Escuela Técnica Superior Ingeniería de Telecomunicación

COMANDOS BÁSICOS

Asterisk

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- *Core show uptime*

```
PBX*CLI> core show uptime
System uptime: 50 minutes, 12 seconds
Last reload: 50 minutes, 12 seconds
```

- *Core show channels*

```
PBX*CLI> core show channels
Channel              Location             State   Application(Data)
SIP/202-0000001e     201@PBX:1            Up      Dial(SIP/201,20)
SIP/201-0000001f     (None)               Up      AppDial((Outgoing Line))
2 active channels
1 active call
22 calls processed
```

- Core show channel SIP/201-00000021

```
PBX*CLI> core show channel SIP/201-00000021
 -- General --
           Name: SIP/201-00000021
           Type: SIP
       UniqueID: 1604428921.63
       LinkedID: 1604428921.62
      Caller ID: 201
 Caller ID Name: 3CX
Connected Line ID: 202
Connected Line ID Name: ZOIPER
Eff. Connected Line ID: 202
Eff. Connected Line ID Name: ZOIPER
    DNID Digits: (N/A)
       Language: es
          State: Up (6)
  NativeFormats: (ulaw)
    WriteFormat: ulaw
     ReadFormat: ulaw
 WriteTranscode: No
  ReadTranscode: No
 Time to Hangup: 0
   Elapsed Time: 0h0m14s
      Bridge ID: e8095a8e-d5bd-438d-94a8-b761be3801b1
 --     PBX    --
        Context: PBX
      Extension:
       Priority: 1
     Call Group: 0
   Pickup Group: 0
    Application: AppDial
           Data: (Outgoing Line)
 Call Identifer: [C-00000017]
      Variables:
BRIDGEPVTCALLID=s08oyTUyPccBq1ArwhpoEA..
BRIDGEPEER=SIP/202-00000020
DIALEDPEERNUMBER=201
SIPCALLID=061aa7287d1623823e0fbad6302200a0@192.168.0.35:5060
  CDR Variables:
level 1: clid="3CX" <201>
level 1: src=201
level 1: dcontext=PBX
level 1: channel=SIP/201-00000021
level 1: lastapp=AppDial
level 1: lastdata=(Outgoing Line)
level 1: start=1604428921.604115
level 1: answer=1604428925.154371
level 1: end=1604428925.154905
level 1: duration=3
level 1: billsec=0
level 1: disposition=8
level 1: amaflags=3
level 1: uniqueid=1604428921.63
level 1: linkedid=1604428921.62
level 1: sequence=36
```

- *Core show applications*

```
PBX*CLI> core show applications
    -= Registered Asterisk Applications =-
        AddQueueMember: Dynamically adds queue members.
              ADSIProg: Load Asterisk ADSI Scripts into phone
                AELSub: Launch subroutine built with AEL
                   AGI: Executes an AGI compliant application.
         AlarmReceiver: Provide support for receiving alarm reports from a burglar or fire alarm panel.
                   AMD: Attempt to detect answering machines.
                Answer: Answer a channel if ringing.
      AttendedTransfer: Attended transfer to the extension provided and TRANSFER_CONTEXT
          Authenticate: Authenticate a user
            BackGround: Play an audio file while waiting for digits of an extension to go to.
      BackgroundDetect: Background a file with talk detect.
         BlindTransfer: Blind transfer channel(s) to the extension and context provided
                Bridge: Bridge two channels.
            BridgeWait: Put a call into the holding bridge.
                  Busy: Indicate the Busy condition.
  CallCompletionCancel: Cancel call completion service
  CallCompletionRequest: Request call completion service for previous call
       CELGenUserEvent: Generates a CEL User Defined Event.
         ChangeMonitor: Change monitoring filename of a channel.
           ChanIsAvail: Check channel availability
       ChannelRedirect: Redirects given channel to a dialplan target
               ChanSpy: Listen to a channel, and optionally whisper into it.
             ClearHash: Clear the keys from a specified hashname.
            ConfBridge: Conference bridge application.
            Congestion: Indicate the Congestion condition.
         ContinueWhile: Restart a While loop.
       ControlPlayback: Play a file with fast forward and rewind.
             DAHDIScan: Scan DAHDI channels to monitor calls.
              DateTime: Says a specified time in a custom format.
                 DBdel: Delete a key from the asterisk database.
             DBdeltree: Delete a family or keytree from the asterisk database.
               DeadAGI: Executes AGI on a hungup channel.
                  Dial: Attempt to connect to another device or endpoint and bridge the call.
               Dictate: Virtual Dictation Machine.
             Directory: Provide directory of voicemail extensions.
                  DISA: Direct Inward System Access.
              DumpChan: Dump Info About The Calling Channel.
                  EAGI: Executes an EAGI compliant application.
                  Echo: Echo media, DTMF back to the calling party
              EndWhile: End a while loop.
                  Exec: Executes dialplan application.
                ExecIf: Executes dialplan application, conditionally.
            ExecIfTime: Conditional application execution based on the current time.
             ExitWhile: End a While loop.
             ExtenSpy: Listen to a channel, and optionally whisper into it.
           ExternalIVR: Interfaces with an external IVR application.
              Festival: Say text to the user.
              FollowMe: Find-Me/Follow-Me application.
               ForkCDR: Forks the current Call Data Record for this channel.
              GetCPEID: Get ADSI CPE ID.
                 Gosub: Jump to label, saving return address.
               GosubIf: Conditionally jump to label, saving return address.
                  Goto: Jump to a particular priority, extension, or context.
```

- *Core show application goto*

```
PBX*CLI> core show application goto

  -= Info about application 'Goto' =-

[Synopsis]
Jump to a particular priority, extension, or context.

[Description]
This application will set the current context, extension, and priority in the
channel structure. After it completes, the pbx engine will continue dialplan
execution at the specified location. If no specific <extension>, or <extension>
and <context>, are specified, then this application will just set the specified
<priority> of the current extension.
At least a <priority> is required as an argument, or the goto will return a
'-1',   and the channel and call will be terminated.
If the location that is put into the channel information is bogus, and asterisk
cannot find that location in the dialplan, then the execution engine will try
to find and execute the code in the 'i' (invalid) extension in the current
context. If that does not exist, it will try to execute the 'h' extension. If
neither the 'h' nor 'i' extensions have been defined, the channel is hung up,
and the execution of instructions on the channel is terminated. What this means
is that, for example, you specify a context that does not exist, then it will
not be possible to find the 'h' or 'i' extensions, and the call will terminate!

[Syntax]
Goto([[context,]extensions,]priority)

[Arguments]
Not available

[See Also]
GotoIf(), GotoIfTime(), Gosub(), Macro()
```

- *Core show hints (sin llamada)*

```
PBX*CLI> core show hints

    -= Registered Asterisk Dial Plan Hints =-
202@PBX-BLF          : SIP/201               State:Idle            Presence:not_set        Watchers  0
201@PBX-BLF          : SIP/201               State:Idle            Presence:not_set        Watchers  0
----------------
- 2 hints registered
```

- *Core show hints (con llamada)*

```
PBX*CLI> core show hints

    -= Registered Asterisk Dial Plan Hints =-
202@PBX-BLF          : SIP/201               State:InUse           Presence:not_set        Watchers  0
201@PBX-BLF          : SIP/201               State:InUse           Presence:not_set        Watchers  0
----------------
- 2 hints registered
```

- *Sip show peers*

```
PBX*CLI> sip show peers
Name/username             Host                                    Dyn Forcerport Comedia    ACL Port     Status      Description
201/201                   192.168.0.14                             D  No         No             56256    OK (109 ms)
202/202                   192.168.0.11                             D  No         No             64189    OK (6 ms)
2 sip peers [Monitored: 2 online, 0 offline Unmonitored: 0 online, 0 offline]
```

- *Sip show peers 201*

```
PBX*CLI> sip show peer 201


  * Name       : 201
  Description  :
  Secret       : <Set>
  MD5Secret    : <Not set>
  Remote Secret: <Not set>
  Context      : PBX
  Record On feature : automon
  Record Off feature : automon
  Subscr.Cont. : PBX-BLF
  Language     : es
  Tonezone     : <Not set>
  AMA flags    : Unknown
  Transfer mode: open
  CallingPres  : Presentation Allowed, Not Screened
  Callgroup    :
  Pickupgroup  :
  Named Callgr : PBX
  Nam. Pickupgr: PBX
  MOH Suggest  :
  Mailbox      :
  VM Extension : asterisk
  LastMsgsSent : 32767/65535
  Call limit   : 2147483647
  Max forwards : 0
  Dynamic      : Yes
  Callerid     : "3CX" <201>
  MaxCallBR    : 384 kbps
  Expire       : 112
  Insecure     : no
  Force rport  : No
  Symmetric RTP: No
  ACL          : No
  ContactACL   : No
  DirectMedACL : No
  T.38 support : No
  T.38 EC mode : Unknown
  T.38 MaxDtgrm: 4294967295
  DirectMedia  : No
  PromiscRedir : No
  User=Phone   : No
  Video Support: No
  Text Support : No
  Ign SDP ver  : No
  Trust RPID   : No
  Send RPID    : No
  Path support : No
  Path         : N/A
  TrustIDOutbnd: Legacy
  Subscriptions: Yes
  Overlap dial : Yes
  DTMFmode     : auto
  Timer T1     : 500
  Timer B      : 32000
  ToHost       :
```

```
  Addr->IP     : 192.168.0.14:56256
  Defaddr->IP  : (null)
  Prim.Transp. : UDP
  Allowed.Trsp : UDP
  Def. Username: 201
  SIP Options  : (none)
  Codecs       : (ulaw|alaw|gsm|h263)
  Auto-Framing : No
  Status       : OK (110 ms)
  Useragent    : 3CXPhone 6.0.26523.0
  Reg. Contact : sip:201@192.168.0.14:56256;rinstance=1b8ea98bf0ce2208
  Qualify Freq : 60000 ms
  Keepalive    : 0 ms
  Sess-Timers  : Accept
  Sess-Refresh : uas
  Sess-Expires : 1800 secs
  Min-Sess     : 90 secs
  RTP Engine   : asterisk
  Parkinglot   :
  Use Reason   : No
  Encryption   : No
  RTCP Mux     : No
```

- *Sip show channels*

```
PBX*CLI> sip show channels
Peer             User/ANR         Call ID          Format           Hold     Last Message    Expiry     Peer
192.168.0.14     201              6f4452a6118b322  (ulaw)           No       Tx: ACK                    201
192.168.0.11     202              dznHuhrOashZOrW  (ulaw)           No       Rx: ACK                    202
192.168.0.11     (None)           MgOWTCzj42Vos_X  (nothing)        No       Rx: REGISTER               <guest>
3 active SIP dialogs
```

- *Sip show channel*
    - 201 genera el canal de salida

```
PBX*CLI> sip show channel
6f4452a6118b322a2286b08d20eba515@192.168.0.35:5060  dznHuhrOashZOrW5a0QB7Q..
ZDYwY2NhOWRlZjkyYjBmNTMxZTAzZDcyYWRhYjFjZWQ.
PBX*CLI> sip show channel 6f4452a6118b322a2286b08d20eba515@192.168.0.35:5060

  * SIP Call
  Curr. trans. direction:  Outgoing
  Call-ID:                 6f4452a6118b322a2286b08d20eba515@192.168.0.35:5060
  Owner channel ID:        SIP/201-00000023
  Our Codec Capability:    (ulaw|alaw|gsm|h263)
  Non-Codec Capability (DTMF):   1
  Their Codec Capability:   (ulaw|gsm|alaw)
  Joint Codec Capability:   (ulaw|alaw|gsm)
  Format:                  (ulaw)
  T.38 support             No
  Video support            No
  MaxCallBR:               384 kbps
  Theoretical Address:     192.168.0.14:56256
  Received Address:        192.168.0.14:56256
  SIP Transfer mode:       open
  Force rport:             No
  Audio IP:                192.168.0.35 (local)
  Our Tag:                 as76c1e637
  Their Tag:               282d9d37
  SIP User agent:          3CXPhone 6.0.26523.0
  Username:                201
  Peername:                201
  Original uri:            sip:201@192.168.0.14:56256
  Caller-ID:               201
  Need Destroy:            No
  Last Message:            Tx: ACK
  Promiscuous Redir:       No
  Route:                   <sip:201@192.168.0.14:56256;rinstance=1b8ea98bf0ce2208>
  DTMF Mode:               rfc2833
  SIP Options:             (none)
  Session-Timer:           Inactive
  Transport:               UDP
  Media:                   RTP
```

    - 202 genera el canal de entrada

```
PBX*CLI> sip show channel dznHuhrOashZOrW5a0QB7Q..

  * SIP Call
  Curr. trans. direction:  Incoming
  Call-ID:                 dznHuhrOashZOrW5a0QB7Q..
  Owner channel ID:        SIP/202-00000022
  Our Codec Capability:    (ulaw|alaw|gsm|h263)
  Non-Codec Capability (DTMF):   1
  Their Codec Capability:   (ulaw|gsm|alaw)
  Joint Codec Capability:   (ulaw|alaw|gsm)
  Format:                  (ulaw)
  T.38 support             No
  Video support            No
  MaxCallBR:               384 kbps
  Theoretical Address:     192.168.0.11:64189
  Received Address:        192.168.0.11:64189
  SIP Transfer mode:       open
  Force rport:             No
  Audio IP:                192.168.0.35 (local)
  Our Tag:                 as35143774
  Their Tag:               ccac021c
  SIP User agent:          Zoiper rv2.10.11.5-mod
  Username:                202
  Peername:                202
  Original uri:            sip:202@192.168.0.11:64189
  Caller-ID:               202
  Need Destroy:            No
  Last Message:            Rx: ACK
  Promiscuous Redir:       No
  Route:                   <sip:202@192.168.0.11:64189;transport=UDP>
  DTMF Mode:               rfc2833
  SIP Options:             (none)
  Session-Timer:           Inactive
  Transport:               UDP
  Media:                   RTP
```

- *dialplan show PBX*

```
PBX*CLI> dialplan show PBX
[ Context 'PBX' created by 'pbx_config' ]
  Include =>        'PBX-services'                           [pbx_config]
  Include =>        'PBX-BLF'                                [pbx_config]
  Include =>        'PBX-ringgroups'                         [pbx_config]
  Include =>        'PBX-privates'                           [pbx_config]
  Include =>        'PBX-local'                              [pbx_config]

-= 0 extensions (0 priorities) in 1 context. =-
```

- *dialplan show PBX-service*

```
PBX*CLI> dialplan show PBX-services
[ Context 'PBX-services' created by 'pbx_config' ]
  '*10' =>          1. NoOp()                                     [pbx_config]
                    2. Record(PBX/locucion-203:wav)               [pbx_config]
                    3. Wait(2)                                    [pbx_config]
                    4. Playback(PBX/locucion-203)                 [pbx_config]
                    5. Hangup()                                   [pbx_config]
  '*98' =>          1. NoOp(VoiceMail access from ${CALLERID(num)}@${CONTEXT}) [pbx_config]
                    2. Set(JITTERBUFFER(fixed)=default)           [pbx_config]
                    3. VoiceMailMain(${CALLERID(num)}@${CONTEXT}) [pbx_config]

-= 2 extensions (8 priorities) in 1 context. =-
```