Session Initiation Protocol (SIP Tutorial: SIP to PSTN Call Flow)				
SIP Subscriber	Network			**EventHelix.com/EventStudio 2.5**
SIP Client	VOIP Network		PSTN Network	10-Jun-05 22:33 (Page 1)
Alice	Proxy 1	NGW 1	Switch	

```
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.
```

```
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.
```

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.

SIP 100 Trying (Proxy 1 → Alice)

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

[Alice] Open port 49172 for RTP receive

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

[Proxy 1] 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 INVITE (Proxy 1 → NGW 1)
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.

SIP 100 Trying (NGW 1 → Proxy 1)

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

ISUP IAM (NGW 1 → Switch)
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.

ACM (Switch → NGW 1)

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.

[Switch] 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.

Ringing Tone (Switch → NGW 1)

The switch sends the ringing tone to the subscriber.

SIP 183 Session Progress (NGW 1 → Proxy 1)
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.

[NGW 1] Open port 3456 for RTP receive

[NGW 1] Start sending received PSTN Audio via RTP

Ringing Tone (NGW 1 → Alice)

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

SIP Subscriber	Network			EventHelix.com/EventStudio 2.5
SIP Client	VOIP Network		PSTN Network	10-Jun-05 22:33 (Page 2)
Alice	Proxy 1	NGW 1	Switch	

SIP 183 Session Progress (Proxy 1 → Alice)
Contact = ngw1@a.example.com, media = audio 3456 RTP/AVP 0, attribute = 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.`

ISUP ANM (Switch → NGW 1) — Bob answers the call. The switch sends an ISUP Answer message to NGW 1.

Switch a bidirectional voice path — A bi-directional voice path is switched between the calling and the called subscriber.

SIP 200 OK (NGW 1 → Proxy 1) — NGW 1 responds with SIP OK message to Proxy 1 to indicate that the subscriber has answered the call.
Contact = ngw1@a.example.com, media = audio 3456 RTP/AVP 0, attribute = rtpmap:0 PCMU/8000

SIP 200 OK (Proxy 1 → Alice) — Proxy 1 sends SIP OK message to Alice's SIP client.

SIP ACK (Alice → Proxy 1) — The SIP client acknowledges the receipt of SIP OK message.

SIP ACK (Proxy 1 → NGW 1) — Proxy 1 acknowledges the receipt of SIP OK message.

Switch through bidirectional voice — At this point, NGW 1 bridges the bidirectional RTP path and the bidirectional SIP path.

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

`Alice Hangs Up with Bob.`

SIP BYE (Alice → Proxy 1) — The SIP client sends a BYE message to Proxy 1

SIP BYE (Proxy 1 → NGW 1) — Proxy 1 sends a BYE indicatation to the Network Gateway.

SIP 200 OK (NGW 1 → Proxy 1) — NGW 1 replies back with success, acknowledging the receipt of call release request.

Close RTP port 3456 — NGW 1 releases the RTP port that was being used for communciation with Alice's SIP client.

ISUP REL (NGW 1 → Switch) — NGW 1 signals the call release to the switch via an ISUP Release message.

SIP 200 OK (Proxy 1 → Alice) — In the meanwhile, Proxy 1 acknowleges the BYE message to the SIP client.

Close port 49172 — Voice communciation is over, so the RTP port can be closed.

ISUP RLC (Switch → NGW 1) — The Switch releases the voice call and replies with ISUP Release Complete message.

LEG: Detailed

```
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.
```

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
```

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
```

[Alice] Open port 49172 for RTP receive

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

[Proxy 1] 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 INVITE (Proxy 1 → NGW 1)
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
```

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
```

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.

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.

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.

Ringing Tone

The switch sends the ringing tone to the subscriber.

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 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
```

Open port 3456 for RTP receive

Start sending received PSTN Audio via RTP

Ringing Tone

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

SIP 183 Session Progress

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

Session Initiation Protocol (SIP Tutorial: SIP to PSTN Call Flow (Detailed))

SIP Subscriber	Network			EventHelix.com/EventStudio 2.5
SIP Client	VOIP Network		PSTN Network	10-Jun-05 22:33 (Page 5)
Alice	Proxy 1	NGW 1	Switch	

```
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.

ISUP ANM (Switch → NGW 1)

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

Switch a bidirectional voice path (NGW 1)

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

SIP 200 OK (NGW 1 → Proxy 1)

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>
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
```

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
```

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
```

Session Initiation Protocol (SIP Tutorial: SIP to PSTN Call Flow (Detailed))

SIP Subscriber	Network			EventHelix.com/EventStudio 2.5
SIP Client	VOIP Network		PSTN Network	10-Jun-05 22:33 (Page 6)
Alice	Proxy 1	NGW 1	Switch	

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
```

Switch through bidirectional voice

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

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

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

SIP BYE (Alice → Proxy 1)

The SIP client sends a BYE message to Proxy 1

```
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
```

SIP BYE (Proxy 1 → NGW 1)

Proxy 1 sends a BYE indicatation 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
```

SIP 200 OK (NGW 1 → Proxy 1)

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
```

Close RTP port 3456

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

ISUP REL (NGW 1 → Switch)

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

SIP 200 OK (Proxy 1 → Alice)

In the meanwhile, Proxy 1 acknowleges 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
```

Close port 49172

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

ISUP RLC (Switch → NGW 1)

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