Conference Call Issue

Wednesday, October 30, 2013
2:49 PM

Subject: Redirecting two bridged lines via AMI causes race condition, preventing the second redirect from working properly.

Decription: The scenario is agent is connected to a customer in a queue. Agent redirects customer to ConfBridge via AMI command in application. Agent calls third party and connects., the two channels are bridged. We execute two AMI redirects to put the agent and 3rd party into the ConfBridge that the customer is sitting in. This will happen successfully at times but other times we will have the following issue. The first redirect call will put the 3rd party in the ConfBridge, and the second redirect will hang up the Agent and proceed to dial the 3rd party phone number again. When this happens we have to hang up on this extra DIAL attempt and put ourselves back in the ConfBridge to get things back to where we should be.

We are upgrading from Asterisk 10 to Asterisk 11. This issue happens on 11.5.1 but does not happen in 10.11.

I attached a PDF file that will explain what I have found.

Attached files:

Issuesample --> This shows the DEBUG for a call that has this issue.
nonIssuesample --> This shows the DEBUG for a call that does **not** have this issue.
fullGBBGB --> This shows 5 calls in a row where the first and fourth scenarios are good and the rest are bad.

Issue in Summary

I think the is when the "channel.c: Bridge stops …." happens in comparison to when the "Redirect" is ran.
If the "redirect" runs before the bridge stops, everything works as expected. However, if the redirect is ran after the bridge stops then we have the issue.
In a nutshell, I believe the bridge is being disconnected (or put into a non-existent state) before the second redirect can be run.

Here is a snippet out of the "**good**" case scenarios in the DEBUG files attached

- Notice the Running action 'Redirect' is performed before the highlighted entries that indicate a bridge stops bridged channels.

[Oct 31 15:21:53] DEBUG[2529] manager.c: Running action 'Setvar'
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[Oct 31 15:21:53] DEBUG[2529] manager.c: Running action 'Setvar'
[Oct 31 15:21:53] DEBUG[2529] manager.c: Running action 'Redirect'
[Oct 31 15:21:53] DEBUG[2529] channel.c: Soft-Hanging up channel 'Agent/1127'
[Oct 31 15:21:53] DEBUG[2876][C-00000034] channel.c: Bridge stops because we're zombie or need a soft hangup: c0=Agent/1127, c1=AsyncGoto/SIP/hq-ast-005-trunk-0000002a<ZOMBIE>, flags: No,Yes,Yes,Yes
[Oct 31 15:21:53] DEBUG[2876][C-00000034] channel.c: Bridge stops bridging channels Agent/1127 and AsyncGoto/SIP/hq-ast-005-trunk-0000002a<ZOMBIE>
[Oct 31 15:21:53] DEBUG[2876][C-00000034] channel.c: Hanging up channel 'AsyncGoto/SIP/hq-ast-005-trunk-0000002a<ZOMBIE>'
[Oct 31 15:21:53] DEBUG[2529] manager.c: Examining event:
Event: Bridge
Privilege: call,all
Bridgestate: Unlink
Bridgetype: core
Channel1: Agent/1127
Channel2: AsyncGoto/SIP/hq-ast-005-trunk-0000002a<ZOMBIE>
Uniqueid1: 1383250905.89
Uniqueid2: 1383250906.90
CallerID1:
CallerID2:

Here is a snippet out of the "**bad**" case scenarios in the DEBUG files attached

- Notice the Running action 'Redirect' is performed well after the highlighted entries that indicate a bridge stops bridged channels.

[Oct 31 15:22:27] DEBUG[2529] manager.c: Running action 'Setvar'
[Oct 31 15:22:27] DEBUG[2529] manager.c: Running action 'Setvar'
[Oct 31 15:22:27] DEBUG[2529] manager.c: Running action 'Setvar'
[Oct 31 15:22:27] DEBUG[2529] manager.c: Running action 'Setvar'
[Oct 31 15:22:27] DEBUG[2529] manager.c: Running action 'Setvar'
[Oct 31 15:22:27] DEBUG[2883][C-00000037] channel.c: Bridge stops because we're zombie or need a soft hangup: c0=Agent/1127, c1=AsyncGoto/SIP/hq-ast-005-trunk-0000002c<ZOMBIE>, flags: No,No,Yes,Yes
[Oct 31 15:22:27] DEBUG[2883][C-00000037] res_rtp_asterisk.c: Setting the marker bit due to a source update
[Oct 31 15:22:27] DEBUG[2883][C-00000037] channel.c: Bridge stops bridging channels Agent/1127 and AsyncGoto/SIP/hq-ast-005-trunk-0000002c<ZOMBIE>
[Oct 31 15:22:27] DEBUG[2529] manager.c: Examining event:
Event: Bridge
Privilege: call,all
Bridgestate: Unlink
Bridgetype: core
Channel1: Agent/1127
Channel2: AsyncGoto/SIP/hq-ast-005-trunk-0000002c<ZOMBIE>
Uniqueid1: 1383250940.94
Uniqueid2: 1383250940.95
CallerID1:
CallerID2:

[Oct 31 15:22:27] DEBUG[2884][C-00000038] pbx.c: Result of 'TRANSFER' is '1383250923.92'
[Oct 31 15:22:27] DEBUG[2884][C-00000038] pbx.c: Launching 'Set'
[Oct 31 15:22:27] VERBOSE[2884][C-00000038] pbx.c: -- Executing [18@xfer:1] Set("SIP/hq-ast-005-trunk-0000002c", "CDR(accountcode)=1383250923.92") in new stack
[Oct 31 15:22:27] DEBUG[2529] manager.c: Examining event:
Event: NewAccountCode
Privilege: call,all
Channel: SIP/hq-ast-005-trunk-0000002c
Uniqueid: 1383250947.96
AccountCode: 1383250923.92
OldAccountCode:

[Oct 31 15:22:27] DEBUG[2884][C-00000038] pbx.c: Launching 'Set'
[Oct 31 15:22:27] VERBOSE[2884][C-00000038] pbx.c: -- Executing [18@xfer:2] Set("SIP/hq-ast-005-trunk-0000002c", "CDR(transfertype)=3way") in new stack
[Oct 31 15:22:27] DEBUG[2884][C-00000038] pbx.c: Result of 'CEID' is 'V03'
[Oct 31 15:22:27] DEBUG[2884][C-00000038] pbx.c: Launching 'Set'
[Oct 31 15:22:27] VERBOSE[2884][C-00000038] pbx.c: -- Executing [18@xfer:3] Set("SIP/hq-ast-005-trunk-0000002c", "CDR(ast_id)=V03") in new stack
[Oct 31 15:22:27] DEBUG[2884][C-00000038] pbx.c: Result of 'UNIQUEID' is '1383250947.96'
[Oct 31 15:22:27] DEBUG[2884][C-00000038] pbx.c: Result of 'CALLERTYPE' is 'THIRDPARTY'
[Oct 31 15:22:27] DEBUG[2884][C-00000038] pbx.c: Result of 'ROOM' is '1127-151659'
[Oct 31 15:22:27] DEBUG[2884][C-00000038] pbx.c: Result of 'AGENT' is '1127'
[Oct 31 15:22:27] DEBUG[2884][C-00000038] pbx.c: Result of 'TRANSFER' is '1383250923.92'
[Oct 31 15:22:27] DEBUG[2884][C-00000038] pbx.c: Result of 'ACTION' is '3'
[Oct 31 15:22:27] DEBUG[2884][C-00000038] pbx.c: Result of 'MANDATACDR' is ''
[Oct 31 15:22:27] DEBUG[2884][C-00000038] pbx.c: Launching 'UserEvent'
[Oct 31 15:22:27] VERBOSE[2884][C-00000038] pbx.c: -- Executing [18@xfer:4] UserEvent("SIP/hq-ast-005-trunk-0000002c",
"Conference,DEPRUniqueID:1383250947.96,CALLERTYPE:THIRDPARTY,Room:1127-151659,Agent:1127,OrigUniqueID:1383250923.92,ActionID:3,ManDataCDR:") in new stack
[Oct 31 15:22:27] DEBUG[2529] manager.c: Examining event:
Event: UserEvent
Privilege: user,all
UserEvent: Conference
Uniqueid: 1383250947.96
DEPRUniqueID:1383250947.96
CALLERTYPE:THIRDPARTY
Room:1127-151659
Agent:1127
OrigUniqueID:1383250923.92
ActionID:3
ManDataCDR:

[Oct 31 15:22:27] DEBUG[2884][C-00000038] pbx.c: Result of 'ROOM' is '1127-151659'
[Oct 31 15:22:27] DEBUG[2884][C-00000038] pbx.c: Launching 'ConfBridge'
[Oct 31 15:22:27] VERBOSE[2884][C-00000038] pbx.c: -- Executing [18@xfer:5] ConfBridge("SIP/hq-ast-005-trunk-0000002c", "1127-151659") in new stack
[Oct 31 15:22:27] DEBUG[2884][C-00000038] app_confbridge.c: Trying to find conference bridge '1127-151659'
[Oct 31 15:22:27] DEBUG[2884][C-00000038] bridging.c: Removed channel 0x9d748cc from bridge array on 0x9a295e4, new count is 0
[Oct 31 15:22:27] VERBOSE[2884][C-00000038] res_musiconhold.c: -- Stopped music on hold on SIP/hq-ast-005-trunk-0000002b
[Oct 31 15:22:27] DEBUG[2884][C-00000038] res_musiconhold.c: Destroying MOH class 'default'
[Oct 31 15:22:27] DEBUG[2884][C-00000038] channel.c: Scheduling timer at (0 requested / 0 actual) timer ticks per second
[Oct 31 15:22:27] DEBUG[2529] manager.c: Examining event:
Event: MusicOnHold
Privilege: call,all
State: Stop
Channel: SIP/hq-ast-005-trunk-0000002b
UniqueID: 1383250923.92

[Oct 31 15:22:27] DEBUG[2884][C-00000038] bridging.c: Added channel SIP/hq-ast-005-trunk-0000002b(0x9d748cc) to bridge array on 0x9a295e4, new count is 1
[Oct 31 15:22:27] DEBUG[2884][C-00000038] confbridge/conf_state.c: Changing conference '1127-151659' state from SINGLE to MULTI
[Oct 31 15:22:27] DEBUG[2529] manager.c: Examining event:
Event: ConfbridgeJoin
Privilege: call,all
Channel: SIP/hq-ast-005-trunk-0000002c
Uniqueid: 1383250947.96
Conference: 1127-151659
CallerIDnum: 19
CallerIDname: <unknown>

[Oct 31 15:22:27] DEBUG[2884][C-00000038] bridging.c: Joining bridge channel 0x9c491c4 to bridge 0x9a295e4
[Oct 31 15:22:27] DEBUG[2884][C-00000038] bridging.c: Added channel SIP/hq-ast-005-trunk-0000002c(0x9dbffcc) to bridge array on 0x9a295e4, new count is 2
[Oct 31 15:22:27] DEBUG[2884][C-00000038] bridging.c: Bridge technology softmix wants to read any of formats (slin) but channel has ulaw
[Oct 31 15:22:27] DEBUG[2884][C-00000038] channel.c: Set channel SIP/hq-ast-005-trunk-0000002c to read format slin
[Oct 31 15:22:27] DEBUG[2884][C-00000038] bridging.c: Bridge 0x9a295e4 put channel SIP/hq-ast-005-trunk-0000002c into read format slin
[Oct 31 15:22:27] DEBUG[2884][C-00000038] bridging.c: Bridge technology softmix wants to write any of formats (slin) but channel has ulaw
[Oct 31 15:22:27] DEBUG[2884][C-00000038] channel.c: Set channel SIP/hq-ast-005-trunk-0000002c to write format slin
[Oct 31 15:22:27] DEBUG[2884][C-00000038] bridging.c: Bridge 0x9a295e4 put channel SIP/hq-ast-005-trunk-0000002c into write format slin
[Oct 31 15:22:27] DEBUG[2884][C-00000038] bridging.c: Giving bridge technology softmix notification that 0x9c491c4 is joining bridge 0x9a295e4
[Oct 31 15:22:27] DEBUG[2884][C-00000038] dsp.c: Setup tone 1100 Hz, 500 ms, block_size=160, hits_required=21
[Oct 31 15:22:27] DEBUG[2884][C-00000038] dsp.c: Setup tone 2100 Hz, 2600 ms, block_size=160, hits_required=116
[Oct 31 15:22:27] DEBUG[2529] manager.c: Running action 'Setvar'
[Oct 31 15:22:27] DEBUG[2529] manager.c: Running action 'Redirect'
[Oct 31 15:22:27] DEBUG[2529] channel.c: Soft-Hanging up channel 'Agent/1127'

Here is what the "bad" case scenario looks like from the CLI output.

Agent goes into conference room.
Agent dials 3rd party and connects.
The first redirect is executed correctly in green.
The second redirect does not show on the CLI screen but instead its running (xfer,19,18) DIAL again.

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   -- Executing [8@amdoutside:24] Queue("SIP/hq-ast-005-trunk-0000000d", "TEST,ntT,,,3}") in new stack
  == Begin MixMonitor Recording SIP/hq-ast-005-trunk-0000000d
   -- Started music on hold, class 'silence', on SIP/hq-ast-005-trunk-0000000d
   -- Stopped music on hold on SIP/s9s003-00000006
   -- agent_call, call to agent '1127' call on 'SIP/s9s003-00000006'
   -- <SIP/s9s003-00000006> Playing 'beep.gsm' (language 'en')
   -- Called Agent/1127
   -- Agent/1127 answered SIP/hq-ast-005-trunk-0000000d
   -- Stopped music on hold on SIP/hq-ast-005-trunk-0000000d
   -- Started music on hold, class 'silence', on SIP/s9s003-00000006
  == Spawn extension (xfer, 18, 1) exited non-zero on 'SIP/hq-ast-005-trunk-0000000d'
   -- Executing [18@xfer:1] Set("SIP/hq-ast-005-trunk-0000000d", "CDR(accountcode)=1383162094.22") in new stack
   -- Executing [18@xfer:2] Set("SIP/hq-ast-005-trunk-0000000d", "CDR(transfertype)=3way") in new stack
   -- Executing [18@xfer:3] Set("SIP/hq-ast-005-trunk-0000000d", "CDR(ast_id)=V03") in new stack
   -- Executing [18@xfer:4] UserEvent("SIP/hq-ast-005-trunk-0000000d", "Conference,DEPRUniqueID:1383162094.22,CALL
ERTYPE:,Room:1127-144159,Agent:1127,OrigUniqueID:1383162094.22,ActionID:2,ManDataCDR:") in new stack
   -- Executing [18@xfer:5] ConfBridge("SIP/hq-ast-005-trunk-0000000d", "1127-144159") in new stack
   -- Started music on hold, class 'default', on SIP/hq-ast-005-trunk-0000000d
   -- Stopped music on hold on SIP/s9s003-00000006
   -- agent_call, call to agent '1127' call on 'SIP/s9s003-00000006'
   -- <SIP/s9s003-00000006> Playing 'beep.gsm' (language 'en')
      > Channel Agent/1127 was answered
   -- Executing [19@xfer:1] Set("Agent/1127", "CDR(accountcode)=") in new stack
   -- Executing [19@xfer:2] UserEvent("Agent/1127", "ConfCallOut,UniqueID:1383162122.24,Room:1127-144159,Dialed:6126706994") in new stack
   -- Executing [19@xfer:3] GotoIf("Agent/1127", "1?eight:") in new stack
   -- Goto (xfer,19,18)
   -- Executing [19@xfer:18] Dial("Agent/1127", "SIP/hq-ast-005-trunk/6126706994,,tT") in new stack
  == Using SIP RTP CoS mark 5
   -- Called SIP/hq-ast-005-trunk/6126706994
   -- SIP/hq-ast-005-trunk-0000000e is making progress passing it to Agent/1127
      > 0x9311398 -- Probation passed - setting RTP source address to 192.168.168.44:11536
   -- SIP/hq-ast-005-trunk-0000000e is ringing
   -- SIP/hq-ast-005-trunk-0000000e answered Agent/1127
      > 0x9311398 -- Probation passed - setting RTP source address to 192.168.168.44:11536
   -- Executing [18@xfer:1] Set("SIP/hq-ast-005-trunk-0000000e", "CDR(accountcode)=1383162094.22") in new stack
   -- Executing [18@xfer:2] Set("SIP/hq-ast-005-trunk-0000000e", "CDR(transfertype)=3way") in new stack
   -- Executing [18@xfer:3] Set("SIP/hq-ast-005-trunk-0000000e", "CDR(ast_id)=V03") in new stack
   -- Executing [18@xfer:4] UserEvent("SIP/hq-ast-005-trunk-0000000e", "Conference,DEPRUniqueID:1383162129.26,CALL
ERTYPE:THIRDPARTY,Room:1127-144159,Agent:1127,OrigUniqueID:1383162094.22,ActionID:3,ManDataCDR:") in new stack
   -- Executing [18@xfer:5] ConfBridge("SIP/hq-ast-005-trunk-0000000e", "1127-144159") in new stack
   -- Stopped music on hold on SIP/hq-ast-005-trunk-0000000d
  == Spawn extension (xfer, 19, 18) exited non-zero on 'Agent/1127'
   -- Executing [19@xfer:18] Dial("Agent/1127", "SIP/hq-ast-005-trunk/6126706994,,tT") in new stack     --Notice here that (xfer,19,18) got executed out of the blue.  I did not redirect to extension 19 at this point.
  == Using SIP RTP CoS mark 5
   -- Called SIP/hq-ast-005-trunk/6126706994
   -- SIP/hq-ast-005-trunk-0000000f is making progress passing it to Agent/1127
      > 0x9333ba8 -- Probation passed - setting RTP source address to 192.168.168.44:11148
   -- SIP/hq-ast-005-trunk-0000000f is ringing
hq-ast-v03*CLI> exit
```

Here is my Extensions.conf (xfer Context)

- When redirecting two bridged channels to a ConfBridge, I redirect the channels to exten 18, priority 1 in the "XFER" context.

```
; This context is needed for the transfering capabilities of the CE work station.

[xfer]
exten => 18,1,Set(CDR(accountcode)=${TRANSFER})
exten => 18,n,Set(CDR(transfertype)=3way)
exten => 18,n,Set(CDR(ast_id)=${CEID})
exten => 18,n,Userevent(Conference,DEPRUniqueID:${UNIQUEID},CALLERTYPE:${CALLERTYPE},Room:${ROOM},Agent:${AGENT},OrigUniqueID:${TRANSFER},ActionID:
${ACTION},ManDataCDR:${MANDATACDR})
exten => 18,n,ConfBridge(${ROOM})

exten => 19,1,Set(CDR(accountcode)=${OUTBOUND})
exten => 19,n,Userevent(ConfCallOut,UniqueID:${UNIQUEID},Room:${ROOM},Dialed:${DIALNUMBER})
exten => 19,n,Gotoif($[${LEN(${CHANPREFIX})}>0]?eight:)
exten => 19,n,Gotoif($[${DIALNUMBER:0:3}=800]?toll:)
exten => 19,n,Gotoif($[${DIALNUMBER:0:3}=811]?toll:)
exten => 19,n,Gotoif($[${DIALNUMBER:0:3}=822]?toll:)
exten => 19,n,Gotoif($[${DIALNUMBER:0:3}=833]?toll:)
exten => 19,n,Gotoif($[${DIALNUMBER:0:3}=844]?toll:)
exten => 19,n,Gotoif($[${DIALNUMBER:0:3}=855]?toll:)
exten => 19,n,Gotoif($[${DIALNUMBER:0:3}=866]?toll:)
```

```
exten => 19,n,Gotoif($[${DIALNUMBER:0:3}=877]?toll:)
exten => 19,n,Gotoif($[${DIALNUMBER:0:3}=888]?toll:)
exten => 19,n,Gotoif($[${DIALNUMBER:0:3}=899]?toll:)
exten => 19,n,Dial(${OUTBOUNDTRUNK}${DIALNUMBER},,tT)
exten => 19,n,Hangup()
exten => 19,n(toll),Dial(${LOCALT1TRUNK}/1${DIALNUMBER},,tT)
exten => 19,n,Hangup()
exten => 19,n(eight),Dial(${CHANPREFIX}${DIALNUMBER},,tT)
exten => 19,n,Hangup()
```