SolSwitch™

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SolSwitch Administrator Guide

(version 2.0.140)

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System Overview

SS1000 Hardware Specifications

Form Factor: 1U Rack Mount
Processor: 2 x Intel Xeon E3 3.1GHz
RAM: 8 GB
Storage: 2 x 500 GB (Hardware RAID 1 / Mirroring)
Ethernet: 2 x RJ-45
Solid State Drives (SSD): Optional
Expansion Slots: 1
Dimensions: 16.8" x 14" x 1.7" (W x D x H)
Power: 260W / 120V Power Supply

SS1000 Power Specifications

Idle	Load
Watts: 32.7	Watts: 37
Amps: 0.189	Amps: 0.232
kVA: 0.0327	kVA: 0.037
BTU/hr: 111.57	BTU/hr: 126.24
Test Voltage: 122v	Test Voltage: 122v

SS1000 Hardware Details

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Connecting the SolSwitch

Connecting the Power

Included with the SolSwitch is a North America 120V power cable. Connect the power cable from the SolSwitch to a surge protector or UPS (uninterrupted power supply).

Connecting the Network

On the back of the SolSwitch, you will find two network interfaces. The left interface is eth1 and the right interface is eth0. By default, eth0 is configured statically with IP address: 10.10.10.1/255.255.255.0. Eth1 is configured for DHCP. Connect the interfaces to your network accordingly.

Console Access

In order to change the SolSwitch's default IP address, you will require console access. To do so, connect your keyboard and monitor to their corresponding ports on the back of the SolSwitch. To log into the console interface, ensure that the SolSwitch is powered on. When prompted, enter username: *admin* and password: *admin* and follow the on-screen instructions.

SolSwitch Behind NAT

If your SolSwitch is behind NAT and you intend to use devices on public IP addresses, it is required that you setup port forwarding on your router. The following ports should be forwarded:

UDP 5060 (SIP)
UDP 10000 to 20000 (for RTP/voice)
TCP 80 (for web and provisioning)

You will also need to see the following sections to properly configure the SolSwitch for NAT:

Main Menu → Miscellaneous
Main Menu → NAT Settings

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Accessing the Interface for the First Time

Web Interface
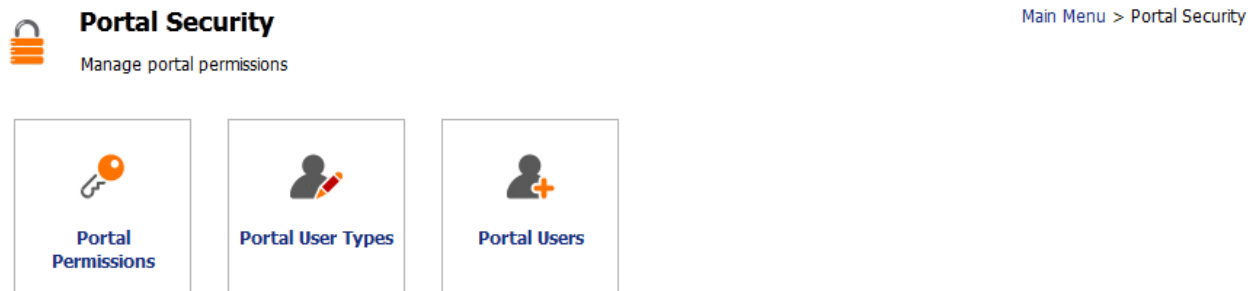
The SolSwitch interface is a web-based portal. By now, you should know the IP address of your SolSwitch. In order to access the interface, open your favourite web browser and enter the IP address in the address bar.

Login

At the login prompt, enter the username and password provided with the SolSwitch to access the Main Menu. It is recommended that you change your username and password. If you do not change the username and password, a large red bar will appear at the top of your screen upon each login until the default password is changed. See the next section for more information.

Setting Up SolSwitch Users and Permissions

From the Main Menu, click the Portal Security icon.

Portal User Types

Set your different types of users so that you can give them particular permissions.

Portal Permissions

In this section, you control permissions for each portal user type by selecting the corresponding checkboxes.

Portal Users

Add, edit or remove portal users as well as designate their portal user type. To add a new portal user, select the Add New User button. To edit or delete an existing portal user, click the corresponding icon next to the user.

Passwords require a minimum of 8 characters, with a minimum of 1 upper-case letter and a minimum of 1 digit.

When a system has multi-tenant licensing enabled, this option controls which users have access to customers other than their own.

CDR Filter limits what the user can search for on the CDR report.

Menu type options are Categorized and Uncategorized. This defines whether or not users will see sub-categories (sub-folders) when logging into the interface.

Remove CDR Recordings is an option for the user to have permission to delete CDR recordings.

Configuring the SolSwitch

From the Main Menu, click the Advanced Tools icon.

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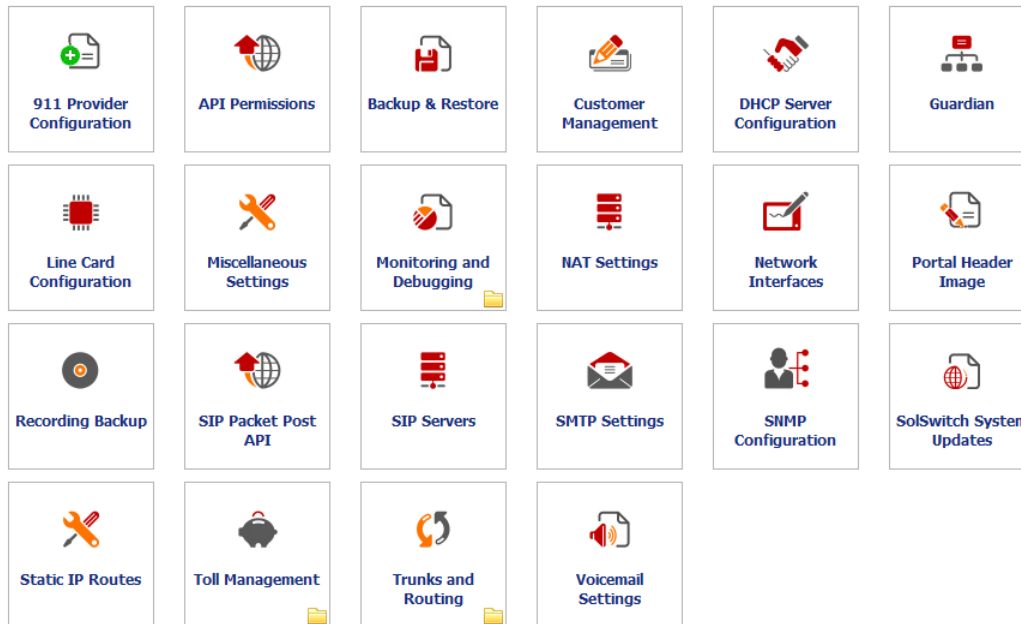
911 Provider Configuration

The SolSwitch has built-in support to control 9-1-1 address information. If you subscribe to 9-1-1 services from Northern 911 or Digital Conceptions, you will be provided with API access credentials which should be entered here.

API Permissions

The SolSwitch comes with a set of API functions available for third party software integration.

To access API Permissions, navigate to Advanced Tools > API Permissions

To view information about the API functions available, click on the "View SolSwitch API function list and usage" link at the bottom of the API Permissions page.

The API Permissions section allows you to specify who can access the API and what they have access to. Permissions are based on IP address and/or API Key.

Click the "Add New Access List" button to add a new entry, or press the Edit icon to modify an existing entry.

For each Access List you must specify the following:

- **Access List Name:** To describe the purposes of this entry.
- **Allow from IP address:** Enter the IP address to allow access. This is recommended, but only required when no API Key specified.
- **API Key:** 256-bit key used for authentication to the API. A key is required when no IP address is specified. Use the Generate Key button to generate a random key.
- **Access to Customer:** Only for multi-tenant platforms. Specify which customer it will have permissions to.
- **API Function Permissions:** Select check boxes for each function that this access list can use. For Digit Validation Log, you must also specify which DV can be accessed.

Clicking the orange question marks will give you more information about each API function.

Backup & Restore

The SolSwitch supports two methods of backing up its data. Only phone system settings, voice prompts, voicemail greetings and messages are contained in this backup. If you are using the system to record calls, see the Recording Backup section for more information.

Automatic FTP Backup: To use automatic FTP backup, you must already have access to a separate FTP server. Enter the FTP server, username, password and path to match your FTP account information. The backup frequency specifies how often (in days) the backup should be run. Backup time specifies the time of day that the backup should be run. The next backup date is the next date that the system will be backed up. This date is re-adjusted based on the selected backup frequency. Press the 'Save

Information' button to save your settings. The 'Backup Now' button performs an immediate backup to the FTP server.

Manual Backup: Manual backups are performed by clicking the 'Backup and Save' button. You will then be prompted to download and save a backup file to your hard drive.

Restoring a Backup: In order to restore a SolSwitch, you must have a backup file of the same version. Click the 'System Restore' button, select the backup file and wait while the file is uploaded and the system is restored.

Customer Management

If you have purchased the Service Provider License, this option will be available. Add, edit and remove customers in this section. The customer number, first name, last name and company fields are required.

The Max SIP Accounts field specifies the maximum number of SIP accounts allowed on a customer's account. Set a value of 0 for unlimited SIP Accounts.

The Max Trunks field specifies the maximum number of concurrent inbound or outbound calls on a customer's account. Internal calls between extensions are not considered trunk calls. Set a value of 0 for unlimited trunk calling.

Toll Plan defines how calls should be billed for this customer. See Toll Management section for more details.

Default Route Group specifies the default outbound route for calls from a customer. See Route Groups section for more details.

Recording Email specifies how the recording will be delivered in the email.

Time Zone field sets the desired time zone which will convert the times throughout the SolSwitch.

DHCP Server Configuration

Enable this option to enable SolSwitch built-in DHCP server on a specific interface. This should be used only when there are no other DHCP servers on the same network.

Guardian

The Guardian protects the SolSwitch from unauthorized access. In this section, you can view and remove blocked IP addresses as well as add IP addresses to the exemption list. The following actions will be monitored by the Guardian.

- Using proprietary techniques, the SolSwitch Guardian ensures that only authorized devices may provision on the SolSwitch.
- The SolSwitch Guardian analyses every SIP registration attempt. Upon detection of an unauthorized attempt, Guardian will automatically alert the administration and immediately block the IP address.
- The SolSwitch Guardian monitors SIP traffic to detect floods or malicious packets and will automatically block the source IP address and alert the administrator.
- The SolSwitch Guardian will detect unsuccessfully web portal login attempts and automatically block access.
- The SolSwitch Guardian enforces strong passwords to prevent easy brute force password discovery.
- The SolSwitch Guardian monitors call patterns and volumes and will alert the administrator if unusual activity occurs.

Line Card Configuration

The SolSwitch supports Digium analog and digital line cards. Line cards are configured in this section.

Miscellaneous Settings

- **Web Address:** If your system is behind a router, specify the external IP address or host name used to access the web interface.
- **Provisioning Server:** Similar to the Web Address, this field defines the external IP address or host name used to provision devices. In most cases, the Provisioning Server address is the same as the Web Address appended with /prov.
- **Time zone:** Specify the correct time zone for the location of the SolSwitch.
- **Main Email:** From address to use when sending out alerts.
- **Alert Notify Email:** Email address to send system alerts to.
- **Default NTP Server:** Where the SolSwitch should get its time.

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Monitoring and Debugging

From the Advanced Tools menu, click the Monitoring and Debugging icon.

CPU Usage: A graph that displays the system's CPU usage for any given time period. Right click the graph to zoom out or click and drag to zoom in.

Customer Usage: This icon will be displayed if the Service Provider License was purchased. Select a customer and time range in order to generate usage graphs. The graphs display inbound, outbound and internal calls. Right click the graph to zoom out or click and drag to zoom in.
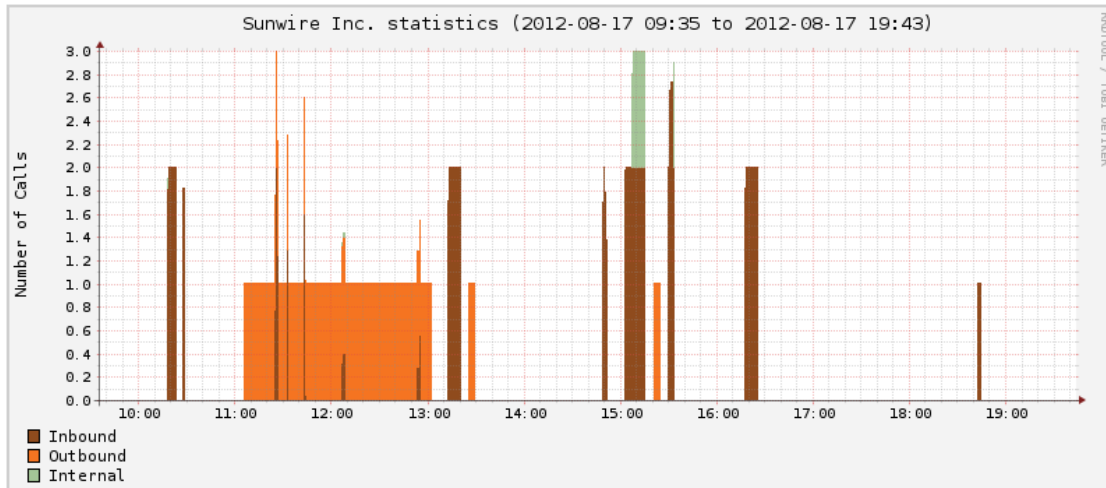
Device Statistics: The device statistics shows information about the SolSwitch, the Interface, IP Table, DNS Library, and Disk Information.

Full Log Viewer: The full log viewer gives you access to the system's full log file in order to troubleshoot specific calls and find its origin, path, route group, etc. To use this tool, follow these directions:

1. Enter whatever you would like to search for in the search field. For example, if you want to troubleshoot a call to a specific number, enter part of or the whole number. Or if you would like to look at a call to a specific device, enter the SIP username in the search field. The results outputted will reflect any reference to your search in the full log file.
2. By looking at the Date/Time, find the corresponding call that you are troubleshooting.
3. See the call in its entirety by clicking the 'ThreadID Channel'. This will change your search results to show anything within that particular Thread ID giving you full details to the entire call in question.

Memory Usage: This section shows a graph of the system memory usage over a specific date/time.
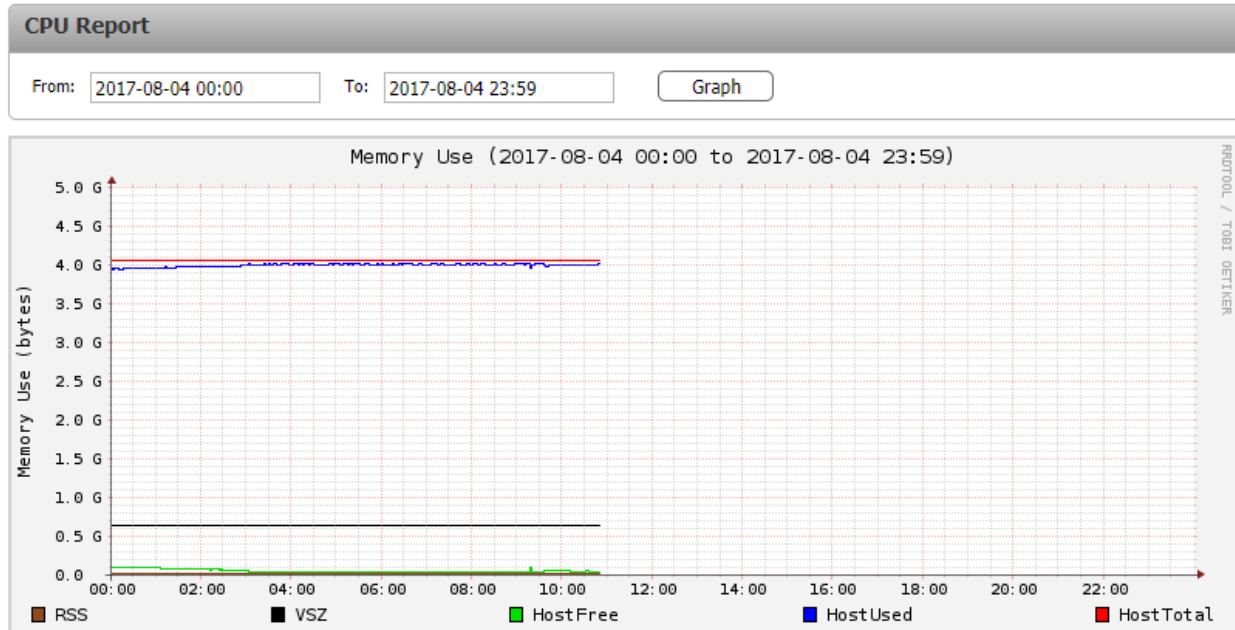
Portal Activity: This section is used to pull web interface activity logs. Data includes date, time, username, customers, IP address, feature accessed, as well as HTTP GET and POST data.
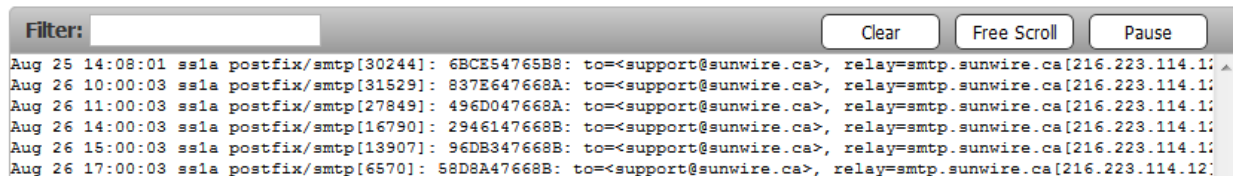
Real-Time Log: Allows the administrator(s) to view real-time call and provisioning activity on the SolSwitch. This information is refreshed every 5 seconds. Click the Clear button to clear the buffer, click the Free Scroll button to prevent the screen from scrolling as activity updates and click the Pause button to stop new activity from being displayed.

Reboot / Shutdown: This section displays all service status for every system as part of the SolSwitch deployment.

SIP and RTP Captures: This section gives administrator(s) access to raw packet traces. Traces are kept for a total of 24 hours.

- **SIP Method:** Select which method to search on
- **Filter URI (User):** Specify the user part of the URI to filter on. This is usually the phone number dialled or the username of the SIP account. Optional but recommended.
- **Filter Source/Dest IP:** These fields are used to filter source and destination IP address.
- **Search Timeframe:** Specify the timeframe to search. A maximum of 10-minute window can be searched at once in order to prevent performance impacts while traces are searched.

SMTP Log: A scrolling log of all SMTP relays including where the email came from and which address it was sent to. This report can be useful when troubleshooting email issues.

System Status: Shows current calls in progress, total processed calls and current uptime. This option also allows you to remotely restart or shutdown the SolSwitch. The Graceful Shutdown button will wait until all calls are completed before shutting down.

Trunk Report: Used to graph incoming and outgoing calls on a specific trunk. Select a trunk and time range in order to generate usage graphs. Right click the graph to zoom out or click and drag to zoom in.

Trunk Reporting and Alerting: This page allows the user to setup scheduled reports to be emailed automatically, in addition to allowing trunk alerts to be sent out based on trunk capacity usage.

Scheduled Reporting, when enabled, allows a frequency of monthly or weekly, on a specific hour, which will report from that day to the previous month/week (based on the frequency). Scheduled reporting will report on all trunks.

Alerting, when enabled, allows alerts to be sent based on how much capacity is being used. In the example below, if trunk capacity is equalled to or greater than 90% it will send an alert to the alerting email. Alerting will alert based on what is enabled, per customer per trunk. There are different alerts that will be sent based on what is enabled. Whether a call fails how many calls failed in the last hours, for which trunk and/or customer, or if calls had got over the threshold.

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Trunk Reporting and Alerting Details

Scheduled Reporting

Enable:	☐
Frequency:	Weekly ▼ Monday ▼
Time:	00:00 ▼
Email(s):	

Alerting

Trunks	☐ Enabled	Capacity	Frequency
Earth	☐	90	15
Iristel	☑	90	15
Dev71	☐	90	15
Vianet	☐	90	15
test	☐	90	15
Verizon2	☐	90	15
Verizon3	☐	90	15

Customers	☐ Enabled
SUNWIRE INC.	☑
NOT SUNWIRE	☑
ROCKTECH	☐
HORIZION	☐
TELCOMP	☐
SUNWIRE TEST LAB	☐
SUNWIRE	☐
Company	☐
FUSHINI	☐

UDP Usage: A graph that displays the amount of UDP ports being used by the system for any given time period. Right click the graph to zoom out or click and drag to zoom in.

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NAT Settings

You must complete this section if your SolSwitch is behind NAT and you intend to use a SIP device outside of the network. Specify the external IP address that should be used for SIP and voice traffic. The Internal Network section specifies all subnets that are internal to the SolSwitch. For example, if your internal network is 192.168.0.0 to .255 then you would specify 192.168.0.0 as Internal Network IP and 255.255.255.0 as the Subnet Mask.

Network Interfaces

This section is used to specify IP information for each of the SolSwitch's network interfaces as well as DNS information. You may specify whether interfaces are static or automatic (DHCP). Secondary (alias) IP addresses on a single interface is supported.

Interface Details

Interface Settings:

Interface:	eth0
Interface enabled:	◉ Yes ◯ No
Address type:	◉ Static ◯ Automatic
IP address:	10.0.201.69
Subnet mask:	255.255.255.0
Gateway:	10.0.201.1
ⓘ Default Gateway:	☐

Save Interface Settings

Interface Aliases

Additional IP Addresses:

IP address	Subnet mask	
192.168.192.1	255.255.255.0	🗎
		Add

Save IP addresses

Portal Header Image

The header image is the image that appears at the top of the SolSwitch portal. This image can be customized. Its size is restricted to 900x104 pixels and must be in a JPEG/JPG, GIF or PNG format. Image file sizes must be below 500KB.

Recording Backup

This section is used to archive call recordings to a separate server. To use this feature, you must have access to a separate FTP server. Enter the FTP server, username, password and path to match your FTP account information. The Day Limit field specifies how many days to keep recordings on the SolSwitch before archiving. After a recording has been archived, it is still accessible through the SolSwitch interface as long as it has not been removed from the archiving server.

SIP Packet Post API

SIP Packet Post API is used to call an external API any time an INVITE packet is received from a specified SIP account. To access this feature, go to Main Menu > Advanced Tools > SIP Packet Post API.

- **Enable:** Enable or Disable this feature
- **POST URL:** API URL to post to. You must prefix the URL with http://
- **API Key:** API Key to add to the post for basic authentication.
- **Fail Notify Email:** Email to notify if POST failed.

To enable API Post on a SIP account, go to Main Menu > VoIP Provisioning > SIP Accounts, select Edit for the SIP account you want to monitor and set "SIP Post API Enable" to Enabled. The change can take up to a minute to take effect.

Here is an example of the POST variables being sent to the API:

```
[APIKey] => sdfSADFAKSJLHDKasd
[SIPAccountID] => 5332
[FromIP] => 192.168.176.34
[InviteTo] => 7055551694
[RawPacket] => INVITE sip:7055551694 (packet truncated for this document)
[AttemptNumber] => 1
[PacketHash] => eed1e35af3d1530385d0983f020a22f1
```

The API must respond with the word "SUCCESS" to avoid retries. The API will be called up to three times if it doesn't receive the response of "SUCCESS". The PacketHash is unique for every INVITE therefore it will stay the same throughout the three attempts.

SIP Servers

If the SolSwitch can be accessed from multiple hostnames or IP addresses, you would specify them here.

SMTP Settings

The SolSwitch sends out emails for alarm notifications, voicemail to email and other features. Specify the SMTP Server to use to send email. Enter the username and password for the SMTP server if the server requires authentication.

SNMP Configuration

The SolSwitch allows the setup of an SNMP community; this is used for an external SNMP service to poll the SolSwitch.

SolSwitch System Updates

When Software Assurance is purchased, your SolSwitch will come with a System Update feature. Under Advanced Tools, this option allows you to check for new SolSwitch versions and update your system accordingly, giving it access to the latest features and bug fixes.

Static IP Routes

If you have multiple subnets or gateways on your network you may need to specify static routing. Use this section to specify static routes.

Toll Management

To use this option, you must have purchased the Service Provider License. This section is used to configure call accounting plans.
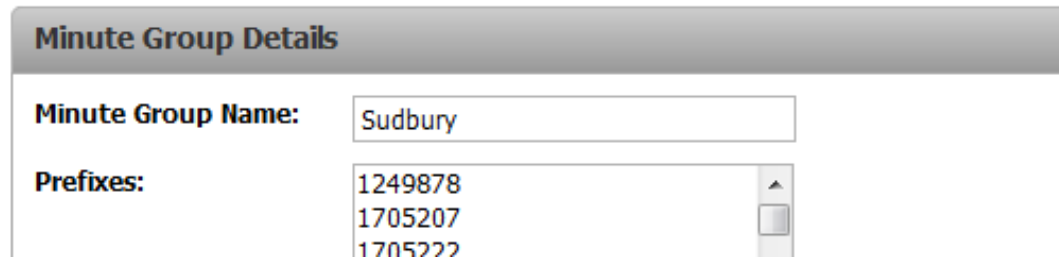
Toll Rate Lists: Specify the names of each individual rate list that you will be offering customers. For example, retail or wholesale.

Toll Destinations: If you have already specified toll destinations, click the Export to CSV button view or edit the data. If this is your first time using this feature, a CSV file must be created and uploaded. For this you may use any spreadsheet application. The CSV must be formatted as follows:

	A	B	C	D	E
1	Area	Prefix	WSCost	Retail	Wholesale
2	Afghanistan - Areeba Mobile	0119377	0.3045	0.3959	0.307545
3	Afghanistan - At Mobile	0119375	0.3141	0.4083	0.317241
4	Afghanistan - AWCC Mobile	0119370	0.3308	0.43	0.334108
5	Afghanistan - Etisalat Mobile	0119378	0.2943	0.3826	0.297243

The first row must be a header row. The first column specifies the Area. The second column specifies the area's calling Prefix. The third column is for informational purposes only but is still required. This column defines your cost for the area. All remaining columns must match the rate lists specified in Toll Rate Lists.

Toll Minute Groups: This section is used to specify areas that may be included in a plan at a special rate. In other words, the Toll Minute Groups will override the previously uploaded Rates List when applicable. To create a minute group, simply choose a name and add all prefixes on individual lines.

Toll Plans: Toll plans are used to create plans that will be assigned to customers. Each plan requires a name and a rate list. All minute groups will be listed here under the Minute Groups section. These fields are used to specify the number of minutes allowed in a given month to that area without incurring the associated rate list costs. If 1,000,000 or more minutes are specified, the system will treat it as unlimited calling. The following options are also available on this page:

Billing Increment: The increments, in seconds that will be used to calculate billing. The first number is the minimum billing time, and the second is the increments to be used after the minimum. For example, with 30/6, a 10 second call will be billed as 30 seconds and a 31 second call will be billed as 36 seconds.

Unlimited Local Calls: Local calls will be determined using the extension Caller ID and rate centers set up on the system. If the external number is not in the same rate center, it will be considered a long-distance call.

Alert Email(s): When an email is provided, an alert will be sent in the scenario that an extension caller ID is invalid or does not match any existing rate centers. Alerts will only be sent if Unlimited Local Calls is enabled.

Create Rate Centers: When selected, if an extension Caller ID does not match an existing rate center, a new rate center will be automatically created. If an email is entered above, an alert will be sent to notify that a rate center was created. This option is only valid if Unlimited Local Calls has been enabled.

Toll Reporting Options: This section is used if you plan to provide PDF or hard-copy reports to customers directly from the SolSwitch. The Name field refers to the company name as it will appear on the report. The Logo specifies the company logo to display on the report. Logos must be JPEG/JPG, PNG

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or GIF format and must be 340 x 110 pixels. The Address Line specifies the company address as it should appear on the report.

Toll Reporting: Use this section to pull reports regarding call accounting. Types of report are as follows:

- Summary by Customer: Display all customers and total monthly charge
- Detail by Customer: Display all customers and all individual calls for the month
- Summary by Extension: Display a summary of all customers and extensions by month
- Detail by Extension (CSV): A CSV report for individual or all extensions
- Detail by Extension (PDF): A PDF report for individual or all extensions

Trunks and Routing

This section is used to configure trunks as well as inbound and outbound routing.
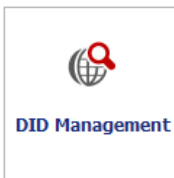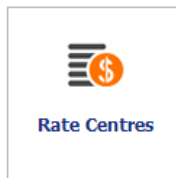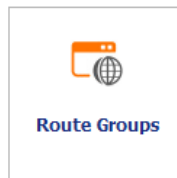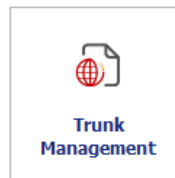
Trunk Management: Trunks are used to connect the SolSwitch to another SolSwitch, PBX or provider. The following options are available for every trunk:

- Trunk Name: Specify the name of the trunk
- Description: Information purposes only
- Call Limit: The trunk will be limited to this many calls. Enter 0 for unlimited.
- Record: Enable this option to record all calls passing through this trunk.
- Incoming Calls
    - o Use Local Routing: If this option is selected, all incoming calls will be compared to the local extension table for termination. This option is usually used when connecting the SolSwitch to another PBX. If there is no match, the system will proceed to 'Use DID table'.
    - o Use DID Table: If this option is selected, all incoming calls will be compared to the local DID table for termination. If there is no match, the system will proceed to 'Use Route Group'

- o Use Route Group: If this option is selected, calls will be routed according to the route group specified. If there is no match, the call will be rejected as invalid number.
- o Digit Manipulation
    - ▪ Add Prefix: The string specified in this field will be prefixed to any incoming DID.
    - ▪ Strip Digits: The system will strip this many digits from the beginning of any incoming DID.
- Outgoing Calls
    - o Use E.164 Numbering: The international standard for phone number formatting. Enabling this will make sure that numbers follow the format.
    - o Unlimited Outbound Calls: enable the trunk to have unlimited outbound calls as long as it is under the call limit (if it is set).
- Allow Dynamic Conf: Allow trunk to use # codes to place conference calls.
- Accessibility: When using SolSwitch with Load Balancing, this option specifies which SolSwitch has access to this trunk.
- Apply toll to trunk: enable calls passing through this trunk to be processed by toll billing.
- Trunk Type: Specifies whether the trunk is SIP or a physical line card. The next option specifies SIP or line card options.

DID Management: Use this section to add or remove DIDs on the SolSwitch. DIDs can be added as a single DID or a range. Rate Center, Description and Notes fields are for informational purposes only. The Customer field specifies the customer to assign the DID to. This field will only be displayed if the Service Provider License was purchased.

The Active field specifies whether or not the DID is active. You also have the ability to strip caller ID name and number from a DID as soon as it arrives onto the system. This is flagged per DID – keep in mind that Caller ID information will still be available in trace files.

Rate Centres: Rate centres are used in route groups (discussed next). This section allows for outbound routing to be area based rather than prefix based. Rate centre information is automatically downloaded from LocalCallingGuide.com. To add a Rate Centre, click the Add New Rate Centre button. Simply enter the Name for the Rate Centre along with any NPA and NXX (Area Code and Exchange) in that area.

To update Rate Centre data, simply click the Edit button on any existing Rate Centre and click the Update Rate Centre button. This will ensure that Rate Centre prefixes are kept up to date.

Route Groups: Route Group tables are used to handle call routing. To add a new Route Group, select the Add New Route Group button. If this will be the default route group, click the check box. A check mark will be displayed next to your default route in the Route Groups list. An option is available for an

incoming call to skip to the next trunk in the Route Group if the current one is busy. To edit an existing route group name, click the Edit icon. To manage a route group, click the Manage icon.

Adding Routes: Each entry in the Route Group table is capable of supporting unlimited routes. To add a new route, click the 'Add New Route' button. The following must be specified:

- Type: Specify whether the SolSwitch should route based on prefix or rate centre.
- Route Prefix: Specify the prefix to match on the dialled number.
- Match Length: Specify the full length of the number to match. Use 0 to match any length.
- Strip Digits: Strip this many digits from the beginning of the dialled number before routing.
- Dial Prefix: Prefix the following string to the dialled number before routing.
- Route To: Specify whether to route to a trunk or another route group table.

Adding Failover Routes: If the primary route fails a failover route will be attempted. To add a failover route, click 'Edit' on an existing Route and then click the Add New Route button. Specify the following information and click the 'Save' button:

- Strip Digits: Strip this many digits from the beginning of the dialled number before routing.
- Dial Prefix: Prefix the following string to the dialled number before routing.
- Route To: Specify whether to route to a trunk or another route group table.

Editing Routes: When editing a route, two additional options become available:

- Require Pass Code: Specifies whether or not a user must enter a pass code before a call is completed to this route. Pass codes are specified per extension (see Extensions section for more details).
- Force Trunk Caller ID: If enabled, this option allows caller ID name and number to be overwritten when a call is placed to this route.
- Managing Priorities: While editing a route with multiple entries, you will be given the option to adjust priorities. Priorities specify the order that the routes are tried.

Add/Edit Route Priority

Prefix	Match Length	Strip Digits	Dial Prefix	Trunk/Route Group	Priority	Edit	Del
1	11	3		RG: 999Route	1	Edit	Del
1	11	0		T: Earth	2	Edit	Del

Add New Route   Save Priorities

Voicemail to Email Settings

This section allows control on how voicemail email messages are generated. Attach Message specifies whether or not the audio file should be attached to the email. The emails body can be configured using the variables provided. Below is an explanation of what each voicemail variable represents:

${VM_NAME} – Mailbox owners name
${VM_DUR} – Length of message
${VM_MSGNUM} – Message number
${VM_MAILBOX} – Mailbox the message is in
${VM_CIDNUM} – Caller ID number
${VM_CIDNAME} – Caller name
${VM_CALLERID} – Caller ID Info ex: "John Smith" <5551234466>
${VM_DATE} – Date the message was left
${VM_TIME} – Time the message was left

Global Voicemail Settings

Attach Message:	Yes
Format:	wav
Max Logins:	3
Email Address:	noreply@sunwire.ca
From Name:	SolSwitch
Email Subject:	New voicemail
Email Body: Variables: ${VM_NAME} ${VM_DUR} ${VM_MSGNUM} ${VM_MAILBOX} ${VM_CIDNUM} ${VM_CIDNAME} ${VM_CALLERID} ${VM_DATE} ${VM_TIME}	Hi, ${VM_NAME}! You have a new message from ${VM_CALLERID}

Save Voicemail Settings

Solutions that shine

VoIP Provisioning

Provisioning is a way of remotely programming specific settings on a device. Provisioning is supported on the following device types: Mitel/Aastra, Cisco, Linksys and Polycom. Provisioning is available via HTTP or HTTPS. Please contact Sunwire for additional information regarding configuring SSL certificates.

VoIP Provisioning
Manage VoIP devices

Main Menu > VoIP Provisioning

Device Provisioning | Directory Profiles | Feature Profiles | SIP Accounts
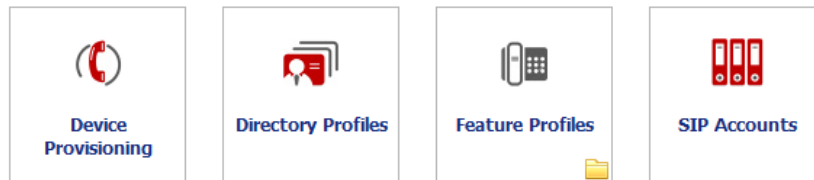
Feature Profiles

Feature profiles are a way to configure specific settings on an individual or on multiple devices. Once feature profiles are created, they must then be selected and attached to the appropriate device in the Device Provisioning section. To add a new feature profile, select the proper device brand and click the 'Add New Profile' button.

Feature Profiles
Feature profiles for different phone types

Main Menu > VoIP Provisioning > Feature Profiles

Aastra Feature Profiles | Cisco Feature Profiles | Grandstream GXV 3350 Feature Profiles | Linksys Feature Profiles | Polycom Feature Profiles | Yealink Feature Profiles
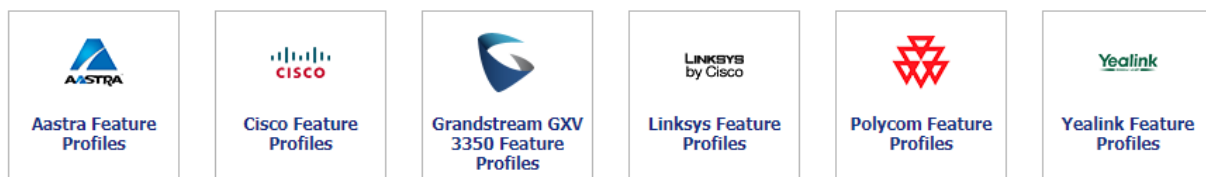
Polycom Feature Profile: The following options are available for the Polycom feature profiles.

- Profile Name: For informational purposes only
- Display Logo on Phone: Whether or not to display the SolSwitch logo on the phone.
- Default Mode: Specify whether handset or headset should be used by default
- Show Missed Calls: Specify whether or not the phone set should display missed calls.
- Enable DND: Allow or disallow Do-Not-Disturb functionality on non-VVX series phones.
- Quick Transfer: Enables the ability to blind transfer someone while on a call.

- Show Queue Stats: This Enables the display of stats on the phones MicroBrowser
- Call Waiting Type
  o Ring: Phone will ring while on a call and receiving an incoming call.
  o Beep: Phone will beep while on a call and receiving and incoming call.
  o Silent: Phone will flash and display incoming call details on LCD.
  o Disabled: Call waiting will be disabled.
- Direct Call Ring Tone: The ring tone to play when a call is routed directly to your extension.
- Queue/Ring Group Ring Tone: The ring tone to play when a call is received via a queue or ring group.
- Soft Keys: Up to 10 soft keys can be specified. Soft keys are used to display specific functions available depending on the state of the device.
  o Name: Label displayed on phone LCD (if supported)
  o Function: Each key can serve the following function
    ▪ Applications: Used for accessing the phones Application page. (Usually the Micro Browser)
    ▪ Buddies: Displays contacts and their current status. Contacts are managed in Directory Profiles. See Directory Profiles section.
    ▪ Callers: Directs user to the callers list
    ▪ Conference: Initiate a conference call.
    ▪ Custom: When specified, the custom text box will appear where a value can be specified. When pressed, the corresponding DTMF for the value will be initiated.
    ▪ End Call: Ends the current call.
    ▪ Hold: Puts the current call on private hold.
    ▪ Hook Flash: Sends a hook flash signal to the remote end. Only supported when remote end is an analog or T1 port.
    ▪ Issue: Allows the ability to report a call issue.
    ▪ Intercom: Pages an extension and places your call on their speaker phone.
    ▪ Login/Logout: Toggles the login/logout roaming feature.
    ▪ New Call: Initiates the dial tone to place a new call.
    ▪ Page: Defaults to extension 8000 which can be setup as a paging group.
    ▪ Park: Places the current call on park.
    ▪ Ready Toggle: Toggles the phone's Do-Not-Disturb status.
    ▪ Record: Records the current call from the beginning.
    ▪ Redial: Directs user to the Outgoing callers list.
    ▪ Speed Dial: Activates the phone's speed dial list.
    ▪ Transfer: Initiate a transfer.
  o Custom: Internal or external number in relation to the specified Function.

- o States: Select the state(s) at which the soft key should appear on the phone.
  - Idle: Display the key when the phone is not in use.
  - Dial tone: Display the key when there is dial tone.
  - Proceeding: Display the key when outbound call is in progress but not yet connected.
  - Alerting: Display the key when an incoming call is ringing.
  - Active: Display the key while on an active call.
  - Hold: Display the key when on private hold.
- o Sorting Soft Keys: Soft keys can be sorted by dragging and dropping the key up or down in the list. Once you are satisfied with your sort order, select the 'Save Ordering' button.

Edit a Polycom Feature Profile

Profile Name: [Main]

General Settings

Display logo on phone: ○ Yes ● No

Default mode: ○ Headset ● Handset

Show Missed Calls: ● Yes ○ No

Enable DND: ● Yes ○ No

Notifications

ⓘ Call Waiting Type: [Ring ▼]

ⓘ Direct call ring tone: [Low Trill ▼]

ⓘ Queue/Ring group ring tone: [Loud Warble 2 ▼]

Mitel/Aastra Feature Profile: You are able to copy one Aastra profile to another, within the list view, as well as import and export Aastra feature profiles. The following options are available for the Aastra feature profiles.

- Profile Name: For informational purposes only
- Default Mode: Specify whether handset or headset should be used by default
- Number of line keys: The number of line keys that should be available on the phone. If 4 is selected, the redial and callers list buttons will be replaced with lines 3 and 4
- Show missed calls: Choose whether or not the number of missed calls should display on the screen
- UI Ring Focus: When on a call, if a call comes in the phone screen can either continue to show the current call or show the information for the incoming call
- Enable Call Waiting Beep: When enabled, the handset will beep when receiving a call while already on a call
- Enable More Key: Choose whether the 'more' key should display on the screen once the softkeys are full (ex, 4 bottom soft keys) or only once there are too many to fit on the screen at once (ex, 5 bottom soft keys)
- Remove Own Extension: When set to yes, the user's own extension will be removed from the Top Soft Keys when provisioning to the phone. This is compatible with the "Own Ext" option.
- Phone Language: The language displayed on the phone screen. Supports English and French.

- Top Soft Keys: Up to 44 top soft keys may be specified. The actual number available will vary based on phone model, additionally presence can only be set on top soft keys and not the bottom soft keys.
  o Name: Label displayed on phone LCD (if supported)
  o Function: Each key can serve the following function
    - BLF (Busy Lamp Field): This option is used to provide live status of other extensions on the system.
    - BLF/Xfer: This option is the same as above, although pressing the button will call the specified Value.
    - Callers List: If used, when the button is pressed, phone call logs will be displayed.
    - Speed Dial: Used to initiate a call to the specified value.
    - Speed Dial/Xfer: This option is the same as above, although it can also be used to transfer an active call.
    - DND: Sets the phone to Do not Disturb.
    - Redial: Directs user to the Outgoing callers list.
    - Directory: Directs user to the directory list.
  o Value: Internal or external number in relation to the specified Function.
- Bottom Soft Keys: Up to 20 bottom soft keys can be specified. The actual number available will vary based on phone model. Soft keys are used to display specific functions available depending on the state of the device. Presence can only be set on top soft keys and not the bottom soft keys.
  o Name: Label displayed on phone LCD (if supported)
  o Function: Each key can serve the following function
    - BLF (Busy Lamp Field): This option is used to provide live status of other extensions on the system.
    - BLF/Xfer: This option is the same as above, although pressing the button will call the specified Value.
    - Callers List: If used, when the button is pressed, phone call logs will be displayed.
    - Speed Dial: Used to initiate a call to the specified value.
    - Speed Dial/Xfer: This option is the same as above, although it can also be used to transfer an active call.
    - DND: Sets the phone to, Do not Disturb.
    - Redial: Directs user to the Outgoing callers list.
    - Directory: Directs user to the directory list.
  o Value: Internal or external number in relation to the specified Function.

- o States: Select the state(s) at which the soft key should appear on the phone.
  - Idle: Display the key when the phone is not in use.
  - Connected: Display the key when a call is in progress.
  - Incoming: Display the key while an incoming call is ringing.
  - Outgoing: Display the key while an outbound call is ringing.
  - Busy: Display the key when the phone is in DND state.

Cisco Feature Profile: The following options are available for the Cisco feature profiles.

- Profile Name: For informational purposes only
- Branding: Used to display the companies brand name on the phone.
- Show Missed Calls: Specify whether or not the phone set should display missed calls.
- Enable DND: Allow or disallow Do-Not-Disturb functionality.
- Enable Call Waiting: Whether or not to allow incoming calls while phone is in use.
- Enable Directory Key: Whether or not to display the Directory soft key.
- Enable Park Key: Whether or not to display the Park soft key.

- Enable Record Key: Whether or not to display the Record soft key.

Yealink Feature Profile: The following options are available for the Yealink feature profiles.

- Profile Name: For informational purposes only
- Show Missed Calls: Specify whether or not the phone set should display missed calls.
- Enable Call Waiting: Whether or not to allow incoming calls while phone is in use.
- Call Waiting Tone: Whether or not to enable a tone when a there is a call waiting
- Call ring tone: To select from 8 different ring tone's
- Display logo on phone: Whether or not to display logo on the phone.
- Default mode: To set the default mode for which to use the phone, Headset or Handset
- Enable DND: Allow or disallow Do-Not-Disturb functionality.
- Bottom Soft Keys
  - Statuses: Will display soft keys based on the phone's current status.

- o Soft Keys: Displays available functions based on the phone's current status
- o Functions:
  - Call Failed
    - Cancel/Switch Call/New Call
  - Call In
    - Call Switch/Answer/Forward/Silence/Reject
  - Connecting
    - Cancel/CallSwitch
  - Transfer Connecting
    - Cancel/Call Switch/Transfer
  - Dialling
    - Cancel/Send/IME/Delete/History/Call Switch/Directory/Line Selection/Pool/Group Pickup/Direct Pickup
  - Ring Back
    - Cancel/Call Switch /Call Completion
  - Transfer Ring Back
    - Cancel/Call Switch/Call Completion/Transfer
  - On Talk
    - Cancel/Mute/Swap/Call Switch/New Call/Answer/Reject/Hold/Private Hold/Transfer/Conference
  - Hold
    - Cancel/Call Switch/Answer/Reject/Transfer/Resume/New Call
  - Held
    - Cancel/Call Switch/Answer/Reject/New Call
  - Transfer to
    - Cancel/Directory/Call Switch/Send/Transfer/IME/Delete
  - Conferenced
    - Cancel/Call Switch/Answer/Reject/Mute/Hold/Split

Edit a Yealink Feature Profile

Profile Name: YealinkProfile

General Settings

Display logo on phone (T2XX):	⦿ Yes ◯ No
Default mode:	◯ Headset ⦿ Handset
Show Missed Calls (T2XX):	⦿ Yes ◯ No
Enable DND (T2XX):	◯ Yes ⦿ No

Notifications

ⓘ Call Waiting:	⦿ Yes ◯ No
ⓘ Call Waiting Tone:	⦿ Yes ◯ No
ⓘ Call ring tone:	Ring 1 ▼

Soft Keys

Item	Status	SoftKeys		
1	Call Failed	Call Switch ▼		📄
		Cancel ▼	📄	📄
		New Call ▼	📄	📄
2	Call In	Answer ▼		📄

Linksys Feature Profile: The following options are available for the Linksys feature profiles.

- Profile Name: For informational purposes only
- Show Missed Calls: Specify whether or not the phone set should display missed calls.
- Enable Call Waiting: Whether or not to allow incoming calls while phone is in use.

Solutions that shine

Grandstream Feature Profile: The following options are available for the Grandstream feature profiles.

- Profile Name: For informational purposes only
- Custom Layout File: Grandstream offers a configuration tool to customize the look and feel of the phone. Upload the file here and it will be applied to all phones with this feature profile. When set, there is an Export button along the top that can be used to download the current layout which can be used for other profiles.
- Enable Call Waiting Tone: Sets the phone system to play the call waiting tone if there is another incoming call.
- Always Ring Speaker: Determines if the speaker will play the ringtone if the speaker channel is not set as default channel.
- Enabled Missed Call Backlight: If set to "Yes", LCD backlight will be turned on when there is a missed call on the phone.
- Enable Missed Call Indicator: If set to "Yes", the LED indicator on the upper right corner of the phone will light up when there is missed call on the phone
- Headset Type: Choose the preferred headset type when answering calls.
- Date Format: Choose the format will be shown on the display.
- Enable Video Call: Enables the video call feature on the phone.
- Enable Transfer: Enables the Transfer function
- Start Video Automatically: Permits the phone system to enable the video feature automatically when it makes an outbound call.
- Remote Video Request: Configures the preference to handle video request from the remote party during an audio call. "Prompt" :A message will be prompted if a video request is received. Users can select "Yes" to establish video or "No" to reject the request. "Accept": Video

request will be accepted automatically, and video will be established. ⬚ "Deny": Video request will be rejected automatically.

- Video Layout: Defines whether to enter full screen when incoming video call is answered.
- Auto Preview: Configures whether to turn on video to preview the video of the caller. If set to "Yes", the user can view the video and hear the caller on the incoming page when there is an incoming call. If set to "Yes with Ringing", the caller can view the video of the caller and hear the ringtone on the incoming page but cannot hear the caller.

Grandstream GXV3350 Feature Profile Details

Export Current Layout

Profile Name:	Grandstream
ⓘ Custom Layout File:	Choose File No file chosen
	Uploading a new file will override the current layout
Enable Call Waiting Tone:	☑
Always Ring Speaker:	☐
Enable Missed Call Backlight:	☑
Enable Missed Call Indicator:	☑
Headset Type:	Normal Headset ▾
Date Format:	MM-DD-YYYY ▾
Enable Video Call:	☑
Enable Transfer:	☑
Start Video Automatically:	☑
Remote Video Request:	Prompt ▾
Video Layout:	Default ▾
Auto Preview:	No ▾

Save Profile Settings

Directory Profiles

Directory Profiles are pre-defined speed dial lists. Once directories are created, they must then be selected and attached to the appropriate device in the Device Provisioning section. To add a new Directory Profile, select the 'Add New Profile' button. To edit or delete an existing profile, click the corresponding icon.

To quickly create similar directories, click the "from" and "to" radio buttons to copy from and paste to another directory.

Import Directory Profile

It is also possible to import a directory through a CSV file. Download the CSV template and fill it out with the required information. If later on you wish to move some items around, you can edit the directory, and move items around by dragging the left icon.

Directory CSV Importer

Importing to: Sunwire Main

CSV: [Choose file] No file chosen   [Upload]

☐ Remove all existing records.

Please **download** the CSV template to fill out and upload back using the form above.

Note: The CSV file **must** have the header line as this tool will always skip the first line in the uploaded file.

The CSV file has some requirments on it listed below:

- Column A is optional - Entry Name
- Column B is required - Entry Contact Number
- Column C is optional - Buddy Watch

Adding/Editing a Directory Profile

- Profile Name: For information purposes and there is an option for Auto Update Profile which recognizes if the name or extension number of an entry in the directory profile is updated or deleted anywhere else in the system. The directory will then update its entries to reflect the changes.
- Add New Directory Listing: To add a new directory listing, click the 'Add New Entry' button.
  - o Label: Text to appear on phone next to button.

- o Contact Number: Phone or extension number
- o Next Speed Dial Index: Order in which the key will be displayed
- o Buddy Watch: Whether or not the SolSwitch should monitor the status of this extension. Only specified on internal extension numbers.
- Sorting Directory Listings: To sort listings in a directory, use the icon to the left of the label to drag and drop the listing in the correct order.

SIP Accounts

A SIP (Session Initiation Protocol) Account is used to authenticate a SIP device such as a telephone or ATA (Analog Telephone Adapter) to the SolSwitch. Any new device must first be added as a SIP account. To add a new SIP account, click the 'Add New SIP Account' button.

- Description: For information purposes only
- SIP Host Type:
  - o Dynamic: Most common authentication-based method used for IP phones or ATA.
    - Username: Specify the username required to authenticate to the SIP Account. SIP Account Usernames cannot contain all digits if it is equal to or less than 11 characters.
    - Password: Specify the password required to authenticate to the SIP Account. Strong passwords are required.
    - Allowed IPs: Allow IPs to use the SIP account
    - Blocked IPs: Block IPs from using the SIP account
  - o Static: Most commonly used for trunks when the SIP peer has a static IP.
    - Name: Specify the name without spaces for this peer
    - Host: Static IP address or host name of the peer
    - Port: Port number that will be used.
    - Domain (optional): Specify the domain name of the SIP registrar.
  - o Register: When the SolSwitch must register to an outside dynamic SIP account.
    - Name / BTN: Name or Billing Telephone Number for this peer
    - Registrar: Remote SIP server IP or host name
    - Domain: Specify the domain name of the SIP Registrar
    - Username: Specify the username for the remote SIP server
    - Password: Specify the password for the remote SIP server
    - Verify Password: Ensure the password was typed the same both times.
- Transport Type: Whether this SIP account should use TCP or UDP. The default is UDP.

- NAT: Whether or not this SIP device will be behind NAT. If enabled, the SolSwitch will always respond to the source IP address and port of the received SIP packets.
- DTMF Mode: Choose inband or rfc2833 in order to ensure settings match SIP device.
- Available Codecs: Choose which codecs should be allowed on this SIP account. The T.38 codec requires G.711 codec to be enabled as well. For video SIP devices, select one or all H.XXX codecs.
- Dynamic Features: Allow incoming calls to send in particular features.
- SIP Post API Enable: Enables the SIP Post API to send a packet capture.
- Alarm Notification: Specify an email address here to receive notifications when the peer changes from online to offline and vice-versa.
- Delete Block: Place a check mark here in order to prevent this peer from being deleted. To delete the peer, the check mark must first be removed.
- Enable Debugging: When enabled, the SolSwitch will ping the peer once per second until the feature is disabled. Ping results may be viewed in graph format under Main Menu → Reporting and Monitoring → SIP Ping Monitor Report.
- Status: The status is displayed for each peer. The following status levels may be displayed:
  - OK: Peer is okay, SIP packet response time displayed in milliseconds.
  - Unreachable: Peer was online but is now unreachable.
  - Offline: Peer is offline.
  - Lagged: SIP response times are in excess of 2 seconds (2,000ms).
- IP: The IP address of the SIP device.
- Trunk: The trunk the SIP account is using.
- Extension: The extension the SIP account is attached to
- Reboot Phone: Allows the remote reboot of the phone.

Device Provisioning

Adding a SIP Device: Click the 'Add Device' button. Select the proper device type from the dropdown menu. See next step.

Adding/Editing Device Details:

- Device Info
  - Name: For information purposes
  - MAC Address/Identifier: The MAC address or identifier of the device you are adding to provisioning.
- Provisioning
  - Authentication: Whether HTTP authentication should be used or not.

o   Username: HTTP authentication username to use.

o   Password: HTTP authentication password to use.
The following encryption options are only support on Cisco SPA-5XX phones.

o   Encryption: Encryption can be disabled or enabled.

o   Encryption Key: When encryption is enabled, the encryption key (string) specified here must be used when provisioning the Cisco phones. When provisioning the Cisco phones to use encryption, you must log in to the Cisco phone's profile rule must be specified as such:

Profile Rule: [--key ENCRYPTION-KEY-HERE]

*Ensure the encryption key is prefixed with [--key and ends with a]

- Profiles
  o   Feature: Select the feature profile for this device. See Feature Profiles section for more details.
  o   Directory: Select the directory profile for this device. See Directory Profiles section for more details.

- SIP Server Info
  o   Primary: The primary SIP server that this device should register with.
  o   Secondary: The secondary SIP server that this device should register with. Please note that some devices (such as Polycom) will register to both SIP servers simultaneously while others (such as Aastra) will only fail over if the primary is unavailable.

- Time Settings (if available)
  o   NTP Server (optional): Specify an NTP server to sync time with.
  o   GMT Offset: Specify time zone for the device.
  o   Time Format: Display in a 12 or 24 hour format.

- Line Keys
  o   Label (if available): Label to display on the phone next to line key
  o   Show Ext # (if available): Whether or not to append the phone's extension number to the label.
  o   SIP Account: Specify the SIP account that should be attached to each of the device's line keys.

Viewing Provisioning Status: Click on the 'Refresh' icon on the corresponding device to view its provisioning status. Administrator(s) also have access to real-time provisioning logs under Main Menu → Advanced Tools → Monitoring and Debugging → Real-Time Log.

Solutions that shine

Quick Refresh					✖
Name	**Ident/Mac**	**Type**	**Last Provisioned**	**Last IP**	**Last Status**
Jason 601	0004f21afda4	Polycom 601	2013-01-08 09:33:02	10.0.201.120	Success

Refresh

Provisioning a Mitel/Aastra IP Phone

Configuring the SolSwitch

See VoIP Provisioning section before configuring a phone.

Configuring the Phone

Use the following directions to configure an Aastra/Mitel phone:

1. Press the settings button
2. Press Advanced (Password 22222, and enter)
3. Select Configuration Server
4. Choose Download protocol HTTP
5. Change HTTP Server to your provisioning server as defined in Main Menu → Advanced Tools → Miscellaneous Settings
6. Change HTTP Port to 80 and HTTP Path to prov
7. Select Save
8. Reboot the phone

Rebooting a Phone

Software changes to a phone may require a reboot. Using the settings button on the phone, press Settings → Restart → Yes. A reboot can also be made through the web portal: Main Menu → VoIP Provisioning → SIP Accounts → Edit.

Solutions that **shine**

Provisioning a Polycom IP Phone

Configuring the SolSwitch

See VoIP Provisioning section before configuring a phone.

Configuring the Phone

Use the following directions to configure a Polycom phone:

9. Press the Menu Key
10. Select Settings
11. Select Advanced (Password 4-5-6, and enter. For V5.6 Polycoms, enter 6-5-4)
12. Select Admin Setting
13. Select Network Configuration
14. Scroll down using the arrow keys and select Server Menu
15. Change Server Type to HTTP
16. Change Server Address to your provisioning server as defined in Main Menu → Advanced Tools → Miscellaneous Settings
17. Select Back twice
18. Select Save & Reboot

Rebooting a Phone

Software changes to a phone may require a reboot. Using the menu button on the phone, press Menu → Settings → Basic → Restart Phone. "Are you sure?" will appear, and select Yes. After time, the IP address will appear with the phone MAC address and phone software revision. A reboot can also be made through the web portal: Main Menu → VoIP Provisioning → SIP Accounts → Edit.

Provisioning a Cisco SPA-122 ATA

Configuring the SolSwitch

See VoIP Provisioning section before configuring an ATA.

Configuring the ATA

Use the following directions to configure a Cisco SPA-122 ATA:

1. Connect a PC to the ATA Ethernet port
2. Connect to the ATA's web interface using the default web address (192.168.15.1)
3. Login using the default username and password (admin/admin)
4. Open the 'Voice' tab
5. Open the 'Provisioning' page
6. Set the profile rule to the following: http://**Client Provisioning Address**/spa$MA.xml
   - o **If you are using Encryption, the profile rule would have to be entered as such:**
     [--key **EncryptionKey**] http://**Client Provisioning Address**/spa$MA.xml
7. Click 'Submit Changes'

Rebooting an ATA

Software changes to an ATA will require a reboot. Unplug the power cable and plug it back in to initiate a reboot.

Provisioning a Grandstream ATA

Configuring the SolSwitch

See VoIP Provisioning section before configuring an ATA.

Configuring the ATA

Use the following directions to configure a Grandstream ATA:

1. Connect a PC to a Router
2. Connect the ATA to the same Router as the PC.
3. Connect an analog phone to the ATA, and wait for a dial tone.
4. Dial *****02** and record the IP address
5. Open the ATA configuration page, by typing that IP address into a web address-bar
6. Login with "admin"
7. Go to advanced settings
8. Under Firmware Server Path

- o Enter: **Client Provisioning Address**/prov
9. Under Config Server Path
   - o Enter: **Client Provisioning Address**/prov
10. Save and reboot the device.

Rebooting an ATA

Software changes to an ATA will require a reboot. Unplug the power cable and plug it back in to initiate a reboot.

Provisioning Other Devices

Please see specific manufacturer documentation for instructions on how to provision the device. See provisioning server address in Main Menu → Advanced Tools → Miscellaneous Settings

Phone System Settings

The Phone System Settings section is where the majority of the PBX and phone features are configured.

Solutions that **shine**

911 Information
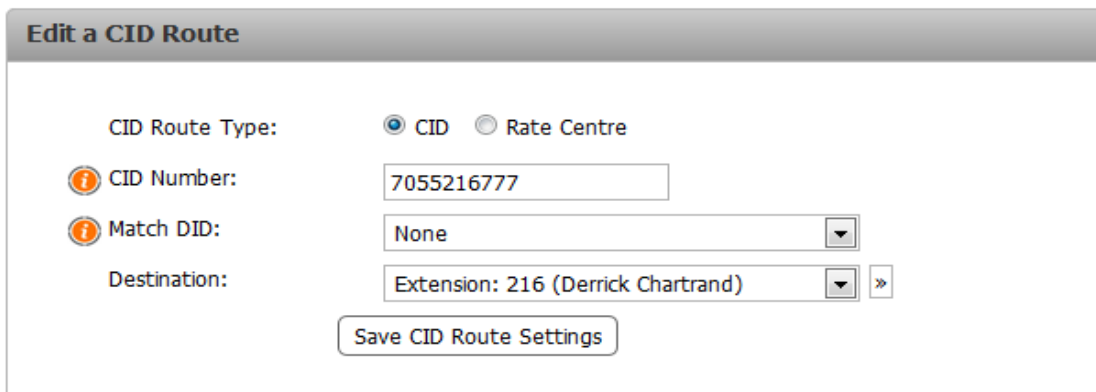
This section should only be used when 911 Provider Configuration has been setup under Advanced Tools. Every DID configured will be displayed here with the ability to create, edit and delete that DIDs 911 record. The records are instantly synched with the 911 provider via their API. This information is used when an extension dials 911. The 911 call taker will see the information attached to their CID number or E911 CID number if one is specified.

Caller ID Routing

Caller ID Routing allows you to route incoming calls based on the CID number or the rate centre when rate centres have been configured in the Advanced Tools under Trunks and Routing.

Adding a CID Route: To add a CID route, click the 'Add CID Route' button.

- CID Route Type
    - o CID: Creates the routing based on the incoming CID number.
    - o Rate Centre: Routes the call based on the Rate Centre the call is coming from.
- CID Number/Rate Centre: If the type chosen is CID, then enter the 10 digit CID number you would like to match. If the type chosen is Rate Centre, select the rate centre you would like to match from the dropdown menu.
- Match DID: This dropdown menu will display all of the existing DIDs. If you want to route a call no matter what company DID was dialled, then set this option to 'None'. Selecting a specific DID will only route the call when that particular DID is called from the CID number/Rate Centre (whichever type is chosen).
- Destination: Select the location you would like the calls to be routed to.

Edit a CID Route

CID Route Type:	⦿ CID ○ Rate Centre
ⓘ CID Number:	7055216777
ⓘ Match DID:	None ▾
Destination:	Extension: 216 (Derrick Chartrand) ▾ »
	Save CID Route Settings

CID Override

The CID Override feature allows you to override incoming caller ID information on the system. The feature works by identifying the CID Number you would like to be modified on incoming calls and assigning a new name and number to that CID information. Anytime an incoming call of the CID Number is made on the system, the SolSwitch will automatically replace the Caller ID information when it gets to the phone.

In the following example, anytime the number 7055216777 called into the SolSwitch, whichever phone is set to ring would display: 7055550000: OFFICE

DID Routing

This section allows you to route incoming calls for each of your DIDs. If you do not see the DID you would like to route, ensure it has been added under the Advanced Tools' Trunks and Routing menu.

In order to route a single DID, click on the 'Edit' button next to that particular DID. From there, simply choose where you would like that DID to be routed to from the dropdown menu. Multiple DIDs can also have their routing modified by using the checkboxes to the left of the DID. Simply select any combination of DIDs and hit any 'Edit' button in order to change the routing on all the selected DIDs.

Extensions

Extensions are assigned to agents/employees of the phone system and can be spread across multiple telephone sets if required. Extensions can be individually added using the 'Add New Extension' button or they can be imported from a CSV file using the 'Import from CSV' file. Once all the extensions have been created, an extension list can also be exported to CSV using the 'Export to CSV' button.

Importing Extensions from CSV: To mass import extensions, along with voicemail and SIP accounts. Simply download the CSV template from the Extension CSV Import page. Fill it out accordingly and upload it back to the system. Below is a list of each column of the CSV file and what information is held in the field.

Exporting Extensions to CSV: There is an option to export all existing extension located on the list extensions page, which allows a user to import the CSV to another system or customer.

Solutions that **shine**

CSV Extension Importer

CSV: [Choose file] No file chosen   [Upload]

Please **download** the CSV template to fill out and upload back using the form above.

Note: The CSV file **must** have the header line as this tool will always skip the first line in the uploaded file.

The CSV file has some requirements on it listed below:

- Column A is required - Extension Number
- Column B is required - Extension Name
- Column C is optional - CID Name
- Column D is optional - CID Number
- Column E is optional - E911 CID Number
- Column F is optional - Directory Searchable (0 for NO, 1 for YES)
- Column G is optional - Default NPA
- Column H is optional - Wrap Time (default: 0)
- Column I is optional - Forward Time (default: 25)
- Column J is optional - Record Call (1 for Always, 2 for On Demand)(default: 2)
- Column K is optional - Record Notify Email
- Column L is optional - Roaming Passcode
- Column M is optional - VM Extension Number
- Column N is optional - VM Forward Email
- Column O is optional - VM to Email Only (0 for NO, 1 for YES)(default: 0)
- Column P is required when M is filled
- Column Q is optional - SIP Username.
- Column R is required when column Q is filled.
- Column S is optional - Extension Active (0 for NO, 1 for YES)(default: 1)

Additional information on what each of these fields do can be found below.

Note: The CSV file <u>must</u> have the header line. The tool will always skip the first line of the CSV file.

Adding Individual Extensions: Individual extensions can be added using the 'Add New Extension' button. Below is a description of each field asked for on the Edit Extension page as well as a screenshot of what a completed extension may look like.

- Extension Active: Choose whether or not the extension is active. If an extension is not active, it cannot be dialled.
- Extension Number: The number to dial in order to access the extension.
- Name: Enter the name of the extension. This name will be used to search the extensions through the directory.

- Directory Searchable: Enabling this will allow callers to find this extension when performing a search using the directory feature.
- Caller ID Name: Enter the caller ID name that will be displayed on external calls.
- Caller ID Number: Enter the caller ID number that will be displayed on external calls.
- E911 Caller ID Number: If 911 Information has been configured, all calls made to 911 will show the 911 provider the number specified in this field.
- CID Internal Override: Use this field to override caller ID number on internal calls. If this field is left blank, the extension number will be used for caller ID number on internal calls.
- Always use Caller ID: By default, a route group can override caller ID. Enabling this option will ensure that the caller ID specified in this extension will always be displayed.
- Default NPA: The default 3-digit area code. This area code will be prefixed on all 7-digit calls.
- Route Group: Specify the route group to use on this extension. Route Groups are setup under Trunks and Routing of the Advanced Tools menu.
- LD Pass code: This pass code will be required to place any long distance call. Password requirements are configured in Route Group settings. If no pass code is entered, this extension will have the ability to place long distance calls.
- Time Rule: Select a specific time rule if this extension is only available during certain times. Time Rules are configured in the Phone System Settings menu.
- Closed Destination: If a time rule is set and its status is closed, calls will be sent to the location specified here.
- Join Message: You can select a System Recording from this list and callers will hear the sound upon reaching the extension.
- Wrap time: After a call is hung up from this extension, no calls will ring this extension until the Wrap time has elapsed. Wrap time is specified in seconds.
- Forward All Calls: You can choose to forward all incoming calls to a specific internal location or any external number. This can also be done from the phone itself using the appropriate *-code.
- Forward Busy or N/A: Where to forward calls to when an extension is busy or not answered.
- Time before forwarding: Specify the amount of seconds before forwarding to the Forward Busy or Forward No Answer location. One ring typically takes 5 seconds.
- Route No Ans. Transfers: When this extension initiates a call transfer, the call will be routed to this location if the destination of the transfer is busy or does not answer. This option overrides the Forward Busy and Forward No Answer settings.
- Record Calls: Can either be set to On Demand or Always. Always will record all calls on that extension whereas On Demand will require the user to press the 'Record' button or #1 during a call.

- Record Notify Email: If a call is recorded, a copy of the recording will be sent to the email address specified here.
- Call Waiting: If enabled, this extension will be able to receive new calls while on an existing call.
- Display Caller ID: Specifies whether or not caller ID is displayed on incoming calls.
- Roaming Pass Code: Entering a pass code in this field allows this extension to be a roaming extension with the ability of logging in and out of that extension from any SIP device. If no pass code is entered, the extension must be logged in manually by specifying its SIP device.
- Distinctive Ring Direct Calls: This allows the extension to ring differently if a direct call is received
- Can Monitor Calls: This option should only be enabled for managers and will allow this extension to monitor calls of others.
- Privacy Enabled: If enabled, this extension cannot be monitored by extensions that have Call Monitoring turned on. Also, all recorded calls will not be made public in the Call Detail Records report; they will only be emailed to the address specified in the Record Notify Email field.
- Paused Presence will show the extension as busy on other phones when paused.
- Block Internal Calling will prevent the extension from calling any other extensions on its own customer.
- SIP Device: The device to attach this extension to. This will not always be required if an extension is set to be a roaming extension.

Extensions Details

Extension Active?	Yes ▼
Extension Number:	108
Name:	John Smith
Dial-by-Name Searchable:	○ Yes ◉ No
Caller ID Name:	testing
Caller ID Number:	7055216777
E911 Caller ID Number:	
CID Internal Override:	
Make CID Private:	○ Yes ◉ No
Always use Caller ID:	○ Yes ◉ No
Default NPA:	245
Route Group:	Use Default ▼ »
LD Passcode:	
Time Rule:	None ▼ »
Closed Destination:	None ▼ »
Join Message:	None ▼ »
Wraptime:	0
Forward All Calls:	○ Off ○ Internal Destination ◉ External Number
Forward External:	7056985571
Forward Busy/DND/Out:	Voicemail: 1290 (sawyers mailbox) ▼ »
Forward No Answer:	None ▼ »
Forward On SIP Offline:	None ▼ »
Time before forwarding:	25 seconds ▼
Route No Ans. Transfers:	None ▼ »
Record Calls:	On Demand ▼
Record Notify Email:	johnsmith@sunwire.ca
Call Waiting:	Enabled on All Calls ▼
Display Caller ID:	Yes ▼
Roaming Pass Code:	
Distintive Ring Direct Calls:	◉ Yes ○ No
Can Monitor Calls:	○ Yes ◉ No
Privacy Enabled:	○ Yes ◉ No
Paused Presence:	◉ Yes ○ No
Block Internal Calling:	○ Yes ◉ No
SIP Device:	00085D686A0D (00085D686A0D) ▼ »
Parking Lot Group:	Group 1 (50 - 98) ▼
XMPP Account:	

Speed Dial Settings:

Use Global Profile:	None ▼ »
Private List:	Manage

Save Extension Settings

Extension Speed-Dial Settings

Extensions -Dial Keys can be added and managed from your telephone set or through the portal. Speed dials can be from 1 – 999. All speed dials are preceded by '0'. Therefore, to call speed dial 1,

you would simply dial: 01. Below are instructions on setting your speed dial through the phone as well as through the web-portal.

Speed Dial Settings (through phone)

- Configuring your speed dial settings are simple:
  1. Dial: 00
  2. Enter the speed dial number to edit followed by the pound key.
  3. Enter the phone number to be assigned to this speed dial followed by the pound key.
  4. You will hear a recording stating: "Speed Dial Saved."
  5. You will then be brought back to the main menu in order to add more speed dials.
- If you select a speed dial which already has an entry, you will be given three dial options.
  1. Edit the existing entry.
  2. Delete the existing entry.
  3. Playback the number assigned to the existing entry.

Speed Dial Settings (through web-portal)

- Use Global Profile: This dropdown menu is a list of existing directory profiles that can be used for speed dial settings. For more on directory profiles, see the VoIP Provisioning → Directory Profiles section.
- Private List: Here is where you can manage your private speed dial list. You can use a global profile in conjunction with the private list. If there are any duplicate entries between the two, the private list will take priority. Ensure to always add a '0' before your speed dial. Adding a 'w' will wait for a pickup before entering the next digits.

Fax to Email

Fax to Email is a feature where incoming faxes to a specified number will be converted to PDF and emailed to a particular address or addresses. In order to add a Fax-to-Email, there must be an unassigned DID available.

Fax Switch

The Fax Switch feature is for a DID that shares a Fax and a Phone; the fax feature will check and route according to whether the incoming caller is a phone or a fax. Assign a Fax extension, a phone extension, and an extension for the Fax switch, then apply it to the DID in question.

Find-Me Follow-Me

The SolSwitch has the ability to look for a user by calling a succession of extensions or numbers while keeping the incoming call on hold. You can have as many find-me numbers as you like. In order to add a Find-Me Follow-Me route, click the 'Add New' button. When a Find-Me number is called, the user is given the option to accept or reject the call. Rejecting immediately goes to the next number in the list or the overflow (whichever is set to be next).

- Extension Number: Specify the Find-Me Follow-Me's extension number.
- Name: Enter the name you would like to use for the Find-Me Follow-Me. This is for reference purposes only.
- Caller ID Type: Check whether you would like the Type to be pass through or Specified
- Caller ID Name: Enter a name to display when dialling external numbers.
- Caller ID Number: Enter a number to display when dialling external numbers.

- Follow Me Numbers: The numbers that should be called. Multiple numbers can be added to this field and they will all sequentially be called in the order that they appear.
- Overflow: In the event that none of the attempted numbers were answered, the Find-Me Follow-Me will overflow to the location specified here.
- Time Rule: Set hours of operation for the Find-Me Follow-Me.
- Closed Destination: If the time rule is closed this option will send calls to specified destination.
- Step Time: The amount of time, in seconds, before attempting the next Find-Me number in the list.
- Join Message: Whether or not to playback the incoming status message before the system starts the Follow-Me steps. This message reads the following:
  - "The person you are calling is unavailable. I will try to locate them for you. Please hold while I try to locate the person you are calling."
- Ask for Name: If set to Yes, the caller will be asked to record their name so that it can be announced to the recipient of each of the Follow-Me numbers when answered.
- Proceeding Message: This option allows for a message to play while a calling is being placed.
- Unreachable Message: If all of the Follow-Me steps have been exhausted or the call recipient has selected not to be reachable, an unreachable status message will be played back to the caller.
- Record: If checked, the system will automatically record the call if it is answered.

Interactive Voice Response (IVR)

IVR, also known as auto attendant, is a computerized phone system that enables a person, typically a telephone caller, to make a selection from a voice menu. The selection is made using keypad entries. Dialling an existing extension at any point while in the IVR will send the caller to that particular extension.

The phone system plays pre-recorded voice message and the person on the call typically presses a series of digits on the telephone keypad in order to select the correct option associated with the voice prompt. IVRs will often point to each other in order to drill down into a menu. When creating a new IVR, follow the instructions below.

- Extension Number: Enter an available extension number for your IVR.
- Name: Enter a name you wish to associate with your IVR for reference purposes (i.e. Daytime IVR, Closed IVR, French IVR, etc).
- System Recording: Select the pre-recorded System Recording you would like to use for when people enter the IVR. See the System Recording section for more details on how to setup a System Recording.
- Time Rule: Each IVR can be set to have its own time rule in order to route calls based on whether or not a Time Rule is open or closed.
- Closed Destination: The call will be sent to this location if a time rule is specified and that rule is closed. If there is no time rule, this field should be set to 'None'.
- Timeout: The length of time before your message is replayed or the time before the caller is sent to the Overflow Destination.
- Overflow Destination: If a caller does not select any of the available options, the call will be sent to the overflow destination following the timeout time.
- Max Repeat: If no available option is selected by the caller, the System Recording will be replayed this many times on the same interval that is specified in the Timeout field. Once the message has been repeated the specified number of times, it then goes to the Overflow Destination.
- Enable Dial-By-Name: This will allow users to press the pound (#) key at anytime during the IVR in order to access the company directory and allow people to search for a person by dialling any combination within their name.
- Dial-By-Name Language: The Dial-By-Name feature supports English and French languages.

Once all these options have been filled out, hit the 'Save IVR Settings' button. Once the IVR is saved, a new box to the right called 'Menu Options' will appear. This menu is used to enter dial options and route calls based on the option selected by the caller. Valid options are 0 to 99 only; excluding the parking lots.

API Digit Validation

API Digit Validation (DV) is similar to an IVR except that the location that the call is routed to is determined by an external API.

For example, a user calls into the DV and is asked to enter their telephone number. The digits entered by the caller are sent to an external API for validation. The call is then routed to the desired location based on the response from the API.

The fields to configure a DV are as follows:

- **Extension Number:** Enter an available extension number for your DV.
- **Name:** Enter a name to describe the function of this DV.
- **System Recording:** Select the pre-recorded System Recording you would like to use for when a user enters the DV. See the System Recording section for more details on how to setup a System Recording.
- **Digit Length to Validate:** You may pre-validate the digits entered based on the length. For example, a valid phone number would be 10 to 11 digits. Therefore, for a phone number, in this section you would enter "10 through 11". Any other digit length entered will be rejected and the user will be asked to try again until "Retry Attempts" is reached.
- **URL of API Command:** This is the URL that the SolSwitch will call in order to validate the digits. This URL must be prefixed by http:// or https://. The method used to call the API is POST. The following variables will be sent:

- o Extension Number – Extension number of the DV
- o CDRCallID – CDR Call ID can be used to obtain additional information about the call
- o CIDNum – Caller ID number of the caller
- o CIDName – Caller ID name of the caller
- o DID – DID that the caller used to dial into the system
- o Digits – Digits that were entered which require validation
- **API Failure Alert Email:** In the event that the API does not respond, or does not provide a valid response, send an alert to this email address. You may enter multiple email addresses by separating them with commas. See "API Response" section below to understand how the API should respond.
- **Retry Attempts:** In the event that the user does not enter the correct digit length, this option specifies how many times to retry. This option is also used to specify how many times to retry the API in the event that it does not respond.
- **Overflow Destination:** Enter the location where the call should be sent in the event that the API did not respond correctly, or when the response does not match specified responses. See "API responses" below for more information.

API Responses

Once a DV is saved, a Behaviour section will appear on the right side. Use this to configure how to handle API responses.

The external API should be programmed to respond with a single word in clear text. In the Response section, you would enter all the possible words and specify the location to route each word.

For example, if we wanted to validate a phone number, we may have two responses. The first response may be "SUCCESS" which would route to the Main IVR. The second response may be "FAIL" which would route to another DV indicating that the user entered an invalid number.

All calls sent to a DV are logged in a special DV Log table. The log may be queried by using the DV log API function. See Advanced Tools > API Permissions for more information about how to use the SolSwitch API.

Meet-Me Conference (Basic)

This feature allows you to create a conference room which allows multiple people to call in and join a phone meeting. People can join a conference room by dialling the extension associated with the room. A DID can also be routed directly to a Meet-Me room in order to avoid having callers enter an extension number. Meet-Me Rooms have the option of being grouped together. Pointing a DID to a Meet-Me Room that is part of a group will allow you to join the desired room by simply entering the user or administrator PIN for that room. A basic Meet-Me room allows anyone to enter with the same pass code (with the exception of the administrator). Meet-Me rooms that are part of a group will require PINs to be unique throughout all rooms in that group. Anytime a new PIN is saved in any room in any group, a check is performed in order to prevent conflicts. In order to create a Basic Meet-Me Room, click the 'Create Conference' button and select 'Basic' as the Conference Type.

- Extension Number: Enter an available extension number for your room.
- Name: Enter a name you wish to associate with your meet-me room for reference purposes.
- Pin: Enter a pin that users must enter in order to join the conference.
- Administrator Pin: Anyone who logs into the conference using the Administrator Pin will have additional permissions such as muting users, removing users and locking the room.
- Listen Only: Listen only mutes all regular users so that only the administrators may talk. This can also be done by the administrator using *96 in a meet-me room to mute all participants and *97 to un-mute all the participants.

- Lock Room: If checked, this will lock the meet-me room so that no other participants may enter or join.
- Require Admin: This option will not allow any meet-me room to begin until an administrator has joined. Users waiting for the administrator will hear music on hold.
- Announce Participants: If selected, all users who enter a Meet-Me Room will be prompted to record their name followed by the # key. When this person enters the meet-me room, all others in the meet-me room will hear a recording stating: "[Name] has joined."
- Group: When a DID is pointing to a Meet-Me Room that is part of a group, users can join the proper Meet-Me Room simply by entering the unique user or administrator PIN for that group. Grouping allows for multiple Meet-Me Rooms to have the same PIN as long as they are in different groups. Grouping options are as follows:
  o Ungrouped: Default; the meet-me room does not belong to any group.
  o New Group: If you wish to create a new group for that meet-me room to belong to, select this option and a text-box will appear allowing you to create a Group name.
  o Select from Existing Groups: Any previously created group will show up. If the meet-me room is to belong to an existing group, simply select it from the list.
  *Groups will be removed from the existing groups list when they no longer hold any Meet-Me Rooms.
- Max Participants: Specify a limit of participants to have in the Meet-Me Room.
- Record: Meet-Me Rooms can be recorded in their entirety. There are three options for this feature:
  o Disabled: No recording will occur on this call.
  o Once: Will only record 1 occurrence of the meet-me room. If the meet-me room is currently in progress, only that call will be recorded. Any calls thereafter, the record option will be changed to 'Disabled'. If no call is currently in progress, then only the next call to take place will be recorded.
  o Always: All calls in this meet-me room will be recorded.
- Record Email: If the record option above is enabled the email addresses entered here will receive a notification message with an encrypted download link to the call recording of the entire Meet-Me room. Multiple emails can be added by separating them with a comma.

Meet-Me Conference (Advanced)

This feature allows you to create a conference room which allows multiple people to call in and join a phone meeting. Advanced allows users and administrators to have their own pins if they choose to. In order to create an Advanced Meet-Me Room, click the 'Create Conference' button and select 'Advanced' as the Conference Type. Advanced rooms are setup similarly to Basic rooms although Pins are not specified until after the room is created. Therefore, in order to configure Pins in an Advanced Meet-Me Room, users must enter all the necessary information and select the 'Save Conference' button. Once this is done, a new table will be visible where conference users can be specified as well as their Pin and permission level.

Meet-Me Conference (Feature Codes)

There are feature codes available for Meet-Me conferences. These options include: mute/un-mute options, lock/unlock options, participant count options and record options. Please consult the Telephone Feature Codes section of this manual under CALL FEATURES for a complete list.

Miscellaneous Settings

This section sets 2 different settings. The first is Operator Destination.

Operator Destination: When a device from inside the phone system dials '0', this is where the call will be routed. This is useful for hospitality industries where users dial 0 to reach an internal operator.

The second setting here is a 911 Notification tool.
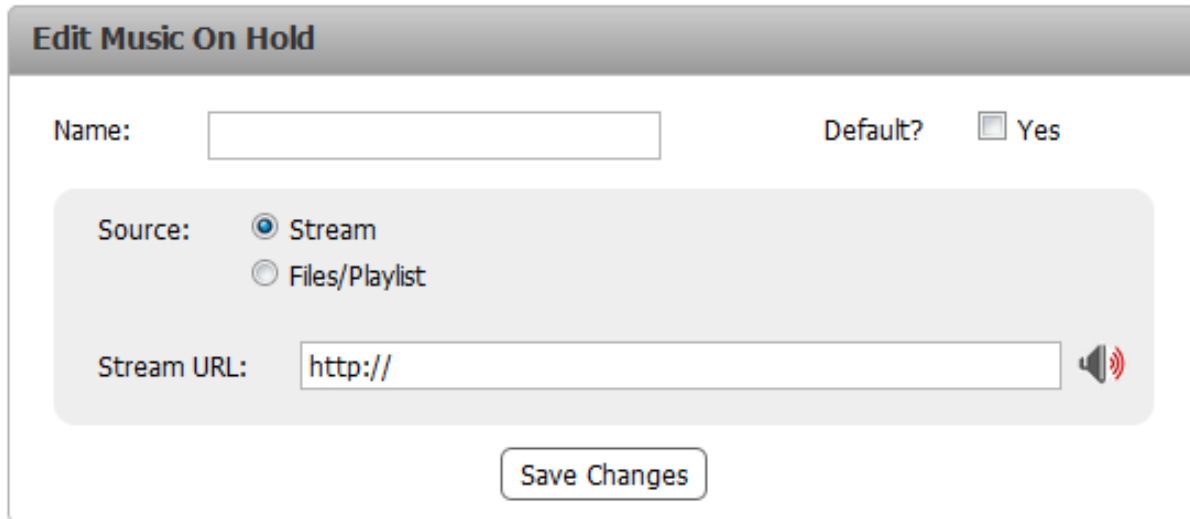
911 Dialled Notification: This setting allows you to configure the system so that if anyone dials 911, a notification is sent out to a destination set here which announces a custom recording from the System Recordings section. The recording would typically be something similar to: "911 was dialled from extension number:" The custom recording is followed by the extension number that dialled 911.

Music on Hold

Music on Hold (MoH) is the music that callers will hear when they are put on hold or parked. The three MoH options are the default royalty-free music, the ability to select your own stream from the internet as well as the option to upload a custom playlist supporting WAV or MP3 files. You can have multiple Music on Hold classes in the event that different types of music should be played at different times. Click the 'Add New Class' button to add different Music on Hold classes.

- Name: For reference purposes.
- Default: If checked, this class will be used as the default unless otherwise specified (in a queue, extension, etc.)
- Source: Select whether this class will be a stream or a playlist.

Adding a Music Stream: If the source is set to 'Stream', you will have the ability to enter a URL as well as listen to the stream to ensure its functionality. Once this is done, hit the 'Save Changes' button.
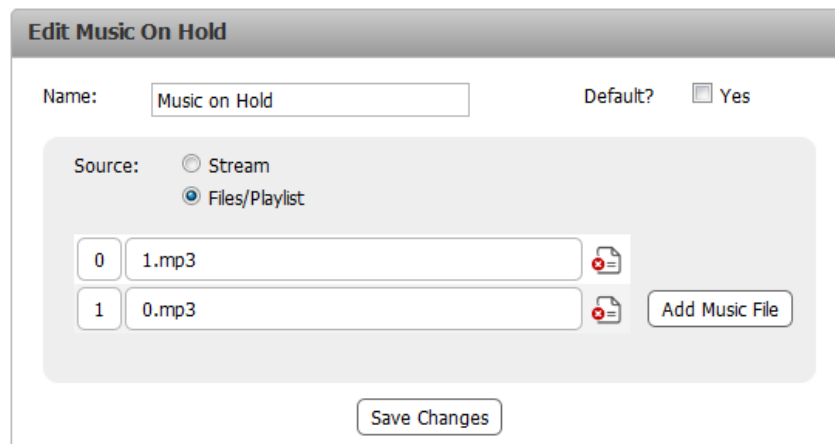
Adding a Music Playlist: If the source is set to 'Files/Playlist', you must first save the class before uploading files. Once the class has been saved, you will have the ability to upload files in WAV or MP3 format. Only one file can be uploaded at a time.
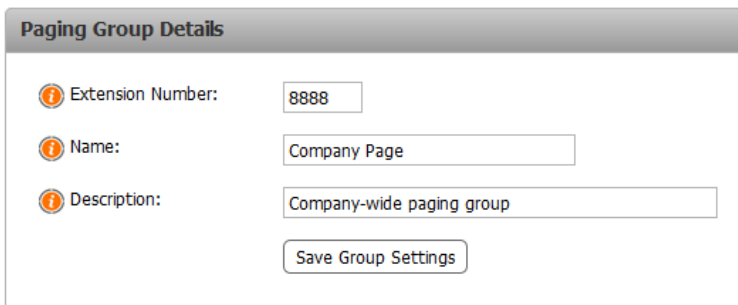
Paging Groups

Paging Groups can be set up to target specific persons or positions within the phone system. For example, if a caller is looking for a sales representative, but no one person in particular, a page can be sent to the entire sales team. Paging groups are often tied to a paging system. Add a paging group by clicking the 'Add Paging Group' button.

Adding a Paging Group: Before adding members to a paging group, it must first be setup and saved by entering the following information.

- Extension Number: The extension number to dial in order to activate that paging group.
- Name: For reference purposes.
- Description: For reference purposes.

Once the paging group has been saved, you will have the ability to add members to the group. Members are added by selecting their extension from the dropdown menu and clicking the 'Add' button.

Parking Lot Details

This section is where you configure the extension range that you would like associated with parking lots.

- Parking Lot Range: Set the range of allowed parking lots (recommended start: 50)
- Reminder Interval: The time between each park reminder.
- Reminder Type: Sets the reminder type.
    - Reminder: The phone will simply remind you that a call is parked in a specific parking lot.
    - Ring Back/Answer: The phone will ring back to you and give you the option to answer the parked call.
- Expiry Time: This option allows you to lock a parking lot for a set amount of time. To allow parking and picking up parks without the parking lot shifting to a new parking lot.

Queues

A phone queue allows calls to sit on hold until a phone agent is ready to handle the call. Queues follow rules for inbound calls and are used as a way to handle incoming calls. Many options are made available so that incoming calls are handled exactly as you need them. Queues can easily be manipulated to suit any system and ensure a better chance of never missing a call. To create a new queue, select the 'Add

New Queue' button from the Queue List. Queue forwarding is also available using the * code *104 followed by the extension number to be forwarded to. Following is a list of details and their functionality associated with a queue as well as an image of the page layout.

Creating a Queue:

- Extension Number: Enter an available extension number for your queue.
- Name: Enter a name you wish to associate with your queue.
- Weight: The weight of the incoming call refers to how much priority to put on calls in this queue compared to other queues.
- Caller ID Prefix: This feature allows agents to know which queue the call is coming from. For example, you could enter 'Sales:' so that the agent knows the call is originating from the Sales Queue.
- Join Recording: Specify if you would like a message to be played when a caller first joins the queue. These messages are configured in the System Recordings section.
- Time Rule: Select a specific time rule if this queue is only available during certain times. Time Rules are configured in the Phone System Settings menu.
- Closed Destination: If a time rule is set and its status is closed, calls will be sent to the location specified here.
- Forward All Calls: This allows you to forward all calls that go to a queue to a specific internal location on the switch. If it is to go to an external number, simply use the Virtual Extension feature.
- Maximum Hold Time: The total amount of time in the queue before the call will be sent to the overflow destination.
- Overflow Destination: Where to send the call when no agents are logged into the queue or when the maximum hold time is reached.
- Ring Members For: How long to ring the agent before trying the next agent in the member list.
- Retry After: Sets the length of time to wait between retrying queue members.
- Service Level: Used for reporting how many seconds before a call is answered and is considered as acceptable service level.
- Wrap time: Length of time before the agent is presented to take another call after finishing a call in this queue.
- Forced Exit Destination: This directs the system how to respond if a customer dials '1' for immediate assistance.
- Ring Strategy: Decides how to distribute calls within the queue. There are three available options:
    - Ring All: Ring all agents in the queue at the same time.
    - Least Recent: Ring the agent that has been waiting the longest for a call.
    - Round Robin: Distribute calls in a circular fashion.

- Fill Calls: If enabled, the queue will attempt to send calls to all available agents at once if there are enough calls in the queue. Compatible with Least Recent and Round Robin ring strategies.
- Hold Type: Whether callers should hear a ringing tone or music while holding in the queue.
- Music on Hold: If the hold type is set to music, this specifies which music on hold to play. Music on Hold is configured in the Phone System Settings.
- Auto-Pause Members: If this option is turned on, an agent will be set to 'Not Ready' if they don't answer a call. They will be required to go back to 'Ready' in order to be presented with subsequent calls. To toggle between Ready and Not Ready, dial: *908.
- Record Calls: Specifies whether calls in this queue should always be recorded or if they should only be recorded on Demand. To record calls on demand, users must either dial: #1 or hit the 'Record' soft-key if it is available.
- Show Prefix for Internal Calls: If this option is turned on, the queue prefix will be displayed when an internal extension calls the queue.
- Allow Callback: This feature allows a caller holding in a queue to press '2' while waiting and a 'Queue Callback' will be created. This means that the caller can hang up and go about their business while keeping their place in the queue. When their call arrives to the front of the queue, the system automatically calls the person back and connects them with the agent.
- Callback Dial Prefix: This field only accepts digits and will prefix the queue Callback number being dialled.
- Callback CID Name: Select the name to display as the Caller ID Name when the system calls the customer back.
- Callback CID Number: Select the number to display as the Caller ID Number when the system calls the customer back.
- Callback Attempts: Select how many times to re-try a callback after an unsuccessful attempt.
- Callback Attempt Interval: Select how long to wait between each dial attempt.
- Callback Fail Notify Email: If a callback fails all of its attempts, a notification will be sent to the email address entered here. You can enter multiple email addresses separated by a comma.
- Use Call Reason Codes: If 'Yes' is selected, at the end of each call, the SolSwitch will request the agent to enter a reason code for that call. Reason codes are specified under Phone System Settings → Reason Codes.
- Use Busy Codes: If 'Yes' is selected, every time a user toggles their ready/not-ready status (*908), they will be required to enter a Busy Reason Code. Reason codes are specified under Phone System Settings → Reason Codes.
- Advanced Announcements: If set to 'No', the queue will use the default announce message style, using the options described below. If set to 'Yes' an extra button will appear giving the option to Manage Announcements, which gives a wider range of customizability for announce messages

(such as having multiple messages with different wait times). See 'Manage Announcements' below.

- Estimated Hold Time: If set to 'Yes', when an Announce Message is played, the system will also announce the estimated hold time of the queue. The estimated hold time is calculated based on an algorithm pertaining to call volume and queue wait time.

- Announce Interval: If an Announce Message is selected, this specifies the interval between replaying the Announce Message.

- Announce Message: The System Recording to use for announcing messages. These are set in the System Recordings page of Phone System Settings.

- Announce Position: If set to 'Yes', will announce the caller's current position in the queue.

- Announce First: Allows changing the order of the announce messages, announce hold time and announce interval.

- Language: This option determines which language to use in systems generated messages. Supports English and French.

Solutions that **shine**

Edit a Queue

Queue Active?	Yes
Extension Number:	3457
Name:	Queue
Weight:	None
Caller ID Prefix:	
Join Recording:	None
Time Rule:	None
Closed Destination:	None
Forward All Calls:	None
Maximum Hold Time:	Unlimited
Overflow Destination:	IVR: 20000 (Main IVR)
Ring Members For:	0 seconds
Retry After:	5 seconds
Service Level:	0 seconds
Wraptime:	None
Forced Exit Destination:	None
Ring Strategy:	Ring All
Hold Type:	Ring
Music On Hold:	None
Auto-Pause Members:	No
Allow Member Forwarding:	No
Record Calls:	On Demand
Show Prefix for Internal Calls:	No
Allow Callback:	● Yes ○ No
Callback Dial Prefix:	
Callback CID Name:	
Callback CID Number:	
Callback Attempts:	1 Attempt
Callback Attempt Interval:	15 minutes
Callback Fail Notify Email:	
Use Call Reason Codes:	No
Use Busy Codes:	No
Advanced Announcements:	○ Yes ● No
Estimate Hold Time:	No
Announce Interval:	Never
Announce Message:	None
Announce Position:	No
Announce First:	Announce Message
Language:	English

Save Queue Settings

After saving a newly created Queue, you will be given the ability to add members to the queue.

Adding Members to a Queue: A new table will now be visible to the right of the Queue Details. Add members by selecting extensions and clicking the 'Add New' button next to that extension. The penalty section here is used to determine a priority on who should be called first. Those with a higher penalty will receive calls after those with the lower penalty. The image below would mean the following:

206 – Jason Legault and 215 – Jeff Rainville would ring first for a period of 20 seconds (Ring Members For option). The queue would then wait 5 seconds (Retry After option) and then ring 216 – Derrick Chartrand for a period of 20 seconds (Ring Members For option). Once there are no other agents to try, the call will be sent to the overflow location.

Manage Announcements: When using Advanced Announcements and selecting 'Manage Announcements' the page will direct to the announcement options. This page has similar settings to the ones described above for announce messages, but allows more customization. The following options are available:

- Estimate Hold Time: If set to 'Yes', when an Announce Message is played, the system will also announce the estimated hold time of the queue. The estimated hold time is calculated based on an algorithm pertaining to call volume and queue wait time.
- Announce Position: If set to 'Yes', will announce the caller's current position in the queue.
- Time to First Message: The first Announce Message will play after this time. If set to never, announce messages will not play.
- Announce Message: The System Recording to use for this announce message. These are set in the System Recordings page of Phone System Settings.
- Time to Next Message: How long to wait before playing the next message in the list. This time starts after this message has finished playing.

- Announce Hold Time/Pos: Select whether the hold time and position messages should be announced before or after this message.

The messages set up on this screen will play in the order they are displayed after the set 'Time to Next Message' has elapsed. Clicking Save Settings will apply these changes and return to the queue edit screen.

Reason Codes

There are different reason codes that could be set and required from users based on the phone action or activity.

Busy Reason Codes

Located under Phone System Settings → Reason Codes, Busy Reason Codes are used when agents toggle their ready/not ready status. Whether or not busy reason codes are required is set in the Queue Details (Phone System Settings → Queues). If the 'Use Busy Codes:' is activated on that Queue, then when a user dials *908 to toggle their status, they will be prompted to enter a Busy Reason Code. Reports can then be pulled using the Queue Agent Not-Ready Report.

Code	Description	Edit	Del
01	Not There		
02	Away		
98	Gone		

Busy Reason Codes — Create Busy Reason Code

Call Reason Codes

Located under Phone System Settings → Reason Codes, Call Reason Codes are used when agents complete a call and are required to enter a reason for the call. Whether or not call reason codes are required is set in the Queue Details (Phone System Settings → Queues). If the 'Use Call Codes:' is activated on that Queue, then when an agent completes a call and the person on the other end of the call hangs up, the system will ask the agent to enter a call reason code. If the agent hangs up before entering a reason code, the SolSwitch will call them back and request it right away. Reports can then be pulled to gather more details regarding the queue, agents and reason codes for specific time periods using the Queue Call Reason Report.

Ring Groups

For companies set up with different divisions or departments, Ring Groups can further improve communication and call efficiency. They are very similar to queues although with a little less customizability. Incoming calls can be redirected to a designated group of extensions, enabling a busy call centre to answer all calls promptly.

Adding a New Ring Group:

- Extension Number: Enter an available extension number for your queue.
- Name: Enter a name you wish to associate with your queue.
- Description: For reference purposes.
- Time Rule: Select a specific time rule if this ring group is only available during certain times. Time Rules are configured in the Phone System Settings menu.
- Closed Destination: If a time rule is set and its status is closed, calls will be sent to the location specified here.
- Forward Busy or N/A: Where to forward calls to when an extension is busy or not answered.
- Time before forwarding: Specify the number of seconds before forwarding to the Forward Busy or Forward No Answer location. One ring typically takes 5 seconds.
- Record Calls: Can either be set to On Demand or Always. Always will record all calls on that extension whereas On Demand will require the user to press the 'Record' button or #1 during a call.

Similar to Queues, once a Ring Group's details have been saved, you will be able to add members to the Ring Group. Ring Groups have no penalties and ring all of its members at the same time.

Scheduled Routing

The Scheduled Routing feature allows you to created Scheduled Routes based on date spans. Multiple scheduled routes can be configured to match any schedule. To create a new Scheduled Route, click the 'Add Scheduled Route' button. You can also import a list in the form of a CSV file.

Adding a Scheduled Route:

- Extension Number: Enter an available extension number for your scheduled routing.
- Name: Enter a name you wish to associate with your scheduled route.
- Default Route: In the event that a date/time range has not been specified, calls will route to the default location. This field can be typed in for quick auto-complete selections.
- Alert Email(s): An alert email will be sent if no time specifications are met, meaning that the default route is being used. Separate additional emails with a comma.
- Auto Repeat: If selected, the scheduled routing will be automatically regenerated when routes run out.
- Add Route Schedule: Select a date range by inserting a start and end date then click the 'Add Date Range' button.
    - Once a date range has been added, times ranges for that date range can be added.
    - Select where on the system you would like the call routed from the dropdown menu and click the 'Add New Rule' button.

You can add multiple time ranges per day range as well as multiple day ranges. Additional options allow you to 'Clone' a daily entry for recurring schedules.

Importing a Scheduled Route:
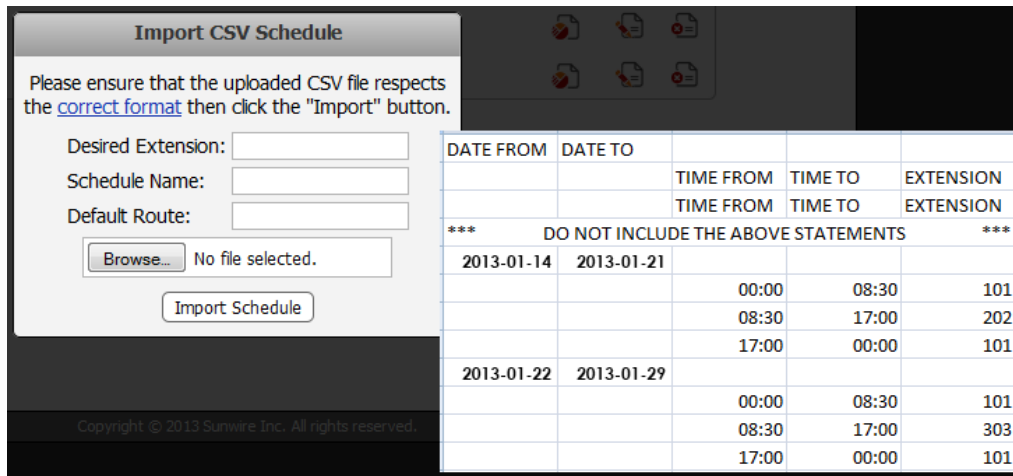
System Recordings

MP3 audio files of IVR or Queue greetings, product announcements, company news, business hours, etc are uploaded here.

Creating a System Recording: Click the 'Add System Recording' button and enter a name for your recording. Set the 'Active:' field to 'Yes' in order to enable the recording.

Uploading a System Recording: If you have a sound file you would like to upload, click the 'Browse' button, select your file and then click the 'Save System Recording' button. The system will accept any standard WAV or MP3 file. The SolSwitch handles sound conversion and bit-rate control so you don't have to.

Recording a System Recording: After naming and activating your system recording, press the 'Save System Recording' button. View the system recording and follow the *-codes to listen, record or toggle the recording.

Time Rules

Time rules are used by the SolSwitch to direct calls to the appropriate extensions, virtual extensions, queue, IVR, ring group, etc. during the specified times as per the created Time Rules. To add a Time Rule, click the 'Add Time Rule Group' button.

- Time Rule Group Name: Enter a name to identify this group of Time Rules.
- Time Rule Description: For reference purposes.
- Toggle Extension: Calling this extension will toggle this Time Rule from open to closed or closed to open.
- Passcode: Based on the selection below, callers will be prompted for this pass code in order to toggle the Time Rule.
- Ask for Passcode: Specify when a pass code should be asked for when toggling the Time Rule.

Click the 'Save Group Info' button.

Once your group has been created, you can begin creating individual rules. To add a rule, click the 'Add Rule' button. You are able to toggle the time rule on and off within the web portal by clicking the Toggle status button. There are two types of rules that can be added.

Range Rules: Range rules specify a start and an end for a rule. It can be selected based on Weekday or by Monday and Day. To create a range rule by weekday, follow the example below.

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The image above shows that this particular range rule will open Monday, Tuesday, Wednesday and Thursday at 8:00AM and will close Monday, Tuesday, Wednesday and Thursday at 5:00PM.

Creating a Range Rule based on Month and Day (such as for holidays); the following format would be followed:

Edit a Time Rule

Rule Type:	Range
Rule Description:	Closed for Canada Day
Rule specificity:	○ By Weekday ● By Month and Day

From:
Month: ☐ January ☐ February ☐ March ☐ April ☐ May ☐ June
☑ July ☐ August ☐ September ☐ October ☐ November ☐ December

Day: 1

Time: 00 : 00

Action: Closed

To:
Month: ☐ January ☐ February ☐ March ☐ April ☐ May ☐ June
☑ July ☐ August ☐ September ☐ October ☐ November ☐ December

Day: 2

Time: 00 : 00

Action: Open

Save Rule

Multiple rules can be used in conjunction with each other in order to adjust the time rule properly.

Instance Rules: An instance time rule does not have a start and end point to the rule. It simply opens or closes a rule based on weekday or based on month and day. Multiple instance time rules can be used as well. If there are instance rules and range rules, the range rules will take precedence over instance rules. In the screenshot below, the instance rule opens the time rule every weekday at 8AM.

Edit a Time Rule

Rule Type:	Instance
Rule Description:	Open Weekdays at 8:00AM
Rule specificity:	◉ By Weekday ○ By Month and Day
Weekday:	☑ Monday ☑ Tuesday ☑ Wednesday ☑ Thursday ☑ Friday ☐ Saturday ☐ Sunday
Time:	08 : 00
Action:	Open

Save Rule

Another instance like the one above could also be made in order to close the Time Rule.

Another example of an instance rule, using the month and day format, would be configured similarly to the following:

Edit a Time Rule

Rule Type:	Instance
Rule Description:	Close on July 1st at Midnight
Rule specificity:	○ By Weekday ◉ By Month and Day
Month:	☐ January ☐ February ☐ March ☐ April ☐ May ☐ June
	☑ July ☐ August ☐ September ☐ October ☐ November ☐ December
Day:	1
Time:	00 : 00
Action:	Closed

Save Rule

Following the configuration of all the rules within the Time Rule group, a 'Rule Calendar' is displayed on the Time Rule group page displaying when the Time Rule is scheduled to be open or closed.

Virtual Extensions

A virtual extension allows you to create an internal extension number that is attached to an outside telephone number. Therefore, in order to reach an external number; internal users would simply need to dial the appropriate extension. To add a new virtual extension, click the 'Add Virtual Extension' button.

Adding a Virtual Extension:

- Extension Number: Enter an available extension number for your virtual extension.
- Name: Enter a name you wish to associate with your virtual extension.
- External Number: An external telephone number that will be associated with this virtual extension.
- Extra Dial String: Enter digits you would like to be dialed after the call is answered.

Solutions that shine

- o This feature is useful in the event that the external number points to an auto-attendant (IVR). Acceptable Characters are: 0-9, *, #, a, A, b, B, c, C, d, D and w (the w is a 0.5 second pause).
- Call Waiting: If set to 'No', this virtual extension will return a busy signal when a call is already in progress.
- Caller ID Type: If set to 'Passthrough', the caller ID will be that specified by the incoming extension. If set to 'Specify', you will have the ability to specify the Caller ID Name as well as the Caller ID Number.
- Time Rule: Select a specific time rule if this virtual extension is only available during certain times. Time Rules are configured in the Phone System Settings menu.
- Closed Destination: If a time rule is set and its status is closed, calls will be sent to the location specified here.
- Record Calls: Checking this box will record all calls directed to this virtual extension. Leaving the box unchecked will not record any call.

Add/Edit a Virtual Extension

Extension Number:	4130	
Name:	John's Cell Phone	
External Number:	7055551234	
Extra Dial String:		
Call Waiting:	Yes	
Caller ID Types:	○ Passthrough ● Specify	
Caller ID Name:	OFFICE FWD	
Caller ID Number:	7055550000	
Time Rule:	None	»
Closed Destination:	None	»
Record Calls:	☑	

Save Virtual Extension

Voicemail

Voicemail on the SolSwitch is treated as a standalone feature. Each voicemail is given its own extension number so that it can be used in any way needed. The Voicemail section under Phone System Settings shows a list of all the voicemails on the system. To add a new voicemail box, click the 'Add Mail Box' button. For a more comprehensive understanding of the voicemail section and all of its options, please see the 'VOICEMAIL' section of this manual.

Adding a Voicemail Box:

- Extension Number: Enter an available extension number for your voicemail. This extension number is the number to dial in order to access the voicemail box directly.
- Mailbox ID: This is typically the actual extension number associated with the voicemail. The mailbox ID is the number dictated by the mailbox system when a user is sent to the mailbox and no greeting has been recorded. This recording reads: "The person at extension XXX is unavailable. Please leave a message after the tone." The mailbox ID is also used when forwarding voicemail messages. When adding a voicemail to an extension, it's highly recommended that the Mailbox ID be the same as the extension you are adding it to.
- Name: Enter a name for your voicemail box; for informational purposes.
- Email: The email address entered here will receive a notification message with the .wav file of the voicemail attached to each email.
- Email Only: Voicemails will be sent to emails but will not be stored locally.
- Email Transcript: A transcript of the voicemail will be included in the voicemail email.
- Force Passcode Change: This option forces the user to change his/her passcode on next voicemail login.
- Exit Destination: This is where a call will be routed if someone dials '0' during the playback of the voicemail greeting.
- Days Valid: The number of days the voicemail will be automatically deleted. A 0 value will require voicemails to be manually deleted.
- Say Caller ID: If this box is checked, the voicemail recording will be preceded by the caller's Caller ID information.
- Say Date/Time: If this box is checked, the voicemail recording will be preceded by the date and time that the message was left at.
- Distribution List: When enabled, any messages left on the voicemail will be distributed to the Notify/Distribution Extensions.

Save the Voicemail Box by clicking the 'Save Voicemail' button. Once this is completed, you will be given the option to configured the 'Notify Extensions' section. This section lists which users will receive 'Message Waiting Indicator' notifications on their devices when a voicemail message is left in that particular mailbox.

Reporting and Monitoring

Agent Stats

The Agent Stats report provides historical data on specific agents (extensions). This can be shown in total numbers or values for individual days. For live monitoring of these same statistics, see the 'Live Agent Stats' report below. The first page of this report allows selection of the date range, display type and the extensions to run the report for.

Agent Stats
Reporting on agent specific statistics

Main Menu > Reporting and Monitoring > Agent Stat

Agent Stats

From:	2017-08-31 00:00
To:	2017-08-31 23:59
Display Type:	Totals

Load Report

Extensions:

☑ - All Extensions -
☐ 100 - Aastra 6867i
☐ 101 - Aastra 6869i
☐ 102 - GrandStream HT701

On loading the report, the statistics are listed for the selected extensions, shown as totals or sorted by date as set by the Display Type.

Agent Stats Report - 2017-08-31 00:00 to 2017-08-31 23:59

Report For: 2017-08-31 00:00:00 - 2017-08-31 23:59:59

Extension	Name	Inbound Calls	Outbound Calls	Total Calls	Average Handle Time	Avg Time To Answer	Avg Service Level
100	Aastra 6867i	0	0	0	00:00:00	00:00:00	NA
101	Aastra 6869i	0	0	0	00:00:00	00:00:00	NA

Call Detail Records

Perhaps one of the more powerful reporting tools on the SolSwitch for general call information, the Call Detail Records shows a breakdown of all the calls on the system during the specified time period. Additional search fields are available and include inbound and outbound calls, extension or routing type and offer an overview of which agents are taking calls from the queue or which calls went to voicemail.

There is also an option to search for recordings only, which will only show calls that were recorded on the system.

Once the search results are displayed, you will have the option to export the results to a CSV file for further processing. If there is a recording of a particular call, a small speaker will be displayed next to that call. For more details regarding the call, click the small '+' button.

Clicking the small '+' button will display details such as when the call was initiated, where the call was routed, who answered it, as well as who ended it. If the call passes through specific IVRs or Queues, it will all be displayed. This is also where you can playback or save call recordings.

CDR Report Builder

The CDR Report Builder is similar to the Call Detail Records although it allows you to customize the type of CDR report you would like. Your custom reports can also be saved in the system for future use. Creating a report is as simple as entering a date range, checking off which columns to include in the report as well as adding optional filter(s) and choosing how to sort the report. To save a report, simply enter a Name for

the report at the bottom of the page and click the 'Save' button. Saved reports will be displayed in the dropdown menu for future use. Clicking the 'Generate Report' button will display the results on the screen. To download a CSV file of the report, use the 'Export CSV' button instead.

Query Details

Date Range

From: [   ]   To: [   ]

Columns to Include   [Check All] [Un-check All]

- ☐ Leg ID
- ☐ Call ID
- ☐ Server ID
- ☐ Start Time
- ☐ Connect Time
- ☐ End Time
- ☐ Call Length

- ☐ Day of the Week
- ☐ Source Name
- ☐ Source Number
- ☐ Source Channel
- ☐ Source Record
- ☐ Source Privacy
- ☐ Dest. Name

- ☐ Dest. Number
- ☐ Dest. Channel
- ☐ Dest. Trunk ID
- ☐ Dest. Record
- ☐ Dest. Privacy
- ☐ Disposition
- ☐ Dialed Number

- ☐ Terminate Side
- ☐ Exit Dialed
- ☐ Global Record
- ☐ Recording Name

Filter(s)

[Add Filter]

Sort By

[None ▼] [Ascending ▼]

[Clear Report]

Save & Load Reports

Name: [   ] [Save]

Saved Reports: [None ▼] [Delete]

[Generate Report] [Export CSV]

Current Calls

The current calls screen displays all active calls currently in place on the system. Each active call has data concerning the source, direction, destination, duration, source channel, destination channel as well as the ability to hang up the call. Only current active calls are displayed here.

Source	Direction	Destination	Duration	Source Channel	Destination Channel	Hang up
206	Dir	7055071309	00:00:08	SIP/jason601-0000014f	SIP/Earth-00000150	

Calls

Extension Time Report

This page gives a detailed summary report of all calls sorted by extension for any specified time period. An example report would be the following:

Agent Report

From: 2014-08-18 00:00
To: 2014-08-18 23:59
Extensions: All Extensions
Group By: None

Load Report

The results would return:

- **Total Inbound Calls & Total Time on Inbound Calls**
- **Total Outbound Calls & Total Time on Outbound Calls**
- **Total Calls & Total Call Time**
- **Total & Average Wrap-Up Time**
- **Total Log-In Time and Log Out Time** (Returns the total amount of time the extension was logged in for or logged out for).

- **Total Break Time** (Returns the length of time agents were on break)
- **Total Time Ready** (Returns the total length of time agents were available for a call).

Live Agent Stats

The Live Agent Stats page shows live details for specific agents (extensions). The following statistics are shown for each agent:

1. Inbound Calls
2. Outbound Calls
3. Total Calls
4. Average Handle Time (Average time spent on calls)
5. Average time to Answer
6. Service Level (Based on service level of queues that the agent is a part of)

Live Agent Stats

Main Menu > Reporting and Monitoring > Live Agent Stats

Agent Statistics [Change Agents]   Last Refresh: 10:00:52am 2017/08/31

Ext	Name	Inbound Calls	Outbound Calls	Total Calls	Average Handle Time	Avg Time To Answer	Service Level
333	Test	0	0	0	00:00:00	00:00:00	NA
802	Test	0	0	0	00:00:00	00:00:00	NA

Live Stats Report

The Live Stats report shows the last updated time and live details for all or select queues. Stats are split into two sections the Queue Statistics, and the current agent status. We are able to see the call taken and missed by the agents in addition to the overall taken/missed calls of the queues.

Queue Statistics 1007 - Sales Queue ▼   Last Refresh: 11:32:04am 2017/08/04

Queue	Qd.	An.	Ab.	VM.	An. Rate	Ab. Rate	Service Level	Current Wait	Calls	Agents	Min.	Wait Time Max.	Avg.	Req.	Callbacks An.	Miss.	Wait.	Min.	Talk Time Max.	Avg.
Sales Queue	0	0	0	0	0%	0%	0	00:00:00	0	1	00:00:00	00:00:00	00:00:00	0	0	0	0	00:00:00	00:00:00	00:00:00
TOTAL	0	0	0	0	0%	0%	0	0	0	1	00:00:00	00:00:00	00:00:00	0	0	0	0	00:00:00	00:00:00	00:00:00

Current Agent Status

Agent	Team	Phone Ext	Status	Duration	Inc. Calls An.	Miss.	Out. Calls	Callbacks An.	Miss.	Log. On	Avail.	Time Management On Inc. Calls	On Out. Calls	Wrap-up	Break
Jason Legault	Customer Support, Queue1, Sales Queue	206	Available	00:00:00	19	0	0	0	0	00:00:00	23:08:33	00:51:27	00:00:00	00:00:00	00:00:00

Live Queue Stats

The Live Queue Stats page will show you live details for all queues, agents and current calls. Queue Statistics include information for each queue relating to:

1. Calls
2. Callbacks
3. Available Agents
4. Answered
5. Abandoned
6. Waiting Time
7. Estimated Wait Time
8. Voicemails
9. Service Level

Queue Member Status shows the last updated time, a list of extensions as well as what queue(s) that extension belongs to, it's current status, if the status is on the phone, which queue the call came from as well as the duration of the call if one exists. In addition, you are able to change what queue stats to display, whether all or some of the queues.

Queue Statistics [Change Queues]   Last Refresh: 11:29:28am 2017/08/04

Ext	Queue	Calls	Callback	Avail. Agnts	Answered	Abandoned	Wait Time	Estimated Wait Time	Voicemail	Service Level
1007	Sales Queue	0	0	1	0	0	00:00:00	00:00:18	0	0
9002	Queue Stats 2	0	0	2	0	0	00:00:00	00:06:55	0	0

Queue Member Status

Name	Queues	Status	Queue/Number	Duration
291 - Software Testing 2	Queue Stats 2	Available		+1 Day
290 - Software Testing	Queue Stats 2	Available		21:01:04
206 - Jason Legault	Sales Queue	Available		+1 Day

Queue Agent Busy Report

This page gives a detailed report of busy reason codes by agent. This report can be filtered by queue (one, multiple or all), by agent (one, multiple or all) or by reason code (one, multiple or all). Based on these filters, the display of the report can be grouped by day or by queue.

Reports grouped by Queue are shown as such:

(Queue Call Reason Report for August 3 2017 – August 3 2017)

Report for: 2017-08-03 to 2017-08-03			
Queue/Extension Name	**#2 - Reasons**	**#996 - Wraptime**	**Total/Time**
Queue Stats 3	**1 / 00:00:07**	**0 / 00:00:00**	**1 / 00:00:07**
Software Testing 2	1 / 00:00:07	0 / 00:00:00	1 / 00:00:07
Queue Stats 4	**1 / 00:00:07**	**0 / 00:00:00**	**1 / 00:00:07**
Software Testing 2	1 / 00:00:07	0 / 00:00:00	1 / 00:00:07
Queue2	**1 / 00:00:07**	**0 / 00:00:00**	**1 / 00:00:07**
Software Testing 2	1 / 00:00:07	0 / 00:00:00	1 / 00:00:07
Queue Stats 2	**1 / 00:00:07**	**2 / 00:02:06**	**3 / 00:02:13**
Software Testing 2	1 / 00:00:07	0 / 00:00:00	1 / 00:00:07
Software Testing	0 / 00:00:00	2 / 00:02:06	2 / 00:02:06

Reports grouped by Day are shown as such:

(Queue Call Reason Report for August 03 2017)

Report for: 2017-08-03			
Queue/Extension Name	**#2 - Reasons**	**#996 - Wraptime**	**Total/Time**
Queue Stats 3	**1 / 00:00:07**	**0 / 00:00:00**	**1 / 00:00:07**
Software Testing 2	1 / 00:00:07	0 / 00:00:00	1 / 00:00:07
Queue Stats 4	**1 / 00:00:07**	**0 / 00:00:00**	**1 / 00:00:07**
Software Testing 2	1 / 00:00:07	0 / 00:00:00	1 / 00:00:07
Queue2	**1 / 00:00:07**	**0 / 00:00:00**	**1 / 00:00:07**
Software Testing 2	1 / 00:00:07	0 / 00:00:00	1 / 00:00:07
Queue Stats 2	**1 / 00:00:07**	**2 / 00:02:06**	**3 / 00:02:13**
Software Testing 2	1 / 00:00:07	0 / 00:00:00	1 / 00:00:07
Software Testing	0 / 00:00:00	2 / 00:02:06	2 / 00:02:06

Queue Alerts

The Queue Alerts page allows configuration of alerts that can be sent based on queue statistics. These alerts are checked every minute on the system, and if any of the items are over the set limit on this page an email will be sent to the specified email(s). The page allows for creation of multiple 'Alert Groups' for the same queue, meaning that different thresholds and emails can be set for the same queue. The Notify Interval can also be set, which limits how often emails will be sent.

Queue Alerts

Queue:	9002 - Queue Stats 2 ▼
Alert Group:	Group 1 ▼
Notify Email(s):	example@sunwire.ca
Notify Interval:	1 minute(s)
Notifications:	
Wait Time:	4 second(s)
Calls in Queue:	2
Callbacks in Queue:	2
Abandon Rate:	25
Talk Time:	30 second(s)
Busy Reason Codes:	⦿ Yes ◯ No

[Save Settings] [Delete Group]

Queue Call Breakdown

A historical report of inbound calls answered and inbound calls abandoned (Inbound calls that hang up before being answered). These stats also display a percentage of service level (S/L). This stat represents the ratio of abandoned calls to completed calls. The S/L Calls number displays the total calls that were answered within the service level time.

Queue Call Reason Report

This page can be filtered by start date/time, end date/time, by queue (one, multiple or all), by agent (one, multiple or all) or by reason code (one, multiple or all). Based on these filters, the display of the report can be grouped by day or by queue.

Reports grouped by Queue are shown as such:

(Queue Call Reason Report for August 3 2017 – August 3 2017)

Agent	Tech	Sales	Wrong Dept	Description	Unknown	Total Calls / Total Time
Jason Legault	1 / 00:01:22	1 / 00:03:25	0 / 00:00:00	0 / 00:00:00	0 / 00:00:00	2 / 00:04:47
John Smith	2 / 00:04:01	1 / 00:02:40	2 / 00:06:56	0 / 00:00:00	0 / 00:00:00	5 / 00:13:37
Room 245	1 / 00:01:42	1 / 00:04:17	2 / 00:06:44	0 / 00:00:00	0 / 00:00:00	4 / 00:12:43
Roman Kolodziejczyk	1 / 00:03:50	0 / 00:00:00	0 / 00:00:00	0 / 00:00:00	0 / 00:00:00	1 / 00:03:50

Queue1 - (August 3rd, 2017 to August 3rd, 2017)

Reports grouped by Day are shown as such:

(Queue Call Reason Report for August 3 2017)

August 3rd, 2017

Agent	Tech	Sales	Wrong Dept	Description	Unknown	Total Calls / Total Time
A Transfer: Recording Test	19 / 01:04:05	25 / 01:11:38	10 / 00:34:19	0 / 00:00:00	0 / 00:00:00	54 / 02:50:02
testing2	0 / 00:00:00	0 / 00:00:00	0 / 00:00:00	0 / 00:00:00	0 / 00:00:00	0 / 00:00:00
Jason Test	1 / 00:01:37	1 / 00:02:09	0 / 00:00:00	0 / 00:00:00	0 / 00:00:00	2 / 00:03:46
vvx300	1 / 00:03:11	1 / 00:04:42	0 / 00:00:00	0 / 00:00:00	0 / 00:00:00	2 / 00:07:53
fax	1 / 00:02:10	1 / 00:02:49	0 / 00:00:00	0 / 00:00:00	0 / 00:00:00	2 / 00:04:59

Queue Service Level Report

This page allows a user to pull a report between two dates and filter by select queues answer/abandoned calls, then displays that pulled record to see all the answer and/or missed calls

Queue Service Level Report - 2017-01-26 00:00 to 2017-01-26 23:59

Queue	Total Calls Abandoned <= 30 secs	Total Calls Abandoned > 30 secs	Total Calls Abandoned	% of NET Abandoned	Total Calls Answered Within S/L	Total Calls Answered After S/L	Total Calls Answered	Total Calls	Service Level
Service Queue	0	0	0	0%	0	0	0	0	0%
Service Level met on **0%** of the calls received.									
Service Level not met on **0%** of the calls received.									
0% of the calls received were abandoned on or before 30 seconds.									
0% of the calls received were abandoned after 30 seconds.									

Answered Service Level

Date/Time	Caller Info	Queue	Call ID	Wait Time	S/L Met

Abandoned Calls

Date/Time	Caller Info	Queue	Call ID	Wait Time	Q > 30

Scheduled Reporting

The scheduled reporting page allows select reports to be set to run at regular intervals. The report will automatically run on the select day/time, for the selected time rage, and the results of the report will be sent via email.

The initial page shows currently scheduled reports, their next run time, and gives the option to Add, Edit and Delete reports.

List of Scheduled Reports   Add New Report

Name	Report	Next Run Date/Time	Edit	Del
Test: Extension Time Report	Extension Time Report	2017-10-05 06:00:00		

Selecting Add or Edit will show the details page that allows the user to set up a wide range of options for when and how the report should run. The following options are available on this page:

- Report Type: The report that should be run. Changing this drop down also changes the options at the bottom of the page, based on what information that report requires. For more details of the specific settings for each report, refer to the section for that report.
- Name: Name for the report, used to identify the reports on the initial page as well as when the results are emailed.

- Email Results To: The email accounts that should receive the results. Multiple email accounts can be added by separating them with a comma.
- Run Report: When the report should run. This has the following options:
  - Weekly: Run the report every week on the selected day and time.
  - Monthly: Run the report every month on the selected day and time.
  - Number of Days: Run the report with a set interval of days, set by the frequency. Ie if the frequency is set to 60, the report will run every 60 days at the selected time.
- Time Span: The span of time that the report should cover when it runs. The following options are available:
  - Previous Weekdays: Allows selection of specific days of the week. Report will run on each of those days individually, using the set time frame if the report has the option for times.
  - Previous Week: The report will run for the previous 7 days.
  - Previous Month: The report will run for the previous calendar month. Ie, if the report is run at any point during May with previous Month selected, the report will run for the full month of April.
  - Previous Number of Days: The report will run for a previous set number of days, starting from the previous day.

Toll Reports

This page pulls toll cost reports, it displays the date time, duration, destination and cost of calls for any given month/year.

Toll Reporting	
Date:	October 2016

Date and Time	Duration	Source #	Destination #	Destination	Cost of Call
October 16th, 2016 21:34:57	00:00:30	306	14704440060	USA - Georgia	$2.77
				Outbound:	$2.77
				Toll-Free:	$0.00
				Total:	$2.77

Queue Graphs

This report generates a graph which represents queue activity between selected dates and times. The report will show the number of calls that were waiting in a queue as well as the number of calls that were spent talking in a queue. Below is a sample report:
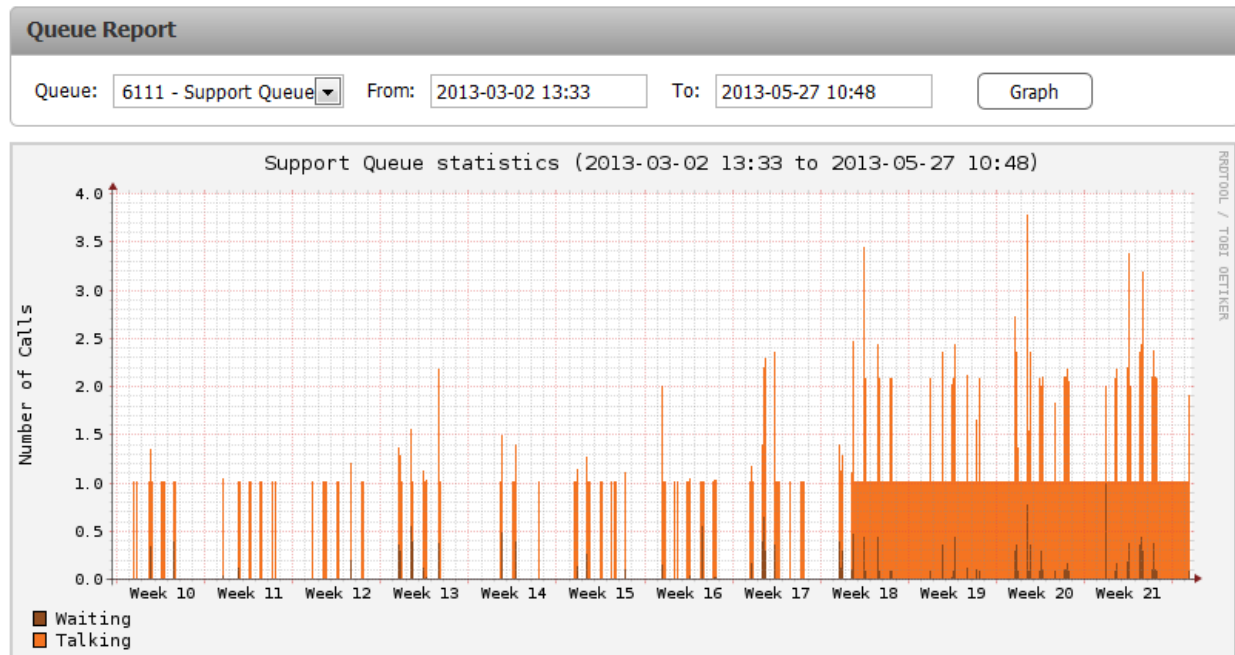
SIP Ping Monitor Report

This feature generates a graph which represents SIP account activity between selected date and times. It will display, in milliseconds, the Pack Loss, Ping response and when a SIP account was offline.

Voicemail Usage Report

The Voicemail usage report shows each voicemail, its name as well as the extension it is assigned to and the size of their mailbox.

Voicemail Usage Report

VM Extension	VM Name	Phone Extension	Mailbox Size
246	Roman Kolodziejczyk	1246	382.09 KB

Tool and Add-Ons

Create Queue Callback

This feature allows you to manually enter a number/caller into a queue as a call-back. As soon as an agent in a queue is available for a call, the system will automatically dial manually entered number and connect the agent.

Outbound Faxes

This page allows you to queue a PDF attachment and send it out the system to any fax number of your choice. To create an Outbound Fax, click the 'Create Fax' button.

Creating an Outbound Fax:

- To Phone Number: The numbers that you would like the fax to be sent to.
- From: Displays a dropdown of users created in the Fax to Email under Phone System Settings, which the outbound fax will be sent from.
- Confirmation Email Address: A notification will be sent to the email address specified here which will announce the results of the fax transmission.
- Maximum Number of Tries: The number of attempts the fax will try to send before stopping.
- PDF to Fax: Upload a PDF of the file you would like to be sent.

SolCast

The SolCast is used as a recurring calling tool where you schedule the valid dates, days, times and retry attempts of a particular batch. You can create as many batches as you like.

Adding a SolCast Batch: To add a new batch, click the 'Create New Batch' button. There are 5-steps to fill out for each batch; outlined on the left in the 'Batch Menu'. Below is a breakdown of each section of the batch menu and its components.

- **Batch Information:** This screen is where you set the main SolCast batch details. Ensure to save the Batch Information before proceeding to the next part of the batch menu.
    - Batch Name: Give the batch you are creating a unique name.
    - Caller ID Name: The name to be displayed when calling numbers.
    - Active: Whether or not the batch will run.
    - Valid Period, Valid Days, and Valid Time: Select the timeframe that you would like the batch active for.
    - Retry Attempts: The maximum number of times to try calling a contact number.

- o Retry Delay: The amount of time to wait before attempting to retry calling a contact number. Note that this time is as of the end of the batch running, not from the last retry attempt.
- o Simultaneous Calls: The number of lines to use at the same time to make multiple calls.
- o DTMF 1 Goes to: Where to send the caller if they press '1' during the message playback.

- **Batch Dial List:** This batch menu is where you set your dial list. You can use the 'Upload CSV' button to upload a list of users to contact. Ensure that the CSV is formatted properly. Click the button to see the proper format.

  Users can also be entered manually using the 'Manual User Entry' form, whose options are listed below:
  - o Name: The user/contact's name.
  - o Phone #: The phone number to call which is associated with the above name.
  - o Fields 1-5: Fields 1 through 5 can be used to enter various contact information such as Email address, street address, billing type, balance due, etc.

  The 'Edit Field Names' button will allow you to change the Field 1, Field 2, Field 3, Field 4, Field 5 labels to custom labels of your choice.

Ensure to save your changes before proceeding to the next phase of the batch menu.

Batch Menu

- Batch Information
- Batch Dial List
- Contact Numbers
- Answered Steps
- Voicemail Steps

- Save Settings
- Exit

Manual User Entry

Name:		Phone #:	
Balance:		Not Used:	
Account:		Not Used:	
Not Used:			

Edit Field Names | Save Record

Batch Dial List

Upload CSV

	#	Phone #	Name	Balance	Account	Not Used	Not Used	Not Used	Attempts	Last Attempt	Log
☐	1	7055551111	Jason	85.55	123456				0	2013-07-04 13:29:06	🖼
☐	2	7055552222	John	45.21	!2312				1	2013-07-05 15:24:32	🖼
☐	3	7055553333	Jim	$12.44					1	2013-07-05 15:27:48	🖼
☐	4	7055554444	Jason	$123.99	12212				1	2013-07-05 15:31:06	🖼
☐	5	7055556666	Sandra	454.22	456123				1	2013-07-05 16:01:58	🖼
☐	6	7055557777	Jill	$45.22	51324				1	2013-07-05 16:03:42	🖼
☐	7	7055558888	Jack	$457.25	1234				1	2013-07-05 16:04:58	🖼
☐	8	7055559999	Derrick	$18.32	1247				1	2013-07-05 16:05:59	🖼

Remove | Edit | Reset Attempts | Export to CSV

- **Contact Numbers:** This area can be used to specify different contact information based on which rate centre is used for the call.

  For example, if you select a specific rate centre such as Toronto, all calls going to the Toronto rate center will show the phone number entered as the CID information. Also, specific recordings can be played based on where the call is being made too. There is no limit to the rate centres that can be added so long as they are configured in your SolSwitch.

  This option can be very useful for large companies with office numbers across multiple cities. This gives the person receiving the call a local number that they can call back. Ensure to save your changes before proceeding to the next menu page.

Batch Contact Numbers ◼

This area can be used to specify different contact information based on which rate centre is used for the call.

In the "Calling Steps" section, a "Contact #" step will use the values below to determine what the customer will hear. If a system recording is selected, it will be played to the customer. If not, the digits of the phone number will be read to them.

Rate Centre	Phone Number	System Recording
Default:	8885216777	None »
Sudbury	7055216777	None »
Select Rate Centre		

Add Rate Centre

- **Answered Steps:** This screen is used to tell the SolCast what actions it should take when a phone call is answered. Ensure to save your changes before proceeding to the next menu page. Below are the options to choose from as well as the function they serve:
  - o Reading: Will play whatever system recording is selected from the reading dropdown.
  - o Field: Will spell out whatever field is selected from the 'Batch Dial List'. For example, if you select 'Balance' then the system will look up that particular field in the 'Batch Dial List' for each entry and read off the balance.
  - o Read Number: Will playback the batch's contact number based on the rate centre being called. If no rate centre is chosen, the default contact number is displayed.

- **Voicemail Steps:** The same steps above apply to calls that are answered by a voicemail system although the steps themselves can be different whether a call is answered or goes to voicemail.

Queue Assignment

A color coded data table representing which agents are assigned to which queues as well as the queue status.

SolSwitch Connect

SolSwitch Connect Accounts

Configure SSC accounts to set up for phone calls and/or messaging.

- User Account: Establish a username and password for the account.
- Messaging: Enable/disable the messaging feature for the account. If enabled, choose the rosters they will participate in.
- Messaging vCard: Set information for the account such as Display Name, Email, Work/Mobile phone number, Department, and Title.
- Misc Settings: Additional optional settings.
    - SMS: Allows SMS to be sent and received via the SSC client.
    - Collaboration features: Enable this to give user access to conferences and screen sharing. Provide a conference number when enabled.
    - SSC Tools: If enabled, it gives the user access to the SSC tools tab in desktop clients.
- SIP Settings: Set the SIP accounts and server here for the account to function. SSC can support up to 2 SIP accounts at once.
- Incoming Call Routing: Handles how certain calls can be routed.
    - Auto Answer: When enabled, incoming calls will automatically be answered.

Solutions that **shine**

o   SSC Desktop/Mobile: Enable or disable incoming calls coming directly or through a queue.

SolSwitch Connect Download

A list of downloadable SSC clients for various platforms including Windows, Mac, Android and iOS.

SolSwitch Connect Rosters

Manage rosters for SSC. These function as a space where members of the roster can communicate with each other via private messaging or chat rooms.

SolSwitch SNMP Details

SolSwitch uses SNMP (Simple Network Management Protocol) to provide access to critical values. Through the portal a user is able to specify the *rocommunity* name to be used to read values with as well as the subnet it is available to using an IP/subnet mask combination.

To do this you would go to **Main Menu > Advanced Tools > SNMP Configuration**. This is the section of the SolSwitch portal where you can define *rocommunity* and who it is accessible by.

SolSwitch OIDs

The SolSwitch uses *MIB-2*. An SNMP walk will provide you with a full list of available SNMP OIDs. The following list of OIDs is SolSwitch specific and will show up as interfaces in most SNMP monitoring software.

- *.1.3.6.1.2.1.2.2.1.8.10 [The result of most recent SIP test; 1 for success, 0 for failure]*
- *.1.3.6.1.2.1.2.2.1.8.11 [Status of High Availability; 1 for working/syncing; 0 for failure or not enabled]*
- *.1.3.6.1.2.1.2.2.1.8.12 [Status of Load Balancing; 1 for working/syncing; 0 for failure or not enabled]*
- *.1.3.6.1.2.1.2.2.1.8.13 [Status of database replication; 1 for working/replicating; 0 for failure or not enabled]*
- *.1.3.6.1.2.1.2.2.1.10.10 [Number of current calls on the system]*

Other OIDs

Other OIDs that the SolSwitch will display are:

- hrSystemUptime; .1.3.6.1.2.1.25.1.1.0
- hrSystemProcesses; .1.3.6.1.2.1.25.1.6.0
- hrMemorySize; .1.3.6.1.2.1.25.2.2.0
- hrProcessorLoad; .1.3.6.1.2.1.25.3.3.1.2

Call Features

Placing a Call

Depending on how you want to place a call, several operations are available.

- To place a simple call, pick up the handset, dial and press the send soft key.
- Speakerphone is available by pressing the speakerphone key, dialling and pressing send
- Paging is set up on a soft key and activated by pressing the appropriate button
- Speed dial is set up on another soft key. Press the button, choose the number and press dial on the soft key.
- Soft keys are set up in software by the system administrator.
- Detailed telephone features that can be set up by the agent are explained in the respective SIP Device's manual.

Receiving a Call

To receive a call, you must pick up the handset, press answer on the soft key (will use speaker phone if handset is hung up), or press the speaker-phone button for speaker phone.

Call Forwarding

This feature can be set up in one of two ways:

1. Dial *72. When asked to enter the number to be forwarded to, do so, then press the # key. The system will say "Call forwarding enabled" when done correctly.
    a. To disable the above call forwarding, simply dial *73.

2. Out-of-office call forwarding can also be setup in the web portal under Main Menu → Phone System Settings → Extensions → Edit Extension menu. Choose the forwarding method and internal extension, external extension or virtual extension.

Recording a Call

Call Recording is done in one of two ways, depending on how the feature is enabled. In the portal under Phone System Settings → Extensions → Edit Extension:

- Calls are recorded automatically or,
- Calls can be recorded on demand with a soft key or by dialling #1. Even in the last 2 seconds of a call, if the record soft key is pressed, it will record the entire call no matter how many transfers may have occurred.
- An email will be sent to an email address you entered in the Edit Extension area and will contain an audio copy of the call.
- You might want to record a call, for example, that could be to protect yourself in the case of verbal orders. A customer could call in and order something and a recording of that call prevents them from saying that they did not order a certain quantity or a colour of something.

Parking a Call

Inbound calls can be parked (put on public hold) by way of a soft key or by dialling ##. This allows you to pick up the call from any other device on the phone system.

- Park Call–## - Parked calls are calls on hold that are available for anyone to pick up.
- Answer Parked (dial lot#) – (Lot # is specified in Main Menu → Advanced Tools)

Do Not Disturb (DND)

All calls directed to the extension will not ring the extension, but will immediately follow a queue rule. This is activated only on a hard key from the phone.

Do Not Disturb – A hard key on certain phones used to ignore incoming calls. Your inbound calls will follow a previously-configured rule such as forwarding or voicemail.

Roaming Extensions

This feature allows an agent to log into any phone on the system and carry the extension with them.

- Log out of the existing telephone. Use dial code *12
- Login to the new phone. Use dial code *11. When asked to enter your extension number, do so, then press pound. When asked to enter your pass code, do so, then press pound.
- *13 will give you a status of your login state

Login/Logout of Queue

Any extension on the system has the ability to log itself in or out of queue using the following codes:

- Login: *16 + (Queue Number) + Dial/Send
- Logout: *17 + (Queue Number) + Dial/Send

Logging in and out of an extension can only be done remotely if the number performing the login or logout has a virtual extension on the system. In order to achieve this, a call must be placed to an IVR. Once the IVR answers, assuming the call display associated with the number calling is visible and a virtual extension is pointed to that number, then the following commands will achieve the desired result:

- Login: Dial 016 + (Queue Number)
- Logout: Dial 017 + (Queue Number)

In the event that the device you are calling from has its caller ID number blocked and the system isn't aware the device is tied to a virtual extension, the following command must be entered:

- Login: Dial 016 + (Queue Number) + * + (3-5 digit Virtual Extension)
- Logout: Dial 017 + (Queue Number) + * + (3-5 digit Virtual Extension)

Agent Ready/Not Ready

All phones by default are set to Ready, if you would like to set yourself to Not Ready you must dial *908 and enter a reason code supplied to you by your company. An agent will not receive calls from a queue until his/her phone is set back to Ready. To return to Ready, simply dial *908 again.

Call Monitoring

This feature is often used to monitor call substance from a managerial level to a caller and an agent, and can be useful in training. An extension must have previously been given authorization to use the monitoring feature. To give permission, go to Main Menu → Phone System Settings → Extensions → Edit Extension and check Can Monitor Calls and click Save. In some cases you'll want to exclude extensions from being monitored and which is called Privacy Exclusion. This can be done in the same menu as above. Simply choose the extension you would like to exclude and select the Privacy Enabled check box and press Save. If supervisors try to monitor an extension that has Privacy Enabled, they will be prompted that they do not have permission to monitor that extension.

A. Silent Mode

Silent monitoring allows a supervisor to monitor an agent without the agent knowing. The agent or caller will not hear the supervisor. To begin using this feature use the star code *111. You will be prompted to enter the extension and press #. When you are finished monitoring, simply hang up.

B. Coaching Mode

Coaching allows a supervisor to coach agents during a call. When supervisors speak, they will be heard by the agent but not the caller. To begin using this feature use the star code *112. You will be prompted to enter an extension and press #. When you are done monitoring, simply hang up.

C. Barge-In Mode

Barge-In mode allows a supervisor to barge into a call. This essentially turns the call into a three-way conference. All parties will be heard when they speak. To begin using this feature use the star code *113. You will be prompted to enter an extension and press #. When you are done, simply hang up.

Paging Groups

Telephone-wide intercom and group-specific paging can be set up. For example, if you would like to page only the sales group, that feature is set up here. Soft keys are used to activate this feature, or by dialling a page extension set up at Main Menu → Phone System Settings → Paging Groups → Edit Paging Group. Page extensions can specify a group to page such as, service, sales, accounting, etc., and only the phones set up in those respective groups will follow their respective rules.

Anonymous Calling

In the event you want to make a call but do not want to display your identity, dial *67, wait for a dial tone and dial the number. Your number will not be displayed on the recipients' phone.

Call Transfer

Transfer – A hard key on certain phones used to transfer attended or blind.

Transfer Attended:

- While on a call, press the transfer button.
- Dial the number to transfer and press send.
- To transfer the call, press Transfer.

Transfer Blind:

- While on a call press the Transfer button.
- Press the Blind button.
- Dial the number to transfer and press Send to send a call to another agent or extension.
- Conference – A hard key or soft key on phones used to allow more than one person to be part of a call.

Telephone Feature Codes

By using the dial codes, features can be enabled or disabled.

Extension Features Codes	Dial Code	Soft Key*	Hard Key**
Extension Login	*11	Login	N/A
Extension Logout	*12	Logout	N/A
Extension Status	*13	Status	N/A
Extension Ready (Queue)	*14	Ready	N/A
Extension Not Ready (Queue)	*15	Not Rdy	N/A
Queue Login	*16 + Queue	N/A	N/A
Queue Logout	*17 + Queue	N/A	N/A
Queue Forwarding	*104 + Ext. #	N/A	N/A
Queue Forwarding Toggle	*104 + Queue #	N/A	N/A
Extension Login/Logout Toggle	*904	In Out	N/A
Extension Ready/Not Ready Toggle	*908	Rdy?	N/A
Access Voicemail system	*98	N/A	Messages
Block name/number per call	*67+Number	N/A	N/A
Call forwarding Enable	*72	N/A	N/A
Call Forwarding Disable	*73	N/A	N/A
Record Call	#1	Record	N/A
Hold Call (for local phone only)	N/A	Hold	Hold
Park Call	##	Park	N/A
Answer Parked (dial lot #)	50-70	Park xx	N/A
Redial	N/A	N/A	Redial
Do Not Disturb	N/A	N/A	Do Not Disturb
Transfer	N/A	Transfer	Transfer
Conference	N/A	Conf	Conference

Intercom Call	*22 + Ext	N/A	N/A
Remote Call Pickup	** + Ext	N/A	N/A

Manager Feature Codes	Dial Code	Soft Key*	Hard Key**
Call Monitoring	*111	Monitor	N/A
Call Coaching	*112	Coach	N/A
Call Barge-in	*113	Barge	N/A
Record Names for Staff Directory	*88 + Ext		

Meet-Me Room Feature Codes	Dial Code
Help Menu: Announces all the * codes and their function.	*1
Reach Operator: When dialed by any participant, they will be redirected from the meet-me room to the local operator as defined in Phone System Settings → Miscellaneous Settings.	*0
Mute / Un-Mute Self: Toggles the mute/un-mute status of only the person who dialed the * code.	*6
Participant Count: This option will privately read out the current participant count for the meet-me room. This option is for administrators only.	*91
Lock / Unlock: This option will lock or unlock the room. This option is for administrators only.	*94
Mute All Participants: This option will mute all participants, except for other admin users. This option is for administrators only.	*96
Un-Mute All Participants: This option will un-mute all participants. This option is for administrators only.	*97
Record Call: Dialing *22 during the call will start to record the call in its entirety. Once all participants have left the call, a secure (MD5 hash based) recording URL will be emailed to the email (addresses) specified in the meet-me room settings.	*22

*Soft keys are programmable and may not show up on all phones
** Hard key labels may vary between models.

Speed-Dial Keys

Speed-Dial Keys can be added and managed from your telephone set or through the portal under Phone System Settings → Extensions.

Voicemail

The SolSwitch includes a voicemail system. Among the features that it offers are:

- Three different types of greetings.
- Notification of new messages through email. The notification message may also include an audio file containing the complete message.
- Mailboxes may be password protected.
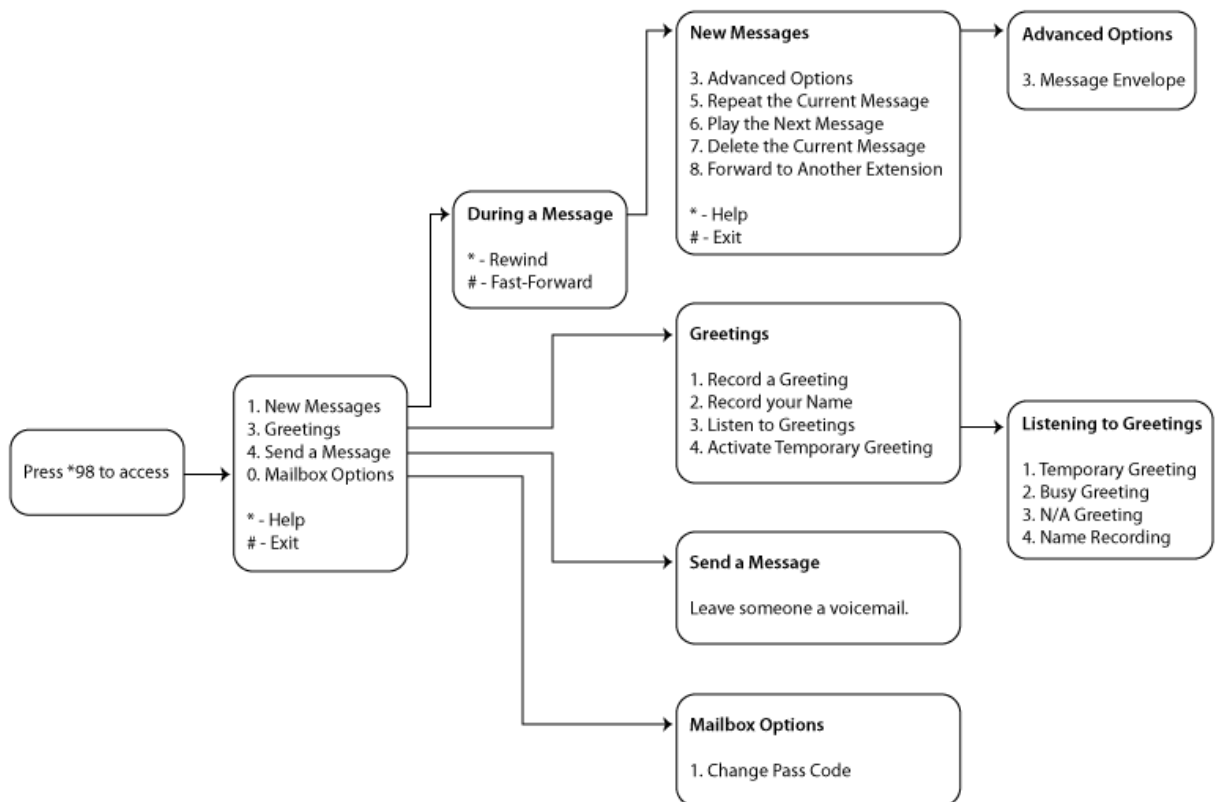- Can optionally allow callers to review and re-record messages.

Accessing the Voicemail

Locally:

- From any phone, press the messages key, or dial *98.
- Alternately, dial the extension number of the voicemail you want to access.

Remotely:

- Dial * when the voice prompt begins to handle your call in your voicemail then enter your pass code.

Types of Greetings

A greeting is intended to let the caller know that you are not able to answer their call. The greeting can also be used to convey other information like when you will be available, other methods to contact you, or other options that the caller can use to receive assistance. The SolSwitch has 3 types of greetings:

A. *Busy Greeting*
If the SolSwitch thinks that you are on the phone, the caller will hear your "busy" greeting. Your busy greeting can either be a generic message, like:

- The person at extension *number* is on the phone.
- *Recorded name* is on the phone or the busy greeting can be a message that you record.

B. *Unavailable Greeting*
If the SolSwitch thinks that you are unavailable (e.g. you don't answer the phone or your phone is not connected to the network) the caller will hear your "unavailable" greeting. The unavailable greeting can either be a generic message, like:

- The person at extension *number* is unavailable.
- *Recorded name* is unavailable. Or the unavailable greeting can be a message that you record.

C. *Temporary Greeting*
You can also record a temporary greeting. A temporary greeting will always be played regardless of your "busy" or "unavailable" status. This could be used, for example, if you are going on vacation and want to inform people not to expect a return call anytime soon, but you do not want to re-record your busy and unavailable greetings when you come back.

Leaving a Message in a Mailbox

If you are not able to answer a phone call, your administrator may have configured the SolSwitch to allow the caller to record a message that you can listen to later. Depending on how the SolSwitch system was configured, the caller may first hear a greeting and/or some instructions before being allowed to record a message. The process of leaving a message looks like this:

1. Play the appropriate greeting. The SolSwitch administrator can control whether a greeting is played or not and can also control whether the busy or unavailable greeting is played (if you record a temporary greeting, the temporary greeting will always be played if the administrator does not turn off greetings).
2. Play some short instructions. Again, the SolSwitch administrator can control the instructions.
3. Play a beep.
4. Record the message, and optionally allow the caller to review the message.

While listening to the greeting or the instructions, the caller can press any of the following buttons:

- "#" Skip the rest of the greeting and instructions and immediately begin recording the message.
- "*" Transfer out of the message recording application. Typically used by the SolSwitch administrator to allow users to access their mailboxes.
- "0" Transfer to an operator.

After the greeting and the instructions have played the caller will hear a beep and then the system will start recording a message. To end the recording, the caller can hang up the phone or press the "#" button.

If the caller hangs up the phone to end the recording, the SolSwitch system will put the message in your "New" folder.

If the caller pressed the "#" button to end the recording and the SolSwitch administrator has configured the system to let the caller review the message, the caller will be presented with the following options:

- "0" Transfer to an operator.
- "1" Save the message.
- "2" Review the message.
- "3" Re-record the message.

If the caller presses "1" the message will be placed in your "New" folder. If the user presses "0" the message will be cancelled and the user will be transferred to the operator.

Logging Into your Mailbox

Your phone may have a special button on it that has been configured to allow access to your mailbox. If not you can access your voicemail by dialling *98.

To access your voicemail externally simply call your extension and press * when your voicemail answers. You will be prompted to enter your pass code.

The First Time You Log In

The first time you log into your mailbox, you may be required to change some settings. In particular, you may be required to:

- Record your name.
- Record your busy greeting.
- Record your unavailable greeting.
- Change your password.

Voicemail Menu Options

As outlined in the above diagram, each menu within the voicemail system has its own set of commands.

A. *Main Menu*
"1" - Listen to messages in the currently selected folder. If you have any new messages, the "New Messages" folder will be selected. If you have no new messages, the "Old Messages" folder will be selected.
"3" - Greetings.
"4" - Send a Message.
"0" - Mailbox Options.
"*" - Repeat the menu options.
"#" - Exit from the voice mail system.

B. *Greetings*
"1" - Record your Greetings.
"2" - Record your Name.

"3" - Listen to Greetings.

"4" - Activate Temporary Greeting.

"#" - Return to Main Menu.

C. Mailbox Options

"1" - Change your password.

"#" - Return to Main Menu.

D. During Message Playback

"*" - Rewind the message by 3 seconds.

"#" - Fast forward the message by 3 seconds.

"0" - Pause the message playback. Press any other button to resume playback.

"1456789" - Pressing any of these buttons will halt the playback of the message and perform the corresponding action from the post-playback options.

E. After Message Playback

"3" - Advanced options.

"4" - Go to the previous message in the folder.

"5" - Replay the current message.

"6" - Go to the next message in the folder.

"7" - Delete or undelete the message.

"8" - Forward the message to another user on the system.

"9" - Save the message to a different folder.

"*"- Replay the prompt.

"#"- Exit the voice mail system.

F. Advanced Options

"3" - Play the message envelope.

G. Advanced Options

The voicemail system will play back the message "envelope". Depending on how the voice mail system has been configured, the message envelope may consist of the following items:

- Position.
- Date and time that the message was received.
- Caller ID information.
- Duration of the message.

Pressing "1" any time during the playback of the message envelope will skip to the message playback. The voice mail system will then play back the message.

Glossary

AGENT – End user using the telephone system.

ATA – Analog Telephone Adapter required when older technology phones are used.

CALL LEG– Call Leg is a discrete segment of a call connection in a VoIP network. A call leg is a logical connection between the router and either a telephony endpoint over a bearer channel or another endpoint using a session protocol.

CDR – Call Detail Record

CID – Caller Identification

DID – Direct Inbound Dial all telephone numbers including toll-free numbers.

DTMF mode – Dual Tone Multi-frequency Signalling Used for telecommunication signalling over analog telephone lines.

IP PHONE – Internet Protocol phone

IVR – Interactive Voice Response Having a recording prompt the caller to choose options.

MWI (Message Waiting Indicator) – Flashing red light on phones that indicates someone left a voice message.

NANP – North American Numbering Plan

NAT – Network Address Translation is the translation of an Internet Protocol address (IP address) used within one network to a different IP address known within another network.

NPA – Numbering Plan Area Code the first three digits of a 10 digit North American Numbering Plan (NANP) number, commonly called the Area Code.

Solutions that shine

NPX – Central Office Code the first three digits following the Area Code in a 10-digit NANP number, commonly called the telephone *exchange* or *prefix*

OCN – Operating Company Number

PARK – Public hold

PBX – Private Branch Exchange

PoE – Power over Ethernet

SIP – Session Initiation Protocol

SL – Service Level used for reporting inbound call-to-answer ratios.

TRUNK – A trunk is a communication line between two switching systems.

TCP – Transmission Control Protocol.

UDP – User Datagram Protocol

VoIP – Voice over Internet Protocol

WEIGHTING – Call Priority