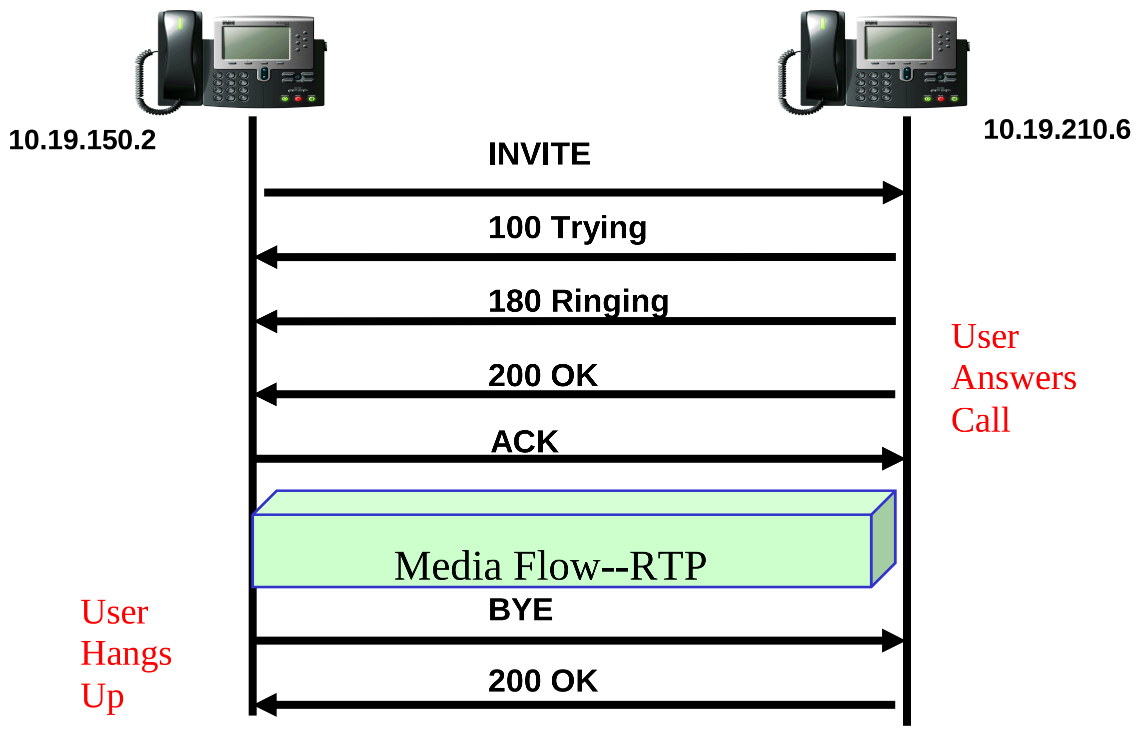
Troubleshooting Calls in the TA900

Overview: This document is designed for support personnel responsible for the installation and maintenance of TA900 series Integrated Access Devices. This guide may refer to additional documents available at ***kb.adtran.com*** and should be obtained for complete documentation. Certain Requests for Comments (RFCs) are referenced and are freely available in downloadable form on the Internet. These referenced documents are not included, but are left to the reader to obtain A basic assumption in this document is that the T1, and Ethernet interfaces are configured and passing data without error.

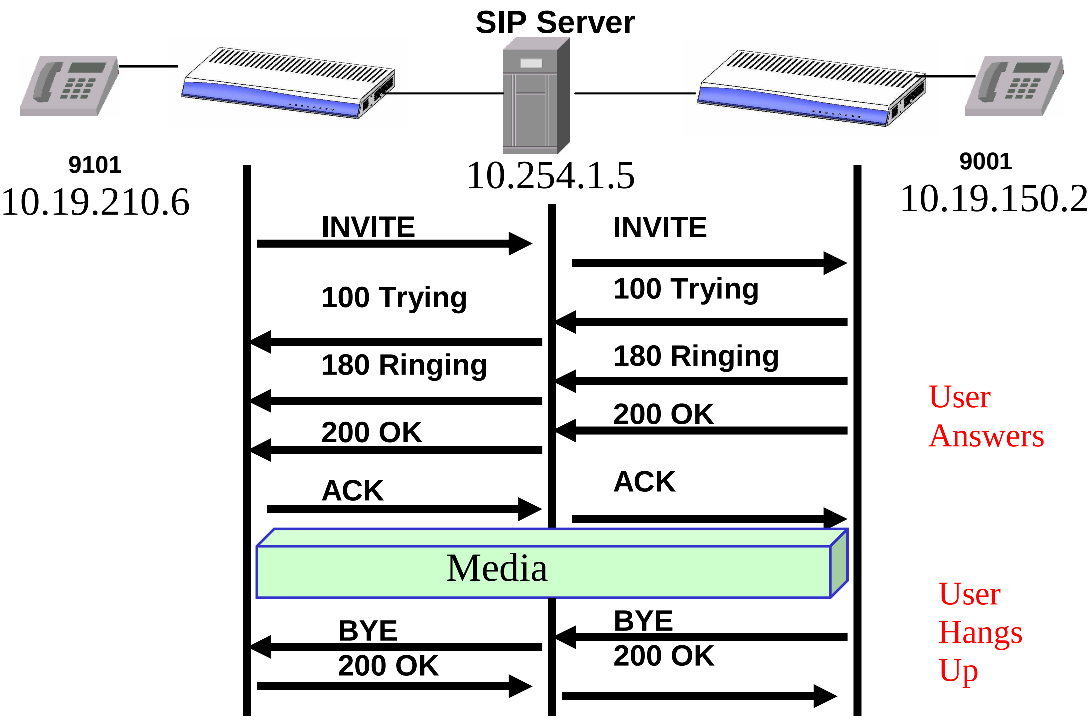
Introduction: VoIP calls consist of a signaling component and a streaming sampled voice component. The signaling component is made up of Session Initiation Protocol (SIP – RFC 3261) messages which include the Session Description Protocol (SDP – RFC 2327) call type and detail descriptors. Together these two protocols negotiate: Call Setup, Codec type, source/destination port numbers for the sampled audio stream, authentication, and provide a method for error handling.

The Real Time Protocol (RTP – RFC 2833) is the standard used to format the actual sampled audio voice call. Below is an example of a simple SIP/RTP call.



While the Virtual Call Flow may be represented by the diagram above, the actual call normally involves the user of a Registrar server. Each phone (User Agent) Registers with the Registrar Server. The Registrar server then keeps up with the IP Address of each user agent with their phone number in order to be able to forward a call request (INVITE) to the correct destination user agent. In most of the applications of the TA900

series products the SIP provider has a softswitch (Popular switches are: Broadsoft, Sylantro, Metaswitch, Asterisk, etc) that acts as a switch, a Registrar Server, and a PSTN Gateway.
In a typical application, assume there are two TA904 IADs, one at each of two locations. Both units are connected to a provider that supplies Voice (SIP based) and Internet Access via the TA904 as in the following diagram.



Registration: In order for calls to route properly, the Softswitch must know the IP address of the user agent. In the case of the TA900, the registration can be for a SIP Trunk, for an analog port, or for an attached digital trunk to a PBX (PRI or RBS) and its associated phone number. The identity in the Softswitch configuration MUST be the same as that configured on the TA900 product. Notice in the simple example below a basic SIP Registation packet. To see this and other sip packets use the following Debug command:

TA904# Debug sip stack message

```
REGISTER sip:10.254.1.5 SIP/2.0
From: <sip:9002@10.254.1.5>;tag=1bf4308-0-13c4-423-11c7d01f-423
To: <sip:9002@10.254.1.5>
 Call-ID: 1c0bf20-0-13c4-423-336f996f-423
 CSeq: 1 REGISTER
 Via: SIP/2.0/UDP 10.19.150.2:5060;
 branch=z9hG4bK-423-102b1e-6b5a4e85
 Max-Forwards: 70
 Supported: 100rel, replaces
 Contact: <sip:9002@10.19.150.2>
 Content-Length: 0
```

Registration details:

SIP Identity – Each endpoint in the SIP implementation is uniquely identifiable by using a Uniform Resource Identifier (URI). The URI is fairly flexible and takes a form very similar to an email address. Many of the rules governing acceptable URIs have evolved as an extension of the email header rules (RFC 2396).

Look at the example registration packet. Notice the From and To fields and their included URI information.

Some examples of acceptable URIs are:

```
      sip:alice@atlanta.com
      sip:alice@192.0.2.4
      sip:9002@10.254.1.5

      General form:

SIP-URI=  "sip:" [ userinfo ] hostport
           uri-parameters [ headers ]
userinfo   =  ( user / telephone-subscriber ) [ ":" password ] "@"
user       =  1*( unreserved / escaped / user-unreserved )

reserved    =  ";" / "/" / "?" / ":" / "@" / "&" / "=" / "+"
                    / "$" / ","
unreserved  =  alphanum / mark
mark        =  "-" / "_" / "." / "!" / "~" / "*" / "'"
                    / "(" / ")"
```

Registration with authentication

The softswitch must be configured with the same password as the TA900

```
REGISTER sip:10.254.1.5 SIP/2.0
From: <sip:9002@10.254.1.5>;tag=1bf4308-0-13c4-423-11c7d01f-423
To: <sip:9002@10.254.1.5>
 Call-ID: 1c0bf20-0-13c4-423-336f996f-423
 CSeq: 1 REGISTER
 Via: SIP/2.0/UDP 10.19.150.2:5060;
 branch=z9hG4bK-423-102b1e-6b5a4e85
 Max-Forwards: 70
 Supported: 100rel, replaces
 Contact: <sip:9002@10.19.150.2>
 Content-Length: 0
```

```
SIP/2.0 401 Unauthorized
 Via:SIP/2.0/UDP 10.19.150.2:5060;
branch=z9hG4bK-423-102b1e-6b5a4e85
 From:<sip:9002@10.254.1.5>;tag=1bf4308-0-13c4-423-11c7d01f-423
 To:<sip:9002@10.254.1.5>
 Call-ID:1c0bf20-0-13c4-423-336f996f-423
 CSeq:1 REGISTER
WWW-Authenticate:DIGEST
realm="test.adtran.com",algorithm=MD5,nonce="1114537059153"
 Content-Length:0
```

The non-encrypted request and 401 unauthorized reply occur first then:

```
REGISTER sip:10.254.1.5 SIP/2.0
From: <sip:9002@10.254.1.5>;tag=1bf4308-0-13c4-423-11c7d01f-423
To: <sip:9002@10.254.1.5>
 Call-ID: 1c0bf20-0-13c4-423-336f996f-423
 CSeq: 2 REGISTER
 Via: SIP/2.0/UDP 10.19.150.2:5060;branch=z9hG4bK-423-102b46-637400ad
 Max-Forwards: 70
 Supported: 100rel, replaces
 Contact: <sip:9002@10.19.150.2>
 Authorization:Digest username="",realm="test.adtran.com",nonce="1114537059153",uri="sip:10.254.1.5",response="80b0f42079d5e543a14045cbc59db47b",algorithm=MD5
 Content-Length: 0
```

```
SIP/2.0 200 OK
Via:SIP/2.0/UDP 10.19.150.2:5060;branch=z9hG4bK-423-102b46-637400ad
From:<sip:9002@10.254.1.5>;tag=1bf4308-0-13c4-423-11c7d01f-423
To:<sip:9002@10.254.1.5>
 Call-ID:1c0bf20-0-13c4-423-336f996f-423
 CSeq:2 REGISTER
Contact:<sip:9002@10.19.150.2>;q=0.5;
expires=3599
 Content-Length:0
```

The encrypted request (note the nonce indicating encryption – MD5 type) and the 200 OK reply.

Registration Troubleshooting

Possible Registration Issues and troubleshooting

Use the commands:

Show sip user-registration: SIP Users are those registering from a SIP IP Phone (either a hard or soft phone)

Show sip trunk-registration: Voice Users and Group Trunks normally register via the SIP trunk to their registrar.

Reachability is often an issue in registration problems. Try to ping the Registrar from the TA900 to verify that the packets from the TA900 are being properly routed. Also verify firewall permissions.

URI common issues: Use the command:

debug sip stack message

to debug sip messages and to see warnings when the sip parser in the TA900 is unable to understand a URI or message construction.

Ex: `1970.01.01 01:28:40 SIP.STACK ERROR  PARSER       SIP Parser Error : Missing URL OTHER PARAM VAL, line 7, column 70`

Generally, unescaped reserved characters or poorly constructed headers result in parser errors. Please contact ADTRAN Technical Support if you are not able to resolve parser errors.

NOTE: It is necessary to double quote escape reserved characters if they appear in the userinfo component of the header (e.g. “j@mes”@adtran.com).

Call Routing: Calls are routed with respect to the TA900. When documentation refers to incoming or outgoing calls the reference is made with the understanding that the call is coming into the TA900 from a SIP Trunk, an analog PoTs trunk, or a PRI/CAS Trunk. So a call might come into the TA900 on a SIP Trunk, but that call is then routed out of a PRI Trunk to an adjoining PBX as an example. Therefore each call consists of two call legs: the inbound leg and the outbound call leg. Inbound and Outbound call direction may effect where certain commands are configured.

Call Endpoints: Calls can be routed to call endpoints such as Voice Users (SIP or Analog Phones), or relayed via trunks to endpoints such as a PRI or CAS Endpoint. All endpoint identities are registered to the Softswitch, so each Phone or SIP Phone needs an identity created on the Softswitch. Small exceptions to this rule are phones collectively used in a Ring Group where the ring group is itself an endpoint and will require an identity. See the KB document TA900 Ring Groups for additional information. Finally, PRI or CAS trunks also have an identity that will be registered with the softswitch.

Trunk Accounts & Group Trunks:

Trunk Account:
Every incoming and outgoing trunk is defined in a trunk account. The trunk account can be setup with the GUI or command line. Examine the following PRI trunk used to deliver calls to and from a PBX local to the TA900 as an example:

```
!
interface t1 0/2
  tdm-group 1 timeslots 1-24 speed 64
  no shutdown
!
interface pri 1
  calling-party override always
  calling-party name “CUSTOMER”
  calling-party number 2569638000
  digits-transferred 7
  connect t1 0/2 tdm 1
  no shutdown
!
isdn-group 1
  connect pri 1
!
voice trunk T01 type sip
  no reject-external
  sip-server primary 10.0.0.1
  authentication username “Globaluser” password “Globalpassword”
  register 2569638000 auth-name “user8000” password “pass8000”
  register 2569639000 register range 2569638101 2569638103 auth-name
“rangeuser” password “rangepassword”
!
voice trunk T02 type isdn
  no reject-external
  connect isdn-group 1
```

From the example above, the PRI setup seems involved, however if we start with the voice trunk account we can see that the voice trunk is connected to the isdn-group 1 which shows a connection to PRI 1. These successive levels offer a greater degree of control when needed, and allows most trunks to be completely constructed.

Voice grouped-trunk
A voice Grouped trunk defines the outgoing properties of one or more trunks. In the above example we saw that the setup of the PRI allows for conversion of whatever the local PBX sends into the TA900 as a calling party name and number will be overwritten. We may only want certain destination numbers accepted into the PRI and PBX. Note also that the SIP Trunk holds the registration for the voice grouped-trunk phone numbers. We will set up these restrictions in the outbound Grouped-trunk PRI as follows:

```
voice grouped-trunk PRI
  no description
  trunk T02
  accept 256-963-8000 cost 0
  accept 256-963-810[123] cost 0
```

Note that the main number and three extensions are permitted to be routed into the PBX.

Dial Plan and Class of Service:

The TA900 dial plan accomplishes two major tasks: helps determine when a call has been completely dialed, and categorizes a call to determine if a particular Trunk or Voice user may be allowed to place such a call (e.g. 900 numbers). Dial completion allows completely dialed calls to be placed immediately without timeout elapse. For instance if the internal call dial plan is four digits starting with 4, the

```
"voice dial-plan 1 extension 4XXX"
```

pattern ensures that the interdigit timeout (10 secs) does not need to elapse before the call is routed.

A public access (Lobby) phone may be connected to an analog FXS port specified in a voice user. That Voice User should have a Class of Service (CoS) that only permits internal extension calls, but no long distance.

Destination routing: The TA900 routes incoming calls to endpoints using a least cost/longest-match algorithm. The incoming call is routed out to the endpoint that has the closest matching accept criteria at the least cost. Cost is the first level differentiator and if the costs are equal for two competing trunk accepts, the most specific or longest match will have the call routed on that trunk. It is imperative that for the most predictable call routing, accept numbers are configured on Group trunks as detailed as possible. Source routing is NOT currently supported.

CODECs: Anytime the call changes format, the Digital Signal Processor comes into use. If a call is received as a SIP/RTP call, and is to be sent out of an analog Foreign Exchange Station (FXS) port, it must be converted. If that call is to be routed from a Softwitch to a standard analog telephone set (common home telephone), a

Coder/Decoder or CODEC (DSP Based Digital to Analog voice converter) is engaged to convert the RTP digital voice to an analog electrical signal that will drive the phone speaker or earpiece. The TA900 supports two CODEC Types: G.711 (toll Quality) or G.729 (Compressed near toll quality). A CODEC pool must be defined and assigned to the SIP trunk(s) for RTP conversion to/from FXS/FXO/PRI/CAS trunks.

```
!
voice codec-list global
  codec g729
  codec g711ulaw
!
voice trunk T01 type sip
  no reject-external
  sip-server primary 10.1.4.5
  registrar primary 10.1.4.5
  register 2565551001
  register 2565551002
  codec-group global
 !
```

Call routing Modes: Voice Feature mode network is the default and only mode used for applications described in this document for software revisions up to and including Rev. 15.

Voice Feature Mode Network – All calls are routed thru the Softswitch. The TA900 in this mode is a User Agent and must allow the softswitch full knowledge of all call legs including a call from one analog port to another. Unless the softswitch is involved, three-way calling and other features may not operate correctly. SIP hard and soft phones also need to be registered and have calls routed to a softswitch for full feature functionality.

Voice Feature Mode Local – Not currently applicable, but planned for future release for "survivability mode."

SIP Trunks:

Debug SIP issues with the debug command: debug sip stack message

Inbound (RX): An inbound SIP call is initiated when the TA900 receives an INVITE message. The SIP Invite includes the source and destination IP addresses, a FROM and TO header, a call sequence number, Capabilities and the SDP for this call that we can use to troubleshoot. Issues may arise when the incoming TO header is not formatted to match the dialing plan or is not formatted in the way a PBX can use the information.

Outbound (TX): An outbound SIP call is initiated when the TA900 receives an analog or PRI/CAS call and then sends a SIP Trunk corresponding INVITE message. Problems occur if the call coming from a PBX does not have the correct calling Party information (ANI), Called Party number, or it is not in a format that matches the configuration of the TA900.

Inbound SIP Call example:

```
19:30:48 SIP.STACK MSG
Rx: INVITE sip:2143500639@66.55.61.174;transport=udp
SIP/2.0
19:30:48 SIP.STACK MSG
Via: SIP/2.0/UDP 206.222.127.38:5060;branch=z9hG4
bK00000dfc0001ea550002
19:30:48 SIP.STACK MSG
Via: SIP/2.0/UDP 10.10.10.2;branch=z9hG4bK-BroadW
orks.10.10.10.2-10.10.11.130V44988-0-718247046-378261406-1169667598602-
19:30:48 SIP.STACK MSG
Record-Route: <sip:2143500639@206.222.127.38:5060
;lr;transport=udp>
19:30:48 SIP.STACK MSG
From: "TOLL FREE CALL" <sip:8009377473@206.222.12
7.38:5060;user=phone>;tag=378261406-1169667598602-
19:30:48 SIP.STACK MSG         To: ". EXCEL STEEL, INC."
<sip:2143500639@as1>
19:30:48 SIP.STACK MSG         Call-ID: BW143958602240107-1441077629@10.10.10.2
19:30:48 SIP.STACK MSG         CSeq: 718247046 INVITE
19:30:48 SIP.STACK MSG         Contact: <sip:206.222.127.38:5060;transport=udp>
19:30:48 SIP.STACK MSG         supported: 100rel
19:30:48 SIP.STACK MSG         max-forwards: 10
19:30:48 SIP.STACK MSG         Allow: ACK, BYE, CANCEL, INFO, INVITE, OPTIONS,
PRACK, REFER, UPDATE, NOTIFY
19:30:48 SIP.STACK MSG         Content-Type: application/sdp
19:30:48 SIP.STACK MSG         Accept: multipart/mixed, application/sdp
19:30:48 SIP.STACK MSG         Content-Length: 00218
19:30:48 SIP.STACK MSG
19:30:48 SIP.STACK MSG         v=0
19:30:48 SIP.STACK MSG         o=BroadWorks 5000076 1 IN IP4 206.222.127.38
19:30:48 SIP.STACK MSG         s=-
19:30:48 SIP.STACK MSG         c=IN IP4 206.222.127.38
19:30:48 SIP.STACK MSG         t=0 0
19:30:48 SIP.STACK MSG         m=audio 32582 RTP/AVP 0 2 18
19:30:48 SIP.STACK MSG         a=rtpmap:0 PCMU/8000/1
19:30:48 SIP.STACK MSG         a=rtpmap:2 G726-32/8000/1
19:30:48 SIP.STACK MSG         a=rtpmap:18 G729/8000/1
19:30:48 SIP.STACK MSG         a=ptime:20
19:30:48 SIP.STACK MSG         a=sendrecv
19:30:48 SIP.STACK MSG
```

Session Border Controllers (SBC) and SIP Grammar

The role of a SIP Proxy and/or SBC is quite simple: relay call signaling to provide additional control and security for public SIP networks. However, not all SBCs are created nor configured equally. This paragraph provides a general introduction to the manipulation of headers within the SIP message. These headers are significant to call processing by the SIP server, SBC, and User Agent (TA900) alike, and are typically fields altered by the SBC. If a certain header is altered or populated with an unexpected address, one or more devices in the SIP network may reject it. These rejections come in the form of a SIP 404 – Not Found message, 604 – Does Not Exist Anywhere, etc. The ip sip grammar command and its arguments serve as a way to alter the population of these headers so they are properly accepted by the SBC and/or SIP server. However, there is no standard methodology to accomplish a particular result. Commands must be entered based on the configuration, header requirements, and expectations of your SBC and/or SIP server. Please contact your SBC manufacturer or administrator for the individual needs in your SIP network. (See KB TA900 SIP Grammar)

PRI PBX Trunk:

An ISDN trunk communicates with the TA900 using the Q.931 message protocol. The protocol debugger in the TA900 allows an English language translation of these messages as shown below. The TA900 supports both a User and a Network role with respect to the attached PBX. In general the Network role is NOT used to attach to a TELCO ISDN circuit to the TA900 at this time.

Configure the ISDN DSX PRI

Configure as shown below:

```
!
interface t1 0/2
  tdm-group 1 timeslots 1-24 speed 64
  no shutdown
!
interface pri 1
  calling-party override always
  calling-party name “CUSTOMER”
  calling-party number 2569638000
  digits-transferred 7
  connect t1 0/2 tdm 1
  no shutdown
!
isdn-number-template 1 prefix 1 national Nxx-Nxx-xxxx
isdn-number-template 2 prefix 011 international $

!
isdn-group 1
  connect pri 1
!
voice trunk T01 type sip
  no reject-external
  sip-server primary 10.0.0.1
  authentication username “Globaluser” password “Globalpassword”
  register 2569638000 auth-name “user8000” password “pass8000”
  register 2569639000 register range 2569638101 2569638103 auth-name
“rangeuser” password “rangepassword”
!
voice trunk T02 type isdn
  no reject-external
  connect isdn-group 1
```

Use the isdn-number-template command to create an entry in the ISDN number type template that is used when encoding the called party and calling party information elements for inbound and outbound ISDN calls. Use the no form of the command to delete the configured entry. Variations of this command include

the following:
isdn-number-template <template id> prefix <number> abbreviated <pattern>
isdn-number-template <template id> prefix <number> international <pattern>
isdn-number-template <template id> prefix <number> national <pattern>
isdn-number-template <template id> prefix <number> network-specific <pattern>
isdn-number-template <template id> prefix <number> subscriber <pattern>
isdn-number-template <template id> prefix <number> unknown <pattern>

In the example above, if we receive a number such as 01152031121 we may need to send it to the PBX coded as an international call type and send the number as 52031121. If a prefix is present in the command as above, the prefix is removed before the call is forwarded. As in the debug below, the 1 is removed and the call is tagged as national and sent into the ISDN network.

Debugging the PRI Trunk:

Use the debug command: debug isdn L2-formatted
to see ISDN Q.931 messages.

A sample is shown below. Notice the Calling Party Number and Called Party number in this initial SETUP message. Proper formatting of the Number type, Number Plan, and manipulation of these with the ISDN-number-template affects call completion.

```
L2-Formatted===================================================
L2-Formatted      Sent = Sapi:00  C/R:C Tei:00
L2-Formatted      Ctl:INFO     Ns:59   Nr:10
L2-Formatted      Prot:08  CRL:2  CRV:1F40
L2-Formatted       M - 05 SETUP
L2-Formatted      IE - 04 BEARER CAPABILITY   Len=3
L2-Formatted             80 Xfer Cap.:SPEECH
L2-Formatted             90 Xfer Rate:64k
L2-Formatted             A2 Layer 1:u-Law
L2-Formatted        IE - 18 CHANNEL ID          Len=3
L2-Formatted             A1 Primary Rate
L2-Formatted                Intfc ID:IMPLICIT
L2-Formatted                Pref/Excl:PREFERRED
L2-Formatted                D-Chan Indicated:NO
L2-Formatted                Chan. Sel:FOLLOWS
L2-Formatted             83 Numb/Map:NUMBER
L2-Formatted             96 Channel:22
L2-Formatted        IE - 6C CALLING PARTY #     Len=12
L2-Formatted             21 Numb. Type:NATIONAL
L2-Formatted                Numb. Plan:ISDN/Telephony
L2-Formatted             83 Presentation:ALLOWED
L2-Formatted                Ph.# 3096559985
L2-Formatted        IE - 70 CALLED PARTY #      Len=8
L2-Formatted             80 Numb. Type:UNKNOWN
L2-Formatted                Numb. Plan:UNKNOWN
L2-Formatted                Ph.# 5892832
L2-Formatted===================================================
```

Analog Voice Users:

Analog users include regular analog telephone sets, fax machines, and modems. The three configurations differ and the performance can be significantly affected by the setup.

Voice or telephone user:
A Voice user is defined and an FXS port is associated with that user. When a SIP call leg and an analog call leg are activated as a complete call, the DSP (CODEC) is invoked to convert the SIP associated RTP Stream to/from analog voice. In addition the TA900 voice state machine must convert SIP Messages to analog events such as ringing voltages, dial tone, busy, and other call progress tones.

By using the following debug commands, it is possible to see this cause and effect.

Debug interface fxs
Debug voice tones
Debug voice summary
Debug sip stack summ

```
13:31:37 TONESERVICES.EVENTS fxs 0/1 - empty - DialTone Generation: Request
resource
13:31:37 TONESERVICES.EVENTS fxs 0/1 - dsp 0/1.1 - DialTone Generation: DSP
channel allocated for the resource
13:31:37 TONESERVICES.EVENTS fxs 0/1 - dsp 0/1.1 - DialTone Generation:
constructed
13:31:37 TONESERVICES.EVENTS fxs 0/1 - dsp 0/1.1 - DialTone Generation:
starting
13:31:37 TONESERVICES.EVENTS fxs 0/1-dsp 0/1.1 - DialTone Generation: TDM map
13:31:37 TONESERVICES.EVENTS fxs 0/1-empty - Tone Detection: Request resource
13:31:37 TONESERVICES.EVENTS fxs 0/1-dsp 0/1.1 - Tone Detection: DSP channel
allocated for the resource
13:31:37 TONESERVICES.EVENTS fxs 0/1-dsp 0/1.1 - Tone Detection: constructed
13:31:37 TONESERVICES.EVENTS fxs 0/1-dsp 0/1.1 - Tone Detection: starting
13:31:37 TONESERVICES.EVENTS fxs 0/1-dsp 0/1.1 - Tone Detection: TDM map
!! Here the local phone goes off-hook and receives dialtone
2007.07.27 13:31:37 FXS.0/1 Offhook Detected -1691706980 ms
13:31:40 TONESERVICES.EVENTS fxs 0/1-dsp 0/1.1 - Tone Detection: received digit
(9) event
13:31:40 TONESERVICES.EVENTS fxs 0/1-dsp 0/1.1 - DialTone Generation: stopping
13:31:40 TONESERVICES.EVENTS fxs 0/1-dsp 0/1.1 - DialTone Generation: TDM unmap
13:31:40 TONESERVICES.EVENTS fxs 0/1-dsp 0/1.1 - DialTone Generation: release
13:31:40 TONESERVICES.EVENTS fxs 0/1-dsp 0/1.1 - Tone Detection: received digit
(0) event
13:31:41 TONESERVICES.EVENTS fxs 0/1-dsp 0/1.1 - Tone Detection: received digit
(2) event
13:31:42 TONESERVICES.EVENTS fxs 0/1-dsp 0/1.1 - Tone Detection: received digit
(8) event
!! We touchtone a 9, our first digit which stops dialtone and gathers
!! digits for the call we want to place – to 9 0 2 8
13:31:42 VOICE.SUMMARY voice user 9065 cos allowed the call to Extensions
13:31:42 TONESERVICES.EVENTS fxs 0/1-dsp 0/1.1 - Tone Detection: stopping
13:31:42 TONESERVICES.EVENTS fxs 0/1-dsp 0/1.1 - Tone Detection: TDM unmap
13:31:42 TONESERVICES.EVENTS fxs 0/1-dsp 0/1.1 - Tone Detection: release
13:31:42 VOICE.SUMMARY 9065 is calling T01 (9028).
!! Now we construct and send an INVITE for 9028 to the Softswitch
13:31:42 SIP.STACK MSGSUM  Tx: INVITE sip:9028@10.1.4.5:5060 SIP/2.0
13:31:42 SIP.STACK MSGSUM  Rx: SIP/2.0 100 Trying
13:31:42 SIP.STACK MSGSUM  Rx: SIP/2.0 180 Ringing
```

```
13:31:42 SIP.STACK MSGSUM  Tx: PRACK sip:10.1.4.5:5060 SIP/2.0
!! When we get a 180 ringing we must play Ringback to the local phone
13:31:42 TONESERVICES.EVENTS fxs 0/1- empty- Ringback Generation: Request
resource
13:31:42 SIP.STACK MSGSUM  Rx: SIP/2.0 200 OK
!! this 200 OK is confirmation of the call connecting SIP wise.
13:31:42 TONESERVICES.EVENTS fxs 0/1-dsp 0/1.1 - Ringback Generation: DSP
channel allocated for the resource
13:31:42 TONESERVICES.EVENTS fxs 0/1-dsp 0/1.1 - Ringback Generation:
constructed
13:31:42 TONESERVICES.EVENTS fxs 0/1-dsp 0/1.1 - Ringback Generation: starting
13:31:42 TONESERVICES.EVENTS fxs 0/1-dsp 0/1.1 - Ringback Generation: TDM map
13:31:52 SIP.STACK MSGSUM  Rx: SIP/2.0 200 OK
13:31:52 TONESERVICES.EVENTS fxs 0/1-dsp 0/1.1 - Ringback Generation: stopping
13:31:52 TONESERVICES.EVENTS fxs 0/1-dsp 0/1.1 - Ringback Generation: TDM unmap
13:31:52 TONESERVICES.EVENTS fxs 0/1- dsp 0/1.1 - Ringback Generation: release
13:31:52 VOICE.SUMMARY 9065 is connected to T01 (9028)
!! Ringback is turned off and voice (RTP) cuts through to/from analog
13:31:52 SIP.STACK MSGSUM  Tx: ACK sip:10.1.4.5:5060 SIP/2.0
```

Voice User - Analog port (FXS) config:

The following configuration allows changes to many useful settings:

```
!
interface fxs 0/1
  description "Joes phone"
  alias
  impedance 600r
  rx-gain -3.0
  tx-gain -6.0
  ring-voltage 50
  signal loop-start
  no shutdown
!
!
voice user 1001
  connect fxs 0/1
  cos day "GLOBAL"
  sip-identity 1001 T01 register auth-name USER1 password PASSWORD1
  codec-group GLOBAL
!
```

Connect fxs 0/1 in the voice user settings ties the physical fxs 0/1 port with the virtual voice user 1001. The 1001 in this case is the user phone number. This user might also have been defined as Voice User 2565551001 if the identity was configured that way in the softswitch. Also the softswitch can be configured to use an alias to accept calls from 1001 or from 256-555-1001. In this instance the registration is setup on the voice user, but it can also be configured on the sip trunk. Analog settings such as incoming or outgoing volume and impedance is set on the fxs port. Default setting for telephones normally do not require modification for volume or impedance.

Modem/Fax User - Analog port (FXS) config:

Modems and Fax machines connected to an analog port on the TA900 series IADS require a slightly different configuration:

```
voice user 3053622021
  connect fxs 0/12
  password "1234"
  description "Fax Machine"
  no special-ring-cadences
  forward-disconnect delay 250
  no plc
  no echo-cancellation
  codec-group FAX
!
```

The main difference in the configuration is that modems and faxes have their own echo canceller and normally send a disabling tone for analog lines. Turning off the canceller insures that it is off and does not falsely enable again. Also, the forward disconnect option assures that the line will “Hang-up” when the far end disconnects first. This is accomplished by removing line current on the loop for a set time (250 milliseconds here). Most modems and faxes detect this a a command to hang up. Finally, plc is a Packet Loss Concealment Algorithm that does more harm than good on modem circuits. What makes voice sound better on a voice call adversely affects modem and fax operation. Fax and modem calls must always come in on G.711 encoded packets.

Analog troubleshooting:

If we debug interface fxs 0/1 we can see even momentary breaks in the loop current that indicate physical issues on the punch block, or line current level problems. Always make sure the correct signaling is setup and not loop start signaling in a ground start application.

Some modems and fax level and impedance requirements differ from the default TA900 settings. Debug voice tones and debug voice summary can help identify when the TA900 is having issues detecting the proper DTMF digits. Varying the TX and RX gain on the FXS port as well as adjusting the impedance can help create a better match when there is a problem detecting tones or when faxes fail to complete.

Ring Groups:

Ring groups are used on the TA900 to successively ring multiple physical lines into a PBX when the same DID (Called Party) is dialed. It is usually better to set this up on the softswitch, but if need be can be configured on the TA900 (See KB article TA900 Ring Groups).

Called and Calling Party Number/Name adjustments:

DNIS Manipulation: The Softswitch may send 10 digit phone-numbers down the SIP Trunk, but the local PBX may be set up to operate with a 4 digit phone number. The TA900 products provide configuration options that allow the DNIS to be changed into a format that equipment such as a PBX can use.

ANI Manipulation: ANI or Automatic Number Identification services allow the party that is receiving the call to identify the call originator. Caller-ID is a telephone service that can be maintained through a TA900. (See KB article Configuring Caller ID in AOS IADs)

Firewall use and effect on TA900 voice mode operations

A stateful inspection firewall is required for proper public access to the calling capabilities of a TA900. In some instances, the TA900 is deployed with a single or multilink PPP access circuit from a provider. The data portion of that access is normally used to provide Internet access services for the organization using the TA900.

The firewall configured access must "permit" all traffic in and outbound to the Softswitch and SBC in an unencumbered fashion to prevent issues with inbound calls and to keep them from being blocked.