

ScopTELTM IP PBX Software

Troubleshooting

ScopTEL[™] IP PBX Software





- <u>General Troubleshooting</u>
- Log Files
- <u>SIP Response Codes 1XX</u>
- <u>SIP Response Codes 2XX</u>
- <u>SIP Response Codes 3XX</u>
- SIP Response Codes 400-416
- <u>SIP Response Codes 417-439</u>
- <u>SIP Response Codes 440-494</u>
- <u>SIP Response Codes 5XX</u>
- <u>SIP Response Codes 6XX</u>
- OS CLI Commands 1-3
- OS CLI Commands 2-3
- OS CLI Commands 3-3
- <u>Asterisk CLI Commands</u>

- Database Repair Commands
- MySQL Repair is Taking Too Long
- ScopTEL GUI is Slow
- <u>ScopTEL GUI Will Not Restart</u>
- <u>ScopTEL GUI Fatal Error and Dead Call Processing</u>
- Debugging Authentication Failures 1-3
- Debugging Authentication Failures 2-3
- Debugging Authentication Failures 3-3
- Debugging Call Failures with 'database show'
- Debugging Call Failures with SIP Cause Codes
- <u>ScopSTATS System Monitoring Logs</u>

General Troubleshooting



General Troubleshooting:

- Most telephony errors are caused by one of the following causes
 - Bad authentication credentials
 - DNS resolution
 - Class of Service misconfigurations
 - Firewall issues
 - Asterisk bugs
 - User error (DND, call forwarding etc...)
- CLI tools are used to debug most of these issues

Windows Utilities:

• Putty (Windowstm based SSH tool)

http://the.earth.li/~sgtatham/putty/latest/

WinSCP (Windowstm based, Linux file system editor and transfer client)

http://winscp.net/download/

ScopTEL[™] IP PBX Software



Server Logs:

• /var/log/messages

SMTP Logs:

• /var/log/maillog

Hardware Logs:

• dmesg

Asterisk Logs:

/var/log/asterisk/messages

Asterisk Files:

• /etc/asterisk

ScopServ PHP logs:

• /tmp/php2.log

Asterisk Core Dumps (GDB files)

• dir /tmp/core*



SIP Response Codes 1XX

Note: All response codes below are defined by [RFC3261] unless otherwise noted. Registry: Response Code Reference -------Provisional 1xx 100 Trying 180 Ringing 181 Call Is Being Forwarded 182 Queued 183 Session Progress 199 Early Dialog Terminated [RFC6228]



SIP Response Codes 2XX:

Succes	ssful	2xx	
200	OK		
202	Acce	pted	(Deprecated)
204	No No	otifi	cation

[RFC6665] [RFC5839]



SIP Response Codes 3XX

SIP Response Codes 3XX:

Redirection 3xx

- 300 Multiple Choices
- 301 Moved Permanently
- 302 Moved Temporarily
- 305 Use Proxy
- 380 Alternative Service



SIP Response Codes 400-416

SIP Response Codes 400-416:

Request Failure 4xx

- 400 Bad Request
- 401 Unauthorized
- 402 Payment Required
- 403 Forbidden
- 404 Not Found
- 405 Method Not Allowed
- 406 Not Acceptable
- 407 Proxy Authentication Required
- 408 Request Timeout
- 410 Gone
- 412 Conditional Request Failed

[RFC3903]

- 413 Request Entity Too Large
- 414 Request-URI Too Long
- 415 Unsupported Media Type
- 416 Unsupported URI Scheme



SIP Response Codes 417-439:

417 Unknown Resource-Priority	[RFC4412]
420 Bad Extension	
421 Extension Required	
422 Session Interval Too Small	[RFC4028]
423 Interval Too Brief	
424 Bad Location Information	[RFC6442]
428 Use Identity Header	[RFC4474]
429 Provide Referrer Identity	[RFC3892]
430 Flow Failed	[RFC5626]
433 Anonymity Disallowed	[RFC5079]
436 Bad Identity-Info	[RFC4474]
437 Unsupported Certificate	[RFC4474]
438 Invalid Identity Header	[RFC4474]
439 First Hop Lacks Outbound Support	[RFC5626]



SIP Response Codes 440-494:

440 Max-Breadth Exceeded	[RFC5393]
469 Bad Info Package	[RFC6086]
470 Consent Needed	[RFC5360]
480 Temporarily Unavailable	
481 Call/Transaction Does Not Exist	
482 Loop Detected	
483 Too Many Hops	
484 Address Incomplete	
485 Ambiguous	
486 Busy Here	
487 Request Terminated	
488 Not Acceptable Here	
489 Bad Event	[RFC6665]
491 Request Pending	
493 Undecipherable	
494 Security Agreement Required	[RFC3329]



SIP Response Codes 5XX

SIP Response Codes 5XX:

Server Failure 5xx

- 500 Server Internal Error
- 501 Not Implemented
- 502 Bad Gateway
- 503 Service Unavailable
- 504 Server Time-out
- 505 Version Not Supported
- 513 Message Too Large
- 580 Precondition Failure

[RFC3312]



SIP Response Codes 6XX:

Global Failures 6xx

600 Busy Everywhere

603 Decline

604 Does Not Exist Anywhere

606 Not Acceptable





Useful CLI commands page 1:

scopserv yum update executes a full software update scopserv_yum install <package name> installs a package from repositories scopserv_dumpkey displays license and admin password dkms status shows install status of dahdi wanpipe mISDN drivers asterisk -r opens Asterisk CLI for Asterisk command line debugging asterisk -vr opens Asterisk CLI for Asterisk command line debugging with increased verbosity lsof -i :5555 shows which service is using port 5555 rpm -qa <package name> shows package version rpm –e <package name> --nodeps removes package without uninstalling dependencies rpm -qa |grep <package name> shows installed packages version chown -R scopserv:scopserv /<path> changes ownership recursively on a path kill -KILL <pid> kills a process ID



Useful CLI commands page 2:

•	htop	shows all processes and consumption
•	top	shows all processes and consumption
•	ngrep -d any port 25 -W BYLINE	network trace on port 25 to debug SMTP issues
•	nslookup -query=MX <domain.com></domain.com>	Linux DNS lookup for MX record on specified domain
•	rm -rf /root/.ssh/known_hosts	removes all known ssh keys
•	wget <package path=""></package>	CLI command to download a file from path
•	ngrep –d port any 69	network trace on port 69 to display TFTP traffic
•	cat / <path></path>	display contents of file to screen
•	nano / <path></path>	Linux text editor to edit path
•	service <service name=""> restart</service>	restart network without rebooting server
•	service <service name=""> stop</service>	stop service name without rebooting server
•	service <service name=""> status</service>	display service name status



OS CLI Commands 3-3

Useful CLI commands page 3:

reboot	reboot server
shutdown -h now	shutdown server
lynx <website address=""></website>	Linux command line web browser
ping <ip address=""></ip>	Ping an IP address
traceroute <ip address=""></ip>	trace an IP address through multiple hops
tcpdump -nq -s 0 -i eth1 -w /tmp/sip.pcap port 5060	saves a SIP pcap trace to /tmp using eth1
ps auwwwwx grep crond	prints all crond PID's
rpm -eallmatches <package name=""> –nodeps</package>	`removes the package name without dependencies
rpm -e <package name="">nodeps –noscripts</package>	removes a broken package forcibly
route	prints the routing table

The "ifconfig" command allows the operating system to setup network interfaces and allow the user to view information about the configured network interfaces.

Ifconfig

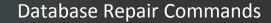
Asterisk CLI Commands



Useful Asterisk CLI commands:

To activate enter asterisk -r from Linux CLI

	ScopTEL™ IP PBX Software	16
•	sip set debug <on off=""></on>	prints SIP messages to the CLI
•	agi set debug on	prints agi debug messages (always useful when creating a support ticket)
•	core show hints	This command lists registered hints
•	hangup request <channel> <all></all></channel>	Request that a channel be hung up. The hangup takes effect the next time the driver reads or writes from the channel. If 'all' is specified instead of a channel name, all channels will see the hangup request.
•	pri set debug off span 	stop intense PRI debugging on span number
•	pri intense debug span 	shows Q.931 and SABME messages
•	pri show spans	displays status of all PRI spans
•	queue show	displays all ACD queues info
•	database show	display astDB info (useful to see if a phone is in DND status
•	sip show peer <peer></peer>	show details of SIP peer
•	sip show peers	show all SIP peers status
•	core show channels	display active channels



Database Repair Commands

• mysqlcheck –A –o –r

Automatically optimizes and repairs mysql tables. If crashed on Slave (HA) then this need to be done on slave and not master

- if you have to rebuild queue_log db :
 - service scopserv_queuelog stop
 - /var/www/scopserv/telephony/scripts/<u>queue2sql.pl</u> --recover /var/log/asterisk/queue_log
 - service scopserv_queuelog start
- Update CDR database direction
 - mysql scopserv (enter) update cdr set calldir='unknown';



MySQL Repair is Taking Too Long

If the Server is not configured in High Availability Mode (standard installation):

- mysql scopserv (enter)
 - truncate cel; truncate phone_history; truncate horde_histories;

If the Server is configured in High Availability Mode:

mysql scopserv_repl (enter)

truncate cel; truncate phone_history; truncate horde_histories;



ScopTEL GUI is Slow

ScopTEL GUI is Slow

• mysqlcheck –A –o –r

Automatically optimizes and repairs mysql tables. If crashed on Slave (HA) then this need to be done on slave and not master

scopserv_yum update

Older packages, especially prior to scopserv_realtime nodejs changes are not fully optimized

• Telephony>Configuration>Reports (CDR/ACD)>Archiving>Enable Archiving :[x]

General Telephony Modules Advanced	Modules Commit Menu Features Code Call Parking Voicemail Virtual Fax Logging and Errors Reports (CDR/ACD)
Recording/Monitoring Sound Manager	Billing Custom Dialplan Actions Provisioning Security
Enable CDR logging?:	✓ Warning: Please note that if you uncheck this option, no CDR reports will be available!
Enable Queue (ACD) logging?:	✓ Warning: Please note that if you uncheck this option, no Queue (ACD) reports will be available!
Date Format display:	2013-12-19
	Default: 2013-12-19
Archiving	
Enable Archiving ? 👔 :	
Archive Mode:	Year(s)
	Default: Year(s)



ScopTEL GUI Will Not Restart

ScopTEL GUI will Not Restart

•	mysqlcheck –A –o –r	Automatically optimizes and repairs mysql tables. If crashed on Slave (HA) then this need to be done on slave and not master
•	service scopserv stop	Stops the GUI service and realtime
•	lsof -i :5555	Displays the PID that is bound to the scopserv service port
•	kill –KILL <pid 5555="" lsof="" of="" output="" –i=""></pid>	Kills the PID bound to the scopserv port
•	service scopserv start	Starts the GUI once the binding port is freed



ScopTEL GUI Fatal Error and Dead Call Processing

- Most often this is caused by a full hard drive
- You must delete files on the hard drive to restore call processing
- Usually this is caused by:
 - Too many local backups
 - Too many recording files in /var/spool/asterisk/monitor/
 - Too many files in /var/log/asterisk/
 - Too many files in /tmp/
- To locate the largest folders do:
 - cd / Change to root directory
 - du * -s
 Print Disk Usage Summary and locate the largest folder
 - Use the 'cd' command to change into the largest directory
 - Use the Linux 'rm' command to remove files
 - Keep drilling down from / and use the 'du * -s' command to locate the largest folders and keep deleting files with the 'rm' command until you have freed enough space in the file system to restore normal service.



Here is a typical authentication failure in Asterisk The reason for this failure is that peer 228 does not exist!

[root@demo ~]# asterisk -vr

[Aug 1 14:41:46] Running as group 'scopserv'

[Aug 1 14:41:46] Connected to Asterisk 1.8.12.0 currently running on demo (pid = 6006)

Verbosity is at least 3

Core debug is at least 3

[2012-08-01 14:50:06] NOTICE[6260]: chan_sip.c:24974 handle_request_register: Registration from '"228"<sip:228@scopserv.local>' failed for '192.168.192.55:57744' - No matching peer found



Here is a typical authentication failure in Asterisk

The reason for this failure is that the password does not match!

[root@demo ~]# asterisk -vr

[Aug 1 14:41:46] Running as group 'scopserv'

[Aug 1 14:41:46] Connected to Asterisk 1.8.12.0 currently running on demo (pid = 6006)

Verbosity is at least 3

Core debug is at least 3

[2012-08-01 14:53:49] NOTICE[6260]: chan_sip.c:24974 handle_request_register: Registration from '"227"<sip:227@scopserv.local>' failed for '192.168.192.55:63316' - Wrong password



Here is a typical authentication success in Asterisk

[root@demo ~]# asterisk -vr

[Aug 1 14:41:46] Running as group 'scopserv'

[Aug 1 14:41:46] Connected to Asterisk 1.8.12.0 currently running on demo (pid = 6006)

Verbosity is at least 3

Core debug is at least 3

[2012-08-01 14:56:00] NOTICE[6260]: chan_sip.c:20812 handle_response_peerpoke: Peer '227' is now Reachable. (7ms / 2000ms)



In this first example a call from 227 to 221 fails to ring 221

[2012-08-01 15:02:05] == Extension Changed 227[default-local] new state InUse for Notify User 221

- [2012-08-01 15:02:05] -- Executing [221@default-default:9] Macro("SIP/227-00000005", "default-default.sIP/221,221,default,,en,u221@default,twWxXkKg,,default_default,Local/0@default-local/n,vm") in new stack
- [2012-08-01 15:02:05] -- Executing [s@macro-default-dial:1] NoOp("SIP/227-00000005", ""CALL TO LOCAL EXTENSION FROM 227(227)"") in new stack

[2012-08-01 15:02:05] -- <SIP/227-00000005>AGI Script agi://127.0.0.1:4573/dial completed, returning 0

- [2012-08-01 15:02:05] -- Executing [s@macro-default-dial:4] Execlf("SIP/227-00000005", "1?Macro(all-vm,default,b,221@default,Local/0@default-local/n,vm)") in new stack
- [2012-08-01 15:02:05] -- Executing [s@macro-all-vm:1] NoOp("SIP/227-00000005", "CALL TO VOICEMAIL") in new stack

[2012-08-01 15:02:05] -- Executing [s@macro-all-vm:2] VoiceMail("SIP/227-00000005", "221@default,b") in new stack

database show shows the extension has enabled the DND feature code!

demo*CLI> database show	
/CustomDevstate/dnd-default-221	: BUSY
/DND/Local/*78@default	: On
/DND/SIP/221	: On



In this second example a call from 227 to 221 fails to ring 221

[2012-08-01 15:24:27] -- Executing [221@default-default:1] Set("SIP/227-00000009", "CDR(tenant)=default") in new stack

- [2012-08-01 15:24:27] -- Executing [221@default-default:9] Macro("SIP/227-00000009", "default-default.sip) dial,SIP/221,221,default,,en,u221@default,twWxXkKg,,default_default,Local/0@default-local/n,vm") in new stack
- [2012-08-01 15:24:27] -- Executing [s@macro-default-dial:1] NoOp("SIP/227-00000009", ""CALL TO LOCAL EXTENSION FROM 227(227)"") in new stack
- [2012-08-01 15:24:27] -- Called SIP/221
- [2012-08-01 15:24:27] -- Got SIP response 302 "Moved Temporarily" back from 192.168.100.100:5060
- [2012-08-01 15:24:27] -- Now forwarding SIP/227-00000009 to 'Local/555@default-default' (thanks to SIP/221-0000000a)
- [2012-08-01 15:24:27] NOTICE[30807]: app_dial.c:892 do_forward: Not accepting call completion offers from call-forward recipient Local/555@default-default-59dd;1
- [2012-08-01 15:24:27] NOTICE[30807]: chan_local.c:899 local_call: No such extension/context 555@default-default while calling Local channel

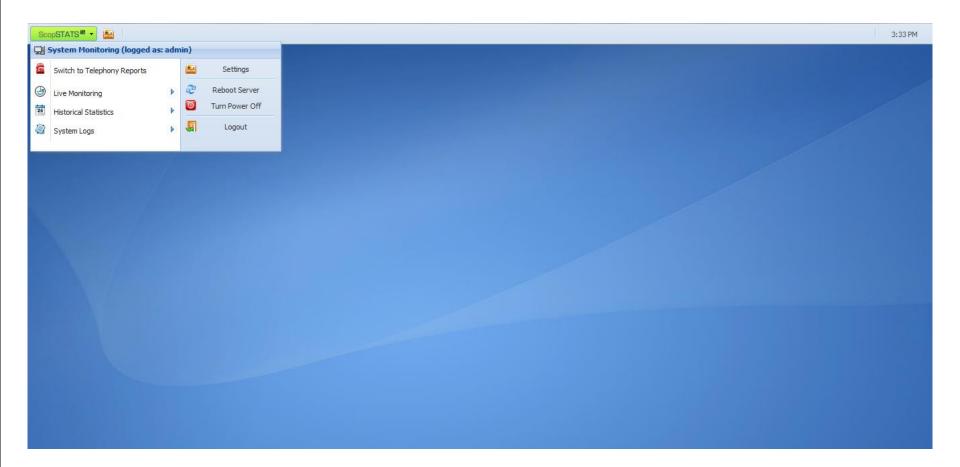
[2012-08-01 15:24:27] NOTICE[30807]: app_dial.c:980 do_forward: Forwarding failed to dial 'Local/555@default-default'

[2012-08-01 15:24:27] == Everyone is busy/congested at this time (1:0/0/1)

The SIP response code shows the extension has enabled a native SIP forward to 555 from the SIP phone! There is no 555 in the dial plan and the forward fails. The user must remove the bad forwarding attempt from their phone.



The ScopSTATS reporting engine can also display System Logs instead of accessing the Linux file system





ScopTEL

