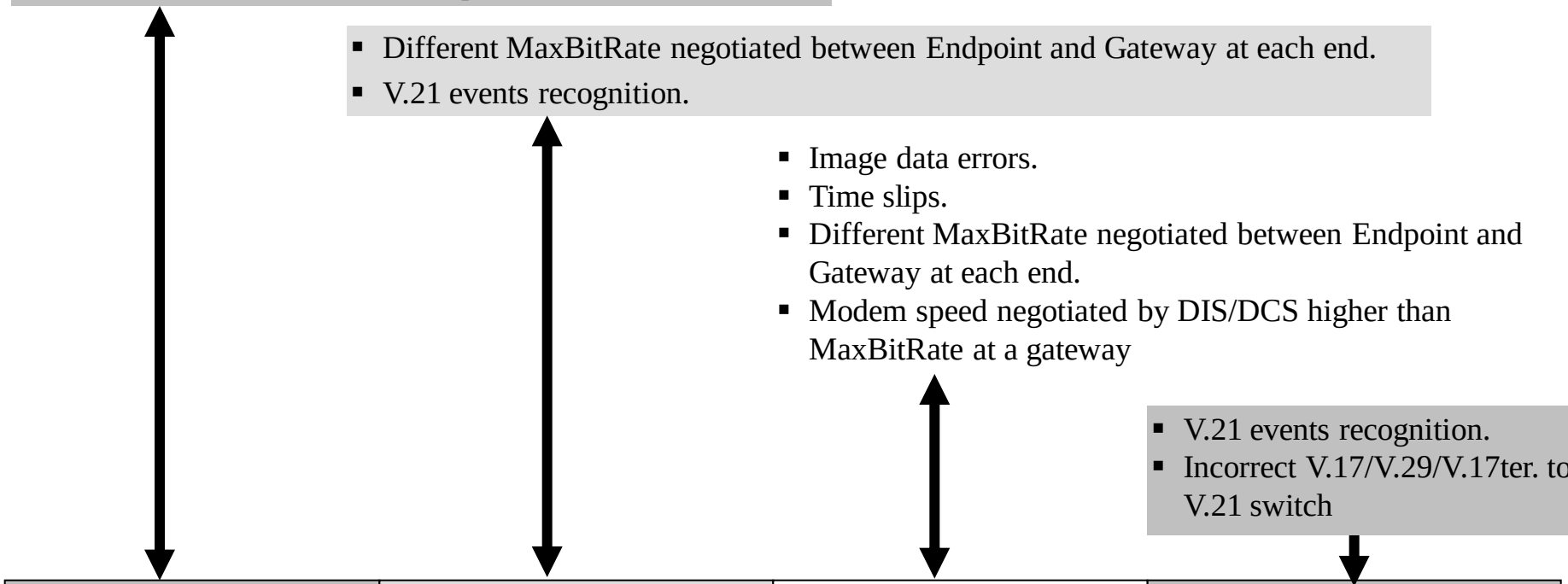
Testing results

- Long SIP/C7 signalling processing – initial 200 OK delay.
- Time shift between re-INVITEs
- T.30 timing problems.
- Media setup problem.
- V.21 events recognition.
- Breaking end-to end SIP signalling during re-INVITE.
- Different FoIP mode at each endpoint.

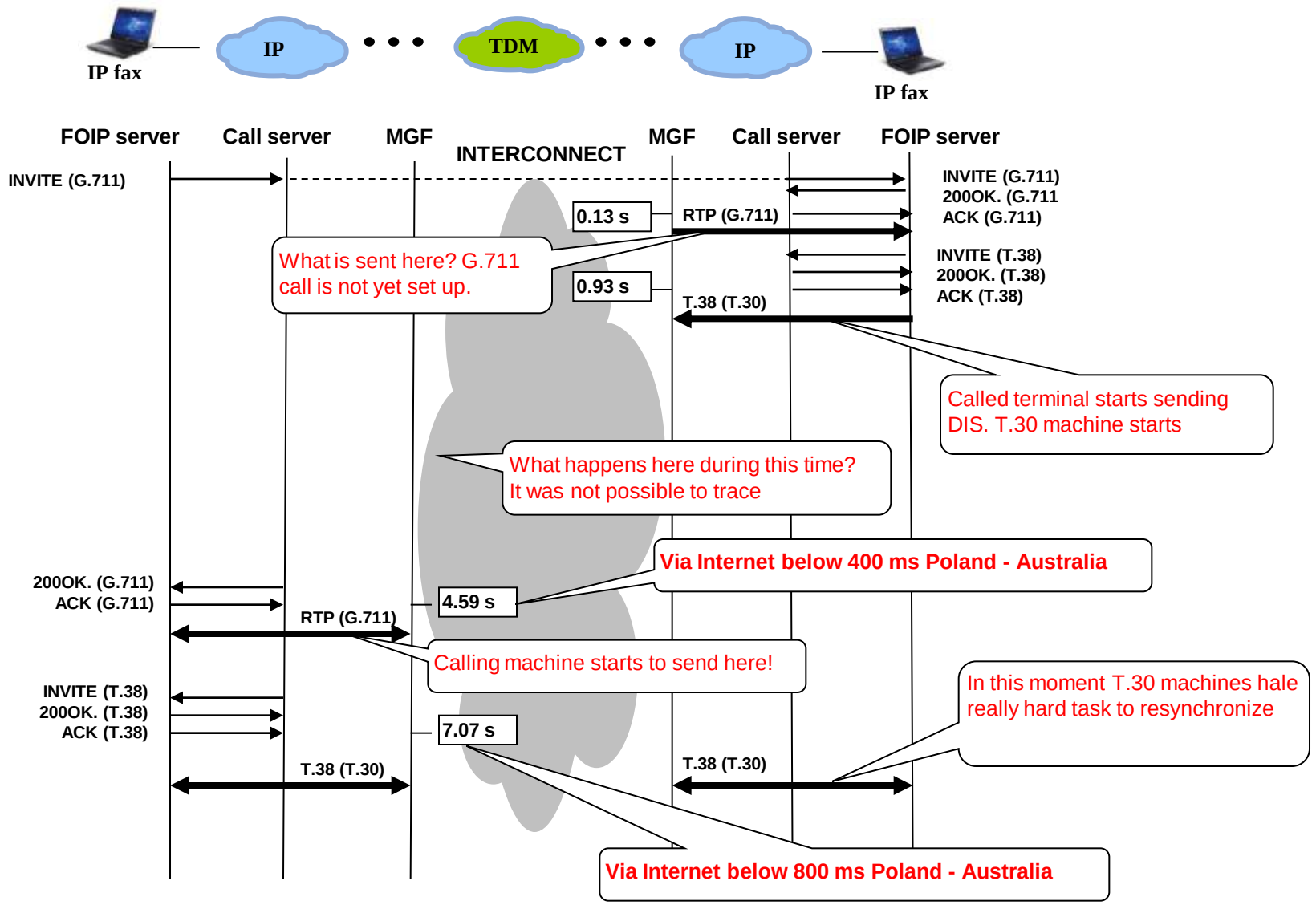
- Different MaxBitRate negotiated between Endpoint and Gateway at each end.
- V.21 events recognition.

- Image data errors.
- Time slips.
- Different MaxBitRate negotiated between Endpoint and Gateway at each end.
- Modem speed negotiated by DIS/DCS higher than MaxBitRate at a gateway

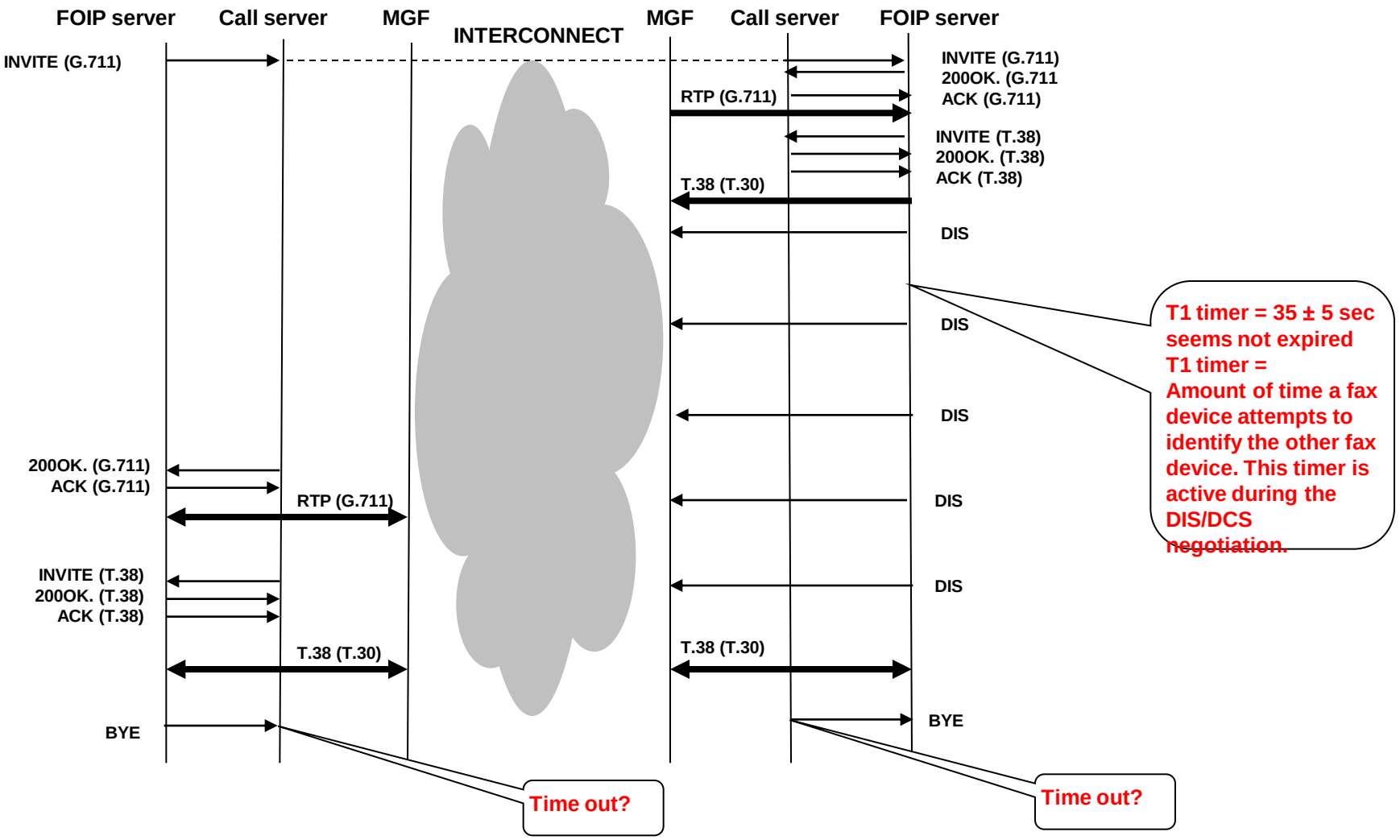
- V.21 events recognition.
- Incorrect V.17/V.29/V.17ter. to V.21 switch



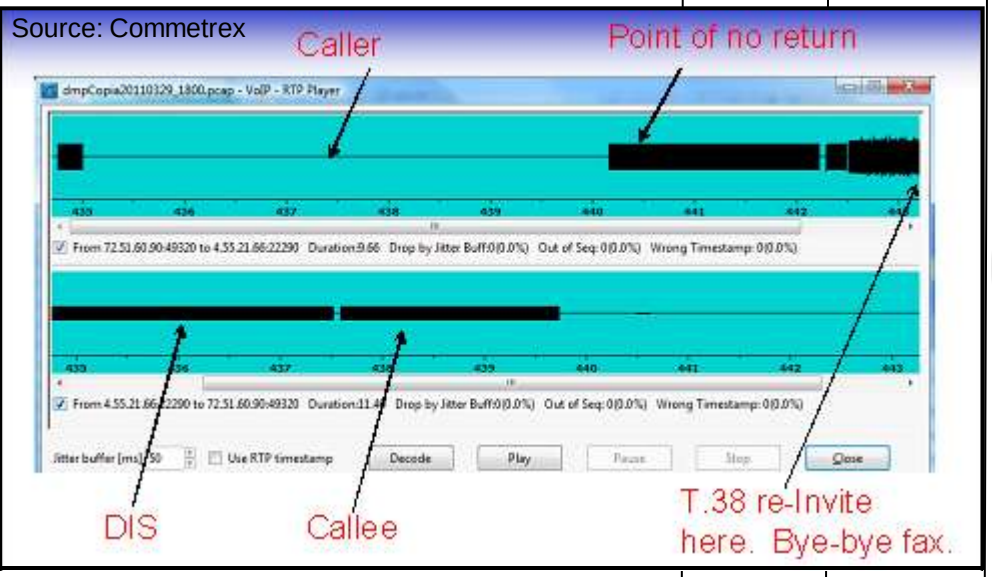
Testing results – signalling delay



Testing results – Call Server time out?



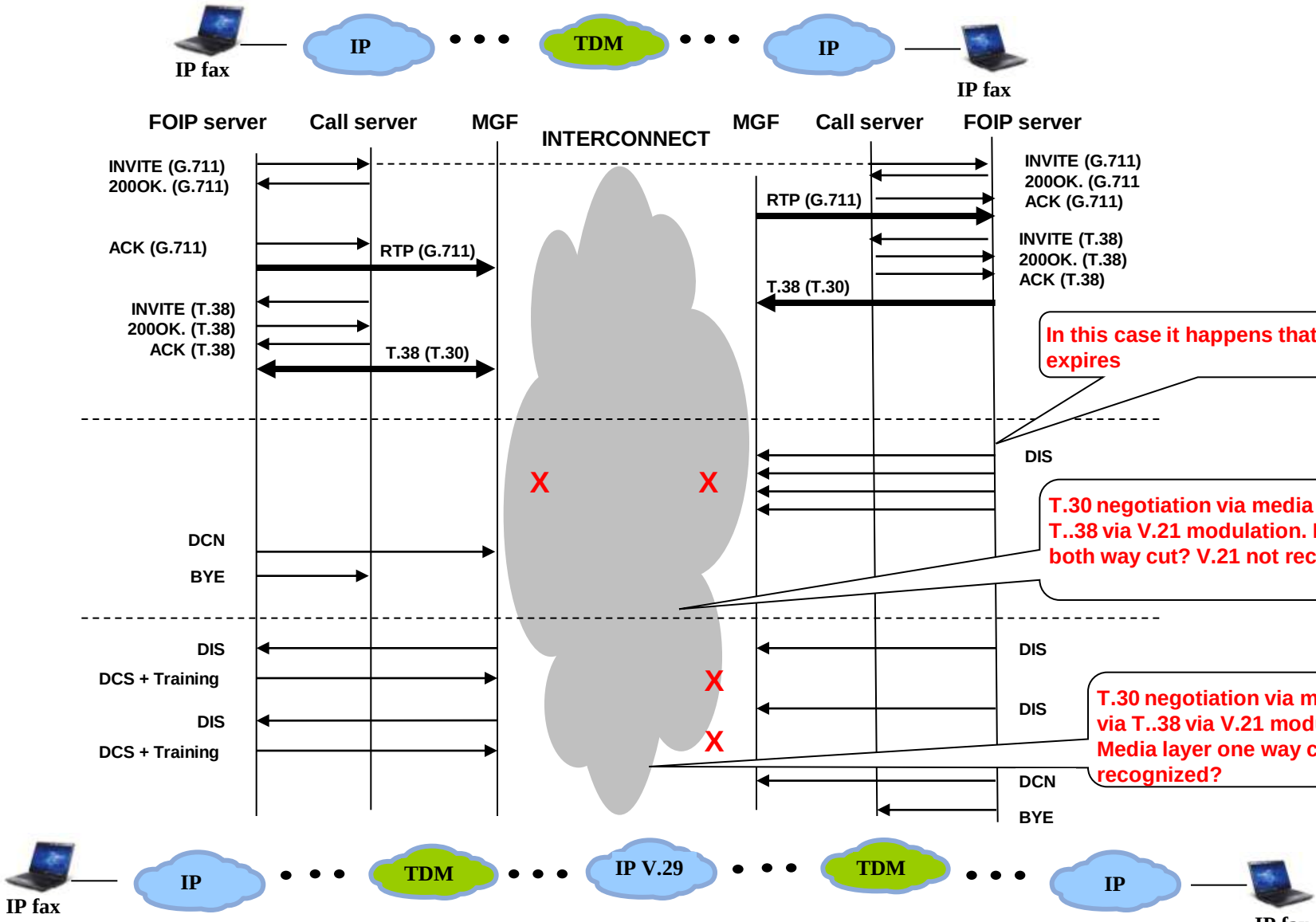
Testing results – late Re-INVITE



FOIP server
Call server
MGF
INTERCONNECT
MGF
Call server
FOIP server

INVITE (G.711)
INVITE (G.711)
200OK. (G.711)
ACK (G.711)
RTP (G.711)
200OK. (G.711)
ACK (G.711)
RTP (G.711)
INVITE (T.38)
200OK. (T.38)
ACK (T.38)
DIS
DCS + Training
No return point
INVITE (T.38)
200OK. (T.38)
ACK (T.38)
T.38 (T.30)
Re-INVITE kills the session
Bye-bye fax
BYE

Testing results – Media setup, V.21?



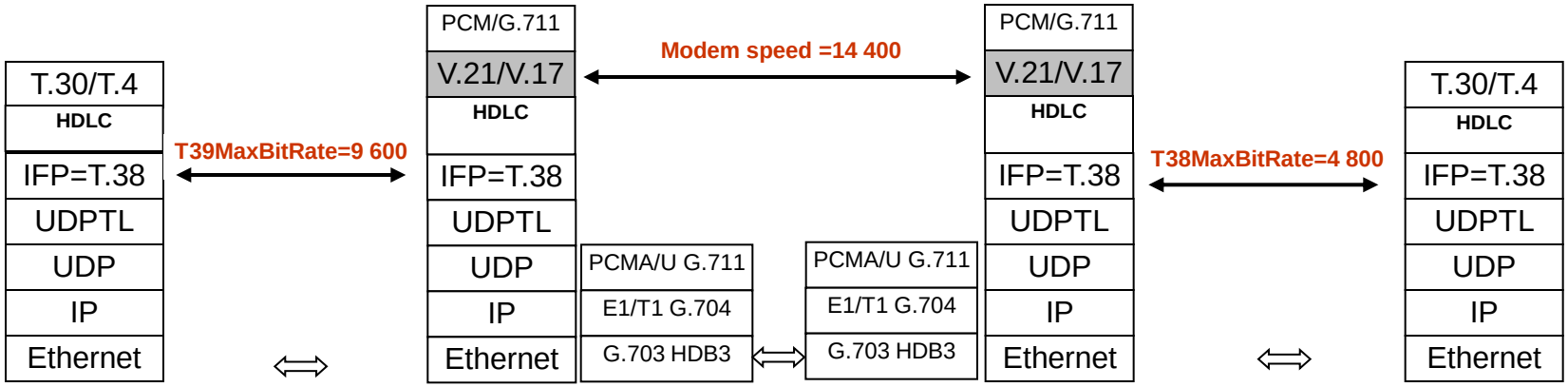
Testing results – SIP not end-to-end

Problems: Different MaxBitRate, Modem speed higher than MaxBitRate

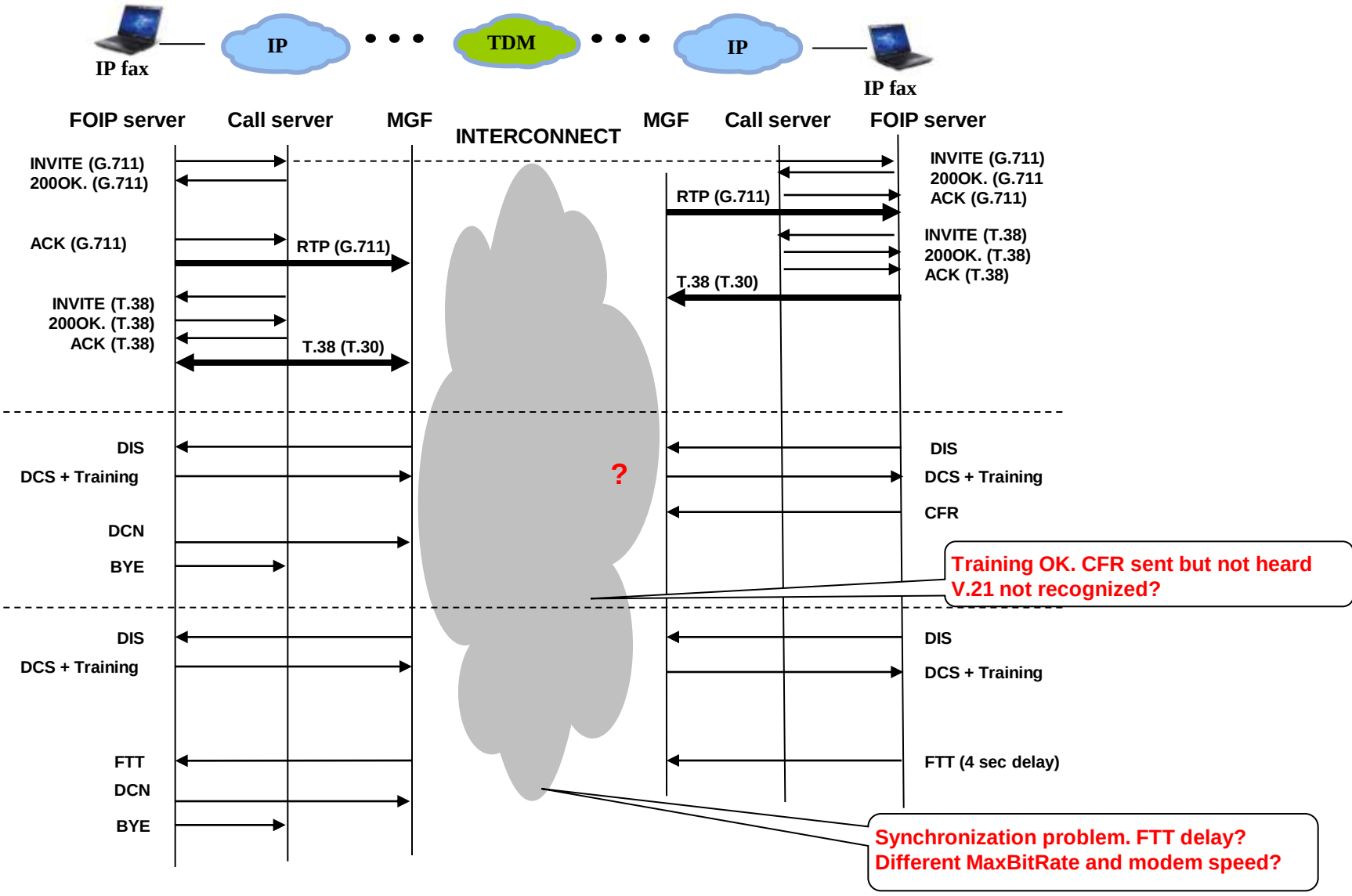
MaxBitRate negotiated by SIP → INVITE/OK in signalling layer

Modem speed negotiated by T.30 → DIS/DCS in media layer

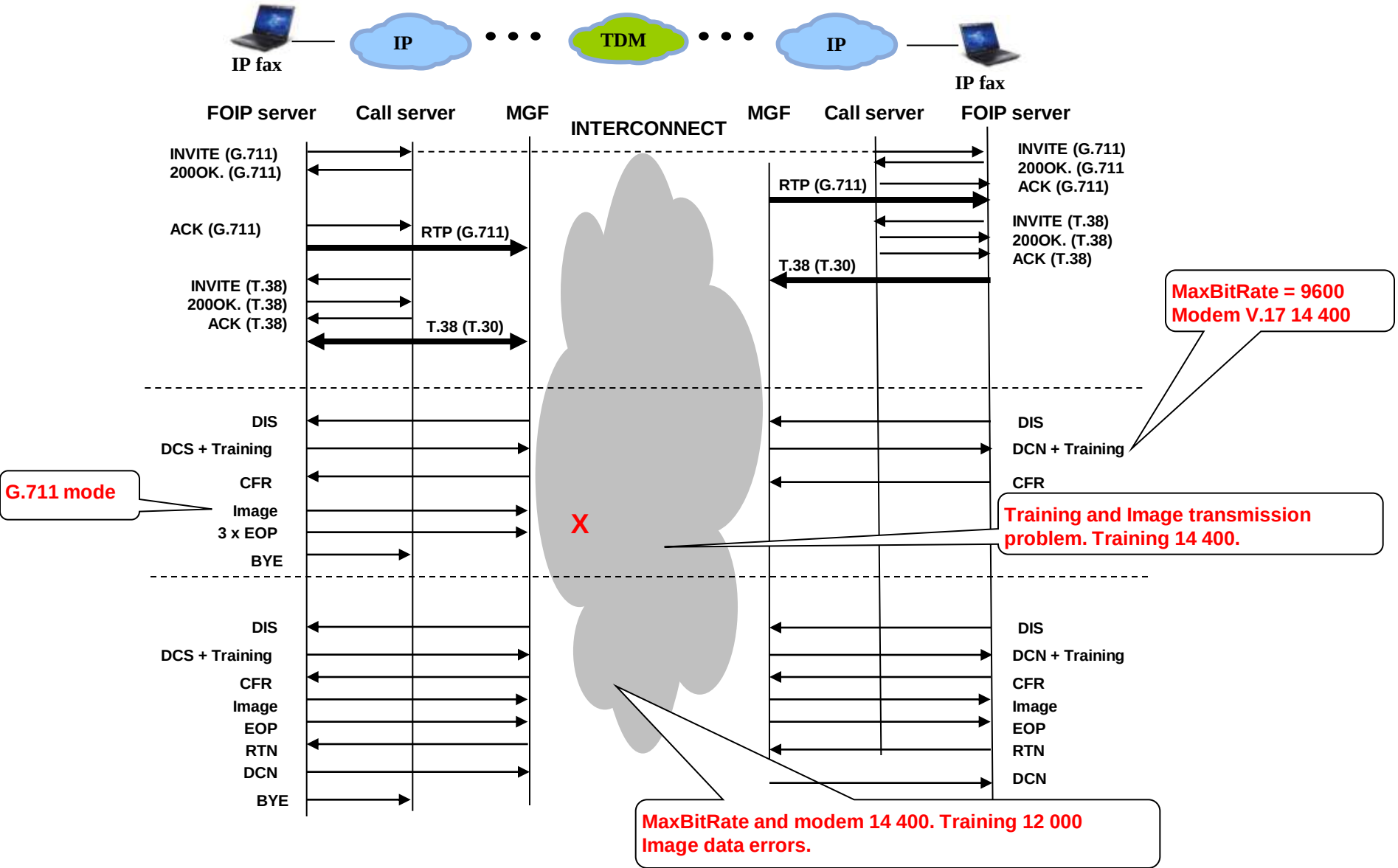
It should be finally agreed by training but …..



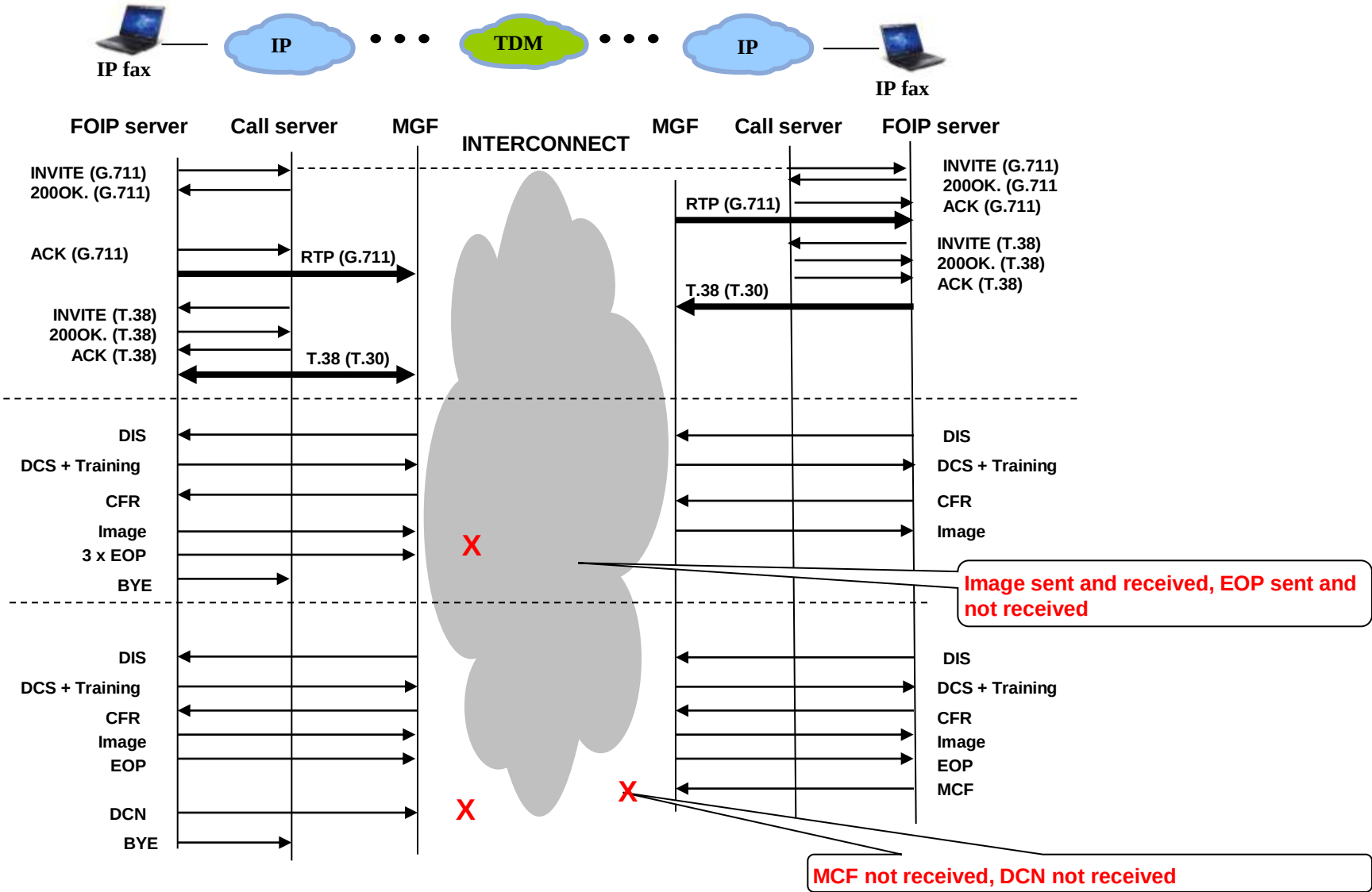
Testing results – Training problems



Testing results – Image errors



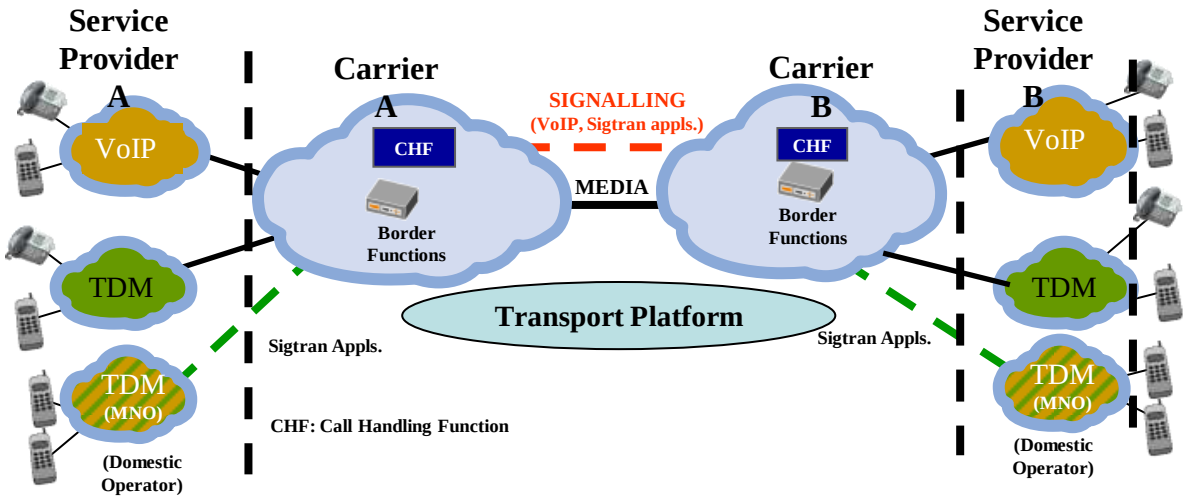
Testing results – Post Image errors



Conclusions

All IP private Interconnect !

1. **Most suspected network elements are:**
   - **Signalling gateways (SIP to C7 and back)**
   - **Media gateways**
2. **Most of this effects could be avoided if fax call originated by IP endpoint is routed via all IP route.**
3. **If all IP route could guarantee QoS that would eliminate transmission errors.**



In all IP testing configuration all T.38 calls were successful. Some FoIP calls in audio mode failed because of burst packet loss

Call handling based on UA capabilities

RFC 3840:

Indicating User Agent Capabilities in the Session Initiation Protocol

RFC 3840 defines how UA can convey its capabilities using feature tags that are:

- added as parameters to the Contact header field
- used to convey information about required UA capabilities.

Feature tag groups: base group, sip.methods, sip.events, sip.extensions, sip.schemes…

RFC 3841:

Caller Preferences for the Session Initiation Protocol (SIP)

RFC 3841 defines:

- how a caller can express preferences using three new request header fields,
  - Accept-Contact,
  - Reject-Contact
  - Request-Disposition
- How the requests with preferences are handled in servers and proxies.
- How are chosen the Uniform Resource Identifiers (URI) a request gets routed to

New feature tag in RFC 3840

Media feature tag name:

- sip.fax

Values appropriate for use with this feature tag:

- t38: The device supports the ITU-T T.38[T38] standard for fax communication.
- passthrough: The device supports the transmission of fax using the ITU-T G.711[G711] audio codec.

Source:

(https://datatracker.ietf.org/doc/draft-hanes-dispatch-fax-capability/)

Example

UA gets registered as fax capable:

REGISTER sip:example.com SIP/2.0
Via: SIP/2.0/TCP bob-TP@example.com;branch=z9hG4bK309475a2
From: <sip:bob-tp@example.com>;tag=a6c85cf
To: <sip:bob-tp@pexample.com>
Call-ID: a84b4c76e66710
Max-Forwards: 70
CSeq: 116 REGISTER
Contact: <sip:bob-tp@example.com;transport=tcp>;+sip.fax="t38„

SIP/2.0 200 OK
From: <sip:bob-tp@example.com>;tag=a6c85cf
To: <sip:bob-tp@example.com>;tag=1263390604
Contact: <sip:bob-tp@example.com;transport=tcp>;+sip.fax="t38"
Expires: 120
Call-ID: a84b4c76e66710
Via: SIP/2.0/TCP bob-TP@example.com;branch=z9hG4bK309475a2
CSeq: 116 REGISTER
Expires: 3600

INVITE with feature tag allows to find appropriate UA:

INVITE sip:UserY@example.com SIP/2.0
From: sip:UserX@operator.com
To: sip:UserY@example.com
Accept-Contact: *;+sip.fax="t38"
Content-Type: application/sdp

Additional clarification needs

1. Draft (and RFC 3841) explains explicitly how to route incoming call inside target domain. **Inter domain routing is not described.**

2. It is a need to draft an „Implementation guide" that will define how +sip.fax feature tag should be used by Class 4 Softswitch (SIP proxy) for intelligent interconnection routing between different domains.

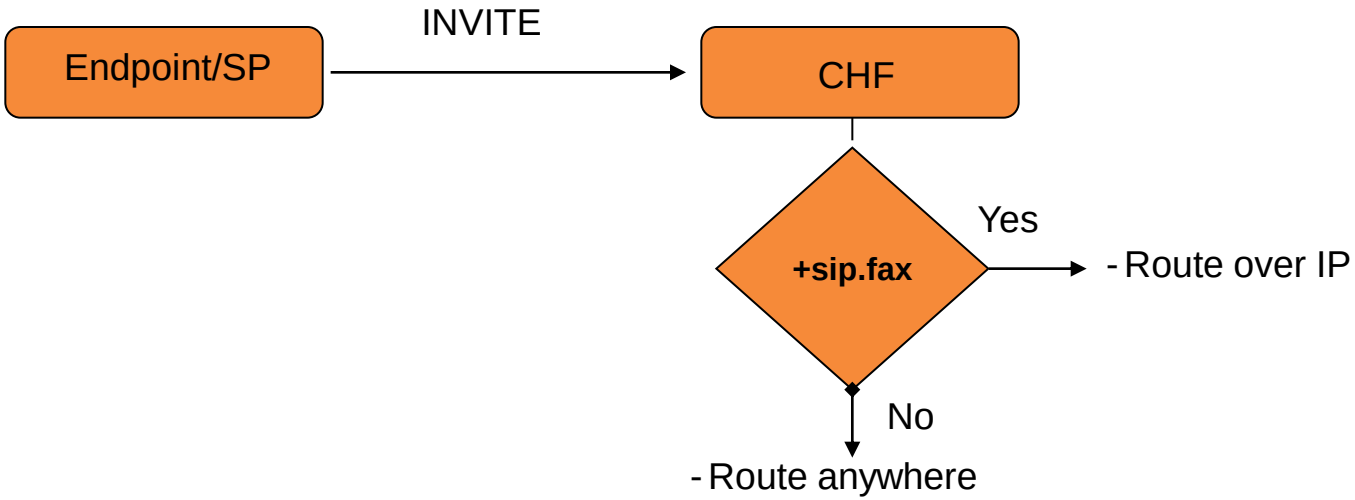
3. Implementation guidelines should describe the way the edge proxies inspect the Contact header in an INVITE request for the +sip.fax feature tag, and then modify their call routing processes accordingly if the tag is present.

4. Implementation of such guidelines would make to some extent an edge proxy non-compliant with RFC 3841, but that should not cause any harm.

Implementation of +sip.fax

Simple algorithm

Many entities involved



Requirements:

- RFC 3840 update and Implementation Guide
- Implementation of +sip.fax in UAC (fax terminals)
- Implementation of intelligent routing in Call Handling Function

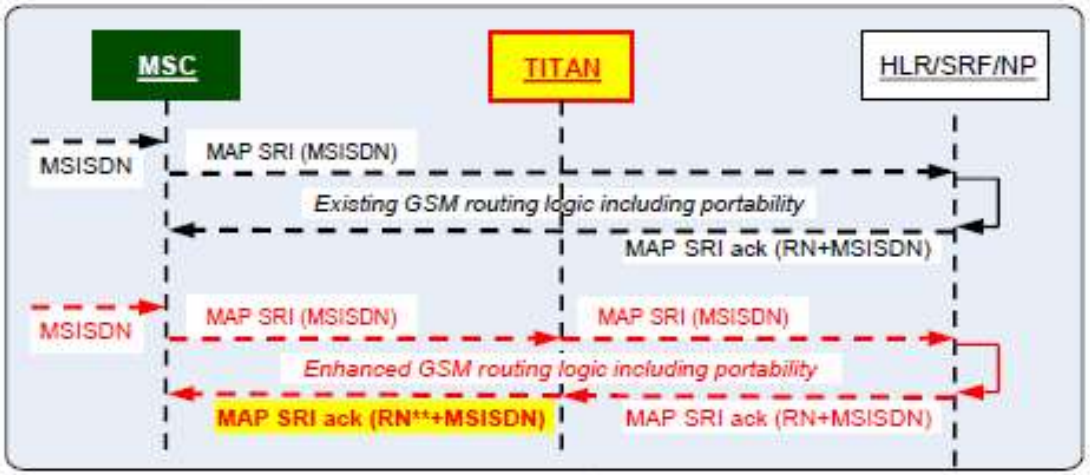
Self learning solution

- 9th IMS World Forum 2012
- Presentation by Pieter Veenstra KPN.

Reference ETSI TR 184 003

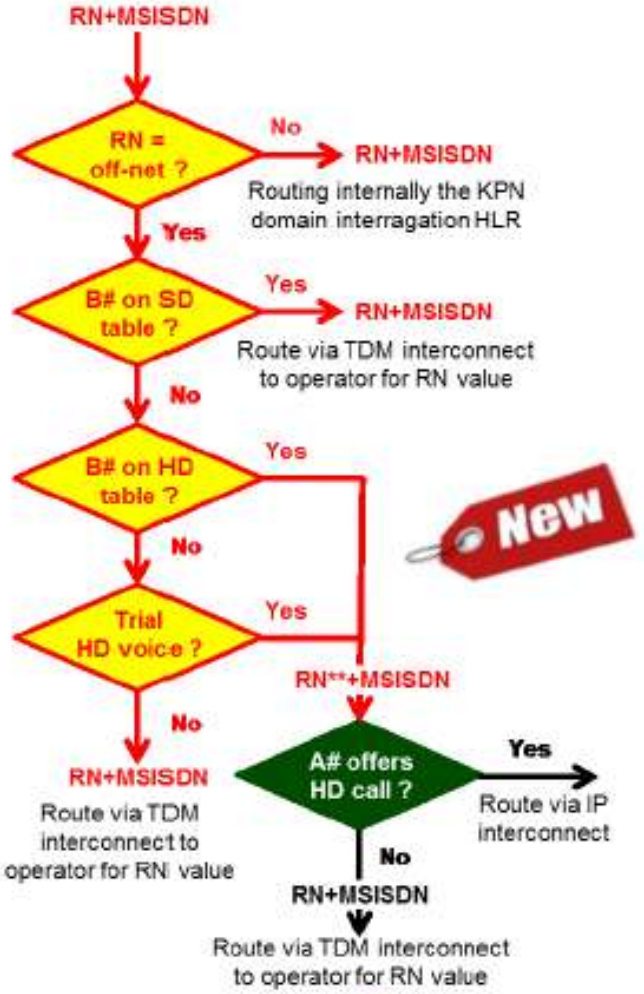
3 IMS Peering considerations IP Interconnection

f) Self-learning HD Voice discovery service – selection IP or TDM?



Origin Network

- TITAN is transparent for MAP SRI
- TITAN checks RN+MSISDN in MAP SRI ack
- Only if potential HD voice call (B-party supports HD) or trialling call for HD voice, then call potentially routed via IP interconnect
- 2 tables in TITAN with self-learned SD voice and self-learned HD voice numbers, respectively (see next slide)
- Trial HD voice calls on max ## call attempts via IP interconnect
- MSC uses RN**+MSISDN in analysis but call routed to TDM interconnect if A-party not HD voice call (no HD codec)

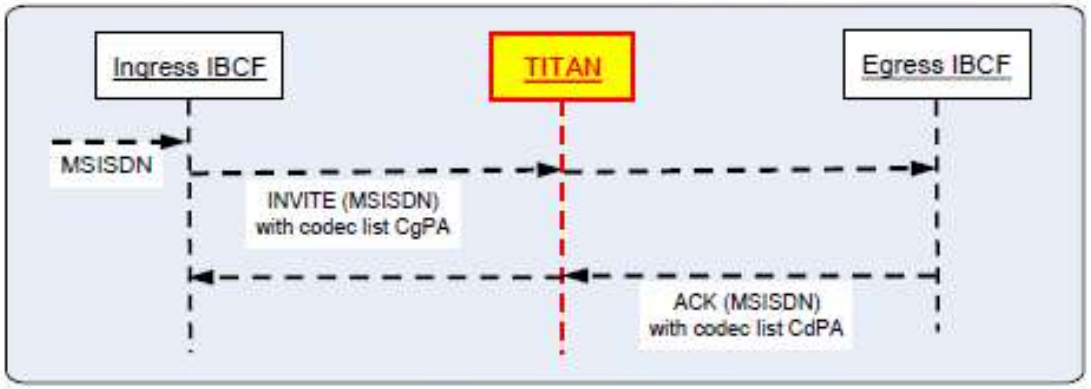


NetNumber acme packet kpn

Source: Pierter Veenstra KPN

3 IMS Peering considerations IP Interconnection

f) Self-learning HD Voice discovery service – contents codec list?



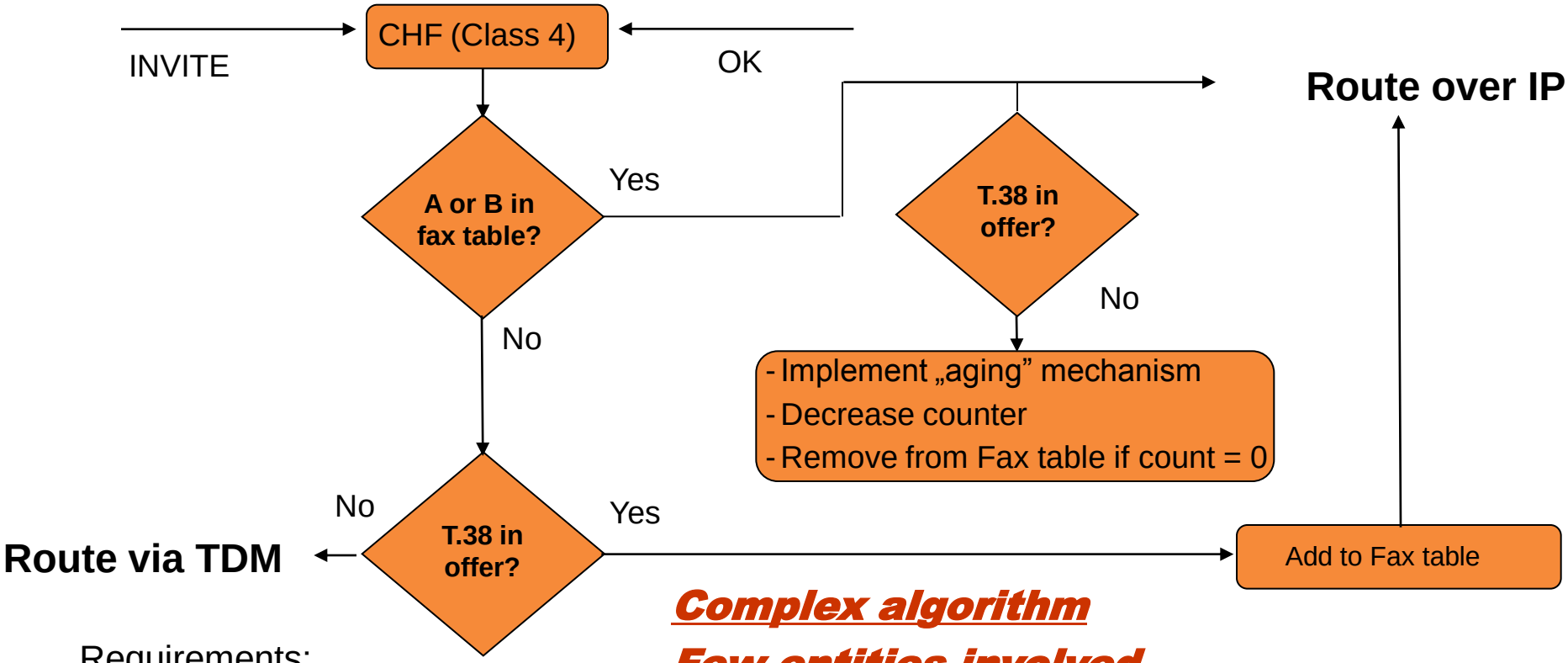
SIP-X Network

- TITAN checks codec list in INVITE
  - If HD codec is included then MSISDN added to table with HD voice numbers
  - If no HD codec included then MSIDN added to table with SD voice numbers
  - Age off mechanism
    - If number on HD table, and for N consecutive calls number no longer supports HD, then delete number from HD table and add number to SD table
    - If number on SD table, and number supplies HD for any call, delete number from SD table and add to HD table
- TITAN reported about codec list in ACK (and other messages)
  - IBCF report TITAN about codec capabilities of MSISDN's of handled calls
  - Catering for 'early offer' scenarios in addition to 'delayed offer' scenarios

NetNumber acme packet kpn

Source: Pierter Veenstra KPN

Self learning algorithm (example)



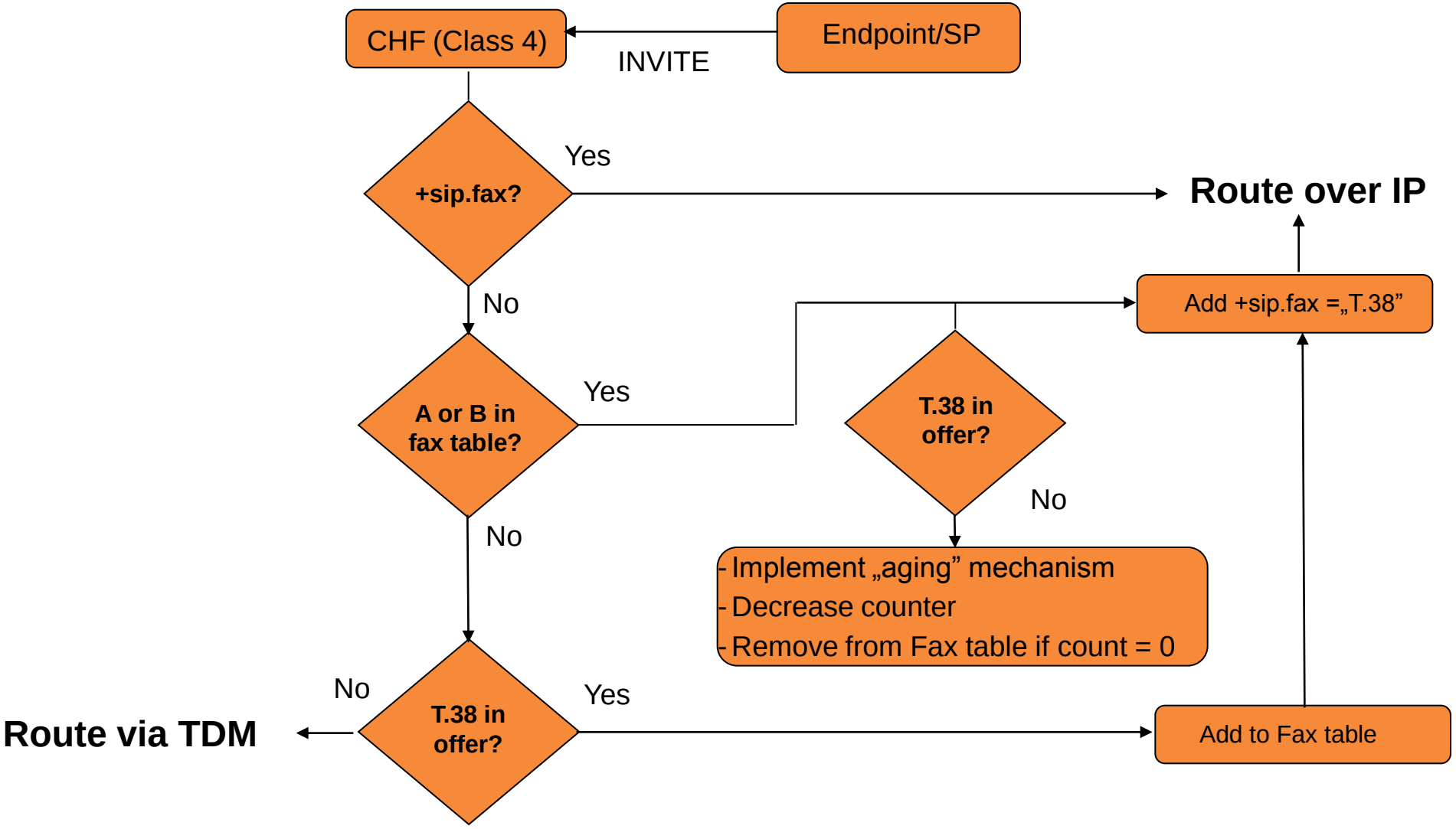
Complex algorithm

Few entities involved

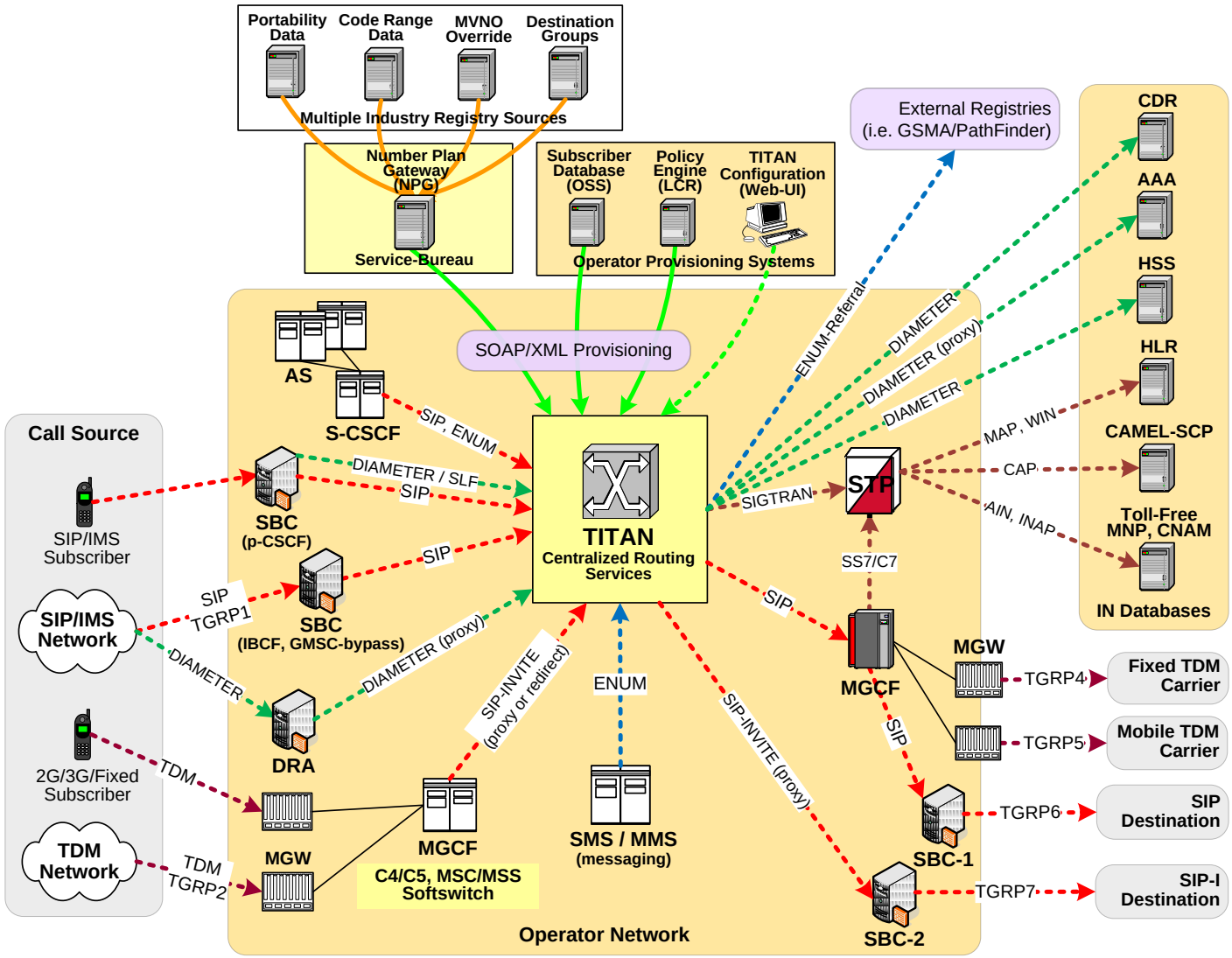
Requirements:

- No standardization needed
- Implementation of self learning algorithm and intelligent routing in Call Handling Function
- Creation and maintenance of „Fax number table”
- Action limited to interconnection segment (only carriers implement modification).

Self learning and +sip.fax (example)

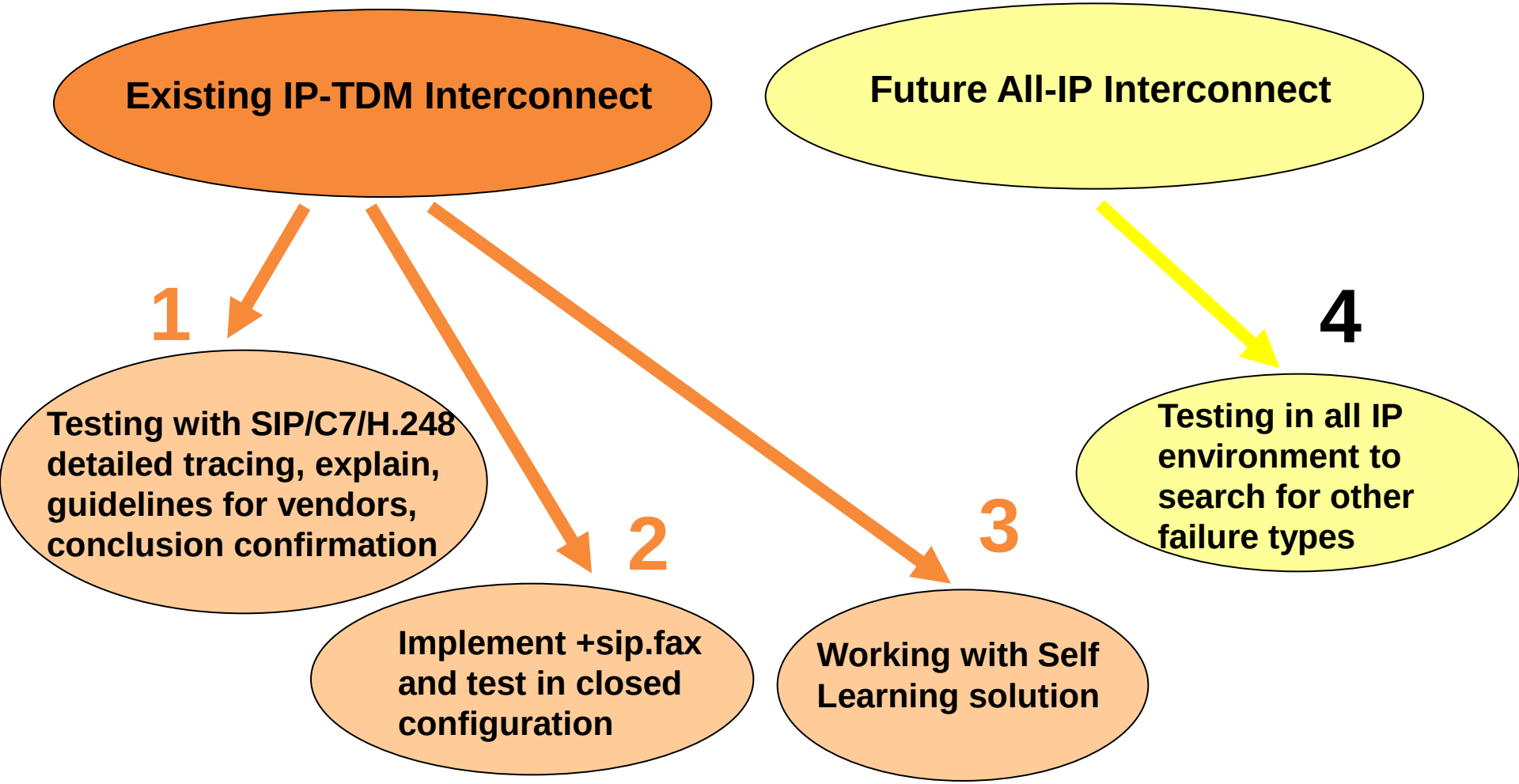


NetNumber TITAN architecture



Source: NetNumber

Next Steps – possible directions



Thank you

Backup

G3 timers

Timer	Value	Description
T0	60 ± 5 sec	Amount of time a calling fax machine waits for an answer from the terminating side. Usually a CED or DIS stops this timer.
T1	35 ± 5 sec	Amount of time a fax device attempts to identify the other fax device. This timer is active during the DIS/DCS negotiation.
T2	6 ± 1 sec	Amount of time a fax device waits to receive a command. This timer also detects the loss of command/response synchronization.
T3	10 ± 5 sec	Amount of time a fax device alerts an operator after a procedural interrupt.
T4	3 sec ± 15%	Amount of time a fax device waits for a response to a sent message.
T5	60 ± 5 sec	Amount of time a transmitting fax device waits for a busy condition on the receiving fax device to clear. This timer is only used during ECM.