

RFC 3666 SIP PSTN Call Flows 2 — SIP to PSTN Dialing

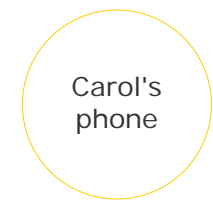
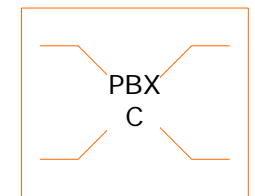
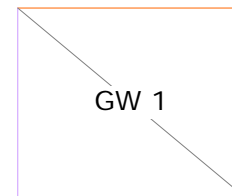
2.2 – Successful SIP to ISDN PBX call

V1.1 – April 29, 2005

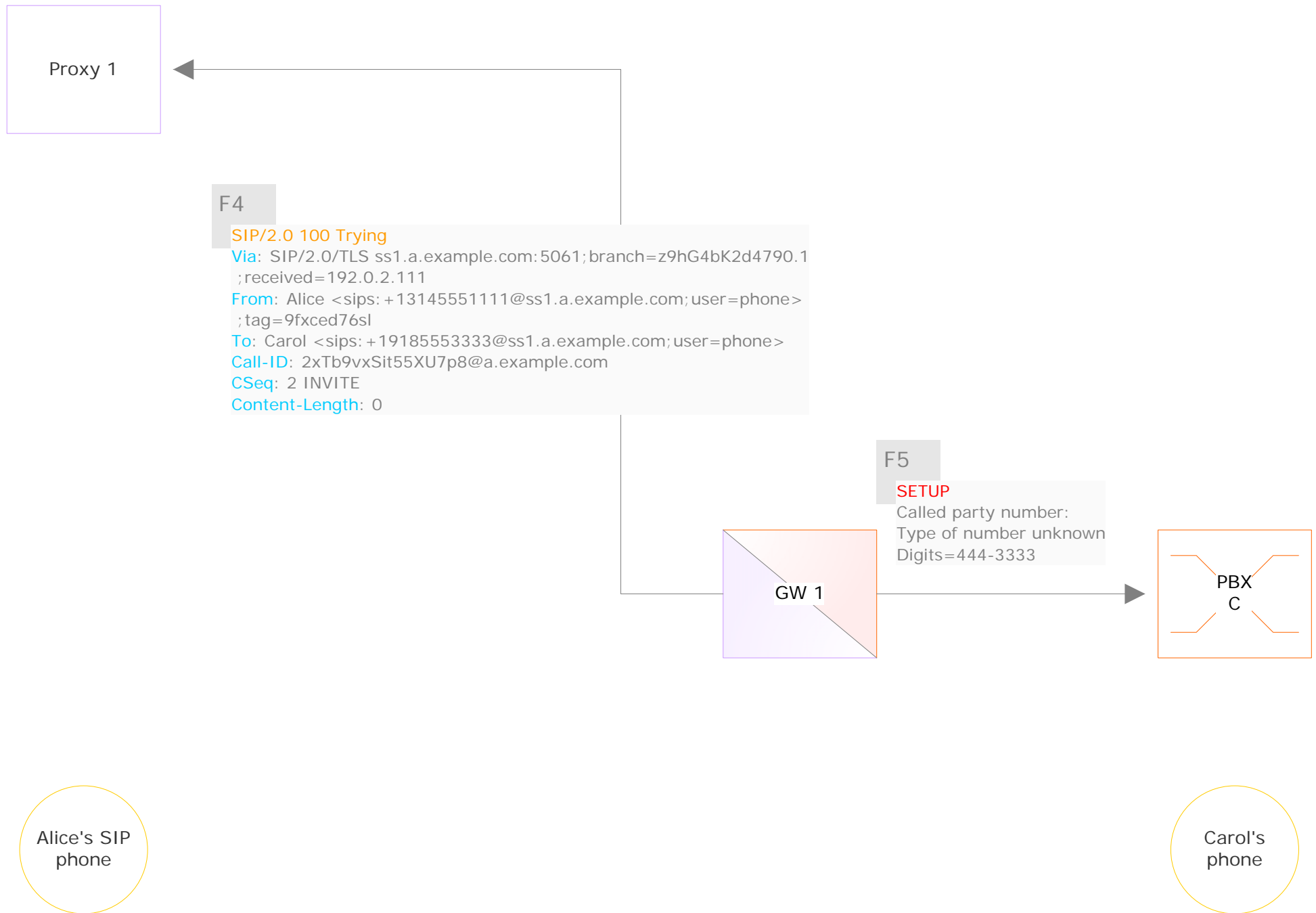


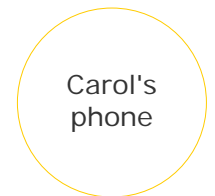
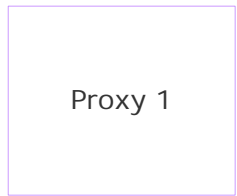
This is a representation, as a slide show, of the SIP examples detailed in RFC 3666 – SIP PSTN Call Flows.

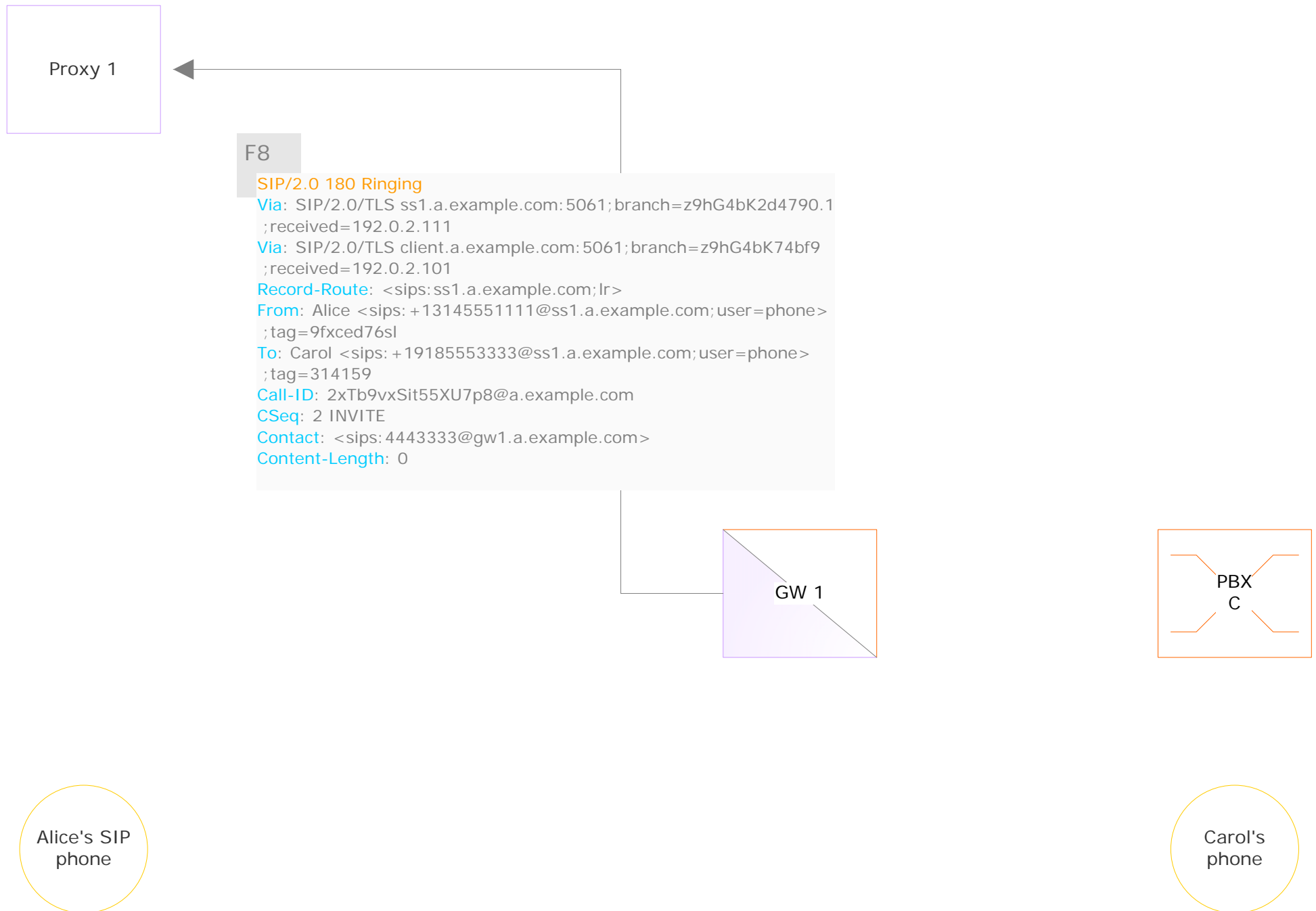
SIP messages are reported in strict conformance with this RFC.

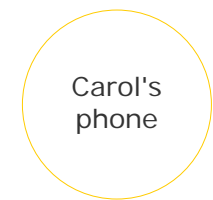
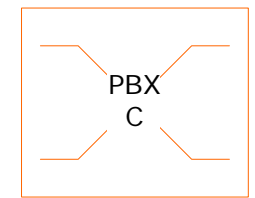
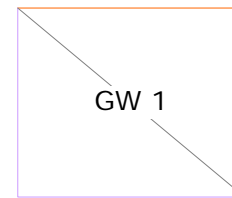


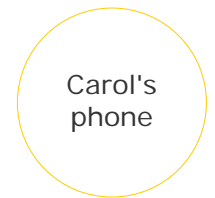
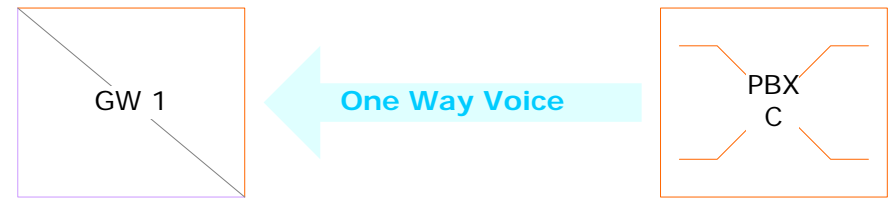
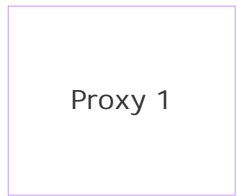


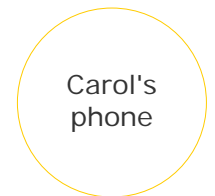
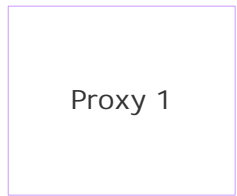




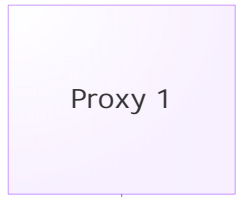












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SIP/2.0 200 OK
Via: SIP/2.0/TLS client.a.example.com:5061;branch=z9hG4bK74bf9
;received=192.0.2.101
Record-Route: <sips:ss1.a.example.com;lr>
From: Alice <sips:+13145551111@ss1.a.example.com;user=phone>
;tag=9fxced76sl
To: Carol <sips:+19185553333@ss1.a.example.com;user=phone>
;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 2 INVITE
Contact: <sips:4443333@gw1.a.example.com>
Content-Type: application/sdp
Content-Length: 144

v=0
o=GW 2890844527 2890844527 IN IP4 gw1.a.example.com
s=-
c=IN IP4 gw1.a.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
    
```

