AudioCodes Quick Reference Guide
Outbound Calls Not Working

Background

There are many reasons why you may experience outbound call failure. This quick reference guide is aimed at helping you to recognize some of the common causes.

First, we need to determine where the call is failing. Is it failing at the AudioCodes SBC, the PBX or on the Carrier side? To find this we will use the AudioCodes Syslog Viewer Application to diagnose issues with the call.

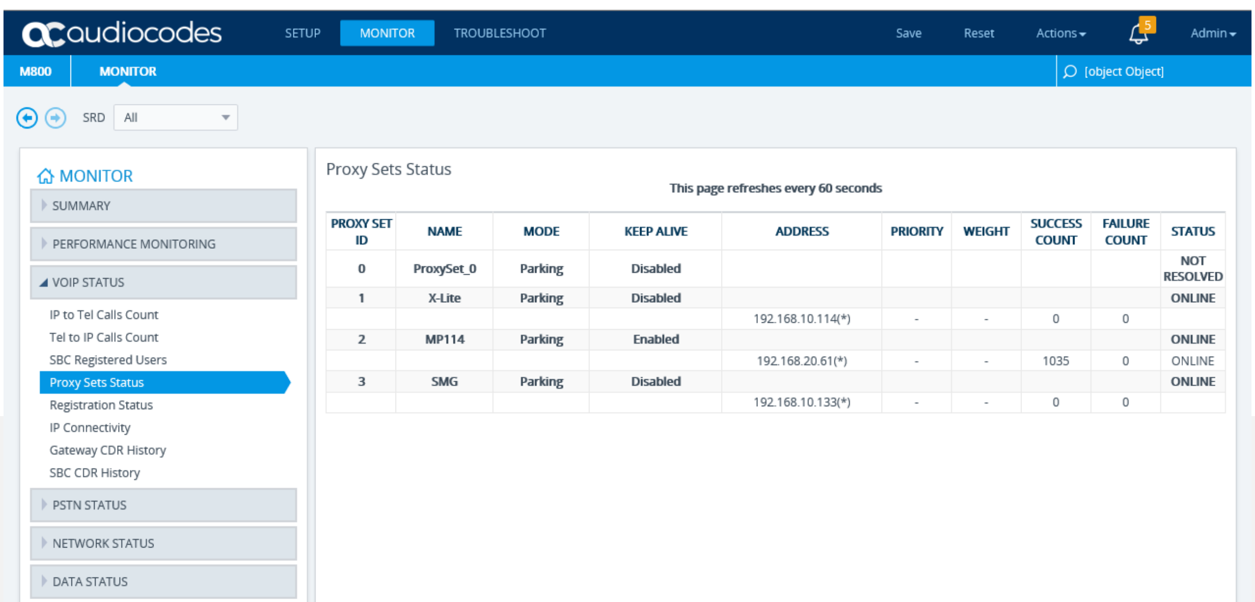
Open Syslog Viewer and make a test call to see if the call is reaching the SBC. If you do not see any evidence of the call coming in, there is probably an issue on the PBX side.

If you can see the call reaching the SBC, check for the response codes sent by the device to determine why the call is failing.

Calls Failing at SBC:

Proxy Set / IP Group Connectivity – If you are set up for proxy keep alive, make sure that the server is responding to the devices Option messages.

To check the Proxy Set Status, go to Monitor > VoIP Status > Proxy Sets Status



If the Proxy set shows offline, an Alarm will also be generated indicating that the IP Group will be temporarily blocked and the device will not attempt to route calls to this Proxy / IP Group. Please check with the provider to determine if they are receiving the Options messages or if they are experiencing an outage.

If the Proxy is online, use Syslogs to determine the cause of the failure.

```
[SID=4924a3:39:1089]  (       lgr_sbc)(     24528)   (#71) SBCRoutesIterator Deallocated. [Time:26-07@11:43:52.703]
[SID=4924a3:39:1089]  (     sip_stack)(     24529)   TransactionUserMngr::HandleNewSIPMessage - Incoming request is rejected [Time:26-07@11:43:52.703]
[SID=4924a3:39:1089]  (     sip_stack)(     24530)   ---- Incoming SIP Message from 192.168.20.61:5060 to SIPInterface #1 (SBC) UDP TO(#1) ---- [Time:26-07@11:43:52.703]
[SID=4924a3:39:1089]  INVITE sip:9999@192.168.20.253;user=phone SIP/2.0
                      Via: SIP/2.0/UDP 192.168.20.61:5060;branch=z9hG4bKac1542389645
                      Max-Forwards: 70
                      From: <sip:601@192.168.20.61>;tag=1c1542379846
                      To: <sip:9999@192.168.20.253;user=phone>
                      Call-ID: 15423792721972018221315@192.168.20.61
                      CSeq: 1 INVITE
                      Contact: <sip:601@192.168.20.61:5060>
                      Supported: em,100rel,timer,replaces,path,resource-priority,sdp-anat
                      Allow: REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,SUBSCRIBE,UPDATE
                      User-Agent: MP-114 FXS_FXO/v.6.60A.342.003
                      Content-Type: application/sdp
                      Content-Length: 430

                      v=0
                      o=AudiocodesGW 1542351257 1542351256 IN IP4 192.168.20.61
                      s=Phone-Call
                      c=IN IP4 192.168.20.61
                      t=0 0
                      m=audio 6030 RTP/AVP 8 96
                      a=ptime:20
                      a=sendrecv
                      a=rtpmap:8 PCMA/8000
                      a=rtpmap:96 telephone-event/8000
                      a=fmtp:96 0-15
                      m=image 6032 udptl t38
                      a=sendrecv
                      a=T38FaxVersion:0
                      a=T38MaxBitRate:14400
                      a=T38FaxMaxBuffer:1024
                      a=T38FaxMaxDatagram:238
                      a=T38FaxRateManagement:transferredTCF
                      a=T38FaxUdpEC:t38UDPRedundancy
                       [Time:26-07@11:43:52.704]
[SID=4924a3:39:1089]  (     sip_stack)(     24531)   New SIPMessage created - (#1) [Time:26-07@11:43:52.704]
[SID=4924a3:39:1089]  (      lgr_flow)(     24532)   (#0)IDSMngr <- (#0): IDSReportEvent [Time:26-07@11:43:52.706]
[SID=4924a3:39:1089]  (      lgr_flow)(     24533)   (#0)MessageRejectionMngr <- (#0): RequestTerminationEvent [Time:26-07@11:43:52.706]
[SID=4924a3:39:1089]  (     sip_stack)(     24534)   New SIPMessage created - (#0) [Time:26-07@11:43:52.707]
[SID=4924a3:39:1089]  (     sip_stack)(     24535)   ---- Outgoing SIP Message to 192.168.20.61:5060 from SIPInterface #1 (SBC) UDP TO(#1) ---- [Time:26-07@11:43:52.708]
[SID=4924a3:39:1089]  SIP/2.0 500 Server Internal Error
                      Via: SIP/2.0/UDP 192.168.20.61:5060;branch=z9hG4bKac1542389645
                      From: <sip:601@192.168.20.61>;tag=1c1542379846
                      To: <sip:9999@192.168.20.253;user=phone>;tag=1c744444377
                      Call-ID: 15423792721972018221315@192.168.20.61
                      CSeq: 1 INVITE
                      Reason: SIP ;cause=500 ;text="General Routing Failure"
```

Routing:

Check the IP 2 IP routing table to make sure there is an appropriate route for the call to traverse the SBC and go to the correct destination.

The IP 2 IP routing table will need to match calls exactly as they are presented to the SBC in order to select the correct route row.

Classification:

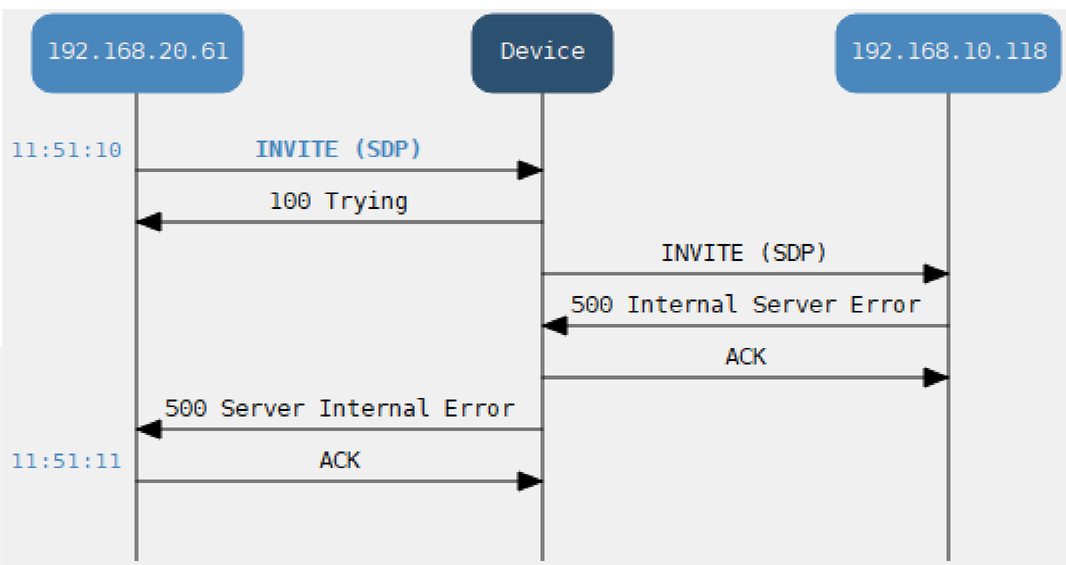
Classification is the process that identifies the incoming call (SIP dialog request) as belonging to a specific SIP entity (IP Group). There are three chronological classification stages, where each stage is done only if the previous stage fails. The device first attempts to classify the SIP dialog by checking if it belongs to a user that is already registered in the device's registration database. If this stage fails, the device checks if the source IP address is defined for a Proxy Set and if yes, it classifies it to the IP Group associated with the Proxy Set. If this fails, the device classifies the SIP dialog using the Classification table, which defines various characteristics of the incoming dialog that if matched, classifies the call to a specific IP Group. The main characteristics of the incoming call is the SIP Interface that is associated with the SRD for which the Classification rule is configured.

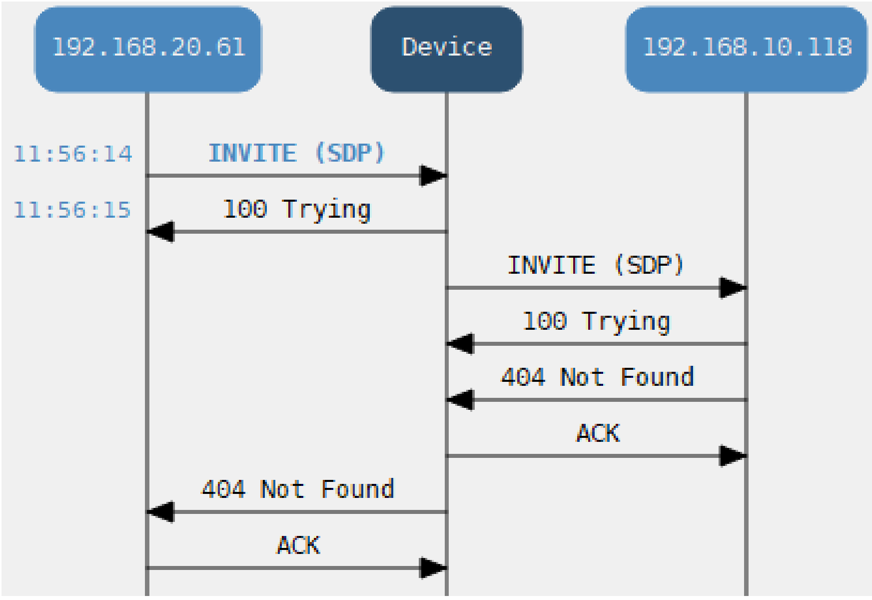
```
[SID=4924a3:39:1385]  (       sip_sbc)(     36488)   SIPSBCMngr::GetStackEP - No user is found in DB [Time:26-07@14:02:09.722]
[SID=4924a3:39:1385]  (       lgr_sbc)(     36489)   (#63) SBCRoutesIterator Allocated. [Time:26-07@14:02:09.723]
[SID=4924a3:39:1385]  (       lgr_sbc)(     36490) ?? [WARNING]  Classification failed. [Time:26-07@14:02:09.725]
[SID=4924a3:39:1385]  (  lgr_stk_mngr)(     36491) !! [ERROR]  SIPAppEngine::NewSBCCallArrived - CMR process FAILED [Time:26-07@14:02:09.725]
LL_END       |SBC       |81086811120720180313l@192.168.20.62              |4924a3:39:1385          |RMT  |192.168.20.62       |5060         |192.168.20.253      |5070
ime:26-07@14:02:09.726]
[SID=4924a3:39:1385]  (       lgr_sbc)(     36492)   (#63) SBCRoutesIterator Deallocated. [Time:26-07@14:02:09.726]
[SID=4924a3:39:1385]  (     sip_stack)(     36493)   TransactionUserMngr::HandleNewSIPMessage - Incoming request is rejected [Time:26-07@14:02:09.727]
[SID=4924a3:39:1385]  (     sip_stack)(     36494)   ---- Incoming SIP Message from 192.168.20.62:5060 to SIPInterface #1 (SBC) UDP TO(#1) ---- [Time:26-07@14:02:09.727]
[SID=4924a3:39:1385]  INVITE sip:9999@192.168.20.253;user=phone SIP/2.0
                      Via: SIP/2.0/UDP 192.168.20.62:5060;branch=z9hG4bKac810878447
                      Max-Forwards: 70
                      From: <sip:601@192.168.20.62>;tag=1c810868685
                      To: <sip:9999@192.168.20.253;user=phone>
                      Call-ID: 81086811120720180313l@192.168.20.62
                      CSeq: 1 INVITE
                      Contact: <sip:601@192.168.20.62:5060>
                      Supported: em,100rel,timer,replaces,path,resource-priority,sdp-anat
                      Allow: REGISTER,OPTIONS,INVITE,ACK,CANCEL,BYE,NOTIFY,PRACK,REFER,INFO,SUBSCRIBE,UPDATE
                      User-Agent: MP-114 FXS_FXO/v.6.60A.342.003
                      Content-Type: application/sdp
                      Content-Length: 428

                      v=0
                      o=AudiocodesGW 810840154 810840153 IN IP4 192.168.20.62
                      s=Phone-Call
                      c=IN IP4 192.168.20.62
                      t=0 0
                      m=audio 6020 RTP/AVP 8 96
                      a=ptime:20
                      a=sendrecv
                      a=rtpmap:8 PCMA/8000
                      a=rtpmap:96 telephone-event/8000
                      a=fmtp:96 0-15
                      m=image 6022 udptl t38
                      a=sendrecv
                      a=T38FaxVersion:0
                      a=T38MaxBitRate:14400
                      a=T38FaxMaxBuffer:1024
                      a=T38FaxMaxDatagram:238
                      a=T38FaxRateManagement:transferredTCF
                      a=T38FaxUdpEC:t38UDPRedundancy
                       [Time:26-07@14:02:09.727]
[SID=4924a3:39:1385]  (     sip_stack)(     36495)   New SIPMessage created - (#70) [Time:26-07@14:02:09.727]
[SID=4924a3:39:1385]  (      lgr_flow)(     36496)   (#0)IDSMngr <- (#0): IDSReportEvent [Time:26-07@14:02:09.728]
[SID=4924a3:39:1385]  (      lgr_flow)(     36497)   (#0)MessageRejectionMngr <- (#0): RequestTerminationEvent [Time:26-07@14:02:09.728]
[SID=4924a3:39:1385]  (     sip_stack)(     36498)   New SIPMessage created - (#69) [Time:26-07@14:02:09.729]
[SID=4924a3:39:1385]  (     sip_stack)(     36499)   ---- Outgoing SIP Message to 192.168.20.62:5060 from SIPInterface #1 (SBC) UDP TO(#1) ---- [Time:26-07@14:02:09.736]
[SID=4924a3:39:1385]  SIP/2.0 500 Server Internal Error
                      Via: SIP/2.0/UDP 192.168.20.62:5060;branch=z9hG4bKac810878447
                      From: <sip:601@192.168.20.62>;tag=1c810868685
                      To: <sip:9999@192.168.20.253;user=phone>;tag=1c2102876080
                      Call-ID: 81086811120720180313l@192.168.20.62
                      CSeq: 1 INVITE
                      Reason: SIP ;cause=500 ;text="Classification Failure"
                      Content-Length: 0
```

Calls Failing at Carrier:

If calls are successfully passing through the SBC, check the syslog messages to verify the call is reaching the correct destination and look for a response from the server indicating why the call may have failed.





Check to make sure that the number is in the correct format that the carrier would like to see. If necessary, create outbound manipulations to put the number in the correct format (i.e. E164).