TECHNICAL SUPPORT NOTE
3-Way Call Conferencing with Broadsoft - TA900 Series

Introduction

Three way calls are defined as having one active call and having the ability to add a third party into that same call so that all can participate in the same conversation. This document will only focus on analog stations as many SIP stations have this ability built in and do not rely on the TA900. The TA900 does not have the ability to provide a three way conference call internally and therefore must rely on some external device to conduct the conference. There are two different ways of doing this depending on the Soft switch that is being used. This document will focus on conferencing using the Info Method compatible with the Broadsoft soft switch.

Overview

This configuration requires the use of the info method to signal to the soft switch when events occur. This is often referred to as “dumb mode” because the TA 900 does not handle any features directly but instead signals events to the soft switch. The only soft switch that supports this mode of operation is the Broadsoft. The network diagram is setup as shown below (figure 1). While this document uses a transfer between units as an example, it should be noted that the message and event flow would be the same for any transfer using Broadsoft, including extensions within the same unit.

Station 256-555-9102 will call 256-555-9101 then flashhook and call 256-555-9103. At this point 256-555-9102 will flashhook again and bring all three parties into the conference. All debug information will be taken off of the unit with IP of 10.19.210.6. For simplicity SIP messages with no relevant information in the body have been summarized.

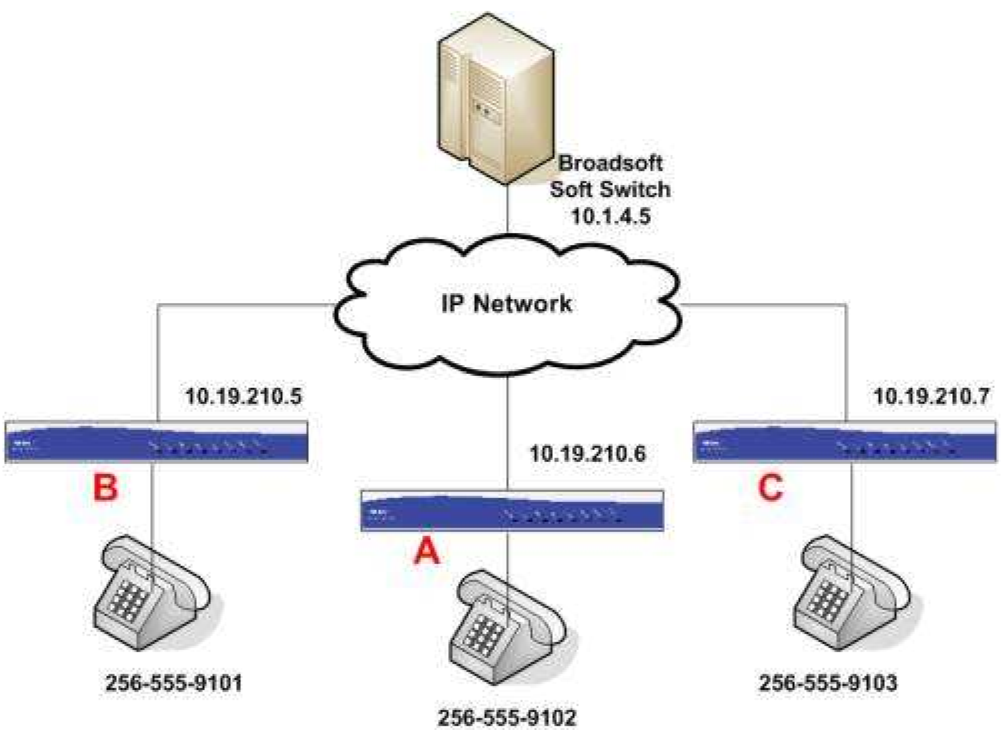

Figure 1. Network Diagram

Configuration

The configuration for this mode of operation is very straight forward. The unit must already have voice users configured and connected to analog stations. The unit must also have a sip trunk defined that is directed to a Broadsoft soft switch. The unit must be in feature mode network. The command “voice flashhook mode transparent” must be entered at global configuration.

```
voice feature-mode network
voice flashhook mode transparent
!
voice codec-list g729g711
  codec g729
  codec g711ulaw
!
voice trunk T01 type sip
  description "Sip Trunk to Broadsoft"
  no reject-external
  sip-server primary 10.1.4.5
  registrar primary 10.1.4.5
  codec-group g729g711
!
voice user 2565559500
  connect fxs 0/1
  password "1234"
  sip-identity 2565559500 T01 register
  codec-group g729g711
!
voice grouped-trunk BROADSOFT
  no description
  trunk T01
  accept $ cost 0
```

Figure 2. Sample voice configuration

Operation

Before any conferencing can be done a call must already exist to the analog station of the TA900 that wishes to initiate the three way call (figure 3). In this example 256-555-9102 calls 256-555-9101. Once this call has been established the user at the analog station 256-555-9102 must flash hook. The TA900 A will receive this flash hook and will then simply relay the flash hook event to the Broadsoft as shown below (figure 4).

```
SIP.STACK MSGSUM  Tx: INVITE sip:2565559101@10.1.4.5:5060 SIP/2.0
SIP.STACK MSGSUM  Rx: SIP/2.0 100 Trying
SIP.STACK MSGSUM  Rx: SIP/2.0 180 Ringing
SIP.STACK MSGSUM  Tx: PRACK sip:10.1.4.5:5060 SIP/2.0
SIP.STACK MSGSUM  Rx: SIP/2.0 200 OK
SIP.STACK MSGSUM  Rx: SIP/2.0 200 OK
SIP.STACK MSGSUM  Tx: ACK sip:10.1.4.5:5060 SIP/2.0
```

Figure 3. Call is placed from 2565559102 (TA900 A) to 2565559101 (TA900 B)

```
SIP.STACK MSG  Tx: UDP src=10.19.210.6:5060 dst=10.1.4.5:5060
SIP.STACK MSG      INFO sip:10.1.4.5:5060 SIP/2.0
                SIP.STACK MSG      From: ""<sip:2565559102@10.1.4.5:5060;transport=UDP>;
SIP.STACK MSG      To: <sip:2565559101@10.1.4.5:5060;transport=UDP>;
SIP.STACK MSG      Call-ID: 33516a0-a13d206-13c4-1726-1009aff5-1726@10.1.4.5
SIP.STACK MSG      CSeq: 3 INFO
SIP.STACK MSG      Via: SIP/2.0/UDP 10.19.210.6:5060;branch=z9hG4bK-1733-5aa26b-5dee1b6e
SIP.STACK MSG      Max-Forwards: 70
SIP.STACK MSG      Supported: 100rel,replaces
SIP.STACK MSG      Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, PRACK, REFER,
                REGISTER
SIP.STACK MSG      User-Agent: ADTRAN_Total_Access_924e/13.01.03.E
SIP.STACK MSG      Contact: <sip:2565559102@10.19.210.6:5060;transport=UDP>
SIP.STACK MSG      Content-Type: application/broadsoft
SIP.STACK MSG      Content-Length: 17
SIP.STACK MSG
SIP.STACK MSG      event flashhook
SIP.STACK MSG
SIP.STACK MSGSUM  Rx: SIP/2.0 200 OK
```

Figure 4. Flash Hook event sent to Broadsoft in Info Message

TA900 A sends an info message to the Broadsoft with a line that states “event flashhook”. This tells the Broadsoft that a flashhook event occurred on that line and it needs to take action. As a result the Broadsoft will issue a reinvite toTA900 A (figure 5). In the SDP of the invite the Broadsoft tells TA900 A to send its RTP to the IP of the Broadsoft media server, not TA900 B which was involved in the previous conversation. This creates a new audio path between the Broadsoft and TA900 A. The analog station on TA900 A will hear dial tone; this is being provided in the RTP stream from the Broadsoft, it is not provided locally. As the user dials digits of the third party to be conferenced in (256-555-9103) the Broadsoft will collect these digits. These digits are sent to the Broadsoft either via RFC 2833 (default) signaling or inband in the RTP.

```
SIP.STACK MSG     Rx: UDP src=10.1.4.5:33541 dst=10.19.210.6:5060
SIP.STACK MSG         INVITE sip:2565559102@10.19.210.6:5060;transport=UDP SIP/2.0
SIP.STACK MSG         Via:SIP/2.0/UDP 10.1.4.5;
SIP.STACK MSG         From:<sip:2565559101@10.1.4.5:5060;transport=UDP>;
SIP.STACK MSG         To:<sip:2565559102@10.1.4.5:5060;transport=UDP>;
SIP.STACK MSG         Call-ID:33516a0-a13d206-13c4-1726-1009aff5-1726@10.1.4.5
SIP.STACK MSG         CSeq:466036345 INVITE
SIP.STACK MSG         Contact:<sip:10.1.4.5:5060>
SIP.STACK MSG         Allow:ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER
SIP.STACK MSG         Supported:100rel
SIP.STACK MSG         Accept:multipart/mixed,application/sdp
SIP.STACK MSG         Max-Forwards:10
SIP.STACK MSG         Content-Type:application/sdp
SIP.STACK MSG         Content-Length:203
SIP.STACK MSG
SIP.STACK MSG         v=0
SIP.STACK MSG         o=BroadWorks 1055 1 IN IP4 10.1.2.6
SIP.STACK MSG         s=-
SIP.STACK MSG         c=IN IP4 10.1.2.6
SIP.STACK MSG         t=0 0
SIP.STACK MSG         m=audio 11916 RTP/AVP 0 2 101
SIP.STACK MSG         a=rtpmap:0 PCMU/8000
SIP.STACK MSG         a=rtpmap:2 G726-32/8000
SIP.STACK MSG         a=rtpmap:101 telephone-event/8000
SIP.STACK MSG         a=fmtp:101 0-11
SIP.STACK MSGSUM  Tx: SIP/2.0 100 Trying
SIP.STACK MSG     Tx: UDP src=10.19.210.6:5060 dst=10.1.4.5:5060
SIP.STACK MSG         SIP/2.0 200 OK
SIP.STACK MSG         From: <sip:2565559101@10.1.4.5:5060;transport=UDP>;
SIP.STACK MSG         To: ""<sip:2565559102@10.1.4.5:5060;transport=UDP>;
SIP.STACK MSG         Call-ID: 33516a0-a13d206-13c4-1726-1009aff5-1726@10.1.4.5
SIP.STACK MSG         CSeq: 466036345 INVITE
SIP.STACK MSG         Via: SIP/2.0/UDP 10.1.4.5;
SIP.STACK MSG         Supported: 100rel,replaces
SIP.STACK MSG         Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, PRACK
SIP.STACK MSG         User-Agent: ADTRAN_Total_Access_924e/13.01.03.E
SIP.STACK MSG         Contact: <sip:2565559102@10.19.210.6:5060;transport=UDP>
SIP.STACK MSG         Content-Type: application/SDP
SIP.STACK MSG         Content-Length: 179
```

```
SIP.STACK MSG
SIP.STACK MSG          v=0
SIP.STACK MSG          o=- 1156279162 1156279162 IN IP4 10.19.210.6
SIP.STACK MSG          s=-
SIP.STACK MSG          c=IN IP4 10.19.210.6
SIP.STACK MSG          t=0 0
SIP.STACK MSG          m=audio 10048 RTP/AVP 0 101
SIP.STACK MSG          a=rtpmap:0 PCMU/8000
SIP.STACK MSG          a=rtpmap:101 telephone-event/8000
SIP.STACK MSG          a=fmtp:101 0-15
SIP.STACK MSGSUM  Rx: ACK sip:2565559102@10.19.210.6:5060;transport=UDP SIP/2.0
```

Figure 5. TA900 A is reinvited to the Broadsoft media server for digit collection

Once the Broadsoft collects the digits that are being dialed (256-555-9103) it will create an invite for that phone number and send the invite to TA900 C. This invite is sent directly from the Broadsoft to TA900 C and will not be seen on TA900 A. Broadsoft will also send a series of reinvites to TA900A during this time so that ringback or other necessary sounds are heard at 256-555-9102. When 256-555-9103 answers the call the Broadsoft will send another reinvite to TA900 A this time telling it to send its media to TA900 C so 256-555-9101 can talk to 256-555-9103. This invite is shown below (figure 6).

```
SIP.STACK MSGSUM  Rx: INVITE sip:2565559102@10.19.210.6:5060;transport= UDP SIP/2.0
SIP.STACK MSGSUM  Tx: SIP/2.0 100 Trying
SIP.STACK MSG     Tx: UDP src=10.19.210.6:5060 dst=10.1.4.5:5060
SIP.STACK MSG          SIP/2.0 200 OK
SIP.STACK MSG          From: <sip:2565559101@10.1.4.5:5060;transport=UDP>;
SIP.STACK MSG          To: ""<sip:2565559102@10.1.4.5:5060;transport=UDP>;
SIP.STACK MSG          Call-ID: 33516a0-a13d206-13c4-1726-1009aff5-1726@10.1.4.5
SIP.STACK MSG          CSeq: 466036346 INVITE
SIP.STACK MSG          Via: SIP/2.0/UDP 10.1.4.5;
SIP.STACK MSG          Supported: 100rel,replaces
SIP.STACK MSG          Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, PRACK
SIP.STACK MSG          User-Agent: ADTRAN_Total_Access_924e/13.01.03.E
SIP.STACK MSG          Contact: <sip:2565559102@10.19.210.6:5060;transport=UDP>
SIP.STACK MSG          Content-Type: application/SDP
SIP.STACK MSG          Content-Length: 224
SIP.STACK MSG
SIP.STACK MSG          v=0
SIP.STACK MSG          o=- 1156279166 1156279166 IN IP4 10.19.210.6
SIP.STACK MSG          s=-
SIP.STACK MSG          c=IN IP4 10.19.210.6
SIP.STACK MSG          t=0 0
SIP.STACK MSG          m=audio 10048 RTP/AVP 18 0 101
SIP.STACK MSG          a=rtpmap:18 G729/8000
SIP.STACK MSG          a=fmtp:18 annexb=no
SIP.STACK MSG          a=rtpmap:0 PCMU/8000
SIP.STACK MSG          a=rtpmap:101 telephone-event/8000
SIP.STACK MSG          a=fmtp:101 0-15
SIP.STACK MSG
SIP.STACK MSG     Rx: UDP src=10.1.4.5:33541 dst=10.19.210.6:5060
SIP.STACK MSG          ACK sip:2565559102@10.19.210.6:5060;transport=UDP SIP/2.0
SIP.STACK MSG          Via:SIP/2.0/UDP 10.1.4.5;
SIP.STACK MSG          From:<sip:2565559101@10.1.4.5:5060;transport=UDP>;
SIP.STACK MSG          To:<sip:2565559102@10.1.4.5:5060;transport=UDP>;
SIP.STACK MSG          Call-ID:33516a0-a13d206-13c4-1726-1009aff5-1726@10.1.4.5
SIP.STACK MSG          CSeq:466036346 ACK
SIP.STACK MSG          Contact:<sip:10.1.4.5:5060>
SIP.STACK MSG          Max-Forwards:10
SIP.STACK MSG          Content-Type:application/sdp
SIP.STACK MSG          Content-Length:184
SIP.STACK MSG
SIP.STACK MSG          v=0
SIP.STACK MSG          o=BroadWorks 1053 2 IN IP4 10.19.210.7
SIP.STACK MSG          s=-
SIP.STACK MSG          c=IN IP4 10.19.210.7
SIP.STACK MSG          t=0 0
SIP.STACK MSG          m=audio 10000 RTP/AVP 18 101
SIP.STACK MSG          a=rtpmap:18 G729/8000
SIP.STACK MSG          a=rtpmap:101 telephone-event/8000
SIP.STACK MSG          a=fmtp:101 0-15
SIP.STACK MSG
```

Figure 6. TA900 A is reinvited to TA900 C

At this point the 256-555-9102 is able to talk to 256-555-9103 only, 256-555-9101 is on hold. It is necessary to bring 256-555-9101 into the call so that all three parties can participate in the conversation. To do this the analog station behind TA900 A again issues a flashhook. This flashhook is again sent to the Broadsoft in an info message just like the previous flashhook shown in figure 4. When the Broadsoft receives this flashhook it knows that both parties involved need to be conferenced together. To do this it sends reinvites to TA900 A, TA900 B, and TA900 C telling them to send their media to the Broadsoft media server. The message that TA900 A receives is shown below (figure 7). This acts as a conference server for all the parties involved so each user can hear the other two.

```
SIP.STACK MSGSUM  Rx: INVITE sip:2565559102@10.19.210.6:5060;transport= UDP SIP/2.0
SIP.STACK MSGSUM  Tx: SIP/2.0 100 Trying
SIP.STACK MSG     Tx: UDP src=10.19.210.6:5060 dst=10.1.4.5:5060
SIP.STACK MSG         SIP/2.0 200 OK
SIP.STACK MSG         From: <sip:2565559101@10.1.4.5:5060;transport=UDP>;
SIP.STACK MSG         To: ""<sip:2565559102@10.1.4.5:5060;transport=UDP>;
SIP.STACK MSG         Call-ID: 33516a0-a13d206-13c4-1726-1009aff5-1726@10.1.4.5
SIP.STACK MSG         CSeq: 466036347 INVITE
SIP.STACK MSG         Via: SIP/2.0/UDP 10.1.4.5;
SIP.STACK MSG         Supported: 100rel,replaces
SIP.STACK MSG         Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, PRACK,
SIP.STACK MSG         User-Agent: ADTRAN_Total_Access_924e/13.01.03.E
SIP.STACK MSG         Contact: <sip:2565559102@10.19.210.6:5060;transport=UDP>
SIP.STACK MSG         Content-Type: application/SDP
SIP.STACK MSG         Content-Length: 224
SIP.STACK MSG
SIP.STACK MSG         v=0
SIP.STACK MSG         o=- 1156279179 1156279179 IN IP4 10.19.210.6
SIP.STACK MSG         s=-
SIP.STACK MSG         c=IN IP4 10.19.210.6
SIP.STACK MSG         t=0 0
SIP.STACK MSG         m=audio 10048 RTP/AVP 18 0 101
SIP.STACK MSG         a=rtpmap:18 G729/8000
SIP.STACK MSG         a=fmtp:18 annexb=no
SIP.STACK MSG         a=rtpmap:0 PCMU/8000
SIP.STACK MSG         a=rtpmap:101 telephone-event/8000
SIP.STACK MSG         a=fmtp:101 0-15
SIP.STACK MSG
SIP.STACK MSG     Rx: UDP src=10.1.4.5:33541 dst=10.19.210.6:5060
SIP.STACK MSG         ACK sip:2565559102@10.19.210.6:5060;transport=UDPSIP/2.0
SIP.STACK MSG         Via:SIP/2.0/UDP 10.1.4.5;
SIP.STACK MSG         From:<sip:2565559101@10.1.4.5:5060;transport=UDP>;
SIP.STACK MSG         To:<sip:2565559102@10.1.4.5:5060;transport=UDP>;
SIP.STACK MSG         Call-ID:33516a0-a13d206-13c4-1726-1009aff5-1726@10.1.4.5
SIP.STACK MSG         CSeq:466036347 ACK
SIP.STACK MSG         Contact:<sip:10.1.4.5:5060>
SIP.STACK MSG         Max-Forwards:10
SIP.STACK MSG         Content-Type:application/sdp
SIP.STACK MSG         Content-Length:205
SIP.STACK MSG
SIP.STACK MSG         v=0
SIP.STACK MSG         o=BroadWorks 1055 3 IN IP4 10.1.2.6
SIP.STACK MSG         s=-
SIP.STACK MSG         c= IN IP4 10.1.2.6
SIP.STACK MSG         t=0 0
SIP.STACK MSG         m=audio 10000 RTP/AVP 18 101
SIP.STACK MSG         a=rtpmap:18 G729/8000
SIP.STACK MSG         a=fmtp:18 annexb=no
SIP.STACK MSG         a=rtpmap:101 telephone-event/8000
SIP.STACK MSG         a=fmtp:101 0-15
```

Figure 7. Broadsoft reinvites TA900 to conferencing media stream

If you experience any problems using your ADTRAN product, please contact ADTRAN Technical Support.

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