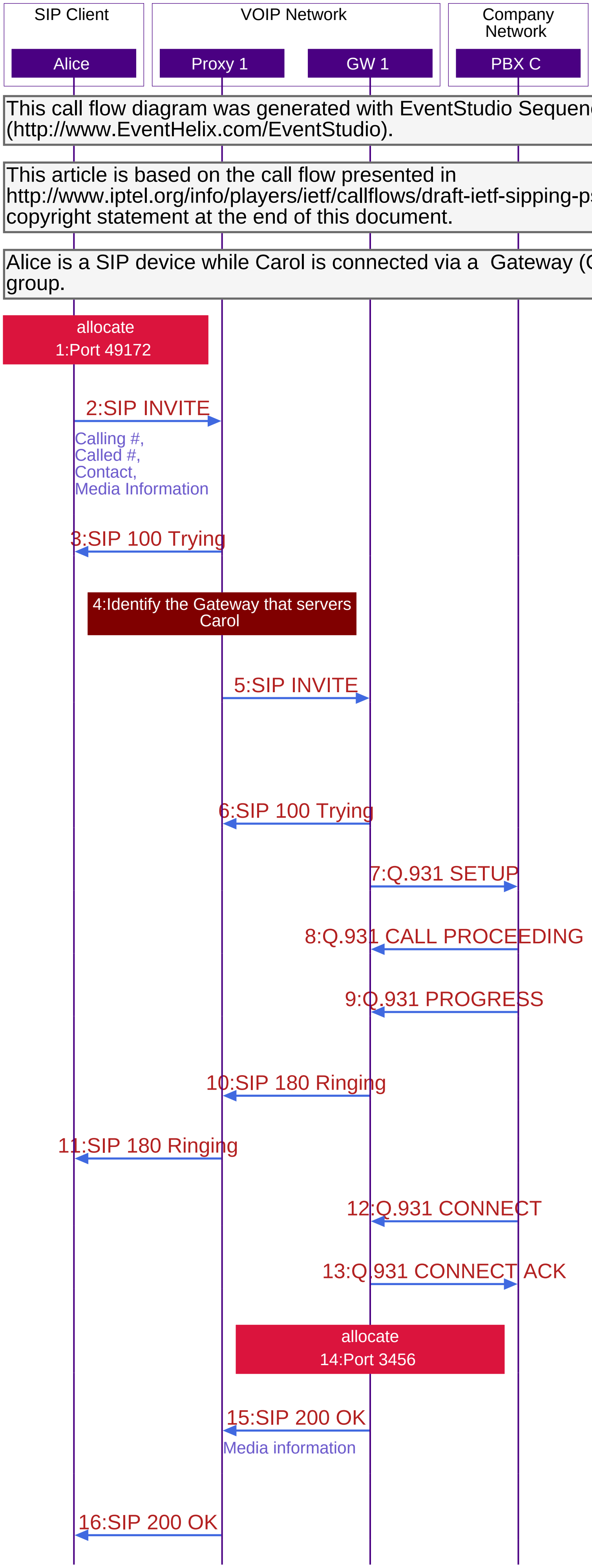


This call flow diagram was generated with EventStudio Sequence Diagram Designer (http://www.EventHelix.com/EventStudio).

This article is based on the call flow presented in http://www.iptel.org/info/players/ietf/callflows/draft-ietf-sipping-pstn-call-flows-02.txt and is reproduced here as per the copyright statement at the end of this document.

Alice is a SIP device while Carol is connected via a Gateway (GW 1) to a PBX. The PBX connection is via a ISDN trunk group.

Alice's PC allocates a port for receiving RTP data. This port number will be included in the SIP Invite.

Alice dials Carol's telephone number (918-555-3333) which is globalized and put into a SIP URI. The message contains information about the RTP port number and the supported voice codecs.

Proxy 1 indicates to the SIP client that it is trying to establish the call.

Proxy 1 looks up the telephone number and locates the gateway that serves Carol. Carol is identified by her extension (444-3333) in the Request-URI sent to GW 1.

The host portion of the Request-URI in the INVITE is used to identify the context (customer, trunk group, or line) in which the private number 444-3333 is valid. Otherwise, this INVITE message could get forwarded by GW 1 and the context of the digits could become lost and the call unroutable.

GW 1 indicates to the Proxy that it is trying to establish the call.

The GW routes the call. Since Carol is served by an ISDN PBX, the Gateway initiates a Q.931 call setup with the PBX.

The ISDN PBX responds with Call Proceeding. This message indicates that the call is in the process of being setup.

The ISDN PBX passes call progress information to the Gateway. This message indicates that the called subscriber is being rung.

The Gateway sends the Ringing indication back to the proxy.

The proxy forwards the ringing indication to Alice's PC.

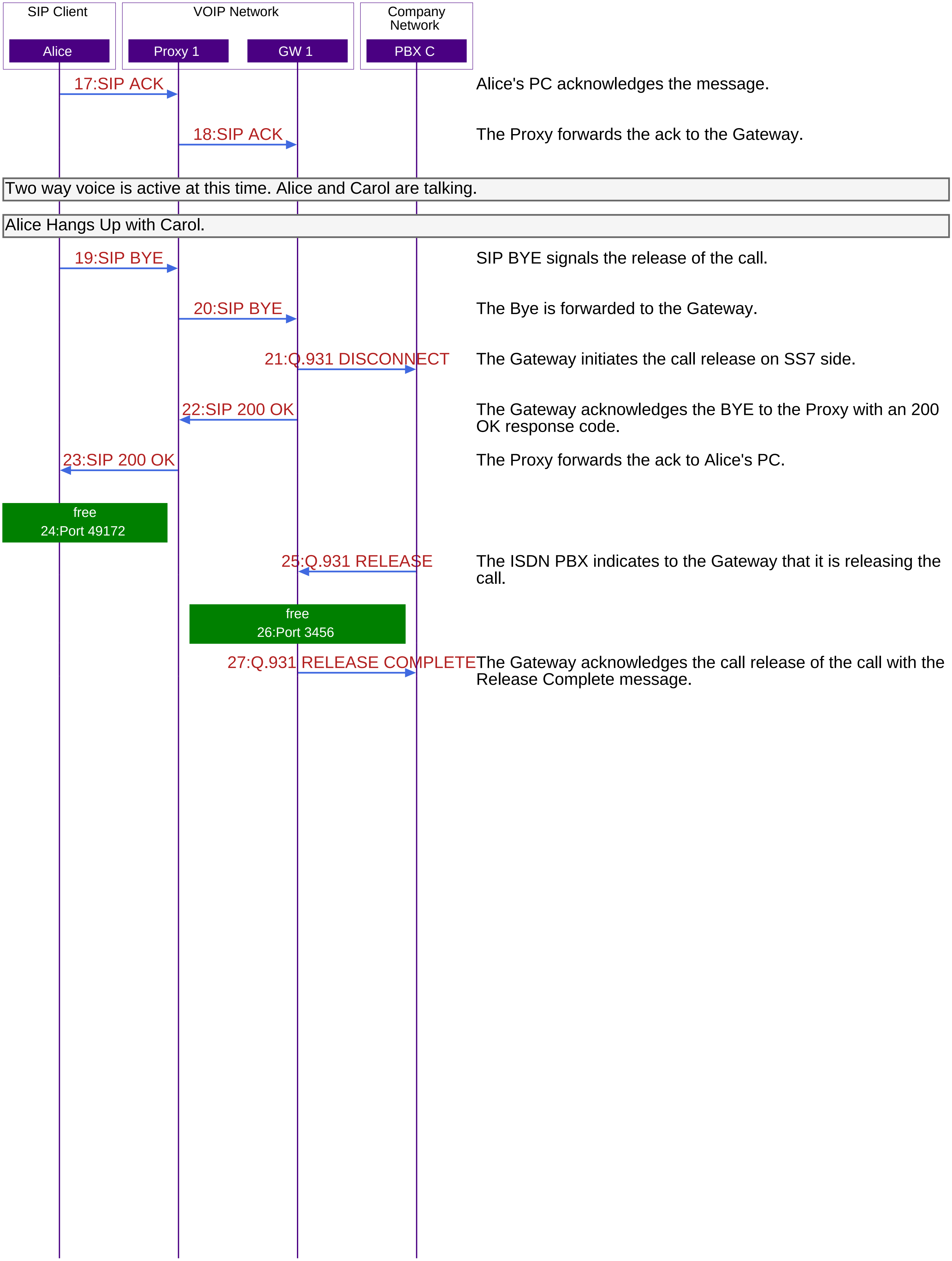
Carol has answered the call. This results in Q.931 CONNECT message being sent to the Gateway.

The Gateway replies with Connect Ack.

The Gateway allocates a port for receiving RTP data from Alice's PC. The port information will be passed to originating subscriber via the "SIP 200 OK" response.

The Gateway indicates to the Proxy that the call is successful. The RTP audio receive port information is also passed in this message.

The Proxy forwards the message to Alice's PC.



Alice's PC acknowledges the message.

The Proxy forwards the ack to the Gateway.

Two way voice is active at this time. Alice and Carol are talking.

Alice Hangs Up with Carol.

SIP BYE signals the release of the call.

The Bye is forwarded to the Gateway.

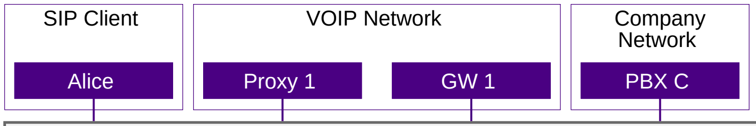
The Gateway initiates the call release on SS7 side.

The Gateway acknowledges the BYE to the Proxy with an 200 OK response code.

The Proxy forwards the ack to Alice's PC.

The ISDN PBX indicates to the Gateway that it is releasing the call.

The Gateway acknowledges the call release of the call with the Release Complete message.



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Alice is a SIP device while Carol is connected via a Gateway (GW 1) to a PBX. The PBX connection is via a ISDN trunk group.

Alice's PC allocates a port for receiving RTP data. This port number will be included in the SIP Invite.

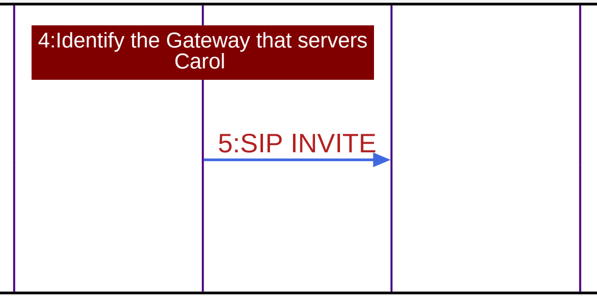
Alice dials Carol's telephone number (918-555-3333) which is globalized and put into a SIP URI. The message contains information about the RTP port number and the supported voice codecs.

```
INVITE sips:+19185553333@ss1.a.example.com;user=phone  SIP/2.0
Via: SIP/2.0/TLS client.a.example.com:5061;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sips:+13145551111@ss1.a.example.com;user=phone> ;tag=9fxced76sl
To: Carol <sips:+19185553333@ss1.a.example.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 2 INVITE
Contact: <sips:alice@client.a.example.com>
Proxy-Authorization: Digest username="alice",
 realm="a.example.com", nonce="qo0dc3a5ab22aa931904badfa1cf5j9h",
 opaque="", uri="sips:+19185553333@ss1.a.example.com;user=phone",
 response="6c792f5c9fa360358b93c7fb826bf550"
Content-Type: application/sdp
Content-Length: 154

v=0
o=alice 2890844526 2890844526 IN IP4 client.a.example.com
s=-
c=IN IP4 client.a.example.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

Proxy 1 indicates to the SIP client that it is trying to establish the call.

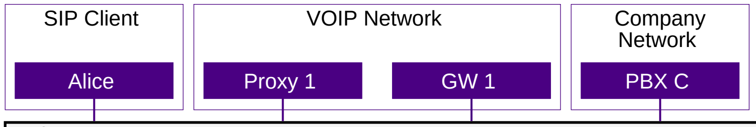
```
SIP/2.0 100 Trying
Via: SIP/2.0/TLS client.a.example.com:5061;branch=z9hG4bK74bf9;received=192.0.2.101
From: Alice <sips:+13145551111@ss1.a.example.com;user=phone>;tag=9fxced76sl
To: Carol <sips:+19185553333@ss1.a.example.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 2 INVITE
Content-Length: 0
```



Proxy 1 looks up the telephone number and locates the gateway that serves Carol. Carol is identified by her extension (444-3333) in the Request-URI sent to GW 1.

The host portion of the Request-URI in the INVITE is used to identify the context (customer, trunk group, or line) in which the private number 444-3333 is valid. Otherwise, this INVITE message could get forwarded by GW 1 and the context of the digits could become lost and the call unroutable.

```
INVITE sips:4443333@gw1.a.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.a.example.com:5061;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TLS client.a.example.com:5061;branch=z9hG4bK74bf9;received=192.0.2.101
Max-Forwards: 69
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>
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 2 INVITE
Contact: <sips:alice@client.a.example.com>
Content-Type: application/sdp
Content-Length: 154
```



```
v=0
o=alice 2890844526 2890844526 IN IP4 client.a.example.com
s=-
c=IN IP4 client.a.example.com
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

GW 1 indicates to the Proxy that it is trying to establish the call.

```
SIP/2.0 100 Trying
Via: SIP/2.0/TLS ss1.a.example.com:5061;branch=z9hG4bK2d4790.1;received=192.0.2.111
From: Alice <sips:+13145551111@ss1.a.example.com;user=phone>;tag=9fxced76sl
To: Carol <sips:+19185553333@ss1.a.example.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 2 INVITE
Content-Length: 0
```

The GW routes the call. Since Carol is served by an ISDN PBX, the Gateway initiates a Q.931 call setup with the PBX.

```
Protocol discriminator=Q.931
Message type=SETUP
Bearer capability: Information transfer capability=0 (Speech) or 16
(3.1 kHz audio)
Channel identification=Preferred or exclusive B-channel
Progress indicator=1 (Call is not end-to-end ISDN;further call
progress information may be available inband)
Called party number:
Type of number unknown
Digits=444-3333
```

The ISDN PBX responds with Call Proceeding. This message indicates that the call is in the process of being setup.

```
Protocol discriminator=Q.931
Message type=CALL PROC
Channel identification=Exclusive B-channel
```

The ISDN PBX passes call progress information to the Gateway. This message indicates that the called subscriber is being rung.

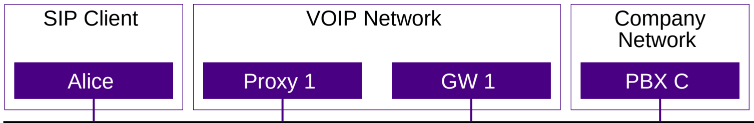
```
Protocol discriminator=Q.931
Message type=PROG
Progress indicator=1 (Call is not end-to-end ISDN;further call
progress information may be available inband)
```

The Gateway sends the Ringing indication back to the proxy.

```
SIP/2.0 180 Ringing
Via: SIP/2.0/TLS ss1.a.example.com:5061;branch=z9hG4bK2d4790.1;received=192.0.2.111
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-Length: 0
```

The proxy forwards the ringing indication to Alice's PC.

```
SIP/2.0 180 Ringing
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>
```



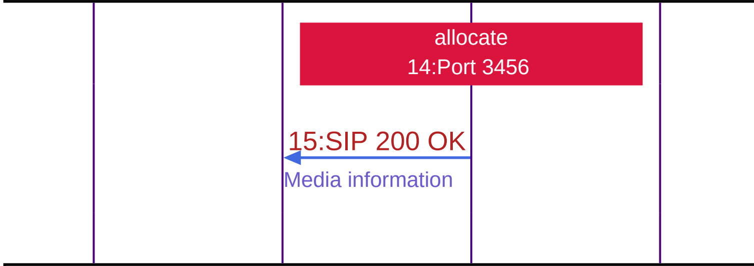
```
Content-Length: 0
```

Carol has answered the call. This results in Q.931 CONNECT message being sent to the Gateway.

```
Protocol discriminator=Q.931
Message type=CONN
```

The Gateway replies with Connect Ack.

```
Protocol discriminator=Q.931
Message type=CONN ACK
```



The Gateway allocates a port for receiving RTP data from Alice's PC. The port information will be passed to originating subscriber via the "SIP 200 OK" response.

The Gateway indicates to the Proxy that the call is successful. The RTP audio receive port information is also passed in this message.

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS ss1.a.example.com:5061;branch=z9hG4bK2d4790.1;received=192.0.2.111
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
```

The Proxy forwards the message to Alice's PC.

```
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
```

Alice's PC acknowledges the message.

```
ACK sips:4443333@gw1.a.example.com SIP/2.0
Via: SIP/2.0/TLS client.a.example.com:5061;branch=z9hG4bK74bf9
Max-Forwards: 70
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 ACK
Content-Length: 0
```

18:SIP ACK

The Proxy forwards the ack to the Gateway.

```
ACK sips:4443333@gw1.a.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.a.example.com:5061;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TLS client.a.example.com:5061;branch=z9hG4bK74bf9;received=192.0.2.101
Max-Forwards: 69
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 ACK
Content-Length: 0
```

Two way voice is active at this time. Alice and Carol are talking.

Alice Hangs Up with Carol.

19:SIP BYE

SIP BYE signals the release of the call.

```
BYE sips:4443333@gw1.a.example.com SIP/2.0
Via: SIP/2.0/TLS client.a.example.com:5061;branch=z9hG4bK74bf9
Max-Forwards: 70
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: 3 BYE
Content-Length: 0
```

20:SIP BYE

The Bye is forwarded to the Gateway.

```
BYE sips:4443333@gw1.a.example.com SIP/2.0
Via: SIP/2.0/TLS ss1.a.example.com:5061;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TLS client.a.example.com:5061;branch=z9hG4bK74bf9;received=192.0.2.101
Max-Forwards: 69
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: 3 BYE
Content-Length: 0
```

21:Q.931 DISCONNECT

The Gateway initiates the call release on SS7 side.

```
Protocol discriminator=Q.931
Message type=DISC
Cause=16 (Normal clearing)
```

22:SIP 200 OK

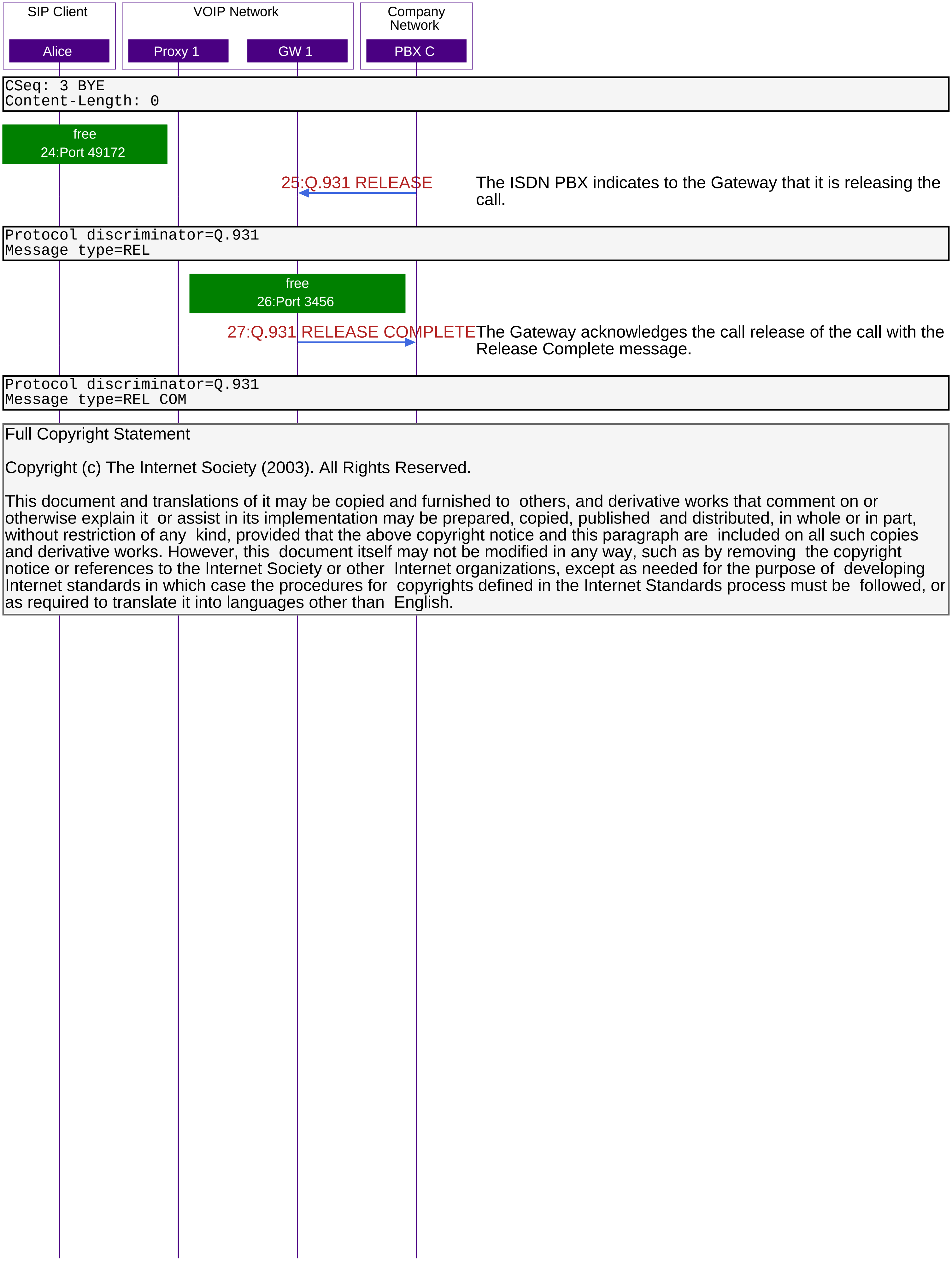
The Gateway acknowledges the BYE to the Proxy with an 200 OK response code.

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS ss1.a.example.com:5061;branch=z9hG4bK2d4790.1;received=192.0.2.111
Via: SIP/2.0/TLS client.a.example.com:5061;branch=z9hG4bK74bf9;received=192.0.2.101
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: 3 BYE
Content-Length: 0
```

23:SIP 200 OK

The Proxy forwards the ack to Alice's PC.

```
SIP/2.0 200 OK
Via: SIP/2.0/TLS client.a.example.com:5061;branch=z9hG4bK74bf9;received=192.0.2.101
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: 3 BYE
Content-Length: 0
```

The ISDN PBX indicates to the Gateway that it is releasing the call.

```
Protocol discriminator=Q.931
Message type=REL
```

The Gateway acknowledges the call release of the call with the Release Complete message.

```
Protocol discriminator=Q.931
Message type=REL COM
```