

Optimum Business
Trunking and the Toshiba®
IPedge® IP-PBX V1.1.3-06
Configuration Guide

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1 Overview

The purpose of this configuration guide is to describe the steps needed to configure the Toshiba IPedge IP-PBX for proper operation Optimum Business Sip Trunking.

2 SIP Trunk Adaptor Set-up Instructions

These instructions describe the steps needed to configure the LAN side of the Optimum Business SIP Trunk Adaptor.

Step 1:

Log on to the Optimum Business SIP Trunk Adaptor

1. Connect a PC to port 4 of the Optimum Business SIP Trunk Adaptor, the silver device labeled Edgewater Networks, 4550 series.



2. Open a Web browser and go to IP Address <http://10.10.200.1>. A login box will appear.

3. Enter login and password and click 'OK'.

Login: pbxinstall
Password: s1ptrunk



Step 2:

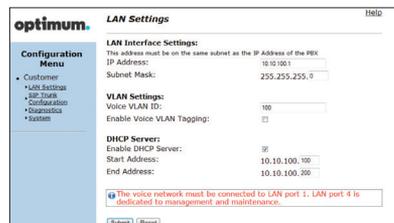
Click on the LAN Settings Link

1. Assign an IP Address to the LAN interface of the SIP Trunk Adaptor. The IP address must be on the same subnet as the IP PBX. This changes the address on port 1 of the Optimum Business SIP Trunk Adaptor.

Note: This will become your local SIP proxy IP address. No other IP addresses will be provided by Cablevision.

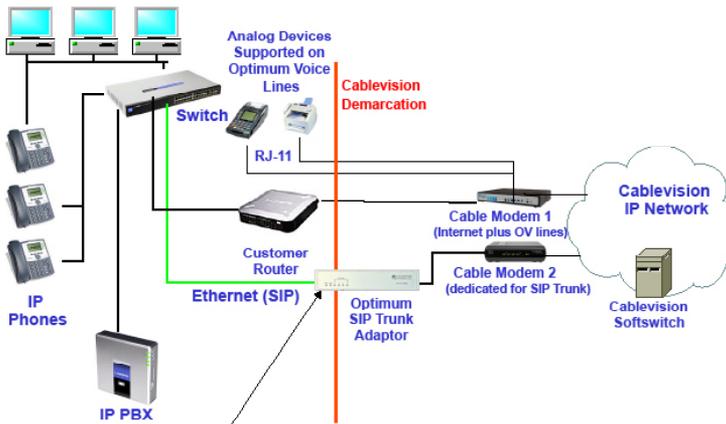
2. Optional: Specify a VLAN for your voice traffic. Click the 'Enable Voice VLAN Tagging' check box. The default VLAN ID is 100.

Note: VLAN 200 should not be used. It is dedicated to port 4 for management.



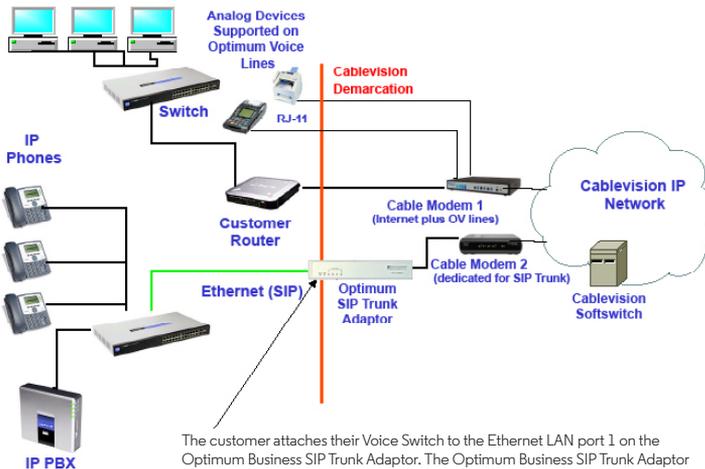
- Optional: Enable the DHCP server. This will allow the SIP Trunk Adaptor to act as a DHCP server, which will provide IP addresses to the voice network, and create a dedicated voice LAN, as per diagram 2.
- Click 'Submit'.

Diagram 1 SIP Trunk Adaptor for IP-PBXs Example: Single LAN Configuration



Using a connection from the customer's LAN, the SIP Trunk Adaptor's address can be a statically assigned private IP address. It may not be assigned a Public IP address.

Diagram 2 SIP Trunk Adaptor for IP-PBXs Example: Separate Voice and Data Networks Configuration



The customer attaches their Voice Switch to the Ethernet LAN port 1 on the Optimum Business SIP Trunk Adaptor. The Optimum Business SIP Trunk Adaptor can be enabled as a DHCP server to provide routing for the separate voice network.

Step 3:

Click on the SIP Trunk Configuration Link

1. Select your IP PBX make and model from the drop-down menu.
2. Specify how the IP PBX will register to the Optimum Business SIP Trunk Adaptor.
3. The Cablevision network only supports Inband DTMF. Click on the check box next to "Convert Inband DTMF" if you cannot configure your IP PBX to send out Inband DTMF. The DTMF tone duration generated by the phones and/or PBX may need to be increased from their default setting. Some phones and/or PBX have a default setting between 180ms to 200ms. This setting is too low. The recommended setting is 600ms.
4. Click 'Submit'.

Step 4:

Diagnostics Link

You can make a test call directly from your phone or use the test call application under the Diagnostics link.

Step 4 continued

Field	Description
Outbound Call Test TelephoneNumber	Specifies an outside phone number to which an outbound call will be initiated. The pilot telephone number of the SIP Trunk will be prepopulated.
Pilot Number	Displays the provisioned pilot number, which is used for outbound and inbound call tests.
Call	Initiates a call outbound to a telephone number entered or inbound to the pilot number displayed.
Inbound Call Test (radio button)	Indicates whether inbound test call will be enabled or disabled. If inbound test calls are enabled, calls made to the pilot number will be redirected to the test UA for fifteen minutes. When the pilot number is dialed, you will hear a test message play.
Submit	Enables or disables the inbound call test.
IP Address to Ping	Verifies basic connectivity to a networking device. Successful ping test results indicate that both physical and virtual path connections exist between the system and the test IP address.
Ping Button	Sends a ping to the IP address specified in the field "IP Address to Ping".
IP Address to Trace	Tracks the progress of a packet through the network. The packet can be tracked through the WAN or LAN interfaces of the adaptor.
Interface (radio button)	Indicates whether a packet will be tracked through the LAN or the WAN.
Traceroute Button	Initiates a traceroute to the specified IP address on either the LAN or the WAN.
Reset	Clears all fields and selections and allows you to enter new information. Reset applies to outbound call test, ping and traceroute.

3 Additional Set-up Information Systems


System
[Help](#)

Configuration Menu

- Customer
 - ▶ LAN Settings
 - ▶ SIP Trunk Configuration
 - ▶ Diagnostics
 - ▶ System

Software Version:
Version 11.6.14.1 -- Fri Jan 4 17:49:28 PST 2013

Hostname:
5164939899

Model:
EdgeMarc 4552

Vendor:
Cablevision

LAN Interface MAC Address:
A8:70:A5:00:D8:18

Registration Status:
The ALG feature is registered. View [license key](#).

System Date:
02/29/2016 15:03:40 UTC

Change Password:

- [pbxinstall](#)

Field	Description
Pbxinstall Link	Select to change the default password for the pbxinstall login ID. Only the password can be changed. The login ID cannot be changed.

Password


Set Password
[Help](#)

Change the GUI password by filling in the fields below. The password must be between 6 and 8 characters in length.

Configuration Menu

- Customer
 - ▶ LAN Settings
 - ▶ SIP Trunk Configuration
 - ▶ Diagnostics
 - ▶ System

Username:

Current Password:

New Password:

Confirm Password:

Field	Description
Username	Specifies the username for which the password can be changed.
Current Password	Specifies the current password.
New Password	Specifies the new password.
Confirm Password	Confirms the new password.
Submit	Applies the settings configured on this page.
Reset	Clears all fields and selections and allows you to enter new information.

4 International Calling

Optimum Voice Business Trunking offers an optional International Calling Service for direct-dialed calls made from the Customer's business or from any phone via the Optimum Voice International Calling remote access number to destinations outside of the United States, Puerto Rico, Canada and the U.S. Virgin Islands at per minute rates. The Customer must login to the Optimum Business Account Center and activate the service on the Optimum Business Trunk Pilot telephone number to activate the service and manage the monthly International spending limit for the account.

Activating International calling on the Pilot TN will enable International calling for all Direct Inward Dial (DIDs) telephone numbers as well. Blocking International calling for one or more DIDs is managed by the customer directly from the PBX phone system configuration. To minimum the exposure to fraudulent calling, It is recommended to limit International calling capability to those DID's that require it and set up an account spending limit that reflects what is necessary to run the business.

It is the Customer (and/or the Customer Agent's) responsibility to properly secure the customer's PBX to prevent the PBX from being compromised and fraudulent calls from being made by unauthorized (internal or external) users. If fraudulent calls are detected, Cablevision reserves the right to disable International Calling until the PBX is properly secured by the customer.

5 PBX Configuration

The steps on the next pages describe the minimum configuration required to enable the PBX to use a SIP Trunk for inbound and outbound calling. This guide provides the configuration and steps for both PBX registration and static (non-registration) modes of PBX operation. Please refer to the Toshiba IPedge product documentation for more information on other advanced PBX features.

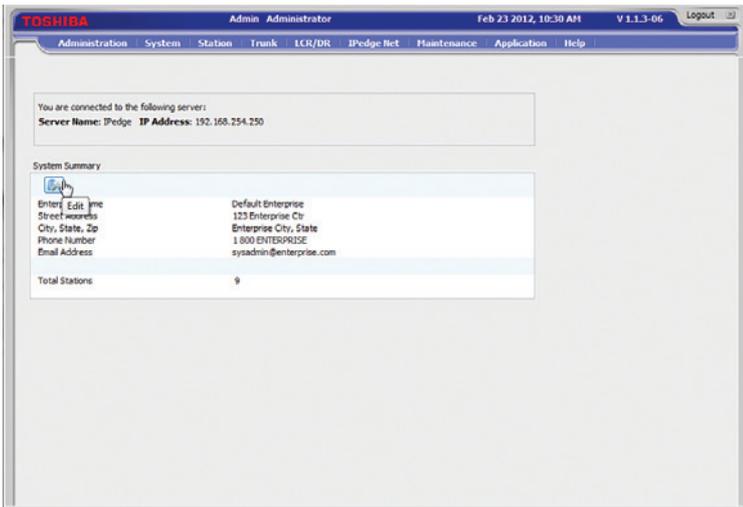
The configuration described here assumes that the PBX is already configured and operational with station side phones using assigned extensions or DIDs. This configuration is based on Toshiba IPedge version 1.1.3-06 and Toshiba IP5022-SD phones, which were used for the certificate testing.

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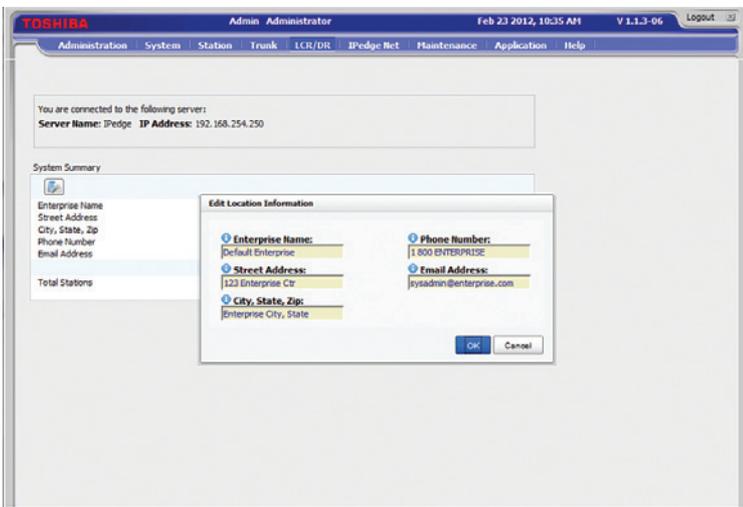
1. The Toshiba IPedge used in testing came with one Ethernet port, with the default LAN IP address set to 192.168.254.250/24. To use Optimum Business SIP Trunk service with the Optimum Business SIP Trunk Adaptor, the Ethernet port and the SIP Trunk Adaptor's LAN port 1 must be in the same LAN segment and have been assigned with IP addresses of 192.168.254.1/24 and 192.168.254.250/24 respectively. In the lab, the same LAN port is also connected to all the IP phones via a PoE switch.
2. The IP phones by default will need IP address assignment. The IPedge PBX does not have a DHCP server, the Optimum Business SIP Trunk Adaptor should have its DHCP server enabled. This is in step 2 of the Optimum Business SIP Trunk Set-Up Guide.
3. To configure the IPedge PBX, connect your PC to the PBX's LAN ports, launch the browser from your PC and enter "**http://192.168.254.250:8080/oamp**" in the address bar to access the Enterprise Manager program, a Web browser based Enterprise Manager application that resides on the IPedge server.
Note: Enterprise Manager can be accessed using Microsoft Internet Explorer version 7 or later, or Mozilla Fire Fox version 5 or later.



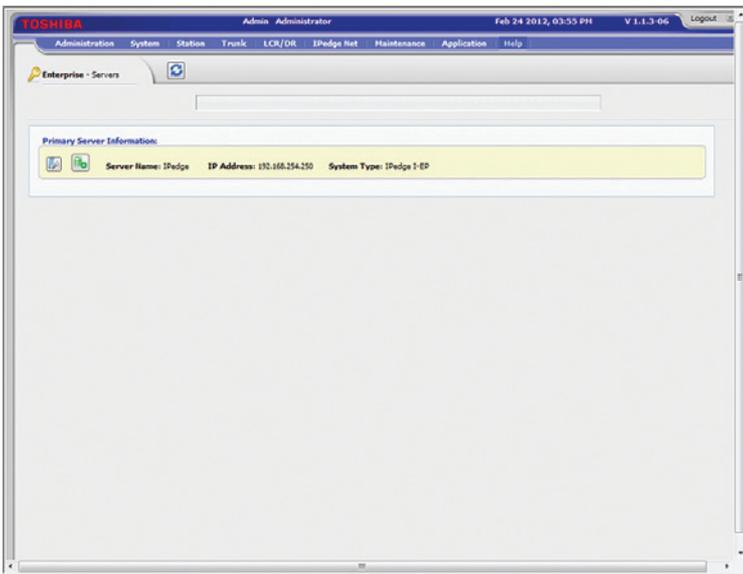
- a. From the login screen, enter "**Administrator**" in the "**Username**" field, enter "**password**" in the "**Password**" field, select the timeout value in the "**Session Timeout**" field, check the "**Remember me**" checkbox (if you want the Enterprise Manager application to automatically log you in on your next visit) and click the "**Submit**" button.



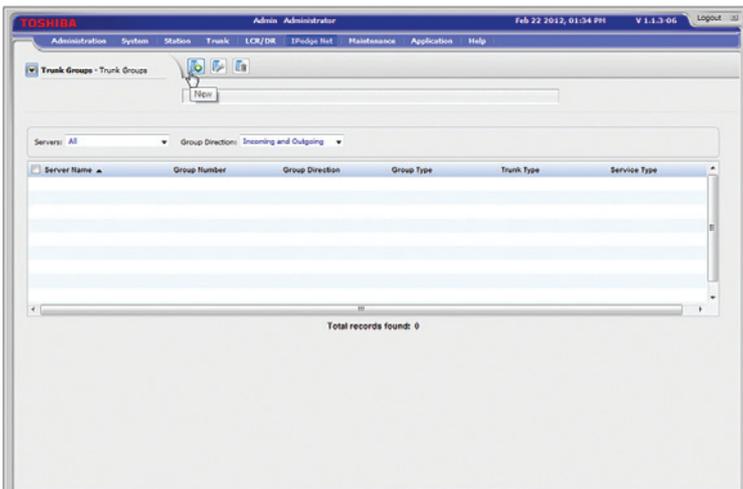
- b. The first screen after login is the System Summary. Click the “**Edit**” icon to enter information about your IPedge server.



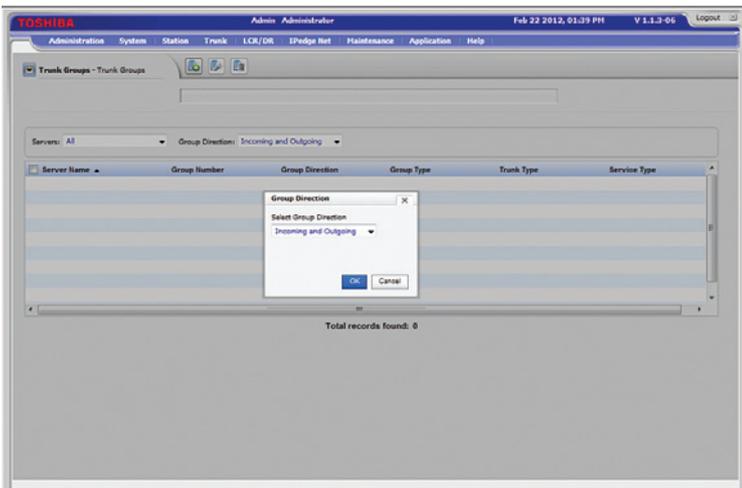
- c. Enter the enterprise name, address, phone number and e-mail address for the IPedge server. Click the “**OK**” button.



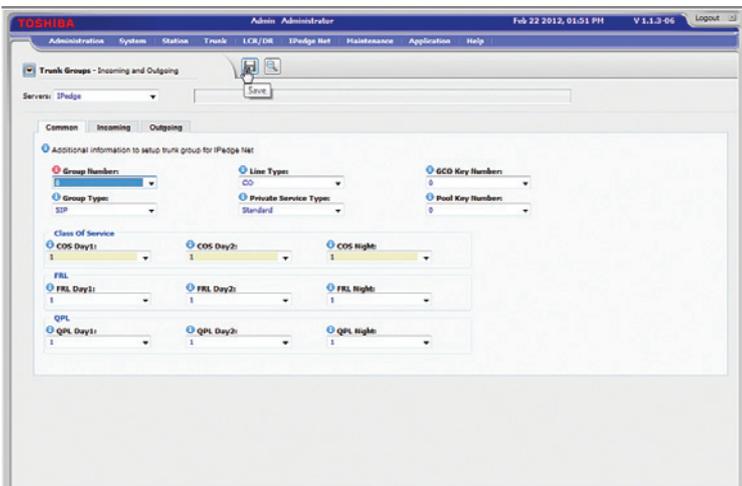
- d. The IPedge servers can have up to three configurations, depending on the hardware platform. Navigate to “**Administration**” > “**Enterprise**” > “**Servers**” to see that this is an IPedge EP server (with only one LAN port), designed to serve up to 40 users.



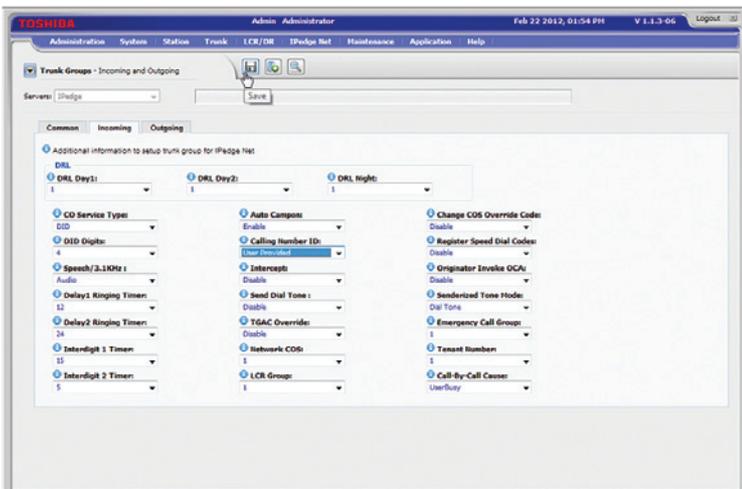
4. Navigate to “**Trunk**” > “**Trunk Groups**” and click on the “**New**” icon to create SIP Trunk Groups for incoming and outgoing lines.



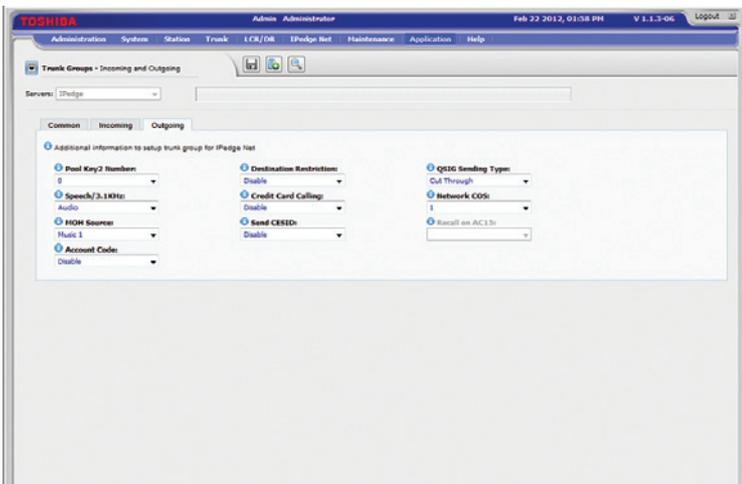
- a. Select **“Incoming and Outgoing”** from the pull-down menu of the **“Select Group Direction”** field. Click the **“OK”** button.



- b. In the **“Common”** tab, select a trunk group number (**“8”** in this example) in the **“Group Number”** field, select **“SIP”** in the **“Group Type”** field, leave other fields as default and click the **“Save”** icon.

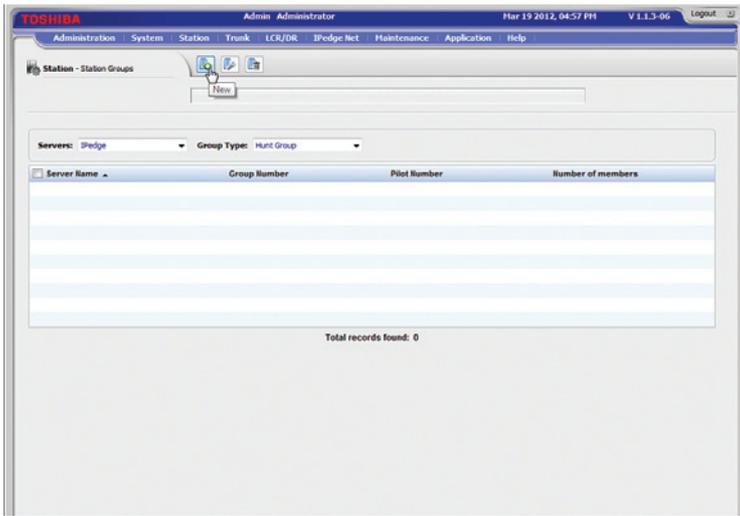


- c. In the **“Incoming”** tab, select **“DID”** in the **“CO Service Type”** field, select **“4”** in the **“DID Digits”** field; leave other fields as default and then click the **“Save”** icon.
Note: The PBX will route the incoming call to an extension based on the last 4 digits of the called number.

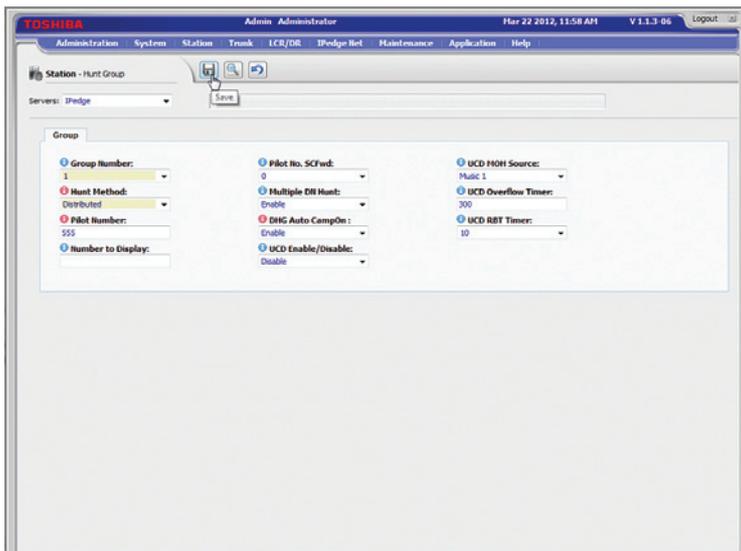


- d. In the **“Outgoing”** tab, leave all fields as default and click the **“Save”** icon.

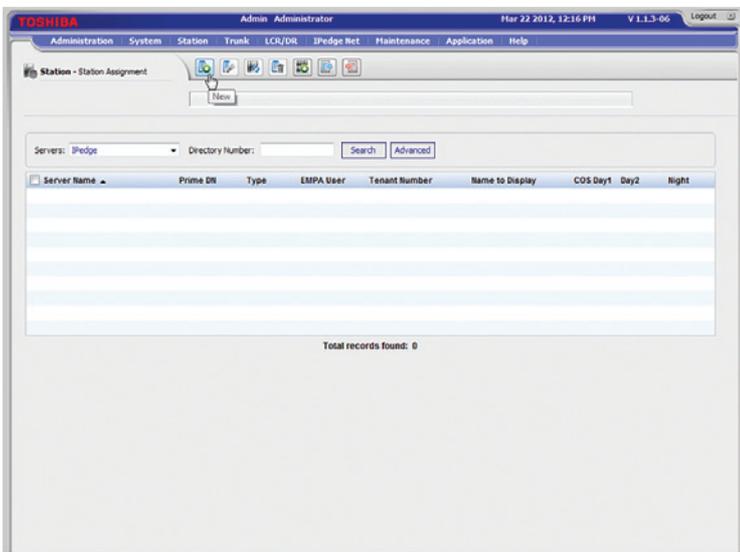
5. Navigate to **“Station” > “Station Group”** to create a hunt group for supporting voicemail and Auto-Attendant.



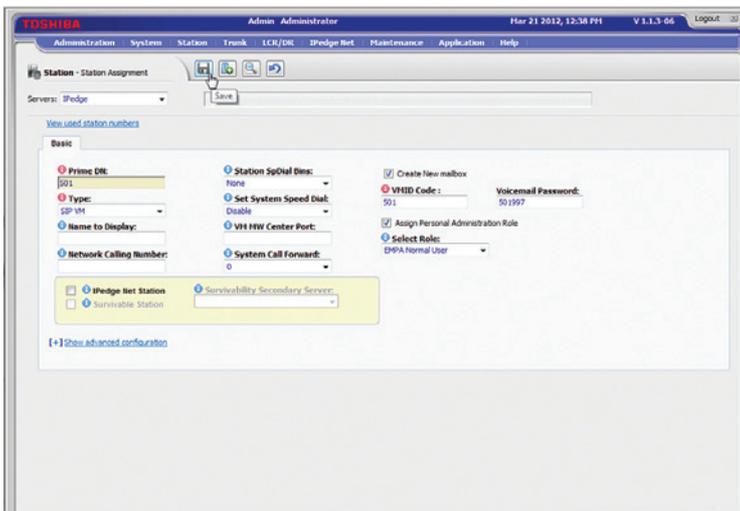
- a. Select **“Hunt Group”** from the **“Group Type”** field’s drop-down list and click on the **“New”** icon.



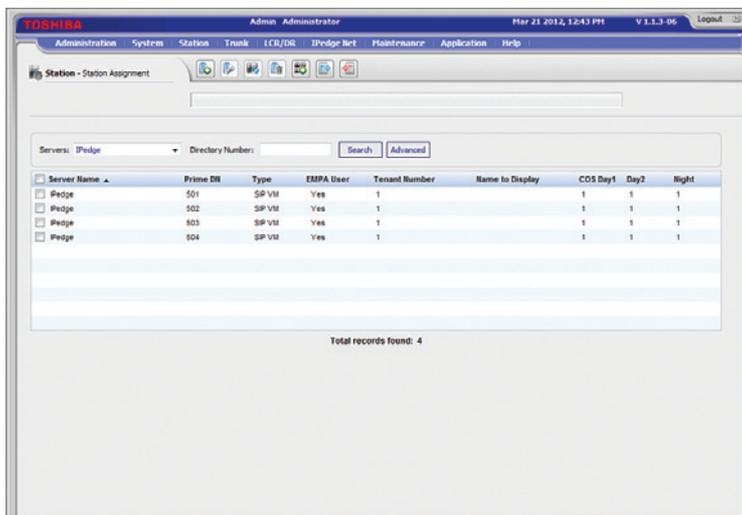
- b. Enter a 3-digit pilot number (“555” in this example) in the **“Pilot Number”** field, select **“Distributed”** from the drop-down list of the **“Hunt Method”** field, select **“Enable”** from the drop-down list of the **“DHG Auto CampOn”** field, leave other fields as default and then click the **“Save”** icon. **Note:** By default, the license will allow the hunt group to have a maximum of 4 ports for simultaneous voicemail access.



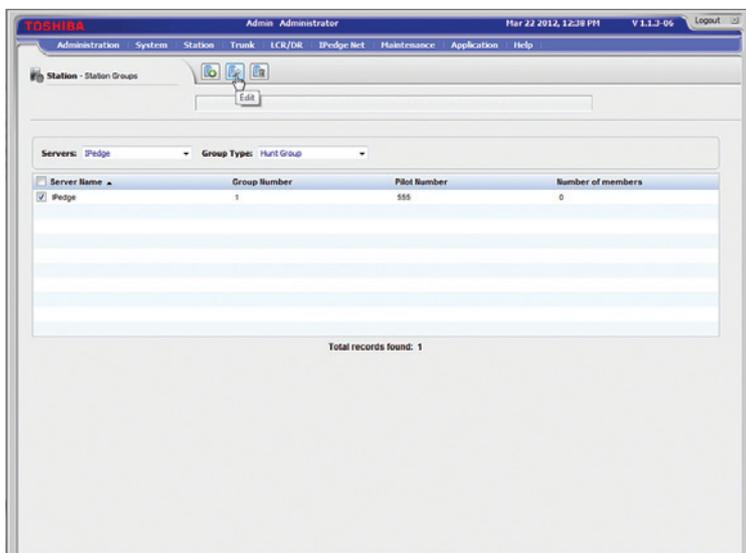
- c. Navigate to “**Station**” > “**Station Assignment**” to create four, 3-digit numbers of “SIP VM” device type for supporting voicemail access. Click the “**New**” icon.



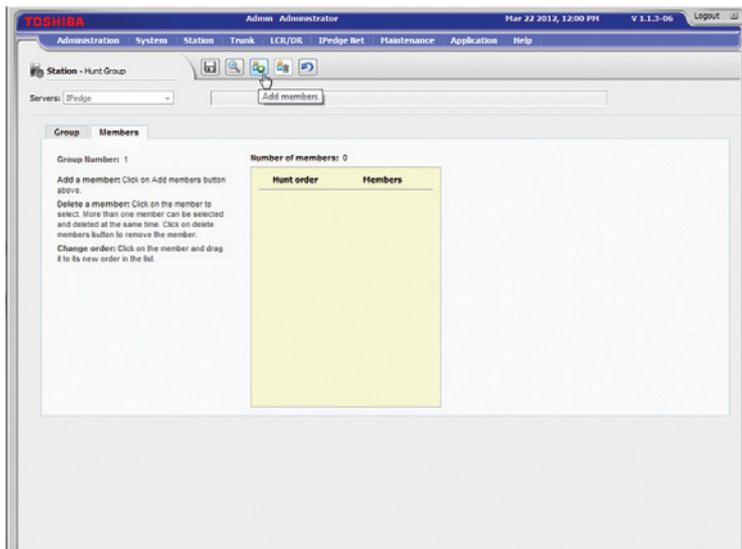
- d. Enter the first 3-digit number (“501” in this example) in the “**Prime DN**” field, select “**SIP VM**” from the drop-down list of the “**Type**” field, check the “**Create New mailbox**” checkbox, leave other fields as default and click the “**Save**” icon. Note that the VMID Code and voicemail password will be automatically generated by the PBX after you get past entering the Prime DN.



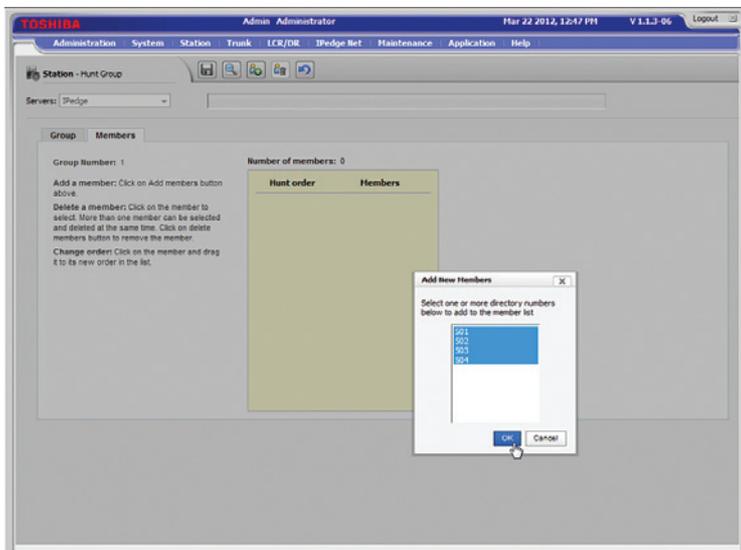
e. After you have set up the four, 3-digit numbers of “SIPVM” device type for supporting voicemail access, your Station Assignment screen should look like the the screen above.



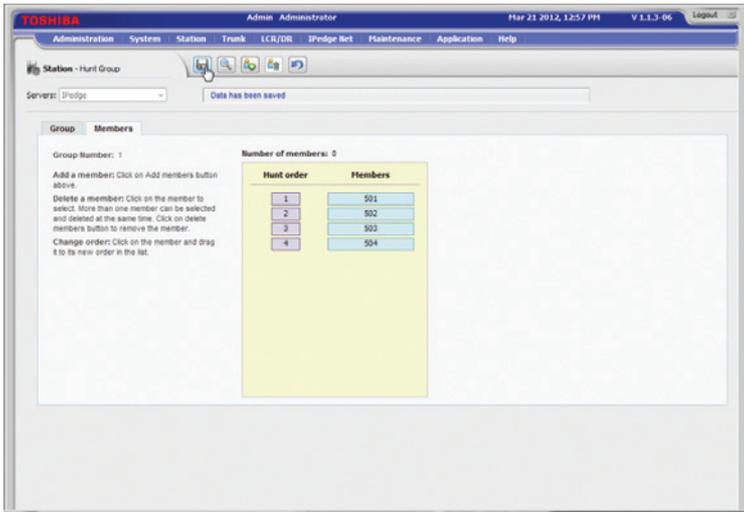
f. Navigate to “Station” > “Station Group” to add the four members to the hunt group for supporting voicemail. Check the checkbox for the **Hunt Group #1** entry and then click the “Edit” icon.



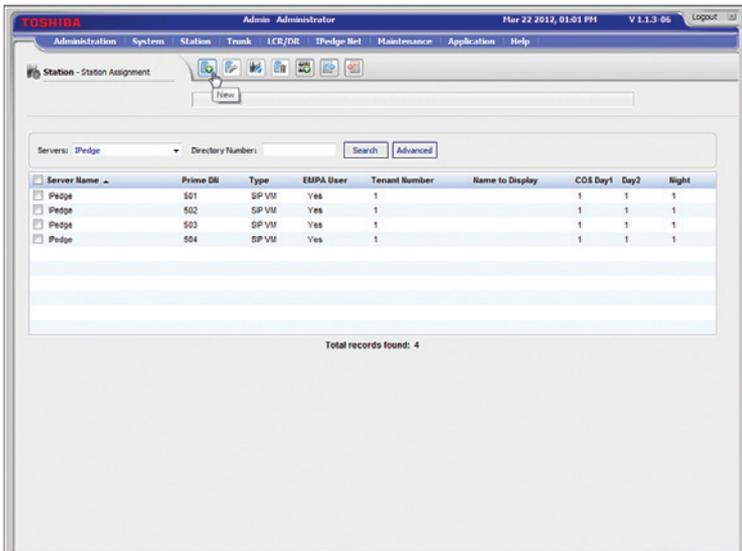
g. Select the **“Members”** tab and click the **“Add members”** icon.



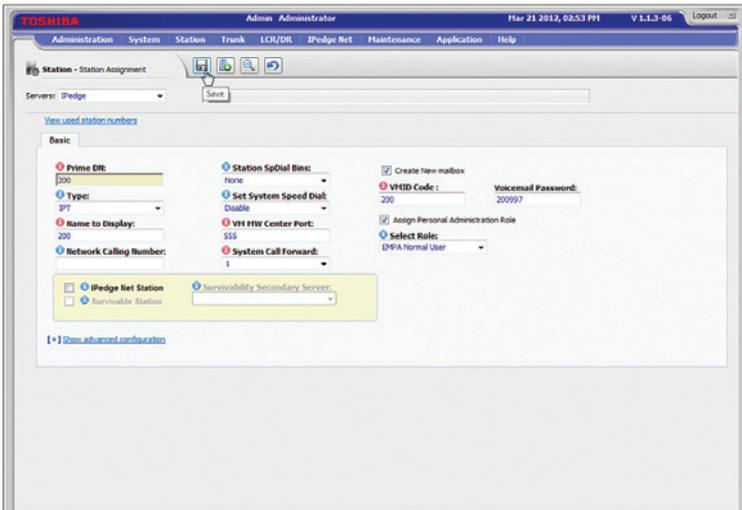
h. Select all four numbers (501 to 504 in this example) and click the **“OK”** button.



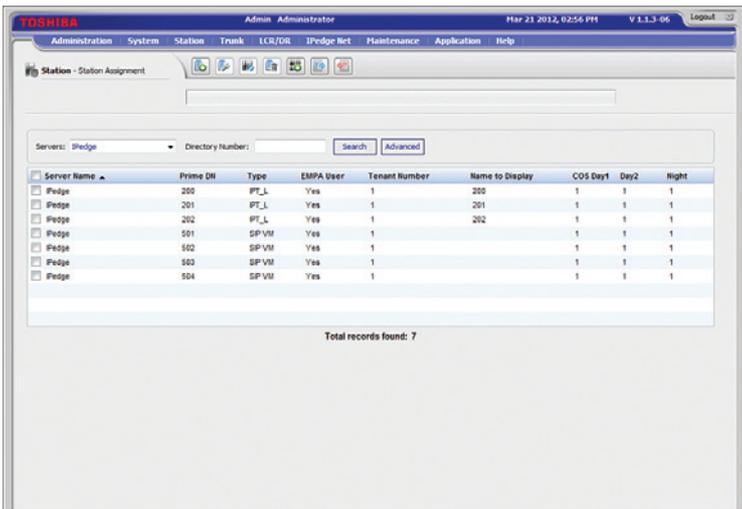
- i. Click the **“Save”** icon.
6. Navigate to **“Station” > “Station Assignment”** to create the 3-digit extensions for the phones.



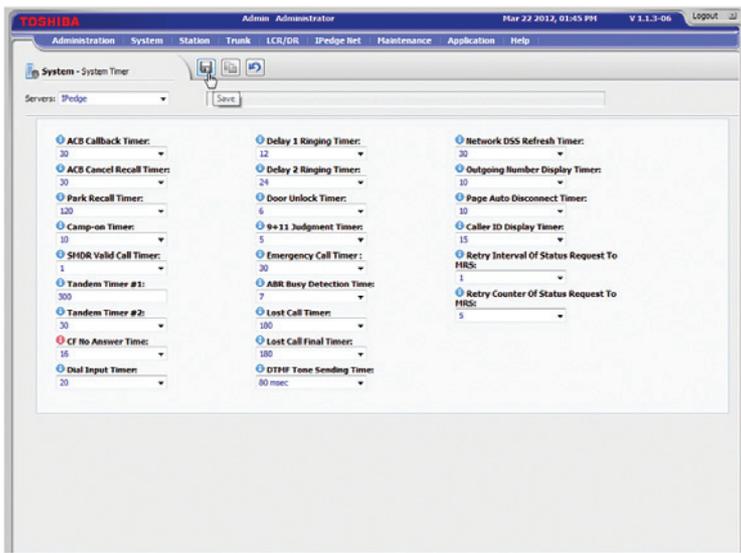
- a. Click the **“New”** icon.



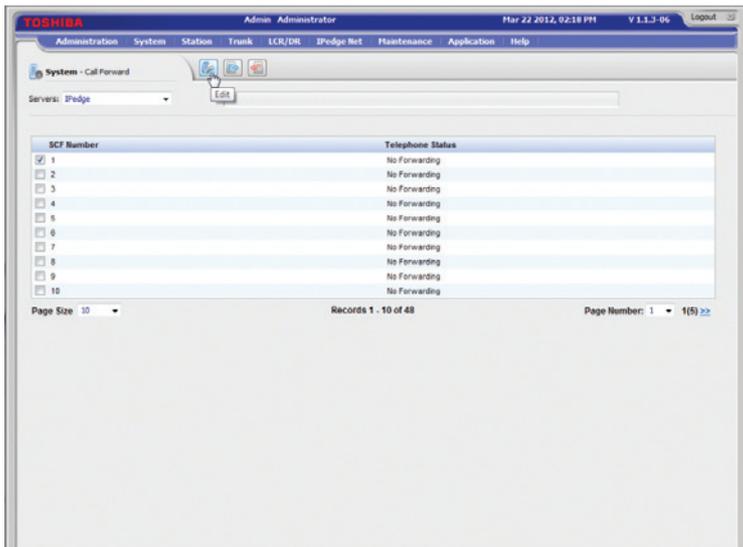
- b. Enter a 3-digit extension number (“200” in this example) in the “**Prime DN**” field, select “**IPT**” from the drop-down list of the “**Type**” field, enter the same extension in the “**Name to Display**” field, enter the Hunt Group #1’s pilot number in the “**VM MW Center Port**” field, select “**1**” (System Call Forward Number 1) in the “**System Call Forward**” field, check the “**Create New mailbox**” checkbox, leave other fields as default and click the “**Save**” icon. Note that the VMID Code and voicemail password will be automatically generated by the PBX after you get past entering the Prime DN. Remember to write down your voicemail password; you will need it to set up voicemail from the phone later.



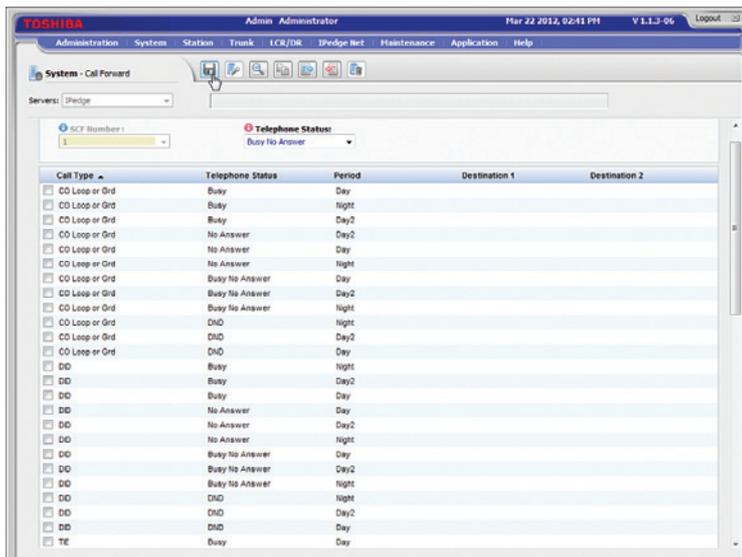
- c. After you have set up the 3-digit extension numbers for the phones (200 to 202 in this example), your Station Assignment screen should look like screen above.



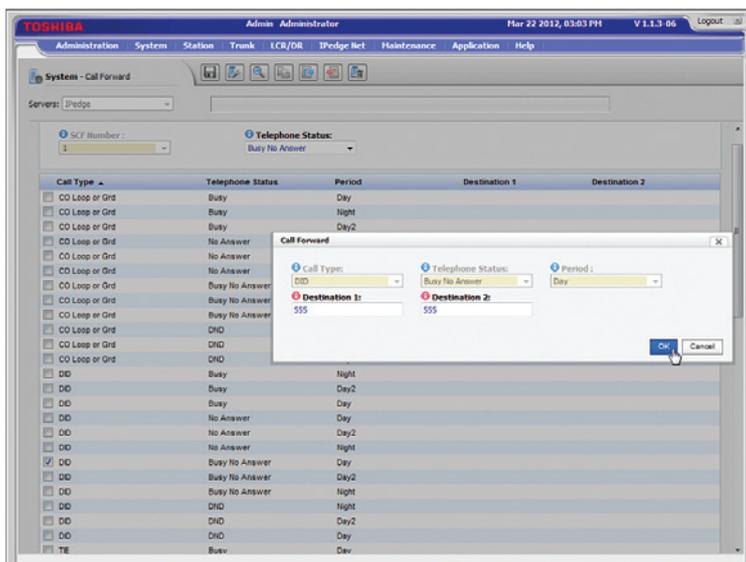
- d. Navigate to “**System**” > “**System Timer**” to configure the PBX to send the inbound call to voicemail after four rings. Select “**16**” (seconds) from the drop-down list of the “**CF No Answer Time**” field and click the “**Save**” icon.



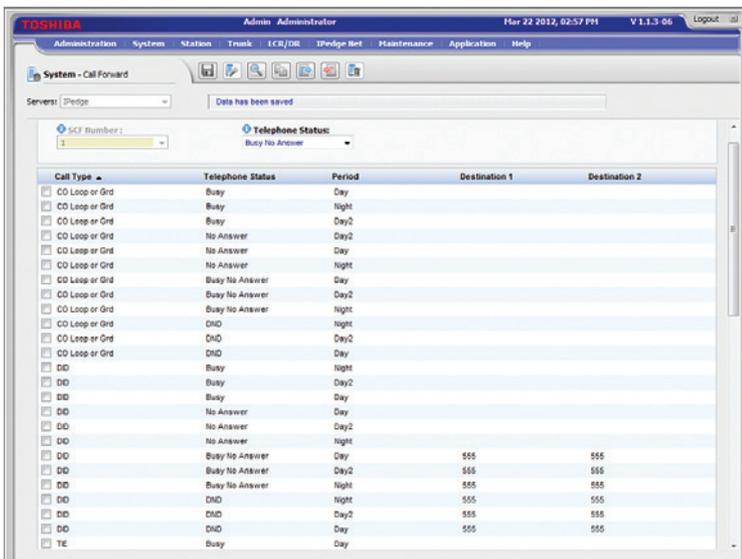
- e. Navigate to “**System**” > “**Call Forward**” to configure the conditions for the PBX to call-forward the inbound calls to voicemail. Check the **SCF Number 1** checkbox and click the “**Edit**” icon.



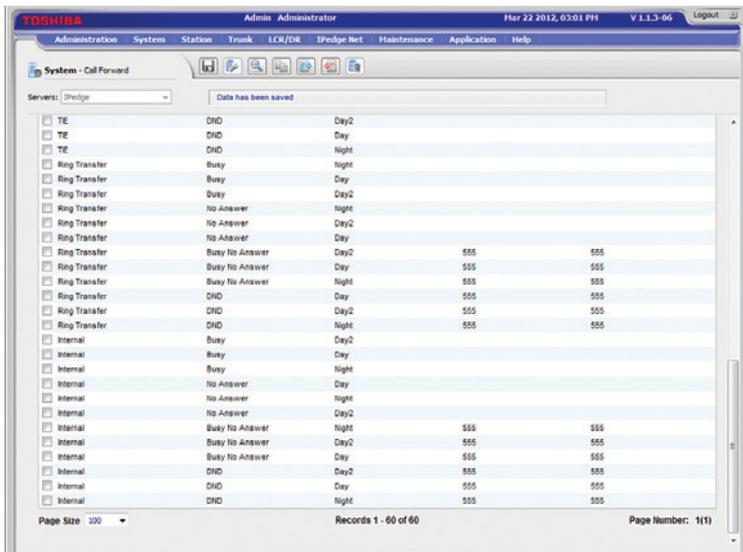
f. Change the name of the SCF Number 1 by selecting **“Busy No Answer”** from the drop-down list of the **“Telephone Status”** field and click the **“Save”** icon.



g. Check the checkbox for the entry where **Call Type=“DID”, Telephone Status=“Busy No Answer”** and **Period=“Day”** and click the **“Edit”** icon. From the pop-up box, enter Hunt Group #1’s pilot number in both the **“Destination 1”** and **“Destination 2”** fields and click the **“OK”** button.



h. When you are done with **Call Type="DID"**, the first half of the SCF Number 1 screen should look like the one above.



i. When you are done with **Call Type="Ring Transfer"** and **Call Type="Internal"**, the second half of the SCF Number 1 screen should look like the one above.

7. Navigate to **“Trunk”** > **“DID”** and then click on the **“New”** icon to map incoming calls to the extensions.

The screenshot shows the Toshiba Admin Administrator interface for configuring a new DID. The top navigation bar includes 'Administration', 'System', 'Station', 'Trunk', 'LCR/DR', 'IPedge Net', 'Maintenance', 'Application', and 'Help'. The main content area is titled 'Trunk - DID' and contains several configuration sections:

- ILG Group Numbers:** A pull-down menu set to '8'.
- DID Numbers:** A text input field containing '8080'.
- Tenant Number:** A pull-down menu set to '1'.
- MOH Source:** A pull-down menu set to 'Music 1'.
- GCO Key Group:** A pull-down menu set to '0'.
- Pooled Key Group:** A pull-down menu set to '0'.
- DID Audio:** A section with three rows of settings:
 - Audio Day1 Dst Type:** A pull-down menu set to 'Dialing Digits'.
 - Audio Day2 Dst Type:** A pull-down menu set to 'Dialing Digits'.
 - Audio Night Dst Type:** A pull-down menu set to 'Dialing Digits'.
 - Audio Day1 Dst Digits:** A text input field containing '200'.
 - Audio Day2 Dst Digits:** A text input field containing '200'.
 - Audio Night Dst Digits:** A text input field containing '200'.
- DID Data:** A section with three rows of settings:
 - Data Day1 Dst Type:** A pull-down menu set to 'No Data'.
 - Data Day2 Dst Type:** A pull-down menu set to 'No Data'.
 - Data Night Dst Type:** A pull-down menu set to 'No Data'.
 - Data Day1 Dst Digits:** An empty text input field.
 - Data Day2 Dst Digits:** An empty text input field.
 - Data Night Dst Digits:** An empty text input field.
- DID/DRIS No. VHID:** An empty text input field.
- DID/DRIS Name:** An empty text input field.
- VH Application Digits:** An empty text input field.

- a. Select the Incoming Line Group number from the pull-down menu of the **“ILG Group Number”** field.
- b. Enter the last 4 digits of the SIP Trunk DID (the pilot DID in this example) in the **“DID Number”** field.
- c. Select **“Dialing Digits”** from the pull-down menu of the **“Audio Day1 Dst Type”** field, **“Audio Day2 Dst Type”** field and the **“Audio Night Dst Type”** field.
- d. If the incoming call should ring an extension, enter the mapped extension in the **“Audio Day1 Dst Digits”** field, **“Audio Day2 Dst Digits”** field and the **“Audio Night Dst Digits”** field.

The screenshot shows the Toshiba Admin Administrator interface for configuring an existing DID. The top navigation bar includes 'Administration', 'System', 'Station', 'Trunk', 'LCR/DR', 'IPedge Net', 'Maintenance', 'Application', and 'Help'. The main content area is titled 'Trunk - DID' and contains several configuration sections:

- ILG Group Numbers:** A pull-down menu set to '8'.
- DID Numbers:** A text input field containing '8083'.
- Tenant Number:** A pull-down menu set to '1'.
- MOH Source:** A pull-down menu set to 'Music 1'.
- GCO Key Group:** A pull-down menu set to '0'.
- Pooled Key Group:** A pull-down menu set to '0'.
- DID Audio:** A section with three rows of settings:
 - Audio Day1 Dst Type:** A pull-down menu set to 'Dialing Digits'.
 - Audio Day2 Dst Type:** A pull-down menu set to 'Dialing Digits'.
 - Audio Night Dst Type:** A pull-down menu set to 'Dialing Digits'.
 - Audio Day1 Dst Digits:** A text input field containing '555'.
 - Audio Day2 Dst Digits:** A text input field containing '555'.
 - Audio Night Dst Digits:** A text input field containing '555'.
- DID Data:** A section with three rows of settings:
 - Data Day1 Dst Type:** A pull-down menu set to 'No Data'.
 - Data Day2 Dst Type:** A pull-down menu set to 'No Data'.
 - Data Night Dst Type:** A pull-down menu set to 'No Data'.
 - Data Day1 Dst Digits:** An empty text input field.
 - Data Day2 Dst Digits:** An empty text input field.
 - Data Night Dst Digits:** An empty text input field.
- DID/DRIS No. VHID:** An empty text input field.
- DID/DRIS Name:** An empty text input field.
- VH Application Digits:** An empty text input field.

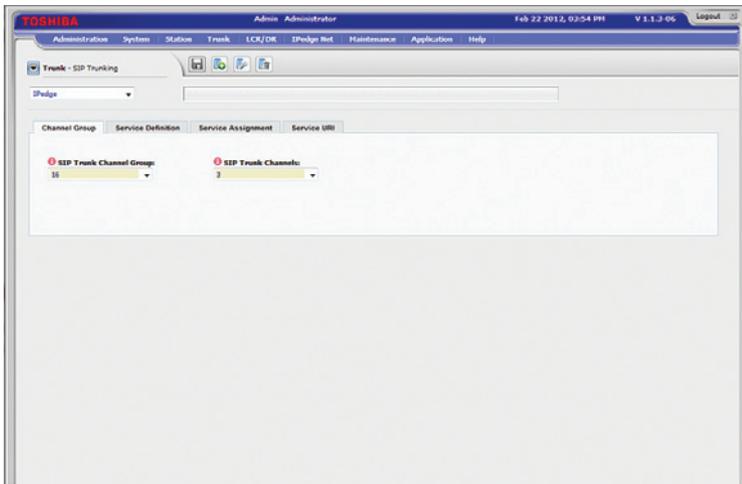
- e. If the incoming call should go to an Auto-Attendant, enter “555” in the “**Audio Day1 Dst Digits**” field, “**Audio Day2 Dst Digits**” field and the “**Audio Night Dst Digits**” field.

Important: The IPedge’s Auto-Attendant can only work with Out-of-Band DTMF tones (RFC-2833). The Cablevision network only supports in-band DTMF tones. In order for the Toshiba IPedge PBX to operate correctly with the Cablevision network, the Optimum Business SIP Trunk Adaptor must be configured to convert out-of-band DTMF tones sent by the Toshiba IPedge PBX to in-band DTMF tones and vice versa.

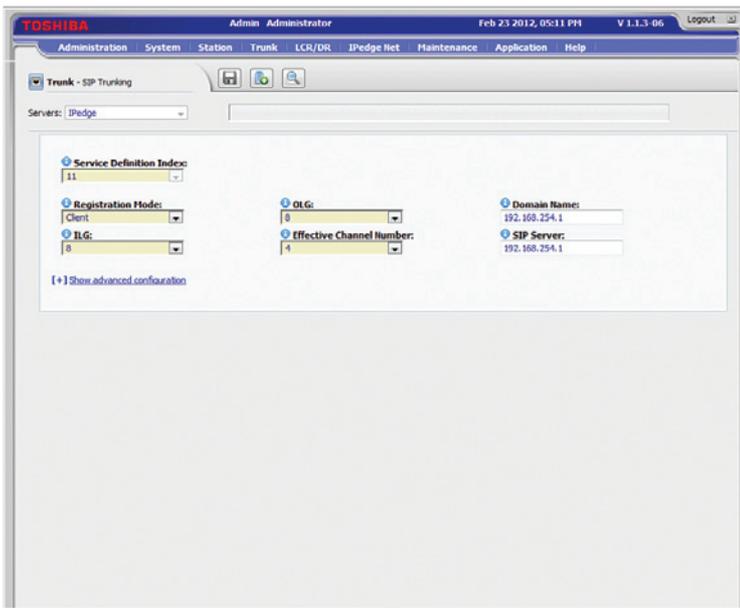
To enable this conversion, follow step 3 in the Optimum Business SIP Trunk Set-Up Guide. On the SIP Trunk Configuration page, you **must** check the **Convert Inband DTMF** checkbox, and click the Submit button to update the setting.

- f. Leave other fields as default and click the “**Save**” icon.
- g. Repeat the steps above until you have mapped all the DID’s to the extensions for inbound calls.

- 8. Navigate to “**Trunk**” > “**SIP Trunking**” to set up SIP Trunk services via the Optimum Business SIP Trunk Adaptor.

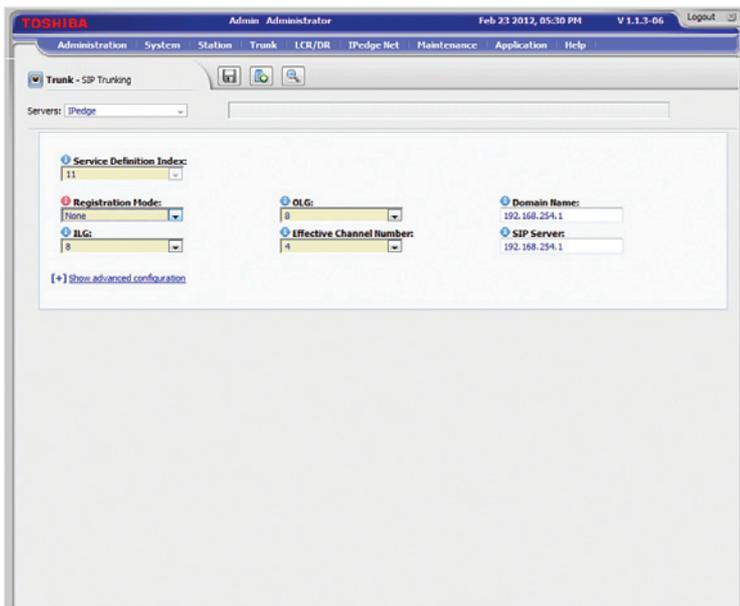


- a. Select the “**Channel**” tab to create a Channel Group for SIP Trunking service.
 - 1) Choose a Channel group number (16 in this example) that has not been used in another section from the pull-down menu of the “**SIP Trunk Channel Groups**” field. Note that the phones will display this channel number when you dial 9 first to make an outside call.
 - 2) Enter “**3**” (because we are using 3 phones for the SIP trunk service) in the “**SIP Trunk Channels**” field.

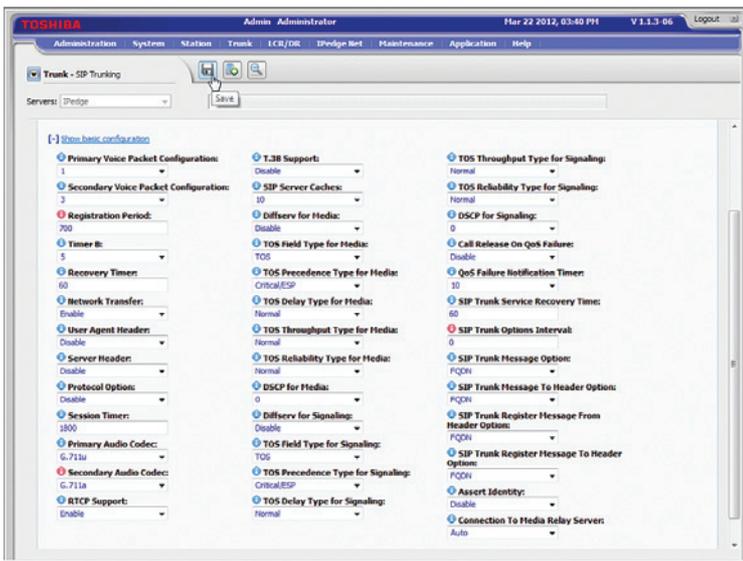


- b. Select the the **“Service Definition”** tab and then click the **“New”** icon.
 - 1) Choose a SIP Trunk Service Definition Index from the pull-down menu of the **“Service Definition Index”** field.
 - 2) You can configure the PBX to communicate with the Optimum Business SIP Trunk Adaptor in Registration mode or Static IP mode.
 - 3) To configure the PBX for Registration mode, select **“Client”** in the **“Registration Mode”** field.

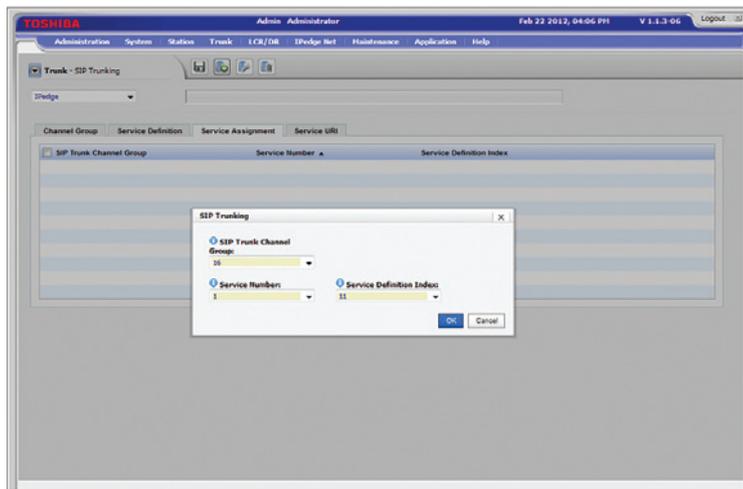
Note: when an outbound call is received by the Optimum Business SIP Trunk Adaptor, it will check to see if the outbound call’s caller ID matches any of the of the SIP Trunk DID’s. If it doesn’t, then the Optimum Business SIP Trunk Adaptor will replace the outbound call’s caller ID with the pilot DID.



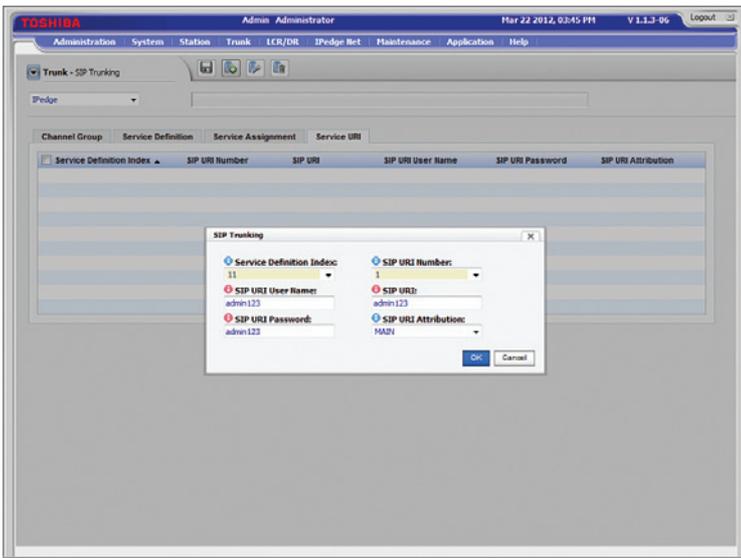
- 4) To configure the PBX for Static IP mode, select **“None”** in the **“Registration Mode”** field. Note that the PBX outbound call may use any of the SIP Trunk DID’s as the caller ID. When a PBX outbound call is received by the Optimum Business SIP Trunk Adaptor, it will check to see if the outbound call’s caller ID matches any of the SIP Trunk DIDs. If it doesn’t, then it will replace the outbound call’s caller ID with the pilot DID.
- 5) Select the Incoming Line Group number in the **“ILG”** field.
- 6) Select the Outgoing Line Group number in the **“OLG”** field.
- 7) Enter the maximum number of calls allowed for the SIP Trunk service in the **“Effective Channel Number”** field.
- 8) Enter the Optimum Business SIP Trunk Adaptor’s LAN IP address in both the **“Domain Name”** and **“SIP Server”** fields. To work with the Optimum Business SIP Trunk service, the LAN port should be on the same subnet as the Optimum Business SIP Trunk Adaptor’s port 1 and use the Optimum Business SIP Trunk Adaptor’s port 1 IP address as it’s sip server or gateway. This is the IP address configured in step 2 of the Optimum Business SIP Trunk Set-Up Guide.
- 9) Click the **“Save”** icon.



- 10) Click the “[+] Show advanced configuration” link and:
 - a. Enter “700” in the “Registration Period” field so that PBX will register more frequent in Registration mode.
 - b. Select “G.711a” in the “Secondary Audio Codecs” field.
 - c. Enter “0” in the “SIP Trunk Options Interval” field so that the PBX will not send Option messages.
 - d. Click the “Save” icon.



- c. Select the the “Service Assignment” tab and then click the “New” icon.
 - 1) Select the SIP Trunk Channel Group number in the “SIP Trunk Channel Groups” field.
 - 2) Select the “Service Definition Index” in the “Service Definition Index” field.
 - 3) Select the “1” in the “Service Number” field.
 - 4) Click the “OK” button.

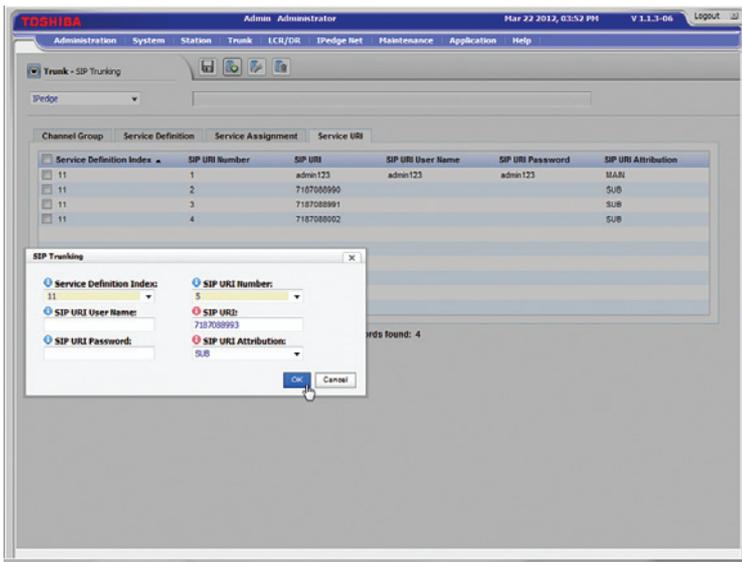


d. Select the the **“Service URI”** tab and then click the **“New”** icon to create the SIP URI (this can be the pilot DID or an alphanumeric string (ie: “admin123”)) for registering with the Optimum Business SIP Trunk Adaptor.

- 1) Select the “Service Definition Index” in the **“Service Definition Index”** field.
- 2) Enter the SIP URI string (“admin123” in this example) in the **“SIP URI”** field and Select “MAIN” in the **“SIP URI Attribution”** field.
- 3) Enter the Authentication username in the **“SIP URI User Name”** field.
- 4) Enter the Authentication password in the **“SIP URI Password”** field.

Note: The PBX will use this SIP URI string for registration only. The Optimum Business SIP Trunk Adaptor must be configured with the same User Name and Password when using Registration mode. This is step 3 in the Optimum Business SIP Trunk Set-Up Guide.

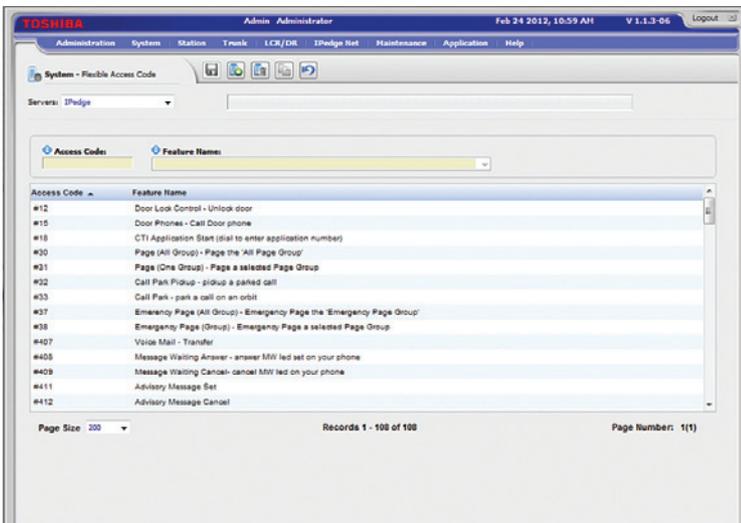
5) Leave other fields as default and then click the **“OK”** button.

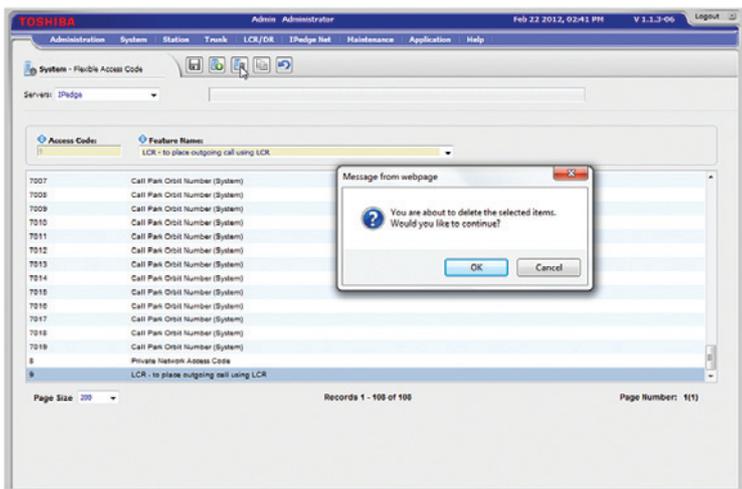


e. Continue to select the the “**Service URI**” tab and click the “**New**” icon to create SIP URI for the SIP Trunk DIDs.

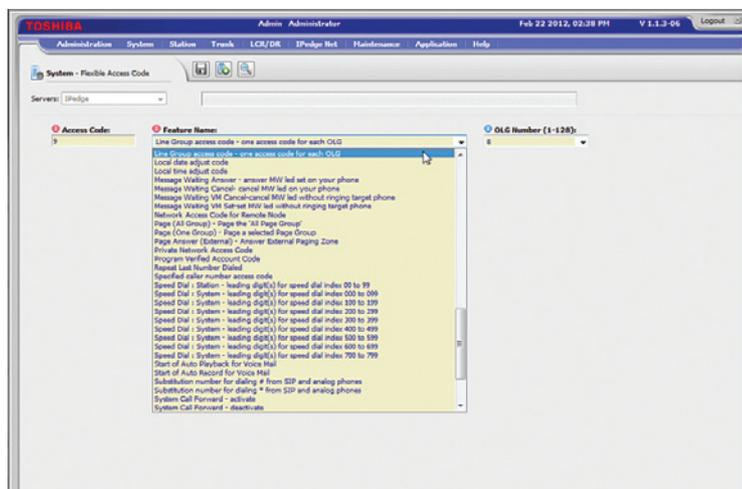
- 1) Select the “Service Definition Index” in the “Service Definition Index” field.
- 2) Enter the SIP Trunk DID in the “SIP URI” field and select “SUB” in the “SIP URI Attribution” field.
- 3) Leave other fields as default and click the “OK” button.

9. Navigate to “**System**” > “**Flexible Access Code**” to configure “9” as the Line Group access code.



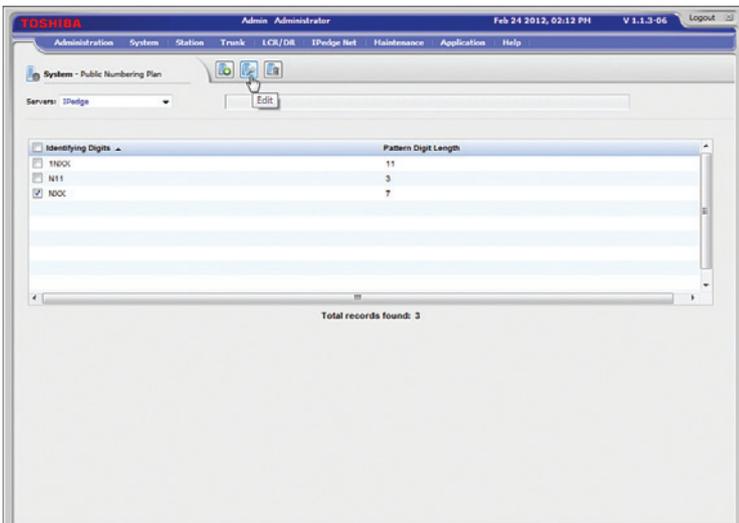


a. If Access Code 9 is not already configured as Line Group access code, delete it. Select the entry for Access code 9, click the **“Delete”** icon and then click the **“OK”** button from the pop-up box.

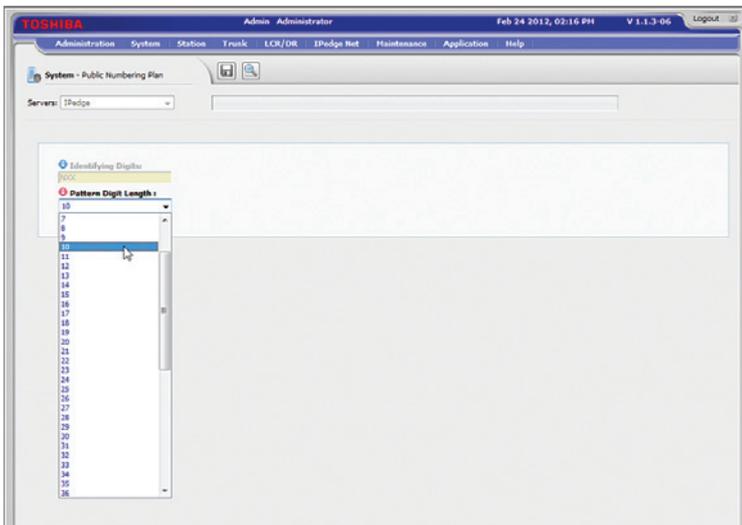


b. To configure Access Code 9 as Line Group access code, click the **“New”** icon, enter **“9”** in the **“Access Code”** field, select **“Line Group access code – one access code for each OLG”** from the pull-down list of the **“Feature Name”** field and click the **“Save”** icon.

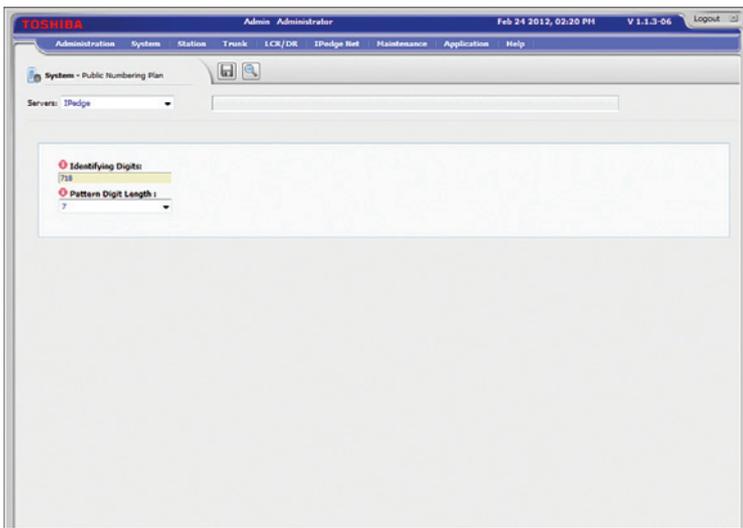
10. Navigate to **“System” > “Public Numbering Plan”** to configure the PBX to allow dialing of **“9”** followed by a 10-digit number. Note that the Identifying Digits of **“NXX”** should be set to **“10”** in the **“Pattern Digit Length”** field.



a. To configure the PBX to allow dialing of “9” followed by a 10-digit number, check the checkbox for the default **Identifying Digits for “NXX”** and click the **“Edit”** icon.

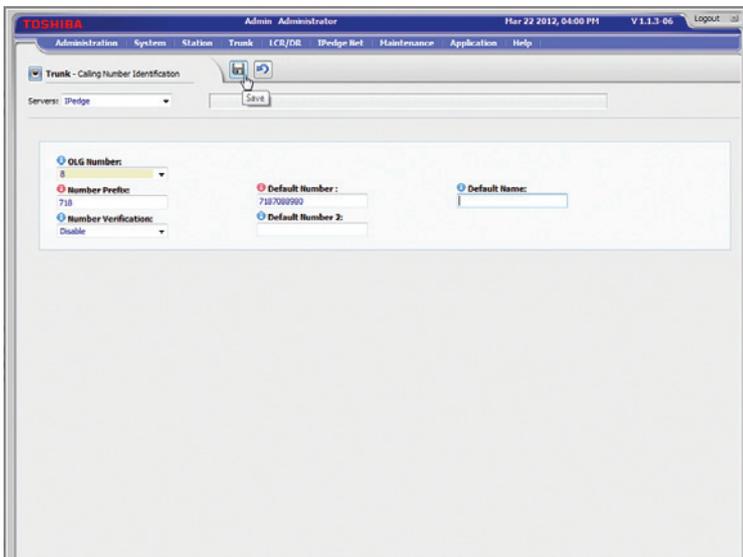


b. Select **“10”** (change from “7” to “10”) from the pull-down list of the **“Pattern Digit Length”** field and click the **“Save”** icon.

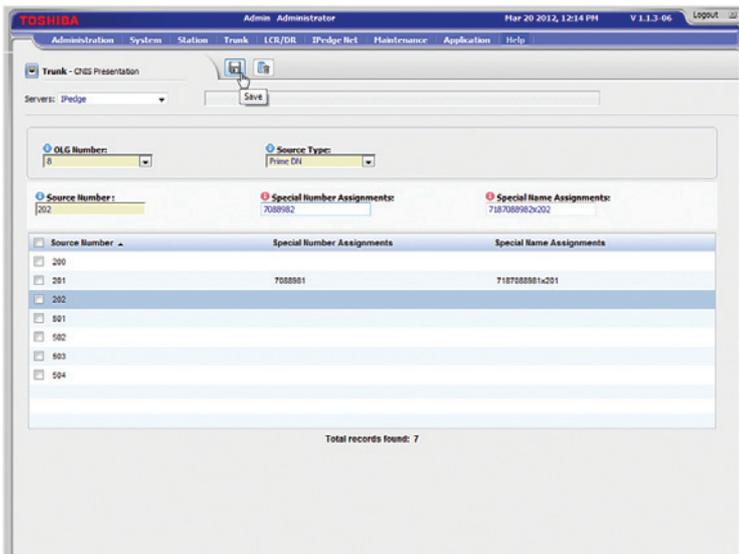


c. To configure the PBX to allow dialing of “9” followed by a 7-digit number, click the “**New**” icon, enter “718” (area code of the SIP Trunk DIDs in this example) in the “**Identifying Digits**” field, select “7” from the “**Pattern Digit Length**” field and click the “**Save**” icon.

11. Navigate to “**Trunk**” > “**Calling Number ->**” > “**Calling Number Identification**” to configure the default Caller ID and the prefix for non-default Caller ID.

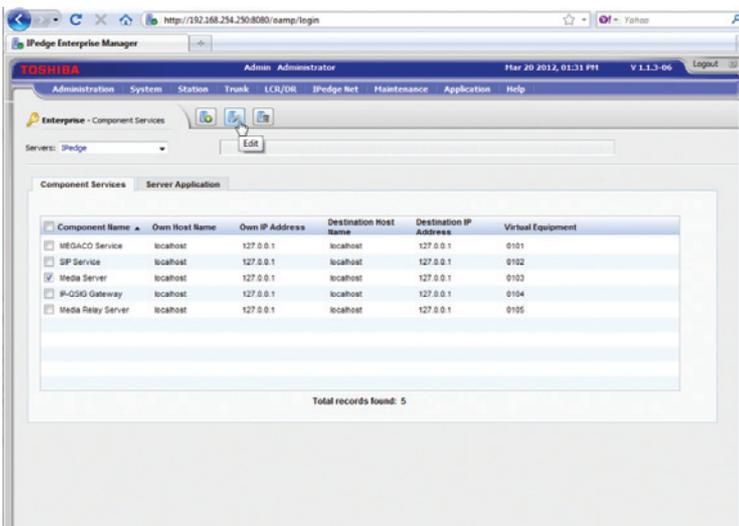


a. Enter the area code of the SIP Trunk DIDs in the “**Number Prefix**” field, enter the pilot DID in the “**Default Number**” field (the PBX will use pilot DID as Caller ID if non-default Caller ID is not configured for the extensions), leave other fields as default and click the “**Save**” icon.

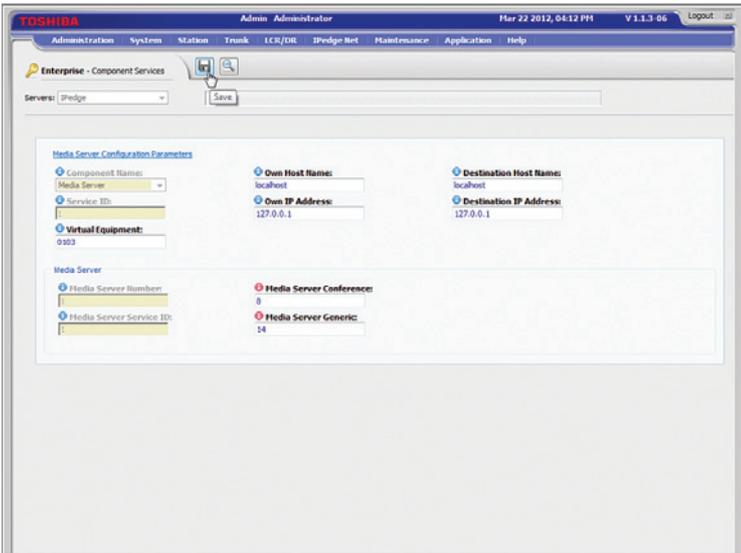


b. Navigate to **“Trunk” > “Calling Number ->” > “CNIS Presentation”** to configure the non-default Caller ID for the extensions. Select the **phone extension entry**, enter the last 7 digits of the SIP Trunk DID you want to use as Caller ID for this extension in the **“Special Number Assignments”** field, enter a name in the **“Special Name Assignments”** field (as of this writing, Toshiba has confirmed that this field is not supported) and click the **“Save”** icon. Note that, when the extension’s **“Special Number Assignments”** field is specified, the PBX will add the area code (configured in **“Trunk” > “Calling Number ->” > “Calling Number Identification”**) to this number and use it as Caller ID for the outbound calls placed from this extension.

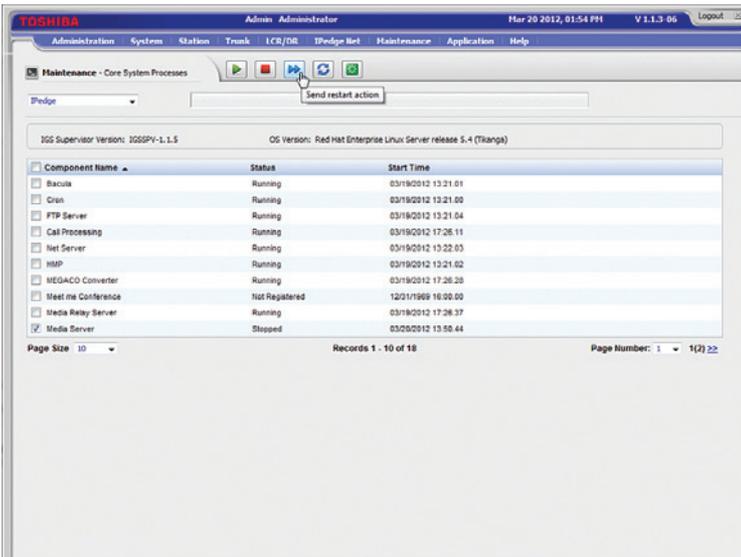
12. Navigate to **“Administration” > “Enterprise” > “Component Services”** to configure the Media Server for supporting conference calls.



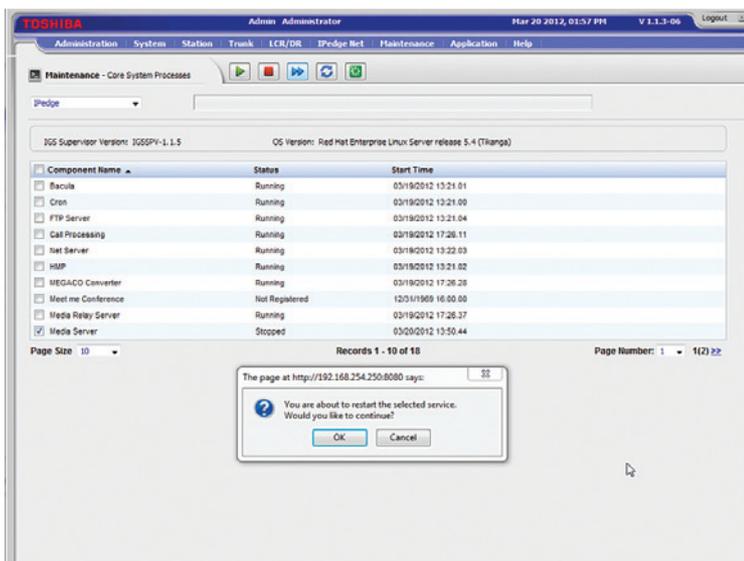
a. Check the **“Media Server”** checkbox and click the **“Edit”** icon.



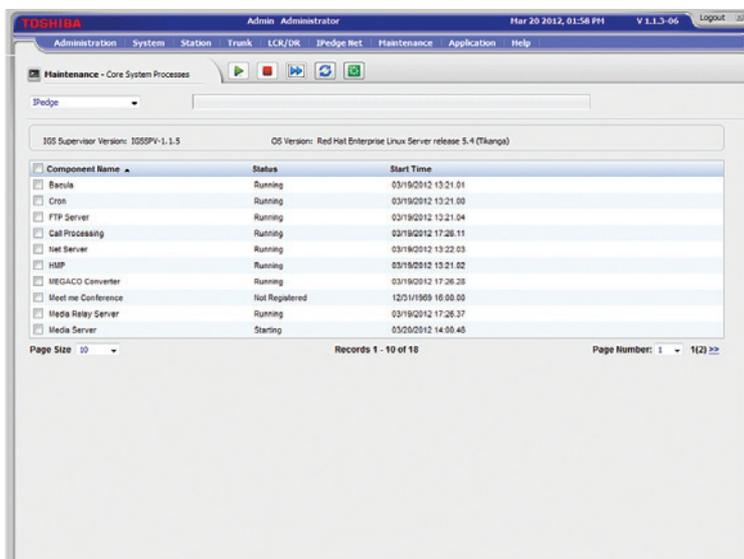
b. Enter **“8”** in the **“Media Server Conference”** field, enter **“14”** in the **“Media Server Generic”** field, leave other fields as default and click the **“Save”** icon.



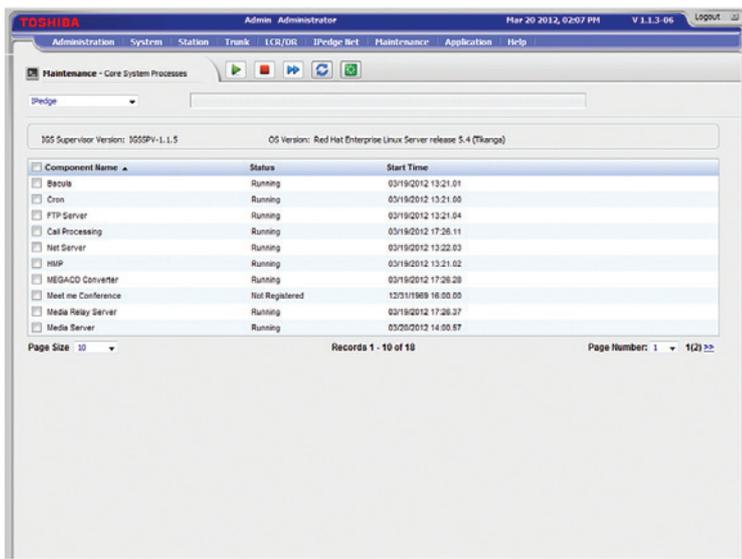
c. Navigate to **“Maintenance”** > **“System Maintenance”** > **“Core System Processes”**, check the **“Media Server”** checkbox and click the **“Send restart action”** icon to start the Media Server process.



d. Click the “OK” button.



e. You may need to restart the PBX to get the Media Server to the running state.



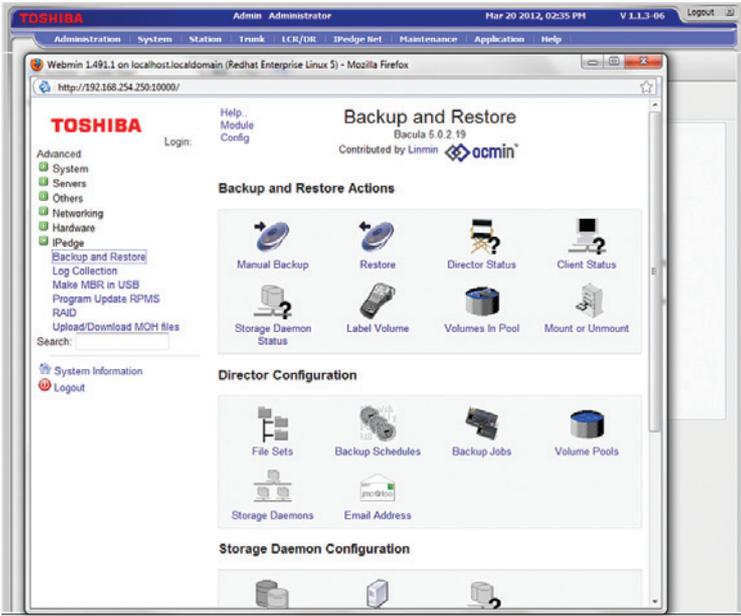
f. The Media Server is running. Now the PBX will allow the LAN phone to support 3-way conference calls with two WAN phones.

13. Configure the LAN phones by connecting the LAN phones to the PBX.

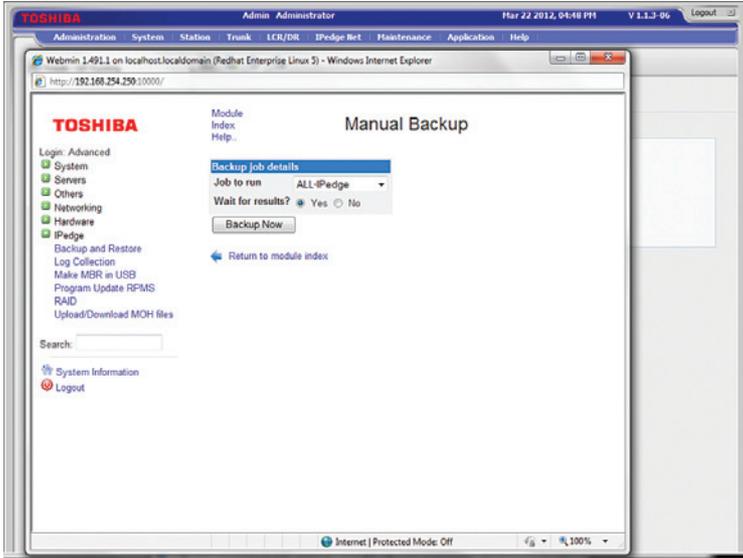
Note: The Toshiba IP5022-SD phones by default will need IP address assignments. Since the Toshiba IPedge is not equipped with a DHCP server, you need to enable the DHCP server option available in the Optimum Business SIP Trunk Adaptor. This is step 2 of the Optimum Business SIP Trunk Set-Up Guide.

- a. The Toshiba IP5022-SD phones communicate with the PBX using the Megaco VoIP protocol and it will pick up any new extension from the PBX as its Station ID if the phone has not been used before. Once the phone has been used, it will insist on using the same Station ID.
- b. To change the IP5022-SD phone's Station ID to match the PBX's extension:
 - 1) Press 3, 6, 9 and the red Hold key at the same time.
 - 2) Press 2 and then press the red Hold key (Feature Button 1-10 on the right side of the phone should be lit in solid red).
 - 3) Press the 6th Feature Button (it should change to flashing red) and enter the new Station ID to match the PBX's extension.
 - 4) Press the red Hold key (The 6th Feature Button should change back to solid green).
 - 5) Press the red Hold key again, lift the handset off the hook for several seconds, then hang up. The phone will restart with the changed Station ID.
- c. Once the phones are up and running, enter the Hunt Group #1's pilot number and voicemail password to set up voicemail.

14. Navigate to “**Application**” > “**Webmin** ->”, click the “**IPedge**” tab and click the “**Backup and Restore**” link to access the Backup and Restore features.



a. To manually back up the configuration, click the “**Manual Backup**” icon.

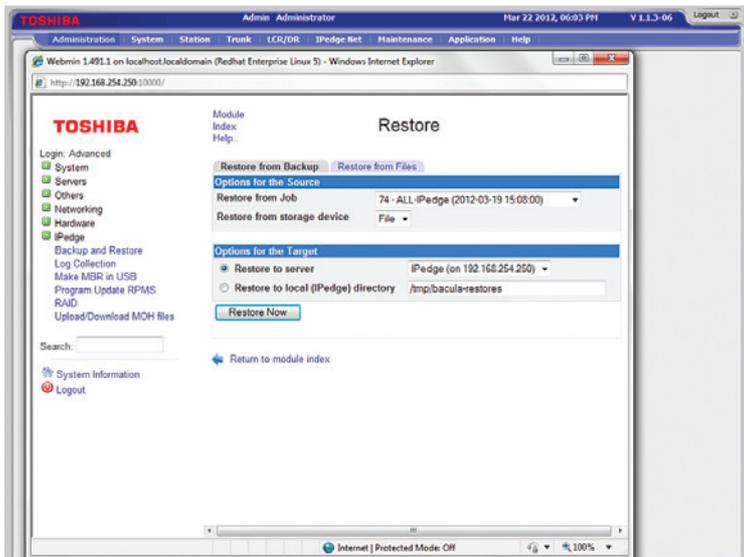


1) Click the “**Backup Now**” button.

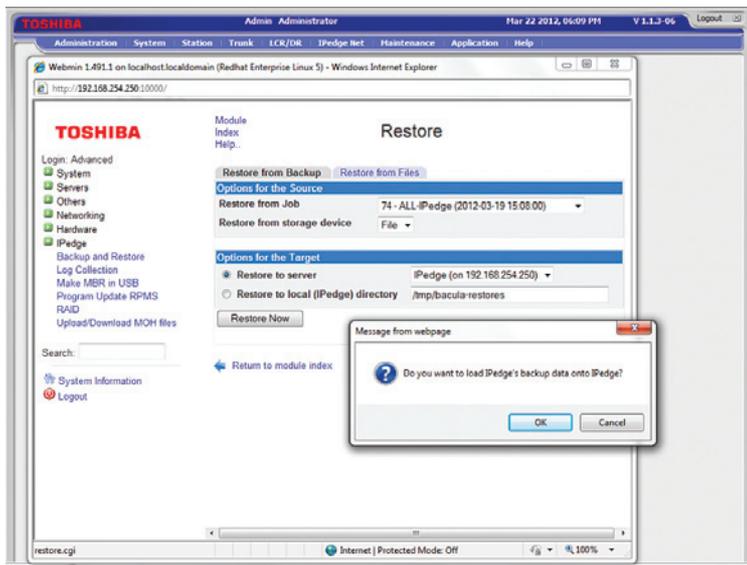


2) Waiting for backup to finish.

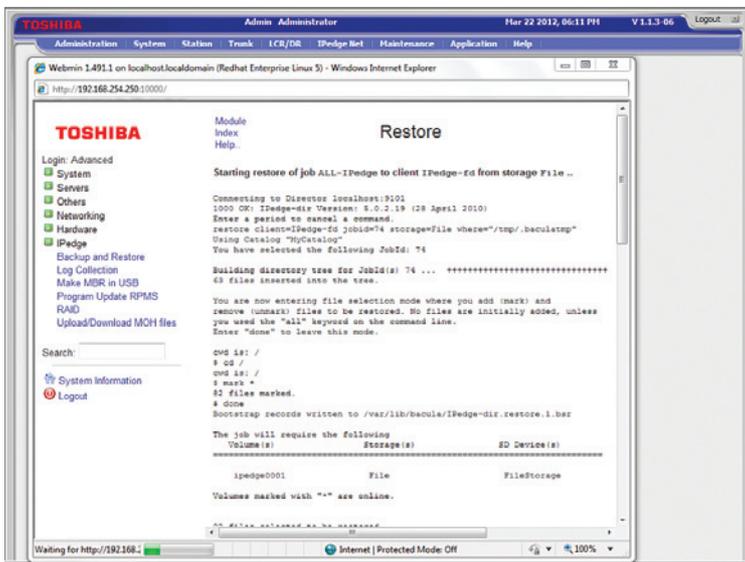
b. To restore from previous manual backup to the server, click the “**Restore**” icon.



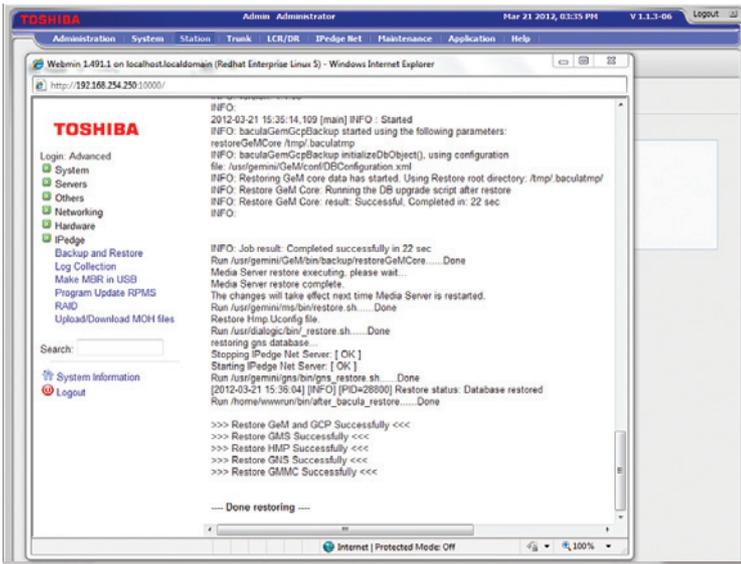
1) From the “**Restore from Backup**” tab, select the backup file to restore from the “**Restore from Job**” field, leave other fields as default and click the “**Restore Now**” button.



2) Click the “OK” button.



3) Wait for Restore to complete.



4) The server is now done with restoring the previous backup.