Asterisk ???????? CLI?????????????????? CLI ?????? logs??????????????

???????????????? putty?? SSH ?????????? root ????????

#asterisk -rvvvvv

Tips:

??? v ???????? logs ?????????? log ????
??? Asterisk CLI ???????????????????
?????? - ? console ?? asterisk -rvvvv ?????????????? CLI> ????????????????? CLI ????????????? logs
?????? - ?????????????? Linux console ?? asterisk -rx "CLI ??"?
?????? Collecting Debug Information for the Asterisk Issue Tracker

---

? ????

?????????? help ???

```
CLI> help voicemail
     voicemail reload Reload voicemail configuration
   voicemail show users List defined voicemail boxes
   voicemail show zones List zone message formats
```

? ??? call-limit ?????? Channel ???

?? call-limit ????? incoming calls ???????????????????????

```
Asterisk -rx "sip show inuse"

* Peer name           In use        Limit
...
provider1              3/0/0         30
provider2              0/0/0         12
...
```

? ??????? Remote UNIX connection, Remote UNIX connection disconnected

(http://www.asteriskdocs.org/en/3rd_E...conf-file.html)

?? /etc/asterisk/asterisk.conf

```
...
[options]
...
hideconnect=yes
...
```

? ?? CLI ????

1. ? extensions.conf ? [globals] ????(??? FreePBX???? globals_custom.conf)
   ${ENV(ASTERISK_PROMPT)}

2. ?? /etc/profile??????
   ASTERISK_PROMPT='%t, %l2, %h*> '
   export ASTERISK_PROMPT

3. ?? Asterisk ????? SSH?

????????

%d Date (year-month-date)
%s Asterisk system name (from asterisk.conf)
%h Full hostname
%H Short hostname
%t Time
%% Percent sign
%# '#' if Asterisk is run in console mode, '>' if running as remote console
%Cn[;n] Change terminal foreground (and optional background) color to specified
A full list of colors may be found in include/asterisk/term.h

On Linux systems, you may also use
%l1 Load average over past minute
%l2 Load average over past 5 minutes
%l3 Load average over past 15 minutes
%l4 Process fraction (processes running / total processes)
%l5 The most recently allocated pid

? ??/??Asterisk??

????/??

CLI> core stop now

CLI> core restart now

??????????/??

CLI> core stop when convenient

CLI> core restart when convenient

???????/??

CLI> core stop gracefully

CLI> core restart gracefully

? ???? DTMF Debugging

??? dtmf ????????? DTMF Debugging ???????????

[2010-05-15 17:33:11] DTMF[24654]: channel.c:2191 __ast_read: DTMF begin '2' received on SIP/osslab-b76020b0
[2010-05-15 17:33:11] DTMF[24654]: channel.c:2195 __ast_read: DTMF begin ignored '2' on SIP/osslab-b76020b0
[2010-05-15 17:33:12] DTMF[24654]: channel.c:2116 __ast_read: DTMF end '2' received on SIP/osslab-b76020b0, duration 60 ms
[2010-05-15 17:33:12] DTMF[24654]: channel.c:2172 __ast_read: DTMF end '2' has duration 60 but want minimum 80, emulating on SIP/osslab-b76020b0
[2010-05-15 17:33:12] DTMF[24654]: channel.c:2224 __ast_read: DTMF end emulation of '2' queued on SIP/osslab-b76020b0
[2010-05-15 17:33:12] DTMF[24654]: channel.c:2191 __ast_read: DTMF begin '2' received on SIP/osslab-b76020b0

??????? DTMF ??????????????

????? ?? /etc/asterisk/logger.conf??? dtmf

[logfiles]
console => dtmf
full => notice,warning,error,debug,verbose,dtmf

??????

CLI> logger reload
CLI> logger show channels
CLI> logger set level DTMF off <== only for Asterisk 1.6.x

Tips:

- ? ?? BUG#0017922?Asterisk 1.6.x ?????? console ??? DTMF log?????
  logger set level DTMF off ?

? ?????????? Outbound Route ????

???????????? console ???

#asterisk -rx "dialplan show" | tr -d "\r" | perl -00 -ne 'print if /Context..outrt/'

? ????????? codec

#asterisk -rx "core show translation recalc"

??????? codec ????? - ????????

```
[root@elastix ~]# asterisk -rx "core show translation"
         Translation times between formats (in milliseconds) for one second of data
          Source Format (Rows) Destination Format (Columns)

          g723 gsm ulaw alaw g726aal2 adpcm slin lpc10 g729 speex ilbc g726 g722
     g723    -  10    2    2        4     2    1     7   12    36   33    4    -
      gsm   22   -    4    4        6     4    3     9   14    38   35    6    -
     ulaw   20  10    -    1        4     2    1     7   12    36   33    4    -
     alaw   20  10    1    -        4     2    1     7   12    36   33    4    -
 g726aal2   22  12    4    4        -     4    3     9   14    38   35    1    -
    adpcm   20  10    2    2        4     -    1     7   12    36   33    4    -
     slin   19   9    1    1        3     1    -     6   11    35   32    3    -
    lpc10   23  13    5    5        7     5    4     -   15    39   36    7    -
     g729   22  12    4    4        6     4    3     9    -    38   35    6    -
    speex   25  15    7    7        9     7    6    12   17     -   38    9    -
     ilbc   24  14    6    6        8     6    5    11   16    40    -    8    -
     g726   22  12    4    4        1     4    3     9   14    38   35    -    -
     g722    -   -    -    -        -     -    -     -    -     -    -    -    -
[root@elastix ~]#
```

? ?? sip.conf ???

CLI>sip show settings

Asterisk-Cli

Command: sip show settings

Execute

```
Global Settings:
----------------
  SIP Port:               5060
  Bindaddress:            0.0.0.0
  Videosupport:           Yes
  AutoCreatePeer:         No
  Allow unknown access:   Yes
  Allow subscriptions:    Yes
  Allow overlap dialing:  Yes
  Promsic. redir:         No
  SIP domain support:     No
  Call to non-local dom.: Yes
  URI user is phone no:   No
  Our auth realm          asterisk
  Realm. auth:            No
  Always auth rejects:    No
  Call limit peers only:  Yes
  Direct RTP setup:       No
```

? ?? General Settings ???

CLI> core show settings

? ????? Trunk ????

CLI> sip show peers

```
Name/username        Host           Dyn Nat ACL Port    Status
voxalot/xxxx         64.34.173.199      N       5060    OK (257 ms)
voiptalk/xxxx        77.240.48.94               5060    OK (1311 ms)
vbuzzer/xxxx         69.172.204.133             80      OK (257 ms)
```

? ???????????

??? 100 ??

CLI> sip show peer 100

? ??????????SIP????

CLI> sip show channels

? ??SIP Trunk ????

CLI> sip show registry

? ?????????-soft hangup

// ??????????????? Call ID

#>asterisk -rx "sip show channels"

```
Peer           User/ANR   Call ID       Seq (Tx/Rx)  Format        Hold   Last Message
...
xx.xx.xx.xx    *600       6faa39db3ca   00103/00000  0x100 (g729)  No     Tx: ACK
xx.xx.xx.xx    mypstn     b68d164f-e9   00101/00102  0x4 (ulaw)    No     Rx: ACK
```

// ? Call ID ????? Channel ID

#>asterisk -rx "sip show channel **b68d164f-e9**" | grep -i "channel id"

Owner channel ID: **SIP/mypstn-b7d27020**

// ????????

for Asterisk 1.4.x
#>asterisk -rx "soft hangup SIP/mypstn-b7d27020"

for Asterisk 1.6.x
#>asterisk -rx "hangup request SIP/mypstn-b7d27020"

? ????????

```
#>asterisk -rx "sip show inuse"


* User name          In use       Limit
221                 0           50
214                 0           50
213                 0           50
212                 0           50
...
* Peer name          In use       Limit
mypstn              0/0          1
221                 0/0          50
214                 0/0          50
213                 0/0          50
212                 0/0          50
211                 0/0          50
210                 0/0          50
209                 0/0          50
...
```

? ?? sip users ??

??sip???sip trunk (type=user & friend) ?????/????? context?NAT ???

```
#>asterisk -rx "sip show users"

Username         Secret       Accountcode    Def.Context     ACL  NAT
xxxxxx           oooooo                      from-sip-extern  No   Always
xxxxxx           oooooo                      from-trunk       No   No
xxxxxx           oooooo                      from-trunk       No   RFC3581
xxxxxx           oooooo                      from-trunk       No   RFC3581
```

? ???????? CLI Logs ?????

??? CLI ?? Logs ????? CLI?? less ???????? ??????????

```
#>asterisk -rvvvvvv | tee /path/to/your.log
```