SoLink-Lite IP-PBX

Administrator Guide

(Version 1.0)

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1.0 INTRODUCTION

SoLink-Lite IP-PBX Appliance is a low-cost, embedded IP-PBX solution, supporting up to 8 FXO/FXS ports. This makes it ideal for SOHO, SMB (small and medium business), or branch office uses.

SoLink-Lite IP-PBX appliance provides the following functions:

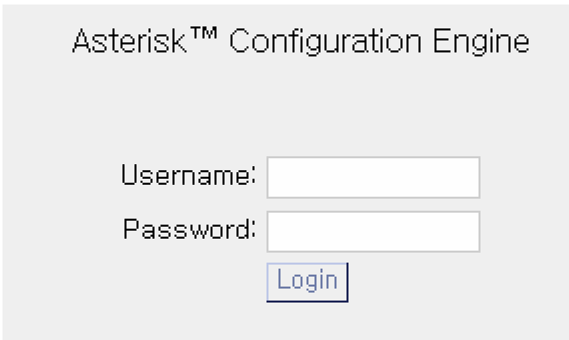
- automated attendant
- operator support
- extension with follow-me support
- call group (or extension group) with group mailbox
- voicemail with email message notification
- fax to email support (for fax receive only)
- call transfer
- call group pickup
- call parking
- music on hold
- conference bridge
- ACD (automatic call distribution)
- directory access
- outbound calling via PSTN or VoIP
- trunk failover support
- multiple user call privileges
- simple IVR (interactive voice response)
- peer IP-PBX support
- multiple language support (English, Cantonese, Putonghua, French, Spanish; only one language may be active at a time)
- call details records (CDR)
- web-based configuration and administration interface
- firmware upgradeable via web interface

The following is a list of all SoLink-Lite IP-PBX appliance series products:

Model	RJ45	Max TEL Ports	RS232 Port	Others	Size (mm)
IP-01	1	1 x FXO/FXS	Yes	-	100 x 100 x 28
IP-02	2	2 x FXO/FXS	No	-	100 x 100 x 28
IP-04	1	4 x FXO/FXS	No	MMC	225 x 120 x 30
IP-08	2	8 x FXO/FXS	Yes	MMC,USB	225 x 120 x 30

2.0 SETTING UP THE IP-PBX

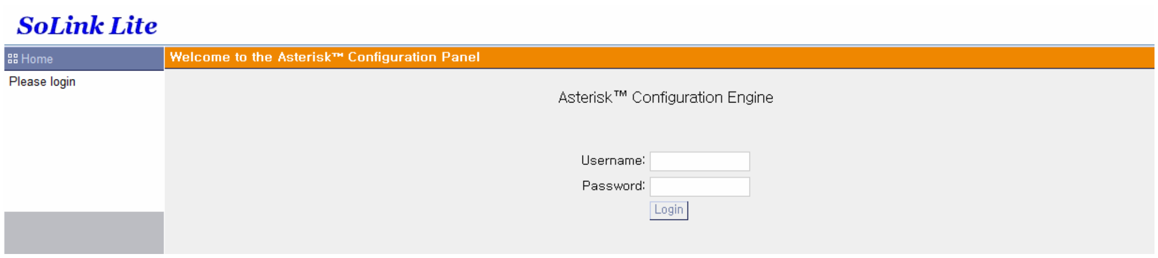
Connect the IP-PBX's WAN port to a network switch. Connect a PC to the same switch. Make sure that the IP address of your PC matches that of the IP address of the IP-PBX (default is 192.168.1.100). Open a web browser and enter the IP address of the unit, the login screen should appear. Default username is admin and password is solink.



3.0 LOGGING ON

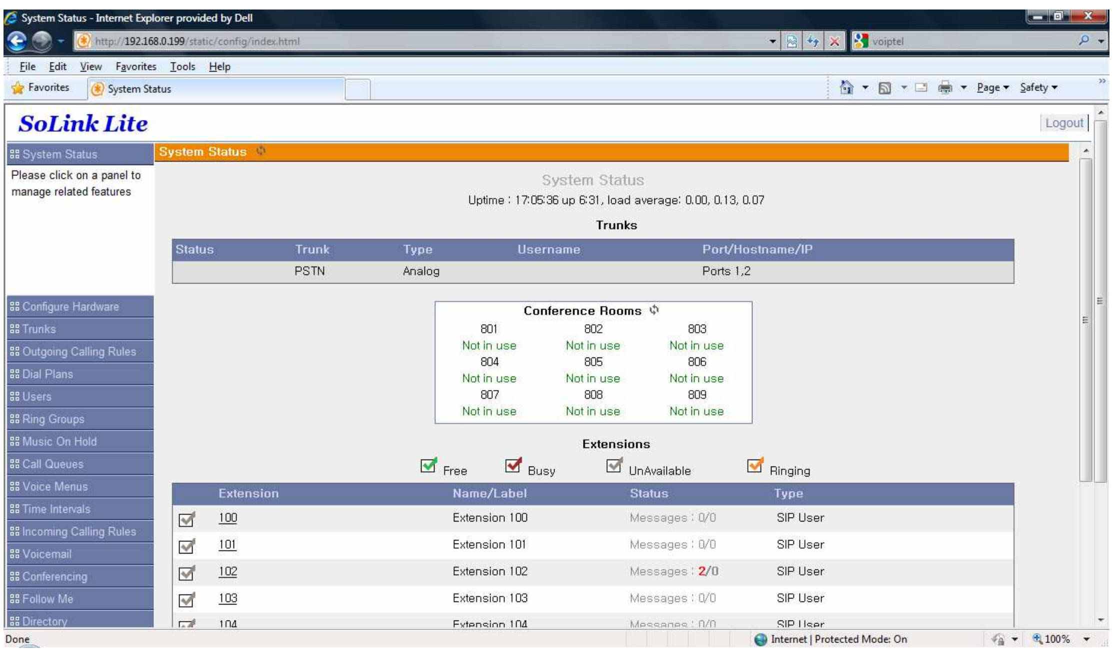
To access the web-based administration interface, simply enter the **Logon ID** and **Password**, and then click the **Login** button. Default IP address is 192.168.1.100 and default username/password is admin/solink.

For first-time logon, the user will be prompted to change the password.



Upon successful logon, the **System Status** screen will appear on the screen. It provides an overview of the system status.

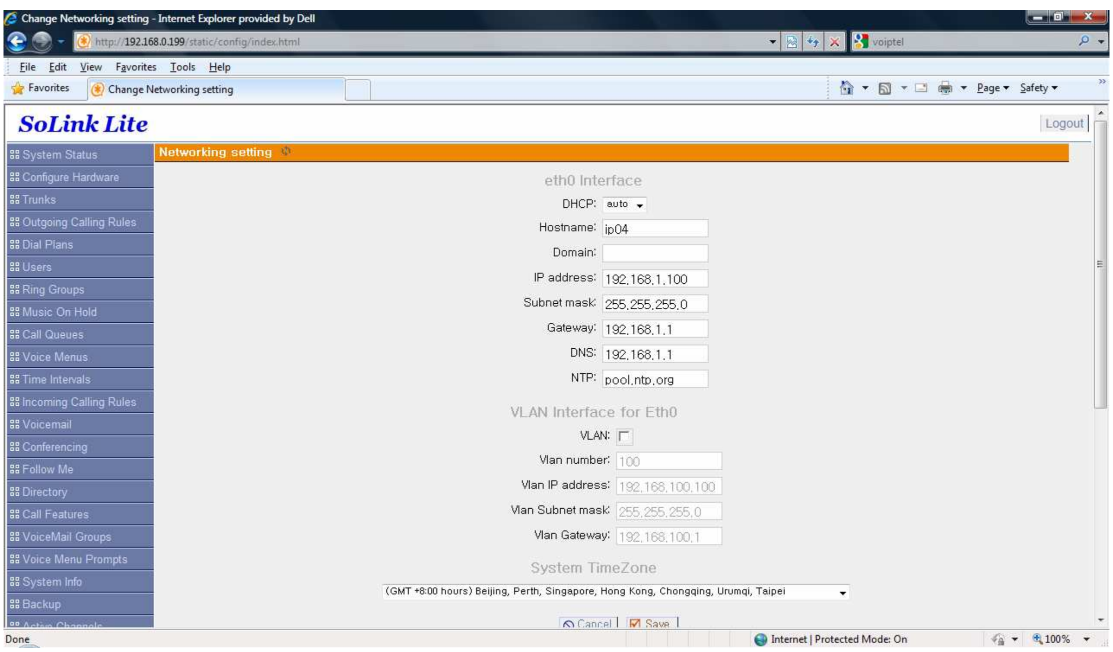
The unit has been pre-configured with default settings and will be ready for use with little or no changes. Please refer to Appendix A for details of the default settings.



4.0 CONFIGURING NETWORK

To configure the network settings, perform the follows steps:

1. Select the *Options* menu.
2. Click on the *Advanced Options* Tab.
3. Click on the *Show Advance Options* button. Now the advanced menus will be shown below the *Options* menu.
4. Select the *Network Settings* menu.
5. Modify the network settings as necessary and click the *Save* button to update the network configuration. Note that there are three types of DHCP Settings:
   a) Yes – the unit will obtain a dynamic IP address from your router
   b) auto – the unit will use the static IP address specified and ping the default gateway. If there is no response from the default gateway, the unit will switch to DHCP mode.
   c) No – static IP address as specified will be used.
6. Now go back to the *Options* menu and select the *Reboot* tab. Click on the *Reboot Now* button to reboot the unit. Note that a power reset is required in certain cases (such as changing from static IP to DHCP mode).
7. You should now be able to access the web admin interface using the revised IP address.



The unit should be ready for use. If you are using the default settings (as described in Appendix A), you may skip the rest of this manual.

5.0 CONFIGURING HARDWARE

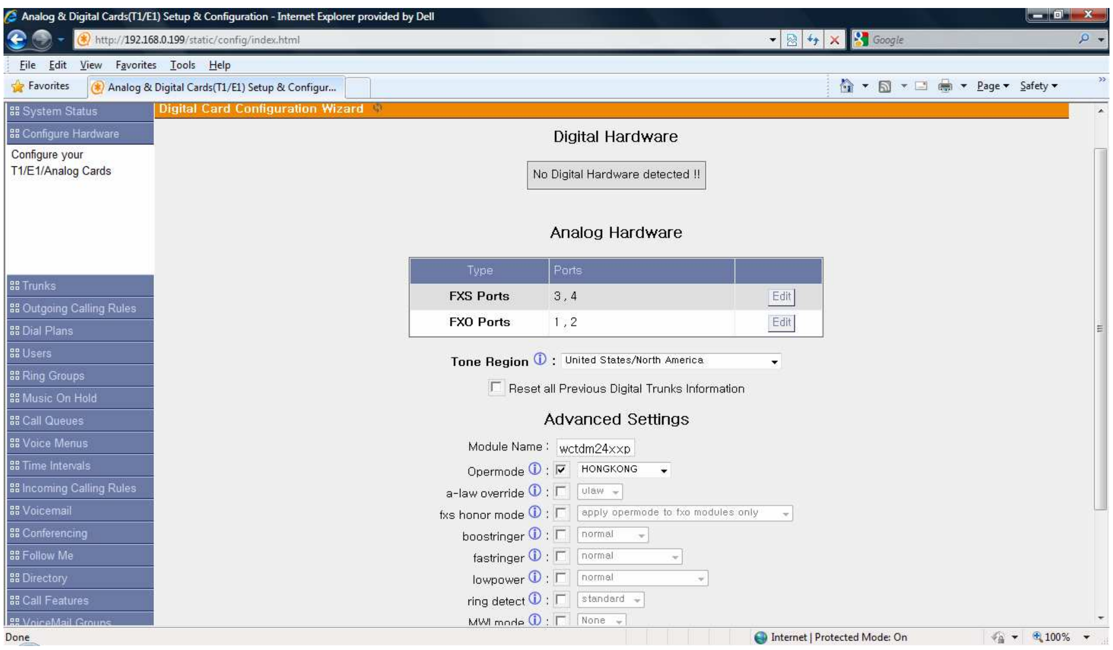
The *Configure Hardware* screen lists all available telephony ports in the unit and allows the user to configure the telephony interface to comply with your local telephony environment.

There are two types of analog telephony interfaces:

- FXO Port
  FXO ports should be connected to the telephone network (i.e. PSTN)
- FXS Port
  FXS Ports should be connected to an analog telephone set

For a detailed description of a field, simply hover on the ⓘ symbol besides the desired field.

To update the changes, simply click on the *Update Settings* button. To cancel, simply click on the *Cancel Changes* button.

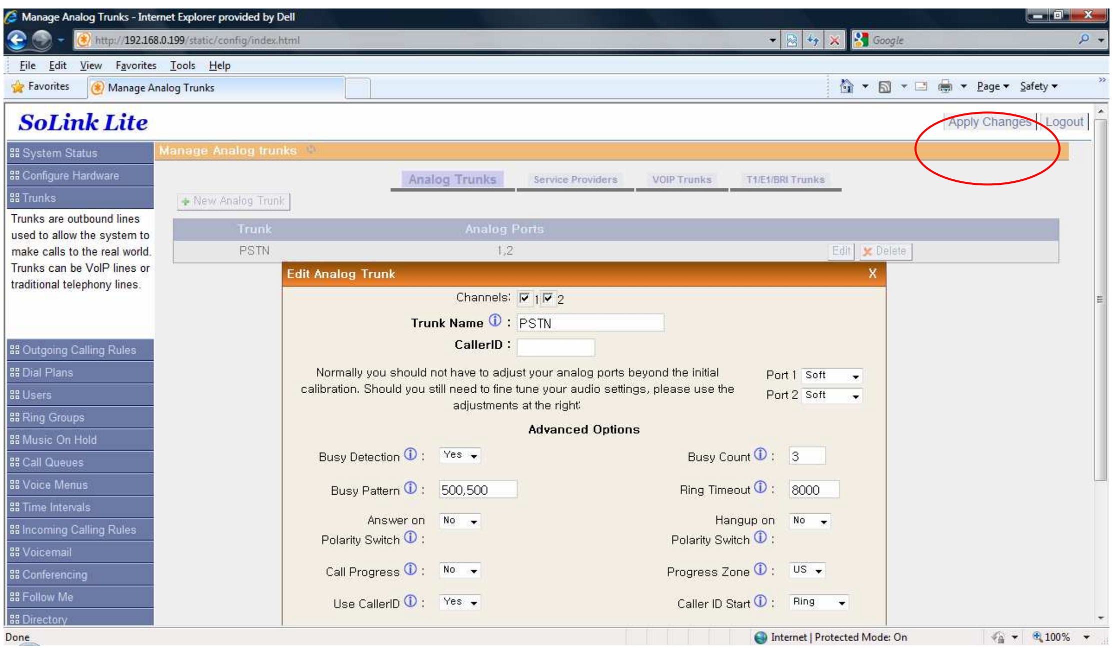


6.0 CONFIGURING TRUNKS

Trunks are outbound lines used to allow the system to make calls to the real world. Trunks can be VoIP lines or traditional telephony lines.

To configure a trunk, simply select the *Trunks* menu, and then select the type of trunk desired. Three types of trunks can be defined:

- Analog Trunks
  These are the analog FXO ports installed in the unit.
- Service Providers
  These are pre-defined VoIP service providers that allows the user to set up an account with these providers easily. Currently, only IAXtel and MTU Nett are defined. If you prefer to use another VoIP provider, set it up using the VOIP Trunks instead.
- VOIP Trunks
  User-defined SIP or IAX VOIP providers.
- T1/E1/BRI Trunks
  Not supported.

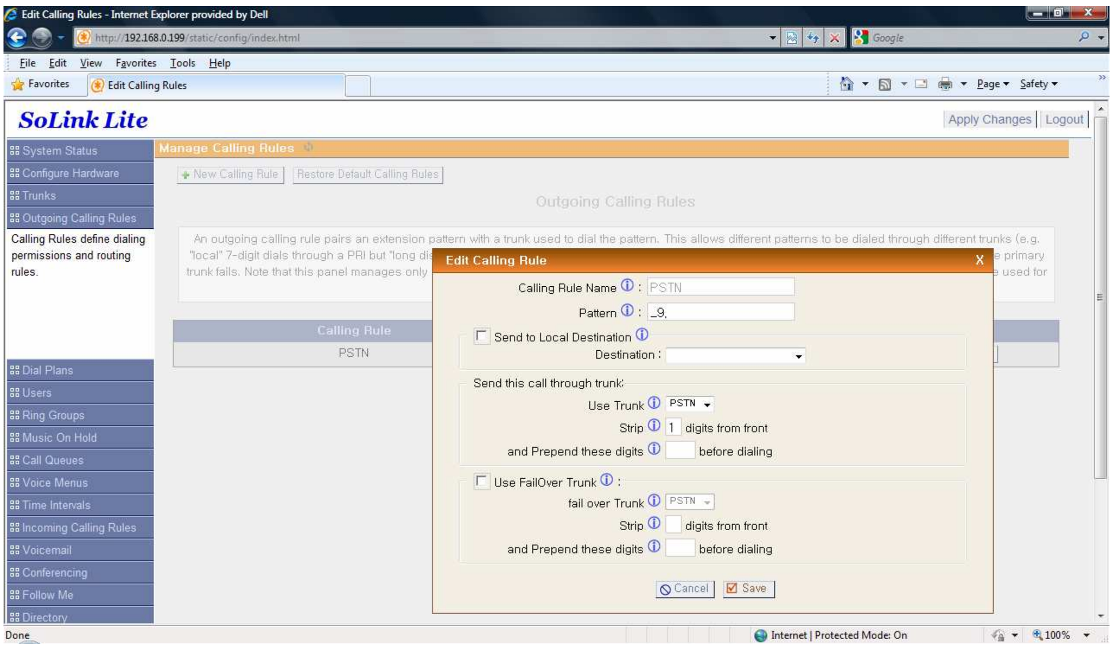


Note that after the trunk configuration (or any other configurations) has been modified, an *Apply Chan*ges button will appear at the top right side of the screen, click on this button to apply the changes to the unit.

7.0 SETTING UP OUTGOING CALLING RULES

Outgoing Calling Rules defines the number(s) that a dial plan is allowed to dial when making outgoing calls. It also defines the associated trunk(s) that will be used for the calling rule.

To configure a calling rules, simply select the *Outgoing Calling Rules* menu. You will then be allowed to define new calling rules and/or update existing rules.



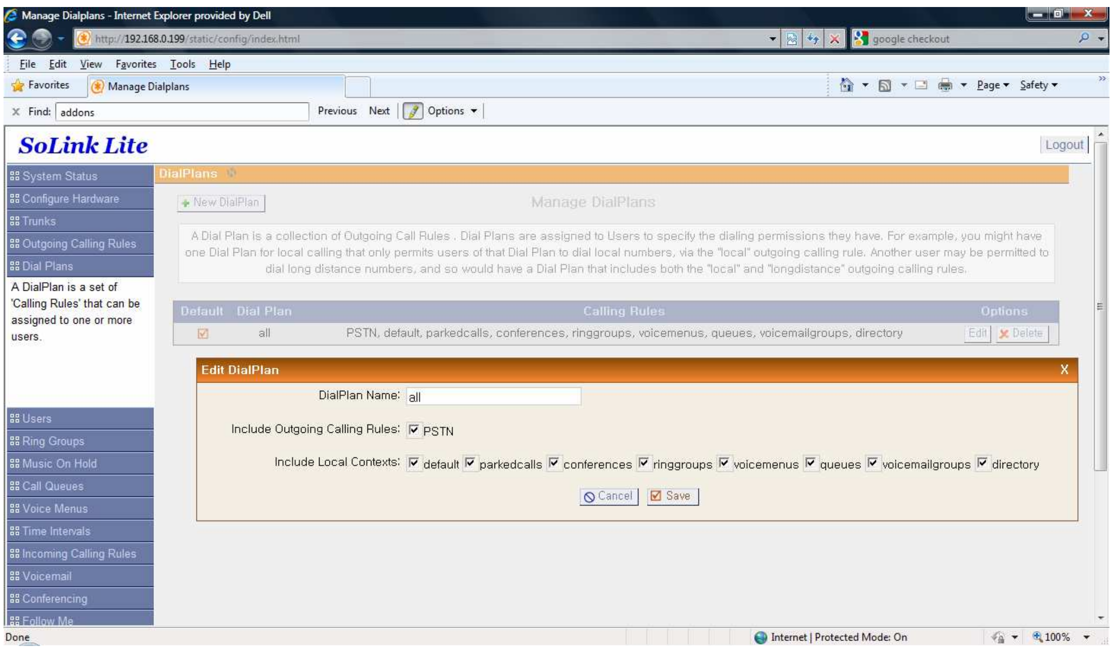
For example, the default calling rule *PSTN* allows user to make call through the FXO trunks by dialing 9 followed by the telephone number desired. This is archived by specifying a dial pattern of _9. (i.e. perform pattern match for dialing number starting with the digit 9 followed by anything; the system will then strip the first digit (i.e. 9) from the dialed number before sending it out through the *PSTN* trunk. No failover trunk is specified in this case.

These outgoing calling rules are then referenced by dial plans to define individual user's call privileges.

8.0 SETTING UP DIAL PLANS

A Dial Plan is a collection of Outgoing Call Rules . Dial Plans are assigned to Users to specify the dialing permissions they have. For example, you might have one Dial Plan for local calling that only permits users of that Dial Plan to dial local numbers, via the "local" outgoing calling rule. Another user may be permitted to dial long distance numbers, and so would have a Dial Plan that includes both the "local" and "longdistance" outgoing calling rules.

To set up a dial plan, simply select the *Dial Plans* menu. You will then be allowed to define new dial plan and/or update existing plans.



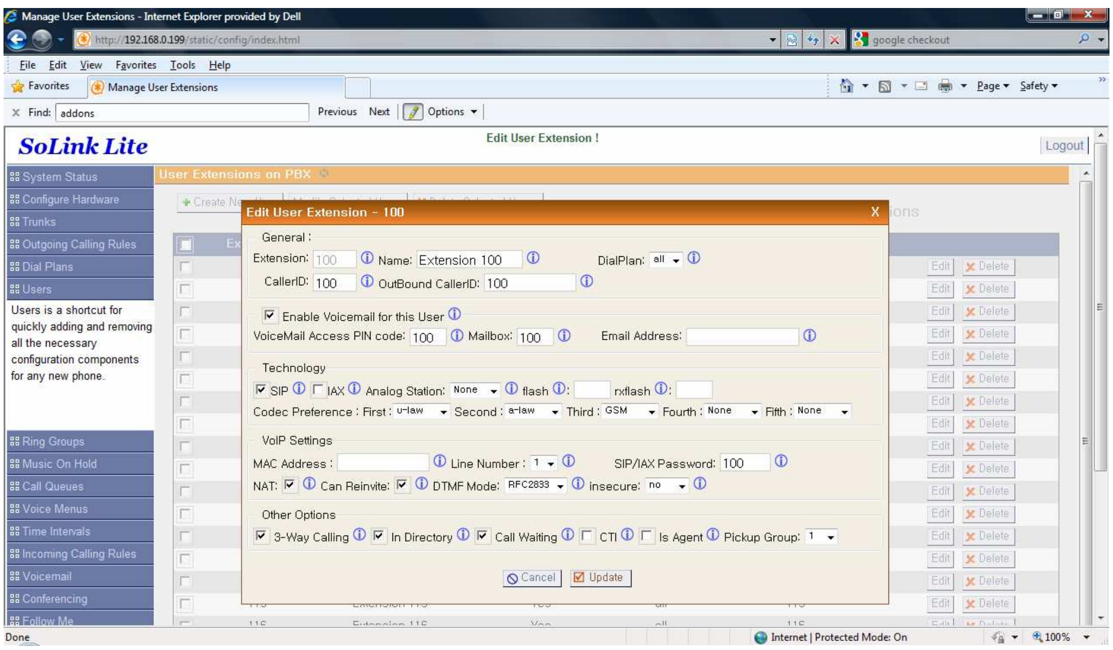
9.0 SETTING UP USERS

Users are individual extensions of the PBX. Each user is assigned a unique extension number and optionally a voice mailbox.

Three types of extension devices are supported:

- SIP – an IP phone or softphone that supports the SIP protocol
- IAX – an IP phone or softphone that supports the IAX protocol
- Analog Station – an analog phone connected to one of the FXS port of the unit

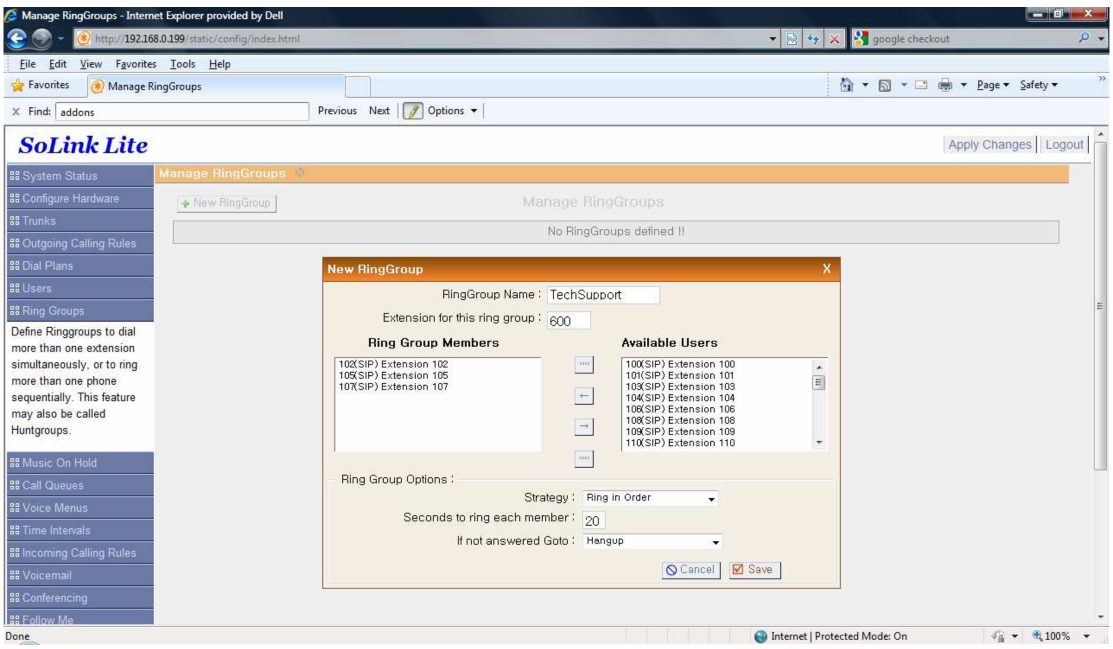
To set up a user extension, simply select the *Users* menu. You will then be allowed to define new users and/or update existing users. Press the *Update* button to update the changes. These changes will be effective once the *Apply Changes* button is pressed.



10.0 SETTING UP RING GROUPS

Ring Groups, also known as hunt groups or extension groups, are groups of related extensions (e.g. technical support, accounting, customer service, etc). Ring groups may be configured to dial all extensions within the group simultaneously or sequentially.

To set up a ring group, simply select the *Ring Groups* menu. You will then be allowed to define new ring groups and/or update existing groups. Press the *Update* button to update the changes. These changes will be effective once the *Apply Changes* button is pressed.



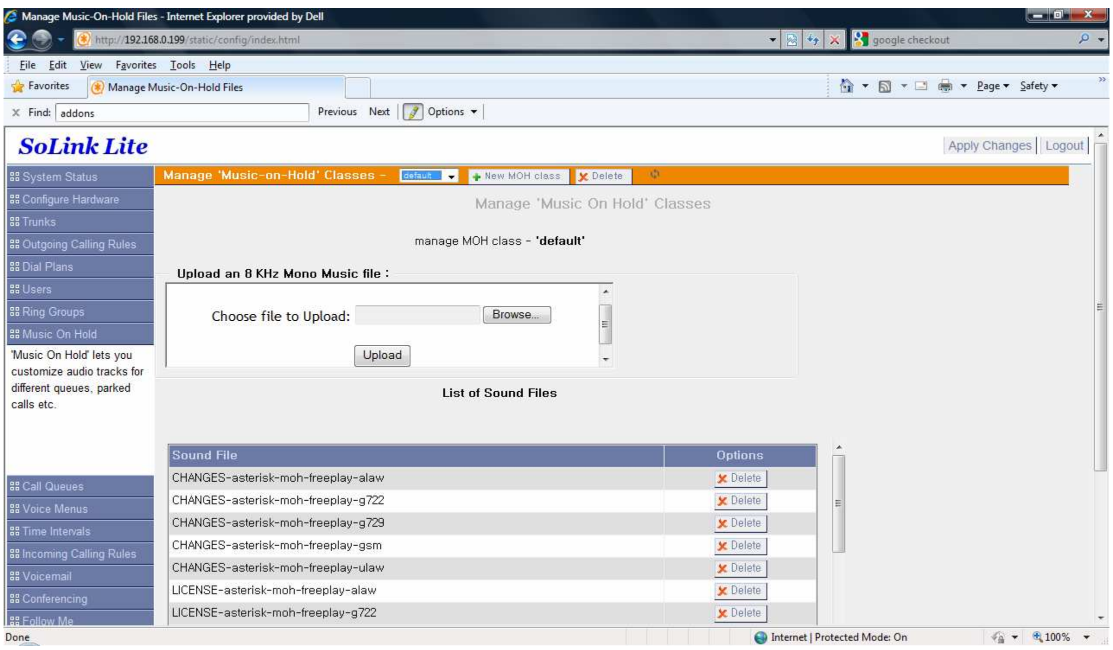
11.0 SETTING UP MUSIC ON HOLD

Music On Hold allows you to customize audio tracks for different queues, parked calls, etc.

To set up a music-on-hold class, simply select the *Music On Hold* menu. To create a new music-on-hold class, simply click on the *New MOH Class* button, enter the name of the new music-on-hold class, and then click on the *Add* button.

To add or delete music files associated with a music-on-hold class, select the desired music-on-hold class and then upload/delete the music file(s) as desired. Click the Apply Changes button to apply the changes.

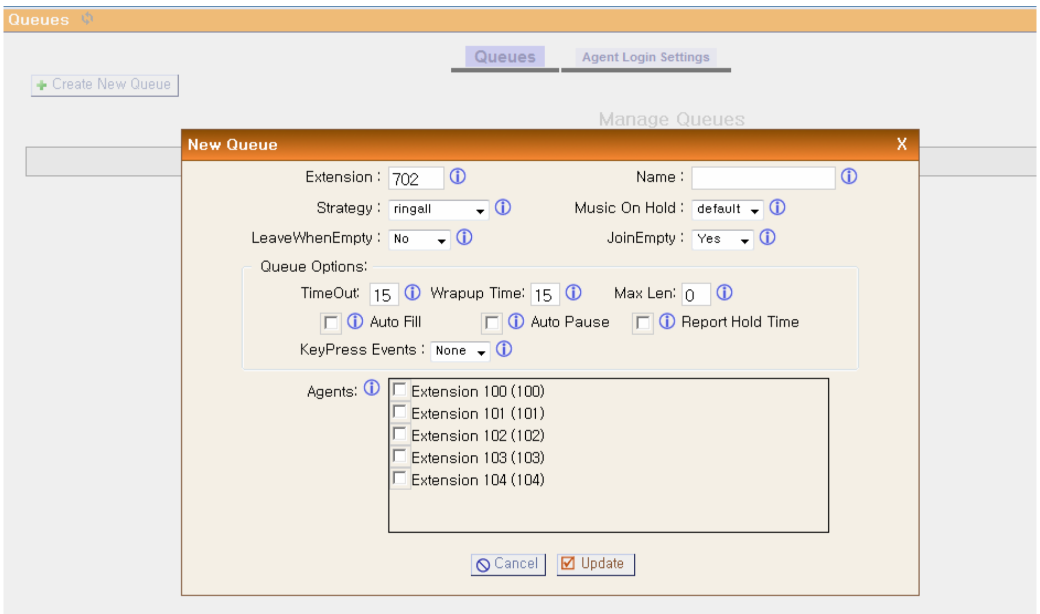
Please note that you cannot upload new MOH files at this point. Uploading new MOH files may be done using SCP or FTP. Please contact support@linksoft.com.hk for details.



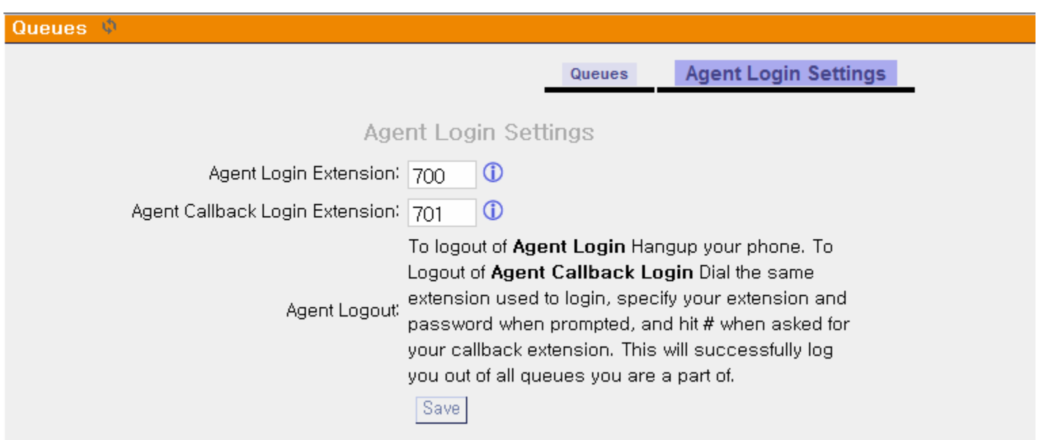
12.0 SETTING UP CALL QUEUES

Call queues allow calls to be sequenced to one or more agents.

To set up a call queue, simply select the *Call Queues* menu and click on the *Queues* tab. You will then be allowed to define new call queues and/or update existing queues. Press the *Update* button to update the changes. These changes will be effective once the *Apply Changes* button is pressed.



An agent must login first before accepting calls from the queue(s). To define the Agent Login and Agent Callback Login Extensions, click on the *Agent Login Settings* tab. Note that an agent will remain on-line if he logged in through the Agent Login extension. If an agent logs in through the Agent Callback Login extension, the call queue will send the call to the agent instead.

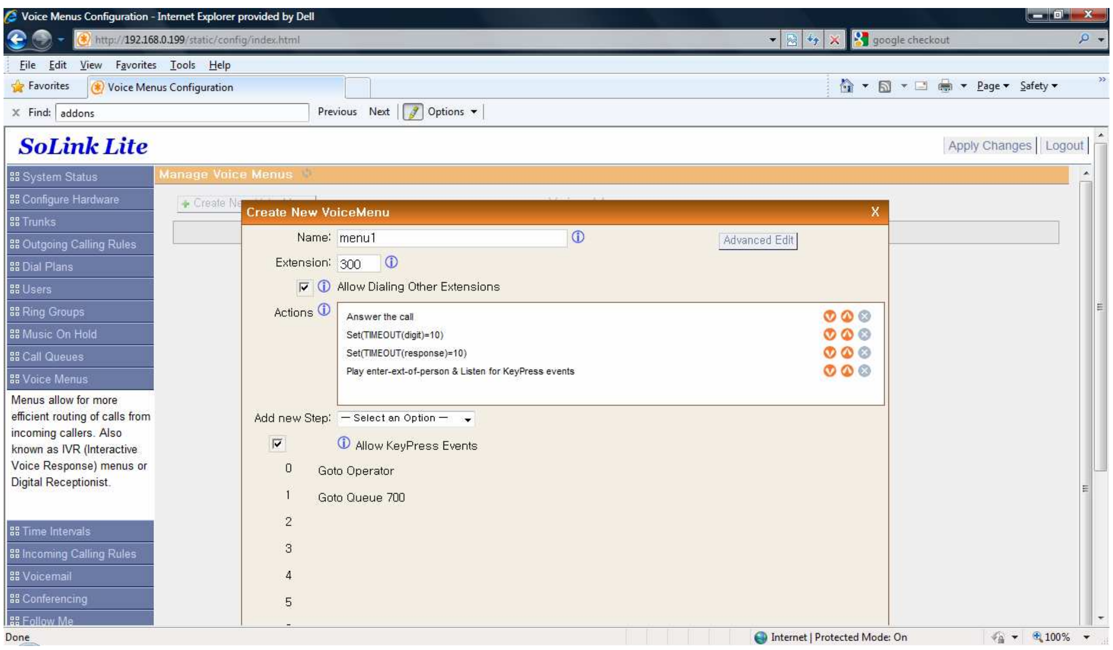


13.0 SETTING UP VOICE MENUS

Menus, also known as IVR (interactive voice response) or Digital Receptionist, provides a better telephony user interface to guide callers to reach the desired destination.

A voice menu typically consists of a sequence of actions (such as playing a menu prompt) and then wait for the caller to press a DTMF key. Depending on the key pressed, it will execute certain action or go to another voice menu. If the *Dialing Other Extensions* option is checked, the caller will also be allowed to dial an extension number other than the ones explicitly defined.

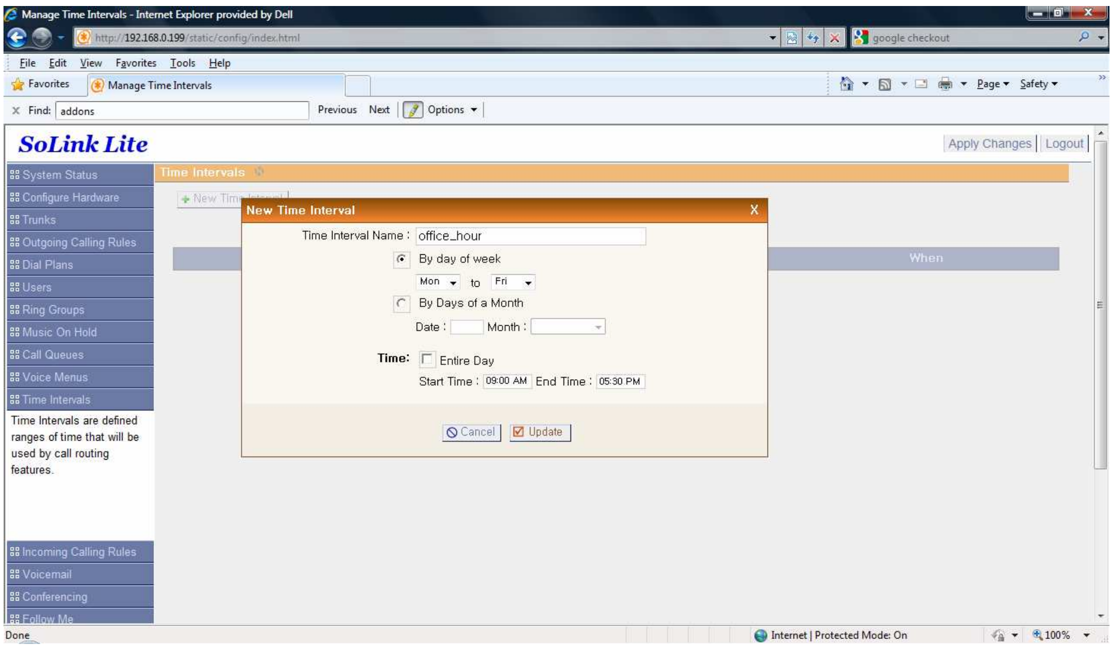
To set up a voice menu, simply select the *Voice Menus* menu and click on the *Create New VoiceMenu* button to create a new voice menu. Enter the desired sequence of actions and the associated command processing parameters. Press the *Update* button to update the changes. These changes will be effective once the *Apply Changes* button is pressed.



14.0 DEFINING TIME INTERVALS

Time Intervals are defined ranges of time that will be used by call routing features.

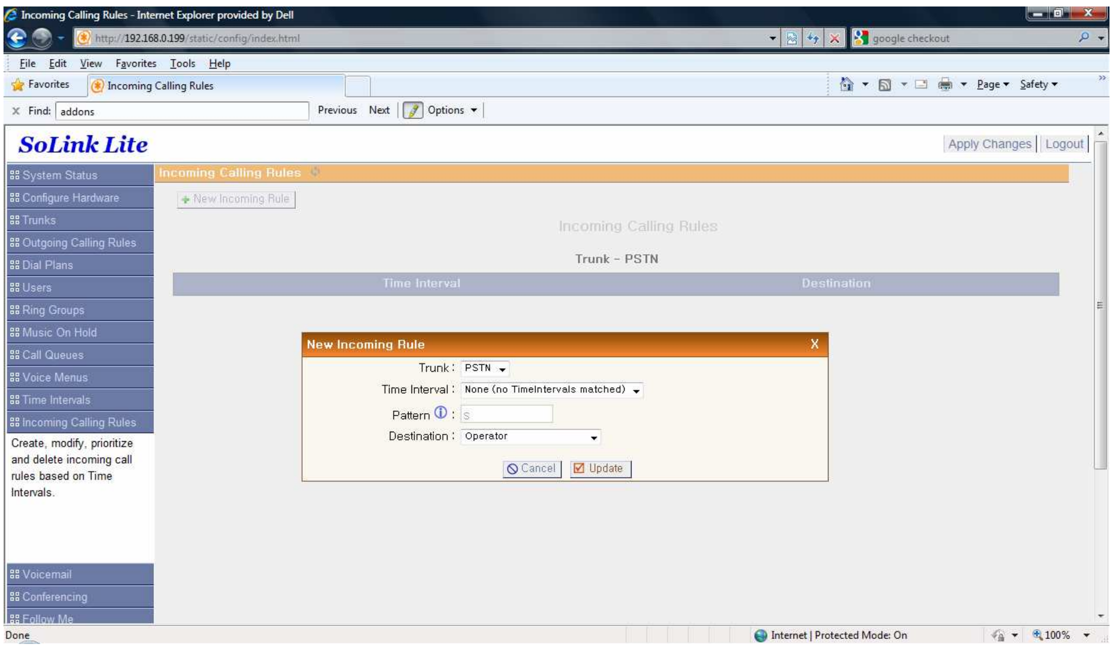
To set up a time interval, simply select the *Time Intervals* menu. You will then be allowed to define new time intervals and/or update existing intervals. Press the *Update* button to update the changes. These changes will be effective once the *Apply Changes* button is pressed.



15.0 DEFINING INCOMING CALLING RULES

Incoming Calling Rules allow the administrator to create, modify, prioritize and delete incoming call rules for handling incoming calls based on the service provider (or trunks), the number called, and the Time Intervals.

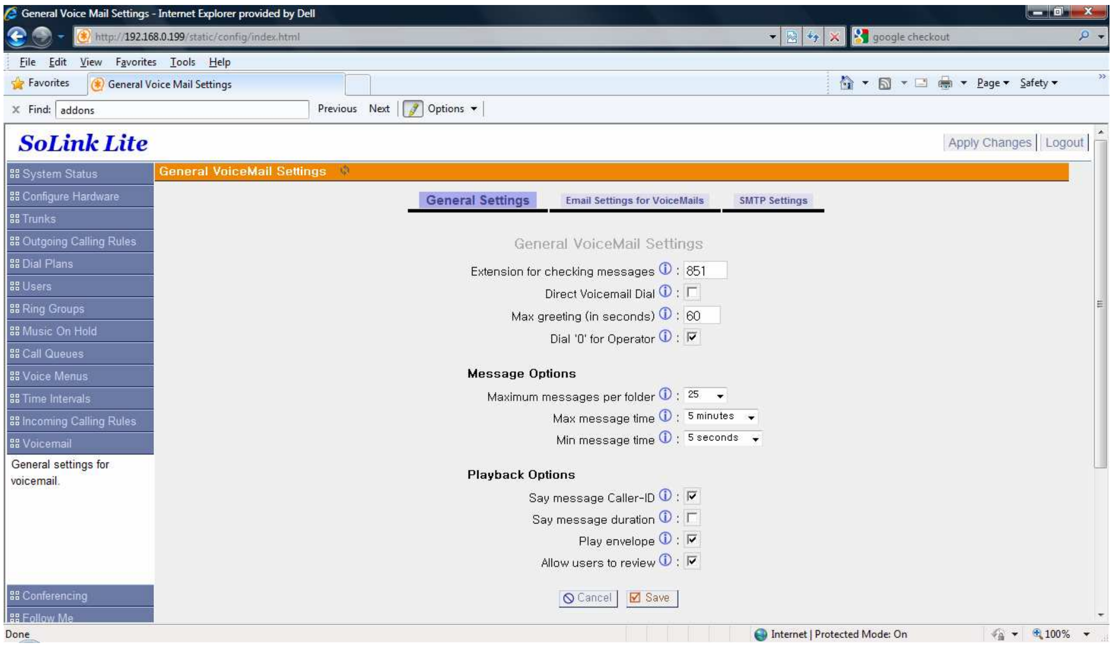
To set up a time interval, simply select the *Incoming Calling Rules* menu. Press the *Update* button to update the changes. These changes will be effective once the *Apply Changes* button is pressed.



16.0 CONFIGURING VOICEMAIL

This screen defines the general settings of voicemail function.

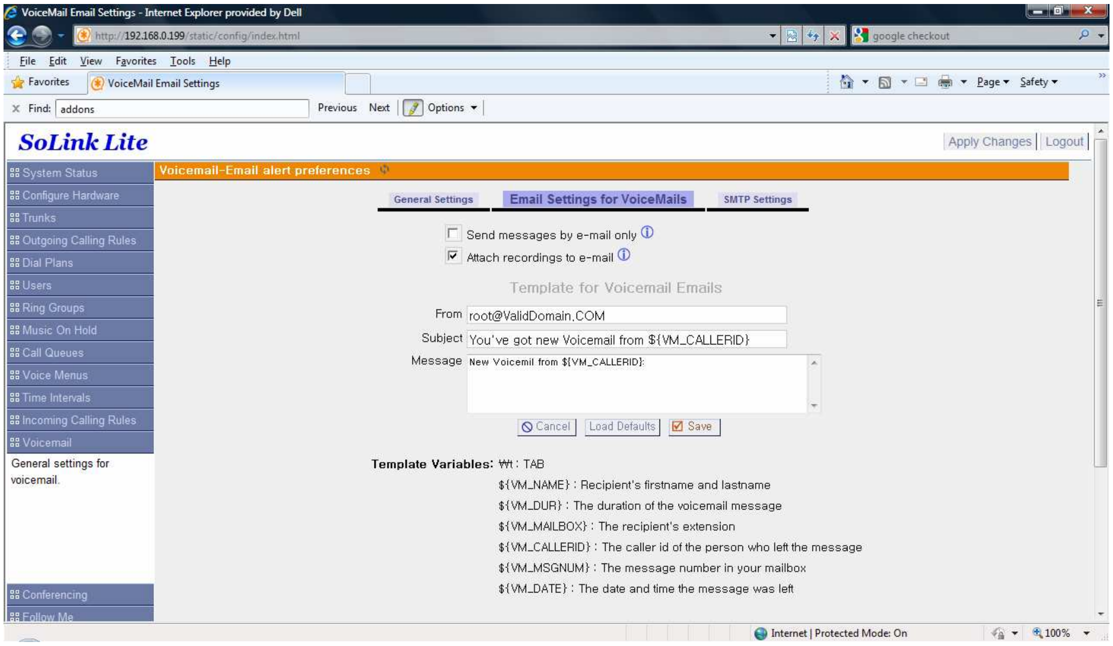
To configure general voicemail settings, simply select the *Voicemail* menu and then select the *General Settings* tab. Press the *Update* button to update the changes. These changes will be effective once the *Apply Changes* button is pressed.

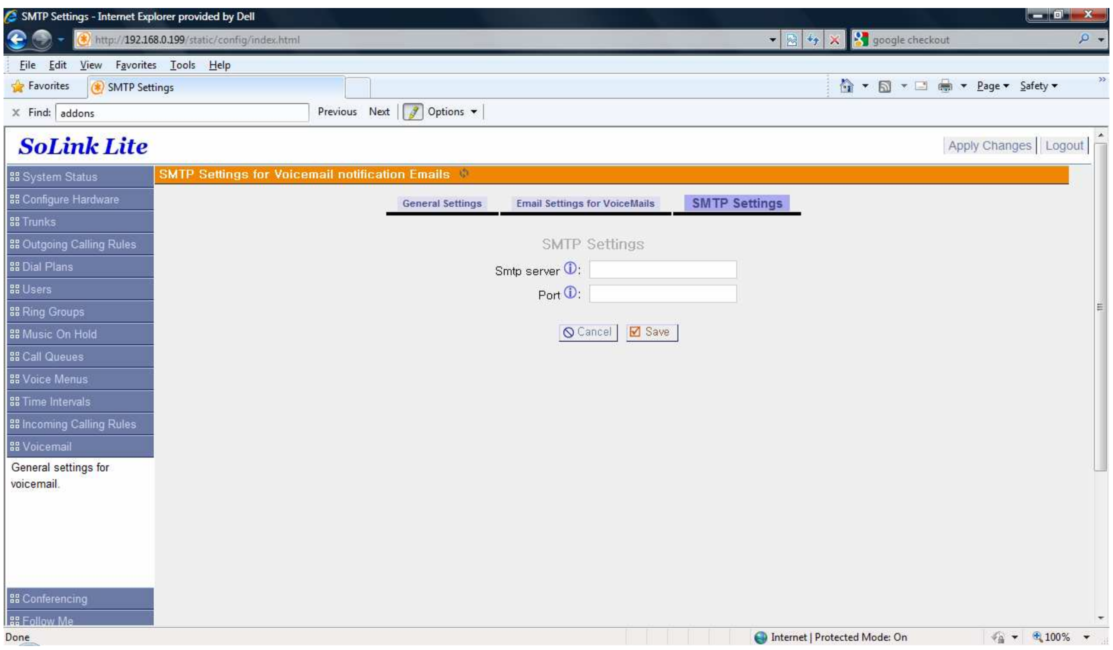


If you would like to enable email message notification of voicemail, set up the *Email Settings for VoiceMails* and *SMTP Settings* tabs.

Note that if your SMTP server needs authentication, you may need to put your username and password in the file ssmtp.conf (via SSH access) as below:
[/etc/ssmtp/ssmtp.conf]

root=user@mycompany.com // mailbox account
mailhub=mail.mycompany.com // smtp server
rewriteDomain=mycompany.com
hostname=user@mycompany.com
AuthUser=user@mycompany.com // mailbox account
AuthPass=mysecret // mailbox password
AuthMethod=LOGIN
FromLineOverride=YES

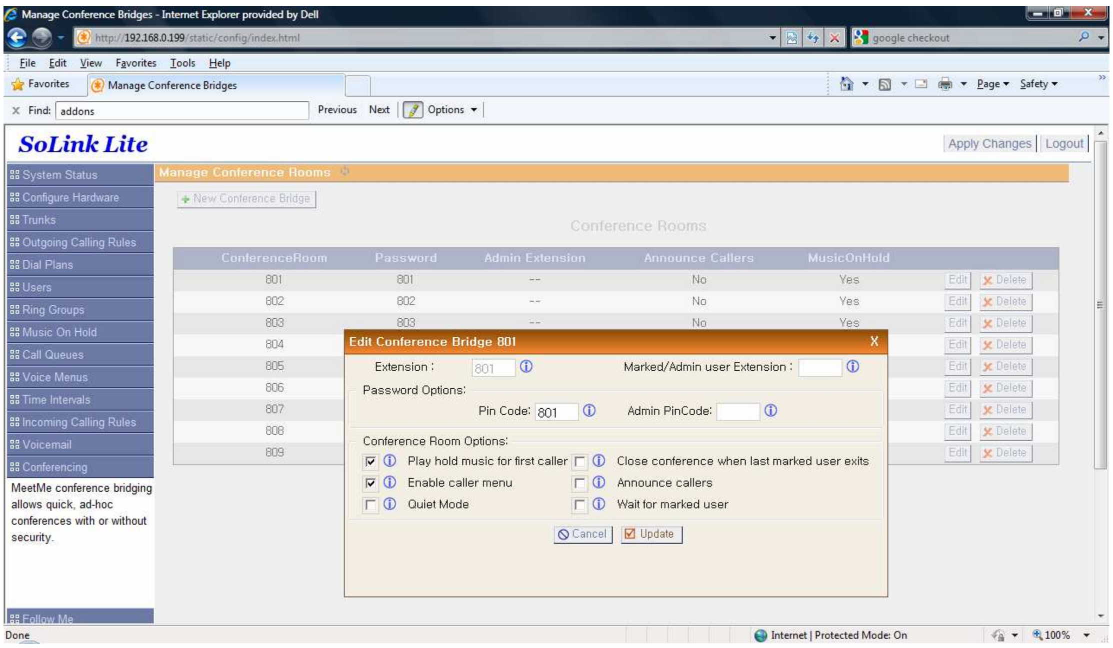

VoiceMail Email Settings - Internet Explorer provided by Dell
SoLink Lite
Apply Changes
Logout
Voicemail-Email alert preferences
General Settings
Email Settings for VoiceMails
SMTP Settings
Send messages by e-mail only
Attach recordings to e-mail
Template for Voicemail Emails
From root@ValidDomain.COM
Subject You've got new Voicemail from ${VM_CALLERID}
Message New Voicemil from ${VM_CALLERID}
Cancel
Load Defaults
Save
Template Variables: \t : TAB
${VM_NAME} : Recipient's firstname and lastname
${VM_DUR} : The duration of the voicemail message
${VM_MAILBOX} : The recipient's extension
${VM_CALLERID} : The caller id of the person who left the message
${VM_MSGNUM} : The message number in your mailbox
${VM_DATE} : The date and time the message was left
System Status
Configure Hardware
Trunks
Outgoing Calling Rules
Dial Plans
Users
Ring Groups
Music On Hold
Call Queues
Voice Menus
Time Intervals
Incoming Calling Rules
Voicemail
General settings for voicemail.
Conferencing
Follow Me


SMTP Settings - Internet Explorer provided by Dell
SoLink Lite
Apply Changes
Logout
SMTP Settings for Voicemail notification Emails
General Settings
Email Settings for VoiceMails
SMTP Settings
SMTP Settings
Smtp server
Port
Cancel
Save
General settings for voicemail.

17.0 SETTING UP CONFERENCE BRIDGE

Meetme Conference Bridging allows quick, ad-hoc conference with or without security.

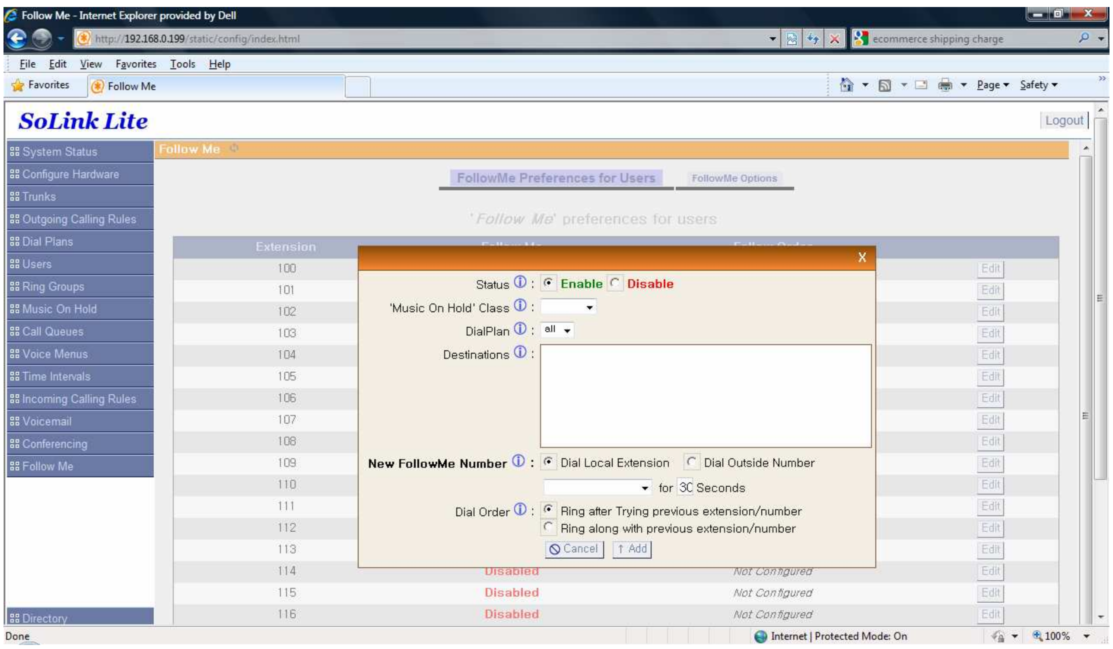
To configure a conference bridge, simply select the *Conferencing* menu option. You will then be allowed to define new conference room and/or update existing rooms. Press the *Update* button to update the changes. These changes will be effective once the *Apply Changes* button is pressed.



18.0 SETTING UP FOLLOW-ME OPTIONS

Follow-me allows callers to reach a destination extension even though extension holder is not at his desk. If there is no answer to the destination extension, the PBX will try to contact the extension by calling a list of “find-me” extensions/numbers in sequence.

To set up the follow-me instruction for an extension, simply select the *Follow Me* menu option and then select the desired extension. You will then be able to edit the list of follow-me numbers for the extension. Press the *Update* button to update the changes. These changes will be effective once the *Apply Changes* button is pressed.



To edit the general options, click on the *FollowMe Options* tab instead.

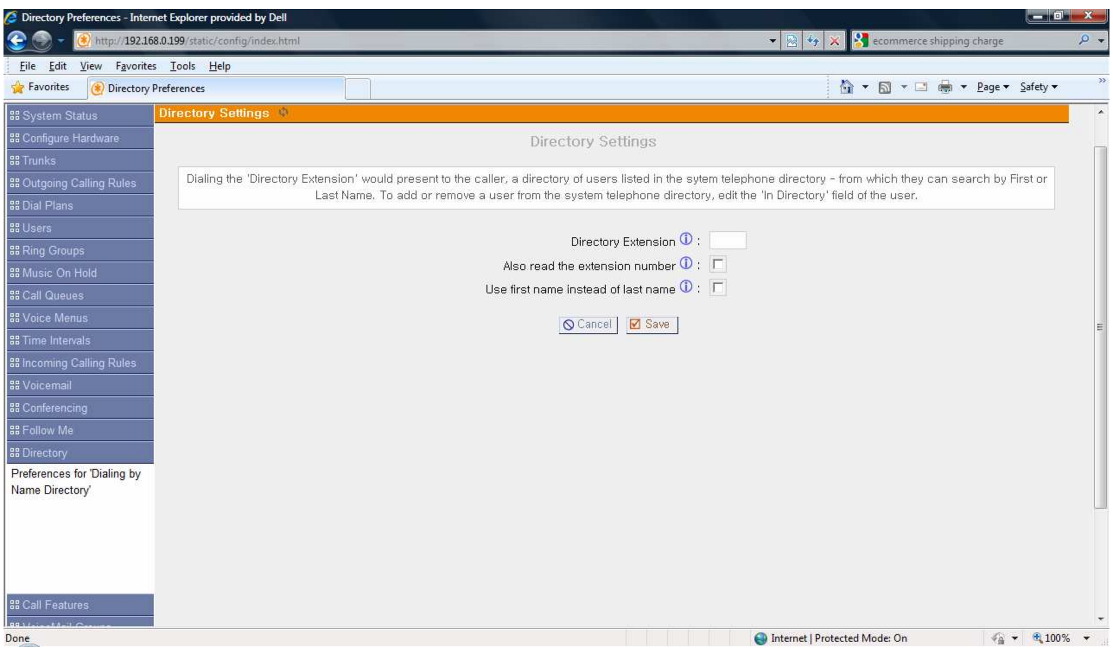
19.0 CONFIGURING DIRECTORY SETTINGS

The *Directory Settings* screen allows the caller to search for an extension by dialing his name.

To configure the directory settings, simply select the *Directory* menu option. Press the *Update* button to update the changes. These changes will be effective once the *Apply Changes* button is pressed.

Dialing the *Directory Extension* would present to the caller, a directory of users listed in the system telephone directory - from which they can search by First or Last Name. To add or remove a user from the system telephone directory, edit the 'In Directory' field of the user.

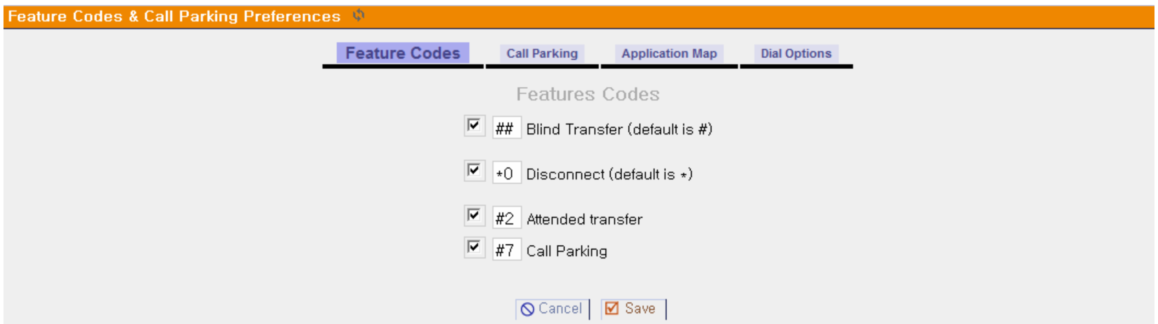
Note that this feature only work with English language and is disabled by default.

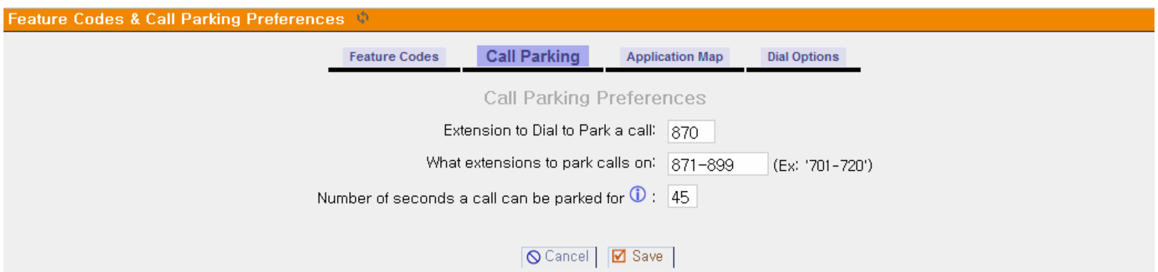


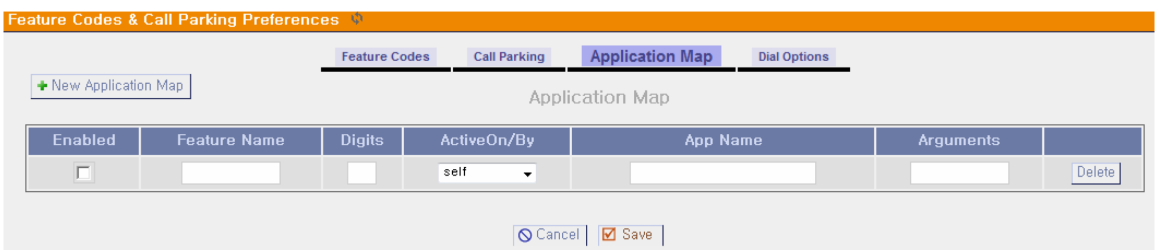
20.0 CONFIGURING CALL FEATURES

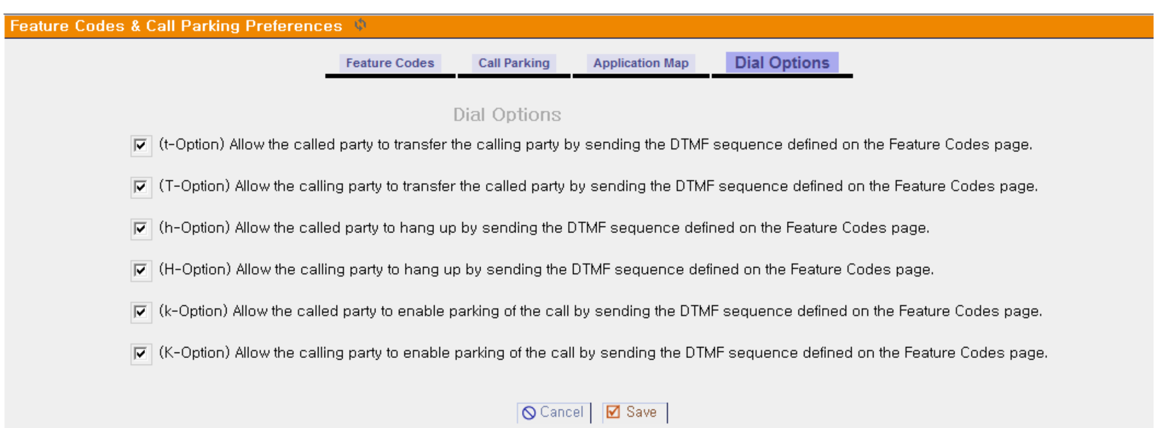
The Call Features screen allows the user to configure various feature codes and calling preferences. These screens are for advanced user only and seldom needs to be changed.

To configure various call features, simply select the *Call Features* menu option and then the desired tab.





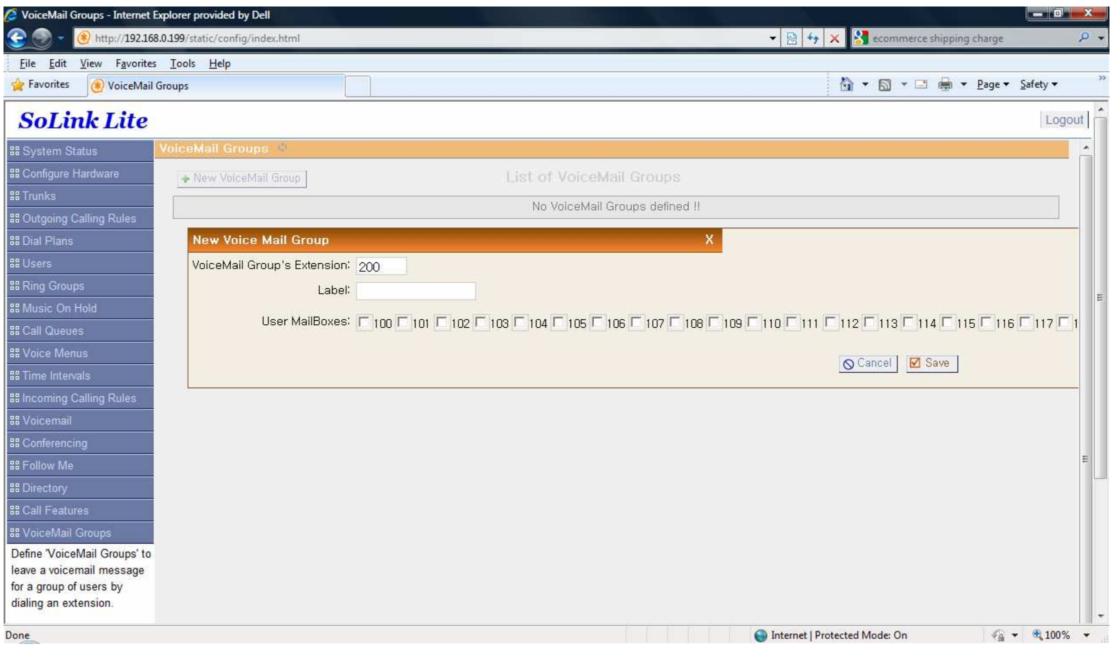




21.0 SETTING UP VOICEMAIL GROUPS

VoiceMail Groups allow the caller to leave a voicemail message for a group of users by dialing an extension.

To set up voicemail groups, simply select the *Voicemail Groups* menu option. You will then be able to add new voicemail groups or update/delete existing groups. Press the *Update* button to update the changes. These changes will be effective once the *Apply Changes* button is pressed.

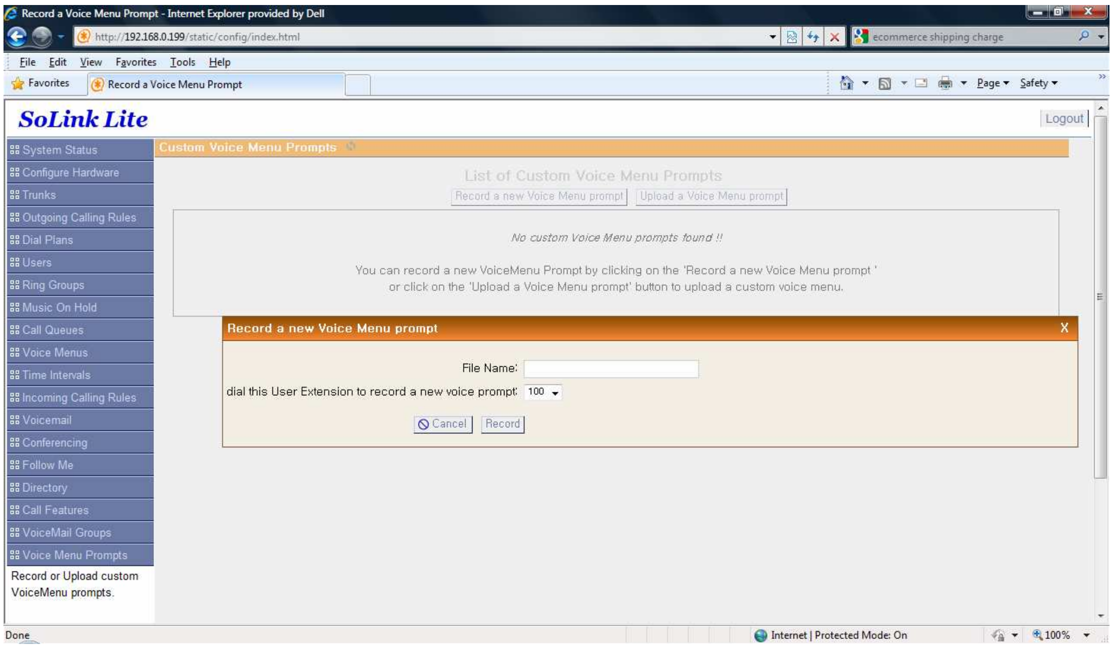


22.0 SETTING UP VOICE MENU PROMPTS

The Voice Menu Prompts screen allows the user to record or upload custom voice prompts for voice menus.

To set up voice menu prompts, simply select the *Voice Menu Prompts* menu option. To record a new voice prompt, click on the *Record a new Voice Menu Prompt* button. The user will then be prompted to enter the filename and the extension to dial for recording. Click on the *Record* button and the system will call the specified extension; the user will then be able to record the voice prompt after picking up the phone. To upload a pre-record voice prompt, click on the Upload a Voice Menu Prompt button. The user will then be prompted to enter the filename to upload. Note that the file should be recorded in WAV format at 8 kHz, 16-bit mono.

Please note that you cannot upload pre-record voice prompts at this point. Uploading pre-record voice prompts may be done using SCP or FTP. Please contact support@linksoft.com.hk for details.

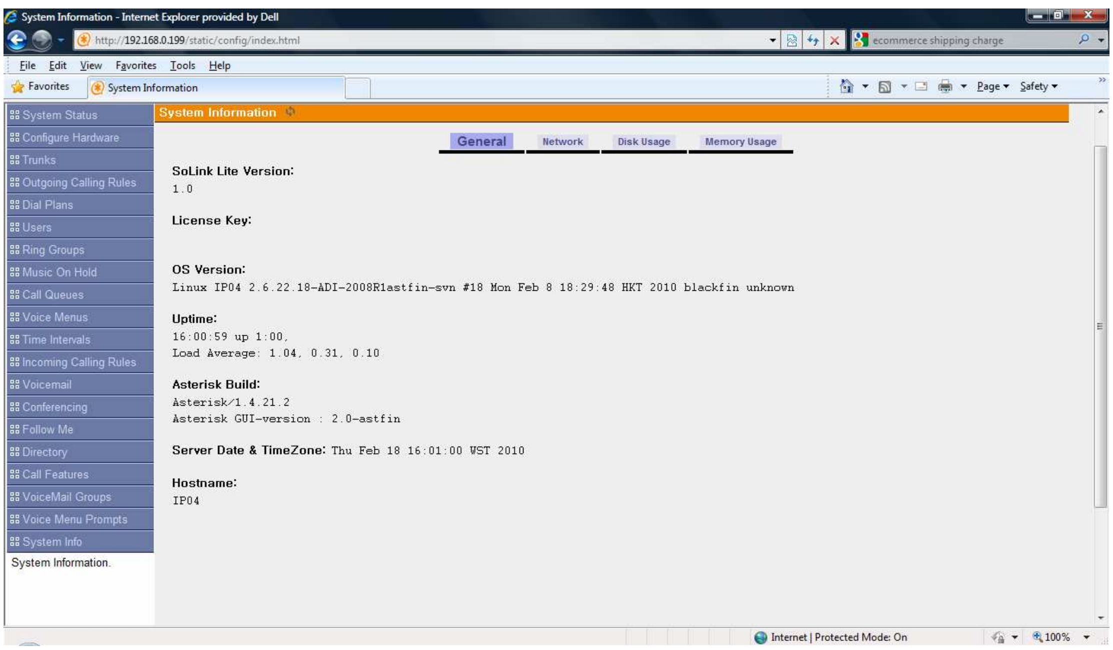


23.0 REVIEWING SYSTEM INFORMATION

The administrator may browse the system status by selecting the *System Info* menu option.

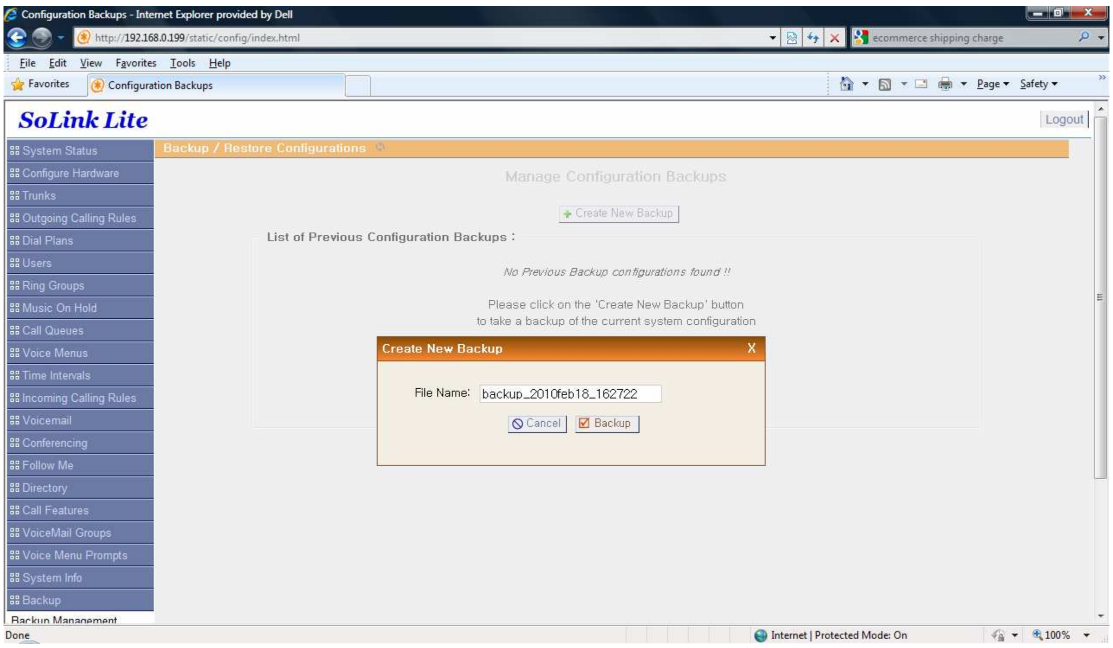
There are four types of system information available:

- General
  contains various system version and license information
- Network
  contains various network interface information
- Disk Usage
  contains disk usage information
- Memory Usage
  contains memory usage information



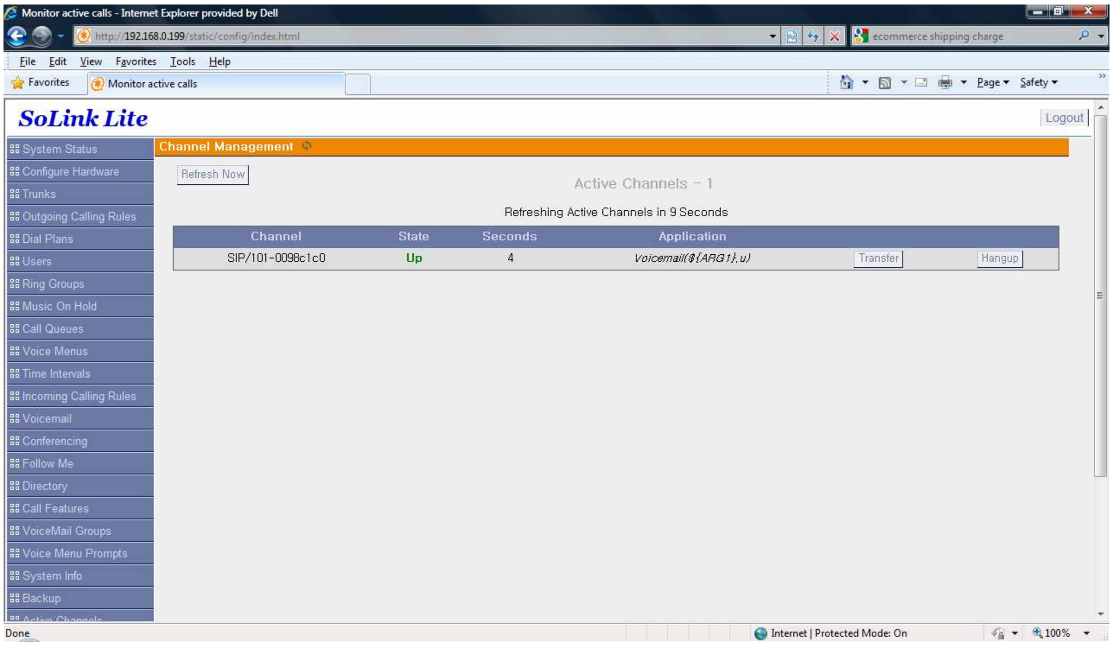
24.0 BACKUP / RESTORE

To backup or restore the configuration files, simply select the *Backup* menu option. Click on the *Create New Backup* button to create a new backup. To restore the configuration from previous backup, click on the *Restore Previous Config* button for the desired backup. To download a backup to the local PC, click on the *Download From Unit* button for the desired backup.



25.0 VIEWING ACTIVE CHANNELS

To display current active channels on the PBX, simply select the *Active Channels* menu option. Press the *Transfer* button to transfer the call to another extension. Press the *Hangup* button to hang up the call.

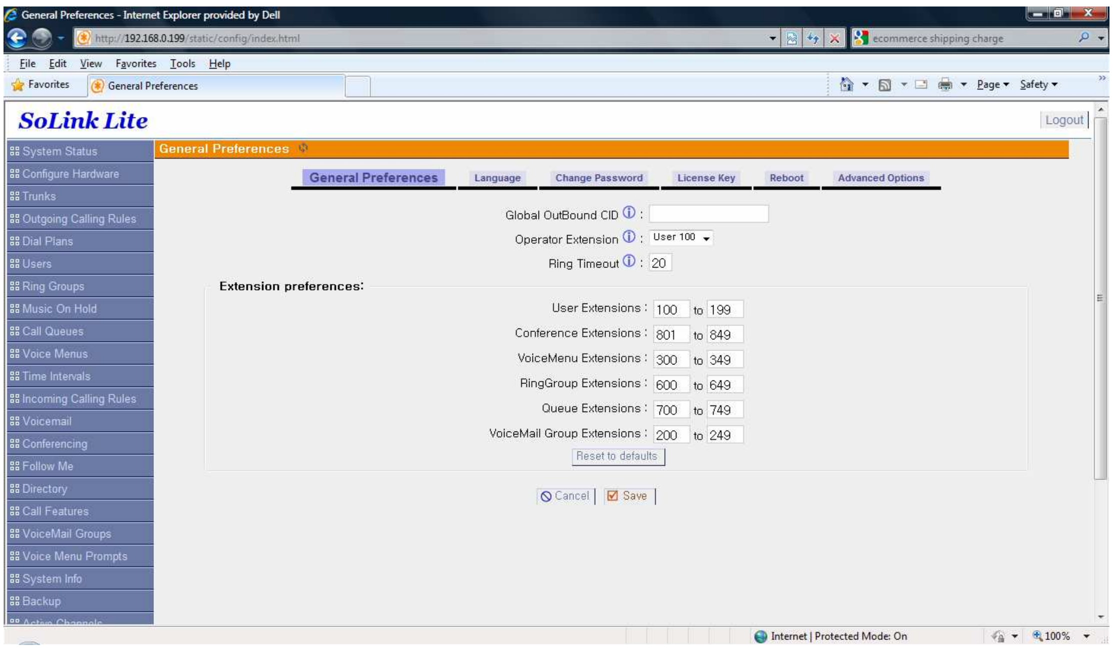


26.0 CONFIGURING MISCELLANEOUS OPTIONS

To configure miscellaneous options, simply select the *Options* menu.

26.1 General Preferences

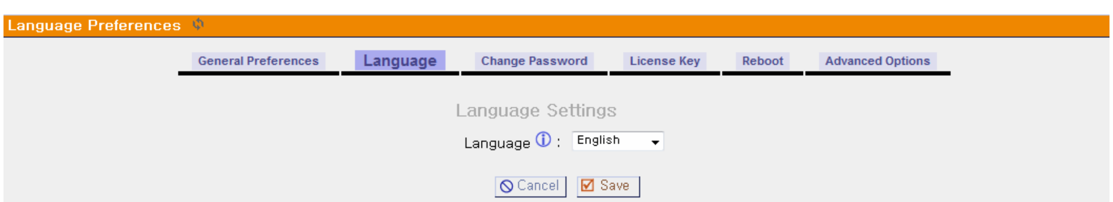
The General Preferences screen allows the administrator to configure global parameters including the global outbound callerid, operator extension, ring timeout, as well as the extension range to be used for various functions.



26.2 Language

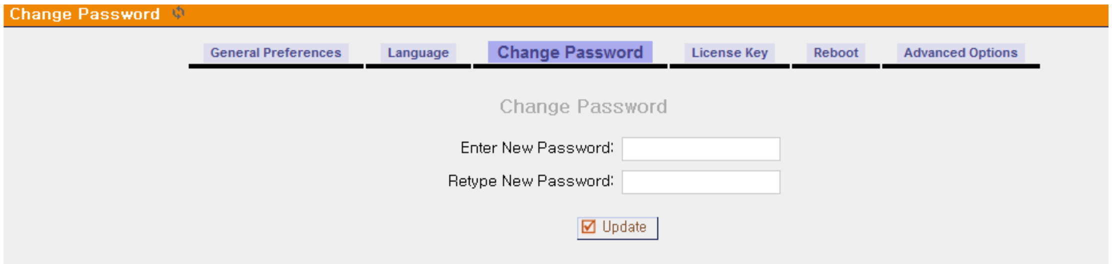
The Language Settings screen allows the administrator to select the language to be used for the unit. Valid language options are English, Cantonese, Mandarin, French and Spanish.

Note that a valid license key is required for downloading of certain languages. Please refer to section 25.4 for license key management.



26.3 Change Password

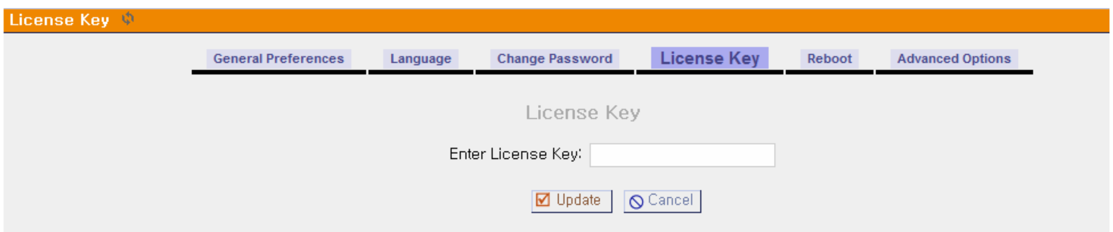
To change the administrator password, click on the *Change Password* tab, enter the new password (twice), and click the *Update* button to save the new password.



26.4 License Key

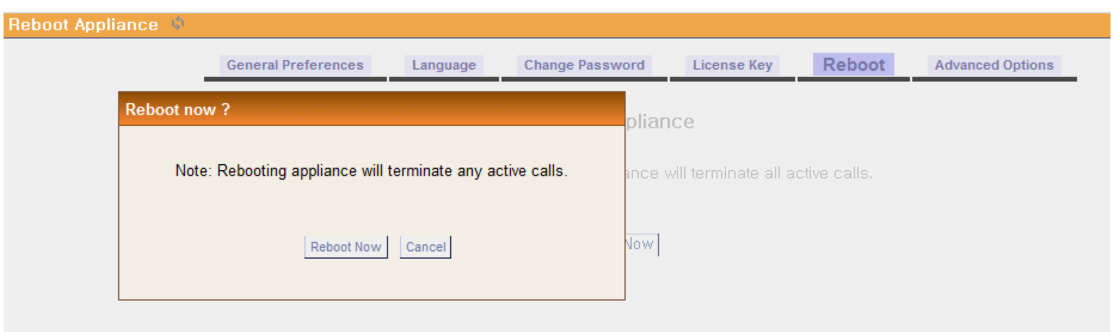
To update the license key, click on the License Key tab, enter the license key and then click on the *Update* button to validate and save the license key.

Note that this option requires a active Internet connection.



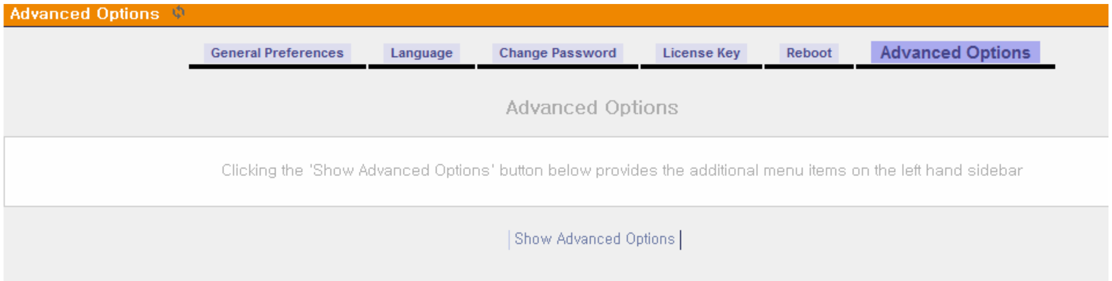
26.5 Reboot

To reboot the unit, click on the *Reboot* tab and then click on the *Reboot Now* button. The unit will be restarted upon confirmation.



26.6 Advanced Options

To show or hide advanced option menus, click on the *Advanced Options* tab, and then click on the *Show Advanced Options* (or *Hide Advanced Options*) button.



The advanced options include:

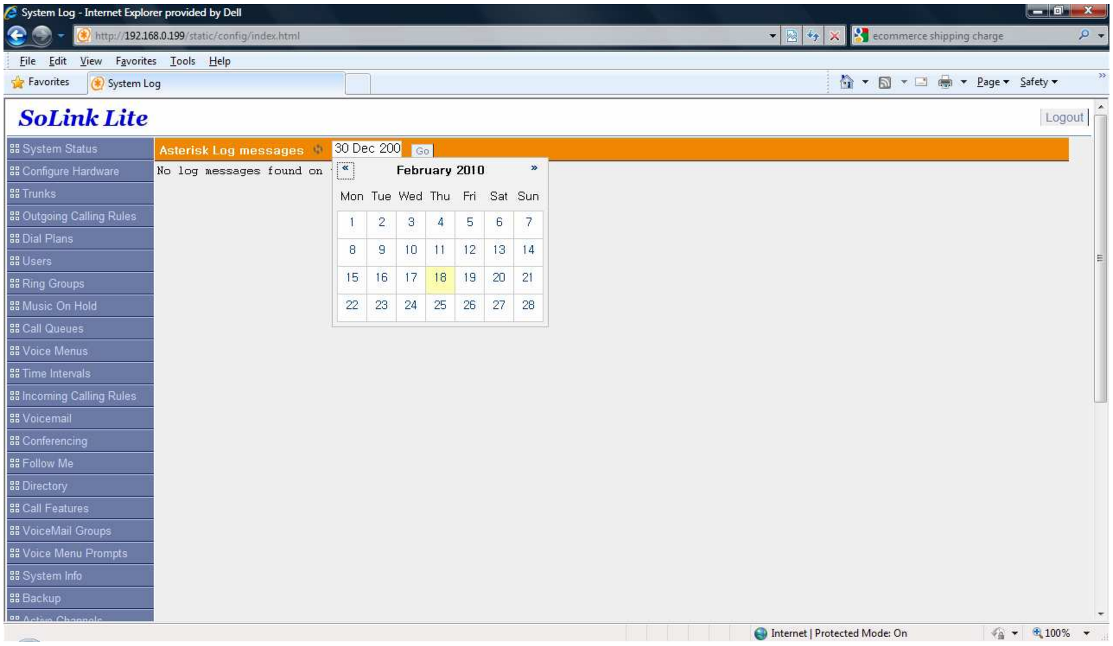
- Asterisk Logs
- Bulk Add
- File Editor
- Asterisk CLI
- IAX Settings
- SIP Settings
- Network Settings
- Firmware Update
- Call Detail Records

These options are intended for advanced users only (particular those who are familiar with Asterisk).

27.0 VIEWING ASTERISK LOGS

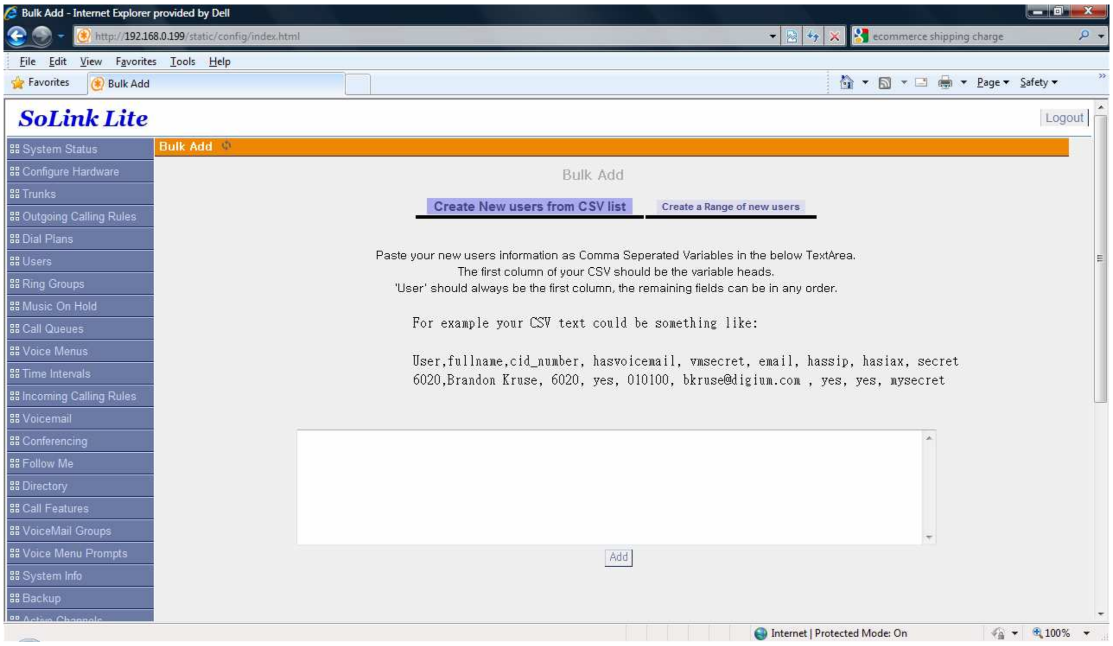
Due to the limited storage in the unit, the logging is turned off by default.

To view the Asterisk Log (after logging is turned on), select the *Asterisk Log* menu option, select the date of the log, and then click the *Go* button to display the log for the date specified.



28.0 ADDING USERS IN BULK

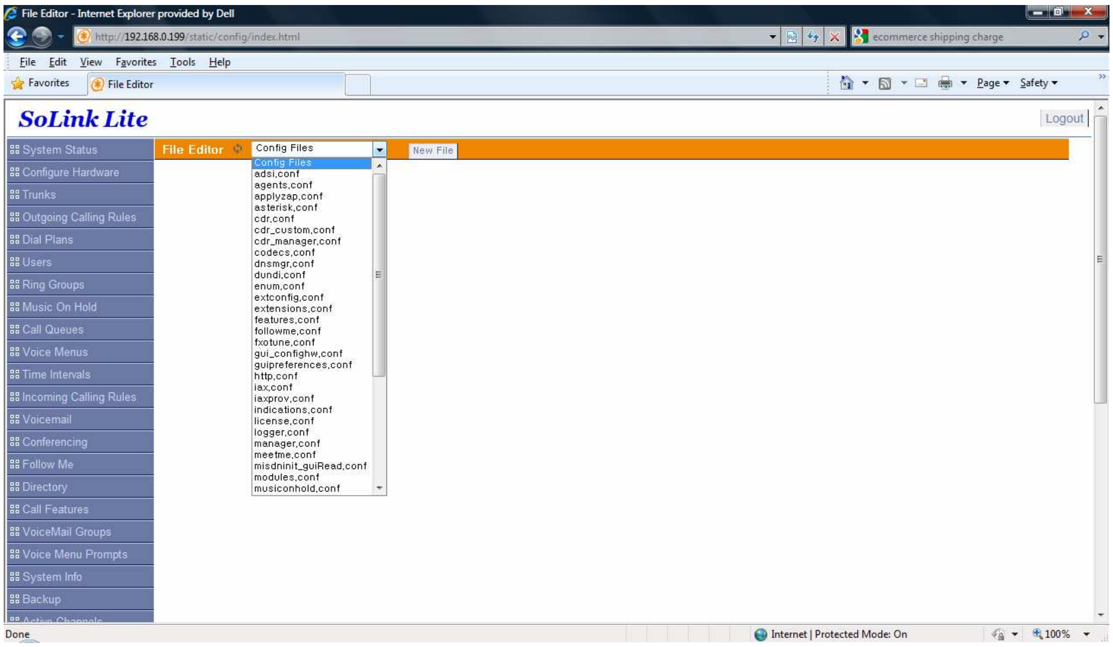
The Bulk Add option allows the administrator to add multiple users to the system in one easy step by importing from a CSV file or creating a range of extensions. Simply select the *Bulk Add* menu option and then select the desired method of creating the extensions.



29.0 EDITING ASTERISK CONFIGURATION DIRECTLY

To edit the Asterisk configuration files directly, select the File Editor menu options and then select the file to be updated.

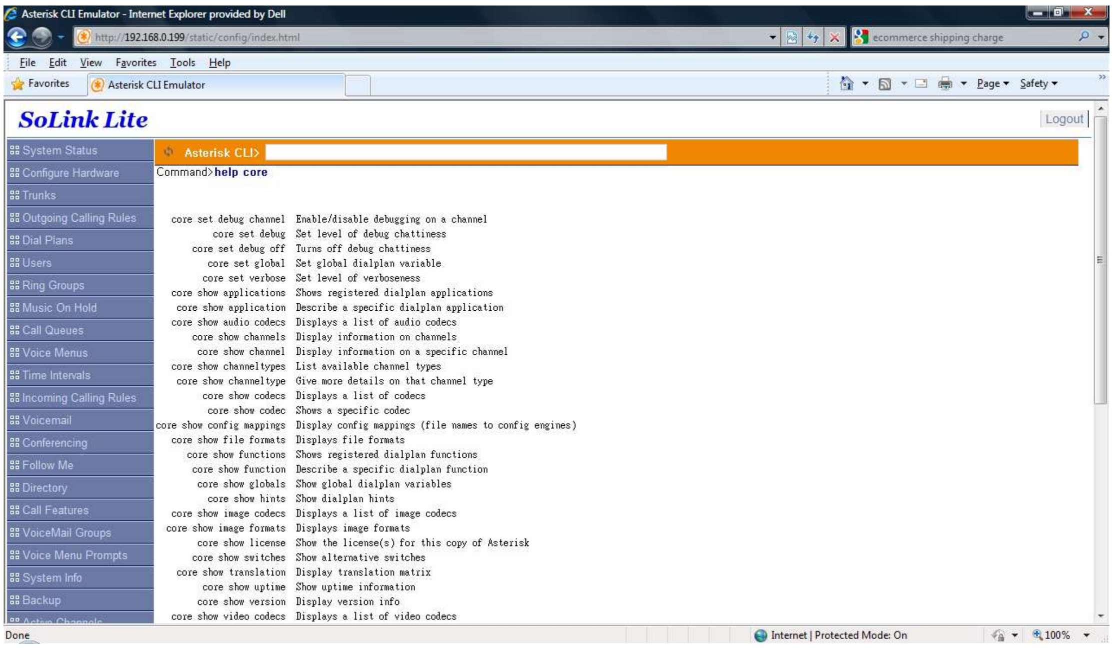
This is for advanced users who are familiar with Asterisk configuration.



30.0 ACCESSING ASTERISK CONSOLE DIRECTLY

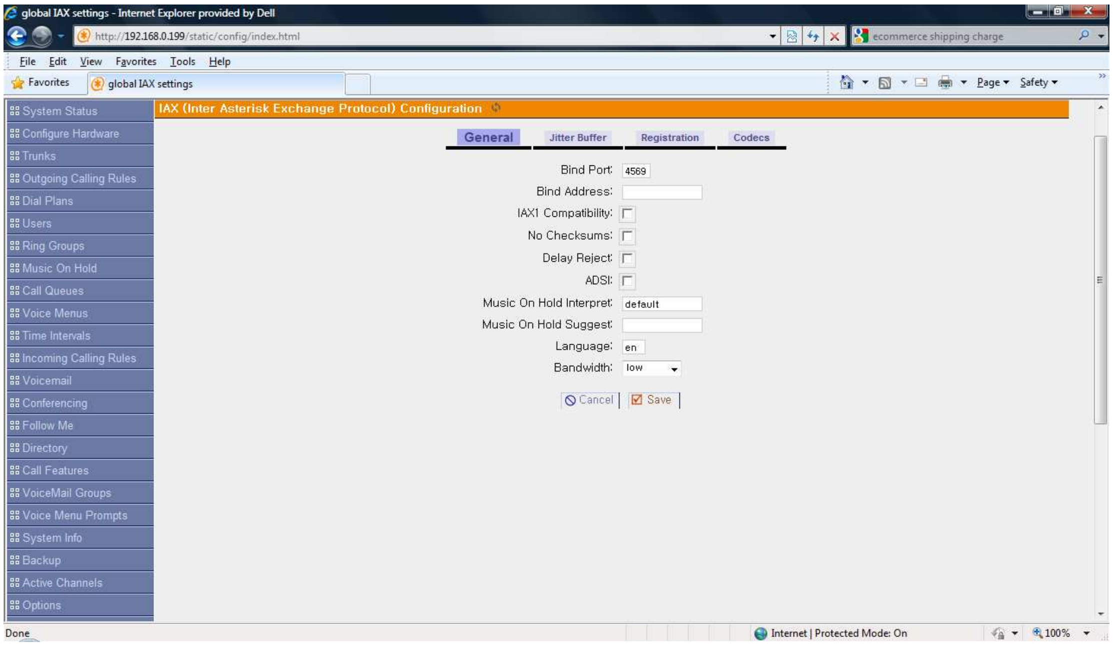
To access the Asterisk console directly, select the *Asterisk CLI* menu option. Type the Asterisk command in the *Asterisk CLI* input box, and press the *ENTER* key to execute the command.

This is for advanced users who are familiar with Asterisk.



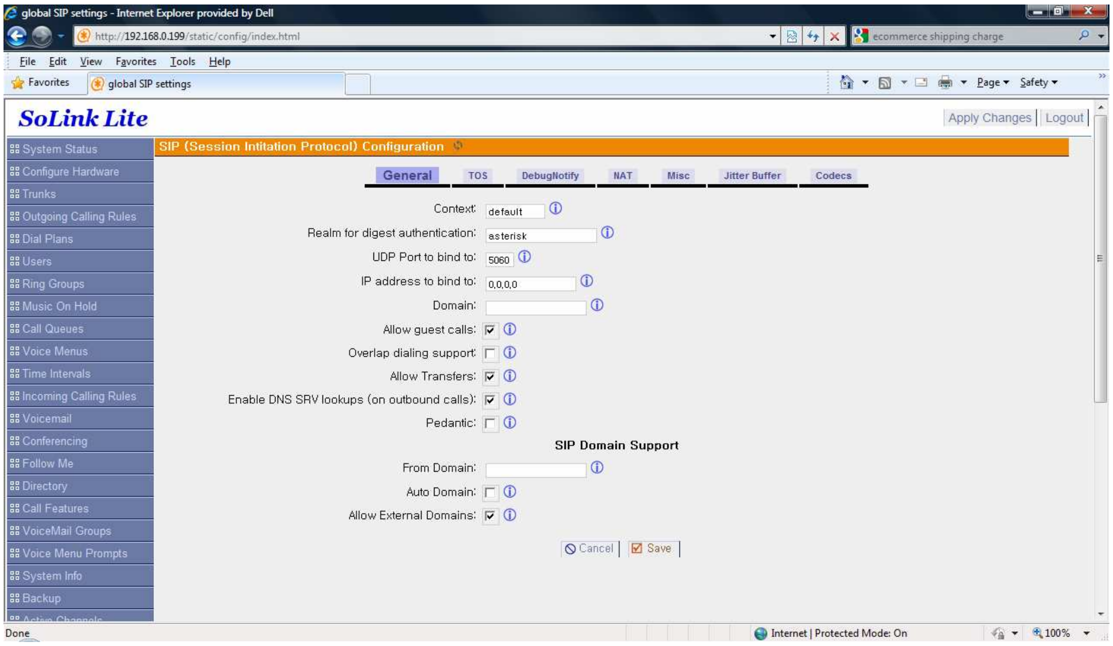
31.0 CONFIGURE IAX SETTINGS

To configure the global IAX Settings, select the *IAX Settings* menu option. Make any changes as desired and press the *Save* button to update the changes. These changes will be effective once the *Apply Changes* button is pressed.



32.0 CONFIGURE SIP SETTINGS

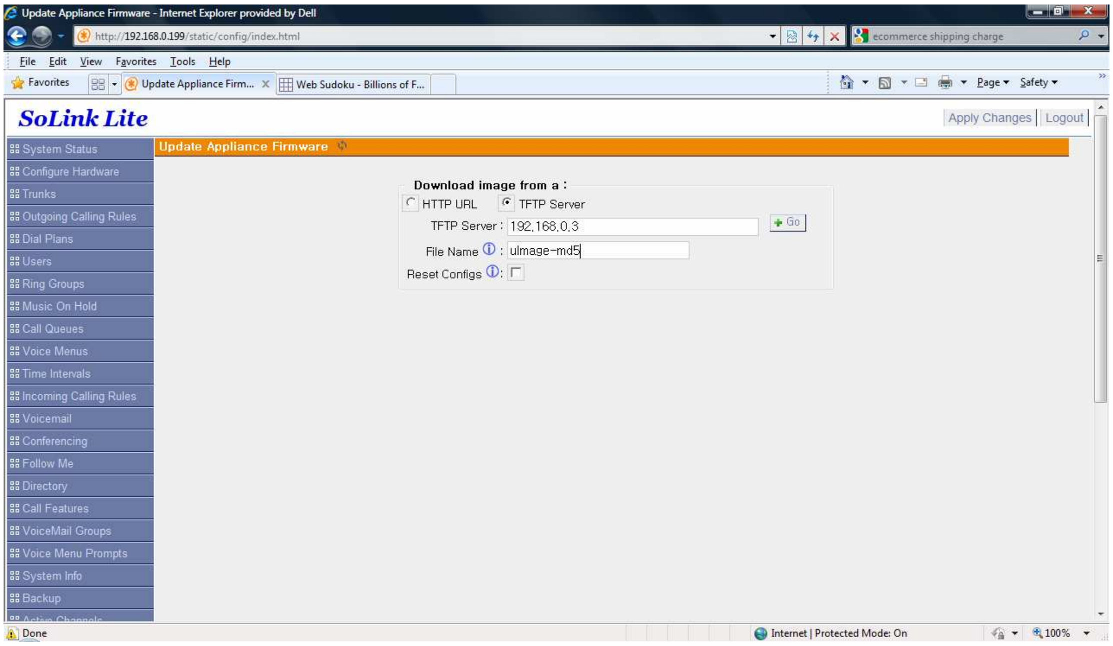
To configure the global SIP Settings, select the *SIP Settings* menu option. Make any changes as desired and press the *Save* button to update the changes. These changes will be effective once the *Apply Changes* button is pressed.



33.0 UPDATING FIRMWARE

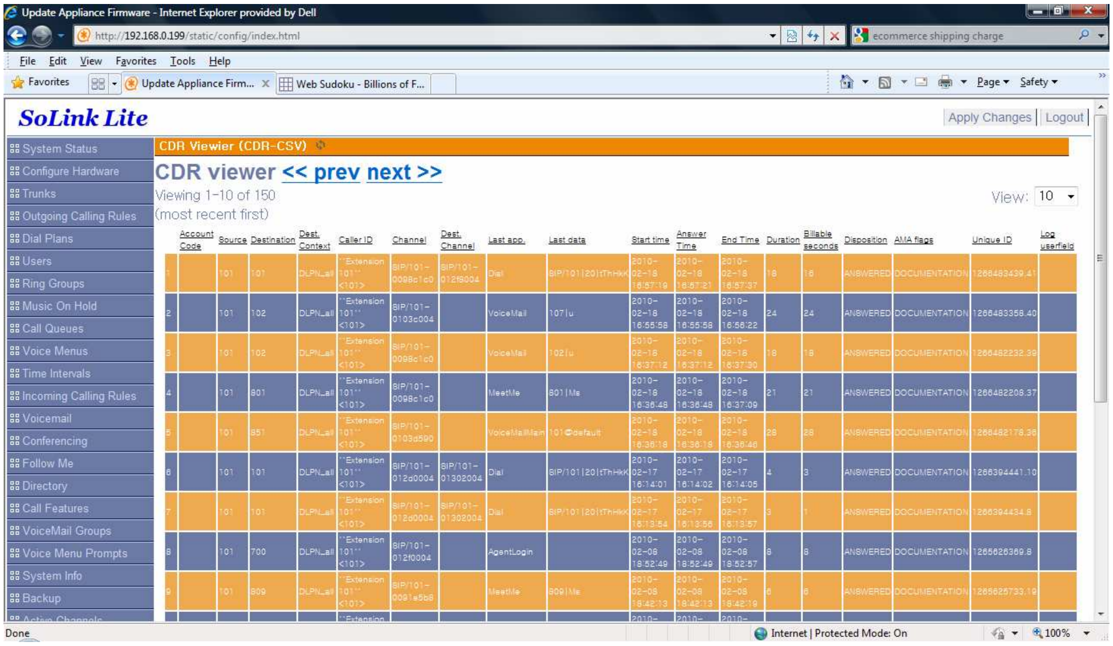
To update the firmware, select the *Firmware update* menu option.

Both HTTP and TFTP update options are available. Select the desired type of server, enter the IP address of the server, and the filename of the firmware. If *Reset Configs* option is checked, the system will reset to the default network and asterisk configuration after the upgrade. Click the *Go* button to initiate the firmware update and reboot the unit after.



34.0 VIEWING CALL DETAIL RECORDS

To view the call detail records, select the *Call Detail Records* menu option.



APPENDIX A DEFAULT SETTINGS

Default Incoming Call Flow
Route to Auto-attendant

User Extension

100 – 131	Extension with voicemail

Operator Extension

100	Operator

Voice Mail Access

851	Voicemail Access

Call Parking

870	Main Parking Extension
871 - 899	Parking Extension

Meetme (Dial-in) Conference

801 – 809	Conference Extensions

ACD

700	Call Agent Login
701	Call Agent Call-back Login / Logout

Trunk

PSTN	traditional PSTN trunk (all FXO ports)

Dial Plan

all	PSTN (prefix with 9)

Feature Keys

*8	Call Pickup
##<ext>	Call Transfer (Blind)
#2<ext>	Call Transfer (Attended)
*0	Call Disconnect
#7	One-touch Call Park

APPENDIX B SUPPORTED CODECS

Codec	Bit Rate (kbps)	Voice quality	Nominal Ethernet Bandwidth (kbps)	Maximum number of simultaneous calls		
				ADSL 1.5M/256k	ADSL 3M/640k	LAN 100Mbps
G.711	64	Excellent	87.2	2	7	1000
G.729	8	Good	31.2	8	19	3000
GSM	13.2	Acceptable	35.4	7	17	2500
Speex	4-48	Good	37.4	6	16	2500

APPENDIX C FIREWALL SETTINGS

The following ports will be used by the SoLink-Lite IP-PBX Appliance:

- UDP 5060 (for SIP registration and signaling)
- TCP 5060 (for SIP registration and signaling for some devices)
- UDP 4569 (for IAX2 signaling)
- UDP 5036 (for IAX signaling)
- UDP 20000 – 21000 (for RTP)
- UDP 4500 – 4999 (for T.38 Fax)
- TCP 80 (for web access)
- TCP 22 (for SSH access) if desired