August 25th, 2013

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

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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

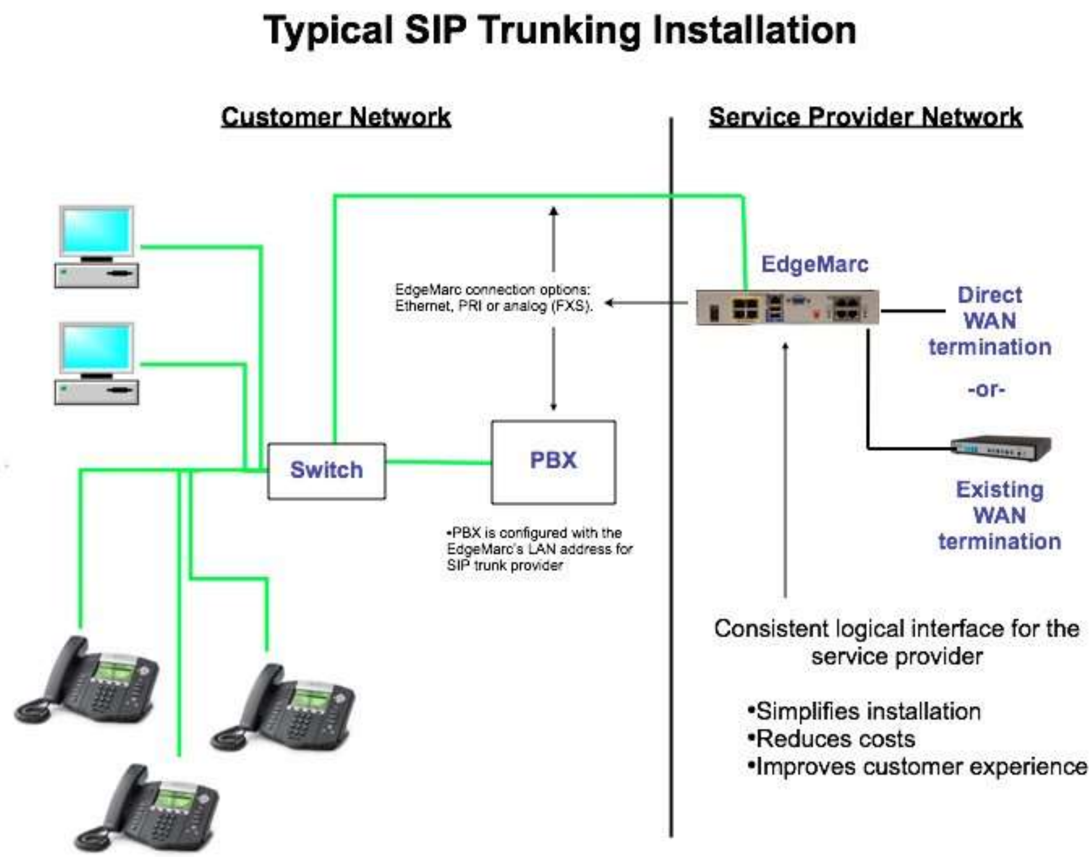


Figure 1 Test Set up

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

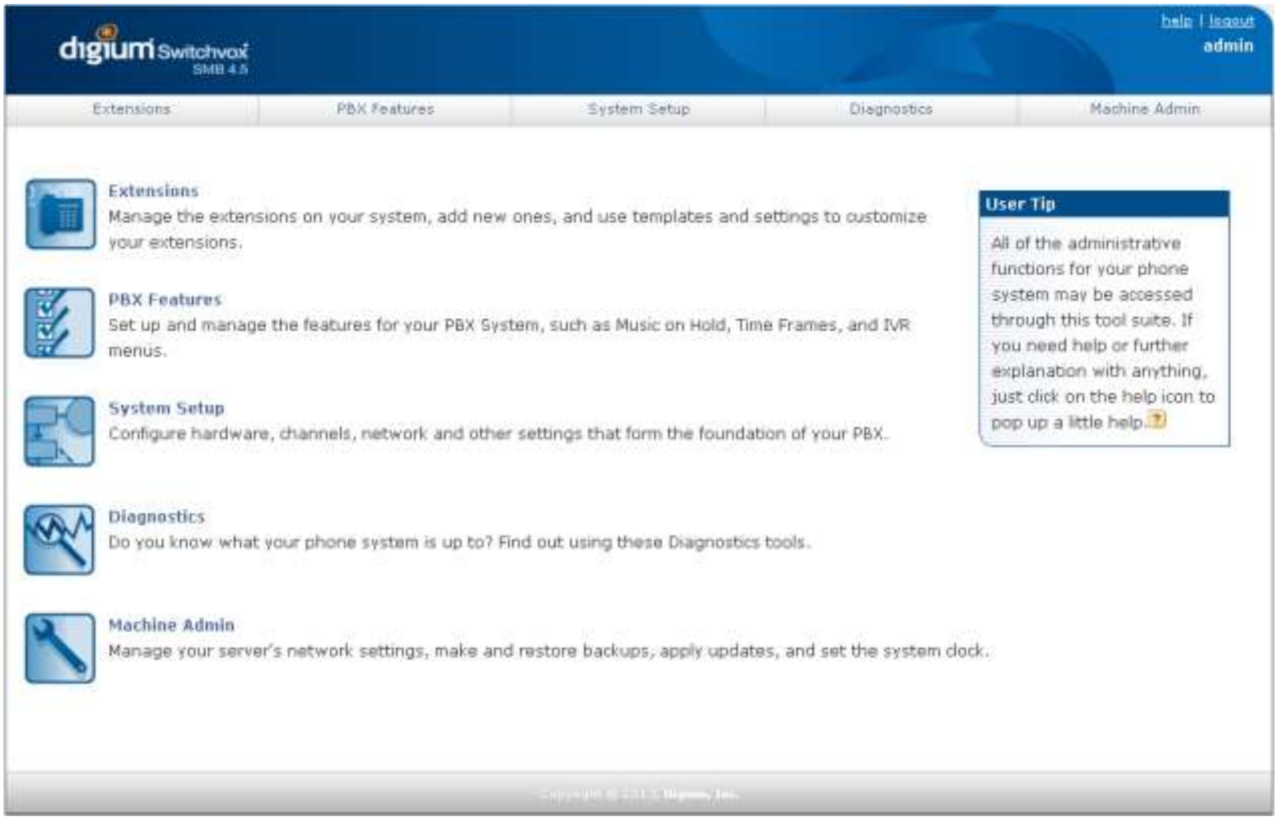
The IP-PBX was shipped with a default IP address of 192.168.1.100/24 for 1 Ethernet port. To work with EM for SIP trunking service, this port should be in the same subnet as EM port 1 and use EM port 1's IP address as its SIP server. All the IP phones communicate with the PBX via this IP address as well, 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.

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.

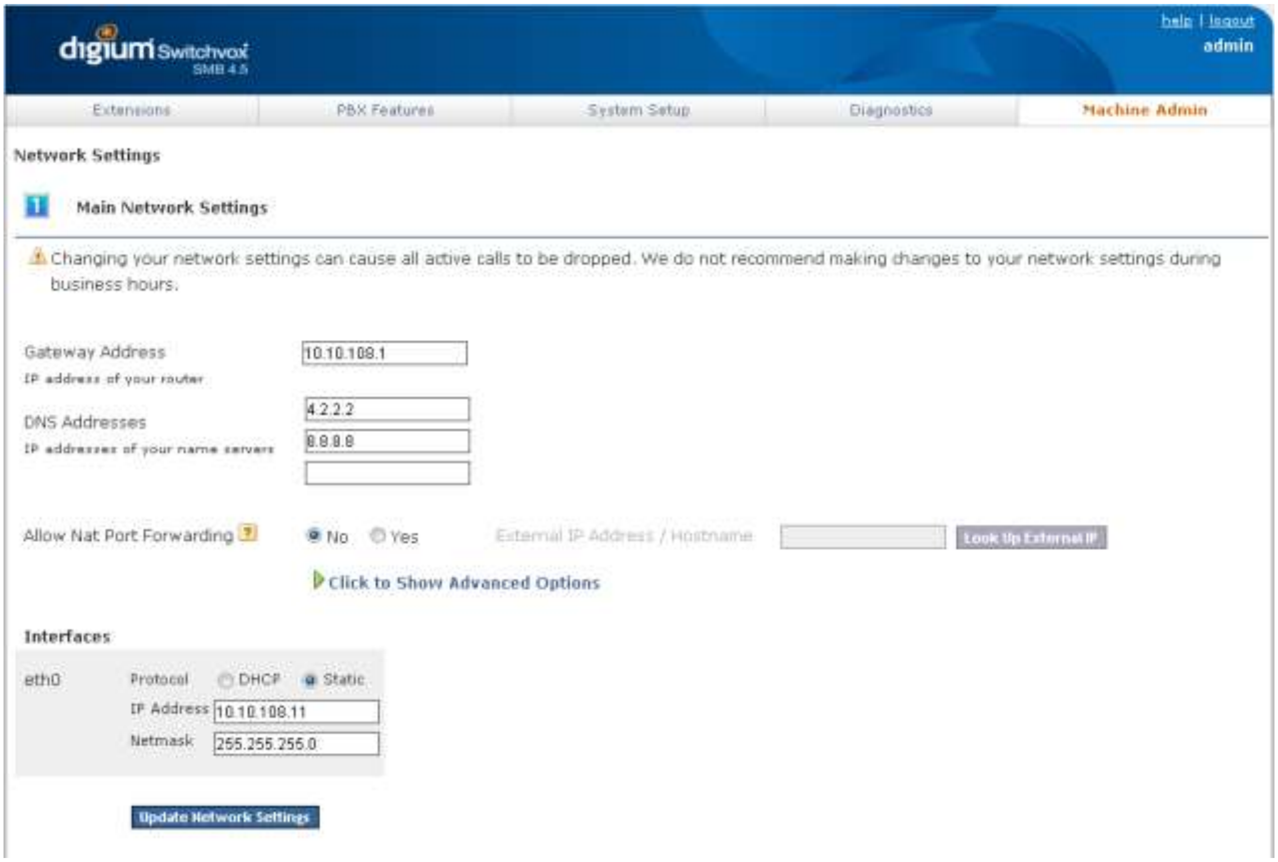
a) Enter EdgeMarc’s IP address in the “Gateway Address” field.

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.



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.

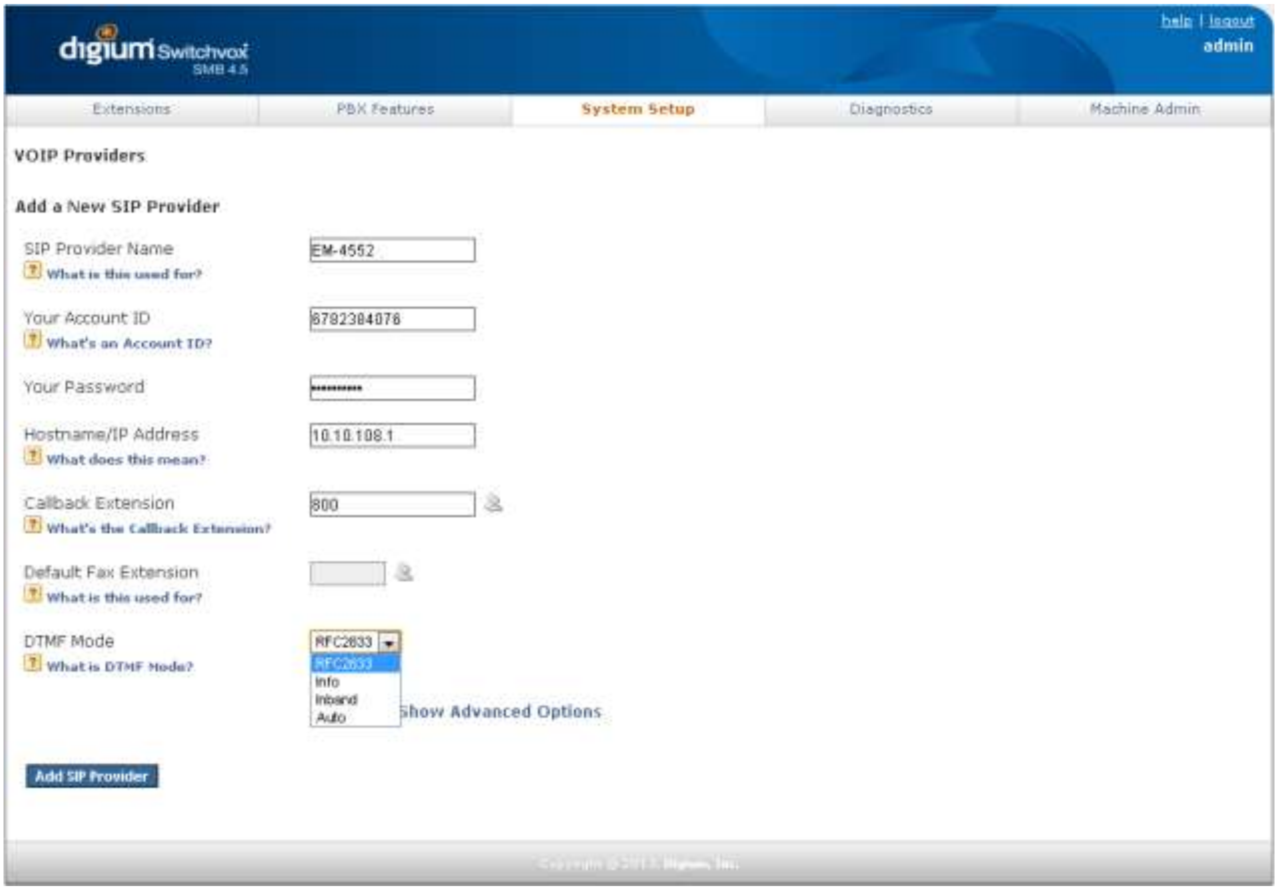
c) Enter password in the "Your Password" field.

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.

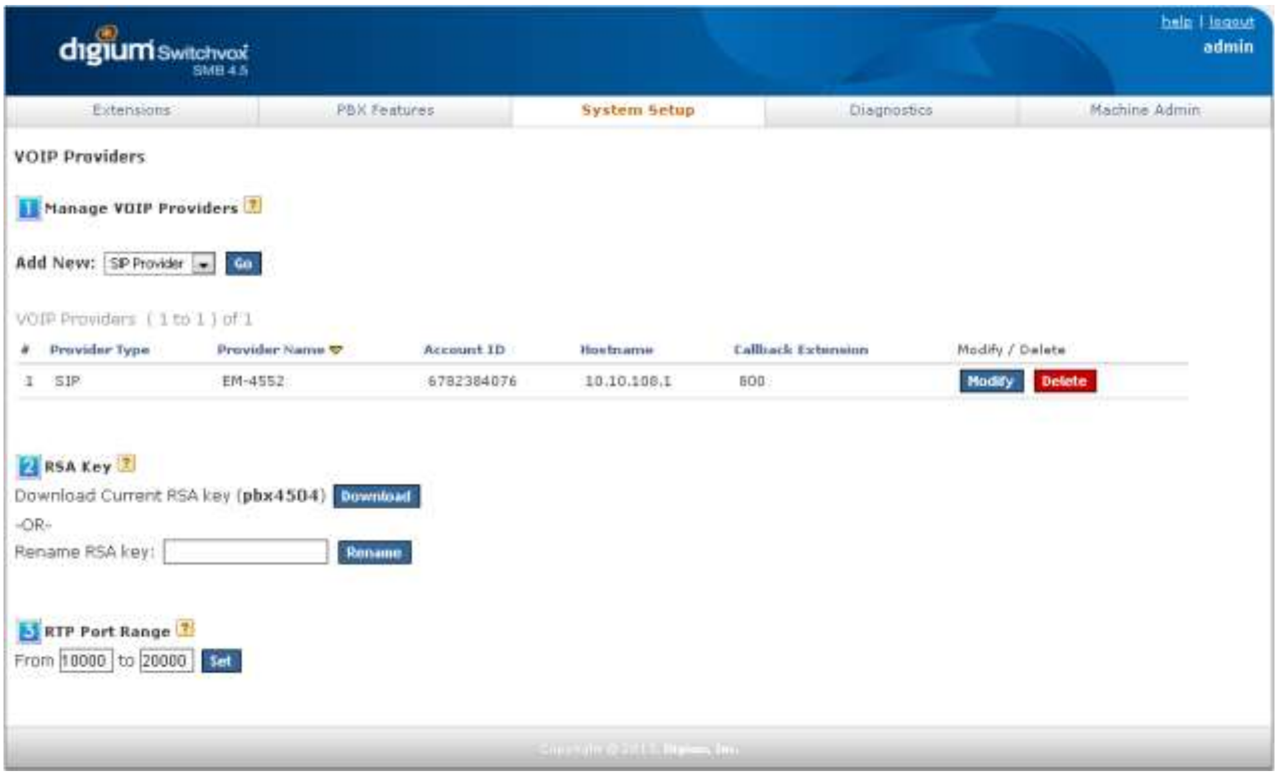
g) Leave other fields as default and hit the "Add SIP Provider" button.


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?
EM-4552
Your Account ID
What's an Account ID?
6782384076
Your Password
Hostname/IP Address
What does this mean?
10.10.108.1
Callback Extension
What's the Callback Extension?
800
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

5.6 Static IP Mode

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

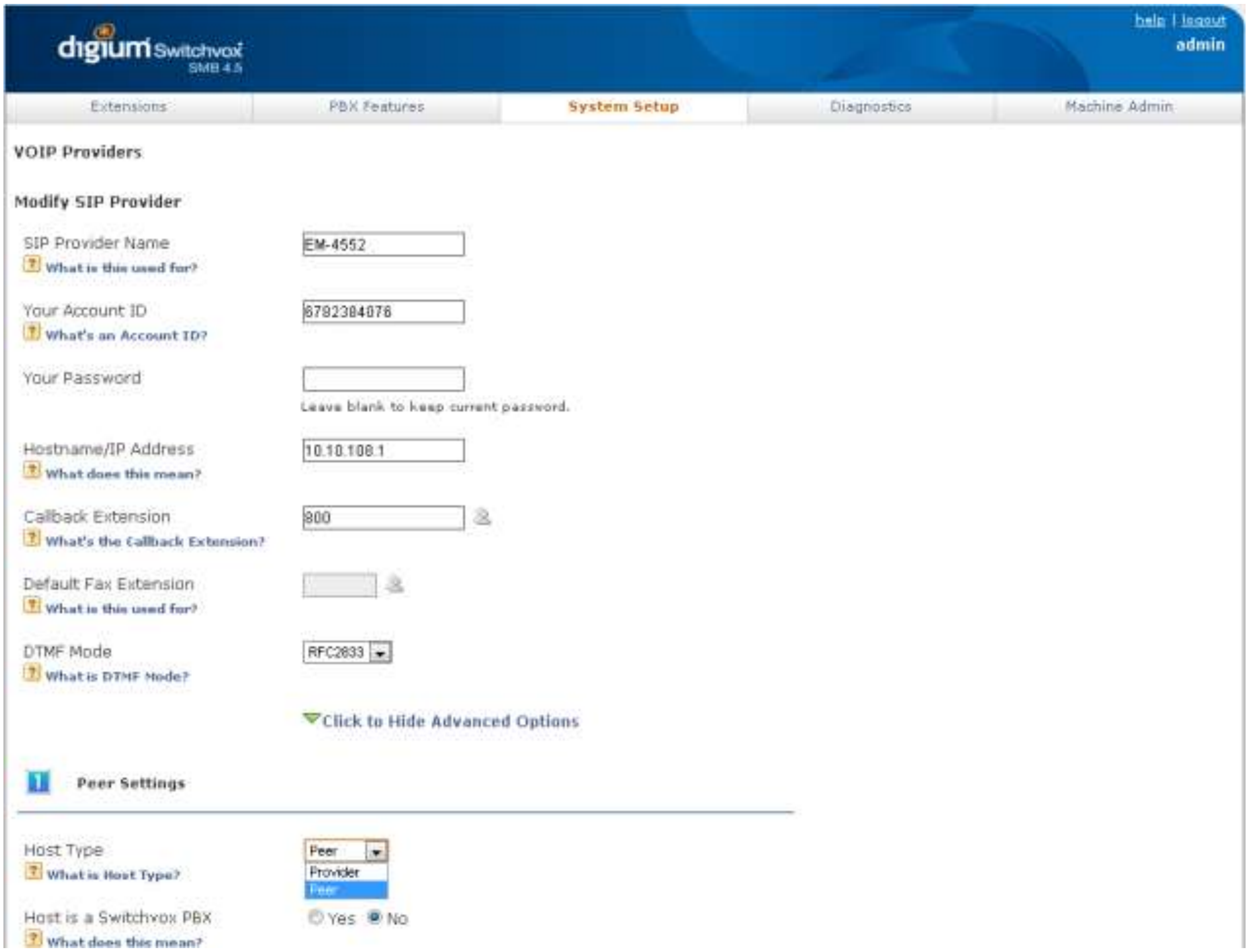
a) Hit the “Modify” button to modify the VoIP Provider account set up for EdgeMarc.



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

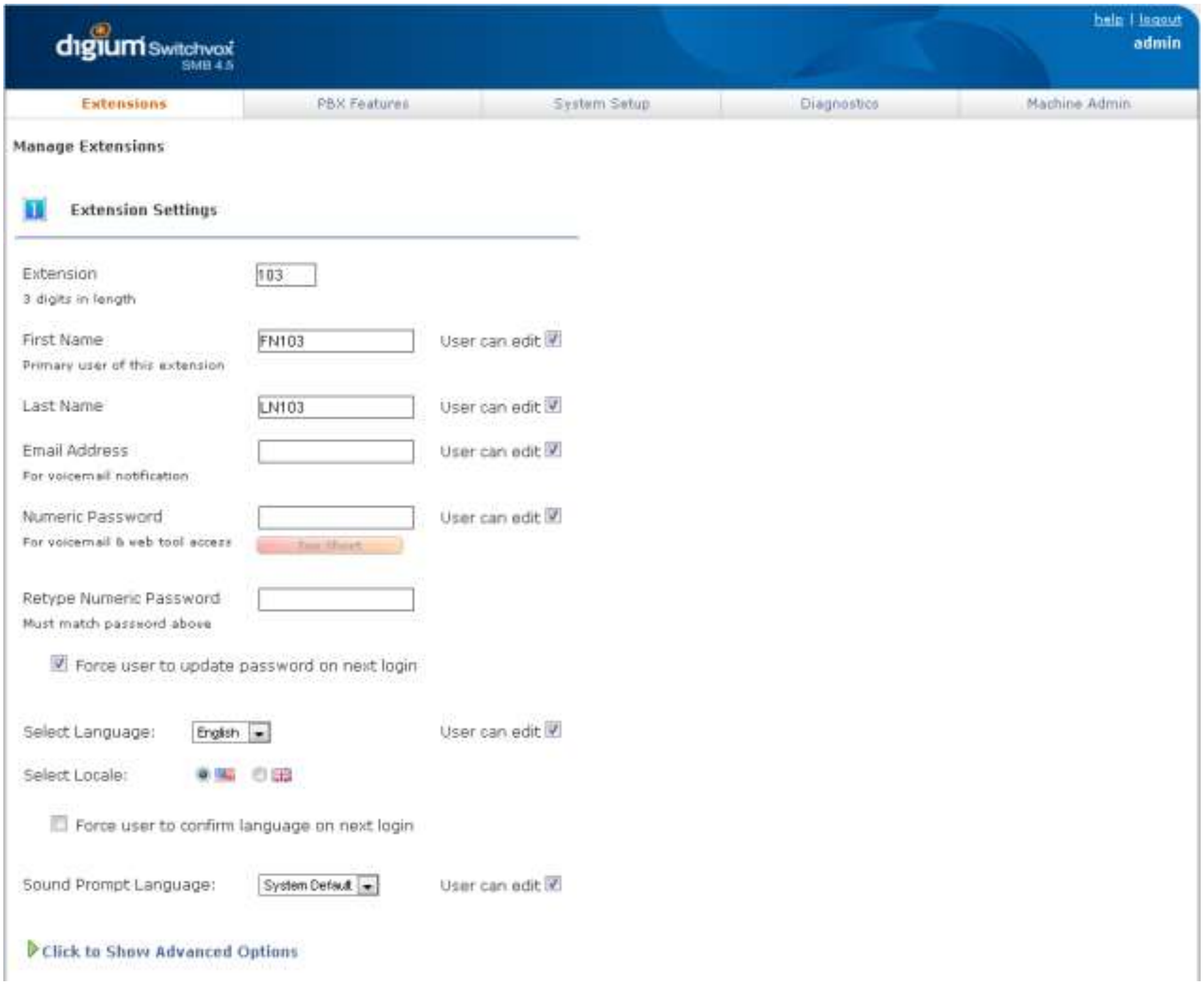
Select "Extensions", select "Manage Extensions" and click on the "Create A New Extension" button to create SIP extensions for the SIP phones.

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

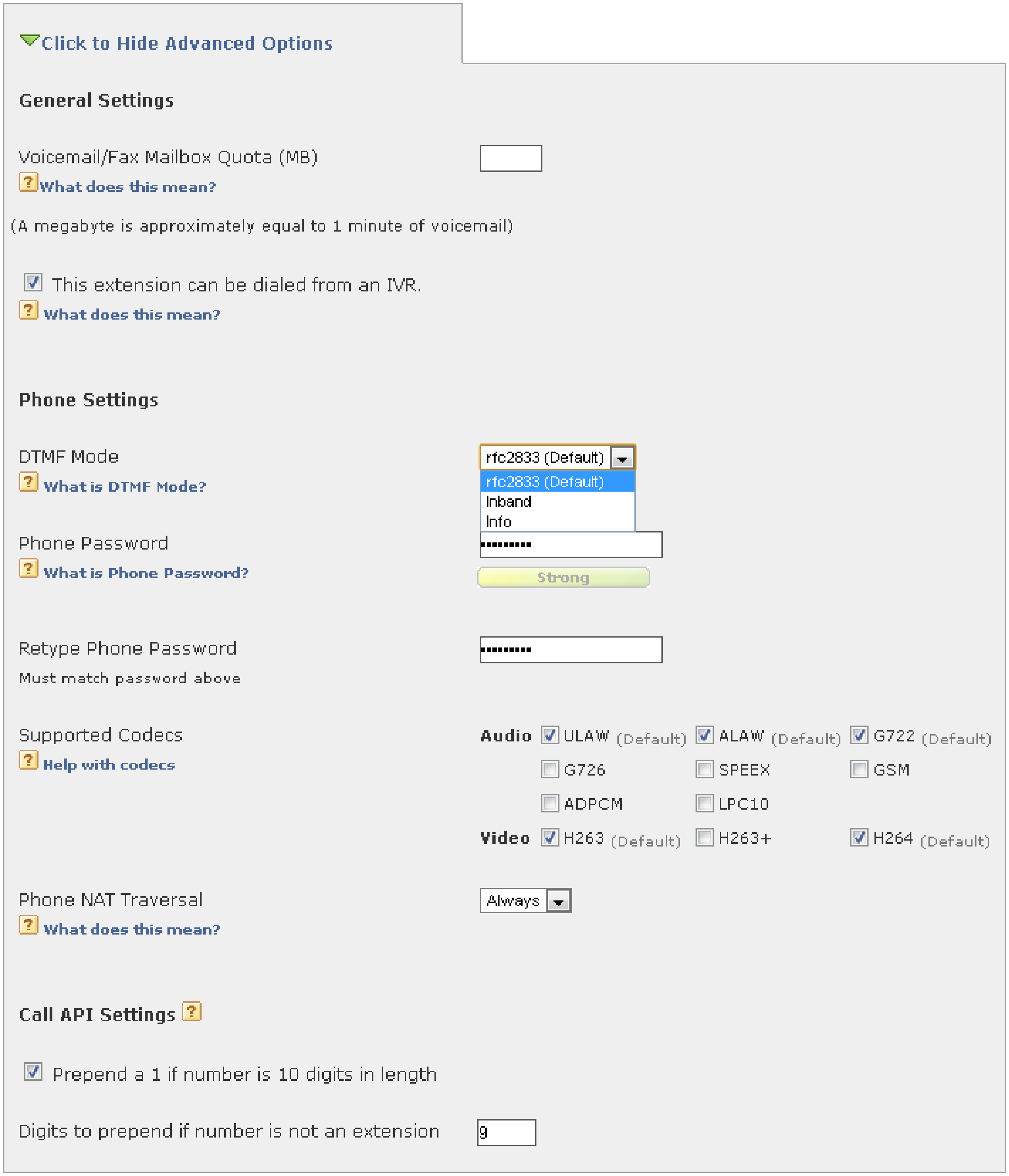
c) Enter the first name of the user in the "First Name" field and enter the last name of the user in the "Last Name" field.

d) Leave other fields as default and hit the "Click to Show Advanced Options" link.

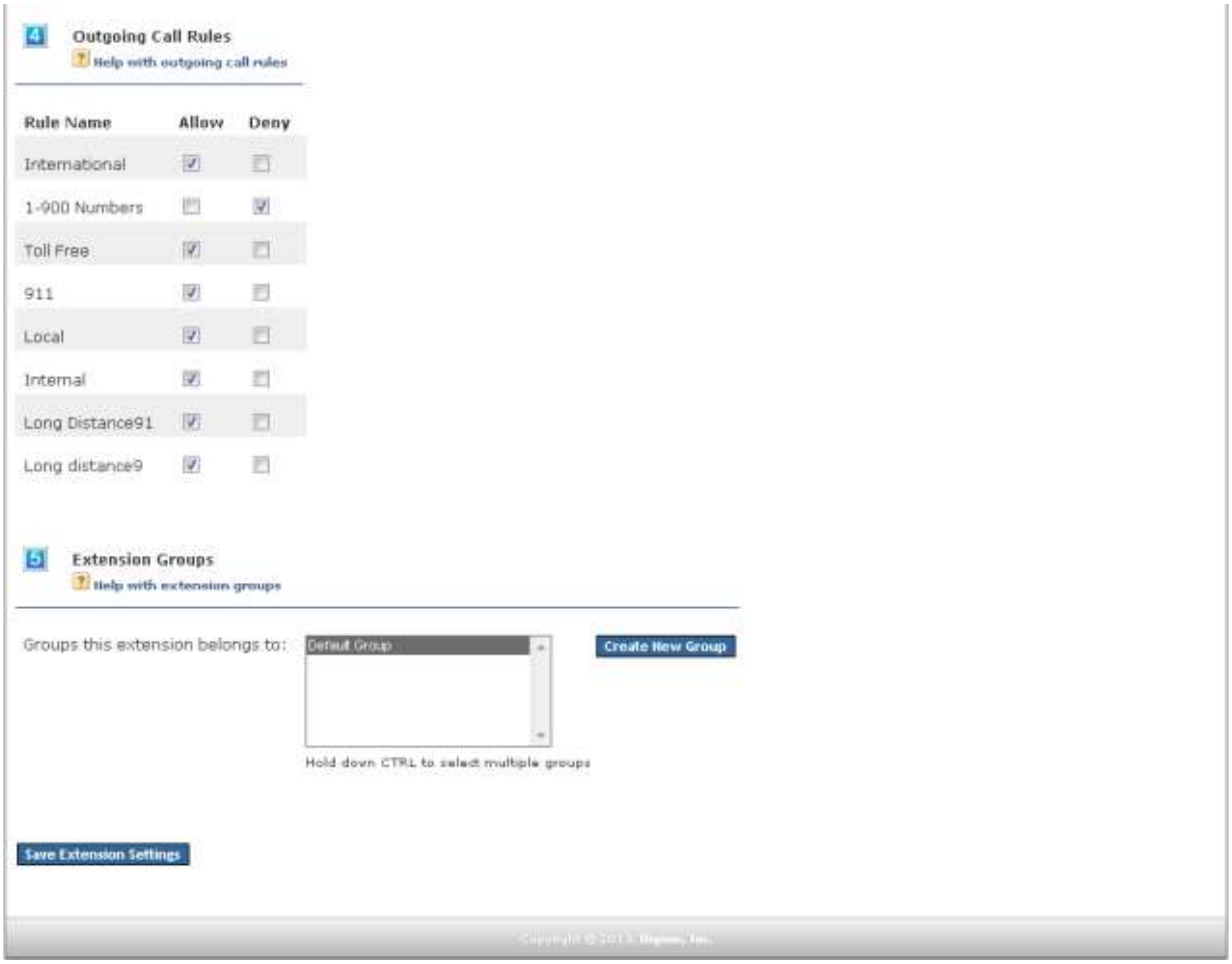


e) Select "rfc2833 (Default)" in the "DTMF Mode" field.

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.

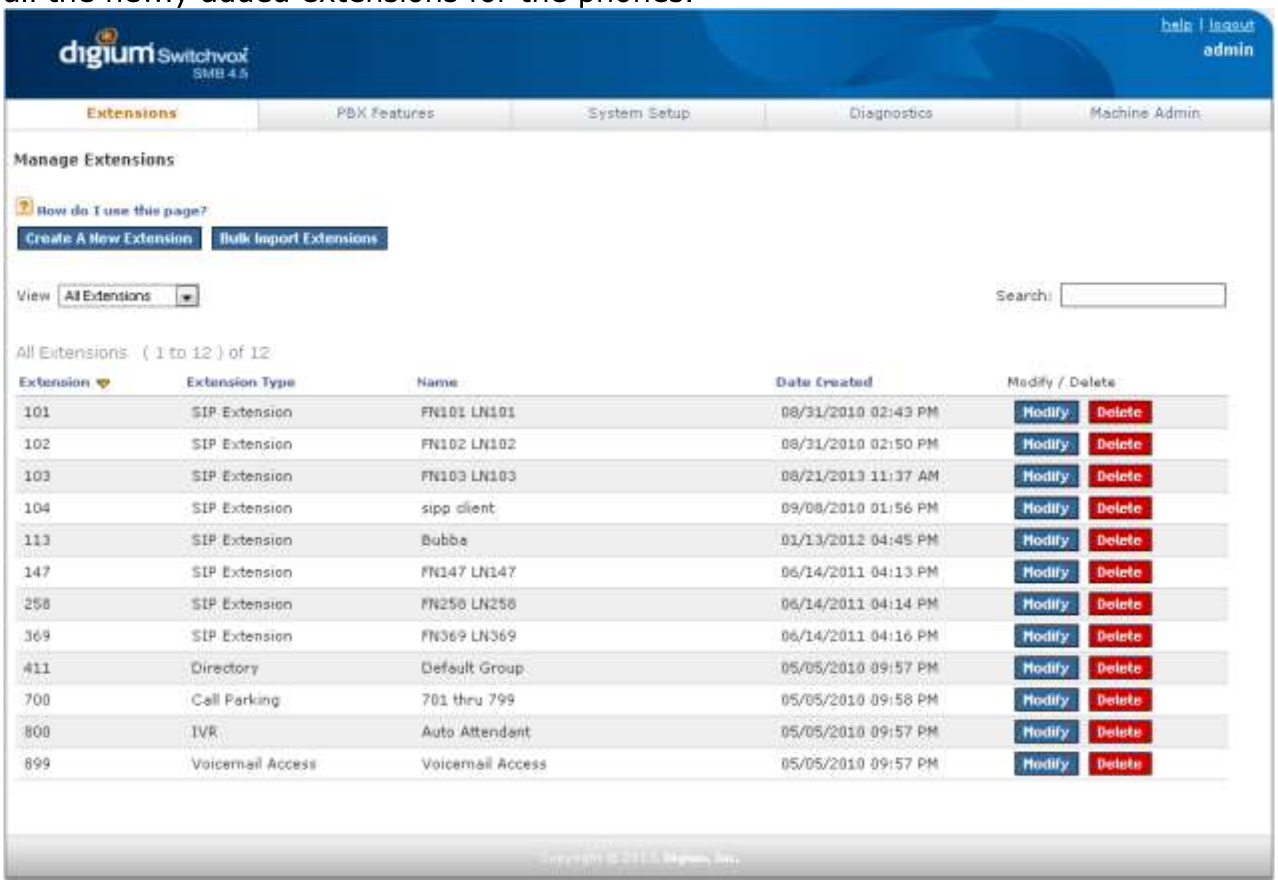

Click to Hide Advanced Options
General Settings
Voicemail/Fax Mailbox Quota (MB)
What does this mean?
(A megabyte is approximately equal to 1 minute of voicemail)
This extension can be dialed from an IVR.
What does this mean?
Phone Settings
DTMF Mode
What is DTMF Mode?
rfc2833 (Default)
rfc2833 (Default)
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?
Always
Call API Settings
Prepend a 1 if number is 10 digits in length
Digits to prepend if number is not an extension
9

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



5.9 Manage Extensions

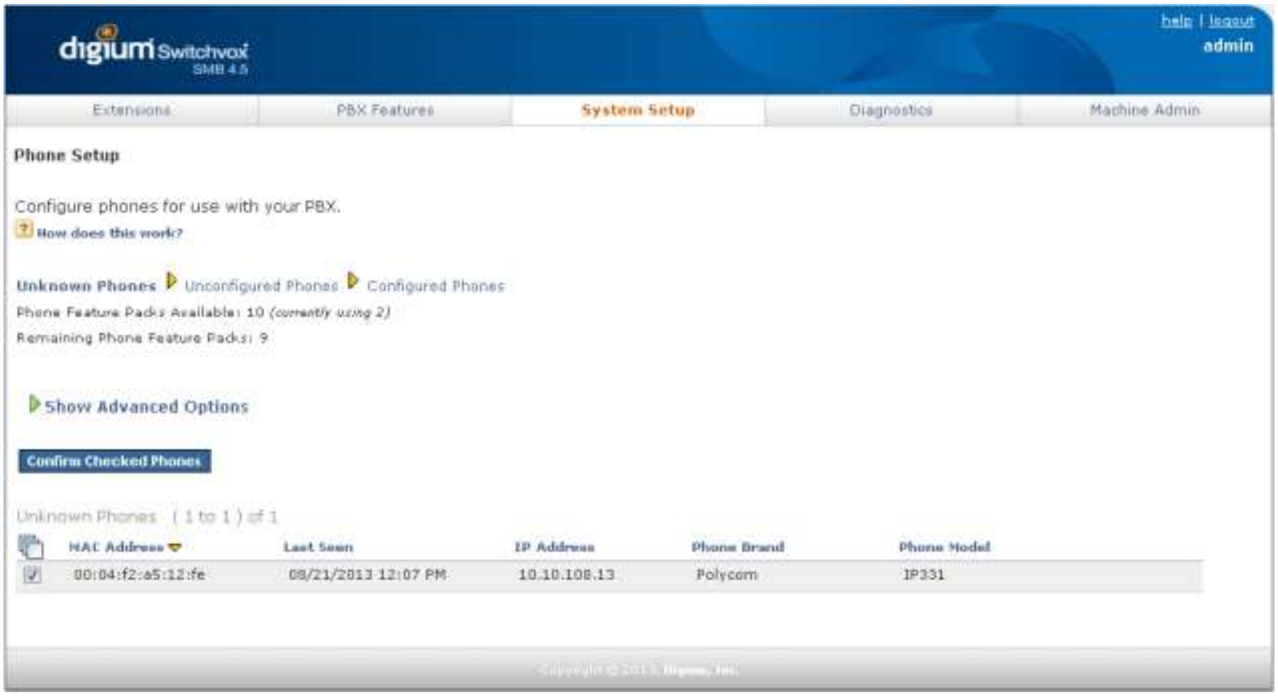
Select "Extension", select "Manage Extensions" to see all the default extensions and all the newly added extensions for the phones.



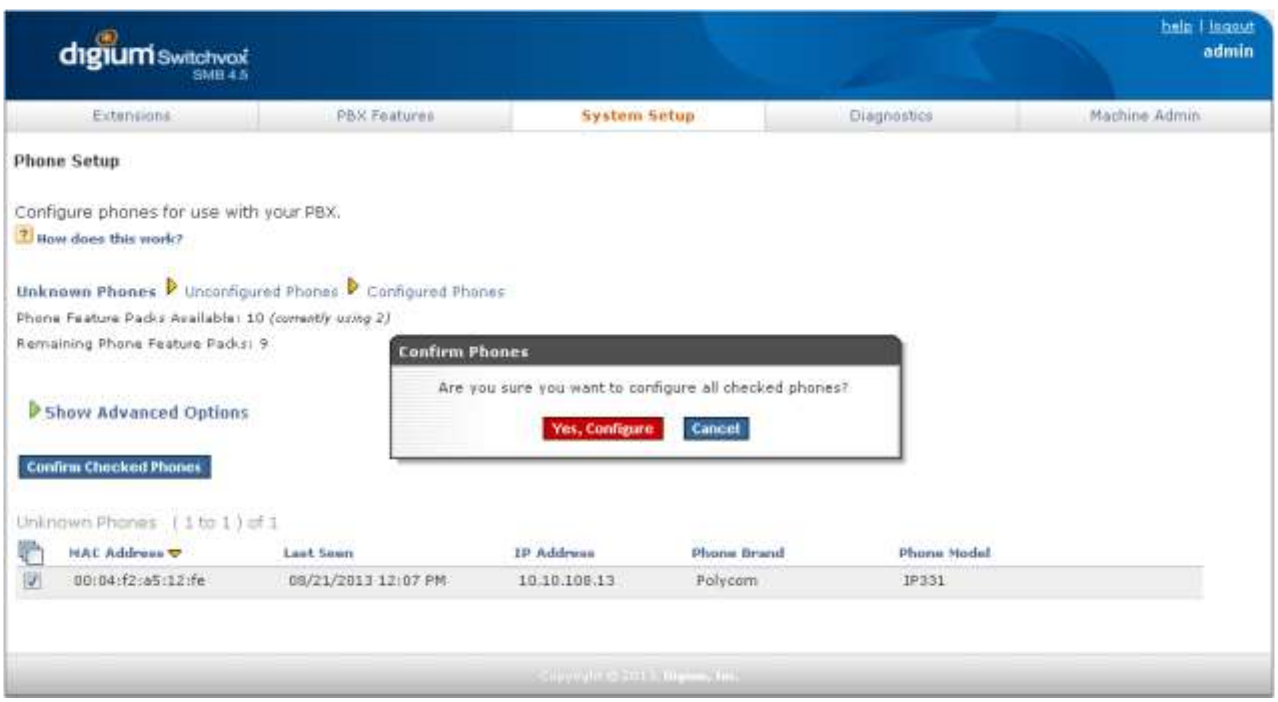
5.10 Phone Setup

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.

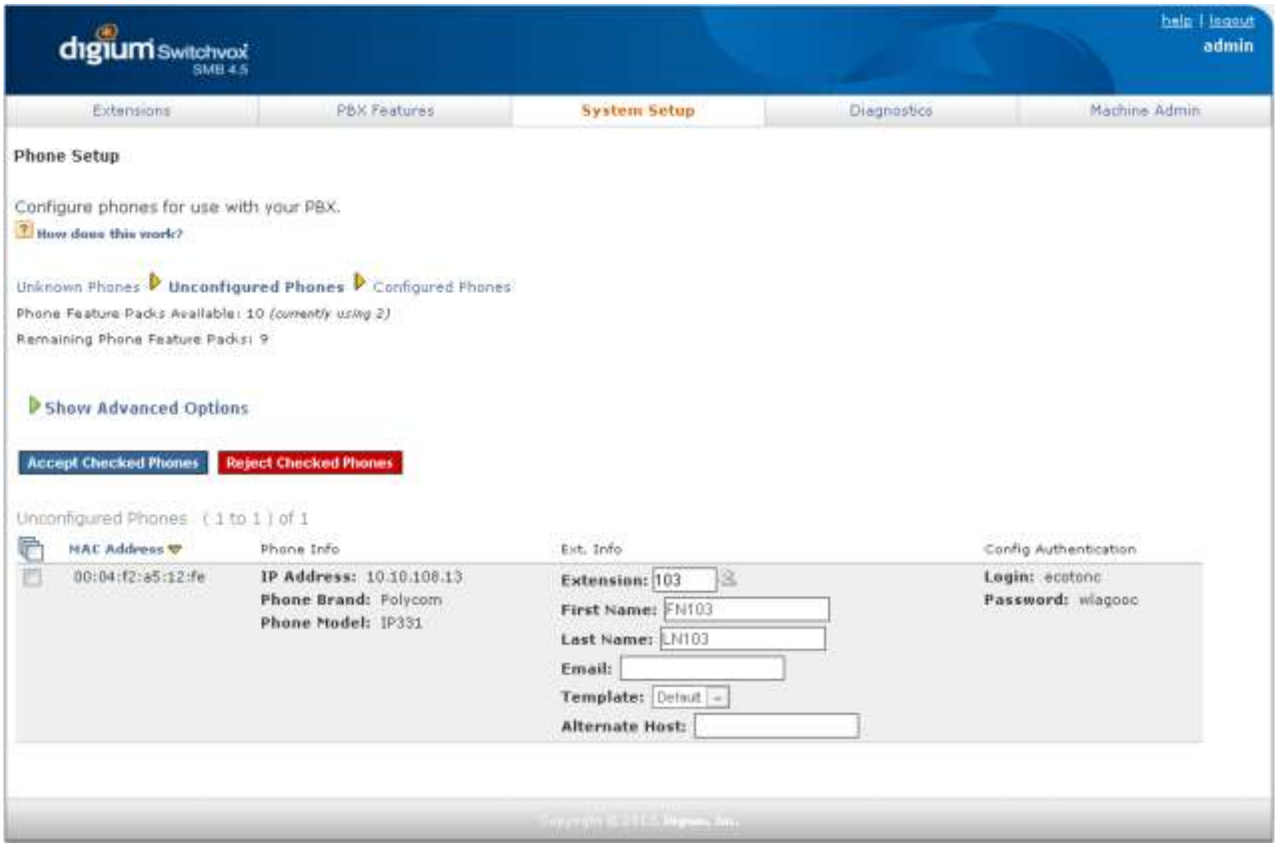
a) To configure the Polycom phone, check the box next to it and click the "Confirm Checked Phones" button. This should move the Polycom phone to the "Unconfigured Phones" list.



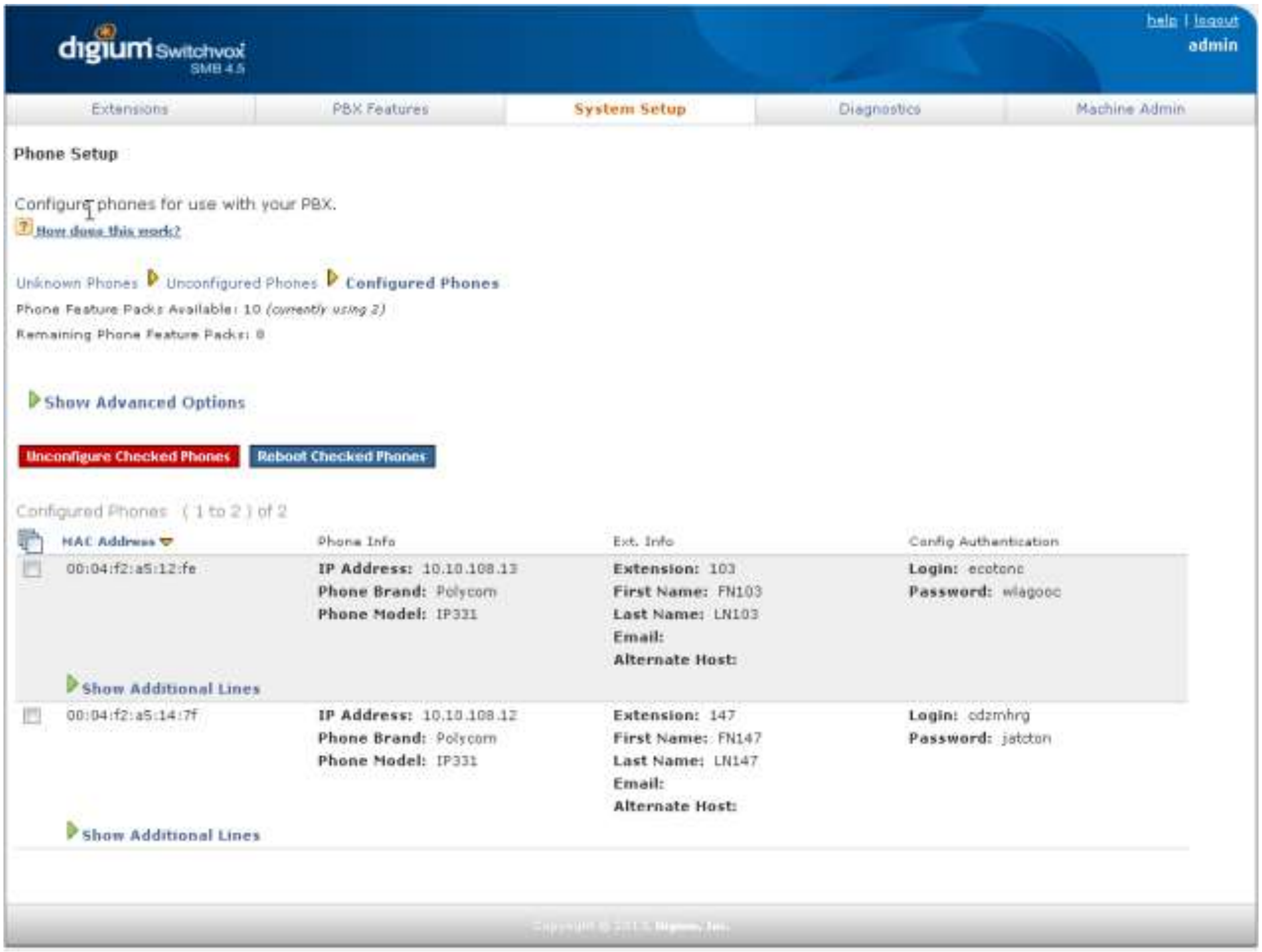
b) Click the "Yes, Configure" button.



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.



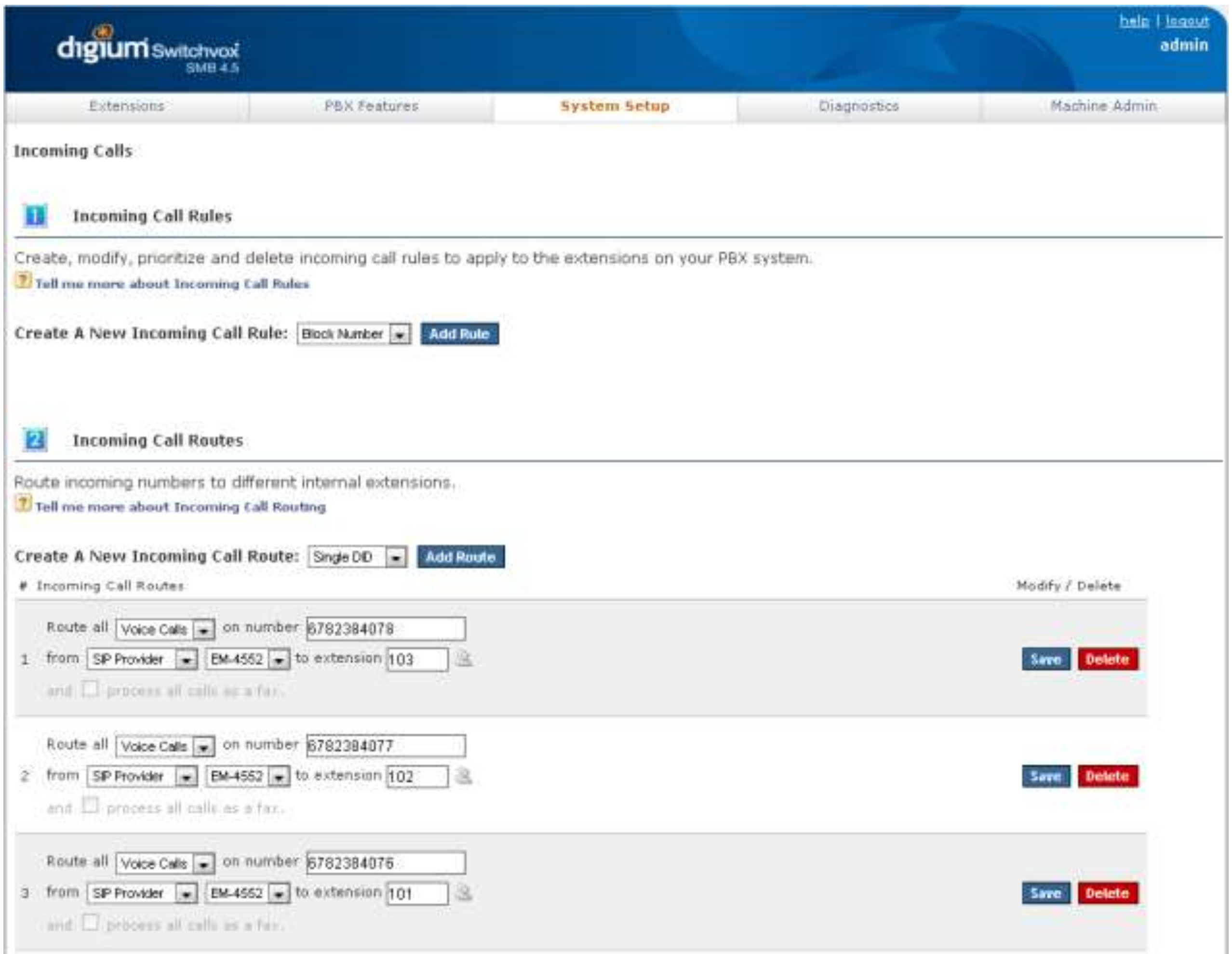
d) Select "System Setup", select "Phone Setup" and click on "Configured Phones" and you should see the phones you just configured are now in the "Configured Phones" list.


digium Switchvox SMB 4.5
help | logout
admin
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: 8
Show Advanced Options
Unconfigure Checked Phones
Reboot Checked Phones
Configured Phones (1 to 2) of 2
MAC Address
Phone Info
Ext. Info
Config Authentication
00:04:f2:a5:12:fe
IP Address: 10.10.108.13
Phone Brand: Polycom
Phone Model: IP331
Extension: 103
First Name: FN103
Last Name: LN103
Email:
Alternate Host:
Login: ecotonc
Password: wlagooc
Show Additional Lines
00:04:f2:a5:14:7f
IP Address: 10.10.108.12
Phone Brand: Polycom
Phone Model: IP331
Extension: 147
First Name: FN147
Last Name: LN147
Email:
Alternate Host:
Login: cdzmhrg
Password: jatcton
Show Additional Lines

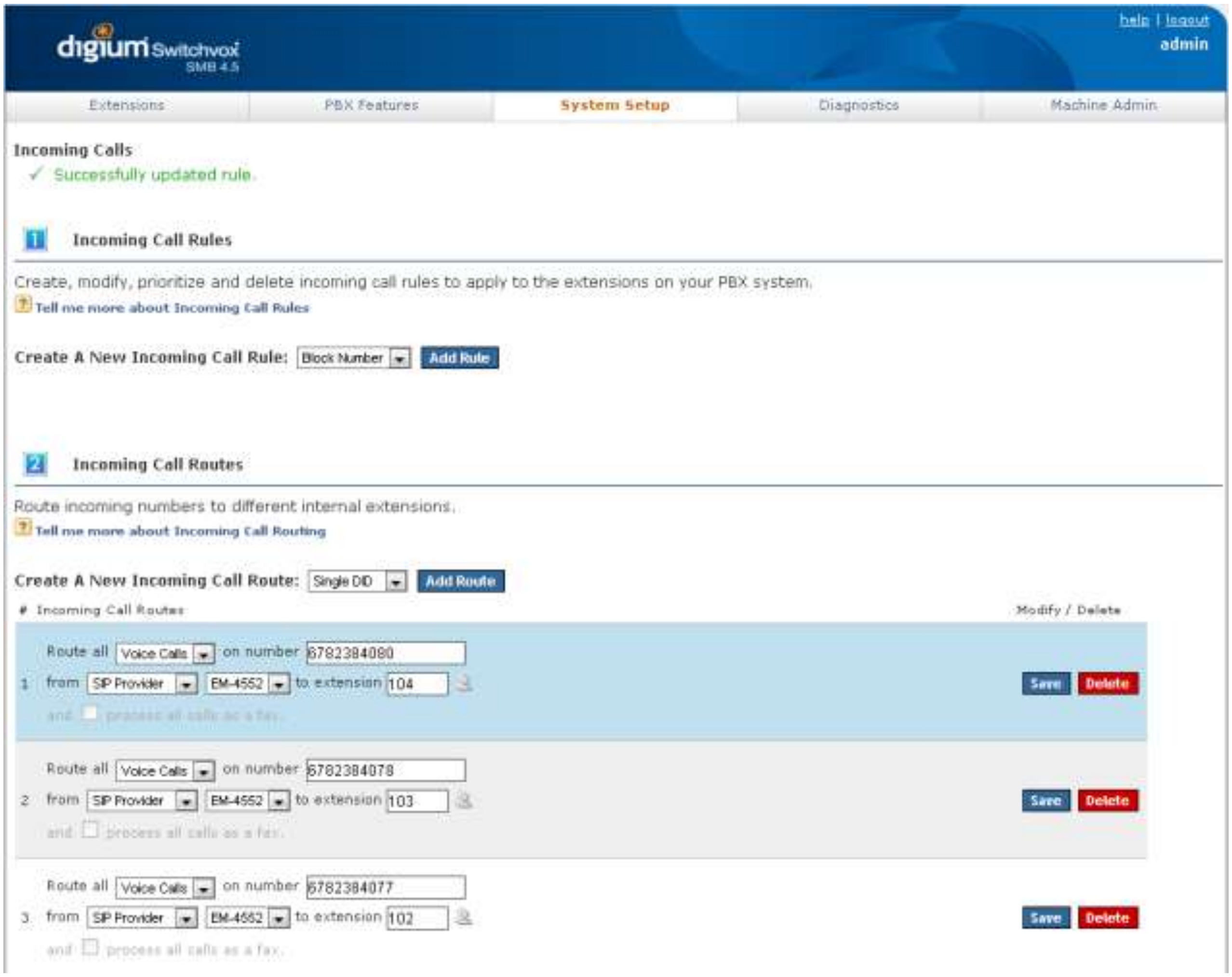
5.11 Incoming Calls

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.

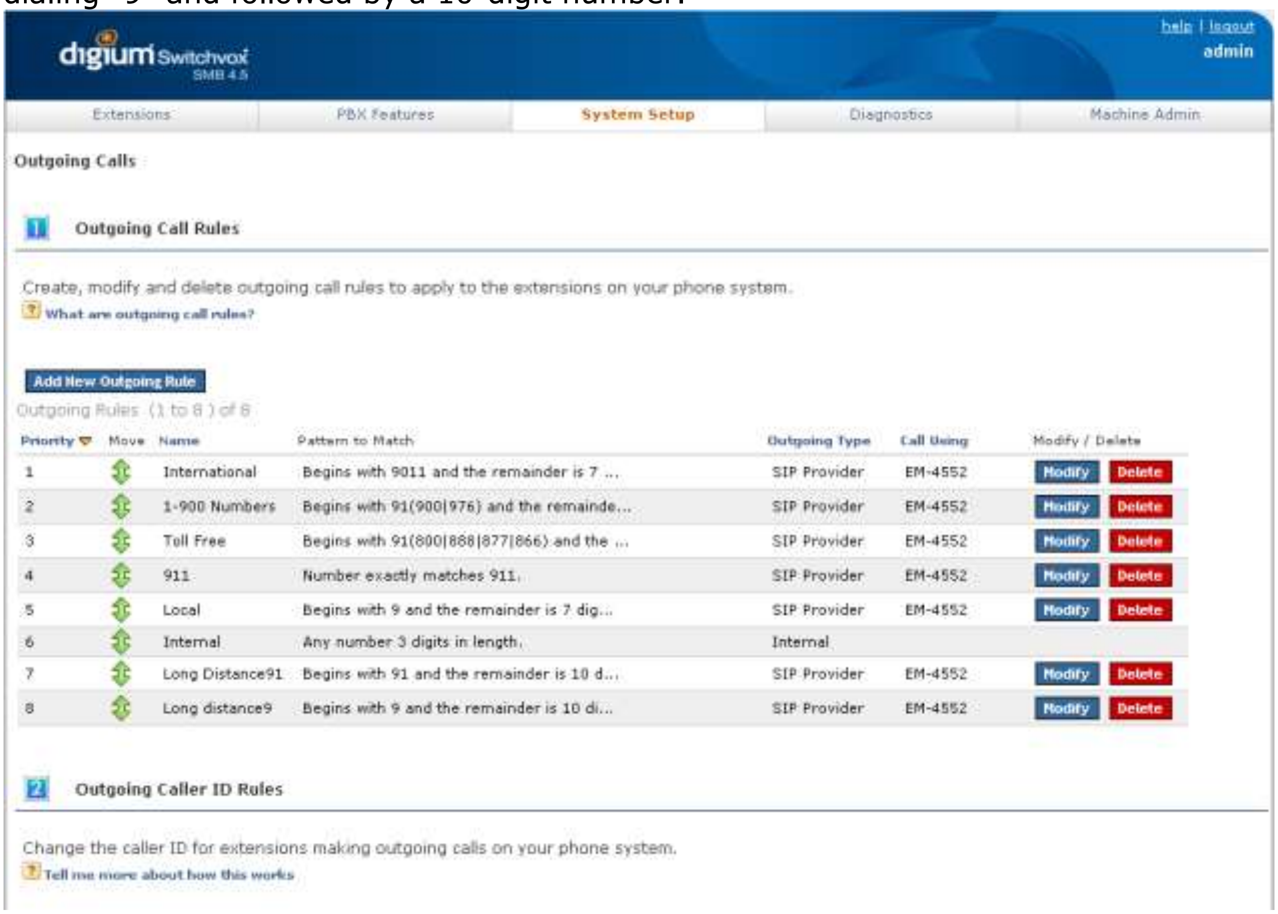


b) From the new entry (high-lighted), select "Voice Calls" in the "Route all" field, enter the assigned DID for the "number" field, select "SIP Provider" in the "from" field, enter the phone extension in the "extension" field and hit the "Save" button.


digium Switchvox SMB 4.5
help | logout
admin
Extensions
PBX Features
System Setup
Diagnostics
Machine Admin
Incoming Calls
Successfully updated rule.
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
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
1 Route all Voice Calls on number 6782384080 from SIP Provider EM-4552 to extension 104 and process all calls as a fax.
Save Delete
2 Route all Voice Calls on number 6782384078 from SIP Provider EM-4552 to extension 103 and process all calls as a fax.
Save Delete
3 Route all Voice Calls on number 6782384077 from SIP Provider EM-4552 to extension 102 and 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.



a) Enter a descriptive name in "Rule Name" field.

b) Select "Yes" for the "Is this rule final?" question.

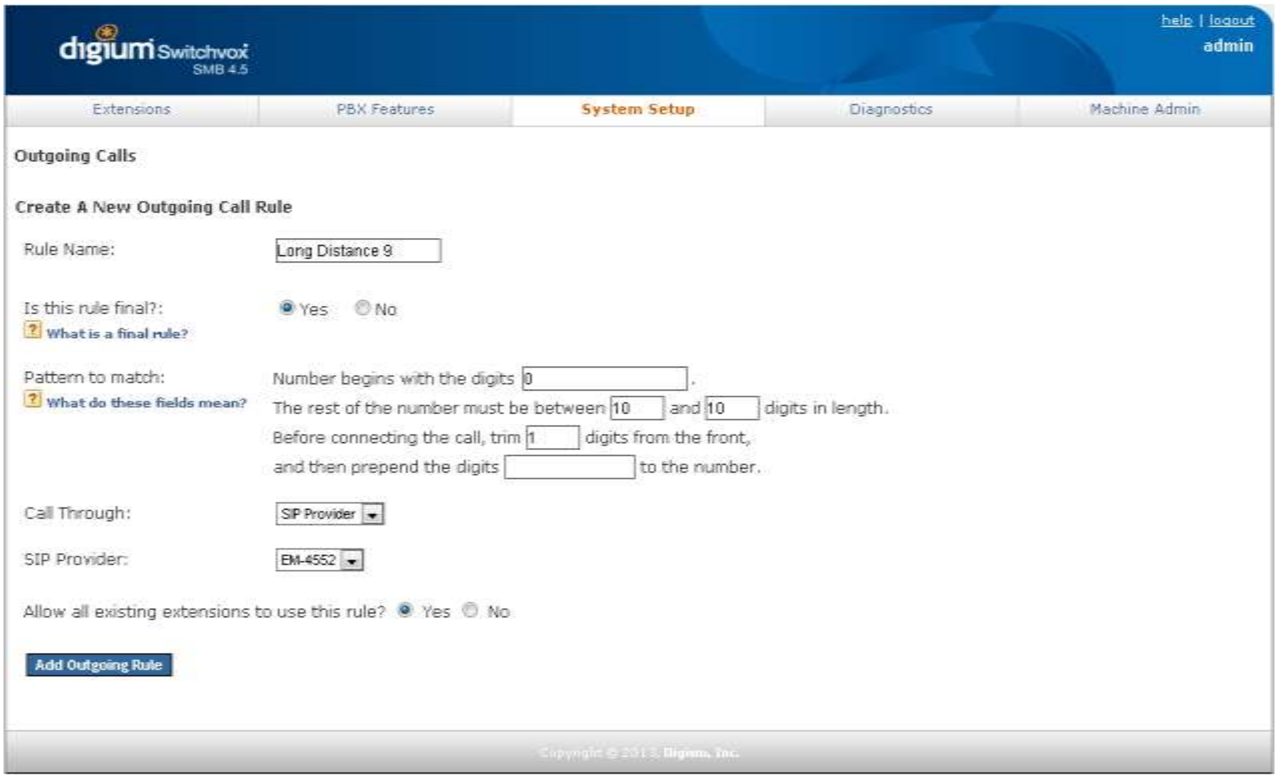
c) For the "Pattern to match" fields, enter "9" as the beginning digit, enter "10" as the exact length of the phone number, enter "1" as the number of digits to trim.

d) Select "SIP Provider" in the "Call Through" field.

e) Select the provider name of EdgeMarc in the "SIP Provider" field.

f) Select "Yes" for the "Allow all existing extensions to use this rule?" question.

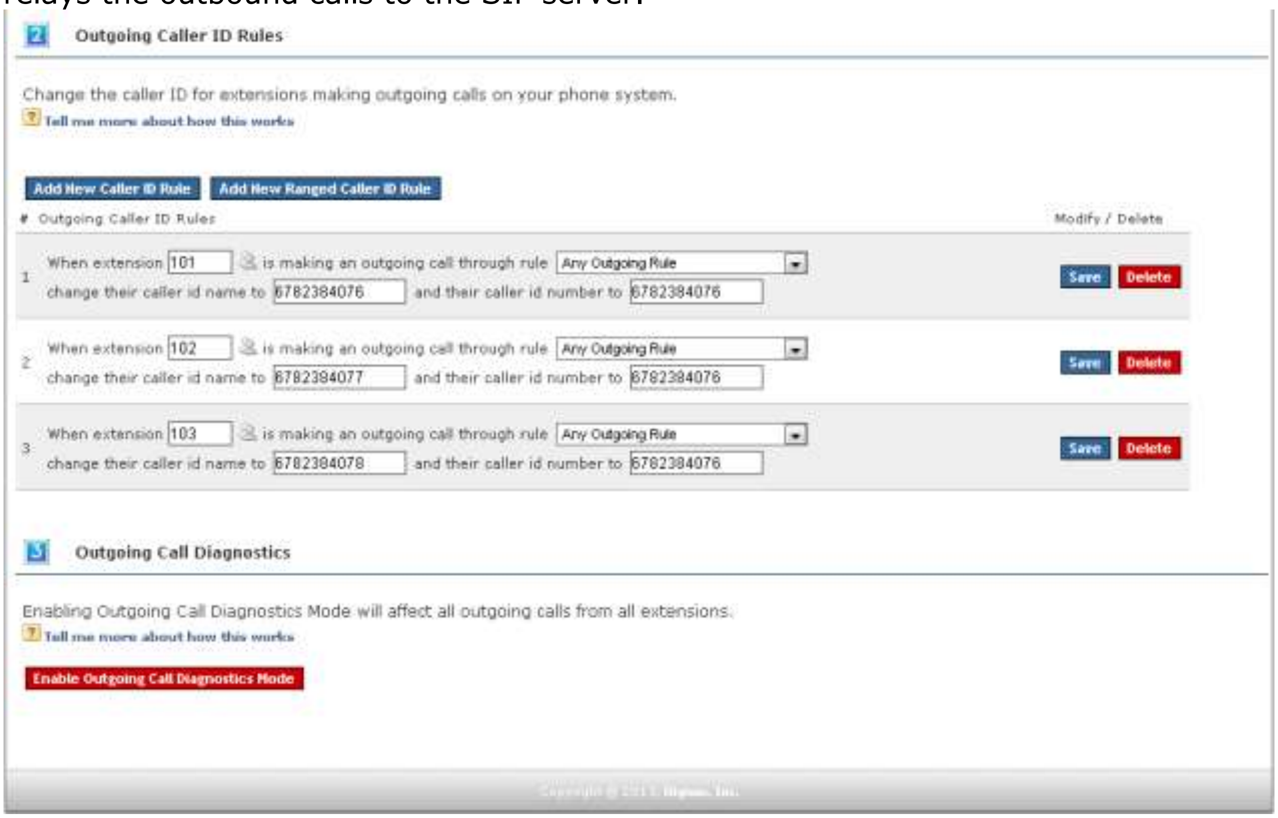
g) Hit the “Add Outgoing Rule” button.



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.



5.14 Static IP Outgoing Caller ID

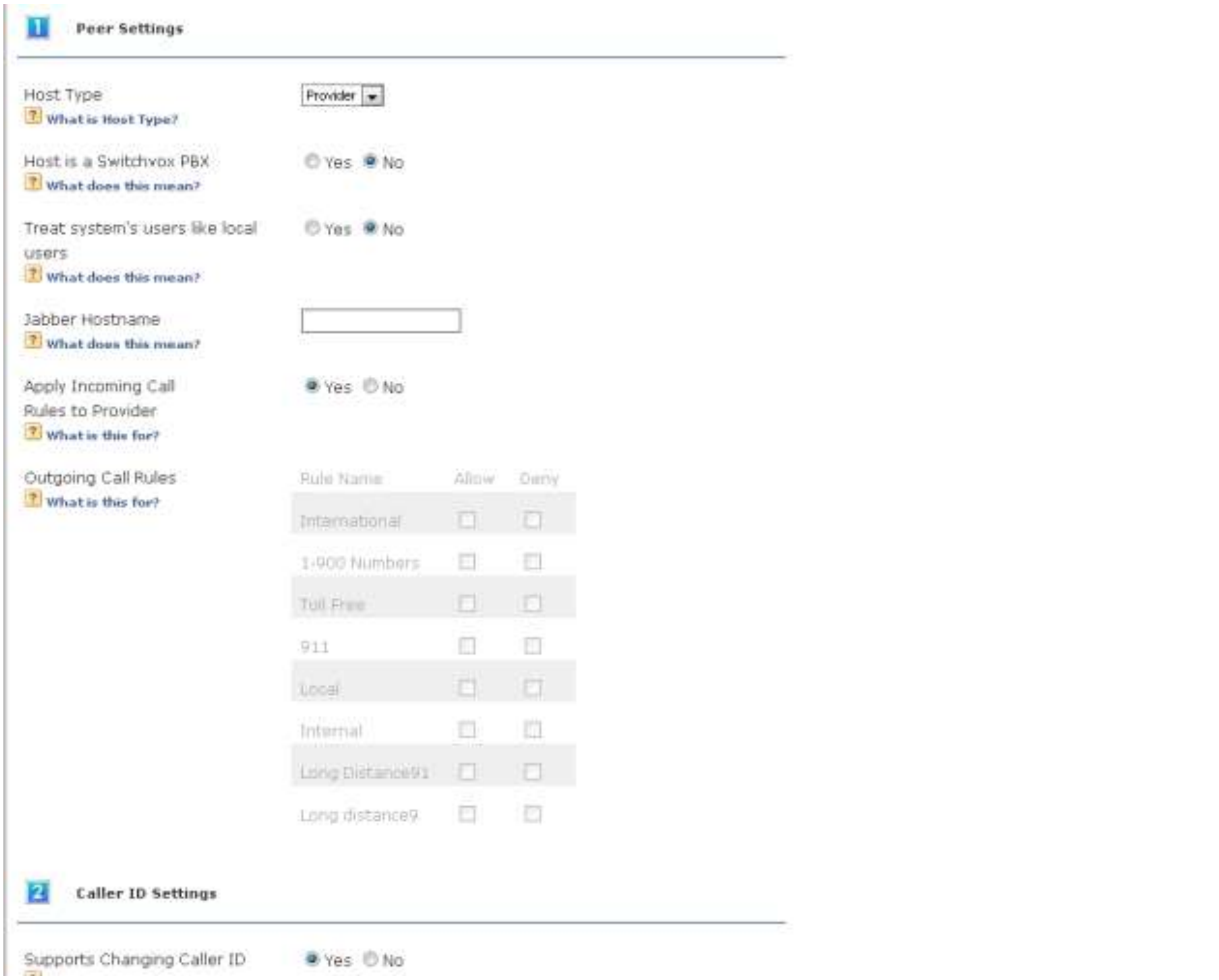
If you have configured the PBX for static IP mode, you may opt to have either the pilot DID or non-pilot DID as the caller ID of the outbound call:

a) Select “System Setup”, select “VoIP Providers” and hit the “Modify” button to modify the VoIP Provider account set up for EdgeMarc.

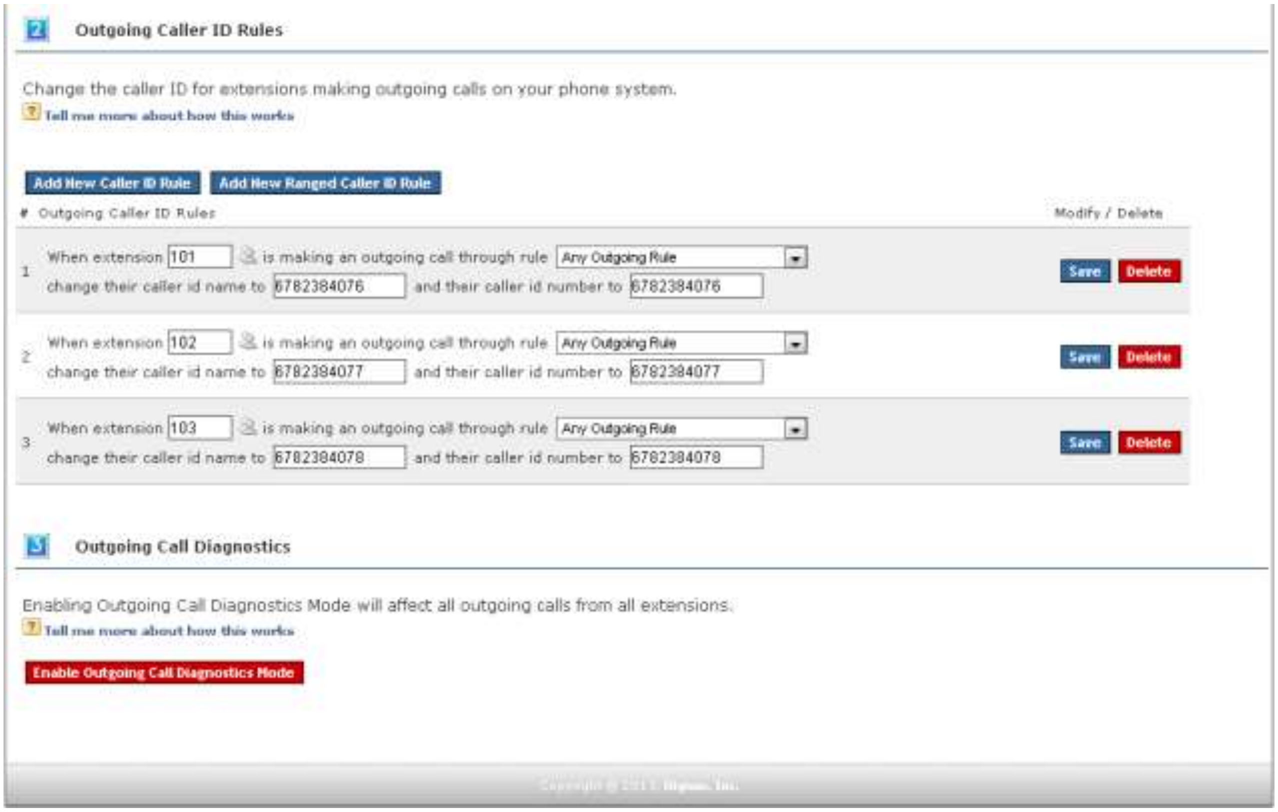
b) Click the “Click to Show Advanced Options” link.

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.

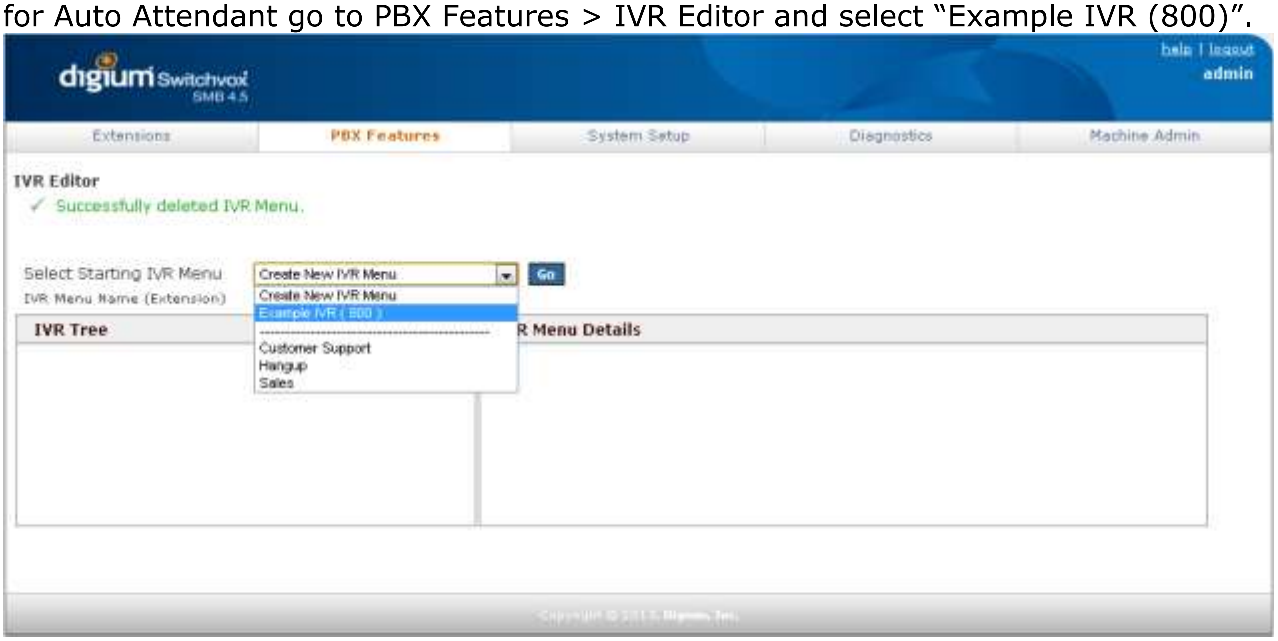


d) Select "System Setup" and select "Outgoing Calls" to set Caller ID for outbound calls in the Outgoing Caller ID Rules section. For each DID assigned to an extension, set the Caller ID number to the same as the assigned DID.

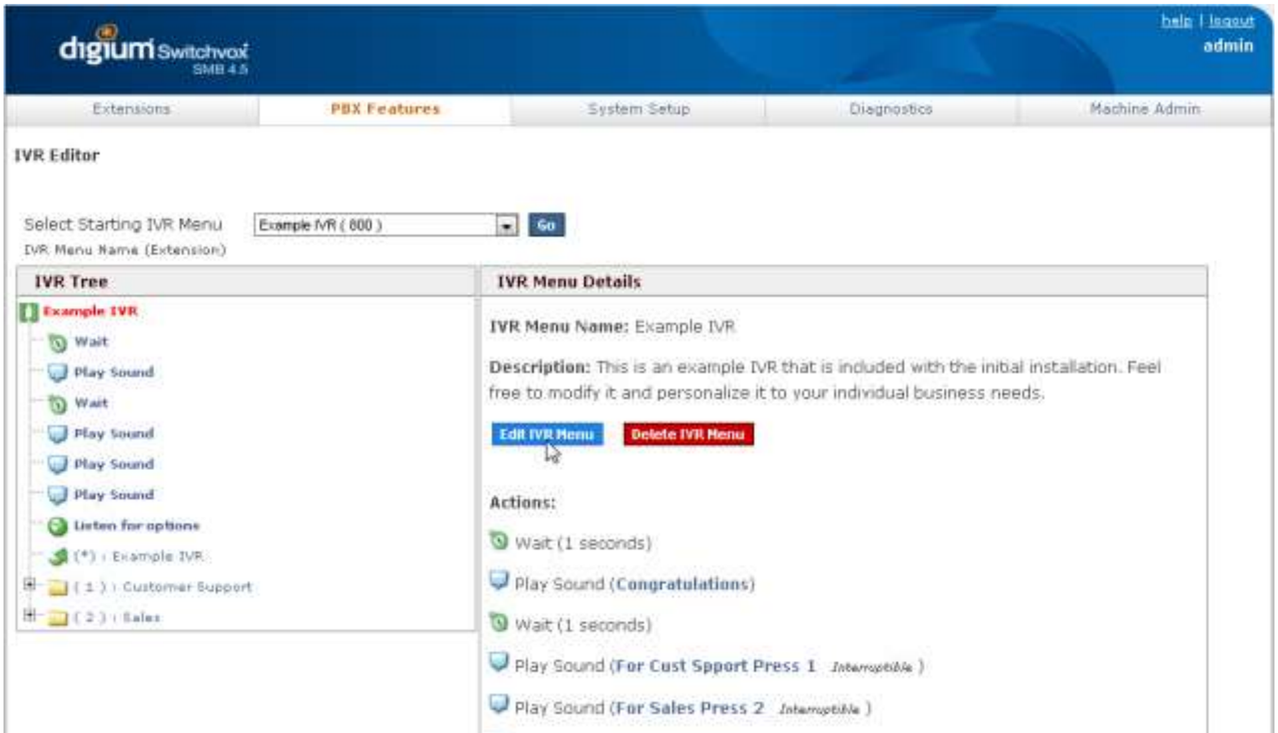


5.15 Auto Attendant

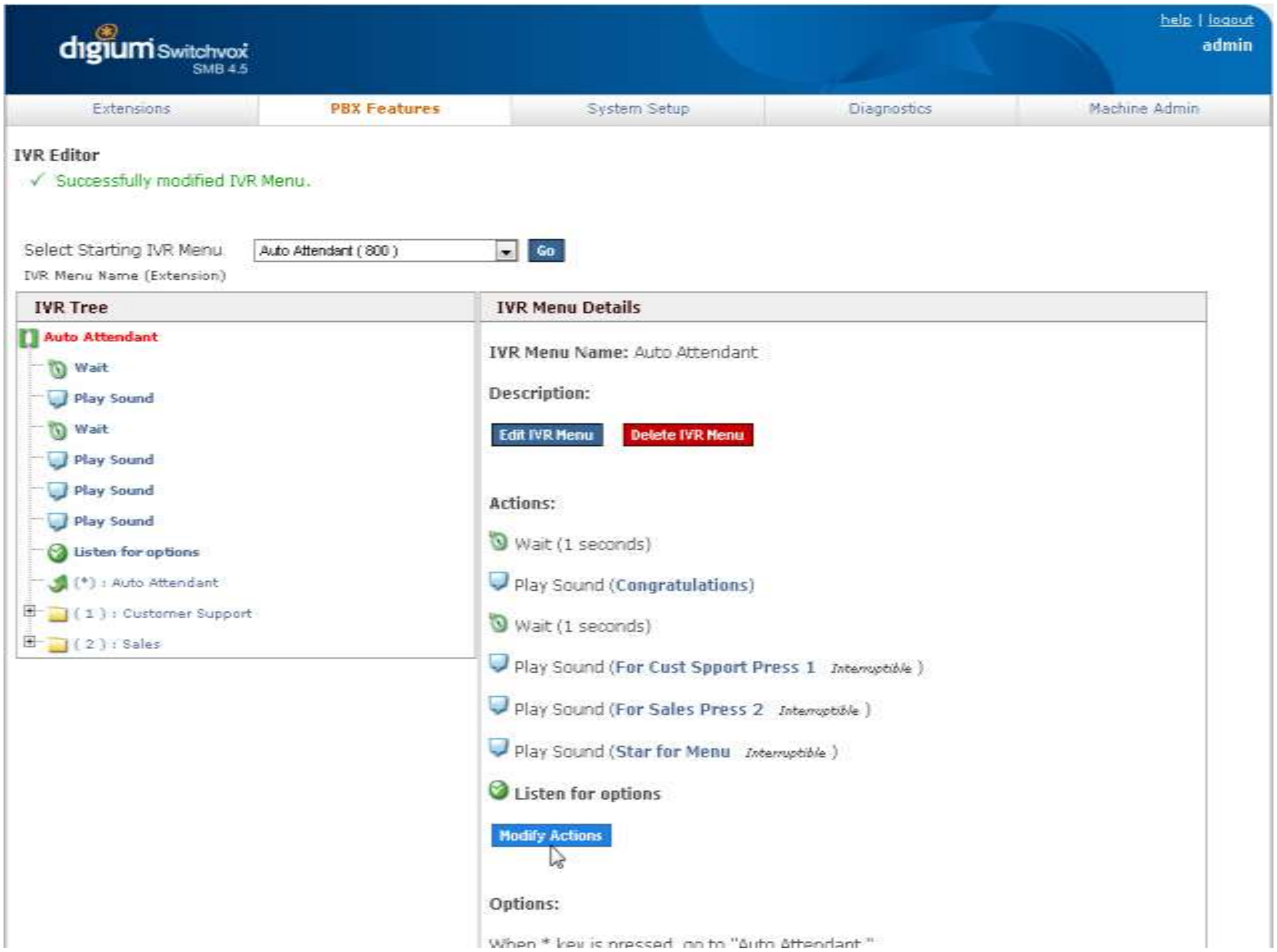
Auto Attendant: Auto Attendant in this PBX is labeled under 'IVR', to setup the PBX for Auto Attendant go to PBX Features > IVR Editor and select "Example IVR (800)".



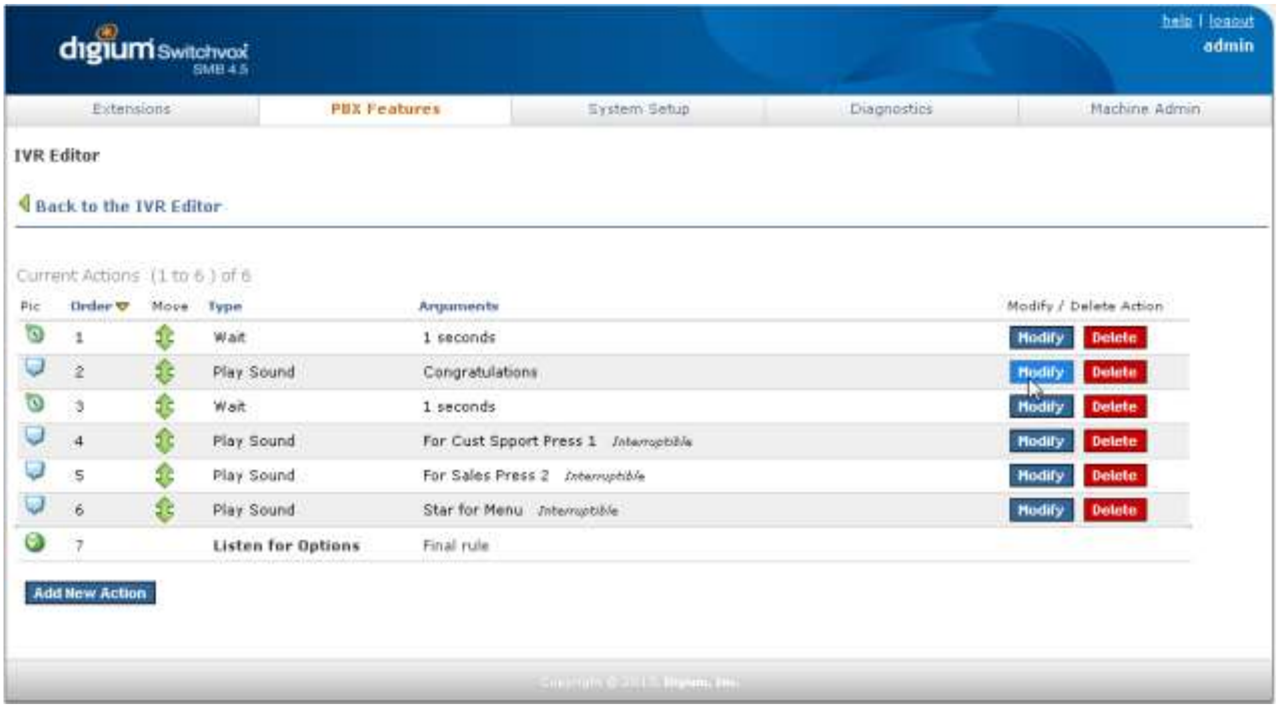
a) Click on “Edit IVR Menu” and change the IVR Menu name to “Auto Attendant” and clear out the Description, then click “Update IVR Menu”



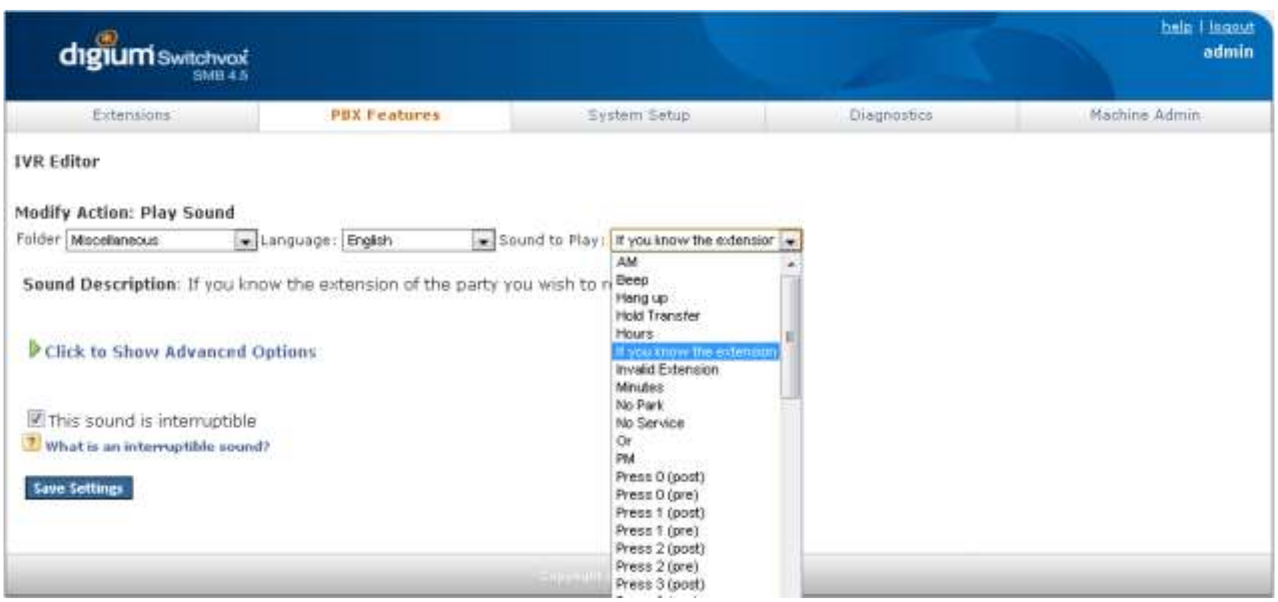
b) Now click on “Modify Actions” to change the Actions layout for extension dialing.



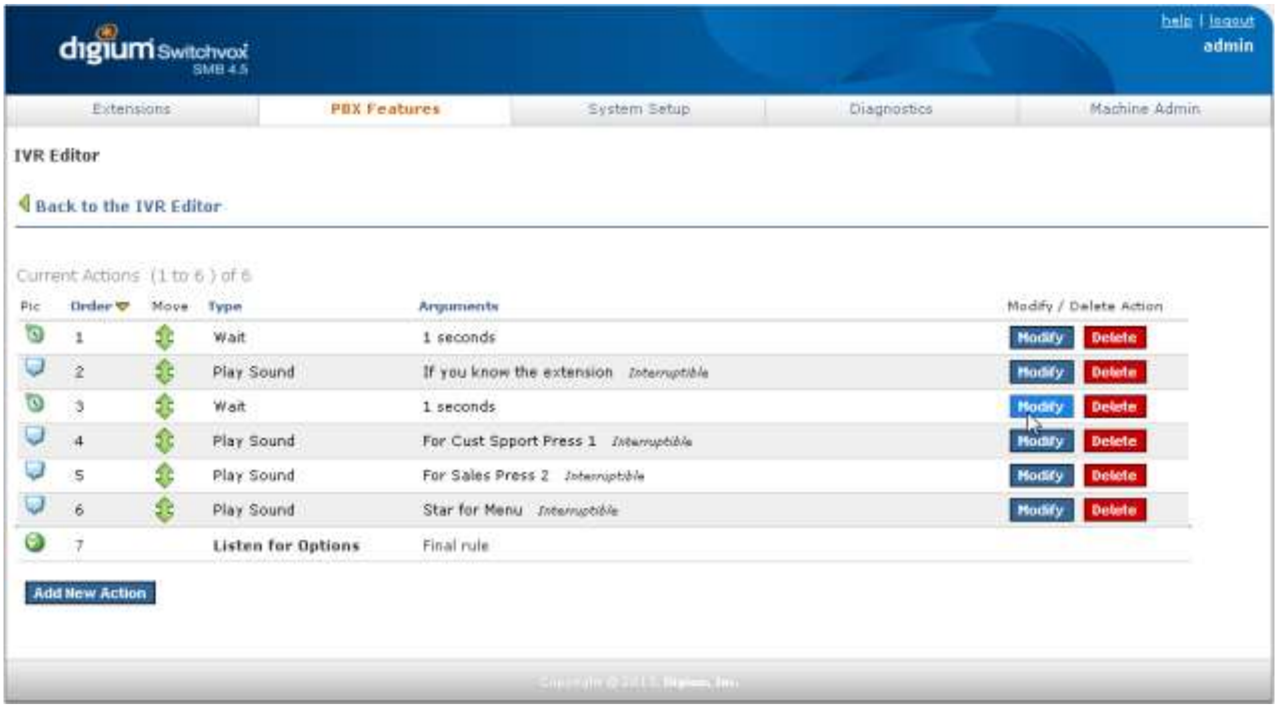
c) Click "Modify" on the second action in the order



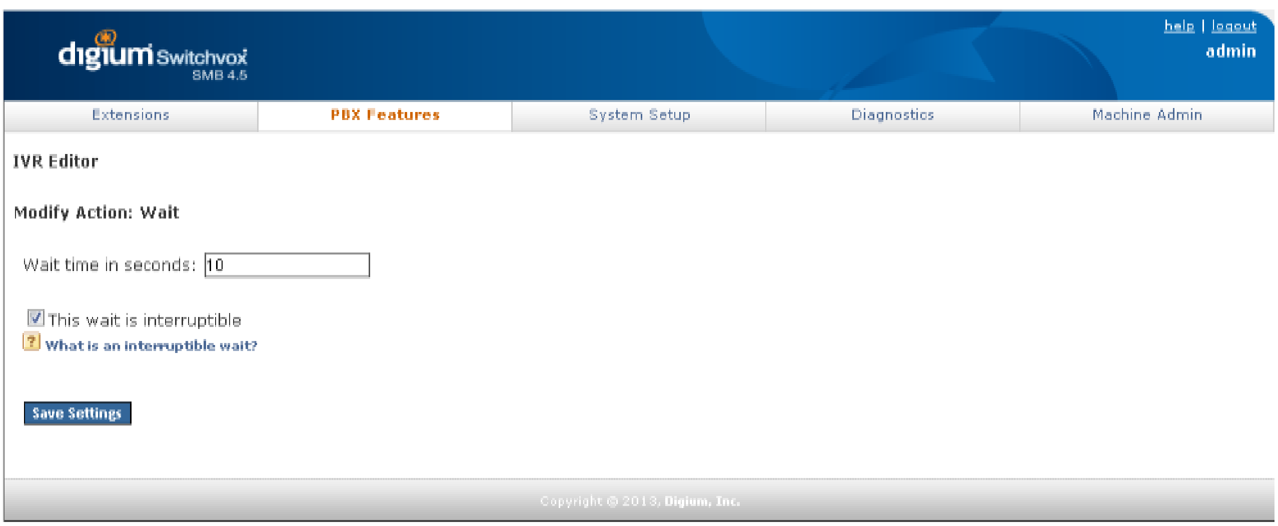
d) Change the Folder to “Miscellaneous” and set the Language to “English” and set the Sound to Play to “If you know the extension”. Make sure “This sound is interruptible” is checked then click “Save Settings”



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

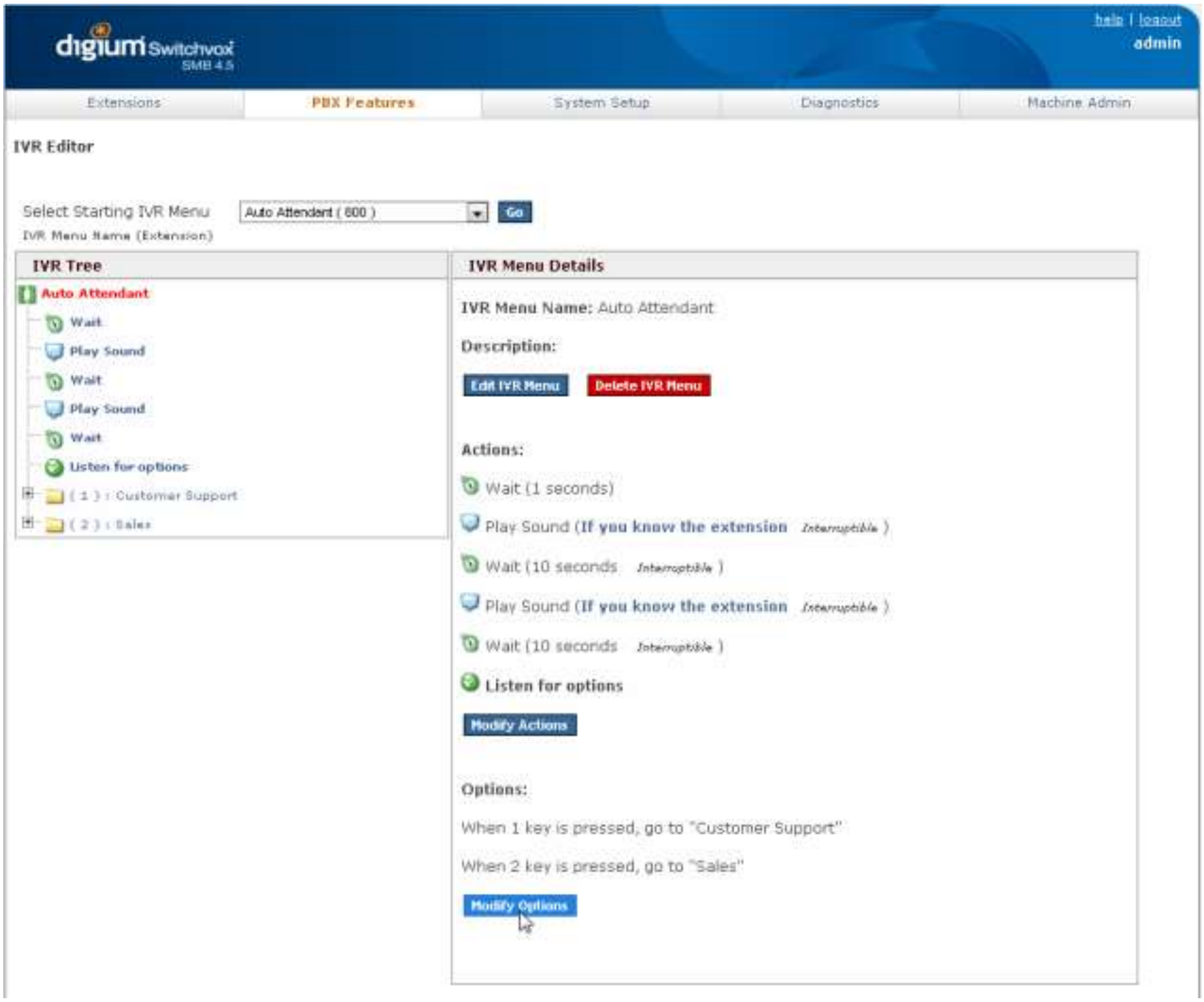


f) Check the box to enable “This wait is interruptible” then click “Save Settings”

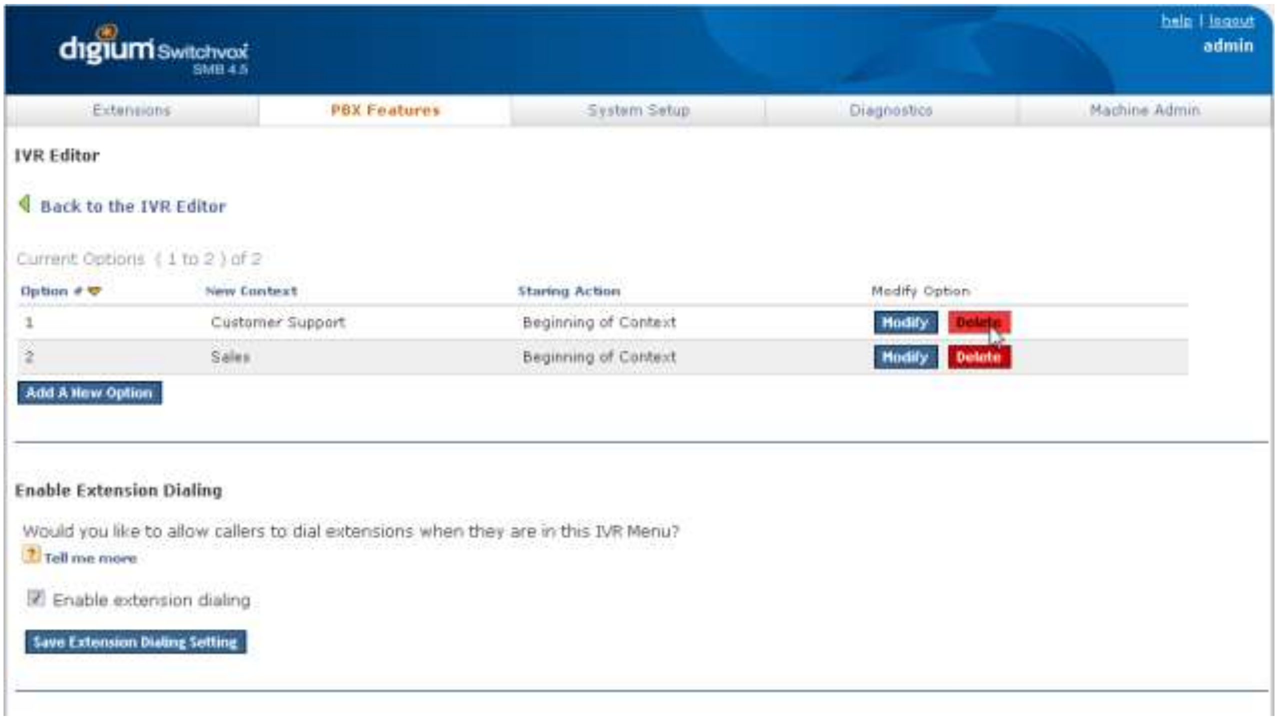


g) Actions 4-6 can be modified to repeat the “Extension” message or they can be deleted

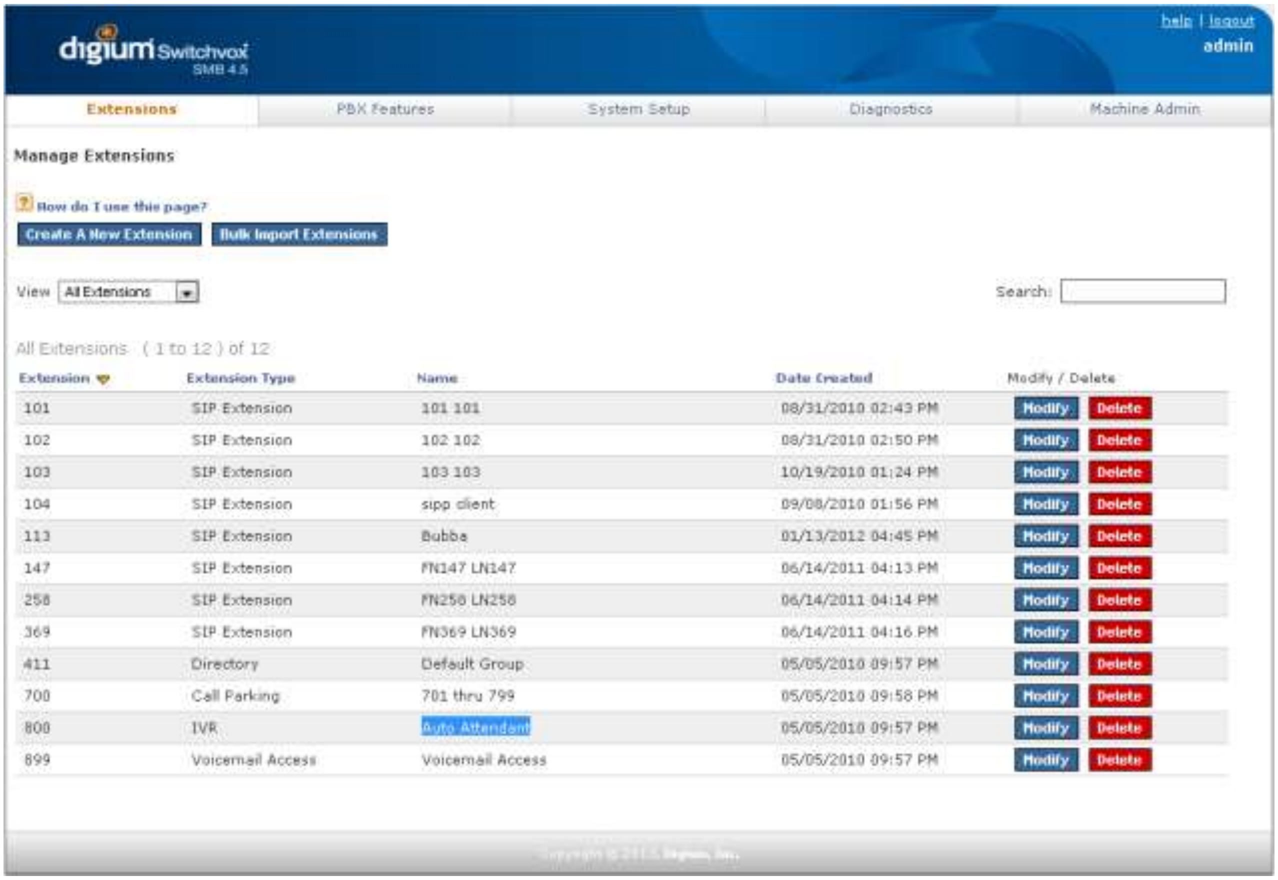
h) Click “Back to the IVR Editor” to go back to the main page, then click “Modify Options”



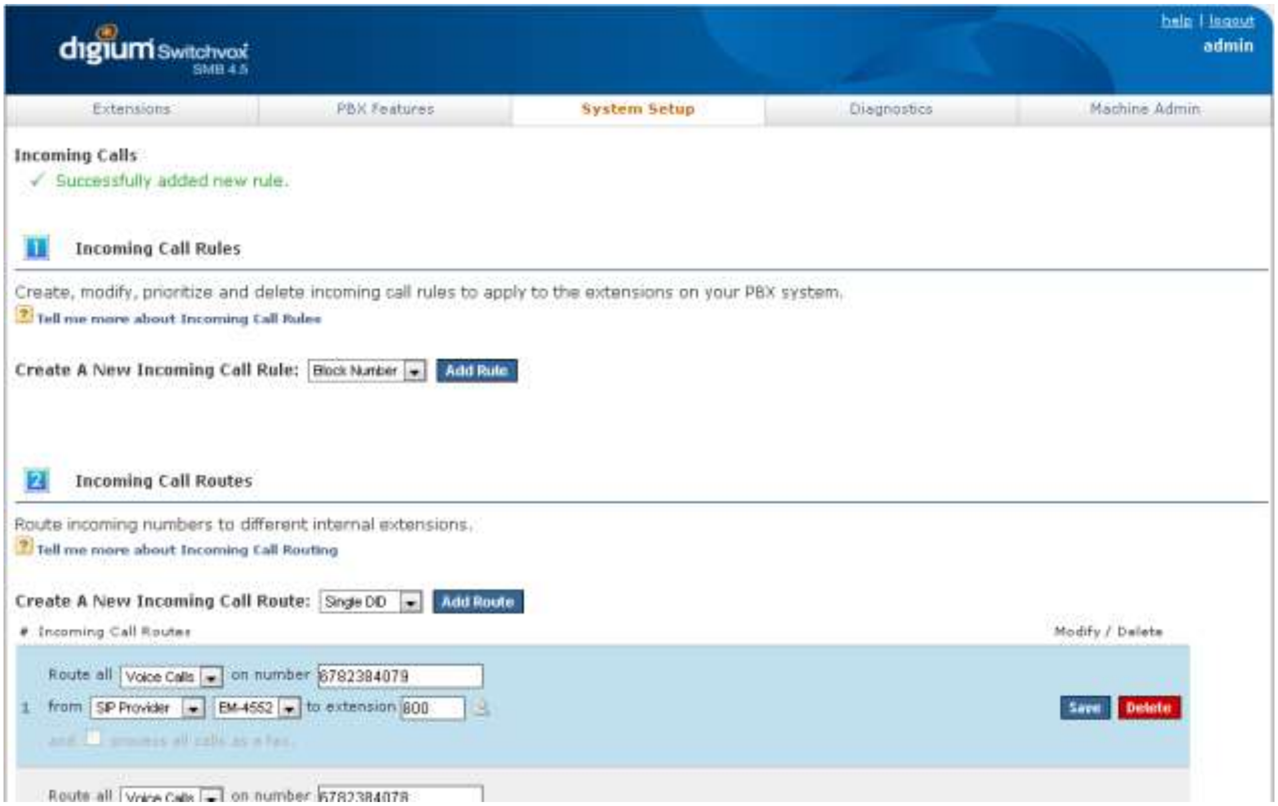
i) Delete the 2 current Options at the top, check the box for "Enable Extension Dialing" and click "Save Extension Dialing Settings"



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



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



For advanced configurations and support please contact the Edgewater Technical Assistance Center support@edgewaternetworks.com or call 408.351.7255.