Advanced System Configuration Guide



Fourth Edition

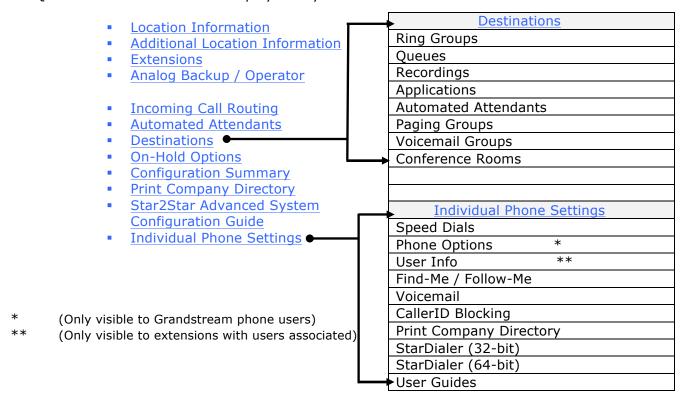
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Introduction To The Guide

This guide is designed to help you get the most out of your Star2Star experience. It provides a detailed explanation for creating successful "end-point results" which if established correctly will deliver the best possible call routing options to meet your business requirements. To achieve complex call routing and realize full feature benefits via our easy to use web portal, this guide

provides detailed explanations, "StarTips" and examples on system setup and configuration. It also includes use of the "Automated Attendant", "Destinations" options and more as listed below. Use Destinations as the building blocks to create the system that will meet your business requirements. The initial system setup will contain a number of default configurations for your use i.e. Ring Group "Ring All (Business Hours)", Ring Group "Ring All (After Hours), Voicemail Access, Company Directory. Create your customized business requirements for call routing / call treatment to best suit your type of business e.g. Specific Ring Groups, Queues, Recordings, Voicemail Groups etc. Always remember to create meaningful names for these groups especially Ring Groups and Queues as the name can be displayed to your extension users on a call:



(Note) Destinations can be created within the Portal without the need to continually click

Save Changes to PBX

. When you have completed your work click

Changes to take effect on the system. Be aware that this can take up to 15 minutes to take effect.

Terminology Guide

Additional Location Information: The Additional Location Information link provides access to detailed configuration information about your system. Use this link to set the following items; Personal Authorization Codes, Interoffice Prefix, Time Zone, Enable Advanced Scheduling, Company Directory Options, Display Five Digit Extension, Key System Mode and Park Positions, Primary DID, Business Hours and After Hours Contact information.

Auto Attendant: An automated attendant is a recorded message that initially greets callers and allows them to dial an extension or navigate a menu to reach the desired person or department. Each digit 0-9, the # key, and the * key can be configured with an action. These actions can include items such as a ring group of members of a department, dialing an extension directory, listening to a recorded message, reaching the "Operator" function. (Note) Any new Auto Attendant menu will require a recording. Please create or upload a recording to enable the menu to work correctly

Analog Backup: Using an ATA this feature, if configured on your system, can provide access to 911 and limited outdialing via a traditional analog line in the event of an Internet failure. Users that have the rights "Allow 411 calls", checked in the Extension settings of the Star2Star Web Portal, can access 411 Directory Services via the analog gateway. Star2Star does not provide access to 411 other than using the analog telephone adapter.

411 (see above)

Directory Services (see above)

Analog Outdial: This feature, if configured on your system, can provide access for users to outdial using the ATA without the system being in a "backup" state i.e. Internet failure. To utilize this feature dial ** followed by the number required. The call will be made using the analog gateway.

CAC (Call Accounting Code): A CAC provides the ability to account for calls based upon number dialed and call duration. This information is reflected in the Call Detail Records (Reports). Typical application would be when a customer charges back to a client. The codes are enabled and added in the Phone System Configuration / Extensions section of the portal. If enabled the system prompts for the Call Accounting Code. However calls can be made without using a CAC simply by pressing # when prompted.

Call Park: Call Park is a feature that enables users to "Park" callers and then handle another call/action. This feature is enabled via the Additional Location Information link. To park a call and retrieve from a different phone perform an attended (Announced) transfer to 7000. The system will play back a retrieval code (7001 – 7099). The system provides up to 99 park positions. When the caller is "Parked" they will hear Message / Music on hold if enabled on your system. To retrieve the call, dial the retrieval code to reconnect to the call. If the call remains parked for longer than the System Call Park Timeout the call will ring back on the originating extension. **(Note)** When using the snom 320 / 360 / 370 phones your system can be programmed to operate in "Key System Mode" when parking calls (provides a visual park appearance). This feature is not available on the snom 300 phone. When a call is active on your phone

press the key to park the call. The system sets a default value of 4 calls that can be parked on your phone in this manner and will be visible to other snom 320 / 360 / 370 phone users. This value can be changed in the Additional Location Information for your system. Each call occupies a different park/line position that will be visible (solid yellow LED) to other phone users on your system. You can retrieve the parked call from the respective park/line position from any snom 320 / 360 / 370 phone on your system. A single button call park (shared park) feature is available on the Cisco® SPA303, SPA502G, SPA504G, SPA508G, SPA509G and the SPA525G2. Set **Key System Mode** on to display shared park appearances on

the SPA508G or SPA509G. You may also display shared park appearances on the SPA500S Attendant Console fitted to the SPA502G, SPA504G and the SPA525G2. **Key System Mode** is set in the

Additional Location Information section for a customer in the Star2Star Web Portal.

Yealink phone models T26P, T28P and T38G in "Key System Mode" have single key park and up to 9 shared park positions available.

Conference Rooms: The system provides the facility to create Conference Rooms that multiple callers can access. Callers are connected together as per a normal conference facility. If configured, users will be prompted for a PIN. To access the room via an external line point a DID to the Conference Room via the Incoming Call Routing option. To access the Conference Room internally use the Internal Direct Dial number.

CNAM: Calling **NAM**e A service that displays the caller's name on the calling party's digital readout. This is similar to caller ID except that the calling party's name is displayed along with the calling number or instead of the calling number.

DEALTRACK: Is the Dealer "Self Serve" application that provides the following functionality:

- Access via an option in the Star2Star web portal "DEALTRACK"
- Self serve
- Streamlined pre-qualification process
- Provides an easy convenient way to track deals
- Ability to upload the Star2Star Sales Quote Calculator
- Upload Quote Calculator Revisions
- Option to "Finalize" with the final revision
- Automatically creates the customer location in the portal
- Ability to start the SNA (StarNet Analyzer) process
- Ability to place a porting pre-qualification i.e. is this number portable?
- Ability to request DID/Number pre-qualification
- Ability to request circuit pre-qualification

Destinations: Ring Groups, Queues, Recordings, Applications, Automated Attendants, Paging Groups, Voicemail Groups and Conference Rooms are all classed as Destinations. Calls can be directed toward Destinations.

DID's / Numbers: A DID (Direct Inward Dial) is a telephone number on the Star2Star system.

Failover: Is a feature that allows another call routing action for Ring Groups, Queues, Recordings, Dial External Number (Application) and Menus. E.g. If a caller reaches a Ring Group that rings for 36 seconds use the Failover option to determine how that call is handled after the initial ring time has expired. You can set 9 levels of Failover. Extensions and Mailboxes are endpoints and as such no further Failover option is configurable.

Find-Me / Follow-Me: Users are provided with powerful functionality to re direct calls via an intuitive web screen. Callers will follow the Find-Me / Follow-Me configuration, if set, to locate their party. There are 3 configuration styles available for this functionality:- Find Me (Immediate) Calls are immediately diverted / forwarded to an alternative number. Find Me (Sequential) Calls can follow a sequence of ringing up to 4 alternative numbers in turn with varying timeout values. Find Me (All) Calls can ring up to 4 alternative numbers simultaneously.

Internal Direct Dial: All Destinations are given an Internal Direct Dial (IDD) number by the Star2Star system. You may use this 3-digit number to locally dial a Ring Group, Queue, Recording, Application, Menu, Paging Group, Voicemail Group and Conference Room to test or use within your application.

Key System Mode: With snom phones (snom 320/360/370) and Cisco SPA 508G / 509G phones, your system can be configured to emulate a squared key system using visual shared park positions by setting "Key System Mode" to on. This feature is enabled via the Additional Location Information link.

Lines: Are the number of concurrent, out of Star2Star network calls, your system is configured with. This is for both outbound and inbound.

Location Information: The Location Information and Additional Location Information links provide access to detailed information about your system. Use this link to enter/modify Location address information, Billing Contact information, Business Hours/Schedule, After Hours Contacts and system configuration options. Initial location Information is automatically created via DEALTRACK.

MOH - Music/Message On Hold Pre recorded music or messages can be uploaded onto the system for playback when callers are on hold. Use the On-Hold Options link to access the up load feature. Many popular formats are supported e.g.

- 128bps MP3
- 160bps AC3
- 160bps MP2
- 16bit Level5 FLAC
- 16bit PCM WAV
- 180bps WMA
- Apple Signed 16bit PCM AIFF
- GSM WAV
- OGG

Multi-Path RCF Remote Call Forwarding (RCF) on business accounts with multiple lines. Multi-Path RCF is available with select Independent Local Exchange Carriers (ILECs) i.e.: Verizon, Competitive Local Exchange Carriers (CLECs) i.e.: Level 3, and Regional Bell Operating Companies (RBOCs) i.e.: AT&T. Not all providers offer this service. Universal "Feature Codes" *XX to activate and *XX to deactivate are pretty standard across the industry, however, can differ from provider to provider. They can usually be found in the local provider's phonebook. There are some providers that do not offer "User Activated" RCF feature and will require a direct call to the providers business office to activate/deactivate for each use. There are a couple providers that only provide RCF services on a "Day-to-Day" basis. This means that each and every day the customer would be required to reestablish the feature, whether via a "Feature Code *XX or requiring direct contact to the provider's business office to reestablish the Multi-Path RCF. When requesting this service to be activated on the account, you must request "Multi-Path RCF", otherwise you will only receive standard "One Path (one call at a time) RCF".

Each provider has their own set of charges for RCF. Some will charge a flat monthly reoccurring fee for enabling this feature on the account, while others may charge a setup fee, reoccurring fee, per path fee, and possibly a minute of use (MOU) fee. These fees and requirement can be identified by calling the providers business office and asking.

New Number: This is the Portal link under PHONE SYSTEM CONFIGURATION that provides access to the Pre Qualification (Prequal) tab to determine if numbers in a specific area are available. You can also request new numbers using the New Number tab. Requests for new customers should be made via DEALTRACK.

Operator: The Operator function, if configured on your system, will determine the action / call treatment for callers who press "0" when listening to a user's voicemail box or the company directory, reaching the Auto Attendant (if enabled) or holding in a queue (if enabled). This is the setting for the default operator action. Pressing "0" on an extension will route the call to the Operator setting. Specific actions can be set for individual queues or Auto Attendant menus.

PAC (Personal Authorization Code): A PAC provides the ability to ensure all phone users enter a code to dial any non-local (> than 7 digits) number. A call cannot be completed without a valid code. This information will be reflected in the Call Detail Records (Reports). This feature is enabled via the Phone System Configuration / Location Information section of the Portal. PAC codes are entered in the same Location Information page.

Paging Group: A Paging Group provides the ability to directly connect to group Members phones via the telephone's external speaker. The member receives an initial alert prior to connection. This feature is particularly useful for company wide alerts or to specific groups of people. (Note) Paging groups are limited to 10 members due to system resource constraints. For larger applications Star2Star recommends the use of external paging equipment that can be connected to the Star2Star system via a separate analog adapter or the snom PA1 paging device. Please contact the support organization for details.

Porting: This is the Portal link under PHONE SYSTEM CONFIGURATION that provides access to the Pre Qualification (Prequal) tab to determine if a specific number(s) can be ported. You can also select numbers to port using the Porting tab. Requests for new customers should be made via DEALTRACK.

Queue: A Queue provides the ability to ring Members in a sequential operation i.e. one at a time. The Queue logic algorithm used is "Round Robin with memory". Callers placed in Queue will hear Message or Music on hold if enabled in the Advanced Settings features available under the On-Hold Options link in the Star2Star portal. Music or Message on hold (MOH) can be set as queue specific. There are 9 levels of Failover available when using Queues. These are set in the same manner as Ring Groups. You can record a message that callers hear prior to entering the Queue. You can display the Queue name to Members on incoming calls. This is particularly useful to distinguish between different types of calls using the phone display e.g. Queue 1, Queue Priority etc. For a more feature rich Call Center Application see StarCenter.

Receptionist Mode: Generally a call will always appear on the first line appearance of a Polycom, Cisco SPA or snom phone unless the system has been configured to point a DID / Line to a specific line appearance on the phone. With a user busy on a call, notification of a subsequent call will be via a call waiting tone (if configured) in the ear and a phone display change. This is not always convenient or practical for a busy administrator. Use the **Receptionist Mode** option (see portal configuration Extensions link for the available phone models) to ensure that incoming calls to the extension will roll down the line appearance keys.

Recording: A Recording is used as a prompt or message that can be played to a caller when entering a Queue or during any stage of Failover in a Ring Group or Queue. A recording can also be used to provide customized access to a Conference Room. The recording can be created by following the prompts and either dialing internally using a StarPhone or by dialing the 800 number and using an access pin. Once recorded the user can play back, save or re record. You may also upload a pre-recorded message in many popular formats e.g.

- 128bps MP3
- 160bps AC3
- 160bps MP2
- 16bit Level5 FLAC
- 16bit PCM WAV
- 180bps WMA
- Apple Signed 16bit PCM AIFF

- GSM WAV
- OGG

Ring Group: A Ring Group provides the ability to ring one or more phones simultaneously. Members are

added to the Ring Group via the Destinations link or when editing menus using the tool set icon adding Members to the Ring Group the system provides the ability to select which line appearance is used on the phone (If your phone has multiple lines). This is particularly useful if a distinctive ring is set on that phone for the line appearance used. Polycom phones provide this feature. If this is set the user will receive a different visual and audible notification based upon which line is calling. The Ring Group can be accessed via an Internal Direct Dial number. This is particularly convenient when testing the configuration. Remember to set the period of time your group will ring before reaching your Failover options. For example a ring time of 36 seconds is approximately 6 rings. The Failover option supports up to 9 levels of failover. Remember that selecting another extension or Voicemail is regarded as an end point and as such no further Failover options can be selected. You can display the Ring Group name to Members on incoming calls.

StarCenter: The innovative and feature rich call center application from Star2Star Communications. This application requires no additional on-site hardware to function. Key features include: Agent login/logout, Advanced Ring Strategies (Round Robin-Skills Based-Best Match-Ring All-Least Recent-Fewest Calls-Random), Multiple (10) Action Keys, Enhanced Queue/Agent Reporting, Flexible Announcements (Position in queue-Wait time-Periodic-Hold Time Announcement to the Agent), Real Time Queue Monitoring, Multi-Q Monitoring, Administrative Monitor/Barge In/Whisper and CallerID Routing. For more details see StarCenter.

StarNet Analyzer (SNA): The Star2Star StarNet Analyzer (SNA) is a Java applet that runs onsite for a minimum of 48 hours (business hours) testing the customers internet connection. We measure several items, including speed, packet loss, jitter, round trip times, which provides you with a graded report card that will also display MoS scoring. The SNA collects several metrics that are critical to voice over IP communications and is therefore a **mandatory** test for all installations.

StarScope: A powerful presence management application with call control from Star2Star Communications. Currently available in two versions: Operator (Used by Operators/Administrators/Managers/Call Center) and Casual (Basic user version). **(Note)** the Operator version provides additional features e.g. Call Steal, multiple extension line appearances. For more details see StarScope.

Star2Star Application Framework System Performance Check (SPC): The Application Framework - like most software packages - has certain system requirements that must be met for smooth operation.

Some older Windows systems, particularly systems with older graphics cards and/or outdated graphics drivers, may not be able to run the Application Framework.

To save time and prevent frustration, our development team created a free program called **System Performance Check (SPC)**. This program quickly checks a Windows PC for compatibility with the Application Framework.

If the **SPC** determines that a system can't run the Framework, it will offer suggestions to help correct the problem.

Location Information

Description

The Location Information and Additional Location Information links provides access to detailed information about your system. The location is automatically created by DEALTRACK. Use these links to enter/modify Location address information, Billing Contact information, Business Hours/Schedule, After Hours Contacts and a number of key system features e.g. Directory Sort Order (you can set this option to play the directory to callers by First Name or Last Name), Multi-Location Directory (you can configure the system to play the directory to callers for all locations associated with your system), Announce Extension on Directory (this announces the selected directory entry to the caller), Key System Mode (Call Park and Retrieval using visible shared park positions on certain phone types), Park Positions (select the number of park positions required), Enable Personal Authorization Codes (authorization code is required when dialing out), Enable Advanced Scheduling (Create detailed routing rules for your DIDs / Numbers).

Setup

- 1. Access the Star2Star portal and select the CONFIGURATION option at the top of the page.
- 2. Select the required location.
- 3. Select the <u>Location Information</u> or <u>Additional Location Information</u> link in the **PHONE SYST EM CONFIGURATION** Menu.

Advanced Settings

- E911 Zone Configuration
- Incoming Call Routing
- Automated Attendants
- Destinations
- StarCenter Manager
- Star2Star Application Framework
- Multi-Queue Monitor
- StarView for StarCenter
- StarCenter Recordings
- Recording Settings
 On-Hold Options
- On-Hold Options
- Custom Devices
 FXS Devices
- Inbound CallerID Blocking
- Asset Management
 StarScope Manager
- Configuration Summary
 Print Company Directory

Individual Phone Settings

- Individual Phone Settings
- 4. Click Edit to edit the relevant Location Information. See examples below.
- 5. Please ensure that the After Hours Contact information is complete and accurate.
- 6. (Note) You must Save Changes at the bottom of this screen when done making changes.

Location Information StarAcademy Tarpon -- Tarpon Main Save Changes to PBX Return Basic customer location CUSTOMER INFORMATION information is created Customer Name StarAcademy Tarpon automatically by DEALTRACK LOCATION INFORMATION Existing Locations Edit a Location Prefix Options Address Service Address Name Tarpon Main Address 600 TALLEVAST RD Address (line 2) STF 202 Tarpon Branch 600 Tallevast Rd Ste 202 Sarasota, Florida 34243 SARASOTA State Florida 34243-3254 Dealer Provides First Line Support? Yes ‡ Default Area Code 941 Lines Billing Contact First Name Captain Address 600 Tallevast Rd Multiple locations defined Last Name Morgan Address (line 2) Suite 202 for this Customer. Phone 9412340001 City Sarasota jdineen@star2star.com State Florida Email Billing Method Credit Card + Zip 34243 Save Changes Return **Additional Location Information** StarAcademy Tarpon -- Tarpon Main Save Changes to PBX CUSTOMER INFORMATION Customer Name StarAcademy Tarpon LOCATION INFORMATION Edit a Location Existing Locations Service Address 600 TALLEVAST RD Tarpon Main SARASOTA, Florida 34243-3254 Name Tarpon Main Tarpon Satellite SARASOTA, Florida 34243-3254 600 TALLEVAST RD Address Address (line 2) STE 202 Tarpon Branch 600 Tallevast Rd Ste 202 Sarasota, Florida 34243 SARASOTA City State Florida 34243-3254 Zip Dealer Provides First Line Support? Yes ‡ Default Area Code 941 Lines Billing Contact First Name Captain Address 600 Tallevast Rd Last Name Morgan Address (line 2) Suite 202 Phone 9412340001 City Sarasota Email jdineen@star2star.com State Florida Billing Method Credit Card ‡ Zip 34243 After Hours Contact Never contact Contact up to 0 + hours before or after Default Business Hours + Set your After Hours Contact Always contact information here. **Primary Contact:** Name: Norman Rockwell Phone: 9412340001 Email: gpotter@star2star.com SMS: Name: Aldus Dumbledore Email: aldusdumb@star2sta SMS: Remember to Save your Email: changes. Save Changes

Extensions

Description

Access the Extensions link to Edit / Reset Voicemail Box extensions. Use the Edit option to change configuration information i.e. User, Extension Number, Extension Name, Enable / Disable Voicemail Box, Outgoing CallerID for each line.

Setup

- 1. Access the Star2Star portal and select the CONFIGURATION option at the top of the page.
- 2. Select the required location.
- 3. Select the Extensions link in the PHONE SYSTEM CONFIGURATION Menu.



Advanced Settings

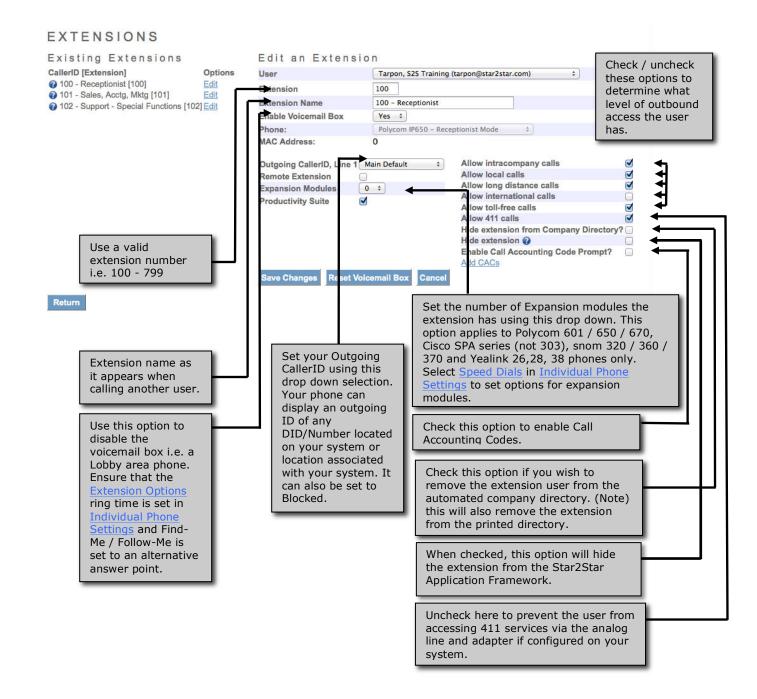
- Incoming Call Routing / Scheduler
- Automated Attendants
- Destinations
- StarCenter Manager
- Multi-Queue Monitor
- StarCenter Recordings
- On-Hold Options
- Inbound CallerID Blocking
- Configuration Summary
- Print Company Directory

Individual Phone Settings

Individual Phone Settings

Extensions (contd)

- 4. Select Edit to amend extension user detail as per the example below.
- 5. (Note) You must select Save Changes or Add Extension at the bottom of the extensions page.
- 6. Remember to Save Changes to PBX take effect. (Note) Changes can take up to 15 minutes to take effect.

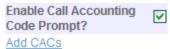


Extensions (contd) - Add Call Accounting Codes

Description

(Call Accounting Code): A CAC provides the ability to account for calls based upon number dialed and call duration. This information is reflected in the Call Detail Records (Reports). Typical application would be when a customer charges back to a client. The codes are enabled and added in the Phone System Configuration / Extensions section of the portal. If enabled the system prompts for the Call Accounting Code. However calls can be made without using a CAC simply by pressing # when prompted.

Setup



- 1. Check the Enable Call Accounting Code Prompt? Option.
- 2. Click Add CACs to Add / View Call Accounting Codes.
- 3. To Add new codes enter the Code: Number and enter a description in the Label: field. See example below.
- 4. To delete existing codes press the 60 icon adjacent to the code you wish to delete.



Analog Backup / Operator

Description

Analog Backup / **Device:** device and action are set up during system installation.

Operator: Use the Operator function to select the action required when a caller presses "0" listening to a user's voicemail box, out of company directory, reaching the Auto Attendant or holding in a queue. This is the setting for the default operator action. Pressing "0" on an extension will route the call to the Operator setting unless the "Personal Operator" feature has been utilized. (See Voicemail guide for details). However specific actions (non- default operator action) can be set for individual queues or menus. (Note) "Out of extension" can be also be disabled.

Setup

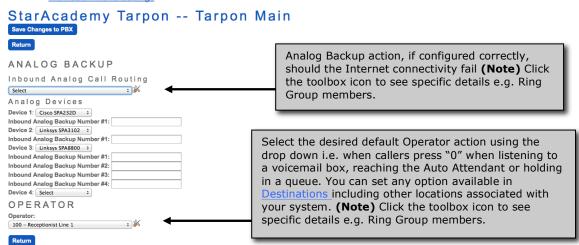
- 1. Access the Star2Star portal and select the CONFIGURATION option at the top of the page.
- 2. Select the required location.
- 3. Select the Analog Backup / Operator link in the PHONE SYSTEM CONFIGURATION Menu. PHONE SYSTEM CONFIGURATION Basic Settings Location Information Additional Location Information Extensions Analog Backup / Operator

Advanced Settings

- . Incoming Call Routing / Scheduler
- Automated Attendants
- Destinations
- StarCenter Manager
- Multi-Queue Monitor
- StarCenter Recordings
- On-Hold Options
- Inbound CallerID Blocking
- Configuration Summary
- Print Company Directory

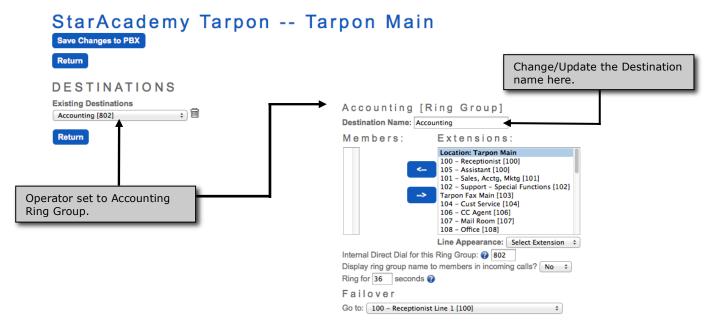
Individual Phone Settings

. Individual Phone Settings

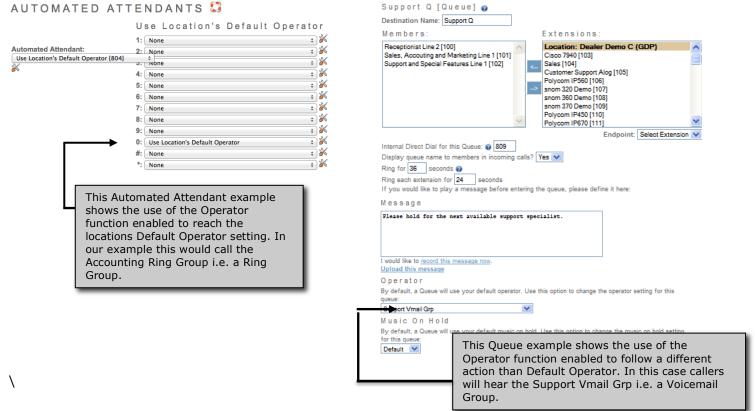


Analog Backup / Operator (contd)

4. In the following example the Operator action has been set to call a Ring Group called Accounting Ring Grp.



- 5. Callers pressing "0" in the Auto Attendant example below will connect to the Locations Default Operator i.e. Accounting Ring Grp.
- 6. Callers pressing "0" in the Support Q (Queue) example below will connect to the Support Vmail Grp (Voicemail Group).



Analog Backup / Operator "StarTips"

- Remember to enable the Operator function in Menus / Automated Attendant if you want callers to have the ability to "0" out.
- You may select an alternative action to the Location Default Operator action for each Auto Attendant Menus.
- Extension Users can set their own "Personal Operator" so that callers pressing 0 when listening to the extension user's voicemail greeting get presented with options specific to that user. (See Voicemail guide for details). This option may be disabled on a per extension basis.
- You may select an alternative action to the Location Default Operator action for each Queue.
- Select Disable in a Menu or Queue if the feature is not required.
- Remember to Save Changes to PBX

 Save Changes to PBX

 (Note) Changes can take up to 15 minutes to take effect.

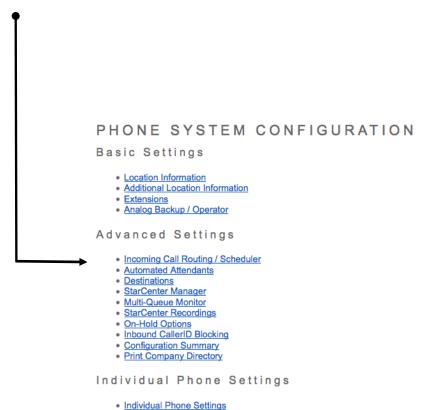
Incoming Call Routing

Description

Click the Incoming Call Routing link to set the appropriate action for routing DID's / Numbers on your system. Calls can be sent to any destination that you have defined e.g. a Menu, Ring Group, Queue, Conference Room or directly to an extension. Calls can also be routed differently based on whether or not the call arrives during company hours or after hours. You can set a different business hours schedule for each individual DID / Number.

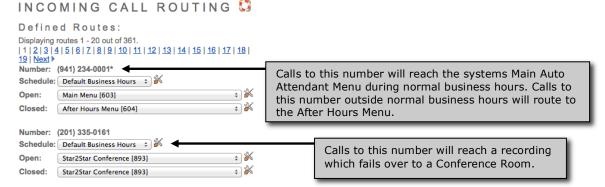
Setup

- 1. Access the Star2Star portal and select the CONFIGURATION option at the top of the page.
- 2. Select the required location.
- 3. Select the <u>Incoming Call Routing</u> link in the **PHONE SYSTEM CONFIGURATION** Menu.

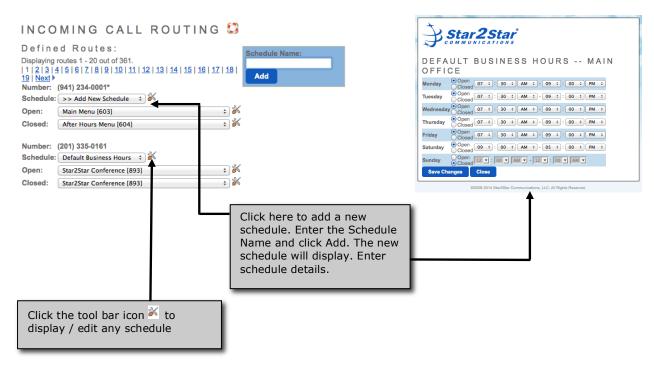


Incoming Call Routing (contd)

4. In the following example the customers system has 2 DID's / Numbers. One points to Conference Intro (Recording) which fails over to a Conference Room defined in Destinations.



- 5. Select the Schedule required for Incoming Call Routing. The default hours are defined in the <u>Location Information</u> link. Default hours are set as 9:00 am 5:00 pm, Monday Through Friday. You may change these values via the <u>Location Information</u>.
- 6. You may also create different schedules to be used for DID's / Numbers. See the example below.



Incoming Call Routing - Advanced Scheduling

The Incoming Call Routing / Scheduler feature provides complete flexibility for Customers wishing to create complex routing rules for a 7-day week. Each number/DID on the system can have its own Incoming Call Routing schedule. Schedules can be as basic as adding a lunch break step for each business day or as detailed as providing multiple steps that differ for every day of the week including the weekend. Each schedule can be given a name that clearly describes its function. By default the schedules are named using the associated number / DID e.g. (941) 234-0001 Schedule. This can easily be changed to reflect a more meaningful name.

(**Note**) when this feature is implemented for the first time, schedules are created for all DIDs / Numbers using the naming convention mentioned above. Previously created Incoming Call Routing rules will continue to function normally. If you turn this option on and create advanced schedules, those schedules will not be available if you subsequently turn the feature off. Scheduling would return to the original preadvanced scheduling state.

Routing schedules can only be created using existing Destinations that have been previously created and are fully functional i.e. Saved to PBX. To learn more about creating Destinations, review the Star2Star Advanced System Configuration Guide which is located in the PHONE SYSTEM CONFIGURATION section of the Star2Star Web Portal.

The full screen view for a schedule displays in 15 minute time segments however Start Time and End Time for each action within a schedule is available in 5-minute time increments.

Saving changes to a schedule followed by the subsequent "Save Changes to PBX" is a now rapid process.

To enable the feature click the Star2Star Web Portal **Additional Location Information** option for the relevant location. Select the **Enable Advanced Scheduling?** option. Set to **Yes.**

Incoming Call Routing "StarTips"

- vert Click the tool bar icon vert to provide details for routing or schedules.
- You may edit the Default Business Hours Schedule in <u>Location Information</u> or by clicking the tool bar icon in the <u>Incoming Call Routing</u> display. Remember to <u>Save Changes</u> for the schedule.
- ho Click the icon ho to display help including a full configuration guide and video.
- Use the Advanced Scheduler Option set in Additional Location Information to create complex routing rules for a 7-day week.
- Remember to Save Changes to PBX

 Save Changes to PBX

Automated Attendants

Description

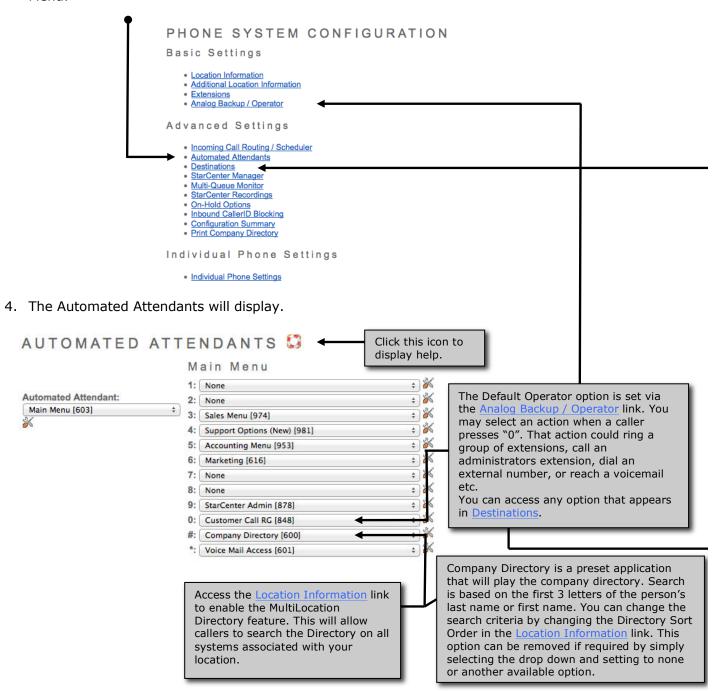
An automated attendant is a recorded message that initially greets callers and allows them to dial an extension or navigate a menu to reach the desired person or department. Each digit 0-9, the # key, and the * key can be configured with an action. These actions can include items such as a ring group of members of a department, dialing an extension directory, listening to a recorded message, reaching the "Operator" function see section **Analog Backup / Operator** etc.

Internal Direct Dial -- By using this extension, you can dial this destination directly from within your company.

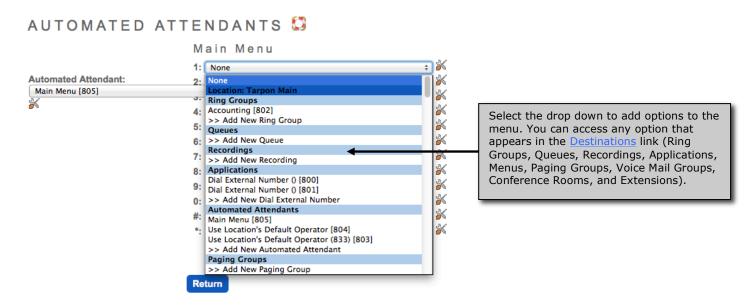
Failover -- For auto attendants and recordings, this time specifies how long the system will wait after your auto attendant message or recording has played. For groups, this time specifies how long the call will ring. After this length of time has been reached, the system will proceed to your failover option.

Setup

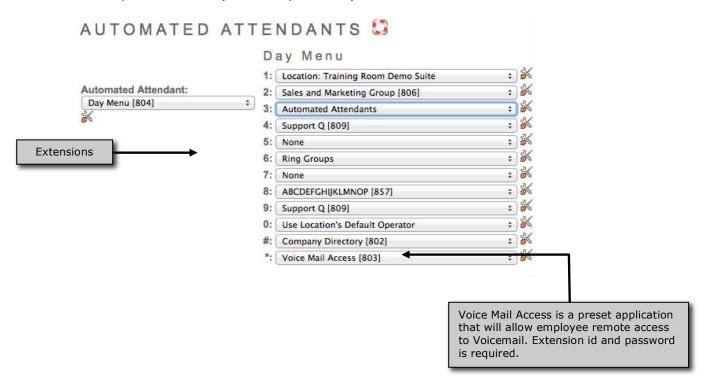
- 1. Access the Star2Star portal and select the CONFIGURATION option at the top of the page.
- 2. Select the required location.
- Select the <u>Automated Attendants</u> link in the PHONE SYSTEM CONFIGURATION Menu.



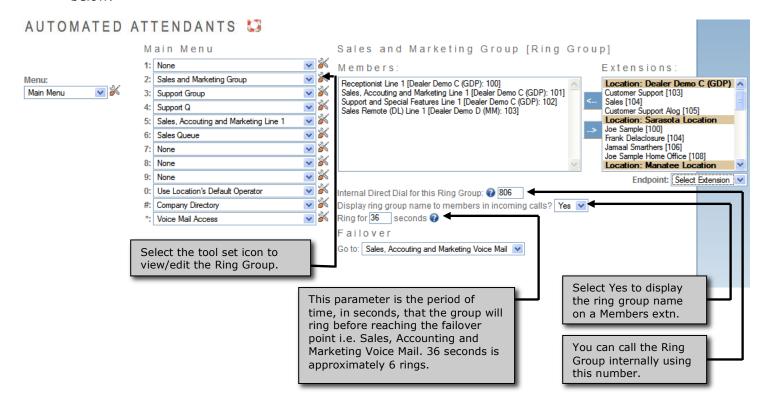
5. Adding options to the menu.



6. More options entered (see example below).

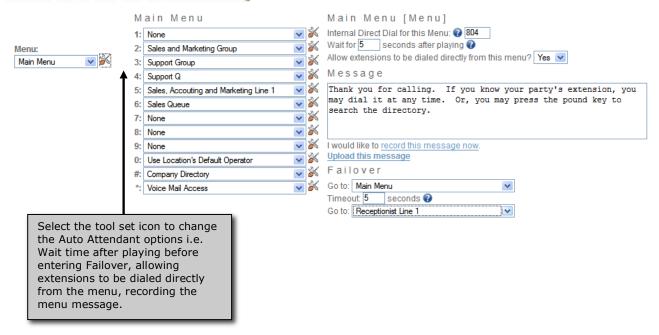


7. View / Edit Ring Group in the menu. You can edit selected options via the menu. See example below.

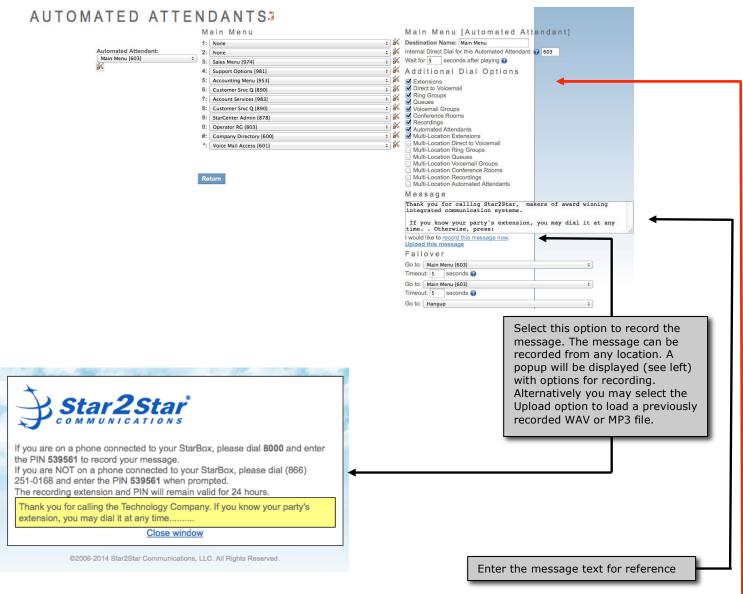


8. This menu example is complete; however it will NOT work until a recording is added.

AUTOMATED ATTENDANTS



9. Creating or loading a recording (Message) for the menu.



10. A number of new Dial Options are available for direct dial through an Automated Attendant.

The new dial through features can be turned on / off at an Auto Attendant level.

- Extensions / Multi-Location Extensions We have always had an option to add a "Dial Extension" option to the Auto Attendant. By default, that feature worked across multiple locations. The feature now allows system admins to turn the "dial across locations" feature on or off.
- · **Direct to Voicemail** A caller can now send a message directly to a user's voicemail by dialing 6xxx (xxx = extension number).
- **Ring Groups** Callers can reach a Ring Group by dialing the [IDD] Internal Direct Dial number of the Destination e.g. 8xx.

- **Queues** Callers can reach a Queue by dialing the [IDD] Internal Direct Dial number of the Destination e.g. 8xx.
- · **Voicemail Groups** Callers can reach Voicemail Groups by dialing the [IDD] Internal Direct Dial number of the Destination e.g. 8xx.
- · **Conference Rooms** Callers can reach Conference Rooms by dialing the [IDD] Internal Direct Dial number of the Destination e.g. 8xx.
- **Recordings** Callers can reach Recordings by dialing the [IDD] Internal Direct Dial number of the Destination e.g. 8xx (this especially useful when accessing a recording that then fails over to a conference room).
- **Automated Attendants** Callers can reach an Auto Attendant by dialing the [IDD] Internal Direct Dial number of the Destination e.g. 8xx.

Destinations

Description

Ring Groups, Queues, Recordings, Applications, Automated Attendants, Paging Groups, Voicemail Groups and Conference Rooms are all classed as Destinations. Calls can be directed toward Destinations.

Setup

- 1. Access the Star2Star portal and select the CONFIGURATION option at the top of the page.
- 2. Select the required customer location.
- 3. Select the Destinations link in the PHONE SYSTEM CONFIGURATION Menu.

PHONE SYSTEM CONFIGURATION Basic Settings Location Information Additional Location Information Extensions Analog Backup / Operator Advanced Settings · Incoming Call Routing / Scheduler Automated Attendants Destinations StarCenter Manager Multi-Queue Monitor StarCenter Recordings On-Hold Options Inbound CallerID Blocking Configuration Summary Print Company Directory

Individual Phone Settings

Individual Phone Settings

Ring Groups

Description

A Ring Group provides the ability to ring one or more phones simultaneously. Members are added to the

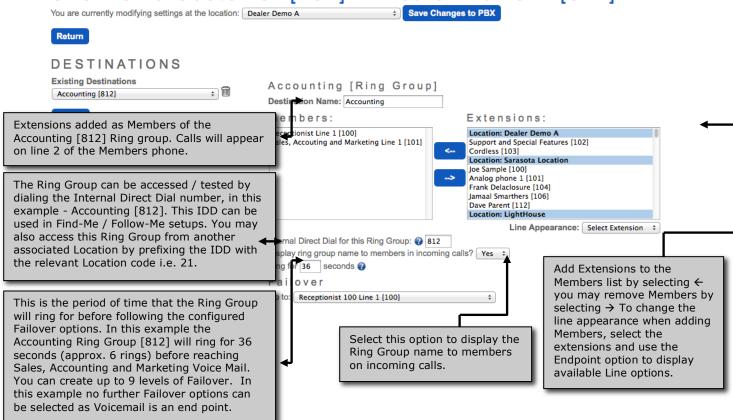
Ring Group via the <u>Destinations</u> link or when editing menus using the tool set icon. When adding Members to the Ring Group the system provides the ability to select which line appearance is used on the phone (If your phone has multiple lines). This is particularly useful if a distinctive ring is set on that phone for the line appearance used. Polycom phones provide this feature. If this is set the user will receive a different visual and audible notification based upon which line is calling. The Ring Group can be accessed via an Internal Direct Dial number. This is particularly convenient when testing the configuration. Remember to set the period of time your group will ring before reaching your Failover options. For example a ring time of 36 seconds is approximately 6 rings. The Failover option supports up to 9 levels of failover. Remember that selecting another extension or Voicemail is regarded as an end point and as such no further Failover options can be selected. You can display the Ring Group name to Members on incoming calls. This is particularly useful to distinguish between different types of calls using the phone display e.g. Operator Ring Grp being an "Operator" call.

Setup

- 1. Access the Star2Star portal and select the CONFIGURATION option at the top of the page.
- 2. Select the required customer location.
- 3. Select the Destinations link in the PHONE SYSTEM CONFIGURATION Menu.
- 4. Select the Add New Ring Group option listed under Ring Groups. 5. Enter a name for the new Ring Group and select Add. DESTINATIONS **Existing Destinations** Select Select Ring Groups Accounting [802] > Add New Ring Group >> Add New Queue Recordings >> Add New Recording Applications DESTINATIONS Dial External Number () [800] Dial External Number () [801] **Existing Destinations** >> Add New Dial External Number >> Add New Ring Group **Automated Attendants** Main Menu [805] Ring Group Name Use Location's Default Operator [804] Use Location's Default Operator (833) [803] >> Add New Automated Attendant Paging Groups >> Add New Paging Group **Voicemail Groups** Give the new Ring Group a meaningful name and select Add to create the group.

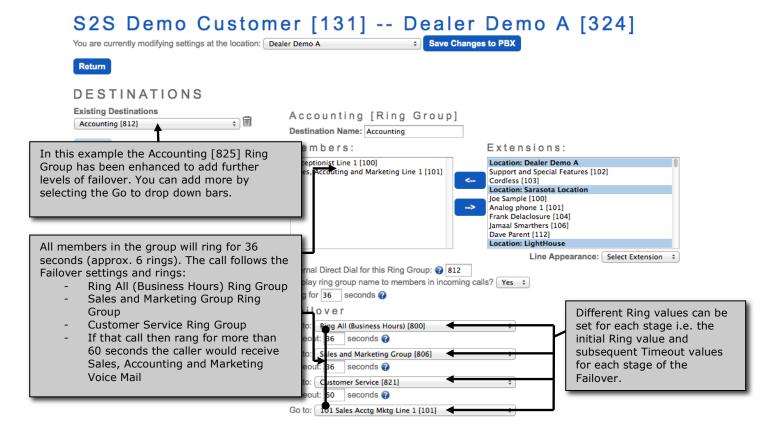
6. The screen will change displaying available extensions to add as Members to your Ring Group after selecting Add. If the customer has multiple locations the system will support adding those extensions as Members of the Ring Group. (**Note**) In the example displayed below 2 extensions have been added as Members of the Accounting Ring Group. Calls to the Accounting [825] Ring Group Members will arrive on the phones Line 2 appearance.

S2S Demo Customer [131] -- Dealer Demo A [324]



Ring Groups (contd)

7. Making further configuration enhancements see example below.



Ring Groups "StarTips"

- Add Members to the Ring Group by selecting
- Remove a Member from the Ring Group by selecting
- You may add members from any locations associated with the main.
- Use the End Point option to point to a specific line appearance on an extension.
- Set the Ring for value (Seconds) for the Ring Group. This is the period of time that the group will ring before reaching the Failover options that have been set. A value of 36 seconds is approximately 6 rings.
- You may set up to 9 levels of Failover options.
- Add Failover options by selecting the Go to Drop down bars.
- An extension or Voice Mail is considered an Endpoint and as such no further Failover option can be set if one of these is used.
- You may use the Internal Direct Dial number to access and Test the Ring Group.
- This IDD number can be used from a different location associated with the main.
- You can display the Ring Group name to Members on incoming calls. This is particularly useful to distinguish between different types of calls using the phone display e.g. Operator Ring Grp being an "Operator" call.
- Remember to Save Changes to PBX

 Save Changes to PBX

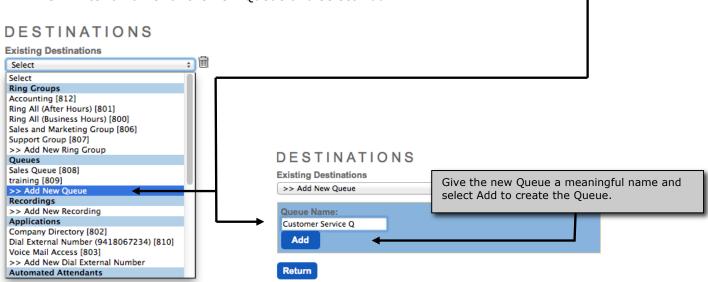
Queues

Description

A Queue provides the ability to ring Members in a sequential operation i.e. one at a time. The Queue methodology used is "Round Robin with memory". Callers placed in Queue will hear Message or Music on hold if enabled in the Advanced Settings features available under the On-Hold Options link. There are 9 levels of Failover available when using Queues. These are set in the same manner as Ring Groups. You can record a message that callers hear prior to entering the Queue. You can display the Queue name to Members on incoming calls. This is particularly useful to distinguish between different types of calls using the phone display e.g. Queue 1, Queue Priority etc.

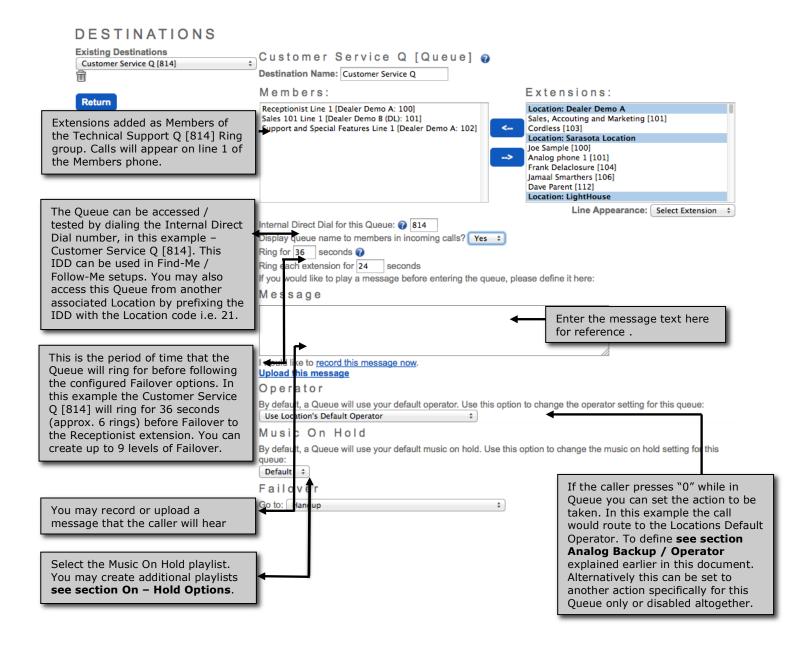
Setup

- 1. Access the Star2Star portal and select the CONFIGURATION option at the top of the page.
- 2. Select the required customer location.
- 3. Select the <u>Destinations</u> link in the **PHONE SYSTEM CONFIGURATION** Menu.
- 4. Select the Add New Queue option listed under Queues. ►5. Enter a name for the new Queue and select Add.



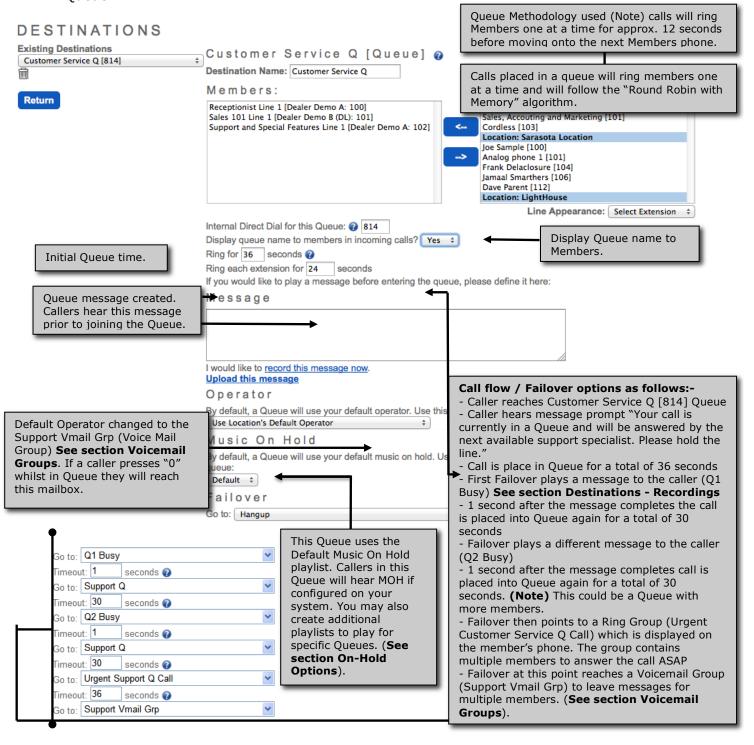
Queues (contd)

6. The screen will change displaying available extensions to add as Members to your Queue after selecting Add. If the customer has multiple locations the system will support adding those extensions as Members of the Queue. (Note) In the following example 3 extensions have been added as Members of the Customer Service Q Queue. Calls to the Customer Service Q [826] Queue Members will arrive on the phones Line 1 appearance.



Queues (contd)

7. This is a more complex example showing multiple Failover points to recordings (**see section Recordings**) and further Queuing. A caller in this Queue will be kept informed if delays are experienced. This Queue example has a message that is played to the caller prior to entering the Queue.



Queues "StarTips"

- ho Add Members to the Queue Group by selecting lacksquare
- Remove a Member from the Queue by selecting
- You may add members from any locations associated with the main.
- Use the End Point option to point to a specific line appearance on an extension.
- Set the Ring for value (Seconds) for the Queue. This is the period of time that the Queue will ring before reaching the Failover options that have been set. A value of 36 seconds is approximately 6 rings.
- You may set up to 9 levels of Failover options.
- You add Failover options by selecting the Drop down bars.
- An extension or Voice Mail is considered an Endpoint and as such no further Failover option can be set if one of these is used.
- Remember to set Timeout values for each stage of Failover. This is the period of time (seconds) before moving to the next stage of Failover.
- You may use the Internal Direct Dial number to access / test the Queue.
- This IDD number can be used from a different location associated with the main.
- Remember to add a Queue message that the caller will hear prior to entering the Queue. For help in creating recordings **See section Recordings Destinations**.
- Remember to add recordings to keep your callers informed if there is wait time.
- You can connect to other Queues. Consider a scenario where a caller is delayed in Queue. Play a recording to keep your caller updated. Place the caller back into another larger Queue i.e. more agents or a Ring Group to ring multiple Members.
- Use the Display Queue Name option to display the Queue Name on the Members phone.
- Remember to advise the caller in that message that there is a "0" out Operator option available if required / programmed.
- Remember to set the Operator to Default or another specific action. Disable if not used.
- When a call is placed in Queue each Queue member's phone will ring one at a time for the # of seconds set in the queue options page. The Queue methodology used is "Round Robin with memory".
- Callers placed in a Queue will hear Music / Message on hold if enabled. **See section On-Hold Options.** For more feature rich call center queue functionality, see StarCenter.
- Remember to Save Changes to PBX

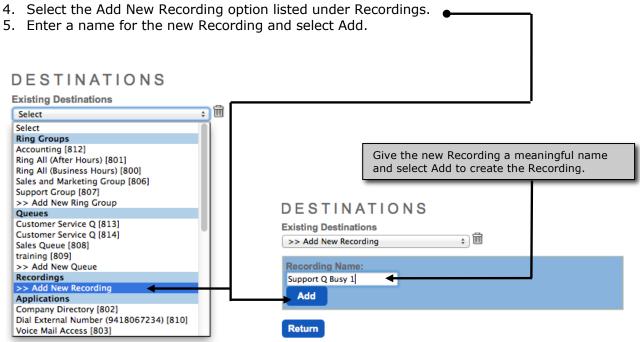
Recordings

Description

A Recording is used as a prompt or message that can be played to a caller when entering a Queue or during any stage of Failover in a Ring Group or Queue. A recording can also be used to provide customized access to a Conference Room. The recording can be created by following the prompts and either dialing internally using a StarPhone or by dialing the 800 number and using an access pin. Once recorded the user can play back, save or re-record. You may also Upload a pre-recorded message in WAV or MP3 format.

Setup

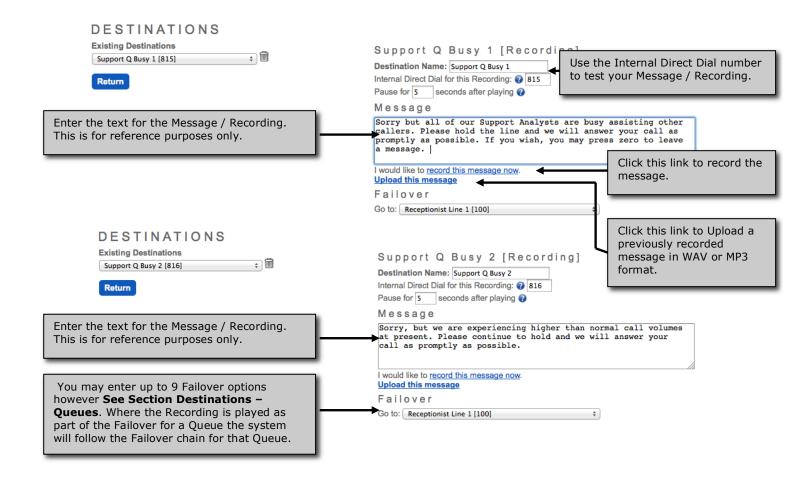
- 1. Access the Star2Star portal and select the CONFIGURATION option at the top of the page.
- 2. Select the required customer location.
- 3. Select the <u>Destinations</u> link in the **PHONE SYSTEM CONFIGURATION** Menu.



Recordings (contd)

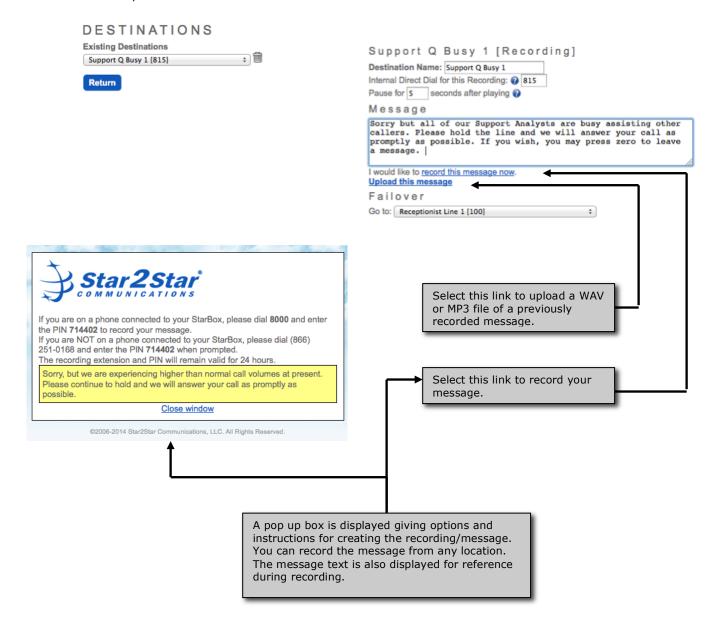
- 6. Enter the text for the Message / Recording into the box provided. This is for reference purposes only.
- 7. If you wish to create the recording select the <u>record this message now</u> link. Follow the on screen instructions to either use a StarPhone or dial an external 800 number and enter the access PIN provided.

If you wish to upload a previously recorded Message in WAV, MP3, AC3, MP2, FLAC, WMA, PCM AIFF or OGG format select Upload this message link. Follow the on screen instructions.



Recordings (contd)

- 8. Creating your own recording/message is simple. Select the <u>record this message now</u> link and follow the instructions set out in the pop up window. You can record the message at any location. See example below.
- 9. Alternatively you may upload a previously recorded message in WAV, MP3, AC3, MP2, FLAC, WMA, PCM AIFF or OGG format.



Recordings (contd)

New procedure for "recording on the fly"

You've always had the ability to add new recordings to your Star2Star system to customize your autoattendants, call queues, and your system recordings. In the past, you were prompted to enter a four-digit extension code to record your new message.

Now, your Star2Star system will assign a unique six-digit PIN for each recording via a help screen pop. You can then either dial extension 8000 on your StarSystem or an 866 Toll Free number via any phone and the system will prompt for the unique pin to create your recording.



Recordings "StarTips"

- Remember to set a Pause value i.e. the length of time the system waits before going to the first failover option.
- You may set up to 9 levels of Failover options.
- You add Failover options by selecting the Drop down bars.
- An extension or Voice Mail is considered an Endpoint and as such no further Failover option can be set if one of these is used.
- Remember to set Timeout values for each stage of Failover. This is the period of time (seconds) before moving to the next stage of Failover.
- You may use the Internal Direct Dial number to access / test the Recording.
- This IDD number can be used from a different location associated with the main.
- You can create a Recording by selecting the <u>record this message now</u> link. Follow the on screen instructions to either use a StarPhone or dial an external 800 number with an access PIN that is provided. Once recorded the message can be listened to, saved or re-recorded.
- Remember the recording extension and PIN provided in the pop up window when creating your own recordings/messages will remain valid for 24 hours.
- You can use a previously recorded Message in WAV or MP3 format select <u>Upload this message link</u>. Follow the on screen instructions.
- Remember that you can create a Recording and add to your system remotely i.e. anywhere that you have internet access to the Star2Star web portal.
- Create a recording to personalize a Conference Room e.g. "Welcome to our Company Conference call. Shortly you will be prompted for your pin number. Enter that pin number and join the Conference". Set the Conference Room as the Failover option for the recording one-second after the recording plays. Configure Incoming Call Routing to point a DID to the recording for your Conference calls.
- Remember to Save Changes to PBX

Save Changes to PBX

Applications

Description

You can enter external numbers for the system to dial "Dial External Number". These "Applications" can be used wherever a Failover selection exists. The Internal Direct Dial number allocated to each Dial External Number by the system may be used in extension Find-Me / Follow-Me configurations. The system automatically creates 2 system wide Applications:

Company Directory-This can be used as a menu option and will automatically play the directory application. This operates requiring the caller to enter the first 3 letters of the person's first or last name. The Directory Sort Order is set in **section Additional Location Information** along with the ability to enable Multi-Location Directory, a feature that allows callers to search the directory on all systems associated with your location. When a successful selection/match has been made the system will call the individuals phone.

Voice Mail Access-This can also be used as a menu option and will provide system users with a generic Voice Mail access to the system. The user is prompted for Mail Box and Password. This is particularly useful for providing system users with Voicemail access when remote from the Star2Star system. Local users can also dial the Internal Direct Dial number from any StarPhone to access their voicemail.

Setup

- 1. Access the Star2Star portal and select the CONFIGURATION option at the top of the page.
- 2. Select the required customer location.

Driver VMG [811]

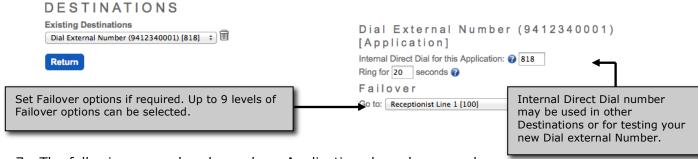
3. Select the Destinations link in the PHONE SYSTEM CONFIGURATION Menu.

4. Select the Add New Dial External Number option listed under Applications.

5. Enter a valid telephone number and select Add. DESTINATIONS **Existing Destinations** Select Sales Queue [808] training [809] >> Add New Queue Enter a valid telephone number and select Add. Recordings Support Q Busy 1 [815] Support Q Busy 2 [816] >> Add New Recording Applications DESTINATIONS Company Directory [802] Dial External Number (9418067234) [810] **Existing Destinations** Voice Mail Access [803] + >> Add New Dial External Number >> Add New Dial Extern **Automated Attendants** After Hours Menu [805] 9412340001 Main Menu [804] >> Add New Automated Attendant Add Paging Groups >> Add New Paging Group Voicemail Groups Return

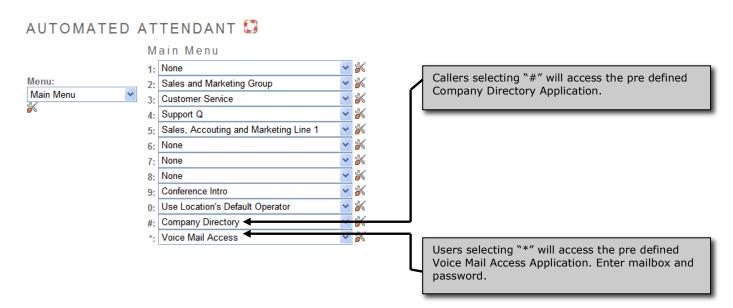
Applications (contd)

6. The example shows a new Dial External Number has been added to the system and is now available for use in applications.



7. The following examples show where Applications have been used.





Applications "StarTips"

- Remember to set a Ring for value i.e. the length of time the system waits before going to the first failover option if set.
- You may set up to 9 levels of Failover options.
- You add Failover options by selecting the Drop down bars.
- An extension or Voice Mail is considered an Endpoint and as such no further Failover option can be set if one of these is used.
- Remember to set Timeout values for each stage of Failover. This is the period of time (seconds) before moving to the next stage of Failover.
- You may use the Internal Direct Dial number to access / test the Application.
- This IDD number can be used from a different location associated with the main.
- Remember to Save Changes to PBX

 Save Changes to PBX

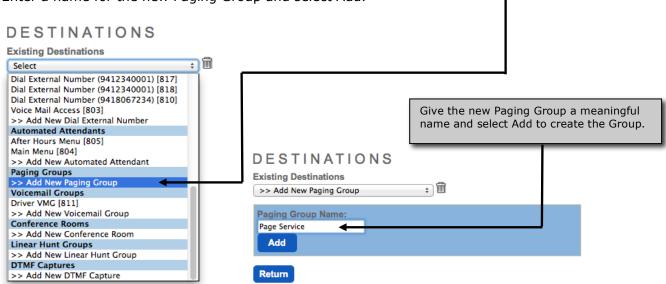
Paging Groups

Description

A Paging Group provides the ability to directly connect to group Members phones via the telephone's external speaker. The member receives an initial alert prior to connection. This feature is particularly useful for company wide alerts or to specific groups of people. (Note) Paging groups are limited to a maximum of 15 members due to system resource constraints. For larger applications Star2Star recommends the use of the snom-pai or external paging equipment that can be connected to the Star2Star system via a separate analog SPA 3102 adapter. Please contact the support organization for details.

Setup

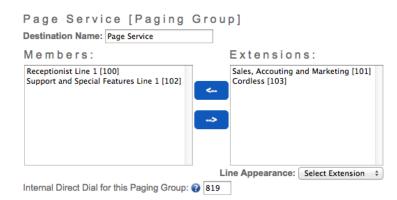
- 1. Access the Star2Star portal and select the CONFIGURATION option at the top of the page.
- 2. Select the required customer location.
- 3. Select the <u>Destinations</u> link in the **PHONE SYSTEM CONFIGURATION** Menu.
- 4. Select the Add New Paging Group under Paging Groups.●5. Enter a name for the new Paging Group and select Add.



- 6. Add members to the new Paging group by selecting —.
- 7. Access the group by dialing the Internal Direct Dial number.

Paging Groups (contd)

DESTINATIONS Existing Destinations Page Service [819] Return Add Extensions to the Members list by selecting ← you may remove Members by selecting →.



Paging Groups "StarTips"

- Add Members to the Paging Group by selecting
- Remove a Member from the Paging Group by selecting
- You may use the Internal Direct Dial number to access / test the Paging Group.
- Member's phones will alert / ring once only as the paging group is accessed.
- (Note). Paging groups are limited to a maximum of 15 members due to system resource constraints. For larger applications Star2Star recommends the use of the snom PA1 or external paging equipment that can be connected to the Star2Star system via a separate analog SPA 3102 adapter. Please contact the support organization for details.
- The snom PA1 is the device used for paging, and can be hooked up directly to speakers, or to a power amplifier on an existing overhead speaker setup.
- Remember to Save Changes to PBX

Save Changes to PBX

Voicemail Groups

Description

A Voicemail Group provides the functionality to deliver messages to multiple extensions simultaneously. The Voicemail indicator will illuminate on all extensions in the group when a caller leaves a message. The message is delivered to each voicemail group member and the voicemail group mailbox. Listening to the voice mail on an extension will only extinguish that extensions voice mail indicator. Additional notification can be sent by using the Email/Pager/Cell Phone option. Alternatively you may create a Voicemail Group that has no members. The message is delivered to the voicemail group mailbox. To retrieve voice mails using this method dial the Internal Direct Dial number for Voicemail Access which is a pre-defined application in the Destination list. Indication that a voicemail has been received can be achieved by setting the Email/Pager/Cell Phone option. (Note) sending notification to a cell phone using text messaging (SMS), differs for each mobile provider. A common list of providers is provided below. However Star2Star cannot guarantee the functionality with a third party device. Contact your mobile provider to ensure that your device supports this feature and to verify the correct format. Please be aware that this is not a free service unless it is included in your mobile plan.

Mobile Service Providers-Email to SMS Gateways (Cell Phone Text Message Alerts)

AT&T Use the format number@txt.att.net

T-Mobile USA Use the format number@tmomail.net

Verizon Wireless Use the format number@vtext.com

Sprint PCS Use the format number@messaging.sprintpcs.com

Sprint PCS - Short Mail Use the format number@sprintpcs.com

Alltel Use the format number@message.alltel.com

Bellsouth Use the format number@bellsouth.cl

US Cellular Use the format number@email.uscc.net

Virgin Mobile USA Use the format number@vmobl.com

Metro PCS Use the format number@mymetropcs.com

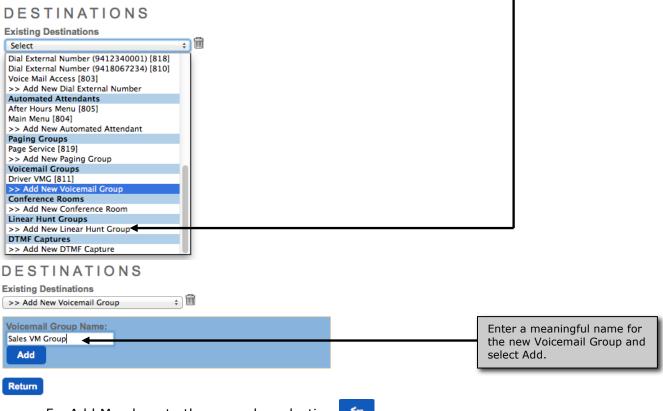
Metro PCS Use the format number@mymetropcs.sms.us

(Note) ensure you enter the correct format for your cell phone. Entering the number alone will not work.

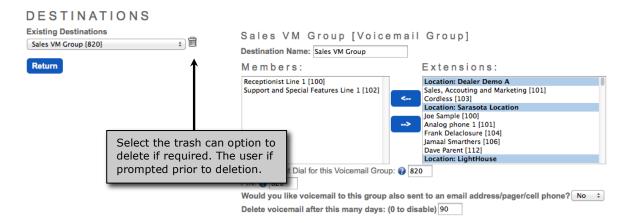
Voicemail Groups (contd)

Setup

- 1. Access the Star2Star portal and select the CONFIGURATION option at the top of the page.
- 2. Select the required customer location.
- 3. Select the Destinations link in the PHONE SYSTEM CONFIGURATION Menu.
- 4. In Destinations locate the Voicemail Groups option. Select Add New Voicemail Group.

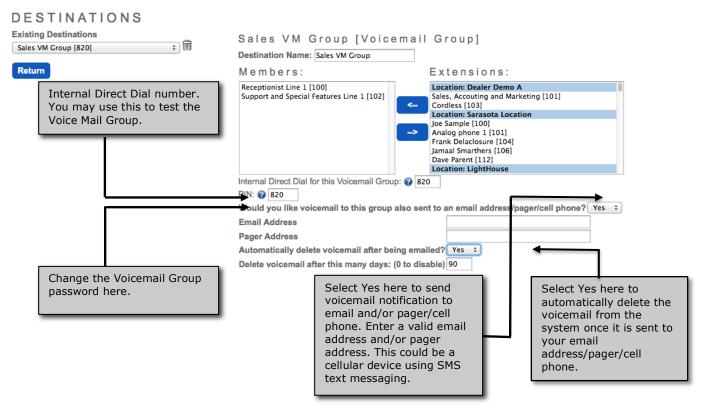


- 5. Add Members to the group by selecting <--
- 6. Remove Members by selecting



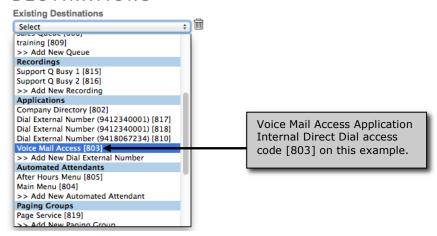
Voicemail Groups (contd)

7. Setting the Voicemail to Email address/pager/cell phone option.



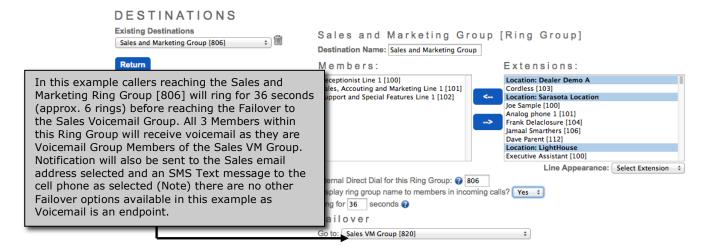
8. To setup up the Voicemail Group mailbox use the Voice Mail Access Internal Direct Dial access code. This is located in <u>Destinations</u>. It is a pre defined system application and will allow access to any system mailbox. The user is prompted for a Mailbox Number and a password. The default password is the same as the mailbox number unless you have changed it previously in the Voicemail Group screen. Create an unavailable message e.g. "You have reached the Sales mail box". When leaving a message in this mailbox the unavailable message will play after which the system will prompt "Please leave your message after the tone". (Note) For additional help in setting up Voice Mail please refer to the Voicemail Guide.

DESTINATIONS



Voicemail Groups (contd)

- 9. The Voicemail Group Mailbox is ready to use.
- 10. The following example uses the mailbox in the customer's environment. Adding the Voicemail Group mailbox as a Failover from a Ring Group.



Voicemail Groups "StarTips"

- Listening to Voicemail Group mail on a Member extension will only extinguish that extensions voice mail indicator.
- You may set up a Voicemail Group without any members. Use the Email/Pager/Cell option for notification.
- Messages left in a Voicemail Group mailbox without Members will be delivered to the Group mailbox number and as such should be accessed via the Voicemail Access Application. This application can be found in Destinations.
- Messages left in a Voicemail Group mailbox with Members will be delivered to the individual Member mailboxes.
- The default password for a new Voicemail Group mailbox is the same as the Internal Direct Dial number unless this has been changed in the Voicemail Group setup screen.
- Remember to record an unavailable message.
- Create a "Company" voicemail group to send mail to all users.
- For additional help in setting up Voicemail please refer to the Star2Star Phone & Features Guide.
- Remember to Save Changes to PBX

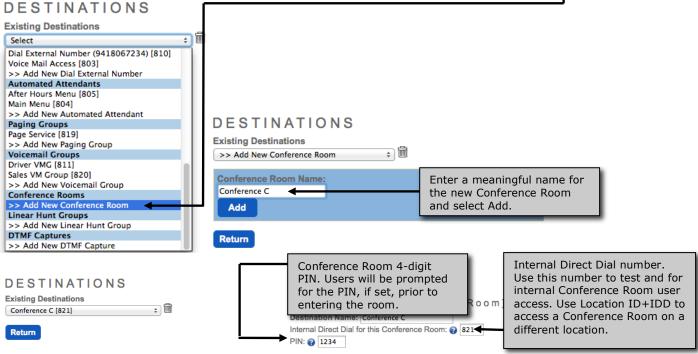
Conference Rooms

Description

The system provides the facility to create Conference Rooms that multiple callers can access. Callers are connected together as per a normal conference facility. If configured, users will be prompted for a PIN. To access the room via an external line point a DID to the Conference Room via the Incoming Call Routing option. You may also connect an incoming caller to the Conference Room using the Internal Direct Dial number.

Setup

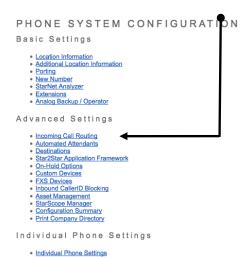
- 1. Access the Star2Star portal and select the CONFIGURATION option at the top of the page.
- 2. Select the required customer location.
- 3. Select the Destinations link in the PHONE SYSTEM CONFIGURATION Menu.
- 4. Select the Add New Conference Room option listed under Conference Rooms •
- 5. Enter a meaningful name for the new Conference Room and select Add.
- 6. You may set a 4-digit PIN for user access.



- 7. Final steps would be to set the Incoming Call Routing to point a line(s) to the new Conference Room.
- 8. Alternatively callers may be transferred to the conference room using the Internal Direct Dial Number.

Conference Rooms (contd)

9. Select the Incoming Call Routing link in the PHONE SYSTEM CONFIGURATION Menu.



10. Select the line required for external access to the Conference Room. See example below.

INCOMING CALL ROUTING 😂 Defined Routes: Conference C [Conference Room] Number: (941) 926-1193* Destination Name: Conference C Schedule: Default Business Hours + Internal Direct Dial for this Conference Room: 2 821 ÷ | % PIN: (2) 1234 Open: Conference C [821] ÷ 🔏 Closed: Conference C [821] Use the drop down bars to select the relevant Conference Room in the Destination list for the required line.

Conference Rooms "StarTips"

- Remember to create a 4-digit PIN if required. Users will be prompted for the PIN prior to entering the Conference Room.
- Internal users can access the Conference Room by using the Internal Direct Dial Number.
- You can transfer an incoming caller into the conference room using the Internal Direct Dial Number.
- Point a DID at the Conference Room for external user access.
- To personalize your Conference Room create a recording "Welcome to our Company Conference call. Shortly you will be prompted for your pin number. Enter that pin number and join the Conference". Set the Conference Room as the Failover option for the recording. Set Incoming Call Routing to point your Conference Room DID to the new recording.
- Remember to Save Changes to PBX

 Save Changes to PBX

On-Hold Options

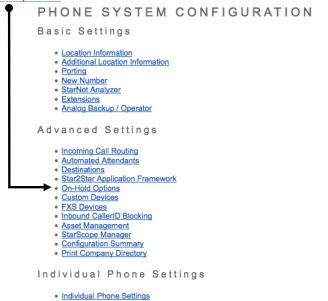
Description

The system provides the facility to Upload On-Hold files without the requirement for any adjunct equipment. Supported file types are WAV, MP3, AC3, MP2, FLAC, WMA, PCM AIFF or OGG. To select sound sources for callers to hear when Queuing, parked or placed on hold, check/enable the relevant file. You may delete files from the system here. The system is pre configured with 3 music on hold source files which reside in the Default Playlist. You may create multiple playlists and upload sound source files to those Playlists. You may select a different Playlist when playing Music / Message On-Hold (MOH) in a Queue. (Note) MOH playlists do not apply to remote extensions.

Use the Call Park Timeout field to enter the amount of time (seconds) that a caller will remain in park. The parked caller will hear music / message on hold if configured. When the timeout is reached the caller will return to the extension that initiated the park. (**Note**) For more detailed information on the Call Park feature please refer to the Star2Star Phone and Features Guide.

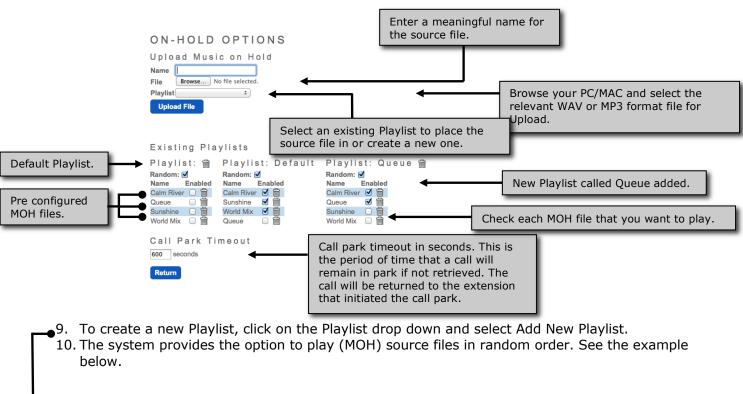
Setup

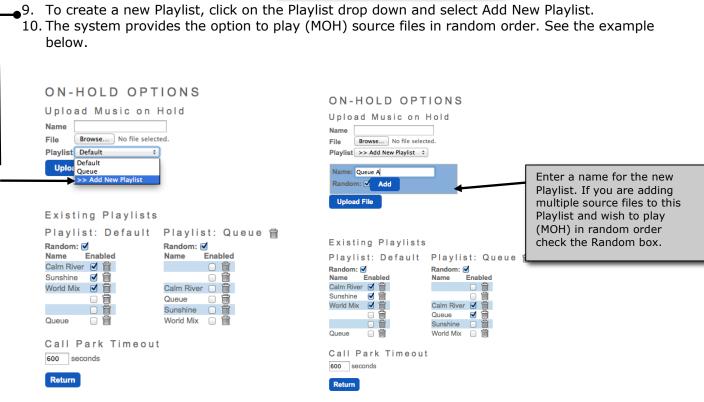
- 1. Access the Star2Star portal and select the CONFIGURATION option at the top of the page.
- 2. Select the required customer location.
- 3. Select the On-Hold Options link in the PHONE SYSTEM CONFIGURATION Menu.



- 4. To upload a new file, enter a meaningful name that will be displayed in the On-Hold Options screen.
- 5. Browse for the file on your PC.
- 6. Select the Playlist required. The system can have multiple playlists that will allow for different Message / Music On Hold (MOH) for Queues.
- 7. Source files that are uploaded will be available for every playlist on the system.
- 8. Check the files that you want callers to hear.

On-Hold Options (contd)





On-Hold Options "StarTips"

- Upload custom On-Hold sources in WAV, MP3, AC3, MP2, FLAC, WMA, PCM AIFF or OGG format.
- Remember to Check / Enable source to play.
- You can create multiple Playlists to upload Music / Message (MOH) source files to.
- The system provides the option to play your (MOH) source files in random order.
- You can disable an On-Hold source by unchecking the box in On-Hold Options.
- Set the Call Park Timeout to an acceptable value for the customer. This is the period of time, in seconds, that a call will remain in park if not retrieved. The call will be returned to the extension that initiated the call park.
- Remember to Save Changes to PBX

Configuration Summary

Description

This Portal option provides a complete configuration listing for your location. The list includes full configuration details for the following:

Call Path

Contains Incoming Call Routing information for Open and Closed hours for all DID's / Numbers.

Destinations

Contains all Destination entries and associated Failover options including timing. The entries are listed in Internal Direct Dialing Order.

Automated Attendant

Contains details for all Menu's configured on your system.

Group Membership

Shows member / extension details for all Ring Groups and Queues.

Extensions

Contains details for extensions; includes phone types, extension name / number, Find-Me / Follow-Me settings including timings.

Misc.

Contains settings for Operator and Analog Backup if configured on your system.

Setup

- 1. Access the Star2Star portal and select the CONFIGURATION option at the top of the page.
- 2. Select the required customer location.
- Select the <u>Configuration Summary</u> link in the PHONE SYSTEM CONFIGURATION Menu.



- Individual Phone Settings
- 4. A complete listing of all configuration settings for your location is displayed. This can be printed via your browser.

Print Company Directory

Description

This Portal option provides a complete company directory listing for your location and any other locations associated with your system. The list includes extension name / number, user email address, user supplementary number. Details are provided for all locations. Extension on other locations associated with your system will include the 2-digit location code required to access that system. (**Note**) Please refer to the Star2Star Phone and Features Guide for further details on Inter-Location Dialing.

Setup

- 1. Access the Star2Star portal and select the CONFIGURATION option at the top of the page.
- 2. Select the required customer location.
- 3. Select the Print Company Directory link in the PHONE SYSTEM CONFIGURATION Menu.

PHONE SYSTEM CONFIGURATION

Basic Settings

Location Information
Additional Location Information
Extensions
Analog Backup / Operator

Advanced Settings

Incoming Call Routing / Scheduler
Automated Attendants
Destinations
StarCenter Manager
Multi-Queue Monitor
StarCenter Recordings
On-Hold Options
Inbound CallerID Blocking
Configuration Summary
Print Company Directory

Individual Phone Settings

www.star2star.com

Individual Phone Settings

Fourth Edition

Individual Phone Settings

Description

This Portal option provides users with the ability to configure their individual phone. (e.g. Find-Me/Follow-Me, Voicemail, Voicemail to Email, Personal Operator.

Setup

- 1. Access the Star2Star portal and select the CONFIGURATION option at the top of the page.
- 2. Select the required customer location.
- 3. Select the Individual Phone Settings link in the PHONE SYSTEM CONFIGURATION Menu.



- 4. The following list of options is displayed. This information is displayed to basic phone users unless Portal Access has been removed.
- 5. Listed below are examples showing slight variations in the Individual Phone Settings. These are:
 - a. Phone Options
 - b. User Info Only displayed when there is a user associated with the phone.
 - c. Extension Options Only displayed when there is no voicemail box.

INDIVIDUAL PHONE SETTINGS

INDIVIDUAL PHONE SETTINGS

- Phone Options
- User Info
 Find-Me / Follow-Me
- Voicemail
- CallerID Blocking
 Print Company Directory
- StarDialer (32-bit)
- StarDialer (64-bit)
- User Guides
- Return to Phone System Configuration

- Speed Dials
- Phone Options
- Find-Me / Follow-Me
- Voicemail
 CallerID Blocking
- Print Company Directory
- StarDialer (32-bit)
- StarDialer (64-bit)
- User Guides
- · Return to Phone System Configuration

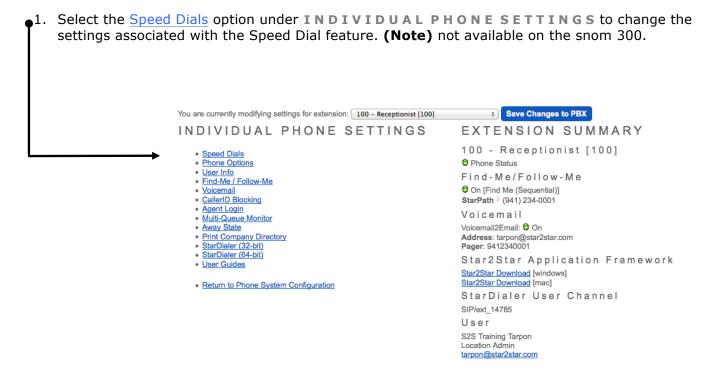
Speed Dials

Description

This option allows the user to add / manage individual speed dial numbers for their phone. Speed Dials are numbers that you dial on a frequent basis or local extension presence. Items that appear in the Destinations list can be selected. This option also provides the ability to add custom numbers and labels for easy identification i.e. those not in Applications as Dial External Numbers. The number of available speed dials will change if a user has an expansion key module associated with the phone. The number of modules can be set in Basic Settings under the PHONE SYSTEM CONFIGURATION Menu. Select the Extensions link. The following phones sold by Star2Star Communications can be fitted with an Expansion Module, sometimes also referred to as an Attendant Console:

Polycom IP 601, IP 650, IP 670 Snom 320, 360, 370 Cisco SPA 5xx Series Yealink T26P, T28P, T38G

Setup

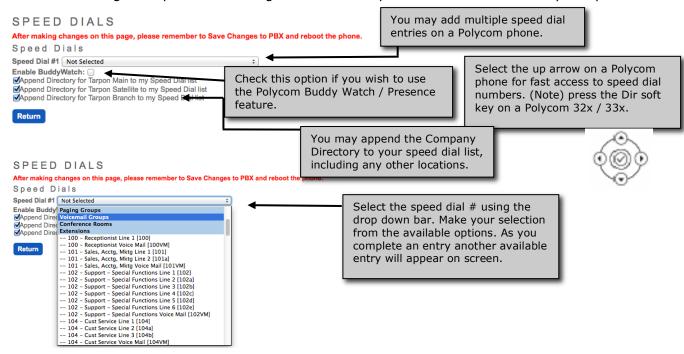


(Note) See the Star2Star Phone and Features Guide for more information.

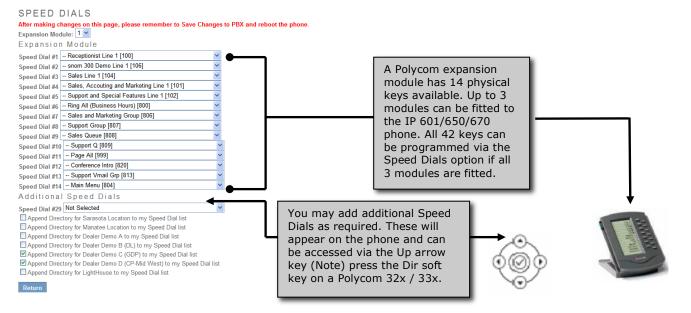
2. Remember to Save Changes to PBX and reboot the phone. To reboot the phone unplug the Internet cable momentarily. The phone will restart. (**Note**) if the phone has a separate power cable connected, momentarily unplug that cable to reboot the phone.

Polycom Phone

3. The following examples show configuration of the Speed Dial feature for a Polycom phone.

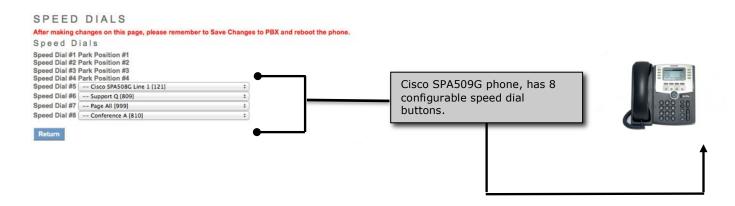


4. The following example shows configuration of the Speed Dial feature for a Polycom IP 601 or IP 650 with 1 Expansion module added. You can also add additional Speed Dials to the phone.

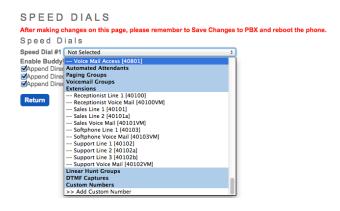


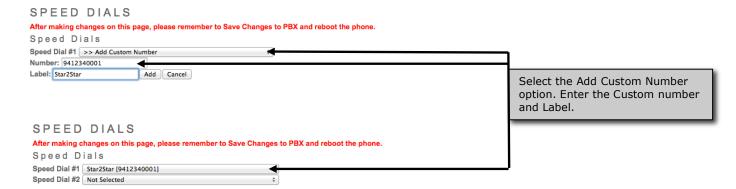
Cisco Phone

1. The following example shows configuration of the Speed Dial feature for a Cisco SPA509G phone in receptionist mode. (**Note**) the phone has 8 physical programmable keys.



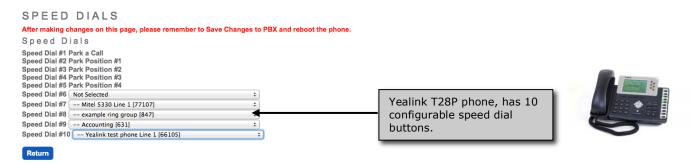
2. To add a Custom Number and label for easy identification, select the Add Custom Number toward the bottom of the selection list. Add the number and associated label. (**Note**) this feature is available for the following phones; Polycom, Cisco, Yealink and snom (not 300).



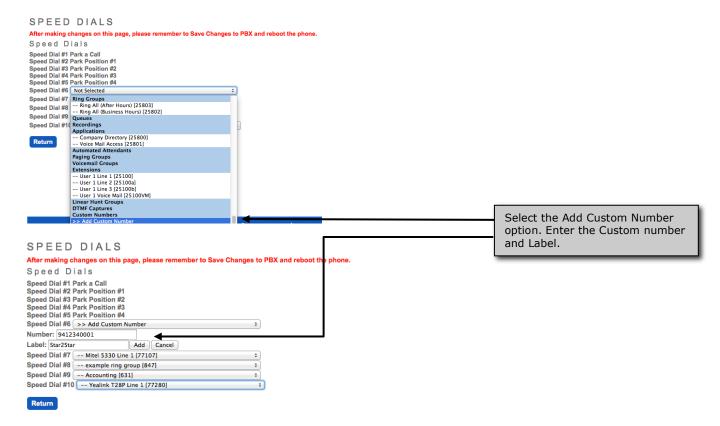


Yealink Phone

1. The following example shows configuration of the Speed Dial feature for a Yealink T28P phone.



2. To add a Custom Number and label for easy identification, select the Add Custom Number toward the bottom of the selection list. Add the number and associated label.

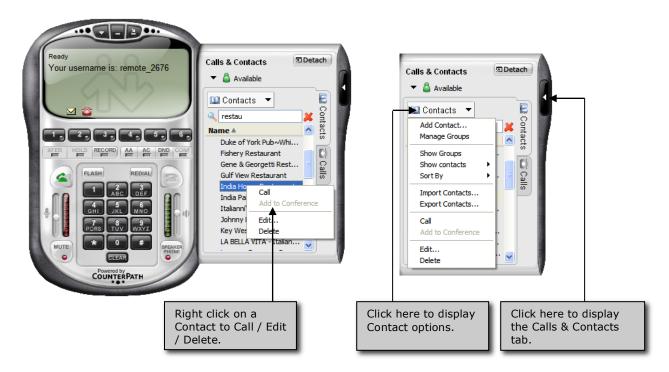


SPEED DIALS

After making changes on this page, please remember to Save Changes to PBX and reboot the phone. Speed Dials Speed Dial #1 Park a Call Speed Dial #2 Park Position #1 Speed Dial #3 Park Position #2 Speed Dial #4 Park Position #3 Speed Dial #5 Park Position #4 Select the Add Custom Number Speed Dial #6 Star2Star [9412340001] option. Enter the Custom number Speed Dial #7 -- Mitel 5330 Line 1 [77107] + and Label. Speed Dial #8 -- example ring group [847]
Speed Dial #9 -- Accounting [631]
Speed Dial #10 -- Yealink T28P Line 1 [77280] + +

Softphone

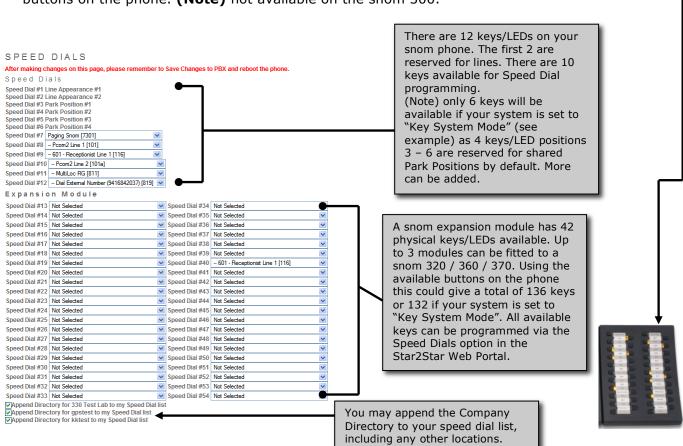
- 1. Speed Dials / Contacts for Softphone (S2S 3000) are set on the phone and not via the Star2Star web portal.
- 2. You may import Contacts to dial from another source i.e. Outlook® or a supported file format.
- 3. Alternatively you may either Add Contacts manually or click in the softphone display during an active call to add that caller to your Contacts.
- 4. Click right mouse on a Contact to dial.
- 5. If your Contacts list contains many entries use the A field to filter the list.



snom Phone

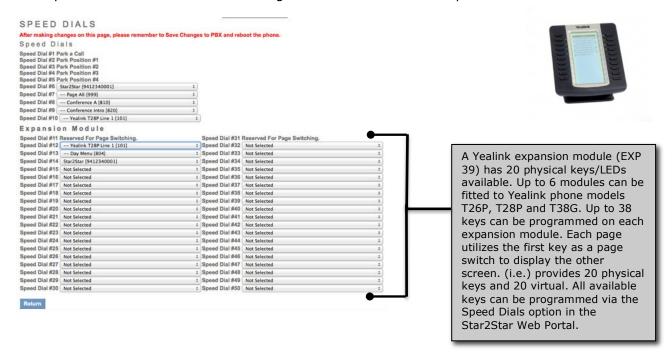
- 1. To access the individual Speed Dial directory entries added via the Star2Star web portal press
 - the Directory button or (snom 320 / 360 / 370) select the **Directory/Dir** soft key. **(Note)** The snom 300 will only provide access to the Company Directory.
- 2. Your Speed Dial entries will be displayed.
- 3. Use the or navigation keys to browse the entries.
- 4. Press over a highlighted or displayed entry to dial.

5. The following example shows the Speed Dial Options for a snom (320 / 360 / 370) phone with Expansion module added. You can also add Speed Dials to utilize the remaining available buttons on the phone. (**Note**) not available on the snom 300.



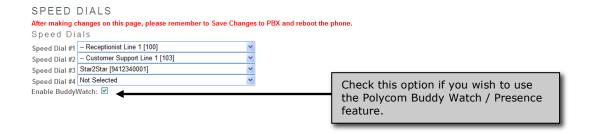
Yealink Phone

1. The following example shows the Speed Dial Options for a **Yealink T28P** phone with 1 Expansion module **(EXP 39)** added. **(Note)** May also be configured on T26P and T38G phones. You may also add Speed Dials to utilize the remaining available buttons on the phone.



Polycom Buddy Watch / "Presence" Feature

Users can also set the Polycom Buddy Watch feature in the Speed Dial screen. Buddy Watch provides users with the ability to monitor the "Presence" of a number of local users. By setting this feature for a particular extension you can quickly determine the status i.e. busy or available. This feature works in conjunction with the Speed Dial / Directory list set on your phone i.e. use the **Polycom Phone** Speed Dial configuration examples to create your list.



Phone Options

This option is only available for certain phone models.

1. Select Phone Options under INDIVIDUAL PHONE SETTINGS to change the settings.

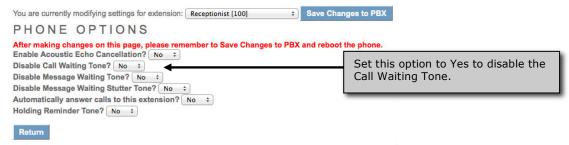


On the Grandstream GXP 2000 it provides the ability to Hide / Display missed calls and Enable / Disable the backlight.



On a Polycom phone, it provides the ability to turn off the Call Waiting Tone which is played to indicate the arrival of a new call when busy. It also provides access to the Acoustic Echo Cancellation setting and other options as shown below.

(**Note**) Acoustic Echo Cancellation should only be used under instruction from the Star2Star Technical Support organization).



On a Cisco phone, it provides the ability to automatically answer calls to an extension and other options as shown below.

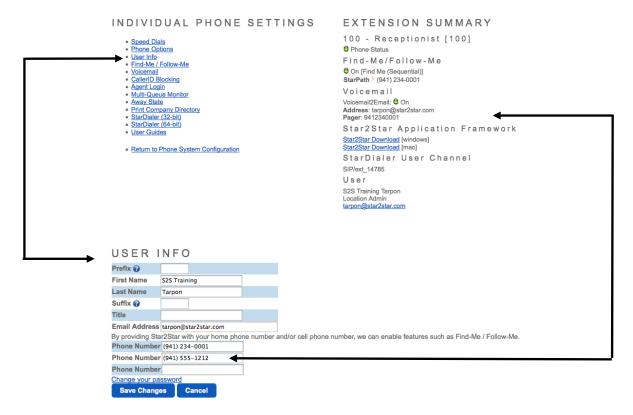


User Info

This option provides the ability to add / update a user's information i.e. Name, Title, Email address, phone number (used in features such as **Find-Me / Follow-Me**) and password. The configuration screen also provides a link for changing the user's password.

(Note) the User Info option will only be available for extensions that have been set up with a user associated.

1. Select the <u>User info</u> option under **INDIVIDUAL PHONE SETTINGS**. This will allow you to provide or update user information. Phone numbers added in this screen will be available for use with the Find-Me / Follow-Me feature. (**Note**) this option is only available if the extension has a user associated with it



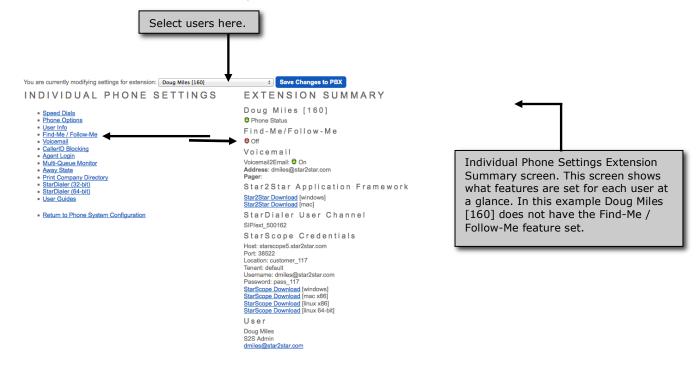
Find-Me / Follow-Me

Users are provided with powerful functionality to redirect calls via an intuitive web screen. Callers will follow the Find-Me / Follow-Me configuration, if set, to locate their party. There are 3 configuration styles available for this functionality:

Find Me (Immediate) Calls are immediately diverted / forwarded to an alternative number. **Find Me (Sequential)** Calls can follow a sequence of ringing up to 4 alternative numbers in turn with varying timeout values.

Find Me (All) Calls can ring up to 4 alternative numbers simultaneously.

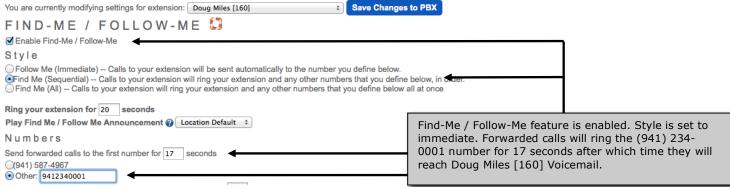
1. Select the Find-Me / Follow-Me option under INDIVIDUAL PHONE SETTINGS



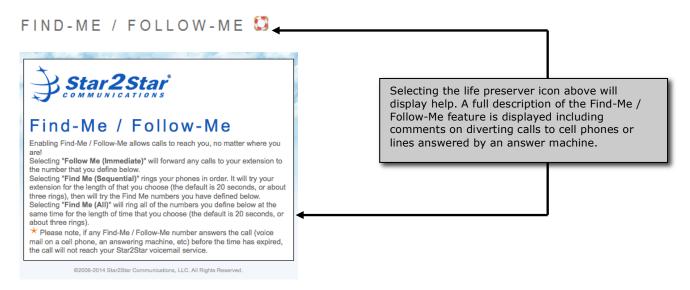
2. Set the Find-Me / Follow-Me feature by checking the option



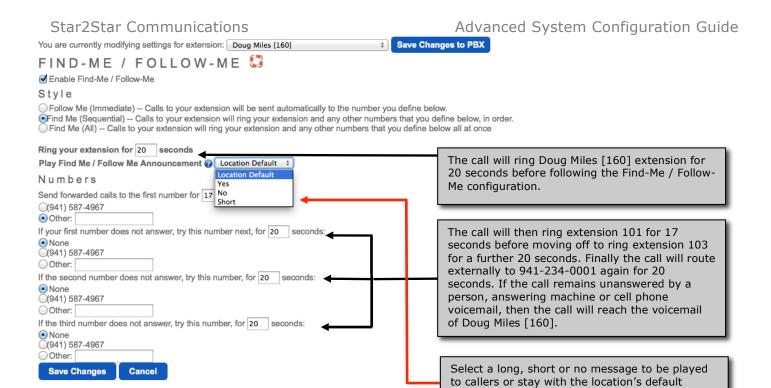
3. In the example below the Find-Me / Follow-Me feature is now set for immediate. Calls to Doug Miles [160] will forward immediately to the alternative number set for a period of 20 seconds. Callers, if unanswered in that timeframe, will receive Doug Miles [160] Voicemail.



4. Select the life raft icon to view help on the Find-Me / Follow-Me feature.

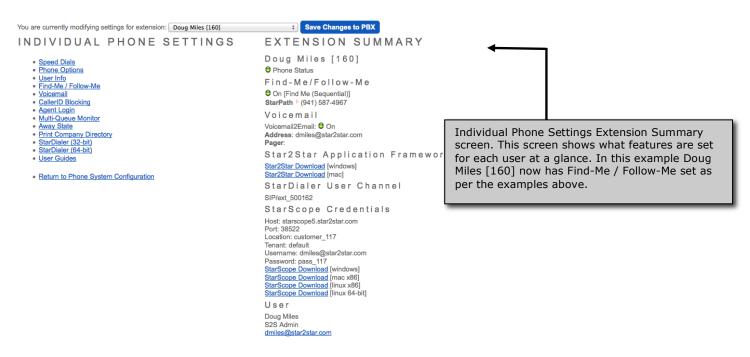


5. Setting the Find-Me / Follow-Me feature to Sequential. See the example below.



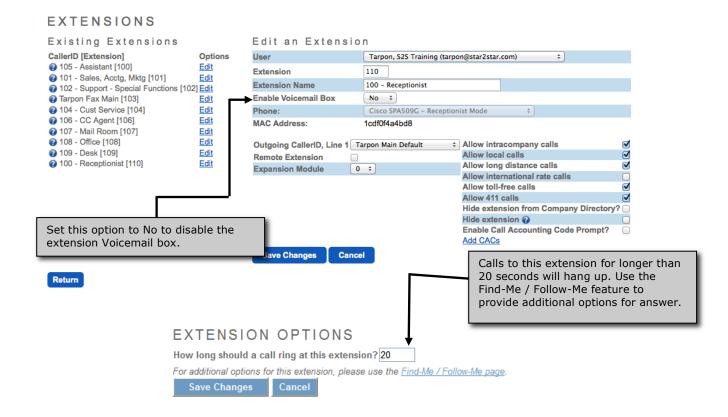
6. View the Individual Phone Settings Extension Summary screen below to see what features are set for your extension.

setting on a per-user basis.



Extension Options

This option is only visible if the extension has been set up without a Voicemail Box i.e. a reception / lobby area phone with no user. See example below. Use this option to determine how long the extension should ring for. Remember that calls to an extension without a mailbox will hang up once the ring time is reached. Use the Find-Me / Follow-Me feature to provide alternate options for answer.

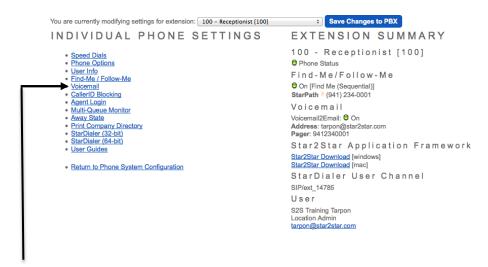


Voicemail

- 1. Select the <u>Voicemail</u> option under **INDIVIDUAL PHONE SETTINGS** to send voicemail to email/pager/cell phone. The following options may also be configured in the same setup screen:-
 - System speaks the caller ID prior to speaking the message.
 - System speaks the Date/Time prior to speaking the message.
 - Determine how long the users phone should ring before the call is sent to voicemail.
 - Determine how many days voicemail will remain before automatic deletion.
 - Set a Personal Operator for your extension
 - Change the user's voicemail pin.

The personal operator feature allows users to customize the call flow when a caller presses "O". Click the Voicemail option in Individual Phone Settings. Click the **Personal Operator** link. (By default, this option is set to "Use Location's Default Operator") Select the defined destination you want. (Note) the destination must be an existing destination such as another extension, voicemail box, ring group, etc. Once you have made your selection, click Save Changes followed by Save Changes to PBX. Refer to Star2Star Voicemail Guide for more details.

Now when a caller presses "O" when listening to your outgoing voice mailbox message, they will be connected to your Personal Operator selection and not the System Default Operator. To disable, select "None" when clicking the **Personal Operator** dropdown.



Star2Star Communications You are currently modifying settings for extension: 100 - Receptionist [100] VOICEMAIL Would you like your voicemail sent to your email/pager/cell phone? Yes Email Address: Pager/Cell Phone Address: Automatically delete voicemail after being emailed? No : Would you like the system to say the Caller ID information before the message? No : How long should the call ring before it is sent to voicemail? (in seconds) 20 Delete voicemail after this many days: (0 to disable) 0

Voicemail (contd)

Change your voicemail PIN

Save Changes Cancel

Personal Operator: **O Use Location's Default Operator

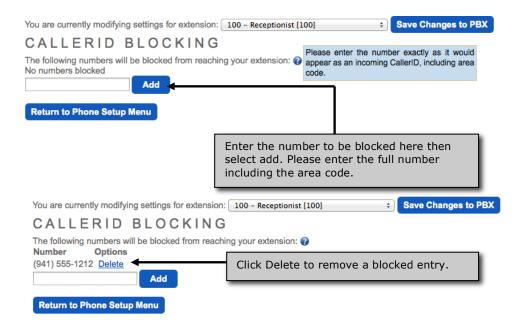
 To configure INDIVIDUAL PHONE SETTINGS to enable the voicemail to email feature, set the option to yes and enter a valid email address as per the example below. You may also set the option to send a message to a Pager/cell Phone Address by entering a valid number in the correct format (See section Voicemail Groups) for help in setting this feature).

You are currently modifying settings for extension: 100	- Receptionist (100) + Save Changes to PBX			
VOICEMAIL				
Would you like your voicemail sent to your email/pager/cell phone? Yes ÷		Set the option to	Set the option to yes and enter a valid	
Email Address:	user@star2star.com	email address.	,	
Pager/Cell Phone Address:	9415551212@tmomail.net			
Automatically delete voicemail after being emailed?	No ‡			
Would you like the system to say the Caller ID information before the message? No ÷				
Would you like the system to say the Date/Time of the message? No :				
How long should the call ring before it is sent to voicemail? (in seconds) 20				
Delete voicemail after this many days: (0 to disable)	0			
Personal Operator: Use Location's Default Operator	‡			
Change your voicemail PIN Save Changes Cancel				
You are currently modifying settings for extension: 100 - Receptionist [100] \$\delta\$ Save Changes to PBX				
VOICEMAIL				
Would you like your voicemail sent to your email/pager/cell phone? Yes ‡				
Email Address:	user@star2star.com	You may also set the sy	stem to send	
Pager/Cell Phone Address:	9415551212@tmomail.net	an alert message to a F	age/Cell Phone	
Automatically delete voicemail after being emailed?	No ‡	Address. (See section	Voicemail	
Would you like the system to say the Caller ID information before the message? No ÷		Groups) for help in set	ting this	
Would you like the system to say the Date/Time of the message? No ‡		feature.		
How long should the call ring before it is sent to voicemail? (in seconds) 20				
Delete voicemail after this many days: (0 to disable) 0				
Personal Operator:				
Change your voicemail PIN				
Save Changes Cancel				



CallerID Blocking

1. Select the <u>CallerID Blocking</u> option under INDIVIDUAL PHONE SETTINGS to block unwanted calls. Blocked callers will hear a fast busy tone for approximately 5 seconds then the call will disconnect. Numbers can be added or removed very easily through this option. (**Note**) this feature is not available when the extension is called via a Ring Group.



Location based caller ID block:

With this feature enabled, callers from blocked numbers will hear a fast busy tone for a few seconds, then the call cuts off. System Administrators can enter blocked number data using comma or line delimited format. You can also use the output from a database / CRM system and paste into the Location CallerID Blocking screen using the required format.

- 1. Select the InboundCallerID Blocking option under ADVANCED SETTINGS
- 2. Remember to Save Changes to PBX when done.

Return

LOCATION CALLERID BLOCKING Calleribs (comma separated or one per line): [\$\frac{1.213-155}{941-321-1654}\$] Enter the number(s) to be blocked here then select Add Blocks. Please enter the full number including the area code by comma separated or one per line. Return LOCATION CALLERID BLOCKING Calleribs (comma separated or one per line): Save Changes to PBX when complete. Add Blocks [\$\frac{1.213-155}{941-321-7654}\$] Save Changes to PBX when complete.

User Guides

INDIVIDUAL PHONE SETTINGS

- Speed Dials
- Phone Options
- User Info
- Find-Me / Follow-Me
- Voicemail
- CallerID Blocking
- Agent Login
- Multi-Queue Monitor
- Away State
- Print Company Directory
- StarDialer (32-bit)
- StarDialer (64-bit)
 User Guides
- Return to Phone System Configuration

Click to provide access to user documentation including FAQ's, Phone and Features Guide, Phone Quick Guides, Voicemail Guide and Phone Features Video.

\P Individual Phone Settings "StarTips"

- Remember to Save Changes to PBX and restart the phone when making changes.
- When using the Find-Me / Follow-Me feature to forward calls to a cell phone setting the ring time will affect whether voicemail will be left on the StarPhone or Cell Phone. Generally if the ring time is greater than 20 seconds the cell phone voicemail will answer the call.
- When setting the Voicemail feature to send a notification to a Pager/Cell Phone Address (**See section Voicemail Groups**) for help on configuration. Please note that sending a message to a cell phone via SMS is not a free service and will incur charges from your mobile provider unless it is included in your plan.
- If you configure an extension without a Voicemail box remember to set the ring time (seconds) in Extension Options. For additional answer options for callers ringing an extension without a Voicemail box use the Find-Me / Follow-Me feature.
- Use CallerID to block unwanted callers. You may remove a block simply by deleting the entry.
- Remember to set a Personal Operator if required.
- Remember to Save Changes to PBX

Multicast Paging for Polycom VVX/Yealink T4x Phone Ranges

Multicast Functionality:

Multicast paging on the Polycom VVX/Yealink T4x phone range is **not constrained** by a limited number of members as is the case with standard "Paging Groups" available on the StarBox 2200, StarBox 2250, StarBox 5500.

Multicast paging on the Polycom VVX/Yealink T4x phone range can be used on StarBox's including the StarLite. The functionality is provided at the "switch" level over "Channels" and has been tested successfully with all Star2Star supplied switches.

Channels:

- Each Multicast Paging Group will be automatically assigned to a Channel.
- There are a total of 25 channels but only 22 can be customized (2 to 23).
- Channel 1 is reserved for Page All.
- An extension can be unsubscribe from the "Page All."
- Channel 24 is reserved for **Priority**.
- Channel 25 is reserved for **Emergency**.

Differences between Priority and Emergency:

- Priority will barge into an active call, over intercom, at the intercom adjusted volume level.
- Emergency will barge into an active call, over intercom, at maximum volume.

Configuring Multicast Paging on the Polycom VVX/YealinkT4x Phone Range:

- First, navigate to the <u>Star2Star portal</u>. Choose "Configuration" from the main menu, then the location for which you want to configure Multicast Paging.
- Click the <u>Additional Location Information</u> link in the <u>PHONE SYSTEM CONFIGURATION MENU</u>
 / Basic Settings.

PHONE SYSTEM CONFIGURATION

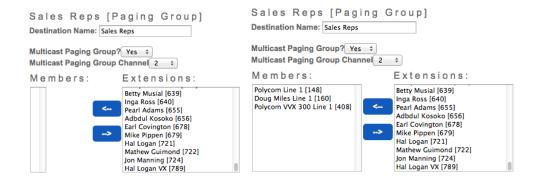
Basic Settings

- Location Information
- Additional Location Information
- Porting
- New Number
- StarNet Analyzer
- Extensions
- Analog Backup / Operator
- Click the "Multicast Paging" dropdown and select "Yes".
- Click "Save Changes" at the bottom of the screen.

Star2Star Communications



- Navigate to "Destinations" in the PHONE SYSTEM CONFIGURATION MENU / Advanced Settings.
- Click "Add New Paging Group".
- Add a name for the Paging Group.
- New groups will default to Multicast groups.
- An existing paging group can be switched to a multicast group.
- Click the "Multicast Paging Group?" drop down and set to "Yes".
- The next available channel is set (note channels 1, 24 and 25 are reserved).
- All Polycom VVX/Yealink T4x Phones will appear in the "Extensions" list.
- Add "Members" as required by selecting <-- or --> to remove.



(Note): Remember to "Save Changes to PBX" when you have completed adding your Multicast Paging Groups. The phones must be rebooted for the multicast paging groups to show up.

Additional Notes:

- When an Administrator converts a pre-existing Paging Group to a Multicast Paging Group, non-VVX phones will be hidden.
- When a user is on a normal phone call and another user initiates a **Priority** or **Emergency** multicast page. The page will be heard over the intercom while the user is still engaged on their regular phone call. They will be able to hear audio from their phone call and the multicast page.
- The user will only be able to see paging groups that they are a member of e.g. extension 100 and 101 are in a paging group. Extension 102 will not be able to see that paging group on their phone. Only extension 100 and 101 will be able to see that paging group.
- A VVX can be removed from the Page All paging group.
 - Login to the Portal.
 - Go to Configuration.
 - Select a Location.
 - Go to Individual Phone Settings.
 - Go to Phone Options
 - Set Unsubscribe From Page All multicast paging group to No.

If a user initiates a multicast page and accepts another incoming call, the multicast page will drop immediately.

Disabling Multicast Paging Group:

- Go to the Portal.
- Go to Additional Location Information.
- Set Multicast Paging to "No".
- Please note that if there are any *Multicast Paging Groups* defined *Multicast Paging* **cannot** be turned off until all Multicast paging groups have been removed.

DTMF Capture Application

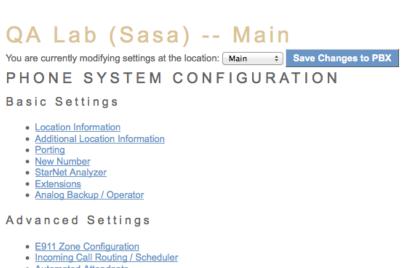
The call is routed to the app (just a regular destination in the Portal)

The user enters some digits (DTMFs)

Digits are validated via one of the Data Validation methods

If digits are a match: route call to **'Success'** destination, otherwise: route to **'Failure'** destination Advanced URL option for data validation

The ability to pass Caller ID Number and Caller ID Name, with either Basic or Advanced URL



- Automated Attendants
- Destinations
- StarCenter Manager
- Star2Star Application Framework
- Multi-Queue Monitor
- StarCenter Recordings
- Recording SettingsOn-Hold Options
- Custom Devices
- FXS Devices
- Inbound CallerID Blocking
- Asset Management
- Starbox Settings
- Configuration Summary
- Print Company Directory

Individual Phone Settings

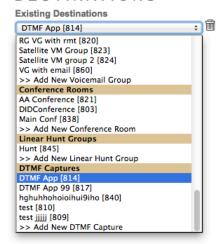
• Individual Phone Settings

QA Lab (Sasa) -- Main

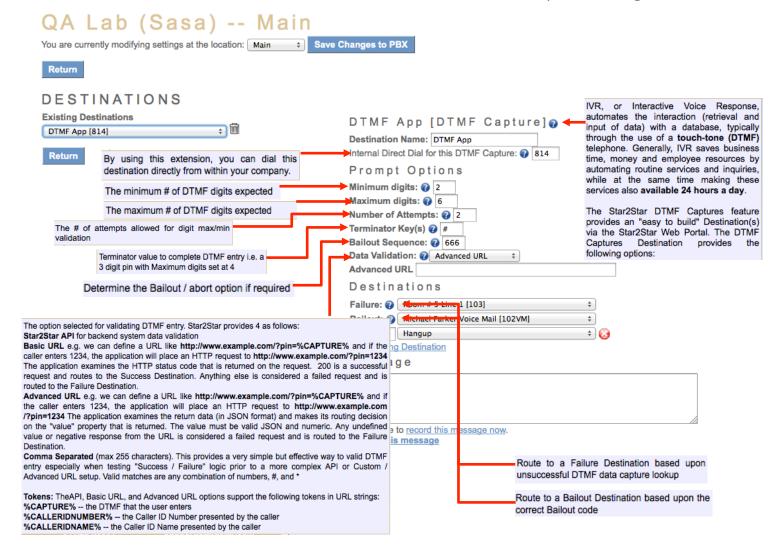
You are currently modifying settings at the location: Main \$\displays Save Changes to PBX

Return

DESTINATIONS



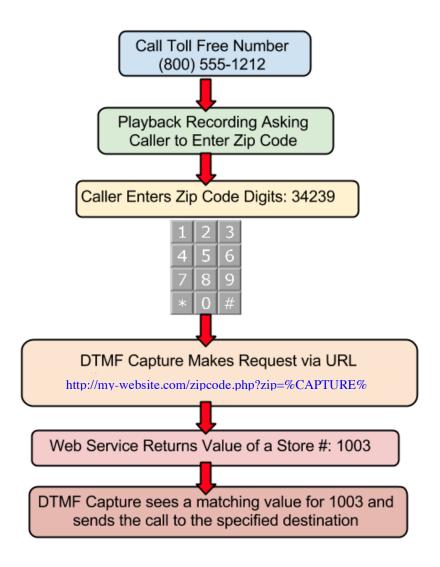
DTMF App [DTMF Capture] ?
Destination Name: DTMF App Internal Direct Dial for this DTMF Capture: 814
Prompt Options
Minimum digits: 2 Maximum digits: 6 Number of Attempts: 2 Terminator Key(s) # Bailout Sequence: 6666 Data Validation: Advanced URL Advanced URL
Destinations
Failure: Room # 5 Line 1 [103] Bailout: Michael Parker Voice Mail [102VM]
1111 Hangup ÷ 🚱
Add Routing Destination
Message
I would like to record this message now. Upload this message



The bailout sequence needs to be within the minimum/maximum digits setting. The first check **initiated** is to verify that the user **made a valid entry** (as defined by minimum/maximum digits and within the defined number of attempts). Assuming that passes, the bailout sequence is checked. Assuming that doesn't match, then it initiates the API / URL / lookup check.

In this scenario, with minimum digits set to 5 and maximum to 6, you could set the bailout sequence to be "66666". You could also lower the minimum digits to 3 and use "666", or you could lower the minimum digits to 1 and use "*", etc.

The particular example below illustrates dialing into a toll free number where the caller would be asked to enter their zip code to find the closest store location of a national retailer. See the call flow diagram below for a visual representation of the call.



The zipcode.php web service as follows:

```
<?php header('Content-type: application/json');
$zipcode = $_REQUEST['zip'];
$con=mysqli_connect('localhost','root','password','database_name');
if (mysqli_connect_errno())
{echo "Failed to connect to MySQL: ". mysqli_connect_error();}
$result = mysqli_query($con,"SELECT * FROM zipcodes where zipcode = '". $zipcode. "'");
while($row = mysqli_fetch_array($result))
{echo '{"value" : ';echo $row[1]; echo ' }';}mysqli_close($con);?>
```

Below is a copy of the database where the lookup is taking place:

mysql> select * from zipcodes;
zipcode store
34232 1001 34234 1001 34236 1002 34237 1002 34239 1003
++ 5 rows in set (0.02 sec)

This is configured in the DTMF capture destination in the portal as shown:

