Troubleshooting Tips for VoIP Measurement

v1.1

To successfully perform the measurement, only 20-30 lines need to be entered in the configuration files, maybe even not that much. Still, there are many kinds of mistakes that can be made, and these are often quite difficult to find. In this document, I describe a few tips that may help.

The point: Let's think it over!

Okay, it does not work. Do not panic, this happens very often. The easiest and fastest way to find the error now is not to search in Google: it is almost certain that the result will not be relevant. It doesn't help much to know what other people's problems were in a similar case: I've helped quite a few students, but I've never seen two identical mistakes. Even more so two identical error messages. So there is no fairy tale, you have to proceed in a nice methodical way.

It's not even certain that there will be just one error, I just counted four errors before the call from MicroSIP to ZoiPer worked.

Local call: what is needed?

The local call does not work, that is, from 300 to 301 and/or vice versa. The first thing to understand is that a local call requires three things to work:

1) One of the SIP clients (further on in this docs: MicroSIP, but there may be others) should be able to register to Asterisk.
2) The other SIP client (further on in this docs: ZoiPer) should be able to register.
3) They should be able to call each other

Let's start with the first point!

MicroSIP registration

If in the lower right corner of the client the phone icon is green and thelabel is Online, then it is good. If not, let's see what could be wrong. First, it is not even certain that they see each other at the IP level. Ping the IP address of OpenWrt from the machine running MicrosSIP. If it works, good. (Actually, if you were able to log in with SSH (e.g. putty), the ping will also work.)

If Ping / putty login / ssh login does not work, you can for example check whether the outside world is visible from OpenWrt. If not, you might have specified the wrong card next to the bridge mode in VirtualBox's network settings.

By the way, in MicroSIP, you can try the registration again by removing the tick from the account name at the top of the drop-down menu on the right, then selecting it again and putting it back.

If you can ping OpenWrt, the SIP messages should get there, too. You can see if this happens, and if so, what the message contains and what the answer is, e.g. with WireShark. Just launching WireShark is not enough. It's tricky, see the beginning of task 3 of the measurement instruction! That way you can already see what's going on, and if you interpret it, you might find out what's wrong.

> For example, I just saw that the REGISTER message goes to OpenWrt, but the virtual machine returns an "ICMP error" packet saying "destination port unreachable". Thinking helped here too. So the registration IP package arrived for the telephone exchange, but no one was waiting for it on the destination port (UDP 5060). How can this be if Asterisk is running? For example, someone changed the SIP port in the config file, but that would be intentional damage. No, the UDP transport was simply not enabled, so Asterisk was not listening on the appropriate UDP port.
>
> Note: I can't know in advance what errors there might be, so I can't write them all down, this is just an example of the suggested way of thinking.

The other way is to let Asterisk help you and tell you what's going on. After running the `asterisk -r` command, the program lists the most important events, such as an incoming registration request and the response to it. You will get a more detailed description, if you start the program with the `asterisk -rvvvvv` command, even if you give the the `pjsip set logger on` command. Of course, this is not enough, it is important to look at carefully and interpret what is displayed! (If you are expecting a lot of messages, it is easier to see where the relevant information begins after the messages have arrived, if you pressing 10-20 Enter ten or twenty times.) These messages are also worth their weight in gold when debugging. For example, it can be found from them that you have mistyped the username.

In Asterisk the `pjsip show endpoints` displays the status of the endpoints, such as whether they are registered.

An other method is to make Asterisk read the `pjsip.conf` és az `extensions.conf` again. Edit and save the file you want to check. It's important to change the date of the file, so even if you haven't changed anything, put a space somewhere and delete it, and then save (or use the Linux touch command). This is necessary because Asterisk will reread only the file with the new date. Start Asterisk, but preferably now without the `v` parameters, just with `-r`. Reread the config files with Asterisk: `core reload`. Then it will print out all the errors found in all the config files. Ignore the following:

```
ERROR[20981]: config_options.c:710 aco_process_config: Unable to load config
file 'cdr.conf'
NOTICE[20987]: sorcery.c:1334 sorcery_object_load: Type 'system' is not
reloadable, maintaining previous values
ERROR[20987]: res_pjsip/config_system.c:261
system_create_resolver_and_set_nameservers: There are no local system
```

```
nameservers configured, resorting to system resolution
WARNING[20981]: pbx.c:8717 ast_context_verify_includes: Context 'local' tries
to include nonexistent context 'parkedcalls'
```

Apart from the four errors above, any other error or warning (ERROR, WARNING, NOTICE) is almost certainly to be taken very seriously!

ZoiPer registration

Of course, it doesn't have to be ZoiPer, it can be any mobile client. Basically, the important ones are described in the previous point here as well.

The phone must be able to ping our OpenWrt. You probably don't have ping on your phone (although you can), but you definitely have a web browser. Enter your OpenWrt IP address, this should bring up the login page. You don't need to log in, just see if it works. If it doesn't work, there is a network problem. If, for example, the phone uses mobile internet and the laptop uses internet which coming into the apartment behind NAT (and which is wired to the router), it probably won't work. If you have WiFi at home, connect your phone to it with!

If there is a network connection, but it was not possible to register, then something was wrong with the login data. Either in the config file or on the mobile. In the latter case, select Settings/Accounts in ZoiPer's ("hamburger") menu (three horizontal lines on the top left corner) and edit the settings a bit. "User name" belongs to the password, and Asterisk searches for the appropriate SIP config in the configuration file based on "Authentication name".

Call

Ha mindkét kliens sikeresen regisztrált, akkor a nehezén e résznek túl is vagyunk. Ha nem megy a hívás, de a regisztrációk igen, akkor esélyes, hogy az `extensions.conf`-ban szúrt el valamit. Ezen a ponton összesen kb. öt sor kell legyen benne, nézze át jól! Az is lehet persze, hogy a `pjsip.conf`-ban rontotta el a context-et, amit ezért nem talál az Asterisk az `extensions.conf` fájlban. A `pjsip.conf`-ban lévő kodekbeállítások elrontása is okozhat hibát persze.

A korábban leírt módon belépve az Asteriskbe az bővebb infókat ír ki a hívás során, amikből azért jó eséllyel rá kell jöjjön, mi a baj.

Trunk registration

The difficulty here is that you only see the home side of things, but not the university side. Until now, you had no such problems, you had the whole system in your hands. One thing is for sure, you must be able to ping `voipmeres.tmit.bme.hu` from both OpenWrt and the machine running the SIP client. If it doesn't work, check if you can ping `www.bme.hu` or `www.tmit.bme.hu`

If you can ping `www.bme.hu`-t but not `voipmeres.tmit.bme.hu` , then see the section on foreign (cross-border) measurement in the measurement instructions! A power outage in the department or building I may also happen, then e.g. `www.bme.hu` is pingeable, but `www.tmit.bme.hu` not.

In the case of trunk configuration, it is also true that if the registration does not go well, then `pjsip.conf` is the suspect (and of course there may also be a problem in the network). If the registration is complete, but the call still does not go through, then both of our configuration files mentioned so far may be faulty.

Inside Asterisk, the `pjsip show registrations` displays the status of the outgoing registrations.

If you have a problem, look for it yourself first! See what Asterisk says when re-reading the config files after updating, what happens in the message exchanges: you can use interactive login to Asterisk or WireShark. Or both. Always think about the results you get, don't try to save on this!

It also happened that a router installed by a service provider (in our case Telekom's Huawei HA35 DSL device) wrote into a SIP packet, we also wrote about this in the measurement instructions.

If it still does not go

If it still doesn't work, feel free to ask the supervisors of the measurement for help! For this, briefly describe what the problem is in the appropriate Teams channel (in the first round, by no means in a message!). Before, of course, try to go through the options above, you can learn the most from them.

For the last task, it is important that the lab supervisor can also login to the server, so that troubleshooting can be even more effective.

Be prepared that the lab supervisor may ask you to explain your problem and answer her questions in a voice call in Teams. You can also be asked to share your screen. However, the camera will probably not be needed.

Good luck!

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