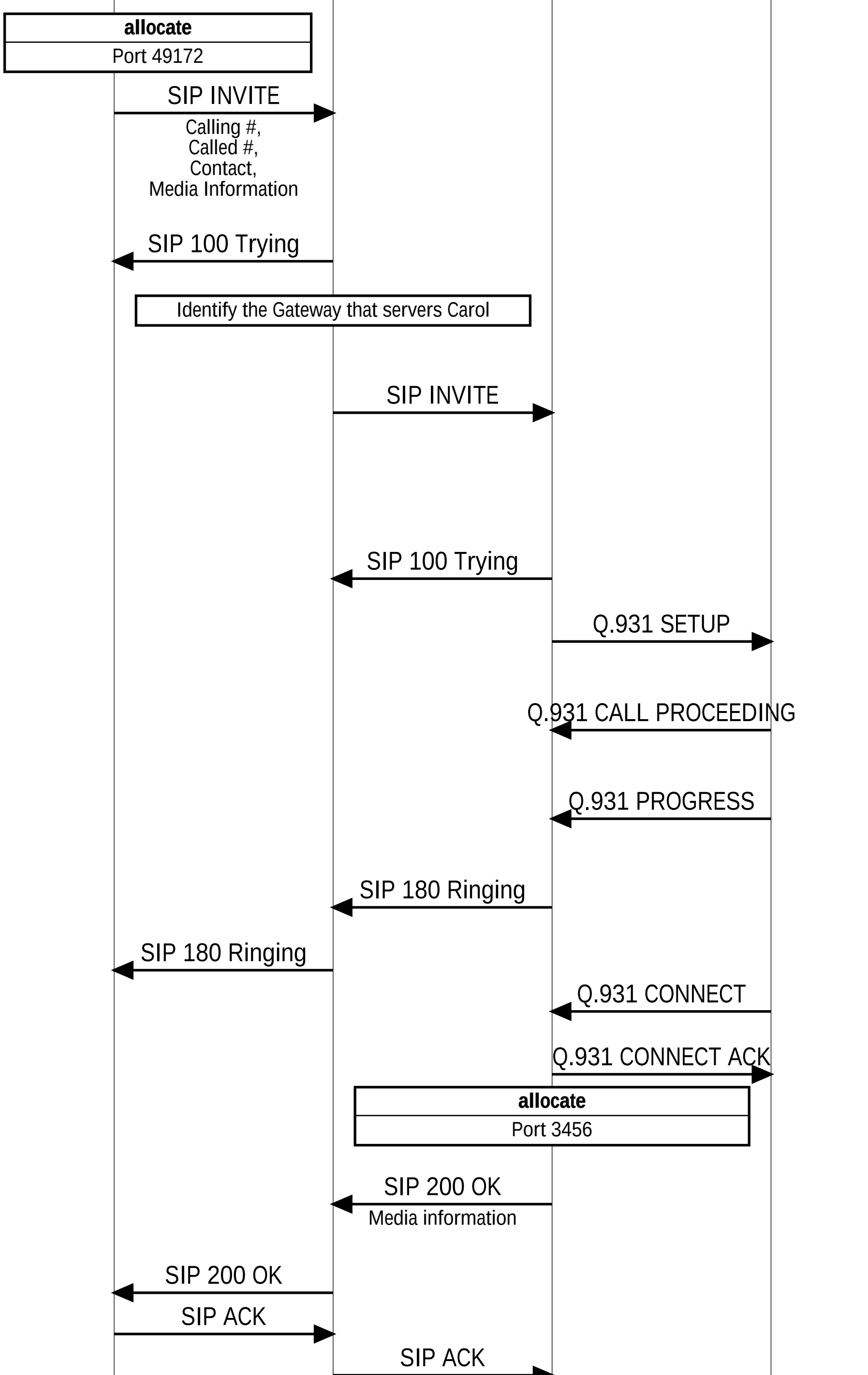
Session Initiation Protocol (SIP Tutorial: SIP to ISDN Q.931 Call Flow (Brief))				
SIP Subscriber	Network			EventHelix.com/EventStudio 2.5
SIP Client	VOIP Network		Company Network	
Alice	Proxy 1	GW 1	PBX C	10-Jun-05 22:14 (Page 1)

```
This call flow diagram was generated with EventStudio Sequence Diagram Designer 2.5 (http://www.EventHelix.com/EventStudio).
```

LEG: Brief

```
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.

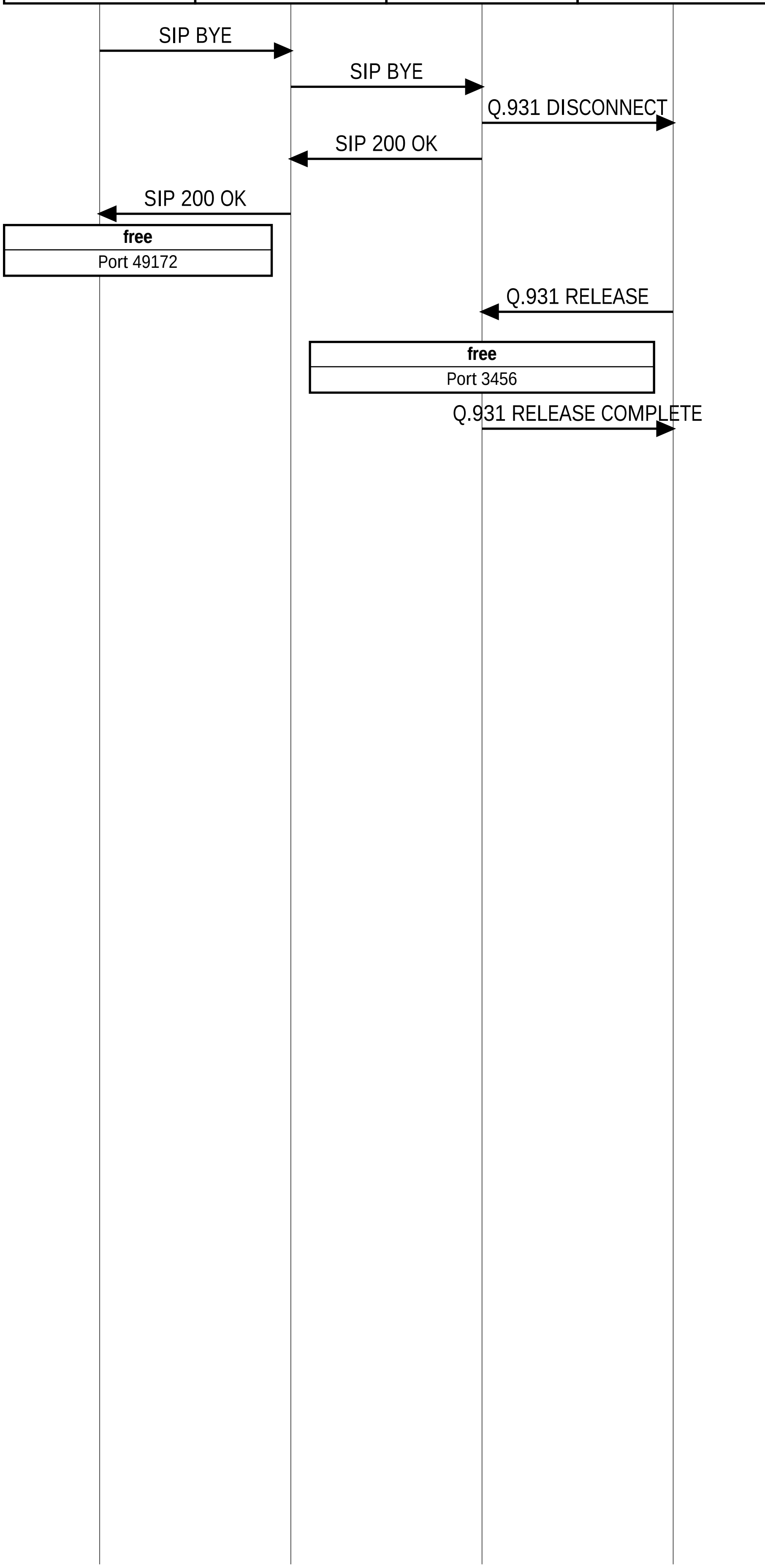
The Proxy forwards the ack to the Gateway.

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

```
Alice Hangs Up with Carol.
```

Session Initiation Protocol (SIP Tutorial: SIP to ISDN Q.931 Call Flow (Brief))

SIP Subscriber	Network			**EventHelix.com/EventStudio 2.5**
SIP Client	VOIP Network		Company Network	
Alice	Proxy 1	GW 1	PBX C	10-Jun-05 22:14 (Page 2)



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 respponse 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.

Session Initiation Protocol (SIP Tutorial: SIP to ISDN Q.931 Call Flow (Detailed))

SIP Subscriber	Network			EventHelix.com/EventStudio 2.5
SIP Client	VOIP Network		Company Network	
Alice	Proxy 1	GW 1	PBX C	10-Jun-05 22:14 (Page 3)

`This call flow diagram was generated with EventStudio Sequence Diagram Designer 2.5 (http://www.EventHelix.com/EventStudio).`

LEG: Detailed

```
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.
```

allocate
Port 49172

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

SIP INVITE (Alice → Proxy 1)
Calling #,
Called #,
Contact,
Media Information

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
```

SIP 100 Trying (Proxy 1 → Alice)

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
```

Identify the Gateway that servers Carol

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.

SIP INVITE (Proxy 1 → 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
```

SIP 100 Trying (GW 1 → Proxy 1)

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

Session Initiation Protocol (SIP Tutorial: SIP to ISDN Q.931 Call Flow (Detailed))

SIP Subscriber	Network			EventHelix.com/EventStudio 2.5
SIP Client	VOIP Network		Company Network	10-Jun-05 22:14 (Page 4)
Alice	Proxy 1	GW 1	PBX C	

```
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
```

Q.931 SETUP

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
```

Q.931 CALL PROCEEDING

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
```

Q.931 PROGRESS

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)
```

SIP 180 Ringing

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
```

SIP 180 Ringing

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
```

Q.931 CONNECT

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

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

Q.931 CONNECT ACK

The Gateway replies with Connect Ack.

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

allocate

Port 3456

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.

SIP 200 OK

Media information

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
```

SIP 200 OK (Proxy 1 → Alice) — 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
```

SIP ACK (Alice → Proxy 1) — 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
```

SIP ACK (Proxy 1 → GW 1) — 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.

SIP BYE (Alice → Proxy 1) — 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
```

SIP BYE (Proxy 1 → GW 1) — 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
```

Q.931 DISCONNECT (GW 1 → PBX C): The Gateway initiates the call release on SS7 side.

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

SIP 200 OK (GW 1 → Proxy 1): The Gateway acknowledges the BYE to the Proxy with an 200 OK respponse 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
```

SIP 200 OK (Proxy 1 → Alice): 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
```

free
Port 49172

Q.931 RELEASE (PBX C → GW 1): The ISDN PBX indicates to the Gateway that it is releasing the call.

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

free
Port 3456

Q.931 RELEASE COMPLETE (GW 1 → PBX C): The Gateway acknowledges the call release of the call with the Release Complete message.

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