



Gear up your PBX

Cut Costs, Boost Profits

Version 20160711

Basic Troubleshooting



Self Help

- Dashboard
- **Verbose** Logging
- Server Activity Log
- Phone System Event Log
- Configure Email Notifications
- WireShark



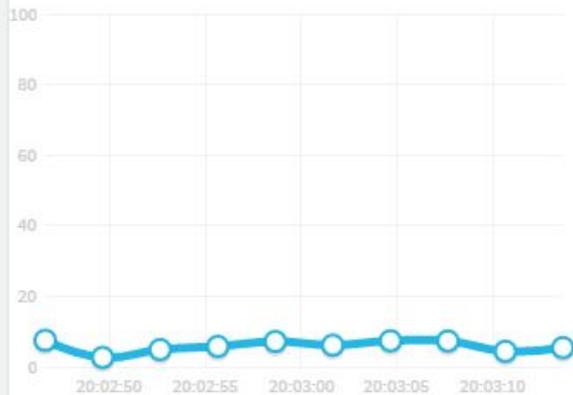
Self Help - Dashboard

- Gives a quick overview over current activity and status of PBX

Dashboard

Help

System Status



Disk Usage

41% used

34.7 GB free

Memory Usage

53% used

2.3 GB free

CPU Usage

5/100%

PBX Status

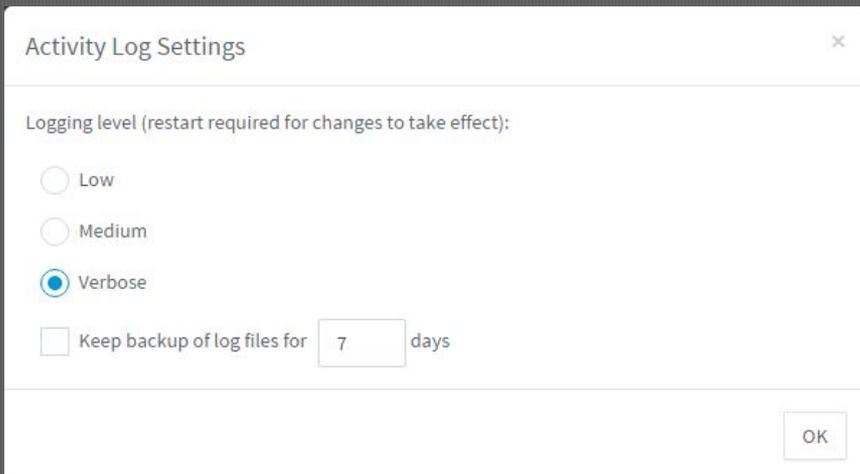
Trunks Up	3/5
Extensions Up	6/10
Number of calls in use	0/8
Blacklisted IPs	1
Event log	Purge
Call history	Purge 223 calls
Chat logs	Disabled
Automatic Backups	OFF
Firewall Check	Firewall test failed
Services	System Extensions

Information

FQDN	.3cx.com.au
IP	Static
PUSH Account	Own Account
Webmeeting FQDN	.3cx.net
Webmeeting MCU	eu006-rbx.3cx.eu
License	Activated PRO 15.0.57336.0
Maintenance	08/18/2017 10:00:00 PM OK
Sim Calls	8
Sim Meeting Participants	25
Outbound Rules	4

Self Help - Verbose Logging

- Default is Low Logging
- Log files use a cycle logging
- Call History (Web Reports) does not cycle
- ALL Log Files will be **CLEANED** after restarting services or PBX
- For debugging logging must be set to Verbose (change requires services to be restarted)
- Log files Path: C:\ProgramData\3CX\Data\InstanceX\Logs



The screenshot shows a dialog box titled "Activity Log Settings" with a close button (X) in the top right corner. Below the title bar, the text "Logging level (restart required for changes to take effect):" is displayed. There are four radio button options: "Low", "Medium", "Verbose", and "Keep backup of log files for". The "Verbose" option is selected, indicated by a blue dot. The "Keep backup of log files for" option is unchecked, and its value is set to "7" days. An "OK" button is located in the bottom right corner of the dialog box.

Activity Log Settings

Logging level (restart required for changes to take effect):

Low

Medium

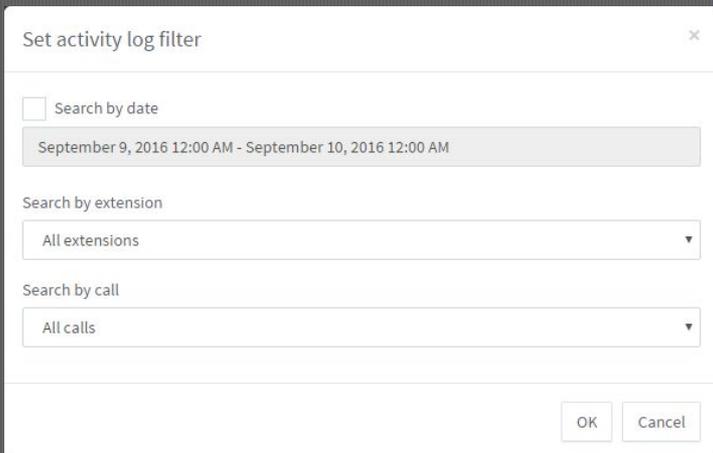
Verbose

Keep backup of log files for days

OK

Self Help - Server Activity Log

- **Basic** SIP Flow Messages
- Provides information on
 - All Phone registrations
 - Interaction with PSTN Gateways & SIP Trunks
 - All calls
- Use Filter for Extension and Call or by Date
- More filter and external analytics can be performed via BinLogViewer



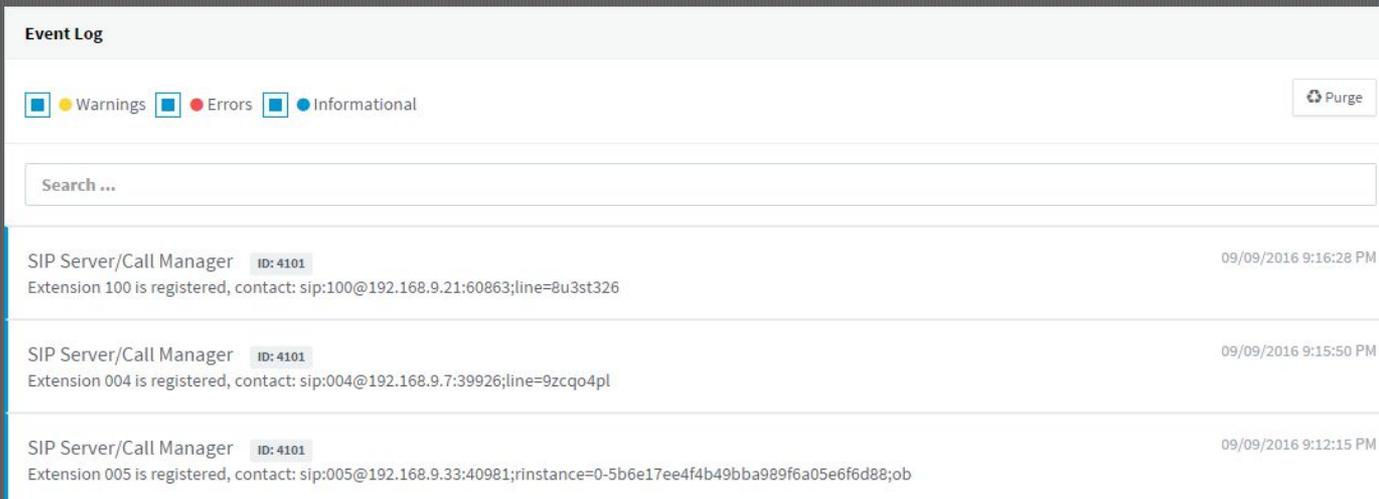
The screenshot shows a dialog box titled "Set activity log filter" with a close button (X) in the top right corner. The dialog contains three main sections for filtering:

- Search by date:** A checkbox is currently unchecked. Below it is a date range input field containing "September 9, 2016 12:00 AM - September 10, 2016 12:00 AM".
- Search by extension:** A dropdown menu is currently set to "All extensions".
- Search by call:** A dropdown menu is currently set to "All calls".

At the bottom right of the dialog, there are two buttons: "OK" and "Cancel".

Self Help - Event Log

- Summarized Event
- Monitoring via
 - 3rd Party Tools
 - E-mail Notifications



The screenshot displays an 'Event Log' interface. At the top left, the title 'Event Log' is shown. Below it, there are filter buttons for 'Warnings' (yellow square), 'Errors' (red square), and 'Informational' (blue square). A 'Purge' button with a circular arrow icon is located on the right side. A search bar with the placeholder text 'Search ...' is positioned below the filters. The main content area contains three log entries, each with a blue vertical bar on the left. Each entry includes the source 'SIP Server/Call Manager', an ID 'ID: 4101', and a timestamp. The first entry is dated 09/09/2016 9:16:28 PM and describes extension 100 registration. The second entry is dated 09/09/2016 9:15:50 PM and describes extension 004 registration. The third entry is dated 09/09/2016 9:12:15 PM and describes extension 005 registration.

Event Description	Timestamp
SIP Server/Call Manager ID: 4101 Extension 100 is registered, contact: sip:100@192.168.9.21:60863;line=8u3st326	09/09/2016 9:16:28 PM
SIP Server/Call Manager ID: 4101 Extension 004 is registered, contact: sip:004@192.168.9.7:39926;line=9zcco4pl	09/09/2016 9:15:50 PM
SIP Server/Call Manager ID: 4101 Extension 005 is registered, contact: sip:005@192.168.9.33:40981;rinstance=0-5b6e17ee4f4b49bba989f6a05e6f6d88;ob	09/09/2016 9:12:15 PM

Self Help - Configure Notifications

- Email Notifications
- Settings > Email
- Multiple addresses possible with “,” separation

Events

Send an Email Alert when the following events occur:

- Someone dials an Emergency Number
- The status of trunk changes
- A trunk failover occurs or max amount of calls available through trunk has been exceeded
- Trunk/Provider responds to Request with an Error code
- The registration status of an extension changes
- The license limit is reached
- An IP has been blacklisted

Self Help - WireShark

- SIP Flow Debugging

Ethernet 2 - VoIP Calls

Detected 2 VoIP Calls. Selected 0 Calls.

Start Time	Stop Time	Initial Speaker	From	To	Protocol	Packets	State	Comments
0.413856	5.649033	192.168.9.148	"Pedro Martinez" < sip:102 < sip:100@192.168.9.213:5060 SIP		SIP	10	COMPLETED	
0.686187	5.591725	192.168.9.213	"Pedro Martinez" < sip:102i < sip:100@192.168.9.213 SIP		SIP	7	COMPLETED	

Total: Calls: 2 Start packets: 0 Completed calls: 2 Rejected calls: 1

Buttons: Prepare Filter, Flow, Player, Select All, Close

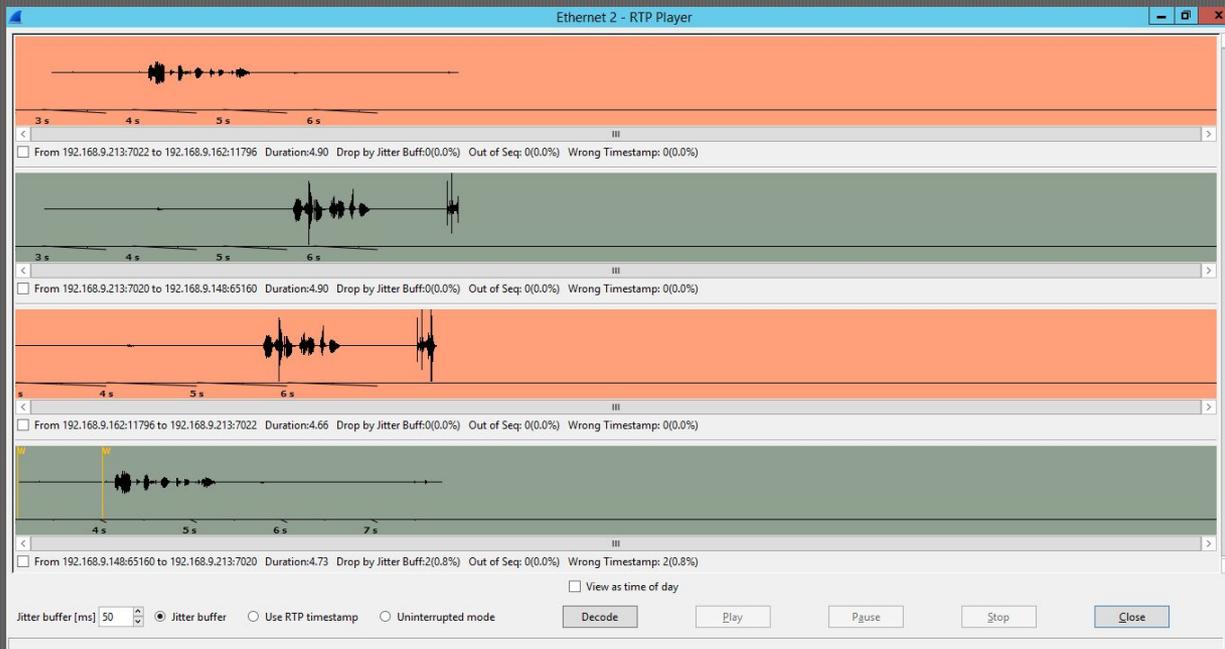
Ethernet 2 - Graph Analysis

Time	192.168.9.148	192.168.9.213	192.168.9.162	Comment
0.413856000	INVITE SDP (g711)			SIP From: "Pedro Martinez" < sip:102@192.168.9.213:5060 To
0.513532000	407 Proxy Auth...			SIP Status
0.558997000	ACK			SIP Request
0.584738000	INVITE SDP (g711)			SIP From: "Pedro Martinez" < sip:102@192.168.9.213:5060 To
0.686187000		INVITE SDP (g711)		SIP From: "Pedro Martinez" < sip:102@192.168.9.213:5060 To:
0.686382000		100 Trying		SIP Status
0.691766000		100 Trying		SIP Status
0.784374000		180 Ringing		SIP Status
0.884576000		180 Ringing		SIP Status
2.633052000		200 OK SDP (g711)		SIP Status
2.641501000	RTP (g711U)			RTP Num packets:144 Duration:2.860s SSRC:0x42A2
2.641602000		RTP (g711U)		RTP Num packets:143 Duration:2.840s SSRC:0x4E6A
2.735311000		ACK		SIP Request
2.735471000		200 OK SDP (g711)		SIP Status
2.765975000	ACK			SIP Request
2.968390000		RTP (g711U)		RTP Num packets:129 Duration:2.559s SSRC:0x185BE2E
3.007917000		RTP (g711U)		RTP Num packets:134 Duration:2.632s SSRC:0x53DE9974

Buttons: Save As, Close

Self Help - WireShark

- Debug of **Audio** issue
- Debug of timing



Self Help - WireShark

The screenshot displays the Wireshark interface with a capture of network traffic on the Ethernet 2 interface. The main pane shows a list of captured packets, including SIP messages (INVITE, ACK, RTP) and other protocols like UDP and TCP. The packet details pane for the selected packet (Frame 1) shows the following structure:

- Ethernet II, Src: Vmware_96:2c:22 (00:50:56:96:2c:22), Dst: HewlettP_44:55:12 (9c:b6:54:44:55:12)
- Internet Protocol Version 4, Src: 192.168.9.213 (192.168.9.213), Dst: 192.168.9.57 (192.168.9.57)
- User Datagram Protocol, Src Port: 3389 (3389), Dst Port: 61370 (61370)
- Data (1129 bytes)

The packet bytes pane shows the raw data of the frame. Below the main capture pane, two analysis windows are open:

- Ethernet 2 - Graph Analysis:** A sequence diagram showing the timing of SIP messages and RTP streams between IP addresses 192.168.9.148 and 192.168.9.213. It includes events like INVITE SDP (g711), 407 Proxy Authentication Required, ACK, 100 Trying, 180 Ringing, 200 OK SDP (g711), and RTP (g711U).
- Ethernet 2 - VoIP Calls:** A summary window titled "Detected 2 VoIP Calls. Selected 2 Calls." showing a table of call details:

Start Time	Stop Time	Initial Speaker	From	To	Protocol	Packets	State	Comm
0.989020	7.757081	192.168.9.148	"Pedro Martinez" <sip:100@192.168.9.213:5000>	<sip:100@192.168.9.213:5000>	SIP	10	COMPLETED	
1.383532	7.693548	192.168.9.213	"Pedro Martinez" <sip:100@192.168.9.213>	<sip:100@192.168.9.213>	SIP	7	COMPLETED	

At the bottom of the VoIP Calls window, it displays: Total Calls: 2, Start packets: 0, Completed calls: 2, Rejected calls: 1. Buttons for "Prepare Filter", "Flow", "Player", "Select All", and "Close" are visible.

- How to: www.3cx.com/blog/voip-howto/sip-traffic-capture

Ask for Help

- When?
 - After Reading Config Guides
www.3cx.com/support
 - After Reading Admin Guide
www.3cx.com/docs/manual
 - After Reading Tech Docs
www.3cx.com/blog/category/docs

Help Requests

- Who?
 - Active NFR Partners
 - Customers with Support Agreement
- About?
 - Supported IP Phones (using default templates)
 - Supported VoIP Providers
 - 3CX products
- How?
 - Via Phone
 - Via Ticket System <http://helpdesk.3cx.com>



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More Training Material at:
www.3CX.com/3CXAcademy

