

CyberData

The IP Endpoint Company


digium[®]
Switchvox[®]

Using CyberData Devices with a Digium SwitchVox PBX



This is a sample of CyberData products, please visit CyberData.net for a complete list.

Document: 931453A

Draft Date: 11/20/2017

Version 1

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Table of Contents

Copyright Notice	1
Document Revision Information	3
Hardware and Firmware Information	3
Summary of Testing Results ~ Video Intercom	4
Summary of Testing ~ Paging Server, Paging Adapter, Speaker, Paging Amp, and Intercom	5
Testing Notes	6
Creating a CyberData Device Template	6
Creating the Extensions	10
Configure the CyberData Device	13
Resources	14
Contact CyberData Corporation	14

Document Revision Information

Version 1 (Original Version)

11/20/2017

Hardware and Firmware Information

Device	Software/Firmware Version
Digium SwitchVox	6.4.2
SIP Paging Adapter	11.6.1
SIP Speaker	11.6.7
Paging Server	12.0.0
Keypad Intercom	11.8.1
SIP Strobe	11.8.0
Video Intercom with Keypad	1.1.0

Summary of Testing Results ~ Video Intercom

#	Test	Result
1	Incorrect Username	Pass
2	Incorrect Password	Pass
3	Minimum Expiry	Pass
4	Maximum Expiry	Pass
5	Default Expiry	Pass
6	Incorrect Username	Pass
7	Incorrect Password	Pass
8	Minimum Expiry	Pass
9	Maximum Expiry	Pass
10	Default Expiry	Pass
11	Simultaneous Registration	Pass
12	DUT to Single Phone Extension - DUT Cancels Before Answer	Pass
13	DUT to Single Phone Extension - DUT Terminates After Answer	Pass
14	DUT to Single Phone Extension - Phone Terminates After Answer	Pass
15	DUT to Single Phone Extension - Mute/Unmute Audio	Pass
16	DUT to Single Phone Extension - Session Refresh	Not Supported
17	Fault Detection - DUT to Single Phone Extension - DUT Cancels Before Answer	Pass
18	Fault Detection - DUT to Single Phone Extension - DUT Terminates After Answer	Pass
19	Fault Detection - DUT to Single Phone Extension - Phone Terminates After Answer	Pass
20	IP-PBX to DUT - Phone Terminates Before Answer	Pass
21	IP-PBX to DUT - Phone Terminates After Answer	Pass
22	IP-PBX to DUT - DUT Terminates After Answer	Pass
23	IP-PBX to DUT - Mute/Unmute Audio	Pass
24	IP-PBX to DUT - Session Refresh	Not Supported
25	Attempt Intercom Call While Intercom Call Is in Progress	Fail
26	Attempt Nightringer Call While Intercom Call Is in Progress	Fail
27	Nightringer Rings During Inbound Call Attempt	Fail
28	DUT to Group Extension - DUT Cancels Before Answer	Fail*
29	DUT to Group Extension - DUT Terminates After Answer	Fail*
30	DUT to Group Extension - Phone Terminates After Answer	Fail*
31	Fault Detection - DUT to Group Extension - DUT Cancels Before Answer	Fail*
32	Fault Detection - DUT to Group Extension - DUT Terminates After Answer	Fail*
33	Fault Detection - DUT to Group Extension - Phone Terminates After Answer	Fail*
34	Phone to Group Extension with DUT Membership - Live Page	Fail*
35	Phone to Group Extension with DUT Membership - Live Page - Mute/Unmute Audio	Fail*
36	Phone to Group Extension with DUT Membership - Live Page - Session Refresh	Not Supported
37	Simultaneous Ring with Phone - Call Cancelled Before Answer	Pass
38	Simultaneous Ring with Phone - Call Answered by Phone	Pass
39	Receive RFC2833 DTMF for Relay Activation	Pass
40	Send RFC2833 DTMF	Pass

* Please see testing notes for description.

Summary of Testing ~ Paging Server, Paging Adapter, Speaker, Paging Amp, and Intercom

#	Test	Result
1	Incorrect Username	Pass
2	Incorrect Password	Pass
3	Minimum Expiry	Pass
4	Maximum Expiry	Pass
5	Default Expiry	Pass
6	Incorrect Username	Pass
7	Incorrect Password	Pass
8	Minimum Expiry	Pass
9	Maximum Expiry	Pass
10	Default Expiry	Pass
11	Simultaneous Registration	Pass
12	DUT to Single Phone Extension - DUT Cancels Before Answer	Pass
13	DUT to Single Phone Extension - DUT Terminates After Answer	Pass
14	DUT to Single Phone Extension - Phone Terminates After Answer	Pass
15	DUT to Single Phone Extension - Mute/Unmute Audio	Pass
16	DUT to Single Phone Extension - Session Refresh	Not Supported
17	Fault Detection - DUT to Single Phone Extension - DUT Cancels Before Answer	Pass
18	Fault Detection - DUT to Single Phone Extension - DUT Terminates After Answer	Pass
19	Fault Detection - DUT to Single Phone Extension - Phone Terminates After Answer	Pass
20	IP-PBX to DUT - Phone Terminates Before Answer	Pass
21	IP-PBX to DUT - Phone Terminates After Answer	Pass
22	IP-PBX to DUT - DUT Terminates After Answer	Pass
23	IP-PBX to DUT - Mute/Unmute Audio	Pass
24	IP-PBX to DUT - Session Refresh	Not Supported
25	Attempt Intercom Call While Intercom Call Is in Progress	Pass
26	Attempt Nightringer Call While Intercom Call Is in Progress	Pass
27	Nightringer Rings During Inbound Call Attempt	Pass
28	DUT to Group Extension - DUT Cancels Before Answer	Pass
29	DUT to Group Extension - DUT Terminates After Answer	Fail*
30	DUT to Group Extension - Phone Terminates After Answer	Fail*
31	Fault Detection - DUT to Group Extension - DUT Cancels Before Answer	Pass
32	Fault Detection - DUT to Group Extension - DUT Terminates After Answer	Fail*
33	Fault Detection - DUT to Group Extension - Phone Terminates After Answer	Fail*
34	Phone to Group Extension with DUT Membership - Live Page	Fail*
35	Phone to Group Extension with DUT Membership - Live Page - Mute/Unmute Audio	Fail*
36	Phone to Group Extension with DUT Membership - Live Page - Session Refresh	Not Supported
37	Simultaneous Ring with Phone - Call Cancelled Before Answer	Pass
38	Simultaneous Ring with Phone - Call Answered by Phone	Pass
39	Receive RFC2833 DTMF for Relay Activation	Pass
40	Send RFC2833 DTMF	Pass

* Please see testing notes for description.

Testing Notes

During the testing of CyberData devices against the Digium SwitchVox there were some anomalies that were experienced. CyberData devices will not function properly if used in a “Group Extension” this is because the SwitchVox sends an unsolicited ‘Update’ message to the device. CyberData Devices do not support the Update method in SIP transactions and because if that they cannot be used in a group call scenario. The SwitchVox system does not support session refresh which should not impact the operation of CyberData devices. If an authentication password is given incorrectly the device will attempt to re-register constantly. This could cause network congestion if it occurred on multiple devices.

Creating a CyberData Device Template

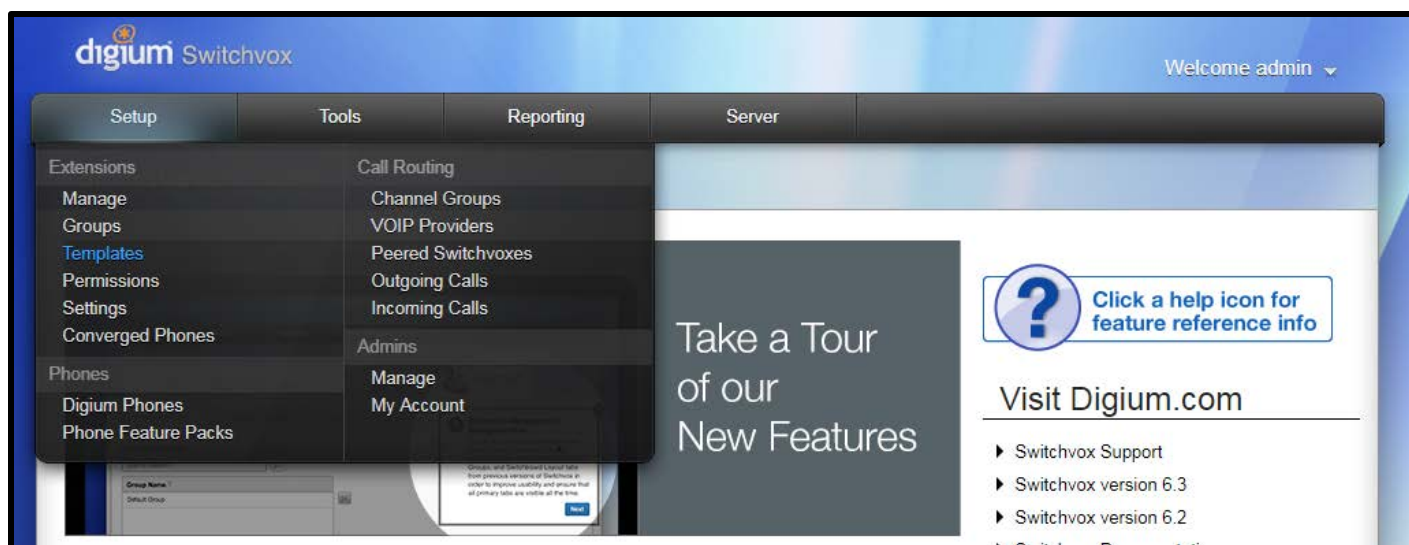
The Digium SwitchVox product range are a robust line of PBX appliances. Their devices range from small mini PCs to full rack mounted systems. The PBX is very user friendly and can be picked up very quickly. Please use the following steps to setup a template for CyberData devices.

1. Navigate to the SwitchVox’s IP address. Make sure do login to the Admin portion of the webpage.

Note To reach the admin portion of the webpage please add ‘/admin’ to the end of the ip address. It should appear like this, “https://IP_ADDRESS_HERE/admin

2. Login using the default login info **admin** for both username and password.
3. Once logged in please hover your mouse over the *Setup tab*.
4. Select the subsection *Templates*.

Figure 1: Creating a Template



5. Press 'Create Extension Template'.
6. Name the template.
7. Then select 'Phone Settings.'

Figure 2: Name the Template

The screenshot shows the 'Modify Extension Template' page in the Digium Switchvox admin interface. The 'Phone Settings' tab is selected. The 'Template Name' field is highlighted with a yellow box and contains the text 'Cyberdata Device'. Other fields include 'Location', 'Language / Locale' (set to 'United States'), 'Timezone' (set to 'System Default'), and 'Default Parking Lot' (set to '701 through 799'). Below these are 'Login Actions' with toggle switches for 'Force change of password on next login' and 'Force language confirmation on next login', both set to 'NO'. At the bottom is a 'Save Extension Template' button with a checkmark.

8. Click on the arrow for 'Advanced Phone Options.'
9. Confirm that *DTMF Mode* is set to **RFC 4733**.
10. Select the following Audio Codecs; **ULAW**, **ALAW** and **G722**.
11. Select the following Video Codec: **H264**.
12. Disable all other codecs.

Figure 3: Codecs used

The screenshot shows the 'Modify Extension Template' interface in the Digium Switchvox admin console. The 'Permissions' tab is selected. The 'Advanced Phone Options' section is expanded, showing 'Config Settings' with 'DTMF Mode' set to 'rfc4733 (Default)'. Below this, the 'Audio Codecs' section shows 'ULAW (default)', 'ALAW (default)', and 'G722 (default)' all turned 'ON'. The 'Video Codecs' section shows 'H264 (default)' turned 'ON'. Other codecs like G726, ADPCM, SPEEX, LPC10, and GSM are turned 'OFF'.

13. Select the 'Permissions' tab.
14. Select the drop-down arrow for 'Use Voicemail Features'.
15. Set 'Can Use Voicemail' to **No**.

Note This will disable all the settings for Voicemail.

Figure 4: Permissions

digium Switchvox Welcome admin ▾

Setup Tools Reporting Server

Modify SIP Extension ?

Profile Information Phone Settings **Permissions** Outgoing Call Rules Assignment

Feature and Configuration Permissions ?

- Change Profile Information (10/10 YES)
- Create Call Rules (6/6 YES)
- Use Voicemail Features (0/5 YES)**
 - ☐ NO Can Use Voicemail Mailbox
 - ☐ NO Receive voicemail attachments
 - ☐ NO Enable Automatic Voicemail Forwarding
 - ☐ NO Customize Voicemail Notifications
 - ☐ NO Screen Voicemail
- Use PBX Features (5/5 YES)
- Use Digium Phone Apps (5/6 YES)
- Change Digium Phone Settings (8/8 YES)
- Use Distinctive Ringtones (2/2 YES)
- Use Calling Features (4/5 YES)
- Change Custom Settings (0/0 YES)

Save SIP Extension ✓

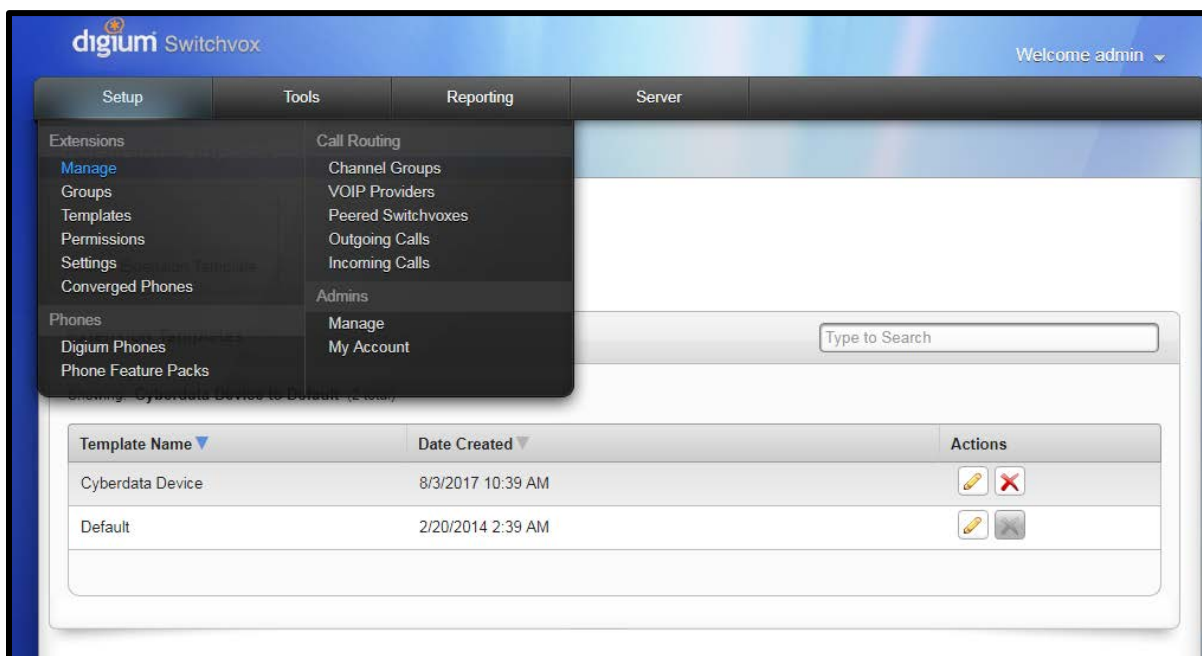
16. Press 'Save SIP Extension.'

Creating the Extensions

Now that the template for the CyberData devices has been created we can now create the extensions that will be used. To begin please press the create extension button and follow these steps.

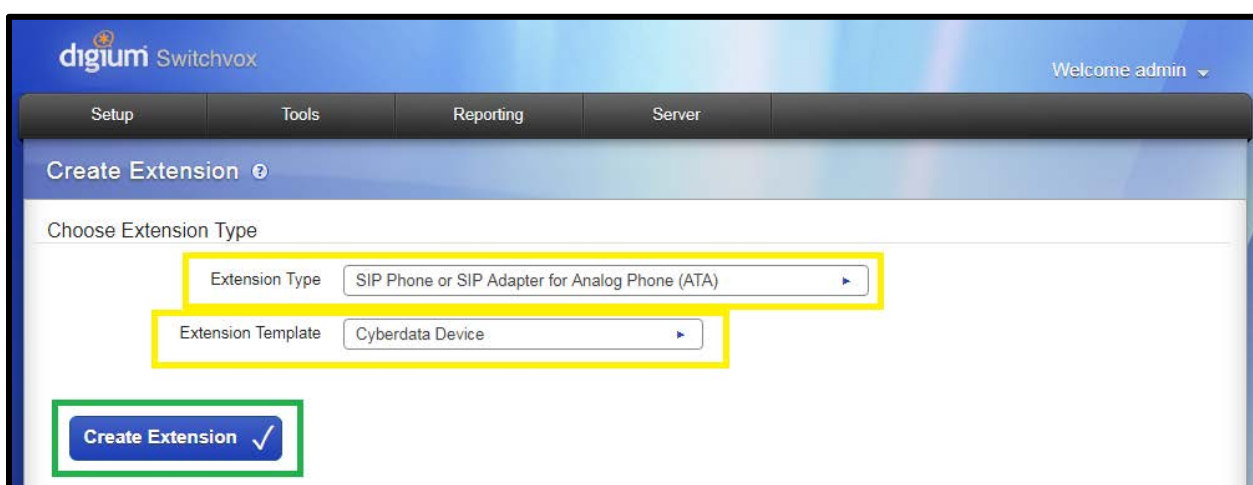
1. Hover over Setup and select manage from the drop-down menu.

Figure 5: Extension Creation



2. Press the **Create Extension** button.
3. Select **SIP Phone or SIP Adapter for Analog Phone (ATA)** for the extension type.
4. Select the template that was created in the last step.

Figure 6: Extension Type



5. Then please press Create Extension.
6. Pick a number for the extension.
7. Give a first and last name for the extension, for example CyberData Paging Server.

Figure 7: Create SIP Extension

The screenshot shows the 'Create SIP Extension' interface in the Digium Switchvox admin console. The 'Phone Settings' tab is selected. The 'Profile Information' section is visible, with the following fields and values:

- Extension: 625
- First Name: CyberData
- Last Name: Paging Server
- Email Address: (empty)
- Location: (empty)
- Title: (empty)
- Language / Locale: United States
- Timezone: System Default
- Password: (empty)
- Retype Password: (empty)
- Numeric PIN: (empty)
- Retype Numeric PIN: (empty)
- Default Parking Lot: 701 through 799

8. Navigate to the Phone Settings.
9. Then go to the Common Settings subsection.
10. Input a password for the device.

Note This password is used to authenticate the device against the server when registering.

Figure 8: Phone Password

The screenshot shows the Digium Switchvox admin interface. At the top, there's a navigation bar with 'Setup', 'Tools', 'Reporting', and 'Server'. Below this is a 'Create SIP Extension' section with tabs for 'Profile Information', 'Phone Settings', 'Permissions', 'Outgoing Call Rules', and 'Assignment'. The 'Phone Settings' tab is active, and within it, the 'Common Settings' sub-tab is selected. The 'General Settings' section contains several options: 'Auto-answer Switchboard initiated calls' (set to NO), 'Phone Password' (masked with dots), 'Retype Phone Password' (masked with dots), 'Phone NAT Traversal' (set to Always), and 'Rapid Dial List Assignment' (set to User Default (Custom)). The 'Phone Password' and 'Retype Phone Password' fields are highlighted with a yellow box. At the bottom left, the 'Save SIP Extension' button is highlighted with a green box.

11. Save the SIP Extension.

Configure the CyberData Device

With the extensions created the final step is to register the CyberData device with the PBX. To configure the CyberData device please use the following steps.

1. Point your browser to the IP address of your device.
2. Login using the default credential, username: **admin** and Password: **admin**.
3. Navigate to the *SIP tab*.
4. Set the **Primary SIP Server** with the IP address of the Digium SwitchVox.
5. Set both the **Primary SIP User ID** and the **Primary SIP Auth ID** as the extension that was used earlier.
6. Set the **Primary SIP Auth Password** as the password that was used for the phone.
7. Finally Save and Reboot the device.

Figure 9: SIP Configuration

CyberData v3.1 Paging Server

SIP Settings

Enable SIP operation: ☒
 Register with a SIP Server: ☒
 Use Cisco SRST: ☐
 Primary SIP Server: 10.0.1.60
 Primary SIP User ID: 625
 Primary SIP Auth ID: 625
 Primary SIP Auth Password: *****
 Backup SIP Server 1:
 Backup SIP User ID 1:
 Backup SIP Auth ID 1:
 Backup SIP Auth Password 1:
 Backup SIP Server 2:
 Backup SIP User ID 2:
 Backup SIP Auth ID 2:
 Backup SIP Auth Password 2:
 Remote SIP Port: 5060
 Local SIP Port: 5060
 Outbound Proxy:
 Outbound Proxy Port: 0
 Disable rport Discovery: ☐
 Buffer SIP Calls: ☐
 Re-registration Interval (in seconds): 360
 Unregister on Boot: ☐
 Keep Alive Period: 10000

Nighthringer Settings

Enable Nighthringer: ☐
 SIP Server: 10.0.0.253
 Remote SIP Port: 5060
 Local SIP Port: 5061
 Outbound Proxy:
 Outbound Proxy Port: 0
 User ID: 241
 Authenticate ID: 241
 Authenticate Password: *****
 Re-registration Interval (in seconds): 360
 Relay rings to multicast: ☐
 Multicast Address: 224.1.2.32
 Multicast Port: 2020

Call Disconnection

Terminate Call after delay: 0

Codec Selection

Force Selected Codec: ☐
 Codec: PCMU (G.711, u-law)

RTP Settings

RTP Port (even): 10500
 Jitter Buffer: 50

Save Reboot Toggle Help

Resources

CyberData Products Page

<http://www.cyberdata.net/voip-category/sip/>

CyberData VoIP Discovery Tool

<http://www.cyberdata.net/assets/common/discovery.zip>

Digium Page

<https://www.digium.com/products/business-phone-systems>

Contact CyberData Corporation



Technical Support

The fastest way to get technical support for your VoIP product is to submit a VoIP Technical Support form at the following website:

<http://support.cyberdata.net/>

Phone: (831) 373-2601, Ext. 333

Email: support@cyberdata.net

Fax: (831) 373-4193

Company and product information is at www.cyberdata.net