15. - SIP and the PSTN

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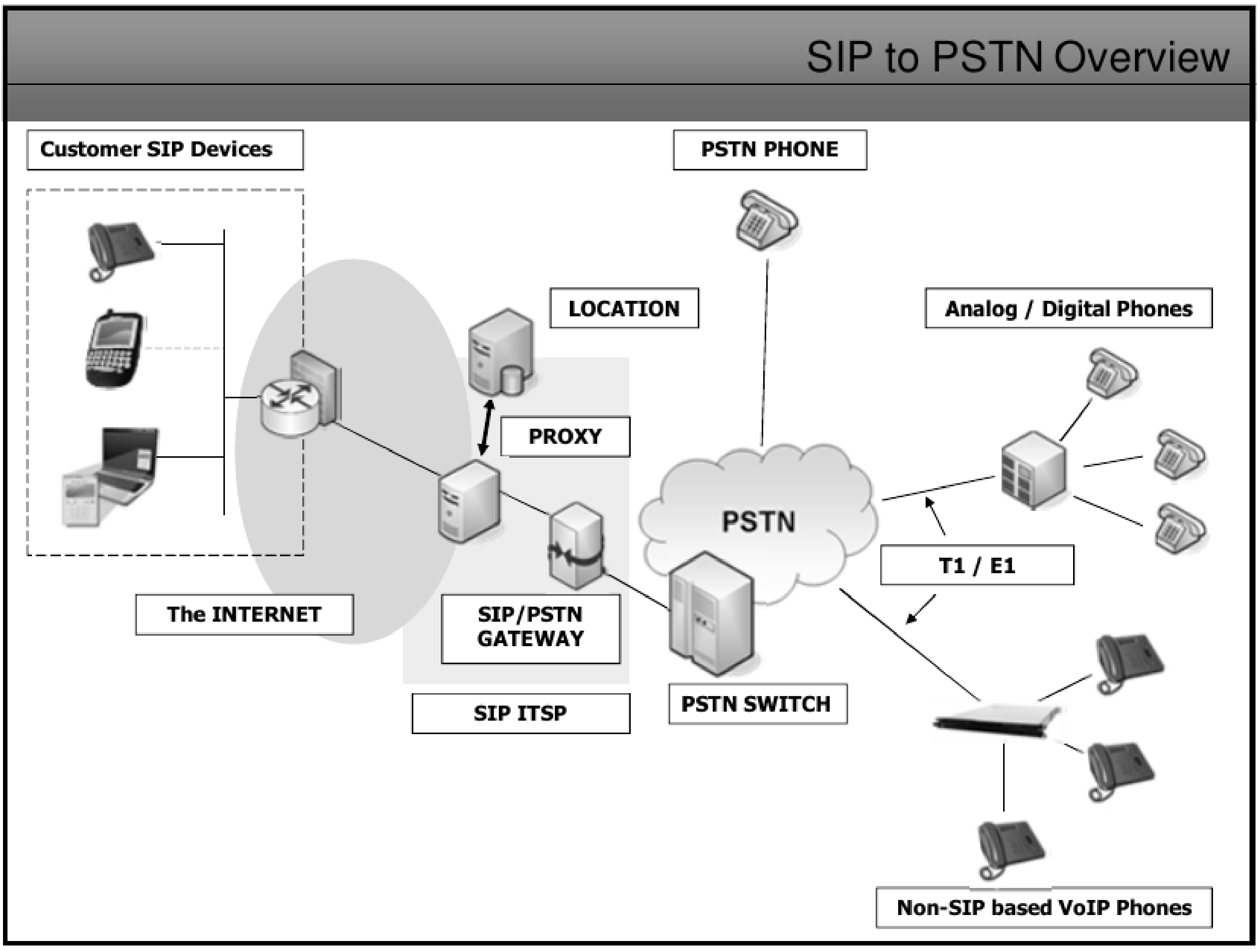
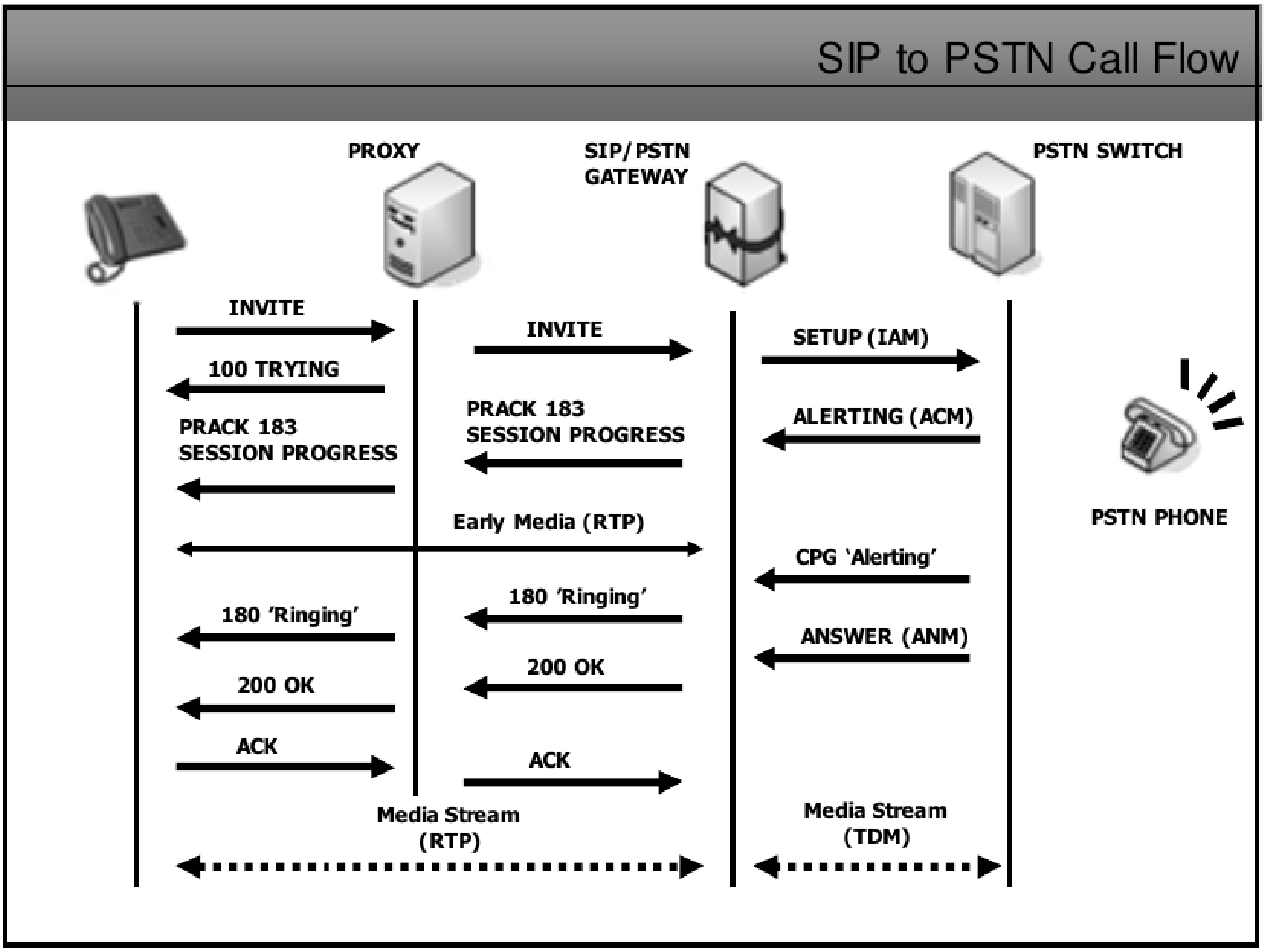
- More on SIP addressing
- DTMF and SIP
- 

Here our SIP phone dial the number of a PSTN device, the proxy will send the invite onto the SIP/PSTN Gateway that will next sent the proper setup message to the PSTN switch then to the telephone.

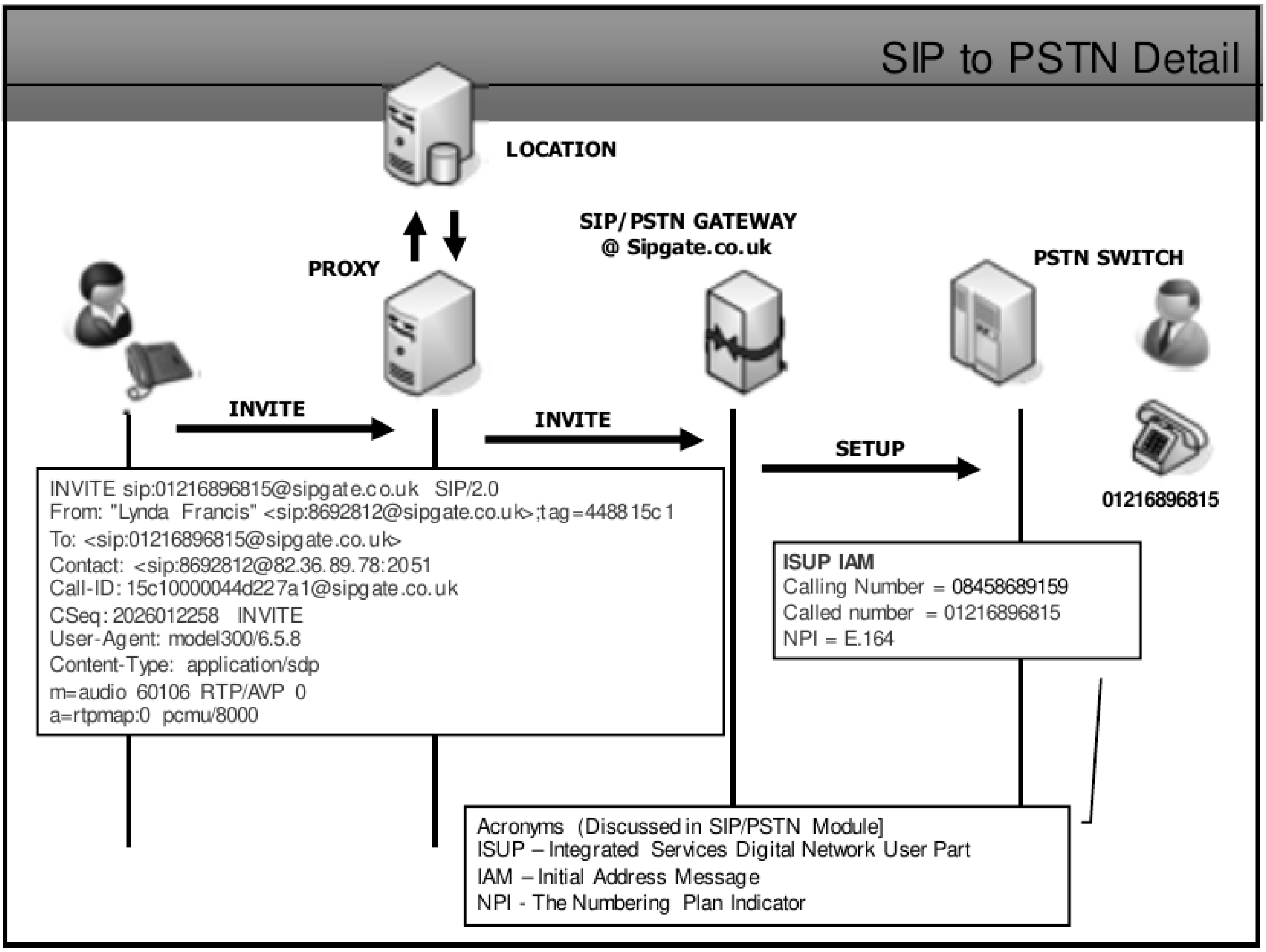
A PRACK 183 inform the SIP phone of the progress of the call setup and setup an Early Media (RTP)

Once the PSTN phone start ringing the alerting 180 Ringing messages are sent back to our SIP phone along with the 200 OK

Our SIP phone will response with an ACK then the media will start to flow, it is up to the SIP/PSTN gateway to convert the media from RTP to TDM and viceversa



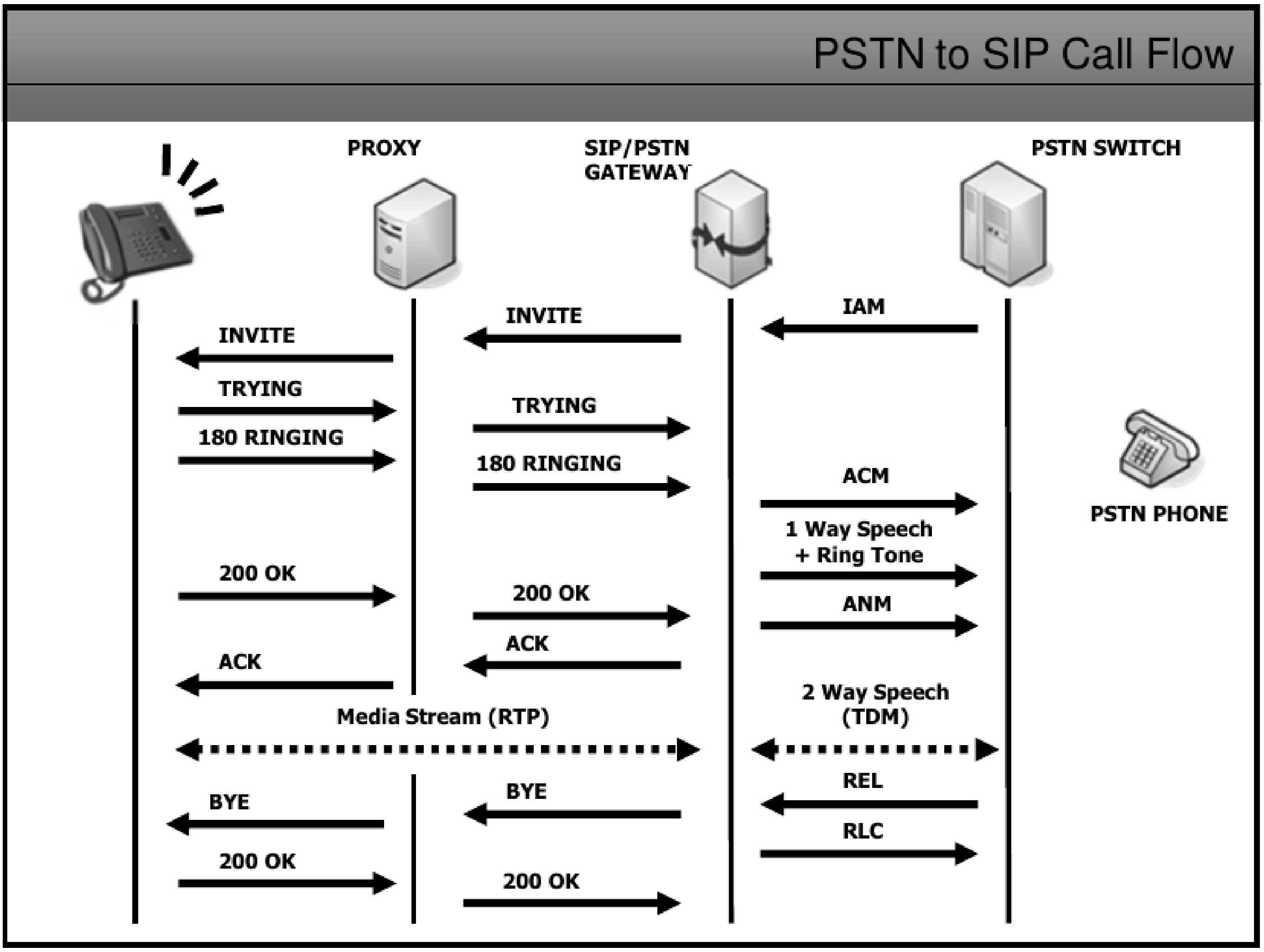
In more detail 01216896815, the SIP proxy uses the location services and determines the number is on the PSTN it then passes the invite to the SIP/PSTN Gateway which thenn by the PSTN Switch goes to Grahams phone. The detail on the next image shows the INVITE along with the contact details and SDP information



Some other sample SIP Codes re: SIP to PSTN call setup attemps are

- 404 Not Found : The number called is unallocated
- 407 Proxy authenticarion Req : Call Rejected
- 480 Temporary Unavailable : No User Response
- 486 Busy Here : Called Party is Busy
- 502 Bad Gateway : Network out of order
- 600 Busy Everywhere : Useer Busy
- 603 Decline : Call Rejected
- 604 Does not exist anywhere : Unallocated number

Here is an example of a PSTN to SIP Call Flow



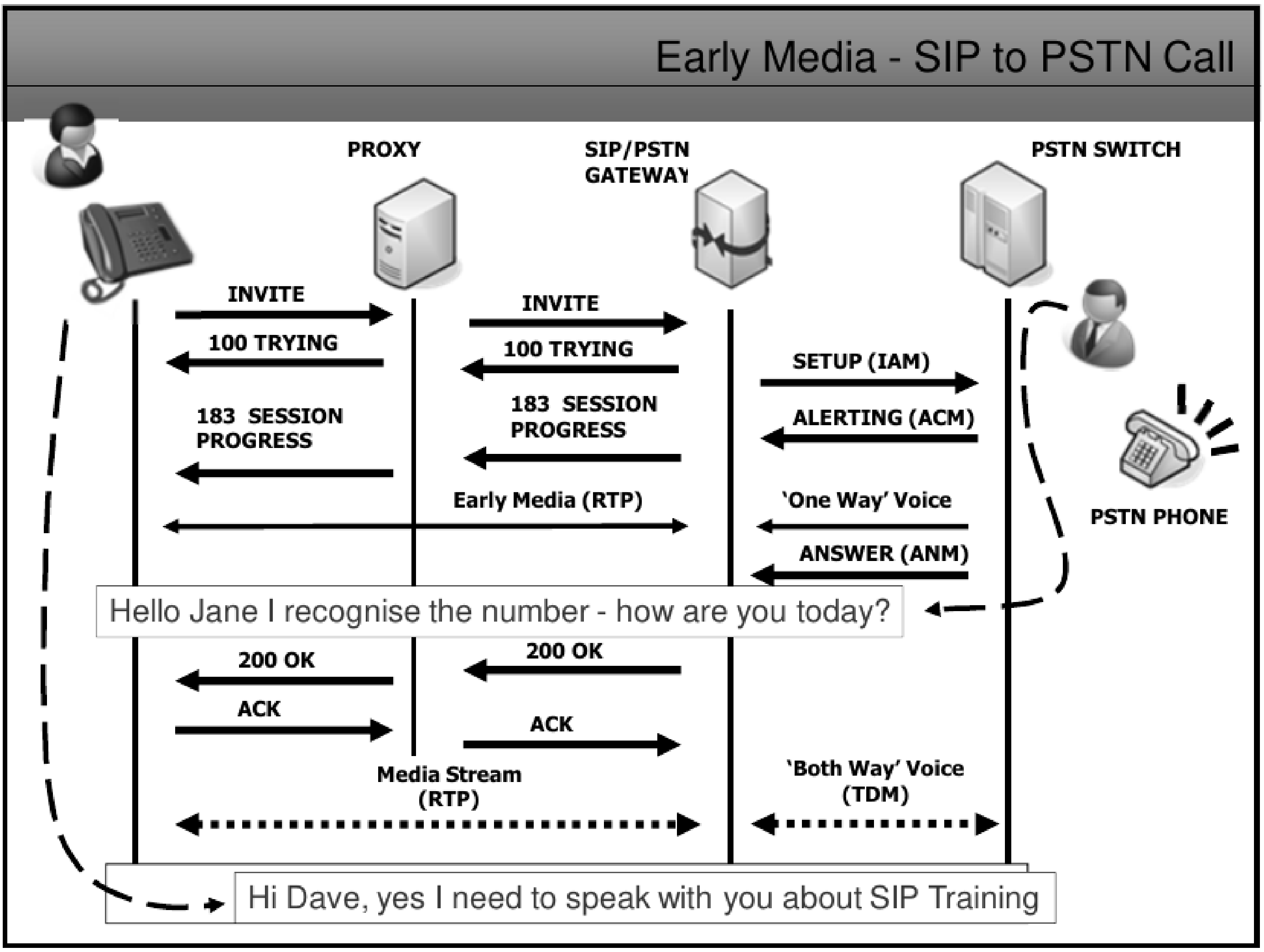
Early Media

To overcome a few problems that arise due to the two different systems such as SIP and the PSTN trying to work together, a concept called 'Early Media' has been introduced

Early media is not required on standard PSTN calls as when a number is dialed a media channel is established so that the caller can hear the ringing tone of the remote device, This also gives companies the opportunity to replace ringing media with corporate messages or other instructions for the caller before they get to speak to a real person.

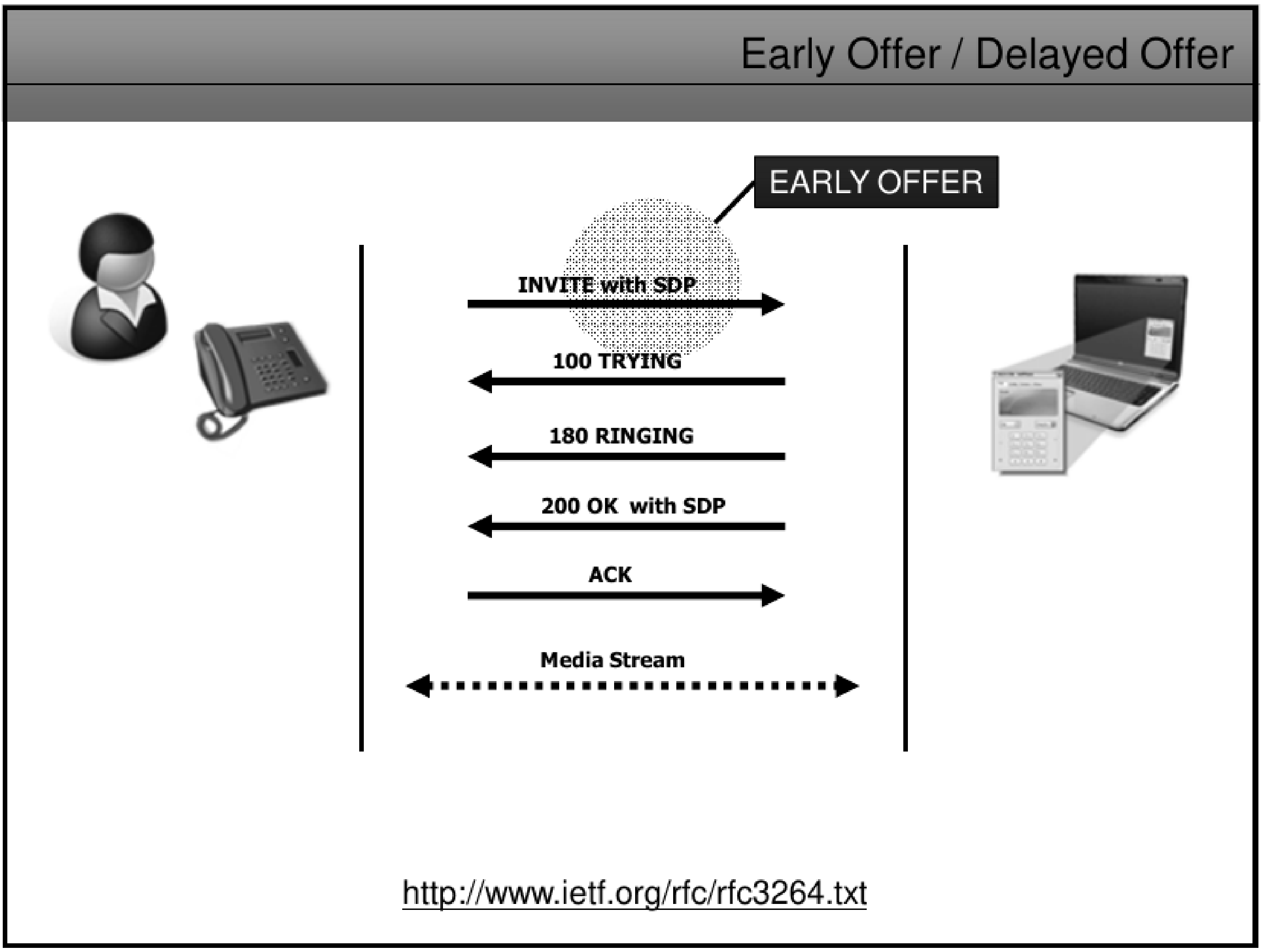
Clipping is a problem where if a person using a PSTN phone answer their phoene adn starts talking straight away without Early Media the SIP phone that is calling them will miss the first part of the conversation as it hasn't yet received a 200 ok message to enable it to setup the RTP media path.

Early media also allows Busy Tone and other announcements to be played to the caller even though the called phone has not been picked up.

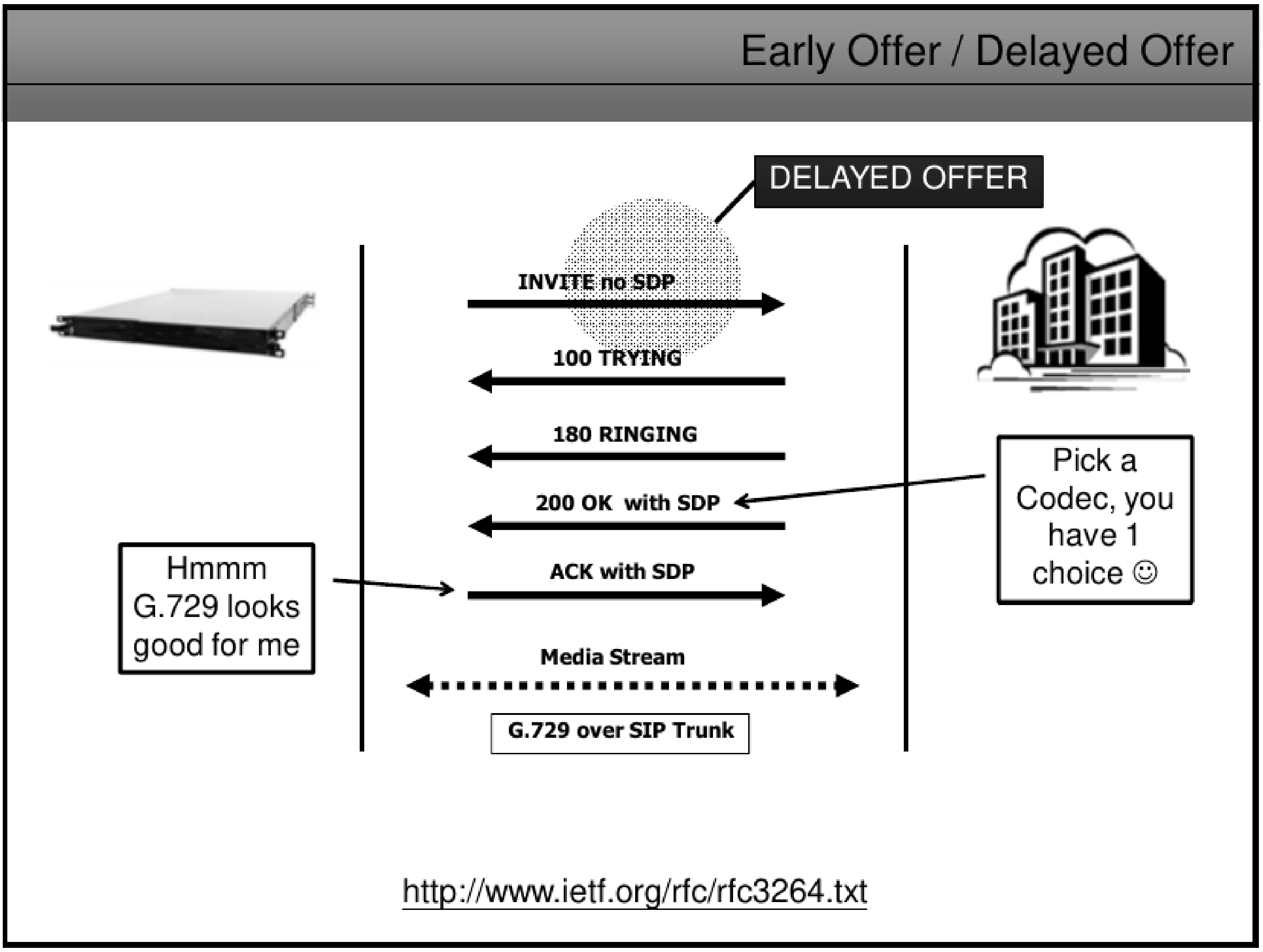


Early Offer / Delayed Offer

With Early offer the invite sends the SDP to being voice inmediately



With delayed offer Invite goes without SDP allowing the other end select a codec to use, if available it will reply with an ACK SDP to begin media.



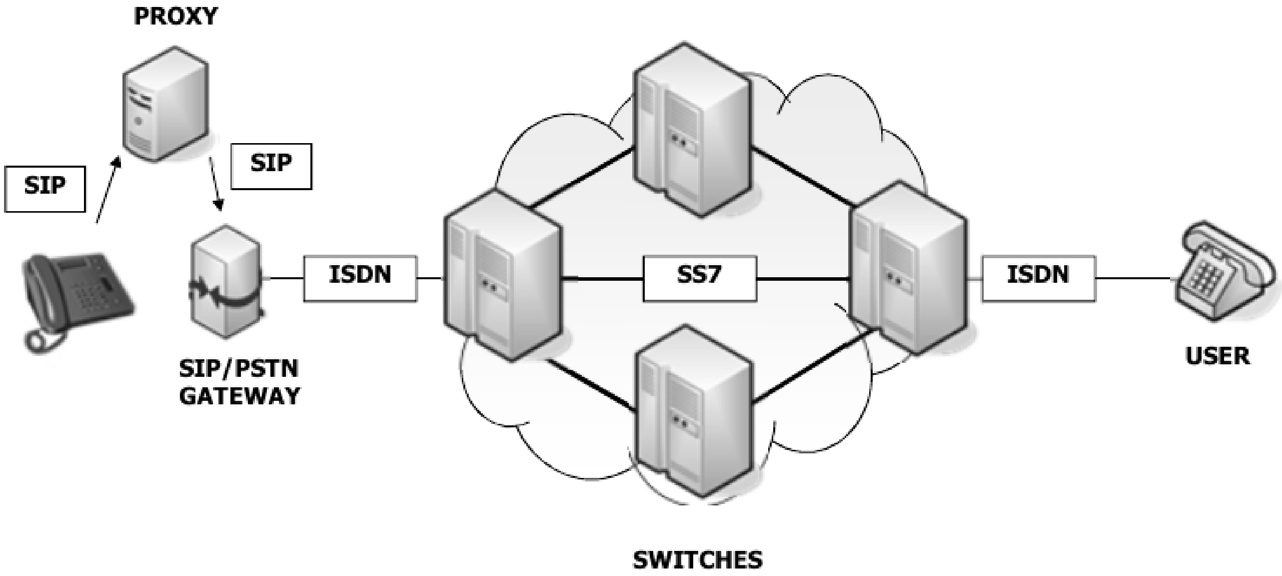
SIP Bridging or SIP-T is a framework that can enable SIP networks to carry legacy telephone signals across an IP based network to another legacy network, This would have a dramatic effect on long distance call charges which is great for end users but of course there are technical issues that need to be addressed.

The legacy SS7 ISUP messages have to be interrogated by the SIP/PSTN gateway and then the information that will help SIP proxies to route the SIP messages is built into the SIP header, while othert ISUP information added as a MIME message body, to ensure that the body is closed to 'unwanted' parties snooping on the network it can be encrypted and this is discussed in the security module course.

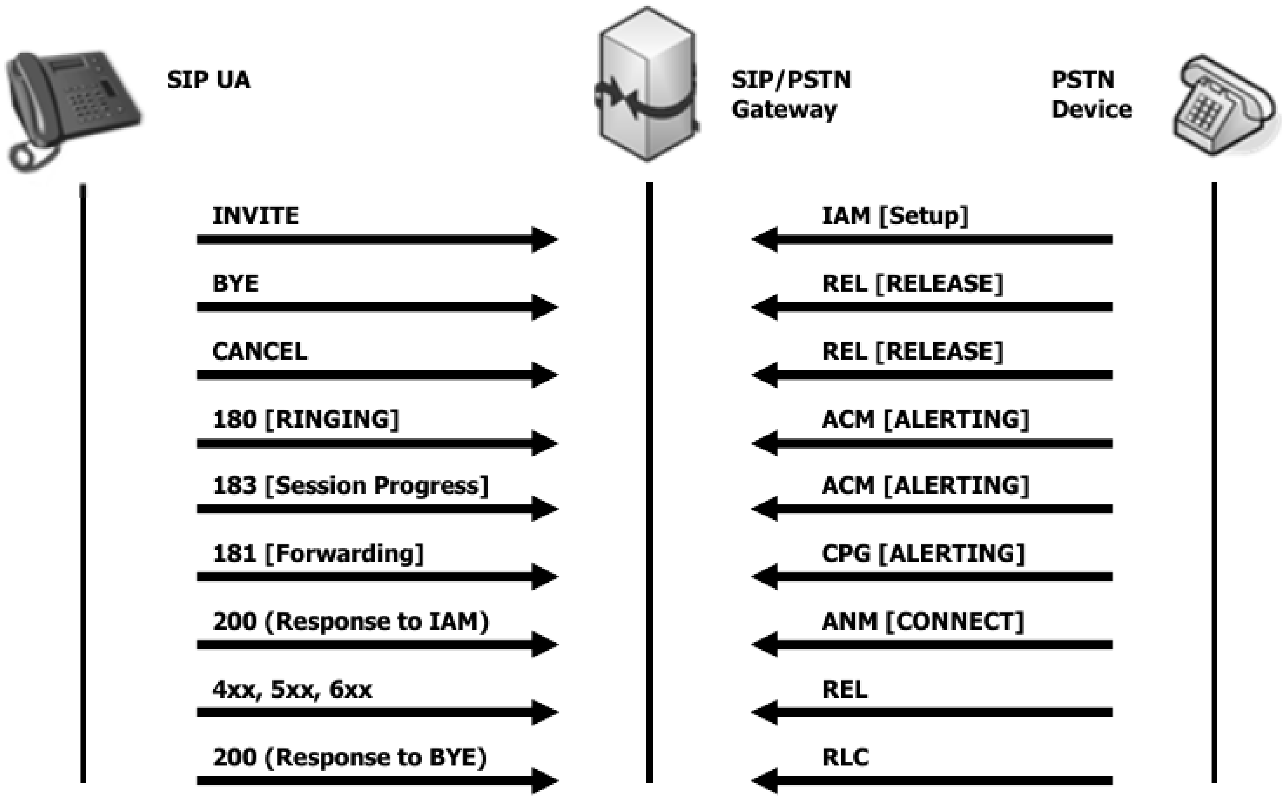
SIP INFO is another SIP-T approach or method that is used for in-call ISUP signaling across an IP network.

In essence SIP-T must provide translation beteen protocols and providee feature transparency across PSTN to SIP to PSTN interconnections.

SS7 (Signalling system 7) sets up network connections between switches within the PSTN whilst ISDN sets up the connections between the user part and the network boundary. They are partners in getting connections setup and we particularly need to address the ISDN User Part (ISUP) Signalling detail so that we can understand more how SIP can take on this role in order to provide connectivity services.



ISUP and SIP messages

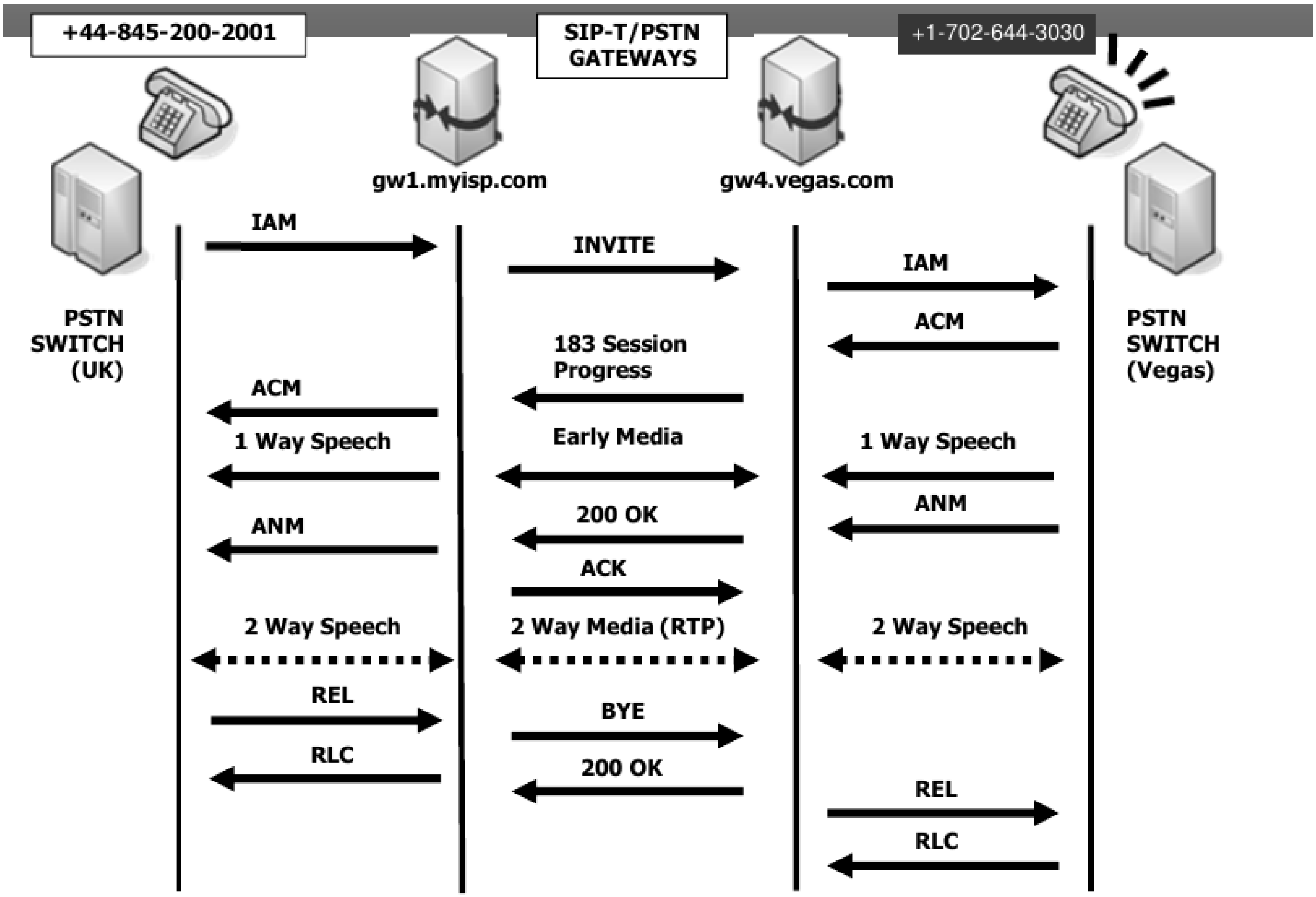


ISDN User part (ISUP) to SIP codes

ISUP Event Code	SIP Message
1: Alerting	180: Ringing
2: Progress	183: Session Progress
3: In-band Information	183: Session Progress
4: Call forward; Line Busy	181: Call is being Forwarded
5: Call forward; No reply	181: Call is being Forwarded
6: Call forward; unconditional	181: Call is being Forwarded

ISUP Cause Code	SIP Message
1: Unallocated Number	404: Not Found
2: no route to network	404: Not Found
17: User Busy	486: Busy Here
18: No User Responding	408: Request Timeout
21: Call Rejected	403: Forbidden (see RFC details)
28: Address Incomplete	484: Address Incomplete
34: No Circuit Available	503: Service Unavailable
38: Network out of order	502: Service Unavailable

Here is an example of a PSTN to PSTN via SIP



ISUP Encapsulation / SDP also known as SIP Bridging

If this is all that a client requires, then all of the ISUP information can be held in the body of the SIP message and the details such as To: From: Contact, etc. never changes as these just relates to the SIP gateway addresses, an example of the SDP portion of the SIP message (mime/isup) is below, This way any ISUP information that cannot be mapped to SIP will be retained and in this case, secured as well.

```
INVITE sip:+1-702-644-3030@vegas.com; user=phone SIP/2.
Via: SIP/2.0/UDP gw1.myisp.com/5060
To: sip:+1-702-644-3030@vegas.com;user=phone
From: sip:+44-845-200-2001@gw1.myisp.com;user=phone
Call-ID: 15c00001-44dec3@myisp.com
CSeq: 1 INVITE
Contact: sip:+1-702-644-3030@vegas.com;user=phone
Content-Type: application/sdp
Content-Length: 166

v=0
o=GW1 2880823527 2880823527 IN IP4 gw1.myisp.com
s=Session SDP
c=IN IP4 gw1.myisp.com
t=0 0
m=audio 4007 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

- The o field show which originating gateway is going to connect to the gateway in vegas and then forward the SIP message.
- The c field also shows originating gateway details

Addressing Notes

If complete encapsulation is not being used then ISUP information needs to be mapped to SIP elements. For example, a call from a PSTN device into a SIP network would complete the SIP FROM field using their E.164 format number such as

```
⊞ From: 01216896815 <sip:01216896815@gw02.uk.sipgate.net>
```

Here, the number 01216896815 is a device on the PSTN and it has gained the @gw02.uk.sipgate.net suffix when entering into the Sipgate network

If the PSTN caller has requested 'privacy' and their number has subsequently been withheld, then the SIP header will look something like the example below

```
⊞ From: "anonymous" <sip:unbekannt@217.10.69.13>
```

```
⊞ To: <sip:448458689159@217.10.79.8>
⊞ Contact: <sip:unbekannt@217.10.69.13>
  Call-ID: 7e7b39720ba012077d5cc3490bcba449@217.10.69.13
```

SIP and DTMF

As well as DTMF Dialling tones there are a lot more tonees and events that need to be considered when carrying them across a SIP/VoIP network

DTMF Tones

Event	encoding (decimal)
0--9	0--9
*	10
#	11
A--D	12--15
Flash	16

Fax-related tones

Event	encoding (decimal)
Answer tone (ANS)	32
/ANS	33
ANSam	34
/ANSam	35
Calling tone (CNG)	36
V.21 channel 1, "0" bit	37
V.21 channel 1, "1" bit	38
V.21 channel 2, "0" bit	39
V.21 channel 2, "1" bit	40
CRdi	41
CRdr	42
CRe	43
ESi	44
ESr	45
MRdi	46
MRdr	47
MRe	48
CT	49

Standard subscriber line tones

Event	encoding (decimal)
Off Hook	64
On Hook	65
Dial tone	66
PABX internal dial tone	67
Special dial tone	68
Second dial tone	69
Ringing tone	70
Special ringing tone	71
Busy tone	72
Congestion tone	73
Special information tone	74
Comfort tone	75
Hold tone	76
Record tone	77
Caller waiting tone	78
Call waiting tone	79
Pay tone	80
Positive indication tone	81
Negative indication tone	82
Warning tone	83
Intrusion tone	84
Calling card service tone	85
Payphone recognition tone	86
CPE alerting signal (CAS)	87
Off-hook warning tone	88
Ring	89

Country specific subscriber line tones

Event	encoding (decimal)
Acceptance tone	96
Confirmation tone	97
Dial tone, recall	98
End of three party service tone	99
Facilities tone	100
Line lockout tone	101
Number unobtainable tone	102
Offering tone	103
Permanent signal tone	104
Preemption tone	105
Queue tone	106
Refusal tone	107
Route tone	108
Valid tone	109
Waiting tone	110
Warning tone (end of period)	111
Warning Tone (PIP tone)	112

Trunk Events

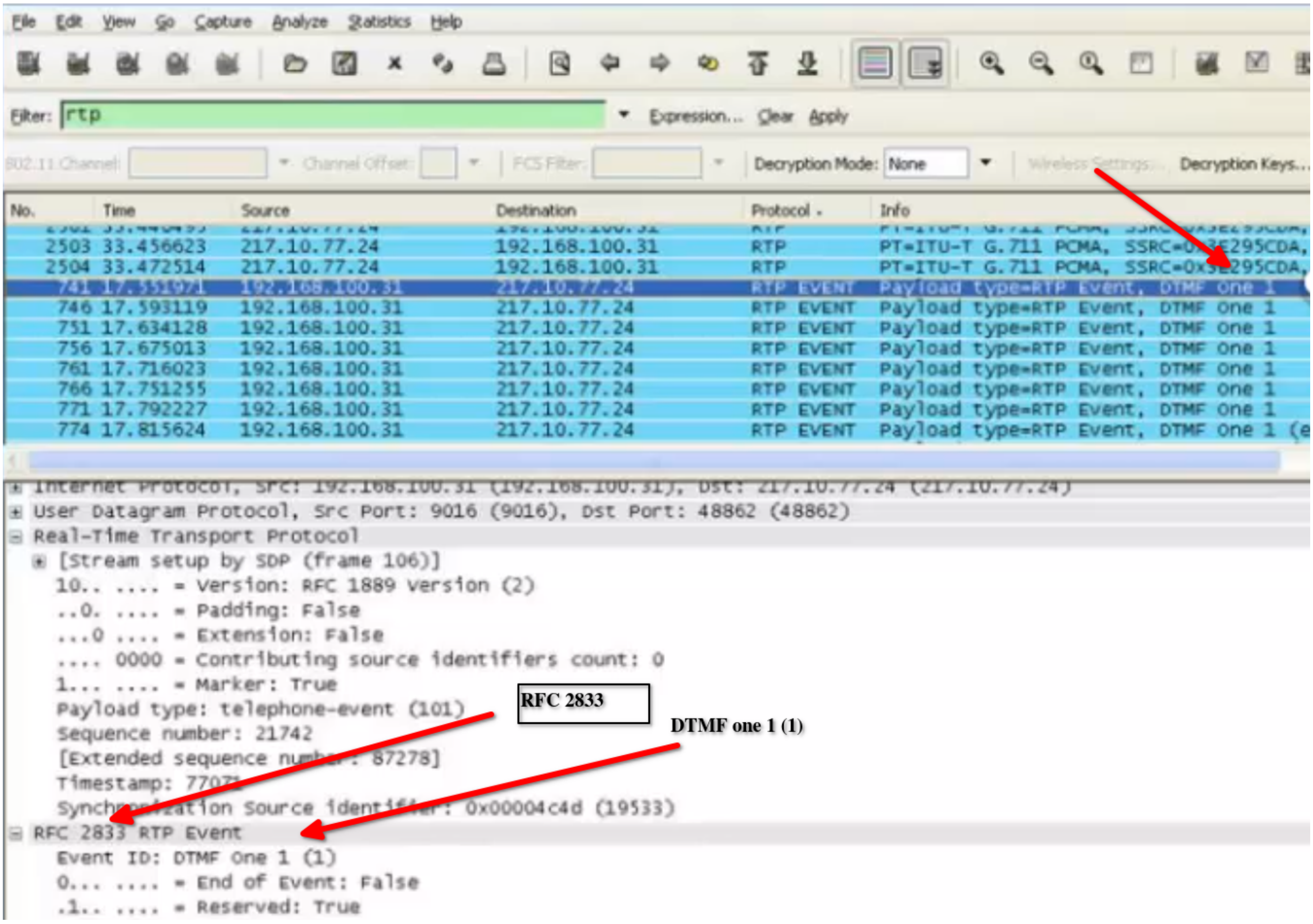
Event	encoding (decimal)
MF 0... 9	128...137
MF K0 or KP (start-of-pulsing)	138
MF K1	139
MF K2	140
MF S0 to ST (end-of-pulsing)	141
MF S1... S3	142...143
ABCD signaling (see below)	144...159
Wink	160
Wink off	161
Incoming seizure	162
Seizure	163
Unseize circuit	164
Continuity test	165
Default continuity tone	166
Continuity tone (single tone)	167
Continuity test send	168
Continuity verified	170
Loopback	171
Old milliwatt tone (1000 Hz)	172
New milliwatt tone (1004 Hz)	173

Different types of DTMF

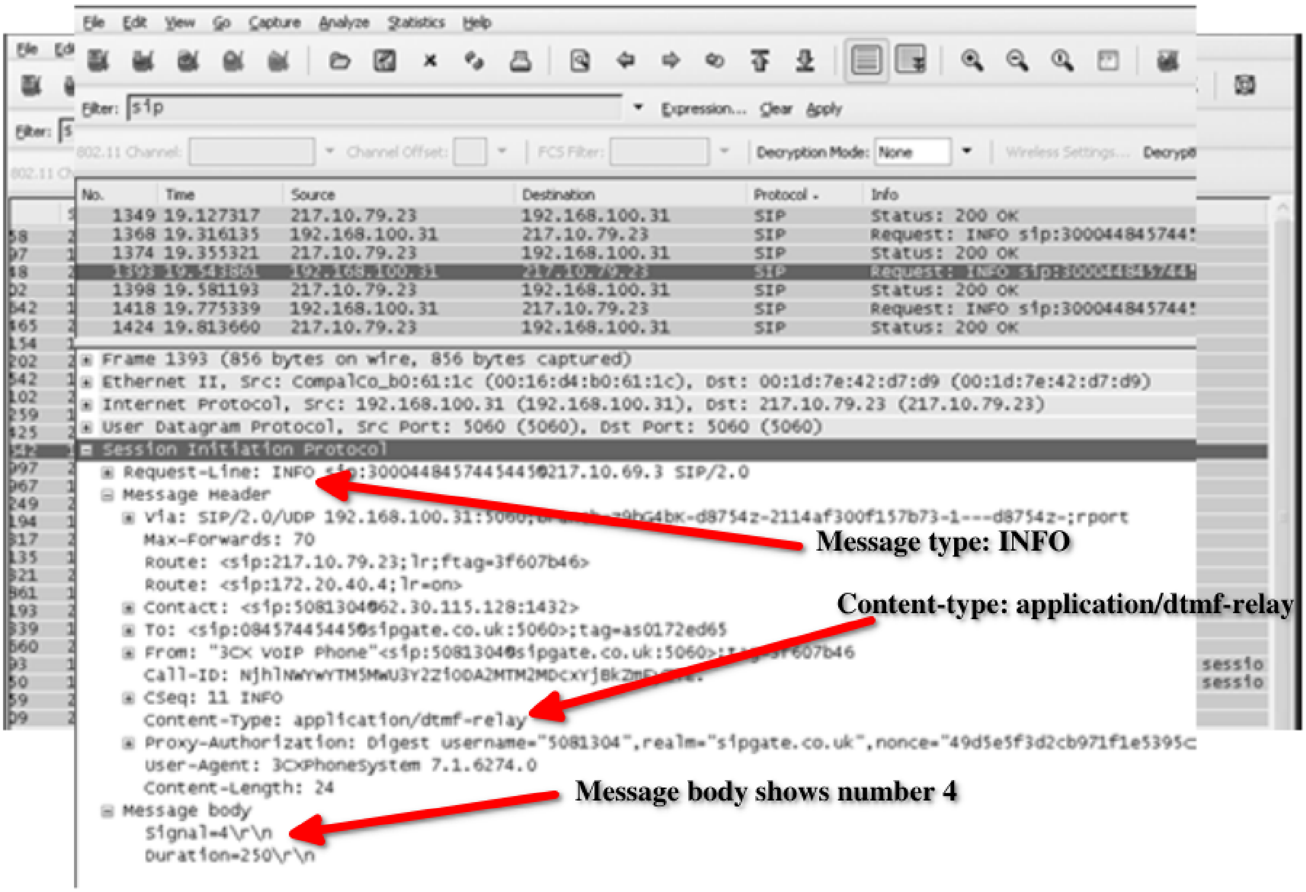
- Inband: DTMF is transmitted in the same RTP stream as the meedia is, thus some tones will be heard by parties in a conversation, it's important to understand that compression codecs such as g.729, g.723 etc, may make the tones unintelligible so this will only really work when using the G.711 codec or better.

- RFC 2833: Known also as Out of band method that takes DTMF out of the RTP stream and into it's on RTP packet, this means that the DTMF codes can survive ok even if the main stream is compressed, the Out-of-band RTP packets theen hold thee various event codes and the tone is -regenerated by an appropriate gateway or SIP UA.
- RFC 4733: Builds on and supersedes RFC 2833 though its taking a while for manufacturers to support it, it actually requires that devices don't have to support every tone and event there is, just simply advertise what they DO support when setting up a connection, example if the payload format uses the payload type number 100, and the implementation can handle the DTMF tones (events 0 through 15) and the dial and ringing tones (assuming as an example that these were defined as events with codes 66 and 70 respectively) it would include the following description in its SDP message:
- ```
  m=audio 12346 RTP/AVP 100
  a=rtpmap:100 telephone-event/8000
  a=fmtp:100 0-15,66,70
  ```
- RFC 4734: More event codes for modem, fax and text telephony signals that get carried in the RTP payload.
- SIP INFO: is used to carry session control information along the SIP signaling path 'during' an existing session, In the example of a phone call to a bank, the session is established but you may get asked to type in an account number, SIP INFO can carry the digits you type without changing the characteristics of the SIP session, Its worth noting that SIP proxies can see and act upon SIP INFO messages and not DTMF Inband or RFC 2833 packets.

DTMF RFC 2833 example
  ```



SIP INFO DTMF example



Summary

- SIP Networks and the PSTN can work well together as long as SIP gateways perform the correct ISUP to SIP mapping in order for a call to reach its destination
- This also is true for SIP to PSTN calls and PSTN to PSTN bridging across a SIP based internetwork
- Gateways can be complex devicees and it wouln't be uncommon to find a single server running gateway, proxy, location, registration service on it
- TRIP is an emerging protocol that promises to do for SIP what BGP did for i nternet in making it easy for SIP calls to be connected to the most appropriate gateway.

- DTMF tones havee been known to be problematic so support for at least RFC 2833 is recommended for all manufacturers and ITSPs
- SIP tu ISUP commands do map fairly well though it takes time to remember what each of the command does and what the responses s hould be, this do not intend to give you all the details on USUP to SIP mapping, check RFC 3398 for more details

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