Asterisk CLI useful commands

- Entering CLI
- Entering CLI with additional debugging
- Running commands outside of CLI
- SIP debugging
- RTP Debugging
- Reloading the complete Asterisk configuration
- Restarting the Asterisk
- List peers
- Active calls
- SIP channels list
- Registered trunks
Entering CLI

Before you can see any of the messages in Asterisk CLI, you need to ssh to the system by using ssh command (if using Linux on your computer) or using putty or similar software if on PC/MAC. After that you can enter the Asterisk CLI via following command:

```
[root@Motion-PBX ~]# asterisk -rvvvvv
```

where number of Vs define the verbosity level of the CLI.

Once inside you will see a lot of useful info print out for all actions on the system, Asterisk related though. You will see:

- Phone calls
- Peer registrations
- Subscribe notification
- Reload of system components (Extensions, Trunks, IVRs, etc.)

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=========================================================================
Connected to Asterisk 13.6.0 currently running on Motion-PBX (pid = 2112)
Motion-PBX*CLI>

Entering CLI with additional debugging

If for some reason you have some inexplicable issues, like Asterisk not being able to start, you can try to run the CLI with different set of switches which should give some application specific debug info which includes start up sequence, database connection, registration retries, etc.

```
[root@Motion-PBX ~]# asterisk -rddddd
```

where number of Ds define the verbosity of these debug messages.

Parsing /etc/asterisk/asterisk.conf
Seeding global EID '00:50:56:8e:23:02' from 'eth0' using 'siogifhwaddr'
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=========================================================================
Connected to Asterisk 13.6.0 currently running on Motion-PBX (pid = 2112)
Core debug was OFF and is now 5.

Important

To turn off the debug messages run this command:

```
Motion-PBX*CLI> core set debug off
Core debug was 5 and is now OFF.
```
Running commands outside of CLI

If you don’t need to be inside CLI, or you need just to execute some command without concern of output from CLI, you can do so by running Asterisk command with following switches being used:

```
[root@Motion-PBX ~]# asterisk -rx "reload"
```

Above will reload Asterisk configuration without going into CLI.

SIP debugging

First important command(s) to know is the SIP debug set of commands which are useful when you need to see the SIP data stream going through Asterisk. Simple command is to enable SIP debugging for one phone with:

```
SIP SET DEBUG PEER PEERNAME
```

Motion-PBX*CLI> sip set debug peer giovelmotion
SIP Debugging Enabled for IP: 151.0.175.186

If for some reason the peer is not registered and the IP of the peer is not known to the asterisk, above command will not work and CLI will not show any SIP messages.

In such case, if you know the IP from which traffic should come, it is better to turn on debugging for that specific IP like this:

```
SIP SET DEBUG IP PEER_IP
```

where PEER_IP is the IP address of the peer which should send traffic to said extension/trunk.

When you finish debugging the SIP stream, you need to turn off SIP debugging since leaving that running clutters the CLI output and you might miss other important information on the system.

Important

To turn off SIP debug run this command:

```
Motion-PBX*CLI> sip set debug off
SIP Debugging Disabled
```

RTP Debugging

One of the primary techniques is to view what is actually getting sent and received by VOIP devices.

If you have issues with audio problems, enable the RTP debug

```
Motion-PBX*CLI> rtp set debug on
RTP Debugging Enabled
```

Example of proper bidirectional RTP traffic

```
Got  RTP packet from 151.0.175.186:55903 (type 00, seq 013414, ts 317314000, len 000160)
Sent RTP packet to 82.215.163.137:10498 (type 00, seq 053165, ts 317314000, len 000160)
Got  RTP packet from 82.215.163.137:10498 (type 00, seq 008373, ts 1558588704, len 000160)
Sent RTP packet to 151.0.175.186:55903 (type 00, seq 020095, ts 1558588704, len 000160)
Got  RTP packet from 151.0.175.186:55903 (type 00, seq 013415, ts 317314160, len 000160)
Sent RTP packet to 82.215.163.137:10498 (type 00, seq 053166, ts 317314160, len 000160)
```
Important

To turn off RTP debug run this command:

```
Motion-PBX*CLI> rtp set debug off
RTP Debugging Disabled
```

**CODEC transcoding list**

If for some reason you have issues with audio problems, some of the messages might indicate codec incompatibilities on the system. In such cases you can see the possible translation paths in Asterisk with following command:

```
Motion-PBX*CLI> core show translation
```

This command will show a table of possible codec transcoding/translation paths that can be followed on the system.

When you see a - sign, it means that transcoding between said codecs is not possible. In most cases, the reason for such issue is missing codec.

```
Motion-PBX*CLI> core show translation
  Translation times between formats (in microseconds) for one second of data
  Source Format (Rows) Destination Format (Columns)
            ulaw  alaw  gsm  g726  g726aal2  adpcm  slin  slin  slin  slin  slin  slin  slin  slin  slin  slin  lpc10
  ilbc       9150 15000 15000
  ulaw  - 9150 15000 15000 15000 15000 15000 17000 17000 17000 17000 17000 17000 17000 17000 17000 17000
  15000 17250 15000
  15000 17250 15000
  15000 17250 15000
  15000 17250 15000
  15000 17250 15000
  g726aal2 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000
  adpcm 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000
  slin  6000  6000  6000  6000  6000  6000  6000  6000  6000  6000  6000  6000  6000  6000  6000  6000
  slin 14500 14500 14500 14500 14500 14500 14500 14500 14500 14500 14500 14500 14500 14500 14500 14500
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  slin 14500 14500 14500 14500 14500 14500 14500 14500 14500 14500 14500 14500 14500 14500 14500 14500
  lpc10 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000
  ilbc 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000
  g722 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000
  testlaw 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000 15000
  15000 17250 15000
```
**Reloading the complete Asterisk configuration**

In cases, and not limited to, where you did manual modifications to Asterisk dialplan, you need to reload the complete configuration of the Asterisk subsystem which can be done by a simple command:

```
Motion-PBX*CLI> reload
```

Or, out of the Asterisk CLI

```
[root@Motion-PBX ~]# asterisk -rx "reload"
```

This will reload all the configuration related to Asterisk telephony engine.

**Restarting the Asterisk**

If reloading of Asterisk is not enough for the changes made, or there is other reason to do so, you can restart complete Asterisk with:

```
[root@Motion-PBX ~]# service asterisk restart
```

**List peers**

When checking the availability of phones/trunks, you can print out a list of the peers on the system:

```
Motion-PBX*CLI> sip show peers
```

The command will print out a list of SIP peers on the system with additional info like online status and IP address from which they connect.

**Active calls**

Watch the complete list of the active channels and active calls

```
Motion-PBX*CLI> core show channels
```

**SIP channels list**

Watch the complete list of the SIP channels

```
Motion-PBX*CLI> sip show channels
```

**Registered trunks**

Check the registration status of the SIP trunks
Motion-PBX*CLI> sip show registry

<table>
<thead>
<tr>
<th>Host</th>
<th>dnsmgr</th>
<th>Username</th>
<th>Refresh State</th>
<th>Reg.Time</th>
</tr>
</thead>
<tbody>
<tr>
<td>voip.eutelia.it:5060</td>
<td>N</td>
<td>0110703070</td>
<td>105 Registered</td>
<td>Tue, 17 Jan 2017 14:39:52</td>
</tr>
<tr>
<td>82.215.163.137:5060</td>
<td>N</td>
<td>xenia_bs2</td>
<td>105 Registered</td>
<td>Tue, 17 Jan 2017 14:39:52</td>
</tr>
</tbody>
</table>

2 SIP registrations.

Note

The trunk registration string must be previously configured in the trunk section of the XCALLY Web GUI