

Osmocom Analog - Feature #4971

OsmoCC Socket Documentation

01/22/2021 10:28 PM - foxrf

Status:	New	Start date:	01/22/2021
Priority:	Immediate	Due date:	
Assignee:	jolly	% Done:	0%
Category:			
Target version:			
Spec Reference:			

Description

It appears that the MNCC socket has been replaced with OSMOCC which does not appear to have ANY documentation around how you are actually supposed to configure call routing with this thing. More specially the osmo-cc-router package attempts to call a bash script "routing.sh" for which there is no sample or documentation going over the syntax. Please add full documentation on how this bash script is supposed to route calls and or provide a detailed example script.

History

#1 - 01/24/2021 10:52 PM - laforge

- Assignee set to jolly

#2 - 01/24/2021 10:54 PM - laforge

I've assigned this to [jolly](#), as he's really the only one (AFAICT) working on osmocom-analog.

As for OsmoCC, there had been an early "concept paper" that jolly shared among the Osmocom developers, but I'm not sure how accurate it still is, and whether he intended it to be publicly distributed.

#3 - 02/12/2021 11:02 AM - atmega

Hey, did you managed to do anyprogress? What OSMO-CC should connect to ?

#4 - 02/12/2021 11:08 AM - atmega

- File IMG_3821.jpg added

I am stuck at socket failed ;-(

#5 - 02/13/2021 07:01 PM - foxrf

On 2021-02-12 05:08, atmega [REDMINE] wrote:

Issue [#4971](#) has been updated by atmega.

File IMG_3821.jpg added

I am stuck at socket failed ;-(

#6 - 02/27/2021 05:49 AM - aldude999

It looks like the routing script needs to echo commands based on the calling number/called number on what to do on what call. Taking a look at call.c under the routing_receive_stdout() function seems to show a list of possible commands. I was able to get it to respond to my calls with the "call cannot be completed as dialed" response by placing the following in the routing.sh script:

```
#!/bin/sh
echo "call interface :"
```

Then set the permissions on the routing script to 775. I'll have to play with some of the endpoint software to see if it gets me any further. It looks like more complex scripts could be built by comparing the arguments osmo-cc-router sends the scripts, it looks like it sends the caller/called number as arguments.

#7 - 02/27/2021 09:06 AM - aldude999

So after fiddling around with the software a bit more, I managed to get a call going through SIP, you have to compile osmo-cc-sip-endpoint, osmo-cc-router isn't actually needed at all. I set up an Asterisk server, but any SIP server should work. I made an account for the cell phone using its phone number as the extension.

One thing I did notice was that if you tried to make a call from a sip client to your analog phone, something along the way drops the phone number you want to ring. Here's the command I ran to get the sip endpoint working:

```
osmo-cc-sip-endpoint -r [SIP_SERVER_IP] -I [OSMOCOM-ANALOG_IP] -A [PHONENUMBER] [SIP_PASSWORD] [DOMAIN/IP] --register [PHONENUMBER]@[DOMAIN/IP] --cc 'screen-called-in "" "[PHONENUMBER]"'
```

The screen-called-in replaces the weird null phone number into the actual phone you want to ring.

Hope this helps out a bit!

#8 - 03/04/2021 10:00 PM - mstamm

- File cnetz.txt added

- File osmo-cc-router.txt added

I am stuck at a somewhat similar problem. I'm trying to set up osmoccom-analog using a LimeSDR and two C-Netz mobiles, and - eventually - I want to implement a SIP connection with a PBX. Using the command line above, I can successfully register osmo-cc-sip-endpoint against my SIP PBX, but I am unable to make or receive any calls, even within osmoccom-analog itself.

Theoretically, osmoccom-analog with the option -x should be able to connect calls from one mobile to another. Both mobiles register successfully, but any number I dial from either mobile always connects the mobile to the waiting/queue music. The call never connects. The numbers are correct, and both mobiles successfully register against osmoccom-analog.

I also tried osmo-cc-router, with the adapted routing.sh above, but the result was the same.

I simply do not know where to look next - I'm attaching the console output for cnetz and for osmo-cc-router.

I believe before I can address the issue of connecting through SIP, I must find out why even osmoccom-analog itself does not connect one mobile to another despite the presence of one OgK and two SpK channels.

Can anyone give me a hint as to where to look from the console output, or is there any other diagnostic I could run? This is one of the unnerving issues where theroretically, everything should work as designed, and actually, it does not.

Thank you in advance!

#9 - 03/04/2021 11:43 PM - aldude999

mstamm wrote:

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Thank you in advance!

I had to run an instance of osmo-cc-sip-endpoint for each phone on the network in a different console. Presumably at some point, osmo-cc-router would be what would set that up for you, but as we don't know how to use it yet, the sip-endpoint has to be set up manually. I also had to change the local port of the osmo-cc-sip-endpoint as well. This was done by changing for instance, -I 127.0.0.1:4203 (the port can be set to any port not in use.), and setting the phone number in the other arguments to the one you want that instance to route calls for. I found osmoccom-analog was able to detect new instances of osmo-cc-sip-endpoints, at least when done locally.

As much as I've looked into the osmo-cc-router, at the moment, it seems useless without either some modifications or some instructions, but I did manage to get things working with osmo-cc-sip-endpoint alone.

#10 - 03/05/2021 12:36 AM - atmega

I am trying to make my osmocom analog NMT to connect to osmo-cc-sip-endpoint using this scheme :

```
osmo-cc-sip-endpoint -r [SIP_SERVER_IP] -I [OSMOCOM-ANALOG_IP] -A [PHONENUMBER] [SIP_PASSWORD] [DOMAIN/IP] --register [PHONENUMBER]@[DOMAIN/IP] --cc 'screen-called-in "" "[PHONENUMBER]"'
```

but all what I get is :

```
tport_vsend(0x7fd1940061f0): Invalid argument with (s=10 */213.218.117.66:5060)
nta: REGISTER (955973299): Invalid argument (22) with */[213.218.117.66]:5060
sip.c: 921 info : Received REGISTER response: 503 Service Unavailable (registration)
```

do I need to deal more with Asterisk or asterisk is not needed? What could be the reason that osmo-cc-sip-endpoint does not login to my sip phone provider? what I do wrong?

anybody could help directly ? please message me on atmega@wp.pl

I dream to call PSTN from my NMT phone :D

#11 - 03/05/2021 07:52 AM - mstamm

aldude999 wrote:

As much as I've looked into the osmo-cc-router, at the moment, it seems useless without either some modifications or some instructions, but I did manage to get things working with osmo-cc-sip-endpoint alone.

Thank you for this explanation. I believe I will park this issue for the moment until I can get the routing of calls directly within osmocom-analog from one C-Netz mobile to another sorted out. I believe there is something fundamentally wrong with my setup, or something I'm just missing, although theoretically, this should work just fine. Calls between mobiles both registered with osmocom-analog should be possible without any other components outside osmocom-analog just by setting the -x command line argument and allocating at least two SpK channels in addition to the OgK channel, and I cannot even get this basic setup to work. I can call a mobile from the console, and the call will be signalled. If I answer the call, I get the "holding queue music". Likewise, if I place a call from one mobile to the other, using its registered number, I only get connected to the holding queue music, and the other mobile does not even ring. The console output is also somewhat unclear to me - at least I cannot readily understand what could be causing the issue. I will register a separate issue and post some more logs, in the hope that someone may help me catch what I'm missing.

#12 - 03/05/2021 05:58 PM - atmega

Is there anybody who could help please? :-)

#13 - 03/06/2021 01:48 PM - laforge

Is there anybody who could help please? :-)

I think aside from jolly nobody in the "Typical osmocom developer team" has been working with OsmoCC so far. And he doesn't seem very active here on the issue tracker. Did you try reaching out directly by e-mail?

#14 - 03/06/2021 02:43 PM - atmega

Yes, I did try. So far no answer ;-(I will wait :-) thank you for your effort :-) Still being excited to call from my NMT phone :-) Regards Jacek

Dnia 6 marca 2021 14:49 laforge [REDMINE] <redmine@lists.osmocom.org> napisał(a):

Issue [#4971](#) has been updated by laforge.

Is there anybody who could help please? :-)

I think aside from jolly nobody in the "Typical osmocom developer team" has been working with OsmoCC so far. And he doesn't seem very active here on the issue tracker. Did you try reaching out directly by e-mail?

Feature [#4971](#): OsmoCC Socket Documentation

- Author: foxrf * Status: New * Priority: Immediate * Assignee: jolly * Category: * Target version: * Spec Reference:

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---Files-----

Screenshot from 2021-01-22 16-20-23.png (175 KB)

IMG_3821.jpg (928 KB)

cnetz.txt (36.7 KB)

osmo-cc-router.txt (7.7 KB)

#15 - 03/07/2021 12:44 PM - atmega

I wonder what do I do wrong :

```
sip.c:1430 debug : Event 29 from SIP stack received (handle=0x55cb861d7980)
sip.c: 921 info : Received REGISTER response: 401 Unauthorized (registration)
sip.c: 943 debug : Register challenge received
sip.c: 178 debug : challenge order received
sip.c: 194 notice : No authentication header found
sip.c: 954 debug : Register failed, starting register timer
sip.c: 956 debug : destroying nua_handle 0x55cb861d7980 (register)
```

using this command line:

```
osmo-cc-sip-endpoint -r mysipnumber@mysipserverip -l localPosmocom-analog -A SipAccountUser SipAccountPass --remote-auth --register
SIPAccountUser@SipServerIp --cc 'screen-called-in "" "myNMTphone"' -P myPublicIP -v 0
```

Any ideas?

#16 - 03/07/2021 10:25 PM - atmega

Guys! I did it!!! My NMT Benefon called from PSTN Voip number. It WORKS. Now working to make outgoing call! Regards to all!

Dnia 6 marca 2021 15:36 atmega@wp.pl <atmega@wp.pl> napisal(a):

Yes, I did try. So far no answer ;-(I will wait :-) thank you for your effort :-) Still being excited to call from my NMT phone :-) Regards Jacek

Dnia 6 marca 2021 14:49 laforge [REDMINE] <redmine@lists.osmocom.org> napisal(a):

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#17 - 03/07/2021 11:04 PM - mstamm

atmega wrote:

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Congratulations!

I made some progress on cnetz and SIP today as well - incoming calls to the SIP account are signalled at the correct mobile, but when I answer the call on my mobile, the connection to the SIP server is not made - the SIP line continues ringing, and there is no connection made. Did you have to use any specific commands or switches, or did this issue not occur in your setup?

#18 - 03/07/2021 11:20 PM - atmega

When I call my public VOIP number osmo-cc-sip-endpoint calls pointed NMT phone over osmocom-analog osmo-cc and when I answer it I can talk! :D At the moment I am working with proper configuration to make outgoing call as somewhere I am missing authentication for VOIP outgoing calls and get SIP respond 403 followed by 407. Will work on it - perhaps tomorrow will post my logs and commands that did worked so far! I am so happy! Dreams come true! Even CallerID works! My Benefon Delta works great! Respect to Jolly!

Dnia 8 marca 2021 00:04 mstamm [REDMINE] <redmine@lists.osmocom.org> napisal(a):

Issue [#4971](#) has been updated by mstamm.

atmega wrote:

Guys! I did it!!! My NMT Benefon called from PSTN Voip number. It WORKS. Now working to make outgoing call! Regards to all!

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#19 - 03/13/2021 01:06 AM - aldude999

I had to make sure the encoding was either Alaw or ulaw, I'm not sure if any other codecs are supported. You can use -D on osmo-cc-sip-endpoint to output more SIP debug info. There's a lot of text, but you can output all the debug text to a file by adding:

2> out.log

to the end of the command.

I also found that NAT settings are extremely important for SIP, and even on a LAN-LAN network, if you don't have your external IP set correctly on the server, and sometimes even the client, audio doesn't always route. FreePBX has a panel in asterisk settings that let you set the external IP. You have to restart after setting it. the SIP Endpoint also has an argument for setting the public IP address, -P. I didn't have this set right at first and there would be no audio for about 30 seconds and then it would hang up automatically.

mstamm wrote:

atmega wrote:

Guys! I did it!!! My NMT Benefon called from PSTN Voip number. It WORKS. Now working to make outgoing call! Regards to all!

Congratulations!

I made some progress on cnetz and SIP today as well - incoming calls to the SIP account are signalled at the correct mobile, but when I answer the call on my mobile, the connection to the SIP server is not made - the SIP line continues ringing, and there is no connection made. Did you have to use any specific commands or switches, or did this issue not occur in your setup?

#20 - 03/18/2021 09:23 PM - atmega

Short update - I am working last few days on make outgoing call possible.. dozens of attempts...

here is what my command looks like :

```
osmo-cc-sip-endpoint --cc 'screen-calling-in "" ""' --cc 'screen-called-in "" "3666090"' --cc 'screen-calling-out "" "3666090"' --cc 'screen-called-out "" ""' -r MYaccount@mySIPprov.com -l 192.168.1.18 -P myPublicIp -R mySIPuser@mySIPprov.com --remote-register -A mySIPuser mySIPpass mySIPprvider.com
```

All what I get is :

```
screen.c: 388 info : Screening outgoing caller ID '3600700':
screen.c: 403 info : -> type = subscriber
screen.c: 411 info : -> present = allowed
screen.c: 421 info : Comparing with rule #1: '*':
screen.c: 517 info : Rule matches, changing outgoing caller ID to '3666090'.
screen.c: 388 info : Screening outgoing dialed number '567':
screen.c: 391 info : -> type = unknown
screen.c: 421 info : Comparing with rule #1: '*':
screen.c: 517 info : Rule matches, changing outgoing dialed number to '567'.
sip.c: 463 info : Sending INVITE (callref 15)
endpoint.c: 923 info : Handle message CC-PROC-IND at state INIT-OUT (callref 15)
sip.c:1146 info : Received INVITE response: 407 Proxy Authentication Required (INVITE (callref 15)
sip.c:1146 info : Received INVITE response: 403 SIP ID!=AUTH ID (callref 15)
endpoint.c: 923 info : Handle message CC-REL-IND at state PROCEEDING-OUT (callref 15)
endpoint.c: 740 info : Changing message to CC-DISC-IND.
endpoint.c: 923 info : Handle message CC-REL-REQ at state DISCONNECTING-IN (callref 15)
```

Always 403 sip error or 407...

is there a problem that my SIP provider does not use AuthID ?

Please guys help! I am dreaming also to make outgoing call! I did make incoming to my sip landline and it works, no way I can make outgoing call

What I do wrong ???

I post my screens - incoming and outgoing call, please help!

when I use config like this

```
osmo-cc-sip-endpoint -r MYSipAccountUsername@mySIPprov-ip -l (mylocalserverip) -A MYSipAccountUsername MYSipPASS --remote-auth
--register MYSippAccountUsername@mySIPprov-ip --cc 'screen-called-in "mySIPacname" "myNMTphoneNumber-3666090"' --cc 'screen-calling-out
"" ""' --cc 'screen-called-out "" "mysipacname"' -P my publicIP
```

- still I get this - but my incoming calls to my SIP public numer works and diverts to NMT phone

```
sip.c: 921 info : Received REGISTER response: 401 Unauthorized (registration)
sip.c: 921 info : Received REGISTER response: 200 OK (registration)
socket.c: 381 error : OsmoCC-Socket failed, socket cause 3.
endpoint.c: 923 info : Handle message CC-REL-REQ at state ATTACH-SENT (callref 2)
screen.c: 388 info : Screening outgoing caller ID '3666090':
screen.c: 403 info : -> type = subscriber
screen.c: 411 info : -> present = allowed
screen.c: 421 info : Comparing with rule #1: '*':
screen.c: 517 info : Rule matches, changing outgoing caller ID to '3666090'.
screen.c: 388 info : Screening outgoing dialed number "":
screen.c: 391 info : -> type = unknown
screen.c: 421 info : Comparing with rule #1: '*':
screen.c: 517 info : Rule matches, changing outgoing dialed number to 'atmega004'.
sip.c: 637 info : Sending INVITE response: 200 OK (callref 8)
endpoint.c: 923 info : Handle message CC-SETUP-COMP-IND at state CONNECTING-IN (callref 8)
sip.c:1343 info : Received BYE (callref 8)
sip.c:1358 info : Sending BYE response: 200 OK (callref 8)
endpoint.c: 923 info : Handle message CC-REL-IND at state ACTIVE (callref 8)
endpoint.c: 740 info : Changing message to CC-DISC-IND.
endpoint.c: 923 info : Handle message CC-REL-REQ at state DISCONNECTING-IN (callref 8)
```

So incoming calls works - but outgoing not... What do I do wrong?

#21 - 03/23/2021 11:04 PM - aldude999

atmega wrote:

```
osmo-cc-sip-endpoint --cc 'screen-calling-in "" "" --cc 'screen-called-in "" "3666090" --cc 'screen-calling-out "" "3666090" --cc 'screen-called-out "" "" -r MYaccount@mySIPprov.com -l 192.168.1.18 -P myPublicIp -R mySIPuser@mySIPprov.com --remote-register -A mySIPuser mySIPpass mySIPprvider.com
```

That looks like it might be wrong, screen calling out is changing your caller ID to the number you set it to, 3666090. If this doesn't match your SIP phone number (the number that shows up when you make a call from your SIP account), they will most likely block it because it looks like you're trying to spoof your phone number. If that matches your actual number you got from the SIP provider that should be fine though.

403 SIP ID!=AUTH ID

The SIP ID doesn't match the authenticated ID. Could be caused by the issue above or something else that we aren't aware of, such as the program not supporting all the possible SIP configurations (there are many more settings possible in SIP than the arguments in osmocc-sip-endpoint, for most cases it seems to be sufficient), or something weird going on with the provider in that they require non-default SIP options.

Files

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