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SIP Trunking using the EdgeMarc Network Services Gateway and the Digium Switchvox IP-PBX

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Table of Contents

- 1 Overview 3
- 2 Prerequisites 3
- 3 Network Topology 4
- 4 Description of Basic Operation and Call Flows 5
- 5 Digium Switchvox PBX Configuration 5
 - 5.1 Default IP Address 6
 - 5.2 Web GUI Access 6
 - 5.3 Username and Password 7
 - 5.4 Network Settings 7
 - 5.5 System Setup 8
 - 5.6 Static IP Mode 10
 - 5.7 Extension Length 11
 - 5.8 Create A New Extension 12
 - 5.9 Manage Extensions 16
 - 5.10 Phone Setup 16
 - 5.11 Incoming Calls 20
 - 5.12 Outgoing Call Rules 22
 - 5.13 Outgoing Caller ID 24
 - 5.14 Static IP Outgoing Caller ID 24
 - 5.15 Auto Attendant 26

1 Overview

The purpose of this knowledgebase solution is to describe the steps needed to configure the Digium Switchvox AA65 IP-PBX for proper operation in a SIP trunking application. Please note that this solution documents the basic configuration needed in the PBX and that the requirements of your specific SIP trunking environment may require modifications to the configuration steps provided in this document

2 Prerequisites

SIP trunking information provided by the VoIP service provider:

- SIP proxy server IP address or DNS name.
- Trunking Direct Inward Dial (DID) phone numbers
 - Calls to the trunking DID(s) are forwarded from the service provider to the wide area network (WAN) IP address of the EdgeMarc. There may be a single "Pilot" phone number used for all inbound calls and/or multiple DIDs depending on the service ordered.
- SIP authentication credentials (optional)
 - Some SIP trunking service providers require a unique username and password to be supplied for IP PBX registrations and/or SIP signaling using P-Asserted Identity (RFC 3325). This knowledgebase solution provides the configuration steps for both PBX registration and static or non-registration modes of PBX operation.
- Digium Switchvox – v23695

3 Network Topology

Typical SIP Trunking Installation

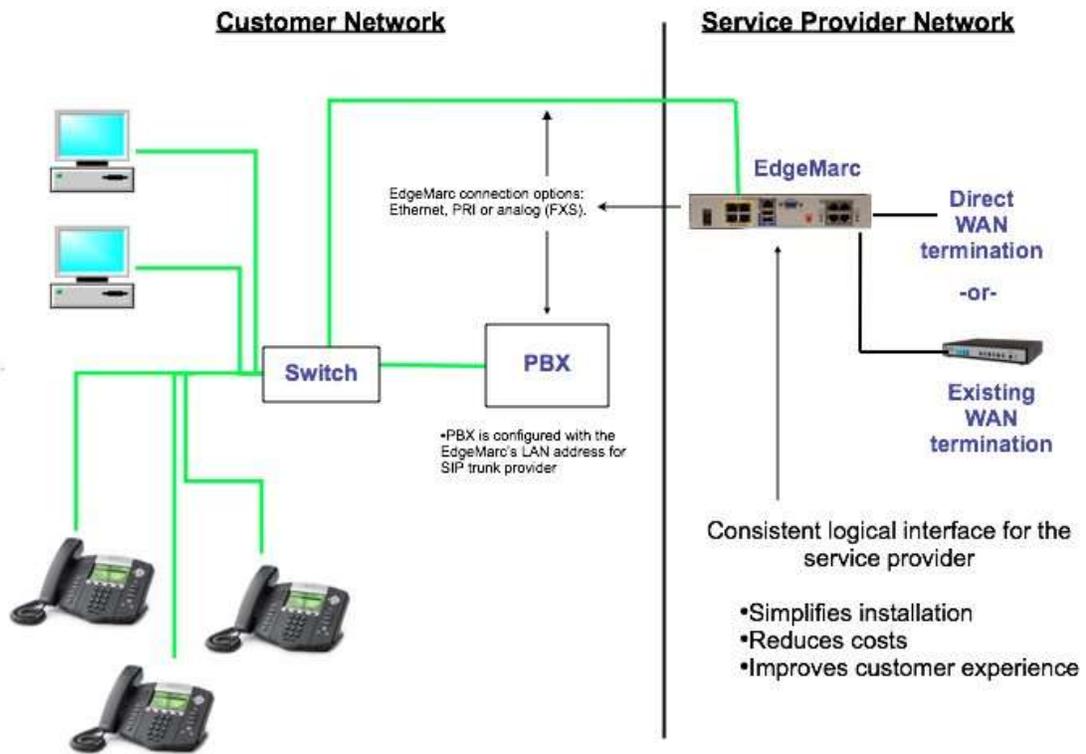


Figure 1 Test Set up

The PBX in the above network topology represents the Digium Switchvox PBX that is connected via its LAN port to the LAN port of the EdgeMarc Network Services gateway. The PBX used in our lab comprises of the following:

Table 1 – PBX Information

Manufacturer:	Asterisk
Model:	Digium Switchvox
Software Version:	23695
Does the PBX send SIP Registration messages (Yes/No)?	Yes
Vendor Contact:	support@netxusa.com

Table 2 – E-SBC Information

Manufacturer:	Edgewater Network, Inc.
Model:	4552
Software Version:	11.6.14

4 Description of Basic Operation and Call Flows

Basic Call Flow:

All phones connect to the Digium Switchvox AA65 PBX. The PBX will interface with the service provider using a SIP trunk.

Internal calls:

- Calls between phones on the LAN
- LAN phone > Digium Switchvox AA65 PBX > LAN phone

Outbound calls:

- Call is initiated by a LAN phone to a WAN phone.
- LAN phone > Digium Switchvox AA65 PBX [SIP trunk] > EM > SIP trunk service provider > WAN phone

Inbound call:

- Call is initiated by a WAN phone to a LAN phone.

WAN phone > SIP trunk service provider > EM > [SIP trunk] Digium Switchvox AA65 PBX > LAN phone

5 Digium Switchvox PBX Configuration

The steps below describe the minimum configuration required to enable the PBX to use a SIP trunk for inbound and outbound calling. Please refer to the Digium Switchvox AA65 product documentation for more information on SIP trunking or other advanced PBX features.

The configuration described here assumes that the PBX is already configured and operational with station side phones using assigned extensions or DIDs. This configuration is based on Digium Switchvox AA65 version 23695.

5.1 Default IP Address

The IP-PBX was shipped with a default IP address of 192.168.1.100/24 for 1 Ethernet port. To work with EM for SIP trunking service, this port should be in the same subnet as EM port 1 and use EM port 1's IP address as its SIP server. All the IP phones communicate with the PBX via this IP address as well, using the SIP VoIP protocol. The IP phones by default will need IP address assignment. For the test setup in the lab, the PBX's default IP address is changed to 10.10.108.11/24 and EM port 1 is set to 10/10/108.1/24, with DHCP server enabled.

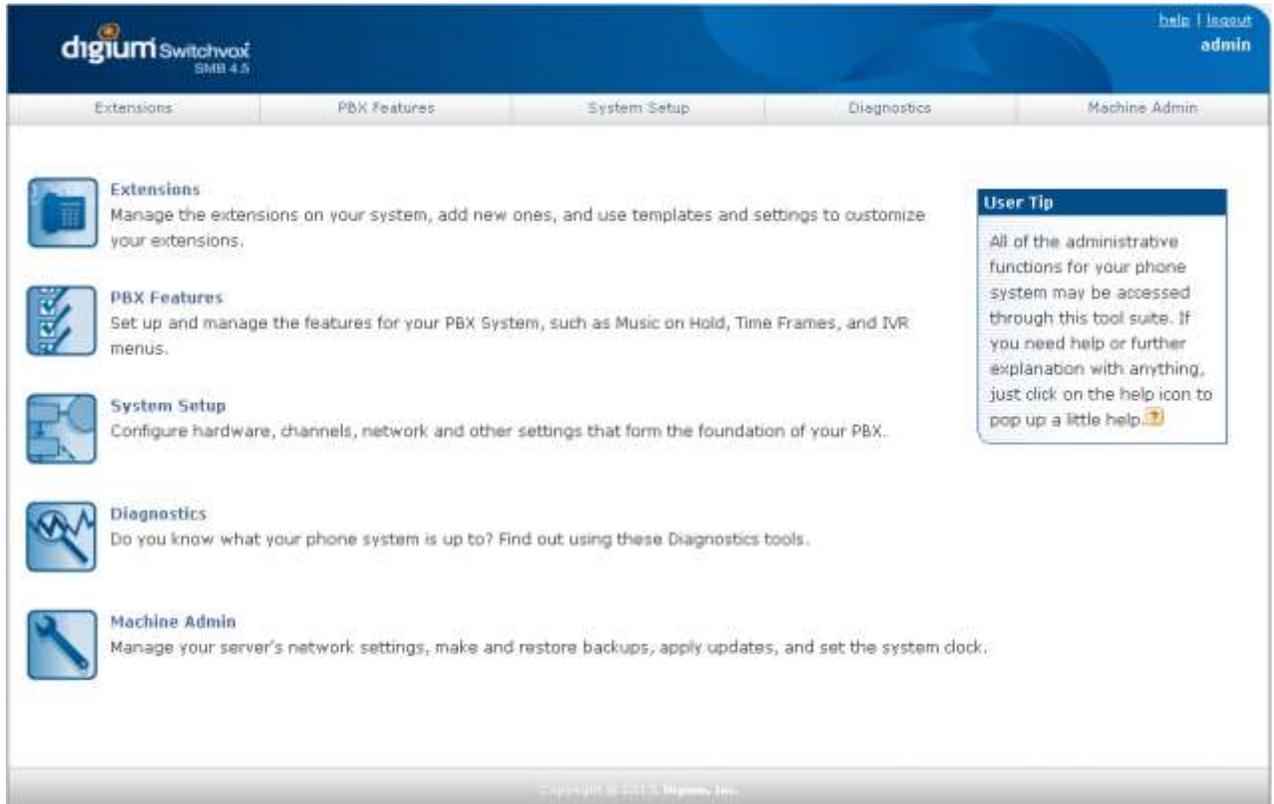
5.2 Web GUI Access

To configure the PBX, run <https://10.10.108.11/admin> on your PC and to access the configuration GUI's login screen.

A screenshot of the Digium Switchvox SMB 4.5 login screen. The page has a blue header with the Digium logo and "Switchvox SMB 4.5" text. Below the header, the text "Please log in to use the Tool Suite" is centered. There are two input fields: "User Name:" and "Password:". Below the password field is a blue "Log In" button. At the bottom of the page, there is a small copyright notice: "Copyright © 2015, Digium, Inc." data-bbox="144 378 929 617"/>

5.3 Username and Password

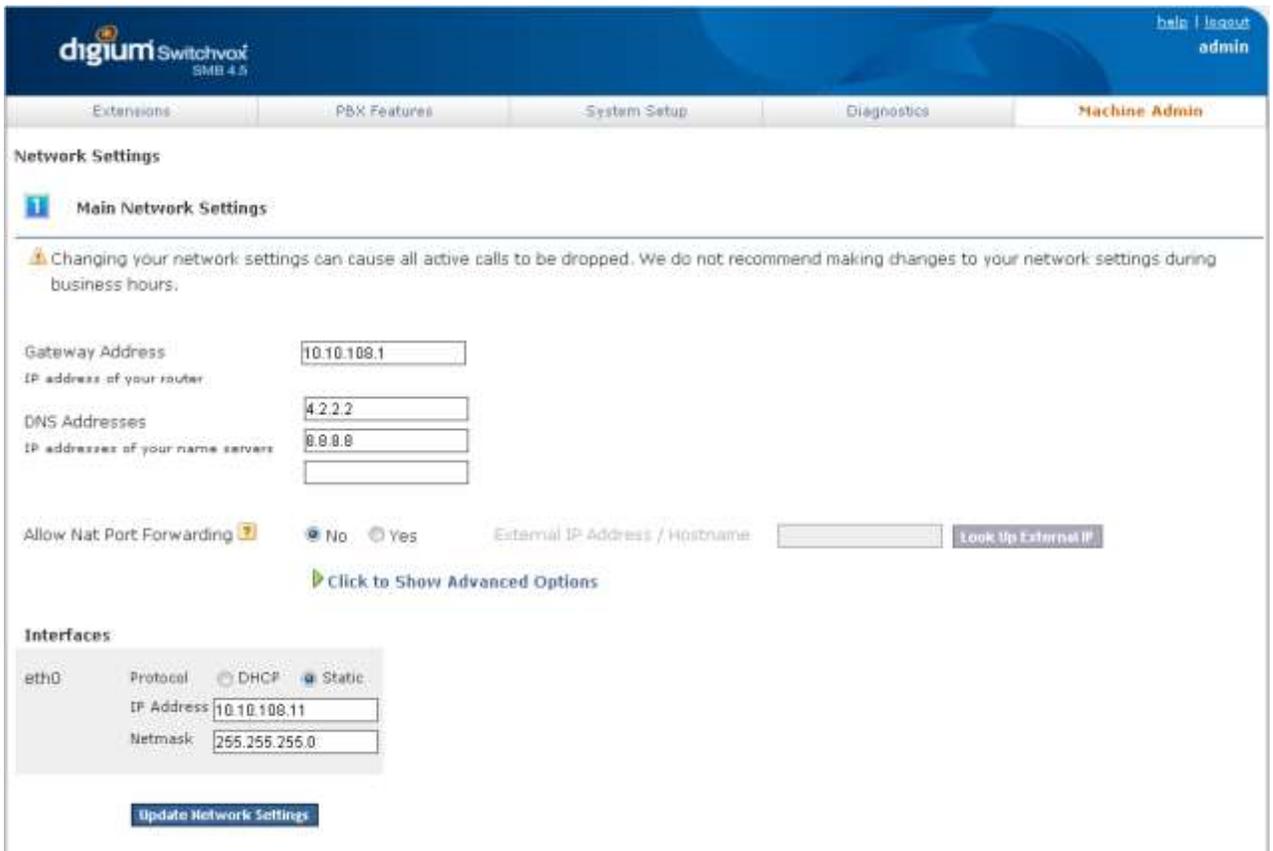
Enter the user name and password for the PBX and hit the “Login” button. The factory default is “admin” for both the user name and password.



5.4 Network Settings

Select “Machine Admin” and select “Network Settings” to set/verify network setting.

- a) Enter EdgeMarc’s IP address in the “Gateway Address” field.
- b) Enter the Primary and Secondary DNS IP address in the “DNS Addresses” fields.
- c) Select “No” for the “Allow Nat Port Forwarding” setting.
- d) In the Interface section, select “Static” for eth0’s “Protocol” field and make sure the IP address and network mask of the PBX are correct in the “IP Address” and the “Netmask” fields.
- e) Hit the “Update Network Settings” button.



Network Settings

1 Main Network Settings

⚠ Changing your network settings can cause all active calls to be dropped. We do not recommend making changes to your network settings during business hours.

Gateway Address
IP address of your router

DNS Addresses
IP addresses of your name servers

Allow Nat Port Forwarding No Yes External IP Address / Hostname [Look Up External IP](#)

[Click to Show Advanced Options](#)

Interfaces

eth0 Protocol DHCP Static
 IP Address
 Netmask

[Update Network Settings](#)

5.5 System Setup

Select "System Setup", select "VoIP Providers", select "SIP Provider" in the "Add New" field and hit the "Go" button to configure EM as the SIP provider, expecting SIP registration from the PBX.

- a) Enter a descriptive name in the "SIP Provider Name" field.
- b) Enter Account ID in the "Your Account ID" field. Note that by default this ID is used for PBX registration and it must match the "User ID" configured on EdgeMarc. In this example, the pilot DID is used.
- c) Enter password in the "Your Password" field.
- d) Enter EdgeMarc's IP address in the "Hostname/IP Address" field.
- e) Enter "800" in the "Callback Extension" field.
- f) Select "RFC2833" in the "DTMF Mode" field.
- g) Leave other fields as default and hit the "Add SIP Provider" button.

digium Switchvox SMB 4.5 [help](#) | [logout](#) | [admin](#)

Extensions | PBX Features | **System Setup** | Diagnostics | Machine Admin

VOIP Providers

Add a New SIP Provider

SIP Provider Name
[?](#) What is this used for?

Your Account ID
[?](#) What's an Account ID?

Your Password

Hostname/IP Address
[?](#) What does this mean?

Callback Extension [?](#) What's the Callback Extension?

Default Fax Extension [?](#) What is this used for?

DTMF Mode
[?](#) What is DTMF Mode?

- RFC2833
- RFC2833**
- Info
- Inband
- Auto

[Show Advanced Options](#)

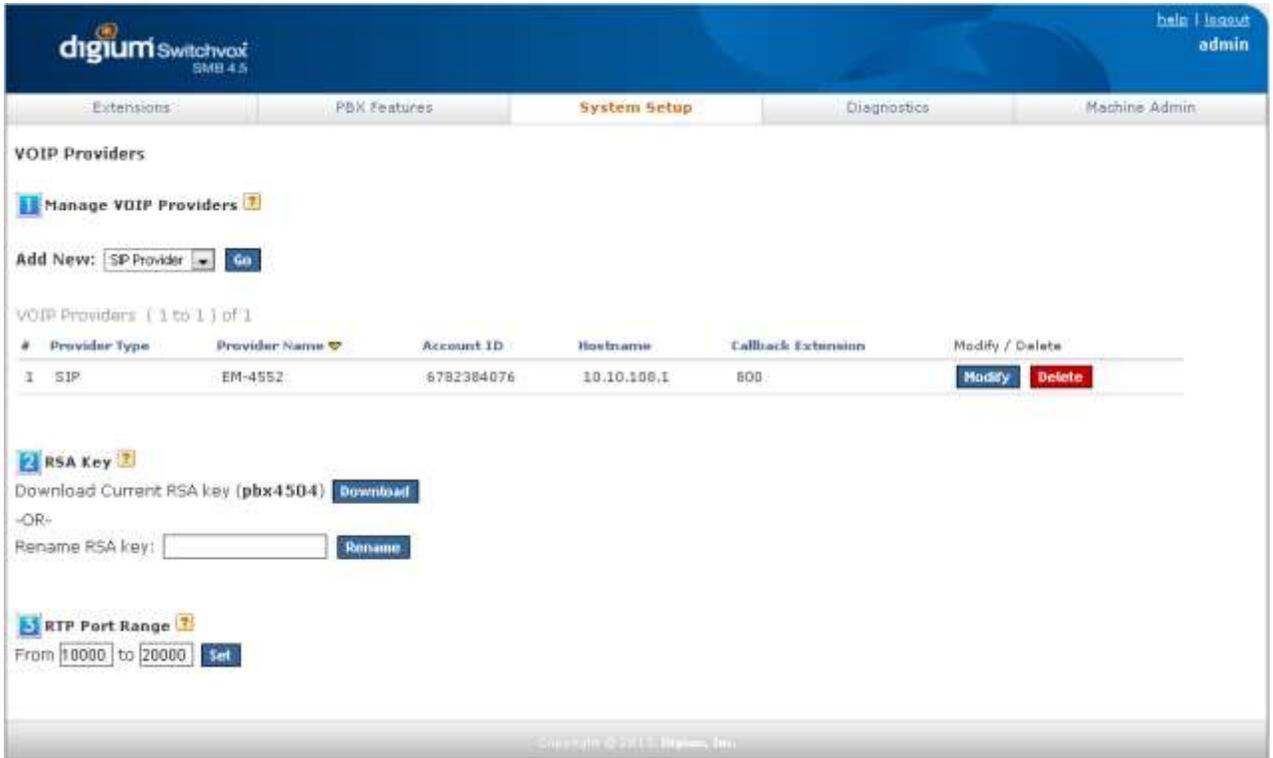
[Add SIP Provider](#)

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5.6 Static IP Mode

If you need to configure the PBX for static IP mode, select "System Setup" and select "VoIP Providers".

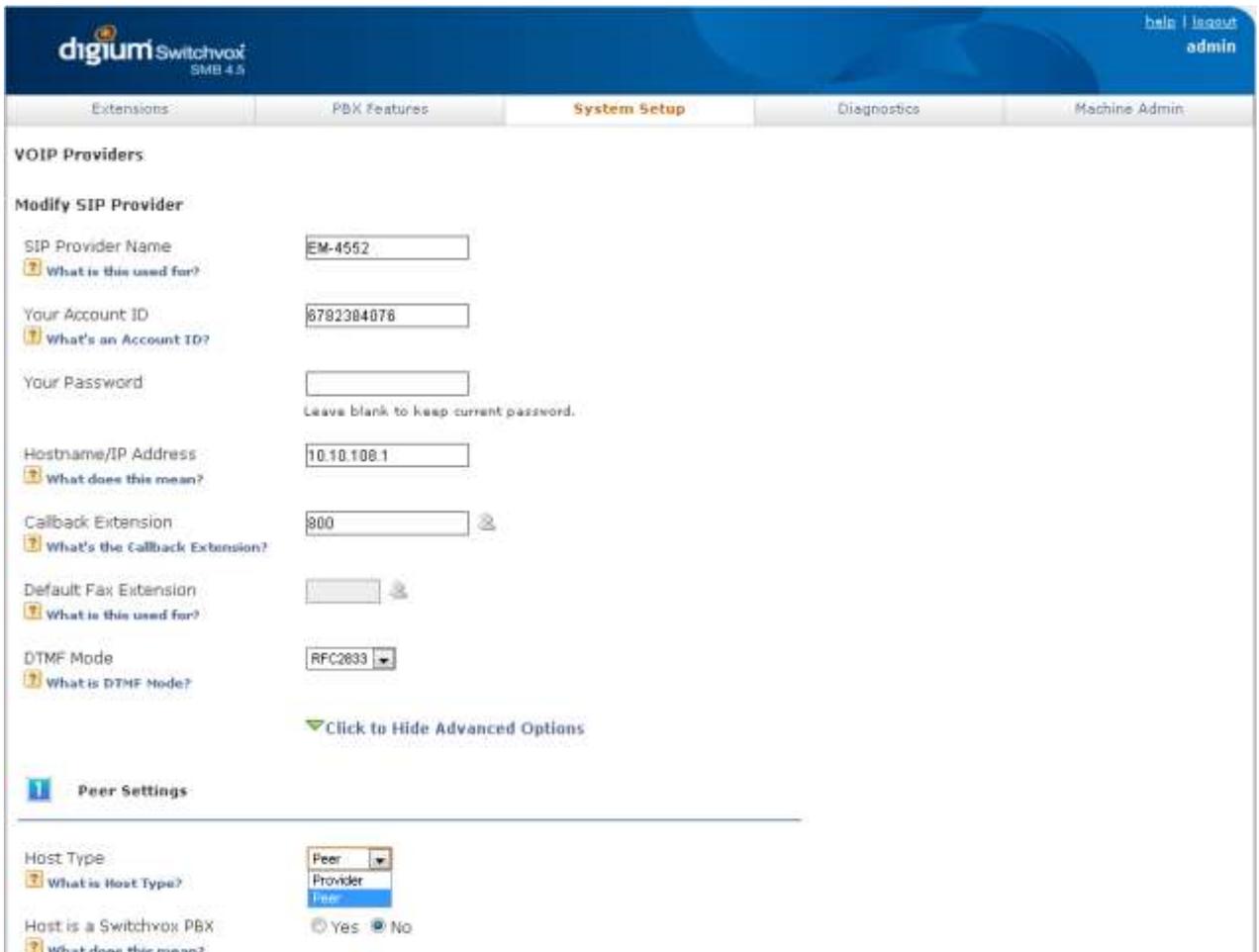
- a) Hit the "Modify" button to modify the VoIP Provider account set up for EdgeMarc.



The screenshot shows the "digium Switchvox SMB 4.5" interface. The "System Setup" tab is active, and the "VoIP Providers" section is displayed. It includes a "Manage VoIP Providers" button, an "Add New" dropdown menu set to "SIP Provider", and a table of providers. The table has columns for #, Provider Type, Provider Name, Account ID, Hostname, Callback Extension, and Modify / Delete. One provider is listed with Account ID 6782384076 and Hostname 10.10.108.1. Below the table are sections for "RSA Key" (with a "Download" button) and "RTP Port Range" (with a "Set" button).

#	Provider Type	Provider Name	Account ID	Hostname	Callback Extension	Modify / Delete
1	SIP	EM-4552	6782384076	10.10.108.1	600	Modify Delete

- b) Click the "Click to Show Advanced Options".
- c) In the Peer Settings section, change "Host Type" field from "Provider" to "Peer". This will change the PBX from default of SIP registration mode to static IP mode.
- d) Leave all other fields as default and hit the "Modify SIP Provider" button at the end of the screen.



5.7 Extension Length

Select "Extensions", select "Extension Settings", select "3" for the "Extension Length" field and hit the "Save Extension Settings" button.

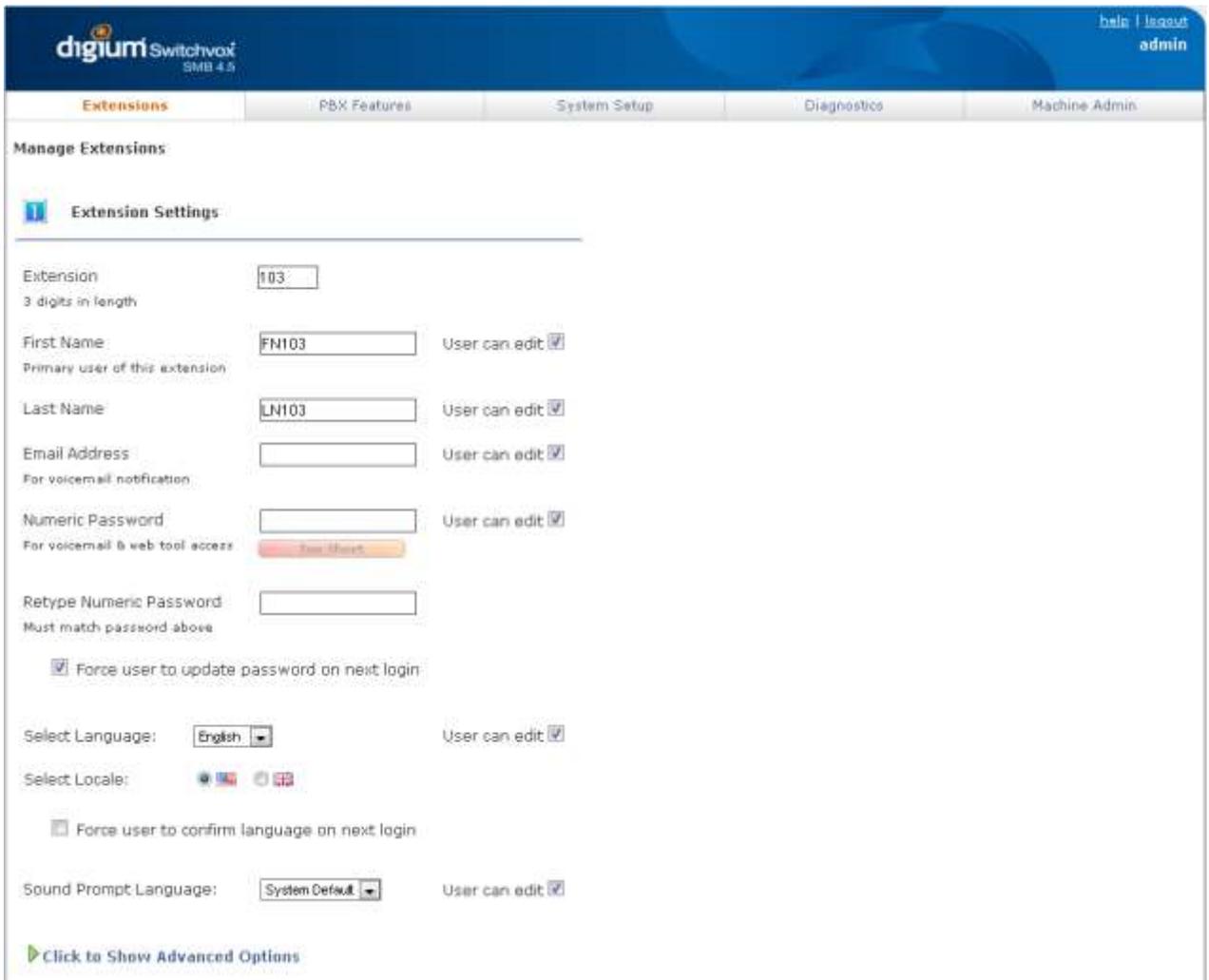


5.8 Create A New Extension

Select "Extensions", select "Manage Extensions" and click on the "Create A New Extension" button to create SIP extensions for the SIP phones.



- a) Leave both the "Extension Type" and the "Extension Template" fields as default and hit the "Create A New Extension" button to create a new extension.
- b) In the Extension Settings section, enter the extension number in the "Extension" field
- c) Enter the first name of the user in the "First Name" field and enter the last name of the user in the "Last Name" field.
- d) Leave other fields as default and hit the "Click to Show Advanced Options" link.



The screenshot shows the 'Manage Extensions' section of the Digium Switchvox SMB 4.5 web interface. The 'Extension Settings' tab is active, showing configuration for extension 103. The fields include:

- Extension:** 103 (3 digits in length)
- First Name:** FN103 (User can edit)
- Last Name:** LN103 (User can edit)
- Email Address:** (User can edit)
- Numeric Password:** (User can edit)
- Retype Numeric Password:** (Must match password above)
- Force user to update password on next login:**
- Select Language:** English (User can edit)
- Select Locale:** (Force user to confirm language on next login:)
- Sound Prompt Language:** System Default (User can edit)

At the bottom of the form, there is a link: [Click to Show Advanced Options](#)

- e) Select "rfc2833 (Default)" in the "DTMF Mode" field.
- f) Enter the same phone password in both the "Phone Password" field and the "Retype Phone Password" field. Note that this password must match the "Authentication Password" setting of the Polycom phone. Also note that "456" has been commonly used as the Polycom phone's password.

▼ Click to Hide Advanced Options

General Settings

Voice/Fax Mailbox Quota (MB)

 [What does this mean?](#)

(A megabyte is approximately equal to 1 minute of voicemail)

This extension can be dialed from an IVR.

 [What does this mean?](#)

Phone Settings

DTMF Mode

 [What is DTMF Mode?](#)

- rfc2833 (Default)
- rfc2833 (Default)
- Inband
- Info

Phone Password

 [What is Phone Password?](#)

Strong

Retype Phone Password

Must match password above

Supported Codecs

 [Help with codecs](#)

- Audio** ULAW (Default) ALAW (Default) G722 (Default)
 G726 SPEEX GSM
 ADPCM LPC10
- Video** H263 (Default) H263+ H264 (Default)

Phone NAT Traversal

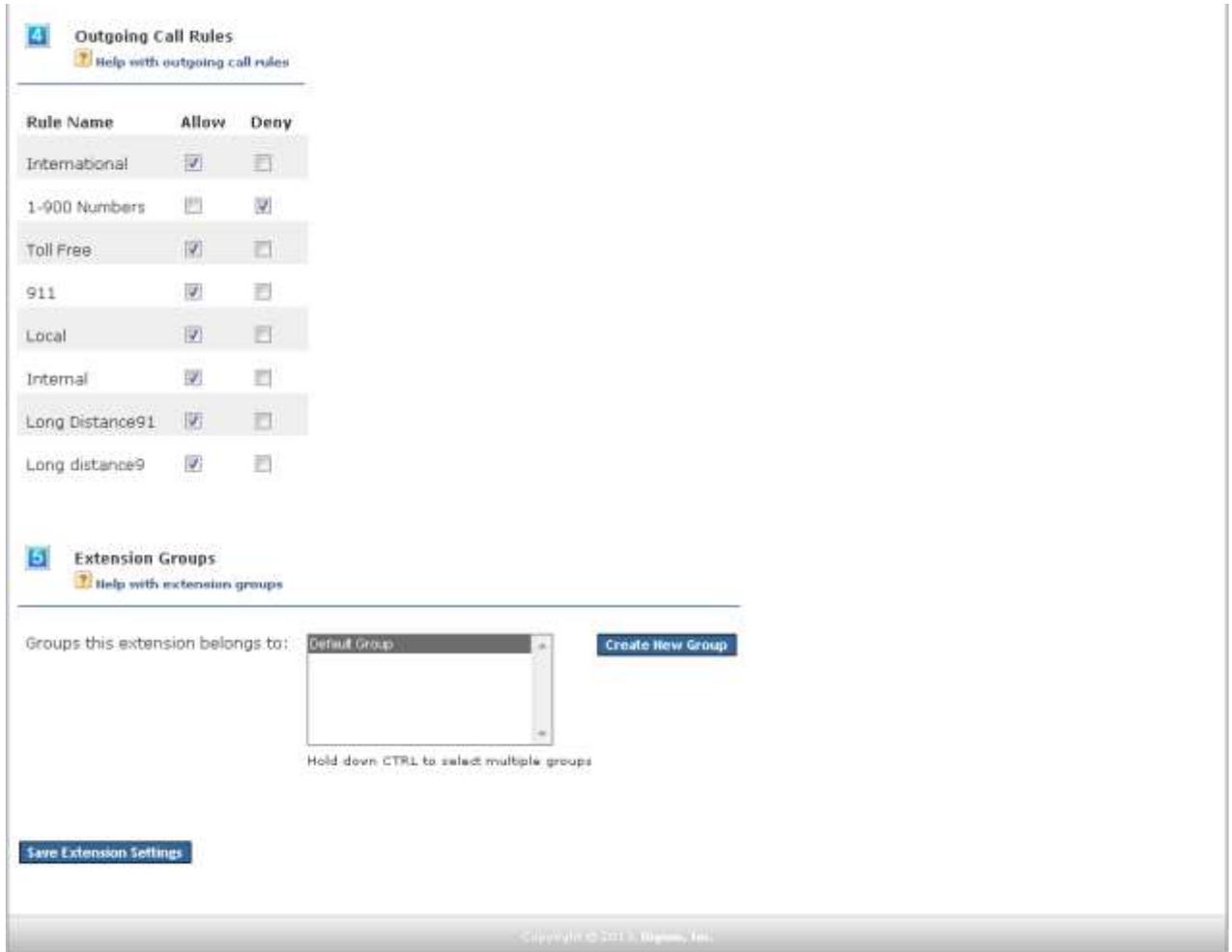
 [What does this mean?](#)

Call API Settings

Prepend a 1 if number is 10 digits in length

Digits to prepend if number is not an extension

g) Leave all other fields as default and hit the "Save Extension Settings" button at the end of the screen.



The screenshot shows a web interface for configuring outgoing call rules and extension groups. It is divided into two main sections: 'Outgoing Call Rules' and 'Extension Groups'.

Outgoing Call Rules

4 Outgoing Call Rules
Help with outgoing call rules

Rule Name	Allow	Deny
International	<input checked="" type="checkbox"/>	<input type="checkbox"/>
1-900 Numbers	<input type="checkbox"/>	<input checked="" type="checkbox"/>
Toll Free	<input checked="" type="checkbox"/>	<input type="checkbox"/>
911	<input checked="" type="checkbox"/>	<input type="checkbox"/>
Local	<input checked="" type="checkbox"/>	<input type="checkbox"/>
Internal	<input checked="" type="checkbox"/>	<input type="checkbox"/>
Long Distance91	<input checked="" type="checkbox"/>	<input type="checkbox"/>
Long distance9	<input checked="" type="checkbox"/>	<input type="checkbox"/>

Extension Groups

5 Extension Groups
Help with extension groups

Groups this extension belongs to: [Create New Group](#)

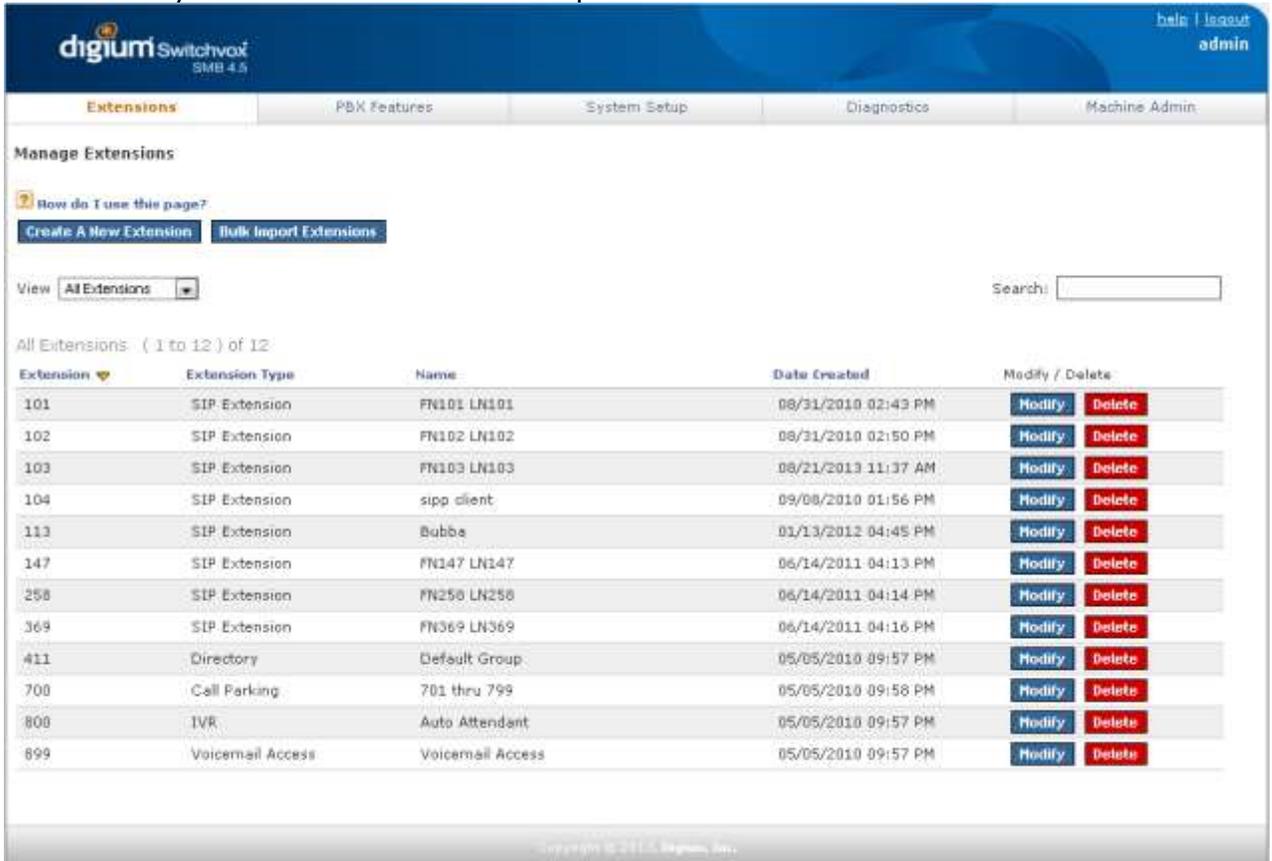
Hold down CTRL to select multiple groups

[Save Extension Settings](#)

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5.9 Manage Extensions

Select "Extension", select "Manage Extensions" to see all the default extensions and all the newly added extensions for the phones.



digium Switchvox SMB 4.5

help | logout admin

Extensions PBX Features System Setup Diagnostics Machine Admin

Manage Extensions

How do I use this page?

Create A New Extension Bulk Import Extensions

View: All Extensions Search:

All Extensions: (1 to 12) of 12

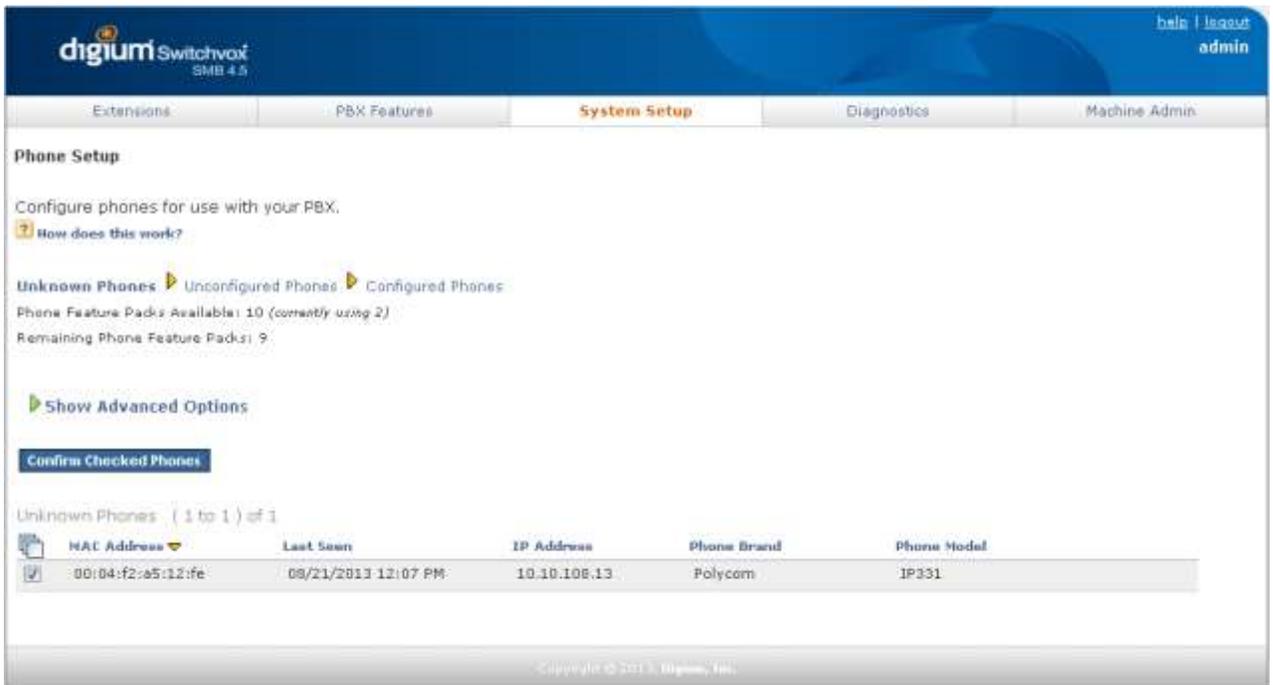
Extension	Extension Type	Name	Date Created	Modify / Delete
101	SIP Extension	FN101 LN101	08/31/2010 02:43 PM	Modify Delete
102	SIP Extension	FN102 LN102	08/31/2010 02:50 PM	Modify Delete
103	SIP Extension	FN103 LN103	08/21/2013 11:37 AM	Modify Delete
104	SIP Extension	sipp client	09/08/2010 01:56 PM	Modify Delete
113	SIP Extension	Bubba	01/13/2012 04:45 PM	Modify Delete
147	SIP Extension	FN147 LN147	06/14/2011 04:13 PM	Modify Delete
258	SIP Extension	FN258 LN258	06/14/2011 04:14 PM	Modify Delete
369	SIP Extension	FN369 LN369	06/14/2011 04:16 PM	Modify Delete
411	Directory	Default Group	05/05/2010 09:57 PM	Modify Delete
700	Call Parking	701 thru 799	05/05/2010 09:58 PM	Modify Delete
800	IVR	Auto Attendant	05/05/2010 09:57 PM	Modify Delete
899	Voicemail Access	Voicemail Access	05/05/2010 09:57 PM	Modify Delete

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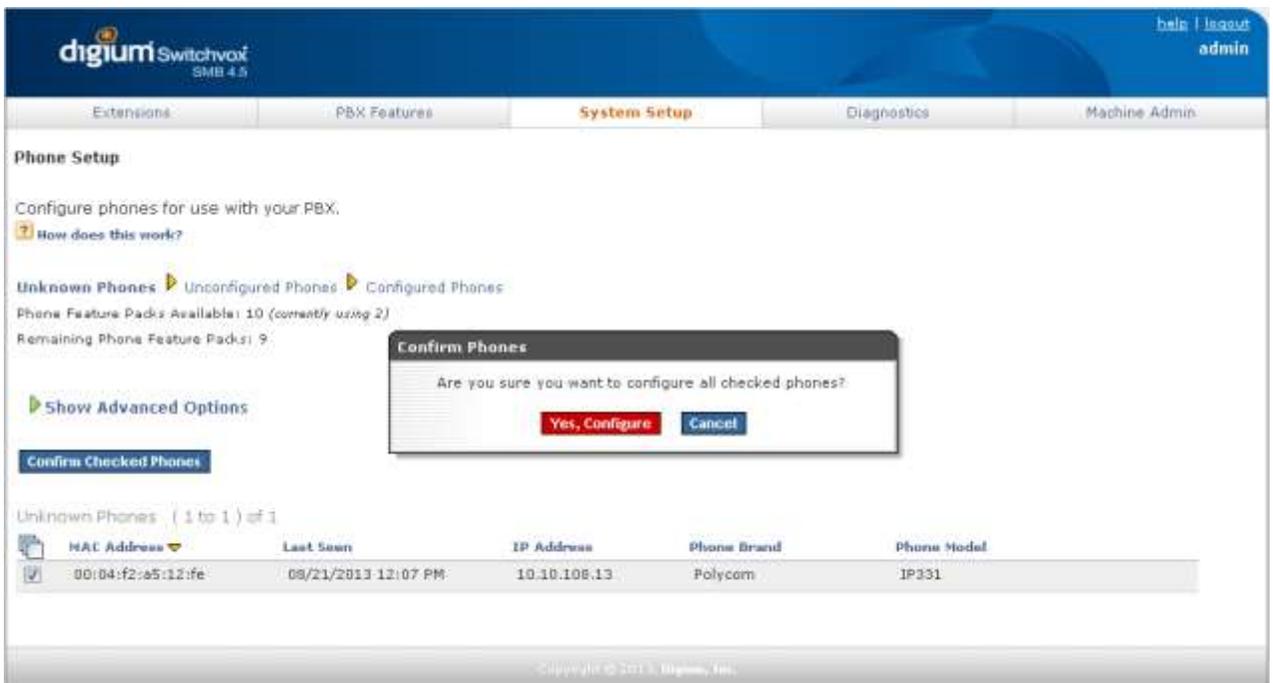
5.10 Phone Setup

Select "System Setup", select "Phone Setup" to assign SIP extensions to the SIP phones. In this example, the PBX has detected a Polycom phone and has added it to the "Unknown Phones" list. If a phone does not show up here, try rebooting the phone. Click on the "How does this work?" link if you need help on configuring the phones for use with your PBX.

- a) To configure the Polycom phone, check the box next to it and click the "Confirm Checked Phones" button. This should move the Polycom phone to the "Unconfigured Phones" list.

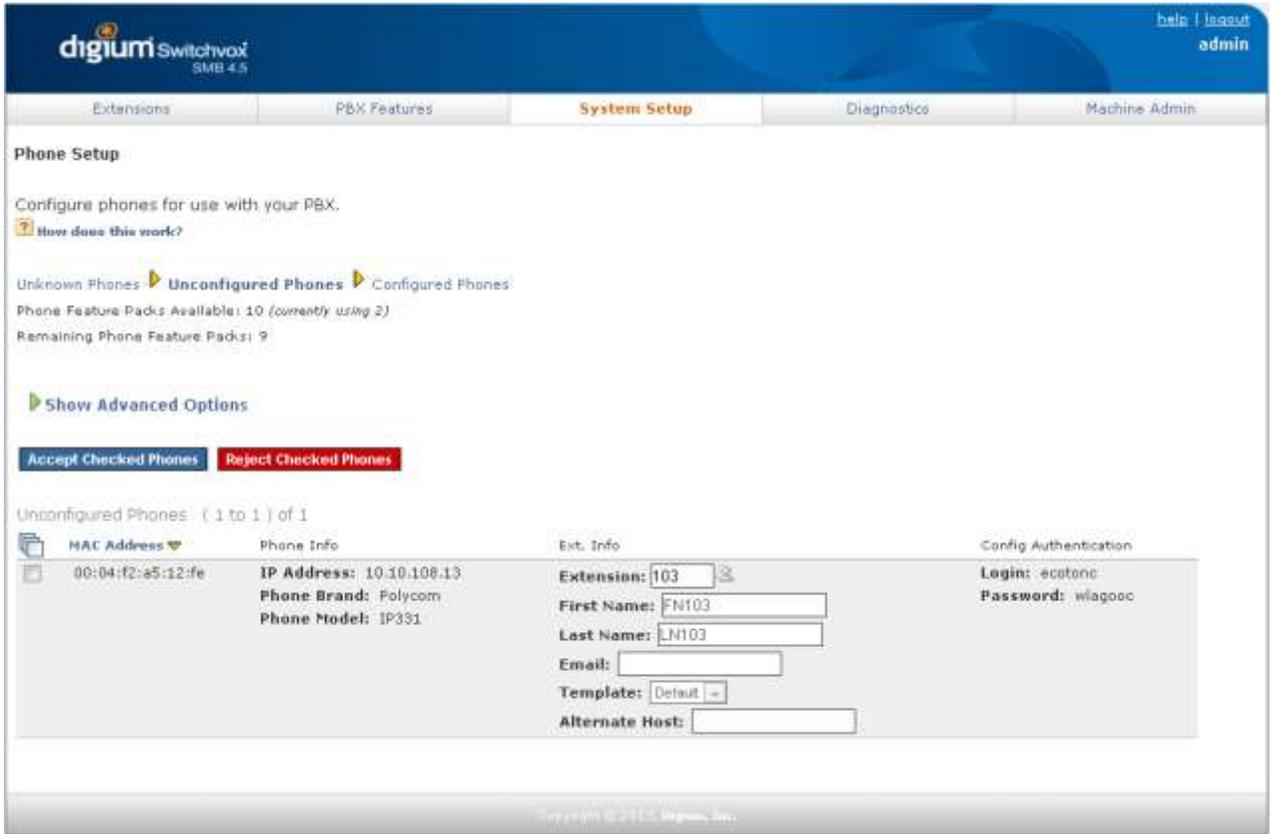


b) Click the “Yes, Configure” button.



c) The phone is now in the “Unconfigured Phones” list. Check the box next to the phone, enter the assigned extension number for the phone in the “Extension”

field, enter the first name of the user in the "First Name" field, enter the last name of the user in the "Last Name" field, leave other fields as default and hit the "Accept Checked Phones" button.



- d) Select "System Setup", select "Phone Setup" and click on "Configured Phones" and you should see the phones you just configured are now in the "Configured Phones" list.

help | logout
admin
digium Switchvox
SMB 4.5

Extensions
PBX Features
System Setup
Diagnostics
Machine Admin

Phone Setup

Configure phones for use with your PBX.
[How does this work?](#)

Unknown Phones ▾ Unconfigured Phones ▾ **Configured Phones**

Phone Feature Packs Available: 10 (currently using 2)
 Remaining Phone Feature Packs: 0

[Show Advanced Options](#)

Unconfigure Checked Phones

Reboot Checked Phones

Configured Phones (1 to 2) of 2

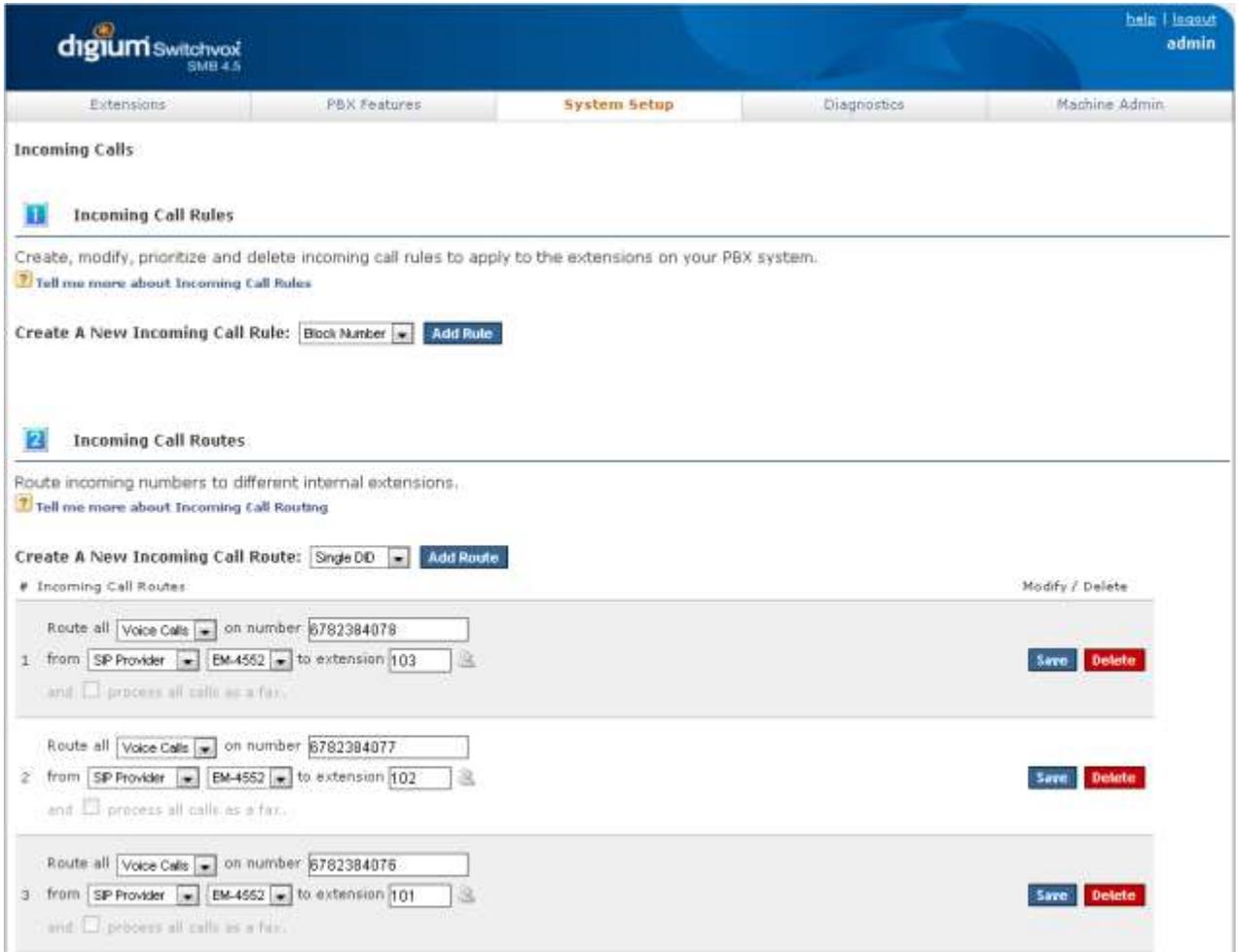
MAC Address	Phone Info	Ext. Info	Config Authentication
<input type="checkbox"/> 00:04:f2:a5:12:fe	IP Address: 10.10.108.13 Phone Brand: Polycorn Phone Model: IP331	Extension: 103 First Name: FN103 Last Name: LN103 Email: Alternate Host:	Login: ecotone Password: wlaqooe
Show Additional Lines			
<input type="checkbox"/> 00:04:f2:a5:14:7f	IP Address: 10.10.108.12 Phone Brand: Polycorn Phone Model: IP331	Extension: 147 First Name: FN147 Last Name: LN147 Email: Alternate Host:	Login: odzmhrq Password: jatcton
Show Additional Lines			

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5.11 Incoming Calls

Select "System Setup" and select "Incoming Calls" to map incoming numbers to different internal extensions.

- a) In the Incoming Call Routes section, select "Single DID" in the "Create A New Incoming Call Route" field and hit the "Add Route" button.



The screenshot shows the digium Switchvox SMB 4.5 web interface. The top navigation bar includes "Extensions", "PBX Features", "System Setup" (highlighted), "Diagnostics", and "Machine Admin". The "Incoming Calls" section is active, showing "Incoming Call Rules" and "Incoming Call Routes". Under "Incoming Call Routes", there is a "Create A New Incoming Call Route" section with a dropdown menu set to "Single DID" and an "Add Route" button. Below this, a table lists existing routes:

#	Incoming Call Routes	Modify / Delete
1	Route all Voice Calls on number 6782384078 from SIP Provider EM-4552 to extension 103 and <input type="checkbox"/> process all calls as a fax.	Save Delete
2	Route all Voice Calls on number 6782384077 from SIP Provider EM-4552 to extension 102 and <input type="checkbox"/> process all calls as a fax.	Save Delete
3	Route all Voice Calls on number 6782384076 from SIP Provider EM-4552 to extension 101 and <input type="checkbox"/> process all calls as a fax.	Save Delete

- b) From the new entry (high-lighted), select "Voice Calls" in the "Route all" field, enter the assigned DID for the "number" field, select "SIP Provider" in the "from" field, enter the phone extension in the "extension" field and hit the "Save" button.

help | logout
admin
digium Switchvox
SMB 4.5

Extensions
PBX Features
System Setup
Diagnostics
Machine Admin

Incoming Calls
✓ Successfully updated rule.

1 Incoming Call Rules

Create, modify, prioritize and delete incoming call rules to apply to the extensions on your PBX system.
Tell me more about Incoming Call Rules

Create A New Incoming Call Rule: Block Number Add Rule

2 Incoming Call Routes

Route incoming numbers to different internal extensions.
Tell me more about Incoming Call Routing

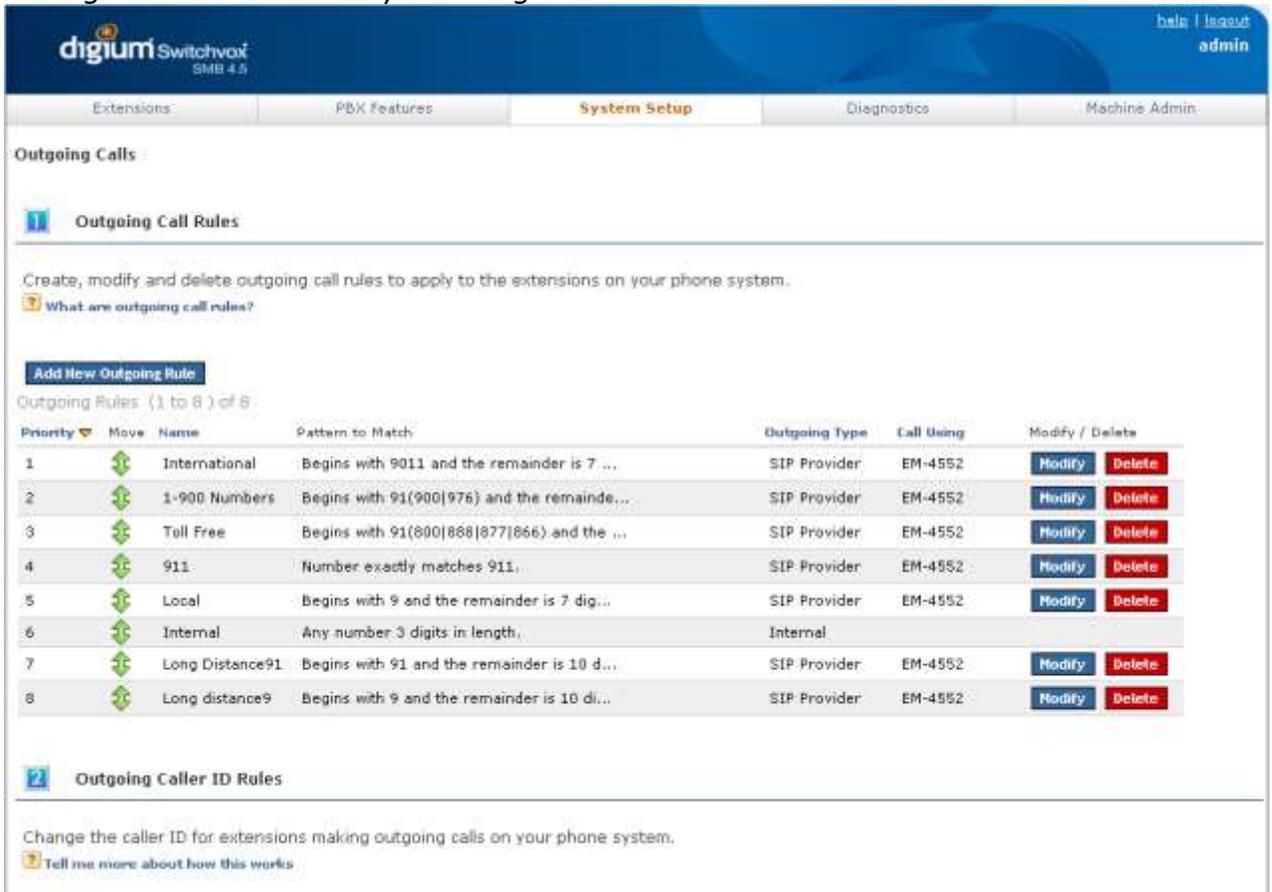
Create A New Incoming Call Route: Single DID Add Route

Incoming Call Routes Modify / Delete

Route all Voice Calls on number 6782384080 1 from SP Provider EM-4552 to extension 104 and <input type="checkbox"/> process all calls as a fax.	Save Delete
Route all Voice Calls on number 6782384078 2 from SP Provider EM-4552 to extension 103 and <input type="checkbox"/> process all calls as a fax.	Save Delete
Route all Voice Calls on number 6782384077 3 from SP Provider EM-4552 to extension 102 and <input type="checkbox"/> process all calls as a fax.	Save Delete

5.12 Outgoing Call Rules

Select "System Setup" and select "Outgoing Calls" to see all the default Outgoing Call Rules from the Outgoing Call Rules section and add new ones if needed. Note that all the default rules had been modified to access SIP trunks via EdgeMarc. Hit the "Add New Outgoing Call Rules" button to create a new rule for accessing the SIP trunks by dialing "9" and followed by a 10-digit number.



digium Switchvox SMB 4.5 | [help](#) | [logout](#) | [admin](#)

Extensions | PBX Features | **System Setup** | Diagnostics | Machine Admin

Outgoing Calls

1 Outgoing Call Rules

Create, modify and delete outgoing call rules to apply to the extensions on your phone system.

[What are outgoing call rules?](#)

[Add New Outgoing Rule](#)

Outgoing Rules (1 to 8) of 8

Priority	Move	Name	Pattern to Match	Outgoing Type	Call Being	Modify / Delete
1		International	Begins with 9011 and the remainder is 7 ...	SIP Provider	EM-4552	Modify Delete
2		1-900 Numbers	Begins with 91(900 976) and the remainde...	SIP Provider	EM-4552	Modify Delete
3		Toll Free	Begins with 91(800 888 877 866) and the ...	SIP Provider	EM-4552	Modify Delete
4		911	Number exactly matches 911.	SIP Provider	EM-4552	Modify Delete
5		Local	Begins with 9 and the remainder is 7 dig...	SIP Provider	EM-4552	Modify Delete
6		Internal	Any number 3 digits in length.	Internal		
7		Long Distance91	Begins with 91 and the remainder is 10 d...	SIP Provider	EM-4552	Modify Delete
8		Long distance9	Begins with 9 and the remainder is 10 di...	SIP Provider	EM-4552	Modify Delete

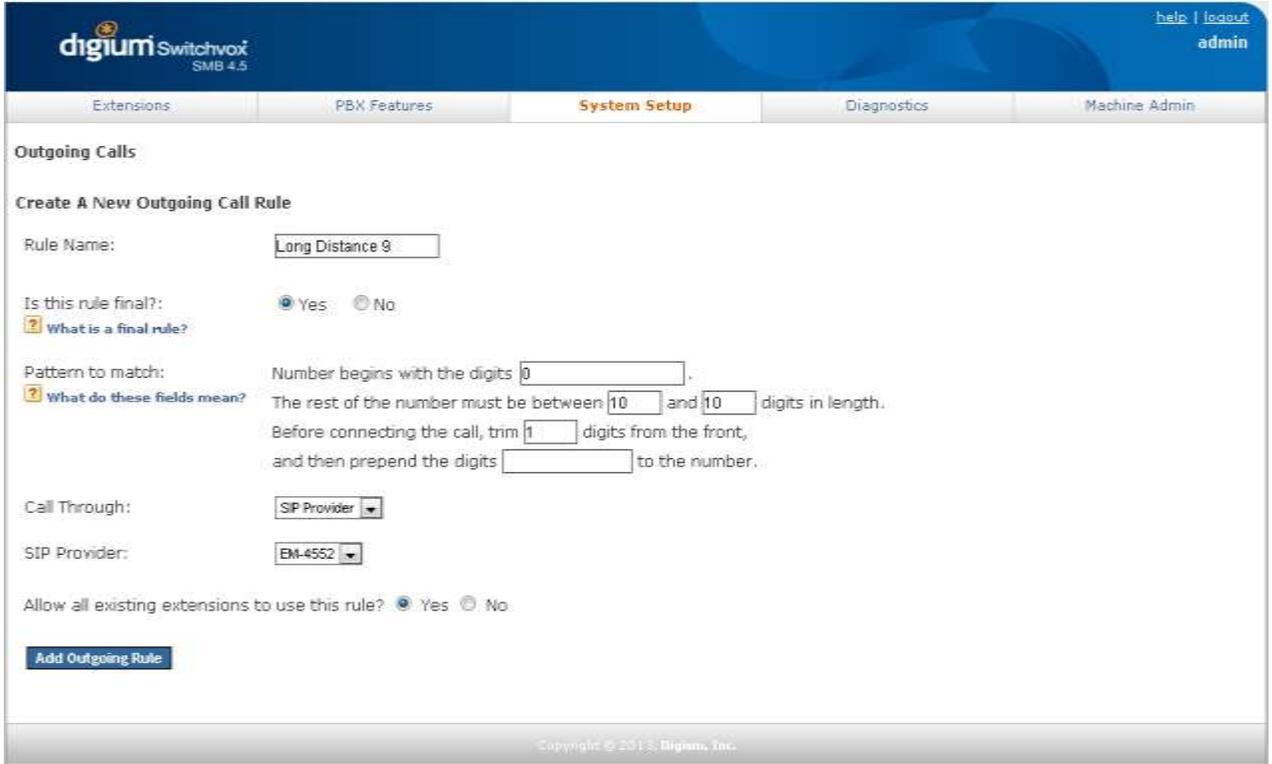
2 Outgoing Caller ID Rules

Change the caller ID for extensions making outgoing calls on your phone system.

[Tell me more about how this works](#)

- Enter a descriptive name in "Rule Name" field.
- Select "Yes" for the "Is this rule final?" question.
- For the "Pattern to match" fields, enter "9" as the beginning digit, enter "10" as the exact length of the phone number, enter "1" as the number of digits to trim.
- Select "SIP Provider" in the "Call Through" field.
- Select the provider name of EdgeMarc in the "SIP Provider" field.
- Select "Yes" for the "Allow all existing extensions to use this rule?" question.

g) Hit the "Add Outgoing Rule" button.

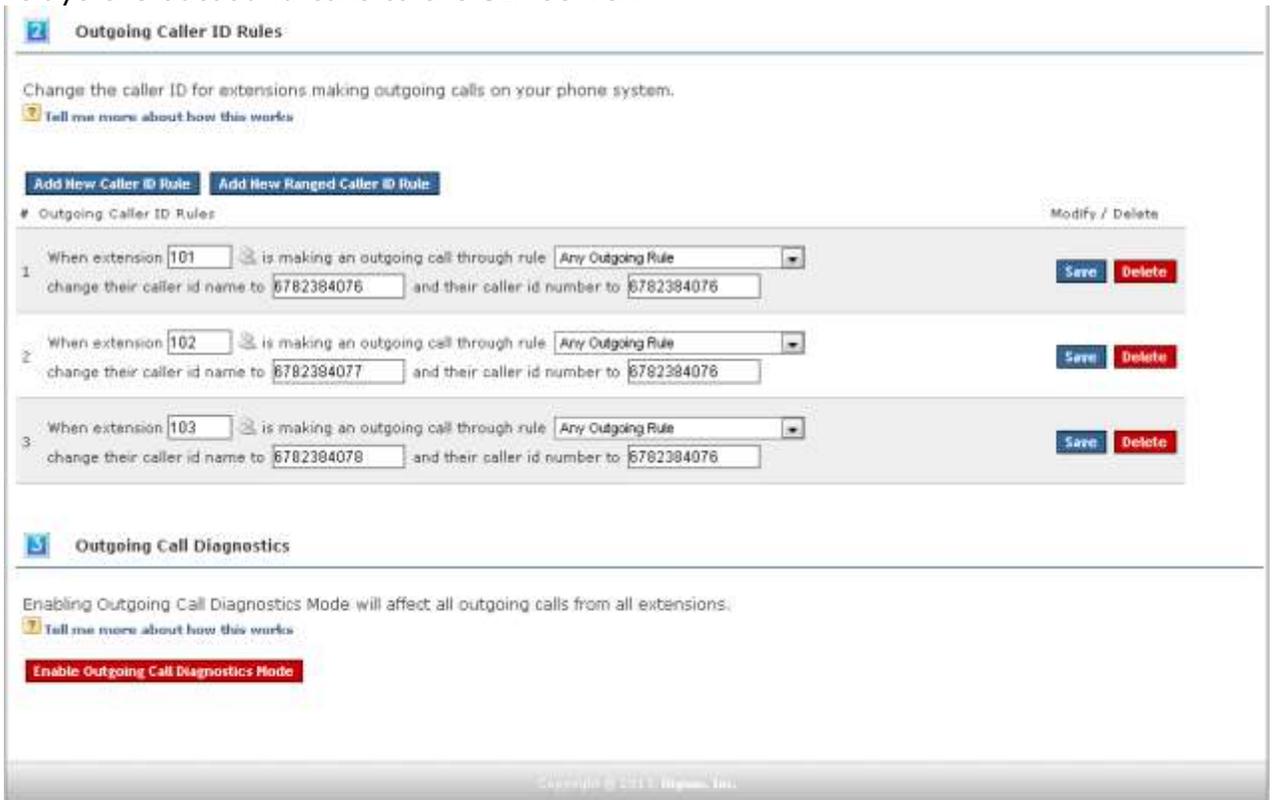


The screenshot shows the Digium Switchvox SMB 4.5 System Setup interface. The page is titled "Outgoing Calls" and "Create A New Outgoing Call Rule". The "Rule Name" field contains "Long Distance 9". The "Is this rule final?" section has "Yes" selected. The "Pattern to match" section includes a "Number begins with the digits" field with "0", a "The rest of the number must be between" field with "10" and "10", and a "Before connecting the call, trim" field with "1". The "Call Through" dropdown is set to "SIP Provider" and the "SIP Provider" dropdown is set to "EM-4552". The "Allow all existing extensions to use this rule?" section has "Yes" selected. An "Add Outgoing Rule" button is at the bottom left. The footer contains "Copyright © 2013, Digium, Inc."

Note: For the Polycom Phones to take advantage of these outgoing call rules, the phones should have the same outgoing call rules defined in its digitmap in the SIP section of the web page. For example, the Polycom phones can make "91+ 10-digit number" call but not the "9+ 10-digit number calls. This is because the phones have a "91xxxxxxxxxx" dialing rule but do not have one for "9xxxxxxxxxx".

5.13 Outgoing Caller ID

Select "System Setup" and select "Outgoing Calls" to set Caller ID for outbound calls in the Outgoing Caller ID Rules section. Note that the Caller ID for any outbound calls must be the same as the Account ID used for PBX registration with EdgeMarc. For each DID assigned to an extension, set the Caller ID number to the same as the Account ID. In this example, the Account ID is the same as the pilot DID. For PBX registration mode, only the pilot DID can be used as the caller ID when EdgeMarc relays the outbound calls to the SIP server.



2 Outgoing Caller ID Rules

Change the caller ID for extensions making outgoing calls on your phone system.
[Tell me more about how this works](#)

[Add New Caller ID Rule](#) [Add New Ranged Caller ID Rule](#)

Outgoing Caller ID Rules Modify / Delete

1	When extension <input type="text" value="101"/> is making an outgoing call through rule <input type="text" value="Any Outgoing Rule"/>	<input type="text" value="6782384076"/>	<input type="text" value="6782384076"/>	<input type="button" value="Save"/>	<input type="button" value="Delete"/>
2	When extension <input type="text" value="102"/> is making an outgoing call through rule <input type="text" value="Any Outgoing Rule"/>	<input type="text" value="6782384077"/>	<input type="text" value="6782384076"/>	<input type="button" value="Save"/>	<input type="button" value="Delete"/>
3	When extension <input type="text" value="103"/> is making an outgoing call through rule <input type="text" value="Any Outgoing Rule"/>	<input type="text" value="6782384078"/>	<input type="text" value="6782384076"/>	<input type="button" value="Save"/>	<input type="button" value="Delete"/>

3 Outgoing Call Diagnostics

Enabling Outgoing Call Diagnostics Mode will affect all outgoing calls from all extensions.
[Tell me more about how this works](#)

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5.14 Static IP Outgoing Caller ID

If you have configured the PBX for static IP mode, you may opt to have either the pilot DID or non-pilot DID as the caller ID of the outbound call:

- a) Select "System Setup", select "VoIP Providers" and hit the "Modify" button to modify the VoIP Provider account set up for EdgeMarc.
- b) Click the "Click to Show Advanced Options" link.
- c) In the Caller ID Settings section, select "Yes" for the "Supports Changing Caller ID" setting. Note that the subsequent changes will work only when "Host

Type” is set to “Peer” (configuration for static IP mode) in the Peer Settings section.

1 Peer Settings

Host Type Provider ▾
What is Host Type?

Host is a Switchvox PBX Yes No
What does this mean?

Treat system's users like local users Yes No
What does this mean?

Jabber Hostname
What does this mean?

Apply Incoming Call Rules to Provider Yes No
What is this for?

Outgoing Call Rules What is this for?

Rule Name	Allow	Deny
International	<input type="checkbox"/>	<input type="checkbox"/>
1-900 Numbers	<input type="checkbox"/>	<input type="checkbox"/>
Toll Free	<input type="checkbox"/>	<input type="checkbox"/>
911	<input type="checkbox"/>	<input type="checkbox"/>
Local	<input type="checkbox"/>	<input type="checkbox"/>
Internal	<input type="checkbox"/>	<input type="checkbox"/>
Long Distance91	<input type="checkbox"/>	<input type="checkbox"/>
Long distance9	<input type="checkbox"/>	<input type="checkbox"/>

2 Caller ID Settings

Supports Changing Caller ID Yes No

- d) Select “System Setup” and select “Outgoing Calls” to set Caller ID for outbound calls in the Outgoing Caller ID Rules section. For each DID assigned to an extension, set the Caller ID number to the same as the assigned DID.

2 Outgoing Caller ID Rules

Change the caller ID for extensions making outgoing calls on your phone system.
[Tell me more about how this works](#)

[Add New Caller ID Rule](#) [Add New Ranged Caller ID Rule](#)

Outgoing Caller ID Rules Modify / Delete

1	When extension <input type="text" value="101"/> is making an outgoing call through rule <input type="text" value="Any Outgoing Rule"/>	<input type="text" value="6782384076"/> and their caller id number to <input type="text" value="6782384076"/>	Save Delete
2	When extension <input type="text" value="102"/> is making an outgoing call through rule <input type="text" value="Any Outgoing Rule"/>	<input type="text" value="6782384077"/> and their caller id number to <input type="text" value="6782384077"/>	Save Delete
3	When extension <input type="text" value="103"/> is making an outgoing call through rule <input type="text" value="Any Outgoing Rule"/>	<input type="text" value="6782384078"/> and their caller id number to <input type="text" value="6782384078"/>	Save Delete

3 Outgoing Call Diagnostics

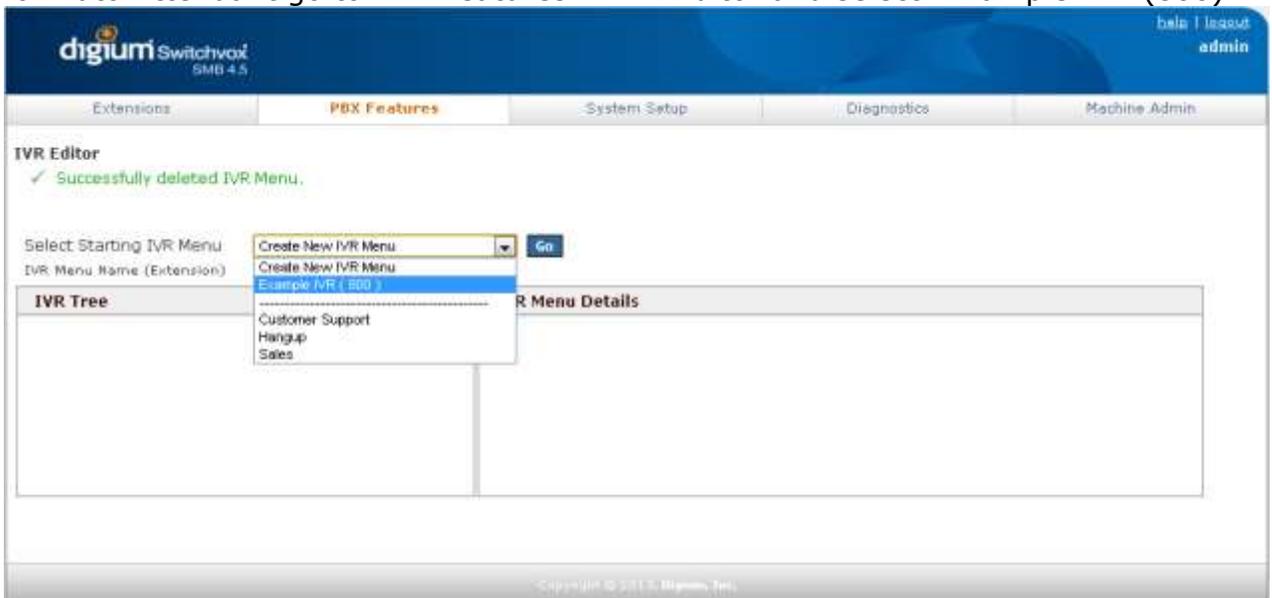
Enabling Outgoing Call Diagnostics Mode will affect all outgoing calls from all extensions.
[Tell me more about how this works](#)

[Enable Outgoing Call Diagnostics Mode](#)

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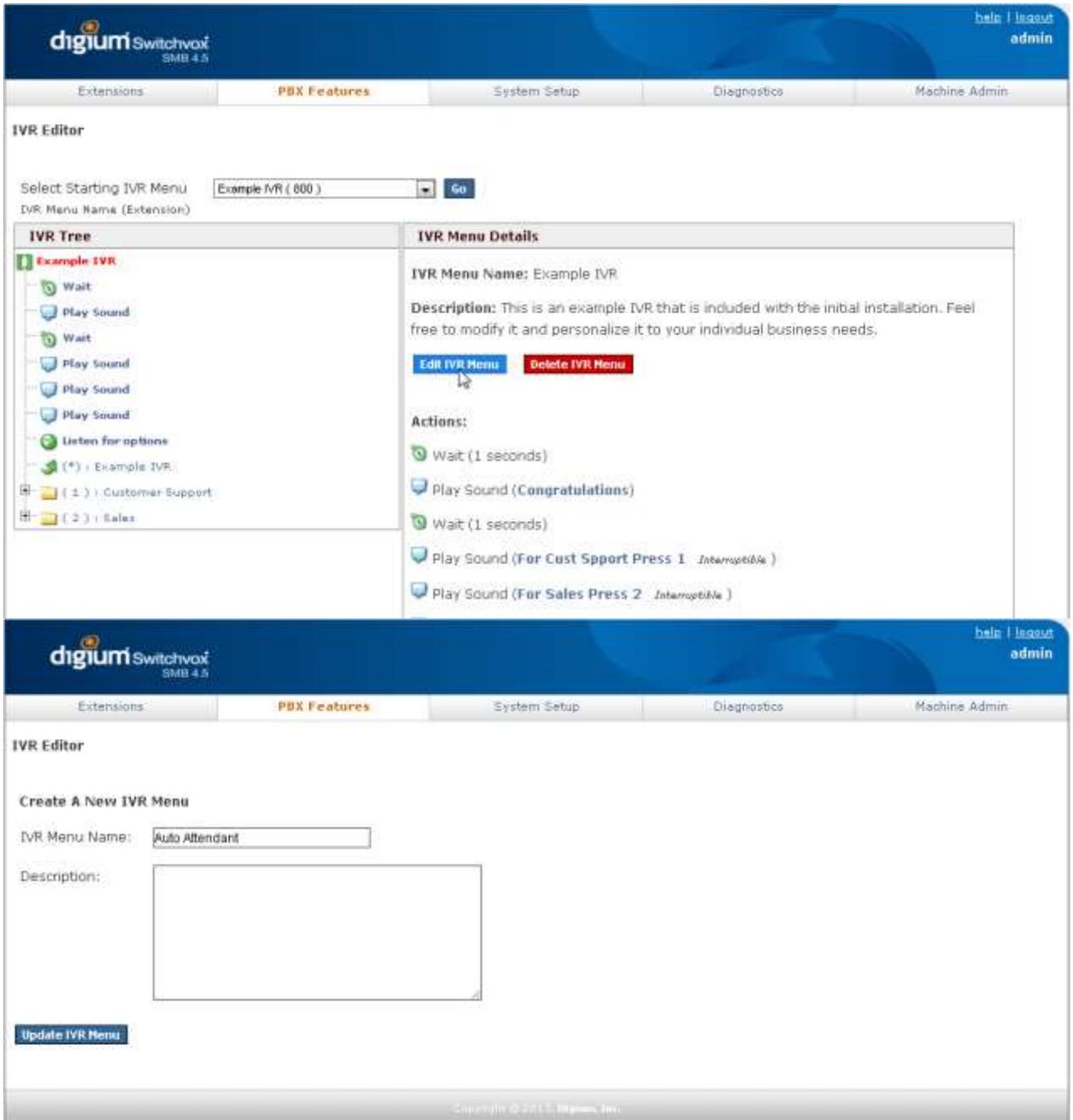
5.15 Auto Attendant

Auto Attendant: Auto Attendant in this PBX is labeled under 'IVR', to setup the PBX for Auto Attendant go to PBX Features > IVR Editor and select "Example IVR (800)".



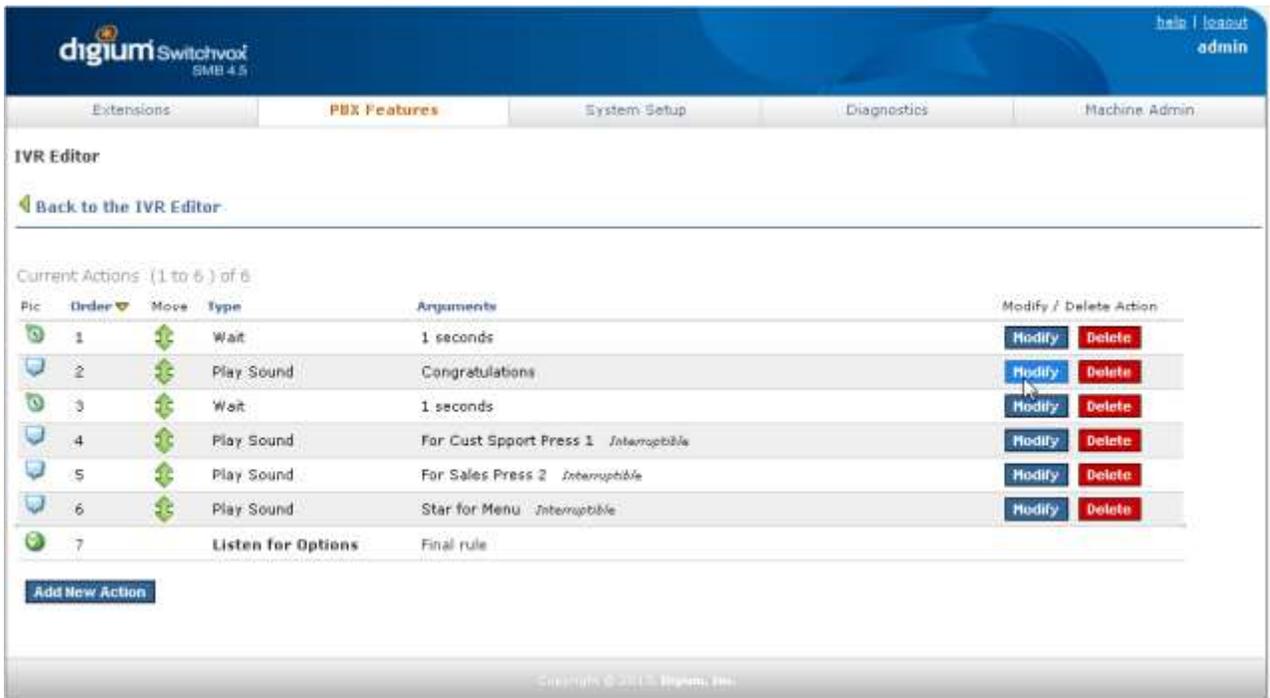
The screenshot shows the Digium Switchvox IVR Editor interface. At the top, there are navigation tabs: Extensions, PBX Features (selected), System Setup, Diagnostics, and Machine Admin. Below the tabs, a message states "Successfully deleted IVR Menu." The main area is titled "IVR Editor" and contains a "Select Starting IVR Menu" section with a dropdown menu and a "Go" button. The dropdown menu is open, showing options: "Create New IVR Menu", "Create New IVR Menu", and "Example IVR (800)". Below this is an "IVR Tree" section with a large empty box. To the right, there is an "IVR Menu Details" section, also empty. The footer of the interface includes the text "Copyright © 2013, Digium, Inc."

- a) Click on "Edit IVR Menu" and change the IVR Menu name to "Auto Attendant" and clear out the Description, then click "Update IVR Menu"

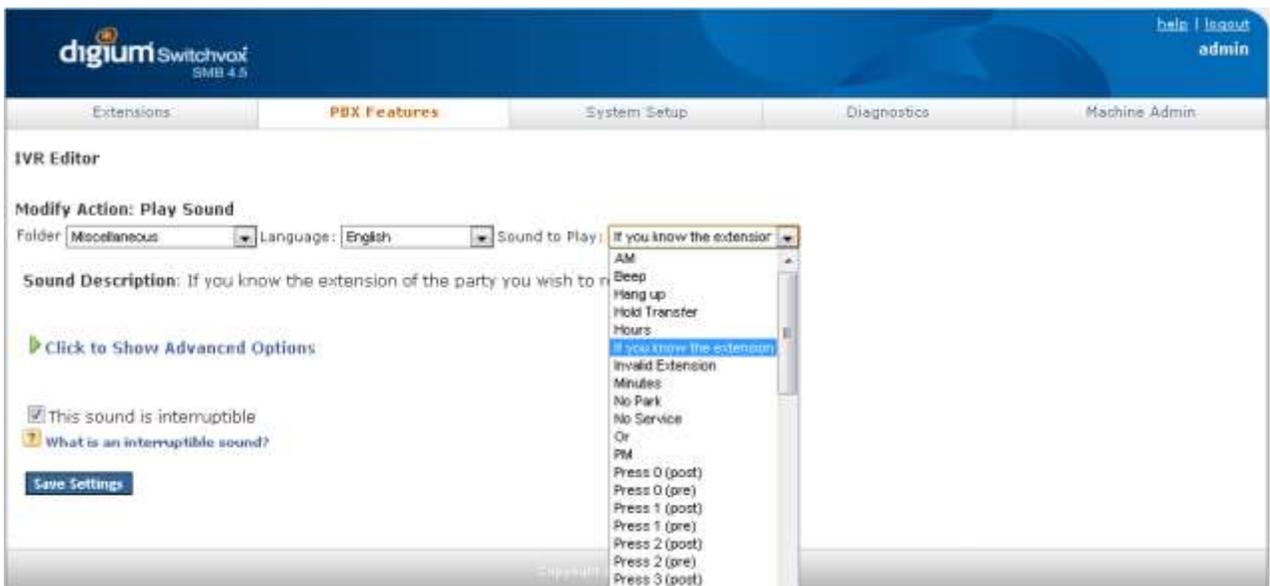


- b) Now click on "Modify Actions" to change the Actions layout for extension dialing.

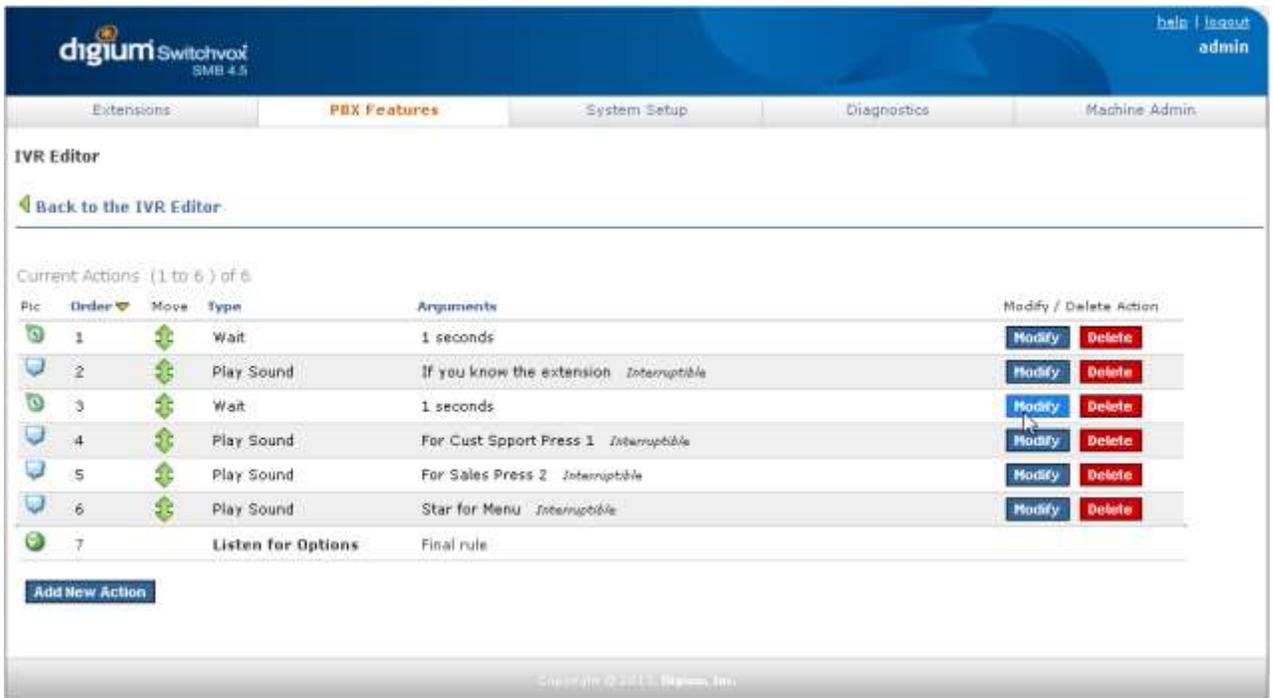
c) Click "Modify" on the second action in the order



d) Change the Folder to "Miscellaneous" and set the Language to "English" and set the Sound to Play to "If you know the extension". Make sure "This sound is interruptible" is checked then click "Save Settings"



e) Click "Modify" on the second Action in the order to change the wait time to 10 seconds.

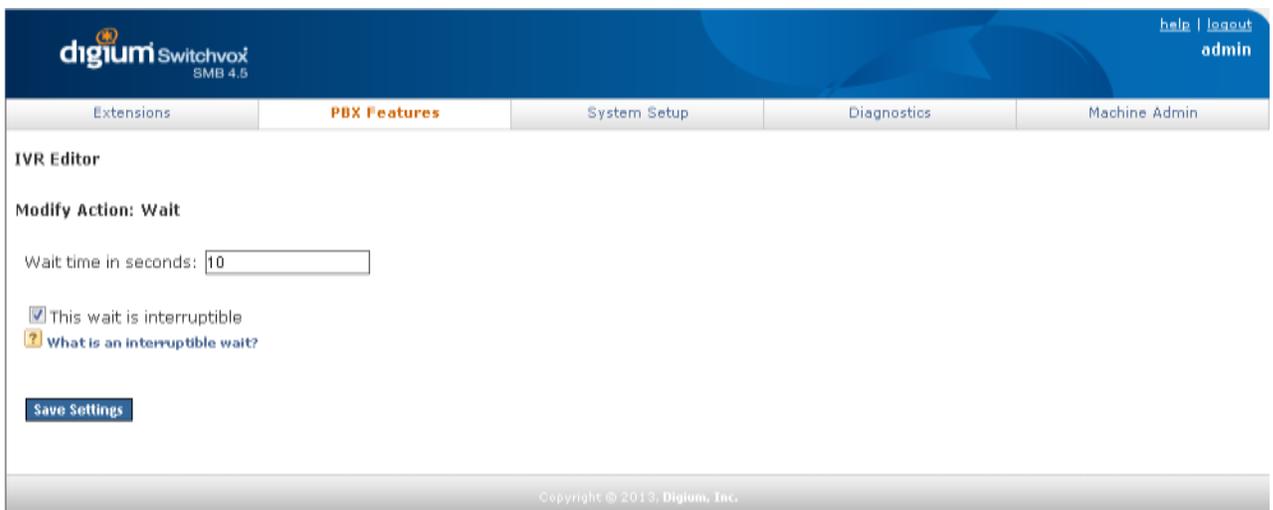


The screenshot shows the 'IVR Editor' page in the Digium Switchvox SMB 4.5 administration interface. The 'PBX Features' tab is selected. A table lists the current actions in the IVR menu:

Pic	Order	Move	Type	Arguments	Modify / Delete Action
	1		Wait	1 seconds	Modify Delete
	2		Play Sound	If you know the extension <i>Interruptible</i>	Modify Delete
	3		Wait	1 seconds	Modify Delete
	4		Play Sound	For Cust Spport Press 1 <i>Interruptible</i>	Modify Delete
	5		Play Sound	For Sales Press 2 <i>Interruptible</i>	Modify Delete
	6		Play Sound	Star for Menu <i>Interruptible</i>	Modify Delete
	7		Listen for Options	Final rule	

Below the table is an 'Add New Action' button. The footer of the page reads 'Copyright © 2013, Digium, Inc.'

f) Check the box to enable "This wait is interruptible" then click "Save Settings"

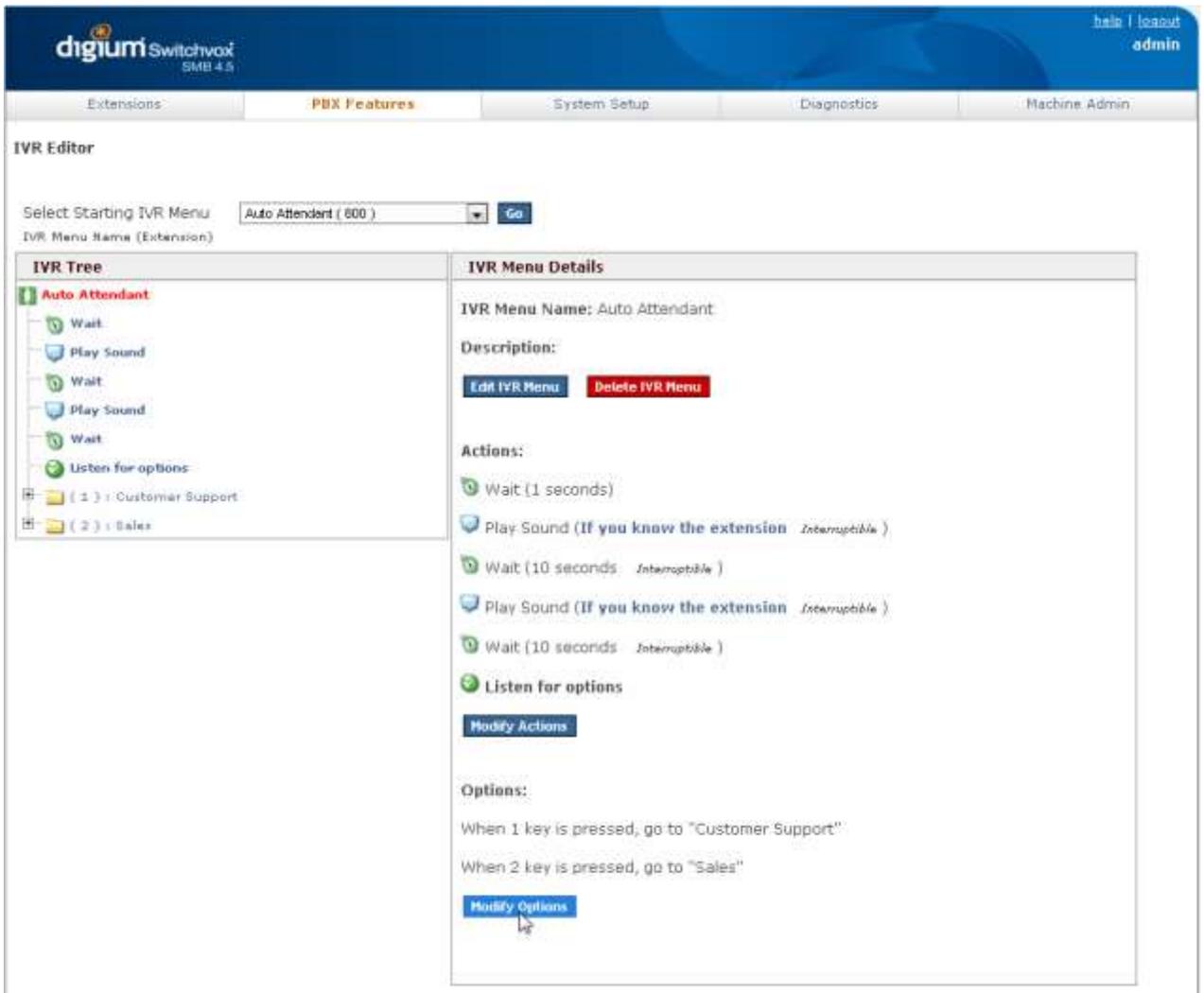


The screenshot shows the 'Modify Action: Wait' screen in the Digium Switchvox SMB 4.5 administration interface. The 'PBX Features' tab is selected. The screen displays the following settings:

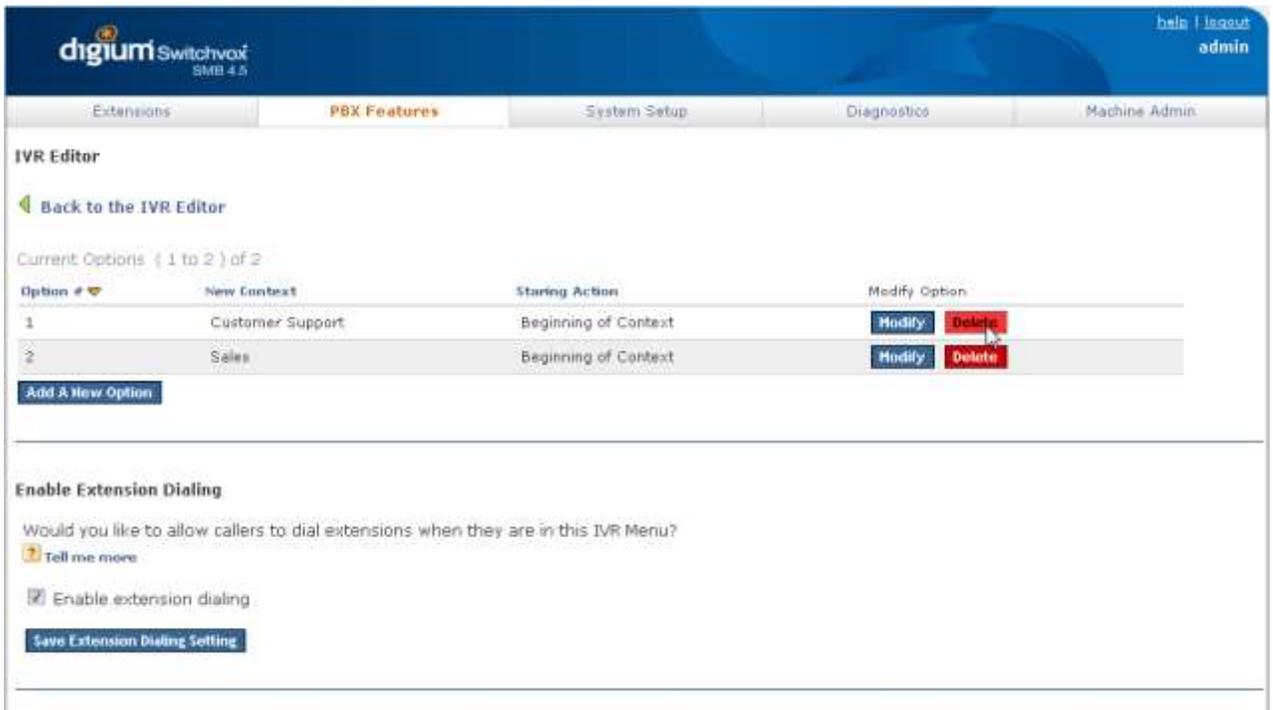
- Wait time in seconds:
- This wait is interruptible
- [? What is an interruptible wait?](#)

At the bottom of the screen is a 'Save Settings' button. The footer of the page reads 'Copyright © 2013, Digium, Inc.'

- g) Actions 4-6 can be modified to repeat the "Extension" message or they can be deleted
- h) Click "Back to the IVR Editor" to go back to the main page, then click "Modify Options"



- i) Delete the 2 current Options at the top, check the box for "Enable Extension Dialing" and click "Save Extension Dialing Settings"



digium Switchvox SMB 4.5

help | logout admin

Extensions **PBX Features** System Setup Diagnostics Machine Admin

IVR Editor

[Back to the IVR Editor](#)

Current Options (1 to 2) of 2

Option #	New Context	Starting Action	Modify Option
1	Customer Support	Beginning of Context	Modify Delete
2	Sales	Beginning of Context	Modify Delete

[Add A New Option](#)

Enable Extension Dialing

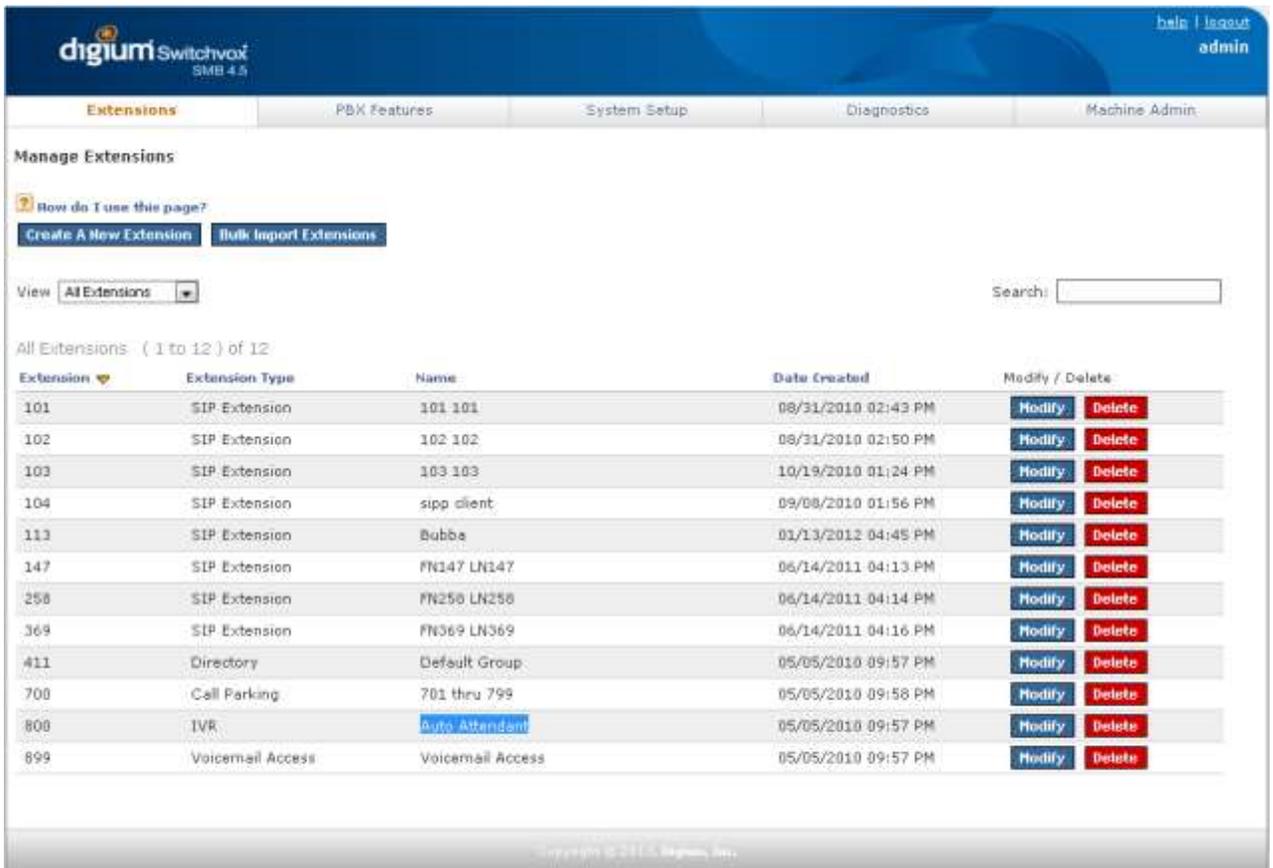
Would you like to allow callers to dial extensions when they are in this IVR Menu?

[Tell me more](#)

Enable extension dialing

[Save Extension Dialing Setting](#)

- j) Go to Extensions > Manage Extensions to make sure that extension 800 is labeled "Auto Attendant"



Manage Extensions

How do I use this page?
[Create A New Extension](#) [Bulk Import Extensions](#)

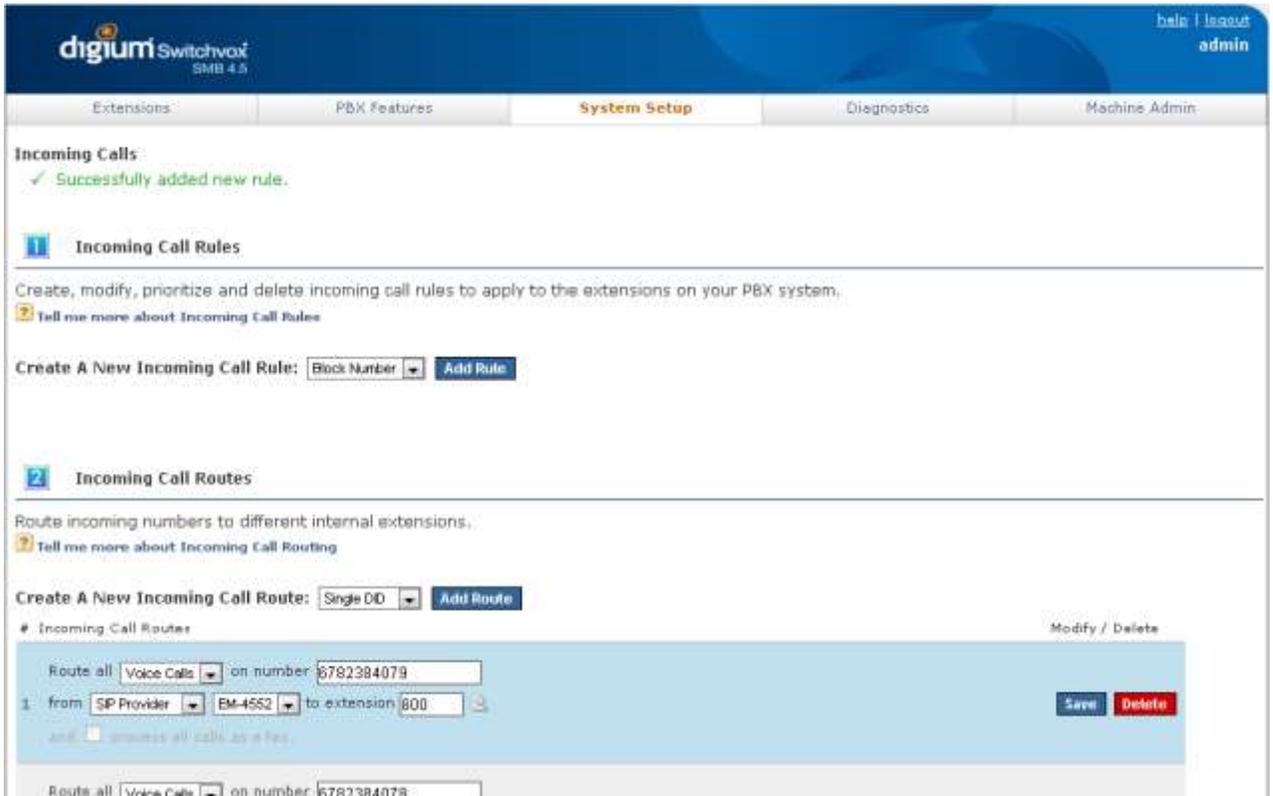
View: All Extensions Search:

All Extensions: (1 to 12) of 12

Extension	Extension Type	Name	Date Created	Modify / Delete
101	SIP Extension	101 101	08/31/2010 02:43 PM	Modify Delete
102	SIP Extension	102 102	08/31/2010 02:50 PM	Modify Delete
103	SIP Extension	103 103	10/19/2010 01:24 PM	Modify Delete
104	SIP Extension	sipp client	09/08/2010 01:56 PM	Modify Delete
113	SIP Extension	Bubba	01/13/2012 04:45 PM	Modify Delete
147	SIP Extension	FN147 LN147	06/14/2011 04:13 PM	Modify Delete
258	SIP Extension	FN258 LN258	06/14/2011 04:14 PM	Modify Delete
369	SIP Extension	FN369 LN369	06/14/2011 04:16 PM	Modify Delete
411	Directory	Default Group	05/05/2010 09:57 PM	Modify Delete
700	Call Parking	701 thru 799	05/05/2010 09:58 PM	Modify Delete
800	IVR	Auto Attendant	05/05/2010 09:57 PM	Modify Delete
899	Voicemail Access	Voicemail Access	05/05/2010 09:57 PM	Modify Delete

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k) Go to System Setup > Incoming Calls and click "Add Route" to assign a DID to extension 800. In this example we used 6782384079.



The screenshot shows the Digium Switchvox SMB 4.5 System Setup interface. The top navigation bar includes 'Extensions', 'PBX Features', 'System Setup' (highlighted), 'Diagnostics', and 'Machine Admin'. The 'Incoming Calls' section shows a success message: 'Successfully added new rule.' Below this, there are two main configuration sections:

- Incoming Call Rules:** A section for creating, modifying, prioritizing, and deleting rules. It includes a 'Create A New Incoming Call Rule' button with a 'Block Number' dropdown and an 'Add Rule' button.
- Incoming Call Routes:** A section for routing incoming numbers to different internal extensions. It includes a 'Create A New Incoming Call Route' button with a 'Single DID' dropdown and an 'Add Route' button.

The 'Incoming Call Routes' section displays a table of existing routes:

Incoming Call Routes		Modify / Delete
Route all	Voice Calls on number 6782384078	
1. from	SP Provider EM-4552 to extension 800	Save Delete
and process all calls as a fee.		
Route all	Voice Calls on number 6782384078	

For advanced configurations and support please contact the Edgewater Technical Assistance Center support@edgewaternetworks.com or call [408.351.7255](tel:408.351.7255).