Alice is a SIP device while Carol is connected via a Gateway (GW 1) to a PBX. The PBX connection is via an 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 gateway 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.
LEG: Detailed

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

This call flow diagram was generated with EventStudio Sequence Diagram Designer 2.5 (http://www.EventHelix.com/EventStudio).
Session Initiation Protocol (SIP Tutorial: SIP to ISDN Q.931 Call Flow (Detailed))

<table>
<thead>
<tr>
<th>SIP Subscriber</th>
<th>Network</th>
<th>Company Network</th>
</tr>
</thead>
<tbody>
<tr>
<td>Alice</td>
<td>Proxy 1</td>
<td>GW 1</td>
</tr>
<tr>
<td></td>
<td></td>
<td>PBX C</td>
</tr>
</tbody>
</table>

SIP/2.0 100 Trying
Via: SIP/2.0/TLS ssl.a.example.com:5061;branch=z9hG4bK2d4790.1;received=192.0.2.111
From: Alice <sip:+13145551111@ssl.a.example.com;user=phone>;tag=9fcxed76s1
To: Carol <sip:+19185553333@ssl.a.example.com;user=phone>
Call-ID: 2xTb9vxsit55s5u7p8@a.example.com
CSeq: 2 INVITE
Content-Length: 0

Q.931 SETUP
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.

Q.931 PROGRESS
The ISDN PBX passes call progress information to the Gateway. This message indicates that the called subscriber is being rung.

SIP 180 Ringing
The Gateway sends the Ringing indication back to the proxy.

SIP 180 Ringing
The proxy forwards the ringing indication to Alice’s PC.

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

Q.931 CONNECT ACK
The Gateway replies with Connect Ack.

allocate
Port 3456

SIP 200 OK
Media information

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.
Alice calls Carol.

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: <sip:ss1.a.example.com;lr>
To: Carol <sip:+19185553333@ss1.a.example.com;user=phone>;tag=314159
Call-ID: 2xTb9vxsit55XU7p8@a.example.com
CSeq: 2 INVITE
Contact: <sip:4443333@gw1.a.example.com>
Content-Type: application/sdp
Content-Length: 144

v=0
o=- 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 200 OK
Via: SIP/2.0/TLS client.a.example.com:5061;branch=z9hG4bK2d4790.1;received=192.0.2.111
Via: SIP/2.0/TLS ss1.a.example.com:5061;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=9fxced76s1
To: Carol <sip:+19185553333@ss1.a.example.com;user=phone>;tag=314159
Call-ID: 2xTb9vxsit55XU7p8@a.example.com
CSeq: 2 INVITE
Contact: <sip: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;received=192.0.2.101
Max-Forwards: 70
Route: <sip:ss1.a.example.com;lr>
From: Alice <sip:+13145551111:ss1.a.example.com;user=phone>;tag=9fxced76s1
To: Carol <sip:+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
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: 70
Route: <sip:ss1.a.example.com;lr>
From: Alice <sip:+13145551111:ss1.a.example.com;user=phone>;tag=9fxced76s1
To: Carol <sip:+19185553333@ss1.a.example.com;user=phone>;tag=314159
Call-ID: 2xTb9vxsit55XU7p8@a.example.com
CSeq: 3 BYE
Content-Length: 0

SIP BYE signals the release of the call.

The Bye is forwarded to the Gateway.
Q.931 DISCONNECT

The Gateway initiates the call release on SS7 side.

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 <sip:+13145551111@ss1.a.example.com;user=phone>;tag=9fxced76sl
To: Carol <sip:+19185553333@ss1.a.example.com;user=phone>;tag=314159
Call-ID: 2xTb9vxSis55UX7p8@a.example.com
CSeq: 3 BYE
Content-Length: 0

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 <sip:+13145551111@ss1.a.example.com;user=phone>;tag=9fxced76sl
To: Carol <sip:+19185553333@ss1.a.example.com;user=phone>;tag=314159
Call-ID: 2xTb9vxSis55UX7p8@a.example.com
CSeq: 3 BYE
Content-Length: 0

Q.931 RELEASE

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

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

Protocol discriminator=Q.931
Message type=REL COM