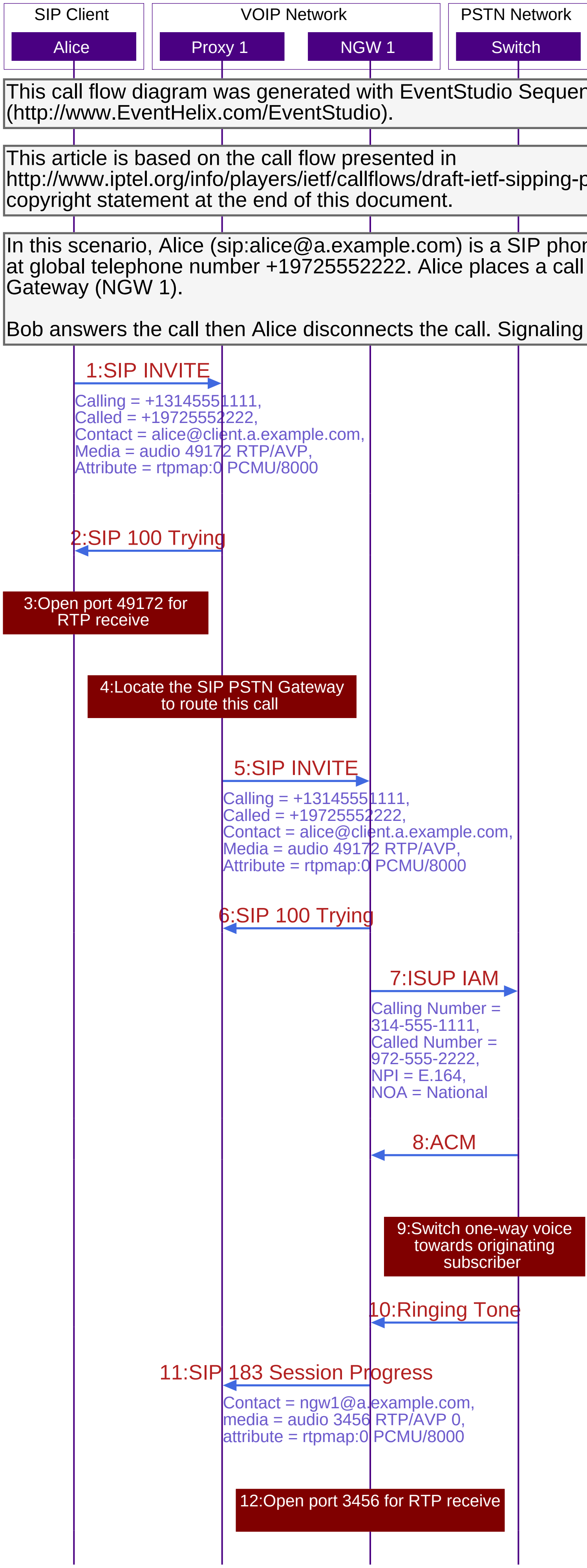
This call flow diagram was generated with EventStudio Sequence Diagram Designer (http://www.EventHelix.com/EventStudio).

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.

In this scenario, Alice (sip:alice@a.example.com) is a SIP phone or other SIP-enabled device. Bob is reachable via the PSTN at global telephone number +19725552222. Alice places a call to Bob through a Proxy Server (Proxy 1) and a Network Gateway (NGW 1).

Bob answers the call then Alice disconnects the call. Signaling between NGW 1 and Bob's telephone switch is ANSI ISUP.



Alice dials the globalized E.164 number +19725552222 to reach Bob. Note that A might have only dialed the last 7 digits, or some other dialing plan. It is assumed that the SIP User Agent Client converts the digits into a global number and puts them into a SIP URI. Note that tel URIs could be used instead of SIP URIs.

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

Client for A prepares to receive data on port 49172 from the network.

Proxy 1 uses a Location Service function to determine the gateway for terminating this call. The location service returns NGW 1 as the gateway.

The call is forwarded to the PSTN Network Gateway NGW 1. The SIP INVITE message is sent to NGW 1.

NGW 1 indicates to the proxy that it is trying to establish the call.

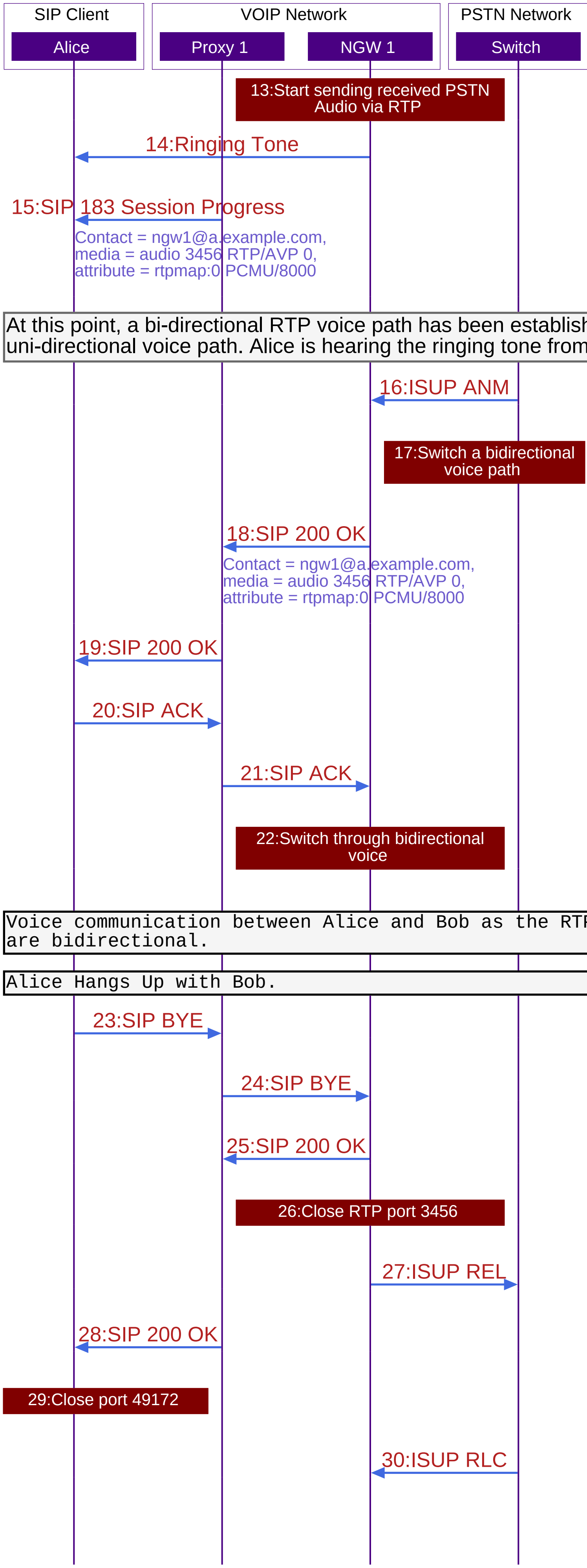
SIP PSTN Gateway sends ISUP Initial Address Message to the PSTN. The calling and the called numbers are included in the message.

Since all the digits were included in the ISUP IAM, the switch replies with ISUP Address Complete message. This signals that the switch has received all the digits and is processing the call.

The switch connects a one way voice path towards Alice. This is needed to send the ringing tone.

The switch sends the ringing tone to the subscriber.

A SIP Session in Progess message is generated when the NGW 1 receives the ACM message from the switch. The "Session Progress" message also contains RTP media information for the RTP port allocated for this call.



SIP PSTN Gateway sends PSTN audio (ringing) in the RTP path to A

At this point, a bi-directional RTP voice path has been established between Alice and NGW 1. The Switch to NGW 1 is a uni-directional voice path. Alice is hearing the ringing tone from the switch.

Bob answers the call. The switch sends an ISUP Answer message to NGW 1.

A bi-directional voice path is switched between the calling and the called subscriber.

NGW 1 responds with SIP OK message to Proxy 1 to indicate that the subscriber has answered the call.

Proxy 1 sends SIP OK message to Alice's SIP client.

The SIP client acknowledges the receipt of SIP OK message.

Proxy 1 acknowledges the receipt of SIP OK message.

At this point, NGW 1 bridges the bidirectional RTP path and the bidirectional SIP path.

`Voice communication between Alice and Bob as the RTP (Alice<->NGW 1) and PSTN (NGW 1<->Switch) paths are bidirectional.`

`Alice Hangs Up with Bob.`

The SIP client sends a BYE message to Proxy 1

Proxy 1 sends a BYE indication to the Network Gateway.

NGW 1 replies back with success, acknowledging the receipt of call release request.

NGW 1 releases the RTP port that was being used for communication with Alice's SIP client.

NGW 1 signals the call release to the switch via an ISUP Release message.

In the meanwhile, Proxy 1 acknowledges the BYE message to the SIP client.

Voice communication is over, so the RTP port can be closed.

The Switch releases the voice call and replies with ISUP Release Complete message.

SIP Client	VOIP Network		PSTN Network
Alice	Proxy 1	NGW 1	Switch

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

In this scenario, Alice (sip:alice@a.example.com) is a SIP phone or other SIP-enabled device. Bob is reachable via the PSTN at global telephone number +19725552222. Alice places a call to Bob through a Proxy Server (Proxy 1) and a Network Gateway (NGW 1).

Bob answers the call then Alice disconnects the call. Signaling between NGW 1 and Bob's telephone switch is ANSI ISUP.

1:SIP INVITE (Alice → Proxy 1)

Calling = +13145551111,
Called = +19725552222,
Contact = alice@client.a.example.com,
Media = audio 49172 RTP/AVP,
Attribute = rtpmap:0 PCMU/8000

Alice dials the globalized E.164 number +19725552222 to reach Bob. Note that A might have only dialed the last 7 digits, or some other dialing plan. It is assumed that the SIP User Agent Client converts the digits into a global number and puts them into a SIP URI. Note that tel URIs could be used instead of SIP URIs.

Alice could use either their SIP address (sip:alice@a.example.com) or SIP telephone number (sip:+13145551111@ss1.a.example.com;user=phone) in the From header. In this example, the telephone number is included, and it is shown as being passed as calling party identification through the Network Gateway (NGW 1) to Bob. Note that for this number to be passed into the SS7 network, it would have to be somehow verified for accuracy.

```
INVITE sip:+19725552222@ss1.a.example.com;user=phone SIP/2.0
Via: SIP/2.0/TCP client.a.example.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 1 INVITE
Contact: <sip:alice@client.a.example.com;transport=tcp>
Proxy-Authorization: Digest username="alice", realm="a.example.com",
 nonce="dc3a5ab25302aa931904ba7d88fa1cf5", opaque="",
 uri="sip:+19725552222@ss1.a.example.com;user=phone",
 response="ccdca50cb091d587421457305d097458c"
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
```

2: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/TCP client.a.example.com:5060;branch=z9hG4bK74bf9;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 1 INVITE
Content-Length: 0
```

3:Open port 49172 for RTP receive

Client for A prepares to receive data on port 49172 from the network.

4:Locate the SIP PSTN Gateway to route this call

Proxy 1 uses a Location Service function to determine the gateway for terminating this call. The location service returns NGW 1 as the gateway.

SIP Client	VOIP Network		PSTN Network
Alice	Proxy 1	NGW 1	Switch

5:SIP INVITE

Calling = +13145551111,
Called = +19725552222,
Contact = alice@client.a.example.com,
Media = audio 49172 RTP/AVP,
Attribute = rtpmap:0 PCMU/8000

The call is forwarded to the PSTN Network Gateway NGW 1. The SIP INVITE message is sent to NGW 1.

```
INVITE sip:+19725552222@ngw1.a.example.com;user=phone SIP/2.0
Via: SIP/2.0/TCP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TCP client.a.example.com:5060;branch=z9hG4bK74bf9;received=192.0.2.101
Max-Forwards: 69
Record-Route: <sip:ss1.a.example.com;lr>
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 1 INVITE
Contact: <sip:alice@client.a.example.com;transport=tcp>
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
```

6:SIP 100 Trying

NGW 1 indicates to the proxy that it is trying to establish the call.

```
SIP/2.0 100 Trying
Via: SIP/2.0/TCP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1;received=192.0.2.111
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 1 INVITE
Content-Length: 0
```

7:ISUP IAM

Calling Number =
314-555-1111,
Called Number =
972-555-2222,
NPI = E.164,
NOA = National

SIP PSTN Gateway sends ISUP Initial Address Message to the PSTN. The calling and the called numbers are included in the message.

8:ACM

Since all the digits were included in the ISUP IAM, the switch replies with ISUP Address Complete message. This signals that the switch has received all the digits and is processing the call.

9:Switch one-way voice towards originating subscriber

The switch connects a one way voice path towards Alice. This is needed to send the ringing tone.

10:Ringing Tone

The switch sends the ringing tone to the subscriber.

11:SIP 183 Session Progress

Contact = ngw1@a.example.com,
media = audio 3456 RTP/AVP 0,
attribute = rtpmap:0 PCMU/8000

A SIP Session in Progess message is generated when the NGW 1 receives the ACM message from the switch. The "Session Progress" message also contains RTP media information for the RTP port allocated for this call.

Notice that the Contact returned by NGW 1 in this and following messages is sip:ngw1@a.example.com. This is because NGW 1 only accepts SIP messages that come through Proxy 1 - any direct signaling will be ignored. Since this Contact URI may be used outside of this dialog and must be routable (Section 8.1.1.8 in RFC 3261 [2]) the Contact URI for NGW 1 must resolve to Proxy 1. This Contact URI is an AOR which resolves via DNS to Proxy 1 (sip:ss1.a.example.com) which then resolves it to sip:ngw1.a.example.com which is the address of

SIP Client	VOIP Network		PSTN Network
Alice	Proxy 1	NGW 1	Switch

NGW 1.

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/TCP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1;received=192.0.2.111
Via: SIP/2.0/TCP client.a.example.com:5060;branch=z9hG4bK74bf9;received=192.0.2.101
Record-Route: <sip:ss1.a.example.com;lr>
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone> ;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 1 INVITE
Contact: <sip:ngw1@a.example.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 146

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.a.example.com
s=-
c=IN IP4 ngw1.a.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

12:Open port 3456 for RTP receive

13:Start sending received PSTN Audio via RTP

14:Ringing Tone

SIP PSTN Gateway sends PSTN audio (ringing) in the RTP path to A

15:SIP 183 Session Progress

Contact = ngw1@a.example.com,
media = audio 3456 RTP/AVP 0,
attribute = rtpmap:0 PCMU/8000

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/TCP client.a.example.com:5060;branch=z9hG4bK74bf9;received=192.0.2.101
Record-Route: <sip:ss1.a.example.com;lr>
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 1 INVITE
Contact: <sip:ngw1@a.example.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 146

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.a.example.com
s=-
c=IN IP4 ngw1.a.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

At this point, a bi-directional RTP voice path has been established between Alice and NGW 1. The Switch to NGW 1 is a uni-directional voice path. Alice is hearing the ringing tone from the switch.

16:ISUP ANM

Bob answers the call. The switch sends an ISUP Answer message to NGW 1.

17:Switch a bidirectional voice path

A bi-directional voice path is switched between the calling and the called subscriber.

18:SIP 200 OK

Contact = ngw1@a.example.com,
media = audio 3456 RTP/AVP 0,
attribute = rtpmap:0 PCMU/8000

NGW 1 responds with SIP OK message to Proxy 1 to indicate that the subscriber has answered the call.

```
SIP/2.0 200 OK
Via: SIP/2.0/TCP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1;received=192.0.2.111
Via: SIP/2.0/TCP client.a.example.com:5060;branch=z9hG4bK74bf9;received=192.0.2.101
Record-Route: <sip:ss1.a.example.com;lr>
```

SIP Client	VOIP Network		PSTN Network
Alice	Proxy 1	NGW 1	Switch

```
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 1 INVITE
Contact: <sip:ngw1@a.example.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 146

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.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
```

19:SIP 200 OK (Proxy 1 → Alice)

Proxy 1 sends SIP OK message to Alice's SIP client.

```
SIP/2.0 200 OK
Via: SIP/2.0/TCP client.a.example.com:5060;branch=z9hG4bK74bf9;received=192.0.2.101
Record-Route: <sip:ss1.a.example.com;lr>
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 1 INVITE
Contact: <sip:ngw1@a.example.com;transport=tcp>
Content-Type: application/sdp
Content-Length: 146

v=0
o=GW 2890844527 2890844527 IN IP4 ngw1.a.example.com
s=-
c=IN IP4 ngw1.a.example.com
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

20:SIP ACK (Alice → Proxy 1)

The SIP client acknowledges the receipt of SIP OK message.

```
ACK sip:ngw1@a.example.com SIP/2.0
Via: SIP/2.0/TCP client.a.example.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
Route: <sip:ss1.a.example.com;lr>
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 1 ACK
Content-Length: 0
```

21:SIP ACK (Proxy 1 → NGW 1)

Proxy 1 acknowledges the receipt of SIP OK message.

```
ACK sip:ngw1@a.example.com SIP/2.0
Via: SIP/2.0/TCP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TCP client.a.example.com:5060;branch=z9hG4bK74bf9;received=192.0.2.101
Max-Forwards: 69
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 1 ACK
Content-Length: 0
```

22:Switch through bidirectional voice

At this point, NGW 1 bridges the bidirectional RTP path and the bidirectional SIP path.

```
Voice communication between Alice and Bob as the RTP (Alice<->NGW 1) and PSTN (NGW 1<->Switch) paths
are bidirectional.
```

```
Alice Hangs Up with Bob.
```

23:SIP BYE (Alice → Proxy 1)

The SIP client sends a BYE message to Proxy 1

SIP Client	VOIP Network		PSTN Network
Alice	Proxy 1	NGW 1	Switch

```
BYE sip:ngw1@a.example.com SIP/2.0
Via: SIP/2.0/TCP client.a.example.com:5060;branch=z9hG4bK74bf9
Max-Forwards: 70
Route: <sip:ss1.a.example.com;lr>
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 2 BYE
Content-Length: 0
```

24:SIP BYE

Proxy 1 sends a BYE indication to the Network Gateway.

```
BYE sip:ngw1@a.example.com SIP/2.0
Via: SIP/2.0/TCP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1
Via: SIP/2.0/TCP client.a.example.com:5060;branch=z9hG4bK74bf9;received=192.0.2.101
Max-Forwards: 69
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 2 BYE
Content-Length: 0
```

25:SIP 200 OK

NGW 1 replies back with success, acknowledging the receipt of call release request.

```
SIP/2.0 200 OK
Via: SIP/2.0/TCP ss1.a.example.com:5060;branch=z9hG4bK2d4790.1;received=192.0.2.111
Via: SIP/2.0/TCP client.a.example.com:5060;branch=z9hG4bK74bf9;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 2 BYE
Content-Length: 0
```

26:Close RTP port 3456

NGW 1 releases the RTP port that was being used for communication with Alice's SIP client.

27:ISUP REL

NGW 1 signals the call release to the switch via an ISUP Release message.

28:SIP 200 OK

In the meanwhile, Proxy 1 acknowledges the BYE message to the SIP client.

```
SIP/2.0 200 OK
Via: SIP/2.0/TCP client.a.example.com:5060;branch=z9hG4bK74bf9;received=192.0.2.101
From: Alice <sip:+13145551111@ss1.a.example.com;user=phone>;tag=9fxced76sl
To: Bob <sip:+19725552222@ss1.a.example.com;user=phone>;tag=314159
Call-ID: 2xTb9vxSit55XU7p8@a.example.com
CSeq: 2 BYE
Content-Length: 0
```

29:Close port 49172

Voice communication is over, so the RTP port can be closed.

30:ISUP RLC

The Switch releases the voice call and replies with ISUP Release Complete message.