IP Office Basic Edition

Web Manager R11.1

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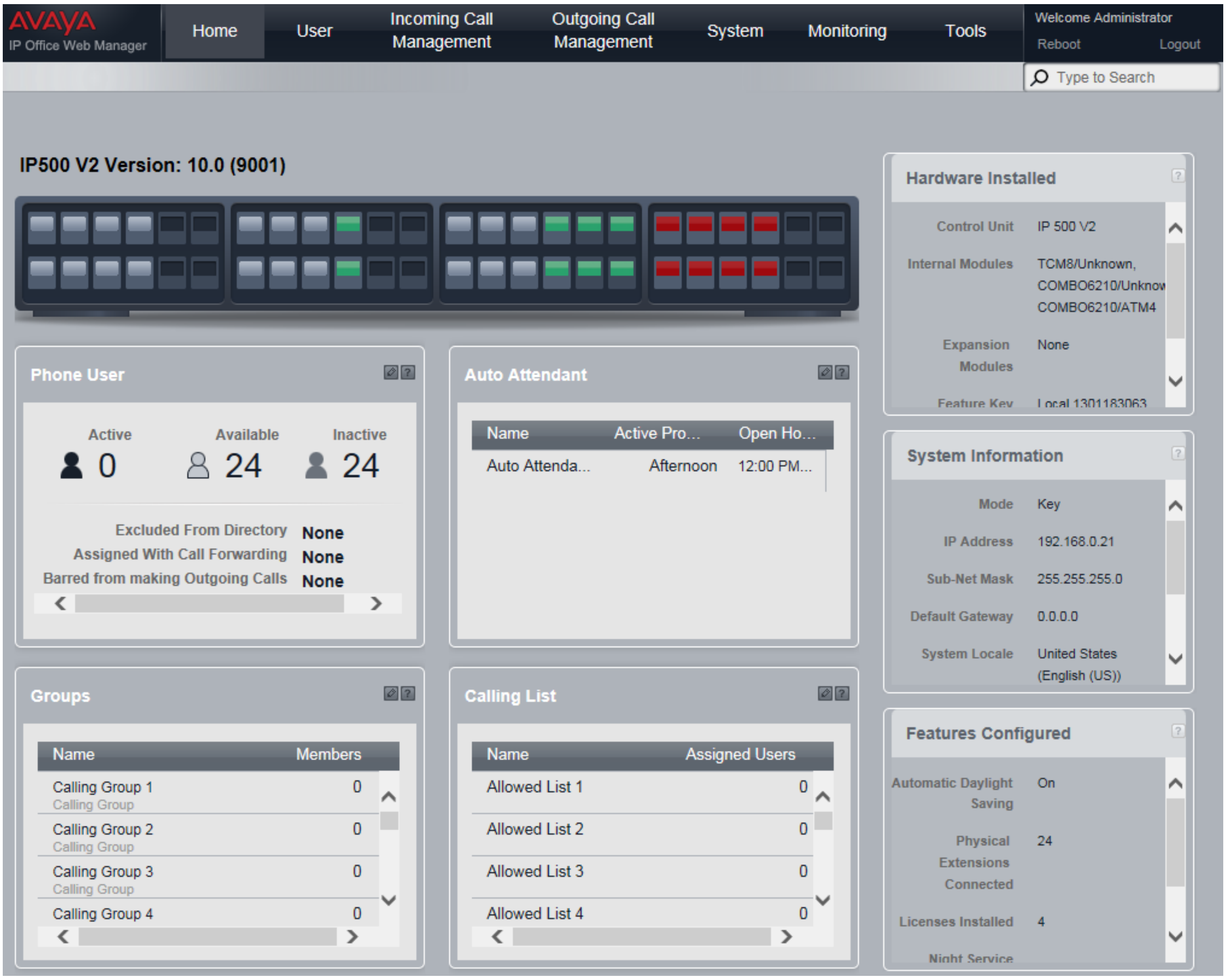
Using Basic Edition Web Manager

1. Using Basic Edition Web Manager

IP Office Basic Edition Release 8.0 and higher systems can be configured from a web browser using Basic Edition Web Manager. This provides access to most system configuration settings.

Basic Edition Web Manager is currently supported with Internet Explorer 10/11, Edge, Firefox, Chrome and Safari 8/9.



Other System Administration Tools

Basic Edition Web Manager can be used for most aspects of system configuration. However, there are a number of other tools that can also be used for system administration. Each tool can perform different functions 238. End users can also make changes to some of their own settings through their phone.

1.1 Enabling Web Manager

For IP Office Release 8.0+, IP Office Basic Edition systems can be managed via web browser. This operation is supported using a range of standard web browsers.

Access to the system is via its IP address and then selecting the ***IP Office Web Management*** link. This documentation covers the recommended initial configuration 134 that can be done via web based management. Full use of web based management is covered in the IP Office Basic Edition Web Base Management manual.

In order to use Basic Edition Web Manager, a number of criteria as listed below must be met. Most of these are applied automatic to a new system installed with IP Office Release 8.0. However, for systems being upgraded to IP Office Release 8.0, additional upgrade steps may be required.

1. The system must be running in IP Office Basic Edition, IP Office Basic Edition - PARTNER Mode or IP Office Basic Edition - Norstar Mode mode. Basic Edition Web Manager is not used for other IP Office modes.
2. The Basic Edition Web Manager files must be present on the System SD card. This can be done in a number of way:
   - By selecting to include those files when prompted to do so while recreating the IP Office SD card using IP Office Manager.
   - By selecting **Upload System Files** when upgrading the system using IP Office Manager.
3. The IP Office system security must allow Basic Edition Web Manager operation:
   - This is done automatically for any new system installed with IP Office Release 8.0 or higher software.
   - This is done automatically for any existing pre-IP Office Release 8.0 system during the upgrade if the system is set to use the pre-IP Office Release 8.0 default password of ***password***.
   - For any system upgraded to IP Office Release 8.0 without first being set back to the default password, either:
     - Using IP Office Manager:
       1. If not already done, select **View | Advanced View**.
       2. Select **File | Advanced | Erase Security Settings (Default)**.
       3. From the Select IP Office dialog, select the required system and click **OK**.
       4. Enter the user name **Administrator** and the password for that account (by default ***password***).
       5. IP Office Manager will confirm if the action was successful or not.
     - Default the system security settings using a DTE cable.

Defaulting the Security on a pre-8.0 IP Office Basic Edition System

In order to allow the security changes necessary for an existing system to support Basic Edition Web Manager after being upgraded to IP Office Release 8.0 or higher, the system must be reset to the default password before being upgraded. This can be done using IP Office Manager or phone based administration.

Using IP Office Manager

1. Start IP Office Manager and receive the configuration from the IP Office system.
2. On the Manager home page, select **Change Remote / Administration Password.**
3. Enter ***password***, the pre-8.0 default, and click **OK**.

Using Phone Based Administration

Refer to the IP Office Basic Edition Phone Based Administration manual. The system administration function #730 is used to set the security password. This function should be used to set the password back to ***password***.

1.2 Displaying a System's IP Address

Logging in 9 to the system using Basic Edition Web Manager requires its IP address. If the system has Avaya telephones connected to it, the following methods can be used to display the current IP address being used by the system.

Using a DS or ETR Phone to Display the System's IP Address

1. With the phone idle, press **Feature** and then dial ***591***. The IP address of the system is displayed.

Using an M-Series or T-Series Phone to Display the System's IP Address

1. With the phone idle, press **Feature** and then dial ***9*81***. The IP address of the system is displayed.

1.3 PC Connection

IP connection to the system is done using the **LAN** port on the back of the system's control unit. During installation, it uses the LAN port to request an IP address from any DHCP server. If there is a DHCP server on the customer's network, that server will give the system an IP address.

If the system was not able to get an address using DHCP when it was first started, it will use the default address ***192.168.42.1/255.255.255.0*** for the LAN port. However, the system is still defaulted as a DHCP client and so will request an address again if it is restarted. Therefore if the system has been started before being connected to the customer's network, it can still be connected and restarted in order to obtain an address from the network.

Normal Network Connection

If the system's control unit is already connected to the customer's network, it probably has an address that is valid on that network, that is an address obtained by DHCP or an address set by the installer.

1. Use the display 8 of an Avaya phone on the system to find out the IP address.
2. Connect your own PC to the customer's network. Most PCs are configured to obtain an IP address using DHCP.
3. Start your web browser and login 9 using the system's address.

LAN Port Direct Connection

If the system is not connected to a customer network, it is most likely using its default address ***192.168.42.1/255.255.255.0***. Connection in this case requires you to know how to temporarily change the IP address settings of your PC.

1. Use the display 8 of an Avaya phone on the system to find out the IP address.
2. Set the IP address of your PCs network port to be a valid address on the same network address range.
    - For example, if the system is using its default address, set your PCs address to 192.168.42.20/255.255.255.0.
3. Connect your PC to the LAN port on the system.
4. Start your web browser and login 9 using the system's address.

WAN Port Direct Connection (Fallback Method)

The WAN port on the rear of the system's control unit is not normally used for any function. However it can be used for Basic Edition Web Manager if it not possible to determine the system's IP address by any other method: For example if the system was given a fixed IP address but only has analog extensions which cannot be used to display that current address.

The WAN port address is always 192.168.43.1/255.255.255.

1. Set the IP address of your PCs network port to be a valid address on the same network address range. For example, set your PCs address to 192.168.43.20/255.255.255.0.
2. Connect your PC to the WAN port on the system.
3. Start your web browser and login 9 using the address 192.168.43.1/255.255.255.0.
4. Once you have logged in, check the actual address of the LAN port. It is shown on the **Switch** 65 menu form.

1.4 Logging In

In order to login you need to know the IP address 8 of the system and to connect your PC 8 to it or the network which it is already on.

1. In a web browser, enter the IP address of the system in the format *http://<IP Address>*, for example ***http://192.168.42.1***.

2. The web page shown displays a number of links, select the ***IP Office Web Management*** link.

    - If the IP Office Web Management link is not shown, there are a number of possible causes; either the system has not been upgraded to IP Office R8 or higher or upload of Basic Edition Web Manager files was not selected during the upgrade or the system is running in IP Office Essential Edition mode which does not use Basic Edition Web Manager.

    - As an alternative you can enter the full address for Basic Edition Web Manager directly. Enter the following address into the browser's address bar, replacing *<IP Address>* with the system's IP address. Note that the address is case sensitive: *https://<IP Address>:8443/webmanagement/WebManagement.html*

3. If the browser responds with a security warning, follow the menu settings displayed for continuing with the connection.

    - If using Internet Explorer 8, you will also see a warning asking *"Do you want to view only the webpage content that was delivered sercurely?"*. Select ***No***. If you select ***Yes***, the System Status page within web management will be blank.

4. When the login menu is displayed, enter the user name and password for system administration.

    - There are two default accounts that cannot be deleted and which have full configuration access. The defaults user names and password for these are:

        - ***Administrator / Administrator***
          This user can access the system configuration to make changes. This is the same account as used for IP Office Manager. This is the account used is **Simplified Login** is selected.

        - ***Business Partner / Business Partner***
          This user can access the system configuration to make changes. They can also setup and configure additional service user accounts 116 for other Basic Edition Web Manager user.

5. Click on **Login**.

6. The home page 31 for the system Basic Edition Web Manager is displayed.

    - Do not use the browsers forward, back and other history functions while in Basic Edition Web Manager. Doing so will require you to log in again.

    - Pages in the systems Basic Edition Web Manager cannot be bookmarked.

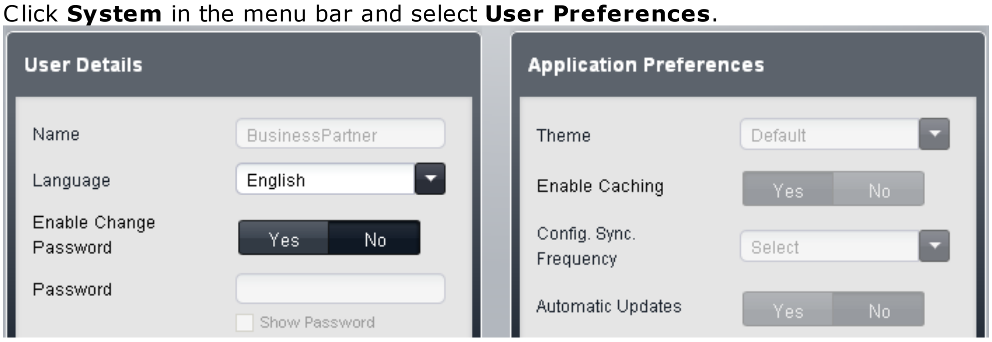
    - You must remember to log out 10 when you have finished editing the configuration. The browser is not automatically logged out after any duration.

1.5 Changing Your Password

Once you have logged in, you can change the name and password used for the login.

Changing Your Name and Password Settings

1. Click **System** in the menu bar and select **User Preferences**.



2. The name and password are shown in the **User Details** panel. Change these to the values that you want to use in future.
3. The other settings relate to how often and when the configuration settings loaded should be updated. Refer to **System | User Preferences** 116.
4. Click **Save**.

1.6 Logging Out

You must remember to log out when you have finished editing the configuration. The browser is not automatically logged out after any duration.

While simply closing the browser will end the Basic Edition Web Manager session, it may be before all the settings that have been changed have been saved to the system. Therefore it is recommended that you always end a Basic Edition Web Manager session by using the log out process below.

1. Click on the **Logout** link shown at the top right of the window.
2. In the confirmation window that appears, click **Yes**.
3. Your Basic Edition Web Manager session is ended and the log in screen is shown.

1.7 Configuring Web Manager Users

If you are logged in with the ***BusinessPartner*** name, you can use Basic Edition Web Manager to create addition Basic Edition Web Manager users. During this process you can define what type of access each user has and the menus that the user can use.

This menu is access by selecting **System** in the menu and clicking on **User Preferences**.

Adding a New Service User

1. Click **System** in the menu bar and select **User Preferences**. Click the edit icon in **Role Based Rights** panel. The existing additional Basic Edition Web Manager user are listed.
2. Click **Add Service User**.
3. Enter and select the values for the service user account.
4. Click **Save**.

Editing a Service User

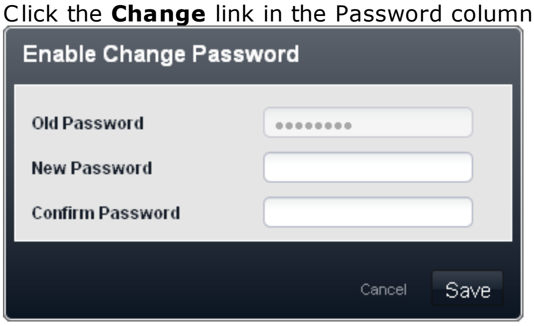
1. Click **System** in the menu bar and select **User Preferences**. Click the edit icon in **Role Based Rights** panel.
2. Double click on the service user's current details.
3. Change the settings as required.
4. Click **Save**.

Deleting a Service User

1. Click **System** in the menu bar and select **User Preferences**. Click the edit icon in **Role Based Rights** panel.
2. Click the **Delete** link adjacent to the service user's details.
3. You will be prompted to confirm the action. Click **Yes**.

Changing a Service User's Password

1. Click **System** in the menu bar and select **User Preferences**. Click the edit icon in **Role Based Rights** panel.
2. Click the **Change** link in the Password column.



3. Enter and confirm the new password.
4. Click **Save**.

Resetting a Service User's Password

1. Click **System** in the menu bar and select **User Preferences**. Click the edit icon in **Role Based Rights** panel.
2. Click the **Reset** link in the Password column.
3. You will be prompted to confirm the action. Click **Yes**.

Chapter 2. Telephony Overview

2. Telephony Overview

Call Routing Modes

The system can operate in either of two modes; ***PBX*** or ***Key***. The selected mode affects the system's outgoing call routing 17 and incoming call routing 18 settings. The default setting is set by the type of System SD card fitted during installation, see below.

Basic Edition Modes

The system can operate as a IP Office Basic Edition, IP Office Basic Edition - Norstar Mode or IP Office Basic Edition - PARTNER Mode system. This is set by the type of System SD card fitted to the system during installation. The modes are all similar in operation. The few differences are related to the different global areas in which they are made available and thus the equipment also supported in those countries. For example IP Office Basic Edition - PARTNER Mode is only available in North American countries and so does not support BRI trunks.

Default Modes

The default mode is set by the type of System SD card fitted to the system:

- **IP Office U-Law**
  A system fitted with this type of card will default to U-Law telephony and ***Key*** mode operation. Intended for North American locales. System's running in this mode are referred to as IP Office Basic Edition systems.
- **IP Office A-Law**
  A system fitted with this type of card will default to A-Law telephony and ***PBX*** mode operation. Intended for locales outside North America. System's running in this mode are referred to as IP Office Basic Edition systems.
- **IP Office Partner Mode**
  A system fitted with this type of card will default to U-Law telephony and ***Key*** mode operation. Supported only in North American locales. System's running in this mode are referred to as IP Office Basic Edition - PARTNER Mode systems.
- **IP Office Norstar Mode**
  A system fitted with this type of card will default to A-Law telephony and ***Key*** mode operation. Supported only in Middle East and North African locales. System's running in this mode are referred to as IP Office Basic Edition - Norstar Mode systems.

2.1 Dial Plan

Extension Numbering

The system can be configured to use either a 2 digit or 3 digit dial plan for user extensions. Note however that changing extension numbering is done through either phone based administration 25 or using the IP Office Manager 6 application.

- For a 2 digit dial plan, the extensions are numbered 10 to 57. This numbering cannot be changed.
- For a 3 digit dial plan, the extensions are numbered from 100 upwards. This numbering can be changed in the range 100 to 579 (the defaults are 100 to 199). In 2 digit mode only 48 extensions are supported, in 3 digit mode a maximum of 100 extensions are supported.
- In both cases, those extensions not matched by physical ports are automatically assigned as phantom extensions 23. That is, extension numbers are assigned to all possible extension regardless of whether the necessary physical card or extension expansion module is installed or not.
- The system assumes that the base control unit is always fully populated with up to 32 extensions, either real or phantom or a mix, to which it assigns extension numbers in sequence. It does this before assigning extension numbers to any real extensions on attached external expansion modules up to the system extension limit. If the system extension limit has not been exceeded, any remaining extension numbers are assigned to additional phantom extensions.

Special Dialed Numbers

The following can be dialed after selecting an **Intercom** button or simply going off hook (for which Intercom is assumed).

Number	Function	Description
0	**Operator**	Calls the first extension in the system.
610 to 657	**Extension Pickup**	Answer the call alerting at another extension. Dial **6** followed by the extension number.
661 to 664	**Group Pickup**	Dial **66** followed by the pickup group number (1 to 4).
6801-6864	**Line Pickup**	Answer the call alerting on a particular line. Dial **68** followed by the line number (01 to 64).
70	**Loudspeaker Page**	Makes a call to the extension configured as the system's **Loudspeaker Paging** 208 extension.
71-74	**Calling Group**	Dial **7** followed by the calling group number (1 to 4). Prefix the number with * to page the group.
75	**Operator Group**	This is supported only on systems with their **Mode** set to ***PBX***. Prefix the number with * to page the group.
76	**Modem**	Modem port 24. Used for remote access for configuration.
771 to 776	**Hunt Group**	Dial 77 followed by the hunt group number (1 to 6). Prefix the number with * to page the group.
777	**Voicemail Collect**	Connects the extension to the extension user's own mailbox.
778	**Remote Voicemail Access**	Connects the extension to prompts to specify the mailbox required. This voicemail code of the mailbox is then requested.
7801 to 7809	**Auto Attendant Access**	Call the auto attendant (1 to 9) specified.
801 to 864	**Idle Line Pickup**	Seize a line in order to then make a call on that line. Dial 8 followed by the line number (01 to 64).
865 to 899	**Seize a Line**	Seize an available trunk in the ARS selector group (65 to 99). This is supported only on systems with their **Mode** set to ***PBX***.
9	**External Call Prefix**	**Key:** Start an outgoing external call. The line used is automatically selected using **Idle Line Preference**.
		PBX: Optional external dialing prefix. The use of 9 can be removed or swapped with 0 (the operator number) using the **Outside Line** 65 setting.
*****	**Page/Direct Call**	Putting * in front of an internal number will attempt to make either a page or direct call. If the target is a group, the call is a page call to all the idle members of the group. If the target is an extension, the call is an auto answered call to that number. If the target cannot auto answer, the call becomes a normal call.
***70**	**Simultaneous Page**	Make a page call to the users in Calling Group 1 and to the extension configured as the system's **Loudspeaker Paging** 208 extension.

Auto Attendant Numbers

Dialing the appropriate number shown in the table below allows recording and playback of the matching auto attendant prompt. The numbers are also displayed in the web manager auto attendant menus.

It is important to understand that callers to an auto attendant hear more than one prompt

1. If the auto attendant's **Emergency Greeting** setting is enabled, the recorded emergency greeting if played.
2. Depending on the time profile being used, the morning, afternoon, evening or out of hours greeting is then played.
3. Finally the morning, afternoon, evening or out of hours menu is played.

	Auto Attendant								
Greeting Prompts	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**
Morning Greeting	7811	7812	7813	7814	7815	7816	7817	7818	7819
Afternoon Greeting	7821	7822	7823	7824	7825	7826	7827	7828	7829
Evening Greeting	7831	7832	7833	7834	7835	7836	7837	7838	7839
Out of Hours Greeting	7851	7852	7853	7854	7855	7856	7857	7858	7859
Emergency Greeting	7861	7862	7863	7864	7865	7866	7867	7868	7869
Action Prompts									
Morning Menu	7841	7842	7843	7844	7845	7846	7847	7848	7849
Afternoon Menu	7871	7872	7873	7874	7875	7876	7877	7878	7879
Evening Menu	7881	7882	7883	7884	7885	7886	7887	7888	7889
Out of Hours Menu	7891	7892	7893	7894	7895	7896	7897	7898	7899
Auto Attendant Access	**7801**	**7802**	**7803**	**7804**	**7805**	**7806**	**7807**	**7808**	**7809**

The auto attendant access numbers shown above allow internal access to an auto attendant. Calls can be transferred to these numbers.

2.2 Outgoing Call Routing

Key Mode

Each phone is configured with 2 Intercom buttons which cannot be changed. It is also configured with line appearance buttons for specific lines using the **Number of Lines** settings and individual button programming.

- Internal calls are made by selecting one of the two Intercom buttons provided on each phone and then dialing the number of another extension or of the system feature required.
- External calls are made by selecting one of the line appearance buttons programmed on the phone and then dialing the external number required.
- If the user dials without first selecting an Intercom or Line button, the user's automatic line selection 144 setting is used to determine which button, if available, gets used.

PBX Mode

Each phone is configured with 3 call appearance buttons (2 only on ETR phones). These can be used to make both internal and external calls. The dialing of an external call can be indicated by the dialing starting with a specific prefix (9 or 0) if required, otherwise any number not matching an internal extension or function is automatically assumed to be external.

The line used for an outgoing external call is determined by a set of **Alternate Route Selection** (ARS) settings:

- ARS Selectors are created. These are groups of lines or selectors for specific functions using any available ISDN line.
- Different classes of call (sets of external number prefixes) are then mapped to those ARS Selectors. The system supports classes for **Emergency**, **National**, **International**, **Cell** and **Toll Free** sets of prefixes. An additional **Local** class is used for any calls that do not match one of the other classes.

When a user dials an external number, it is matched to a selector and uses the function and one of the lines specified by that selector. For SIP trunks set to call by call mode, each call by call entry also has an ARS selector settings which allows it to also be used for outgoing calls.

Line appearances can still be used to make and answer calls on a particular line but are not added by default. They can also be used to select a particular ARS selector for an outgoing call.

Dialing Restrictions

In both modes, the system uses a number of methods to control the external numbers which users are allowed to call.

- **Allowed Number Lists** 188 / **Disallowed Number Lists** 186
  These lists are used in define numbers that can or cannot be dialed. Users are then associated with the different lists.

Allowed Numbers	Each allowed list contains external telephone numbers that members of the list are allowed to dial. The allowed lists to which a user is assigned override any disallowed lists to which they are also assigned. The numbers in the user's assigned allowed lists also override the **Calls Barred** and **Outgoing Call Restrictions** 43 settings that may be applied to a user. There are eight lists, each containing up to 10 numbers. Each number can use the telephone dialing digits 0 to 9, *, # and can be up to 28 digits long. You can also use the ? character as a single digit wildcard.
Disallowed Numbers	Each disallowed list contains external telephone numbers that users who are assigned to the list cannot dial. There are eight lists, each containing up to 10 numbers. Each number can use the telephone dialing digits 0 to 9, *, # and can be up to 28 digits long. You can also use the ? character as a single digit wildcard. Numbers in the disallowed lists of which a user is a member are overridden if they also appear in the allowed numbers lists, emergency number list of which the user is a member and also by marked system speed dials.
Emergency Numbers	You can enter 10 emergency phone numbers into this list. This list is applied to all users and overrides any dialing restrictions that may also be applied to the users. Use of numbers in the list can be viewed using a 911-View/E-View 220 button.

- **Account Codes** 190
  Each user can be configured to need to enter a valid account code whenever they make an external call.
- **Outgoing Call Restrictions** 185
  For each user, the type of external calls that the user is able to make can be configured.
- **Marked Speed Dials** 191
  When a user uses a stored system speed dial number, the actual number dialed is subject to all the call barring methods as if the user had dialed the number directly. However system speed dials set as 'marked speed dials' override any call restrictions.

- **Night Service** 210
  When the system is set to night service, any users in the **Night Service Group** need to enter the system password when making an external call.

2.3 Incoming Call Routing

The options for routing incoming calls depend on whether the system is set to ***PBX*** or ***Key*** mode.

Key Mode

For an incoming external calls on a line, the following options control where the call is presented:

- **Line Appearance Buttons**
  The call will alert on any line appearance buttons that matches the line. Each line has a line number which can be assigned to line appearance buttons on users' phones. Users can answer the call by pressing the alerting line appearance button on their phone.
  - **Number of Lines**
    By default, all analog lines in the system are assigned to line appearance buttons when the system is installed. Lines are assigned for all users starting from button 03 upwards in order of line numbering.
  - **Line Assignment**
    Through individual user button programming, any programmable button can be configured as a line appearance for a particular line.
- **Coverage Destination**
  The **Coverage Destination** setting of each line can be used to select whether an incoming call on that line is also presented to one of the following options in addition to alerting on any matching line appearances. For PRI and BRI trunks, it is not possible to know on which of the trunk's channels incoming calls will arrive. Therefore in most cases, the coverage destination and other settings of each line on the trunk should be set to the same values.
  - **Coverage Extension**
    The call alerts on an intercom button of a selected line coverage extension. The user's call coverage, VMS coverage and call forwarding settings are applied to the call. Any extension can be used as the destination including a phantom extension.
  - **Hunt Group**
    The call is presented, in sequence, to each of the available members of a selected hunt group until answered. Any of the 6 rotary hunt groups can be used as the destination.
- **Auto Attendant Coverage**
  Each trunk or trunk channel can be configured to send unanswered calls to an auto attendant after a set delay (which can be set to 0 for immediate answer). This can be set to operate when the system is in day and or night service. This is done using the **VMS Schedule**, **VMS Delay - Day**, **VMS Delay - Night** and **VMS Auto Attendant** settings of each line.

The following methods can be used to override the normal call routing detailed above:

- **DID Call Mapping**
  For BRI, ETSI PRI and PRI trunks, if the incoming call matches a configured DID and or ICLID number, the **Coverage Destination** setting for the DID/ICLID match is used rather than the line's **Coverage Destination**. DID can also be used on some types of T1 trunk.
- **SIP Call by Call Table**
  For SIP trunks, if the incoming call matches a configured URI, it is presented to the extension or group specified in the SIP line's **Call by Call Table**.
- **Night Service**
  Switching on night service overrides the routing of calls to Coverage Destinations. Instead the calls change to alerting the users who are members of the Night Service group. The settings for auto attendant coverage (VMS Schedule) can also be varied depending on whether the system is in night service or not.

PBX Mode

In PBX mode, a new group, the **Operator Group**, is used as a the default destination for call. This group contains the first extension on the system.

- For analog trunks trunks, the trunk's **Coverage Destination** is defaulted to the ***Operator Group*** but can be changed if required.
- For PRI and BRI trunks all incoming call routing is done by DID Call Mapping. Each DID table has a non-removable default route which is used for any calls that do not match any other specific DID entry. The destination for this default entry is the Operator Group.
- SIP trunks are defaulted to call by call operation, again with a default call by call destination of the ***Operator Group***.

The following new destinations for incoming calls are available:

- **Operator Group**
  This group is the default destination for all incoming calls. The group contains the first extension on the system but can be edited to contain other extensions.
- **Calling Groups**
  In ***Key*** mode these 4 groups are only used internally. In ***PBX*** mode these groups are also available as a destination for trunk calls in the **Coverage Destination** selections, DID Call Mapping and SIP Call by Call tables. A calling group can also be selected as the destination for an auto-attendant transfer.

Night Service Mode

In both modes, when the system is put into night service, all incoming calls except those to specific DID call mapping or SIP call by call destinations, are rerouted to alert the users who are members of the night service group.

2.3.1 Coverage Destination Summary

The table below summarizes the supported destinations for coverage destinations. The options depend on the trunk type and the operating mode of the system.

Coverage Destinations	Key Mode						PBX Mode					
	Alog	BRI	ETSI PRI	PRI	T1	SIP	Alog	BRI	ETSI PRI	PRI	T1	SIP
• **None** If set to ***None***, incoming calls will only alert on user extensions with line appearance buttons that match the line's **Appearance ID**.	✓*	✓*	✓*	✓*	✓*	✓*	✓	–	–	–	✓*	✓*
• **Extension** Route incoming calls to a particular extension.	✓	✓	✓	✓	✓	✓	✓	–	–	–	✓	✓
• **Phantom Extension** A phantom extension can be selected as the destination for calls.	✓	✓	✓	✓	✓	✓	✓	–	–	–	✓	✓
• **Hunt Group** Incoming calls can be routed to one of the 6 rotary hunt groups.	✓	✓	✓	✓	✓	✓	✓	–	–	–	✓	✓
• **Voicemail** Route incoming calls to the systems voicemail to collect messages. This requires the caller to know the mailbox number and passcode.	✓	✓	✓	✓	✓	✓	✓	–	–	–	✓	✓
• **Operator Group** For systems with their **Mode** set to ***PBX***, incoming calls are routed to the **Operator Group**.	–	–	–	–	–	–	✓*	–	–	–	–	✓
• **Calling Group** For systems with their **Mode** set to ***PBX***, incoming calls can be routed to one of these 4 ring all groups.	–	–	–	–	–	–	✓	–	–	–	✓	✓

* = Default destination.

2.3.2 DID/Call-by-Call Summary

The table below summarizes the supported destinations for DID call mapping and SIP call-by-call settings. The options depend on the trunk type and the operating mode of the system.

DID Call Mapping/SIP Call-by-Call Destinations	Key Mode						PBX Mode					
	Alog	**ETSI PRI**	**BRI**	**PRI**	**T1**	**SIP**	**Alog**	**ETSI PRI**	**BRI**	**PRI**	**T1**	**SIP**
• **Extension** Route incoming calls to a particular extension.	–	✓	✓	✓	✓	✓	–	✓	✓	✓	✓	✓
• **Phantom Extension** A phantom extension can be selected as the destination for calls.	–	✓	✓	✓	✓	✓	–	✓	✓	✓	✓	✓
• **Hunt Group** Incoming calls can be routed to one of the 6 rotary hunt groups.	–	✓	✓	✓	✓	✓	–	✓	✓	✓	✓	✓
• **Voicemail** Route incoming calls to the systems voicemail to collect messages. This requires the caller to know the mailbox number and passcode.	–	✓	✓	✓	✓	✓	–	✓	✓	✓	✓	✓
• **76: Modem** The option ***76: Modem*** can be selected to route the call to the systems built in V32 modem function. This is only intended for basic access by system maintainers.	–	✓	✓	✓	✓	✓	–	✓	✓	✓	✓	✓
• **Auto Attendant** Any configured voicemail auto attendants can be selected as the call destination.	–	✓	✓	✓	✓	✓	–	✓	✓	✓	✓	✓
• **Operator Group** For systems with their **Mode** set to ***PBX***, incoming calls are routed to the **Operator Group**.	–	–	–	–	–	–	–	✓*	✓*	✓*	✓*	✓
• **Calling Group** For systems with their **Mode** set to ***PBX***, incoming calls can be routed to one of these 4 ring all groups.	–	–	–	–	–	–	–	✓	✓	✓	✓	✓

* = Default destination for fixed Default DID entry in DID Call Mapping table, ie. matches any call where there is no other specific match.

2.4 Supported Telephones

The following phones are supported by systems running IP Office Release 11.1 software.

Avaya DS Digital Stations

These phones use digital station (DS) ports provided by IP500 base cards (***DS8*** and ***Combo DS6-P2***). They can also use the DS ports provided by Digital Station 16 and Digital Station 30 external expansion modules.

- **Avaya 1400 Series:** ***1403***, ***1408*** and ***1416***.
- **Avaya 9500 Series**: ***9504*** and ***9508***.

Avaya TCM Digital Stations

These phone use ports provided by the IP500 ***TCM8*** base card or by ***DS16A***/***DS30A*** external expansion modules.

- **Avaya M-Series:** ***MT7100***, ***MT7100N***, ***MT7208***, ***MT7208N***, ***M7310***, ***M7310N***, ***M7324*** and ***M7324N***.
- **Avaya T-Series:** ***T7000***, ***T7100***, ***T7208***, ***T7316***, ***T7316E***.
- **Other Phones:** ***Avaya 4100 Series***, ***Avaya 7400 Series*** and ***Audio Conferencing Unit (ACU)***.
- Additional programmable buttons are supported by the addition of button modules on M7324 and T7316E phones.

Avaya ETR Phones

Avaya ETR (Enhanced Tip and Ring) phones are supported on both Avaya PARTNER ACS telephone systems and IP Office systems. On IP Office systems they connect to ETR ports provided by IP500 ***ETR6*** base cards.

- **ACS "Refreshed" Series:** ***ETR6D***, ***ETR18D***, ***ETR34D***.
- **ACS "Euro" Series**: ***ETR6***, ***ETR18***, ***ETR18D***, ***ETR34D***.
- **Avaya DECT Phones:** ***3920***. This DECT phone consists of a paired base station and cordless handset. It connects to ports provided by the ***IP500 ETR6*** base card. Supported in North America only.

Analog Phones

The system supports DTMF analog phones. These connect to PHONE extension ports provided by IP500 base cards (***Phone 2***, ***Phone 8*** and ***Combo DS6-P2***) or external expansion modules (***Phone 16*** and ***Phone 30***). Avaya cannot guarantee the operation of any non-Avaya analog phones on the system. Analog phones can also be connected to ports on the ***IP500 ETR6*** base card.

2.5 Date and Time Setting

By default the system is configured to use network time synchronization using the first analog trunk on the card installed in slot 1 of the system control unit. In that mode it gets its system time and date from the information that the line provider includes as part of the caller ID information. When network time synchronization is being used, system in a North American locale can also be configured to apply automatic daylight saving changes.

If the network time synchronization method above cannot be used on a particular system, it needs to be disabled. The time and date are then set manually. This is all done using a system administrator phone 25.

2.6 Phantom Users

Extension users are created in the system configuration for all possible users regardless of whether they are matched by physical extension ports. Those user extensions without a physical port are referred to as 'phantom' extensions.

The main purpose of phantom extensions is to provide voicemail mailboxes that are not associated with an existing physical extension. These mailboxes can be accessed and used by the auto attendant menus and other functions.

- The system assumes that the base control unit is always fully populated with up to 32 extensions, either real or phantom or a mix, to which it assigns extension numbers in sequence. It does this before assigning extension numbers to any real extensions on attached external expansion modules up to the system extension limit. If the system extension limit has not been exceeded, any remaining extension numbers are assigned to additional phantom extensions.

The Manager application's menus and phone based administration menus allow selection of a phantom user extension number in the same was as for normal physical extension numbers. Phantom extensions are indicated by # in front of the extension number. That includes using a phantom extension as the destination in an auto attendant, trunk DID call map, SIP call by call mapping, etc.

- Calls to a phantom extensions are treated as follows:
  - Calls go immediately to the phantom user's voicemail mailbox. Forwarded or transferred calls go to the mailbox of the user doing the transfer or forward.
  - If the phantom extension is included in a hunt group, they are ignored.
- Callers can use the phantom user's mailbox DTMF breakout settings, if configured, to be transferred to another destination.
- Calls can be transferred to a phantom extension. Since the calls go immediately to voicemail, no transfer return is supported.
- Joining or bridging to a call that has been sent to a phantom extension’s mailbox will drop the phantom extension from the call in the same way it does for a physical extension.
- Calls to a phantom extension cannot be picked up.
- The phantom extensions are supported within the auto attendant actions ***Dial by Name***, ***Dial by Number*** and ***Transfer to Number***.
- Mailbox access for message collection and mailbox configuration is achieved by dialing 778 from any telephone, then entering the phantom extension number and the mailbox access code if it has already been configured. Mailboxes with a configured access code can also be accessed by external calls.
- Phantom extensions can be used as the line coverage extension for a line. In this case, the phantom extension's **VMS Coverage Rings** setting is used before the call goes to the phantom user's mailbox.
- Auto Dial Intercom buttons can be set to route calls to a phantom extension.
- When using the Manager application, when selecting extensions in the various menus, a phantom extension is indicated by a # character. The extensions **Equipment Type** is fixed as ***Phantom***.
- The phantom extension's **Automatic VMS Coverage** setting can be used to disable mailbox operation. If this is done, calls to the phantom extension will hear busy tone.

The following features are specifically are not supported using phantom extensions:

- A phantom extension cannot be configured as any other extension type, ie. loudspeaker, door phone, fax machine or standard extension.
- A phantom extension cannot be configured as a night service alert extension.
- A phantom extension cannot be configured as a hotline extension.
- A phantom extension cannot be added to a hunt group, pickup group or calling group.
- A phantom extension specified as the destination for call forwarding or follow me is ignored. Instead calls will continue to alert at the forwarded user.
- A phantom extension specified as the destination for another extension's call coverage is ignored. Instead calls will continue to alert at the covered extension.

2.7 Modem Access Support

The first analog line port in any system can be used for V32 modem access. The line is switched between modem operation and normal voice call operation by dialing *9000* or through the Modem Enabled option shown in the trunk's advanced setup settings. When operating as a modem, the line cannot be used for normal voice calls.

The modem functionality can also be accessed as extension 76. This can be used as the destination in an auto attendant menu in the DID mapping/SIP Call-by-Call tables of trunks. This allows remote access on lines other than the first analog line. This method can be used without first having to use the Modem Enabled option.

2.8 SIP Trunks

The system can support SIP trunks through its LAN connection. These are configured using IP Office Manager, they cannot be managed through phone based administration.

In order to support SIP trunks, the system must include the following resources:

- **SIP Trunk Licenses**
  These licenses are used to configure the number of simultaneous SIP trunk calls supported, up to a maximum of 20. A The system supports up to 3 channels without licenses.

- **Voice Compression Channels**
  These are required to convert between the audio compression methods used for IP telephony and those used for analog and digital trunks. Each IP500 Combination card (up to 2) installed in the system provides 10 voice compression channels for the system. One voice compression channel is used for each SIP call.

2.9 Phone Based Administration

Many of the system settings can also be programmed from extensions on the system if they are Avaya telephones. The following phone types can be used: ETR18D, ETR34D, M7324, M7310, T7316E, T7316, 1408, 1416, 9504 and 9508. For full details of these function refer to the separate Phone Based Administration manual.

The type of phone based administration possible can be divided into three categories:

- **System Administration**
  This refers to the system administration that can only be done by the first two extensions on the system. That is, the first two extension ports at the left-hand edge of the system's control unit.
- **Centralized Programming**
  This refers to the administration of other user settings that can be done from the first two extensions in the system.
- **Telephone Programming**
  This refers to the administration of their own settings that a user can do from their own phone. All these functions can also be done by centralized programming.

System Administration Functions

This refers to the system administration that can only be done by the first two extensions on the system. That is, the first two extension ports at the left-hand edge of the system's control unit.

Category	Setting	
System Settings	**System Locale**	–
	System Language	–
	System Mode	–
	Default Numbering	#734
	Recall timer	#107
	Wake up Service Button	#115
	Log Caller ID Extensions	#317
		#318
		#319
Key System	**ARS Selectors**	–
	Calls Out	–
	Outgoing Call Prefix	–
Date and Time	**Automatic Daylight Saving**	#126
	Network Time Synch	#128
	System Date	#101
	System Time	#103
Line Settings	**Number of Lines**	#104
	Line Assignment	#301
	Line Coverage Extension	#208
	Assign Line to AA	#210
	Co Disconnect Time	#203
	Ring Pattern	–
Auxiliary Equipment	**Contact Close Grp**	#612
	Type - Contact Close	#613
	Doorphone 1 Extension	#604
	Doorphone 2 Extension	#605
	Doorphone Alert Ext	#606
	Internal Hotline Ext	#603
	Loudspeaker Paging Ext	#617
	Fax Machine Extension	#601

Category	Setting	
Extension Settings	**Automatic Extension Privacy**	#304
	Display Language	#303
	Call Waiting	#316
	Intercom Dial Tone	#309
	External Hotline	#311
	Outside Conference Denial	#109
	Transfer Return Ext	#306
	Override Line Ringing	#324
	Remote Call Forward	#322
Account Codes	**Forced Account Code**	#307
	Forced Account Code List	#409
Voicemail	**VMS Coverage**	#310
	VMS Coverage Rings	#321
	Reset Voice Mail Pwd	#325
	VMS Hunt Delay	#506
	VMS Hunt Schedule	#507
Groups	**Calling Group**	#502
	Group Call Distribution	#206
	Hunt Group	#505
	Night Service Button	#503
	Night Service Grp Ext	#504
	Operator Group	–
	Pickup Group	#501
Ring Settings	**Abbreviated Ringing**	#305
	Call Coverage Rings	#320
	Distinctive Ringing	#308
	Line Ringing	–
	Personal Ring Pattern	#323
	Ringing on Transfer	#119
	Transfer Return Rings	#105

Category	Setting	
Dialing Restrictions and Permissions	**Allowed Lists**	#407
	Allow To	#408
	Disallowed Lists	#404
	Disallow To	#405
	Emergency List	#406
	Outgoing Call Restr	#401
	Set System Password	#403
	Toll Call Prefix	#402

Category	Setting	
Holding Calls	**Hold Timer**	#127
	Music on Hold	#602
System Maintenance	**Clear Backup Alarm**	#123
	Manual Backup	#124
	Memory Card Shutdown/Startup	#733
	Restore	#125
	Copy Settings	#399
	Remote Admin Password	#730
	System Default	#989
	System Copy	#732
	System Reset	#728
	System Shutdown	#729
	System Upgrade	#731

Centralized Programming Functions

This refers to the administration of other user settings that can be done from the first two extensions in the system.

Category	Setting	
Line Settings	**Line Ringing Pattern**	#209
	Auto Line Selection	–
Speed Dials	**Personal Speed Dial**	80-99
	System Speed Dials	600-699
Extension Settings	**Extension Name**	–
	Button Programming	–
	Do Not Disturb Exceptions	700-719

Telephone Programming Functions

This refers to the administration of their own settings that a user can do from their own phone. All these functions can also be done by centralized programming.

Category	Setting	
Speed Dials	**Personal Speed Dial**	80-99
Extension Settings	**Extension Name**	–
	Button Programming	–
	Do Not Disturb Exceptions	700-719

2.10 Voicemail and Auto Attendant Languages

The language used is set by the system language setting. However this can be overridden by the language setting of a particular auto attendant or user. The possible languages are:

- ***Arabic***, ***Cantonese***, ***Danish***, ***Dutch***, ***English (UK)***, ***English (US)***, ***Finnish***, ***French***, ***French (Canada)***, ***German***, ***Italian***, ***Korean***, ***Mandarin***, ***Norwegian***, ***Polish***, ***Portuguese***, ***Portuguese (Brazil)***, ***Russian***, ***Spanish***, ***Spanish (Argentina)***, ***Spanish (Latin)***, ***Spanish (Mexico)***, ***Swedish***, ***Turkish***.

By default not all languages are included in the voicemail/auto attendant prompt files on the system. If the language required is not present, either ***UK English*** or ***US English*** is used. Additional languages can be loaded using IP Office Manager, they cannot be loaded using Basic Edition Web Manager. The languages present by default depend on the type of System SD card used by the system:

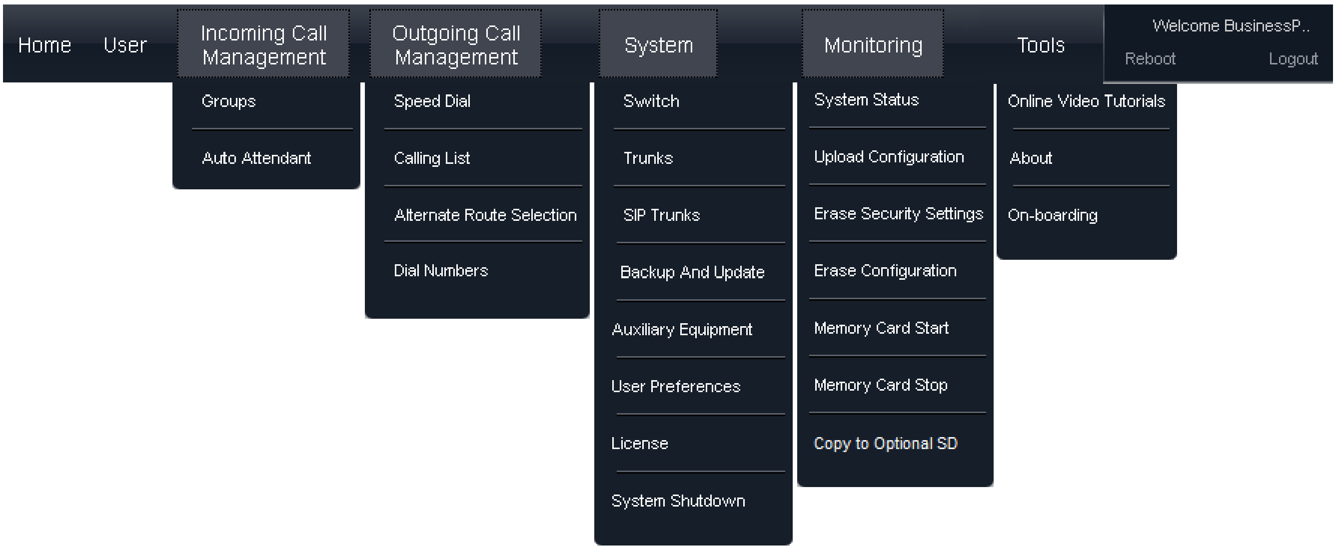
- **IP Office A-Law SD Card:** UK English, French and Spanish.
- **IP Office U-Law SD Card:** US English, Candian French and Latin Spanish.
- **PARTNER SD Card:** UK English, French and Spanish.
- **Norstar SD Card:** UK English, French, Arabic.

Chapter 3. Configuration Menus

3. Configuration Menus

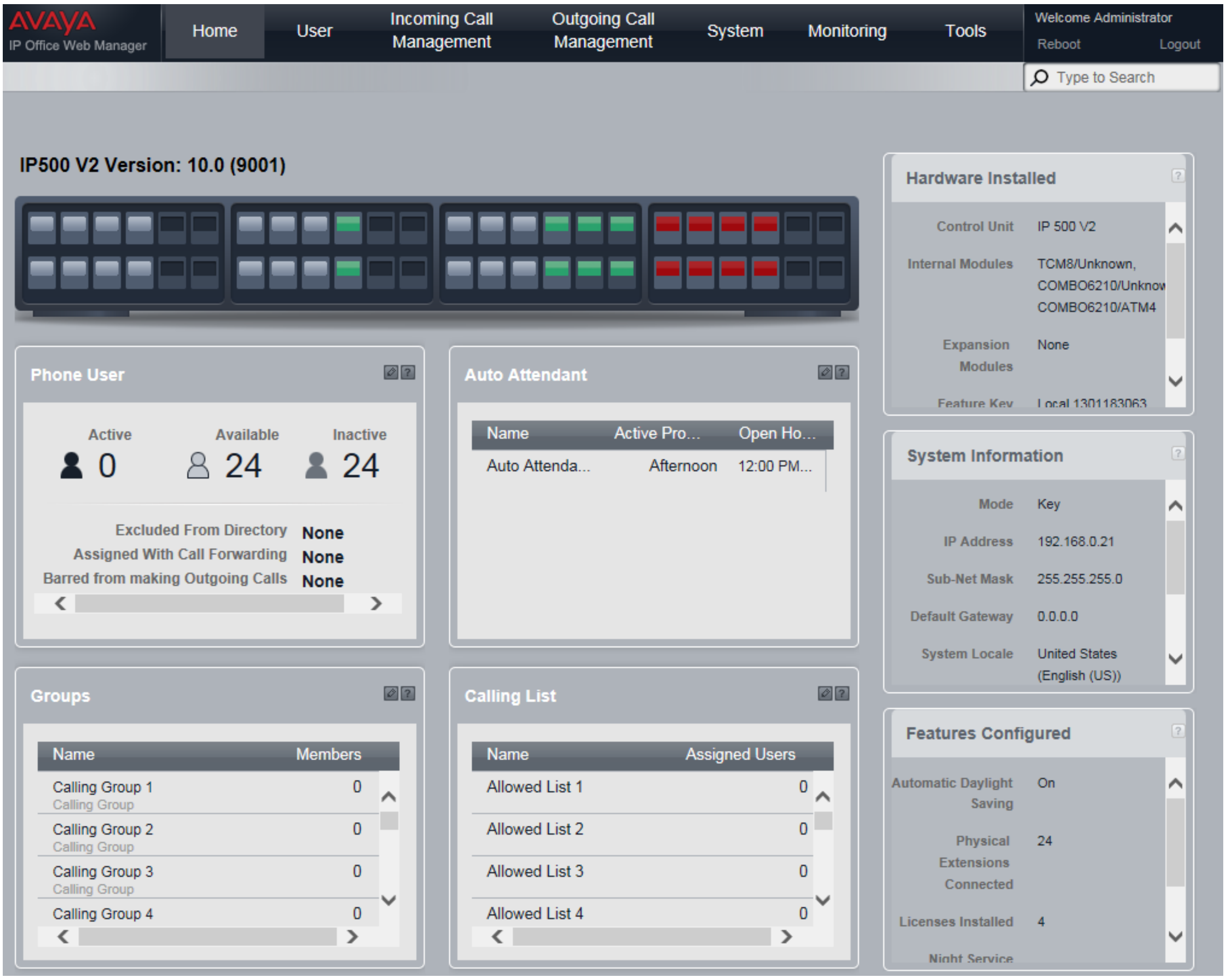
3.1 Configuration Menus

This section provides a summary of the menus accessible from the main menu bar and the options in those menus.



3.2 Home

The home page for Basic Edition Web Manager provides a quick overview of the system and the current system status.



The Dashboard

The dashboard is a representation of the IP Office system, showing the extension and trunks ports. If you hover the cursor over a particular port, a summary windows is shown. For example, for an extension port, the name and extension number of the extension user are shown. Clicking the **Edit** button in the summary will access the appropriate user or trunk settings menu for that port.

Phone User

This panel shows a summary of the installed user extensions. The edit icon can be used to access the Users 33 menu.

Auto Attendant

This panel gives a summary of the auto attendant services (up to 9) currently configured. For each configured auto attendant, the current service being provided by the auto attendant is shown and the hours for that service. Each auto attendant can be configured with different greeting and options for morning, afternoon, evening and out of hours periods.

The edit icon can be used to access the Auto Attendants 48 menu.

Groups

This panel shows a summary of which hunt groups have been configured. The edit icon can be used to access the Groups 46 menu.

- ***Calling Group*** 167
  This type of group can be set as the **Coverage Destination** of a trunk or trunk channel. It can be also the destination of calls routed using DID call mapping or call-by-call settings. Calling Group 1 is also used by the **Simultaneous Page** function (***70**).
- ***Hunt Group*** 167
  This type of group can be set as the **Coverage Destination** of a trunk or trunk channel. It can also be the destination of calls routed using DID call mapping or call-by-call settings.
- ***Night Service Group*** 167
  When the system is put into night service, this group overrides the **Coverage Destination** of all trunks.
- ***Operator Group*** 167
  This option is only available for systems with their **Mode** set to ***PBX***. By default the group contains the first extension on the system. For PRI and BRI trunks, it is fixed incoming destination for calls unless DID Mapping is applied to the call. It can also be selected as the destination for incoming SIP calls.
- ***Pickup Group*** 167
  Users can be pickup calls currently alerting any member of a pickup group. They do not need to be a member of the group.

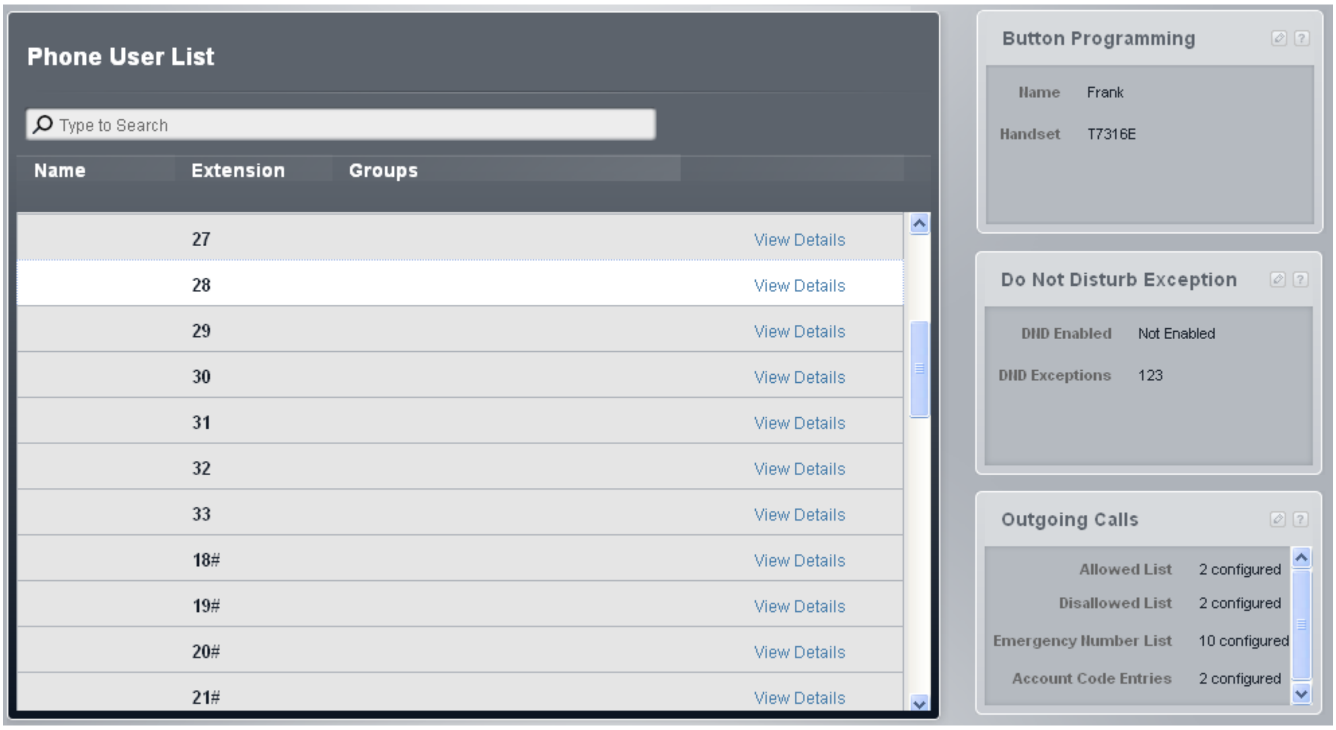
Calling List

This panel shows a summary of the lists that are used to control which numbers users can dial when making outgoing calls. The edit icon can be used to access the List Management 55 menu to edit the settings.

- **Allowed List**
  Allowed lists are used to enter numbers or types of numbers that users associated with the list can dial even if they are restricted from dialing other numbers. Up to 8 such lists, each containing 10 numbers, can be configured.
- **Disallowed List**
  Disallowed lists are used to enter numbers or types of numbers that users associated with the list cannot dial. Up to 10 such lists can be configured. Up to 8 such lists, each containing 10 numbers, can be configured.
- **Emergency Number List**
  This list is used to enter numbers that all users can dial at any time regardless of any other settings that might restrict them from dialing numbers for outgoing calls. Up to 10 numbers can be configured in this list.
- **Account Codes**
  Up to 99 account codes can be entered. In addition selected users can be configured to have to enter an account code whenever they make an outgoing external call.

3.3 User

This menu displays a list of all the phone user extensions supported by the system.



Phone User List

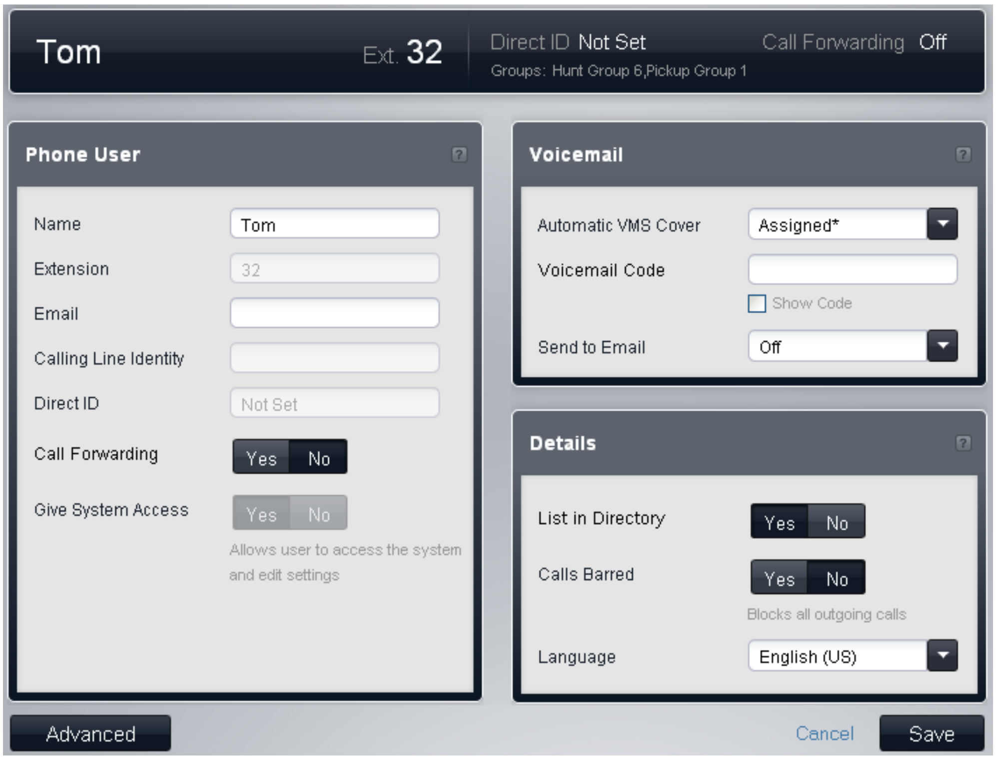
This table lists all the extension users on the system. For any user, the **View Details** button can be clicked to access the user's Details.

The list includes phantom extensions, that is user extensions that do not have a matching physical extension port. These are indicated by a # after the extension number. The system automatically creates a user entry for all users that it could support, regardless of the number of actual extensions.

- **Name**
  The extension user name.
- **Extension**
  The extension number. A # indicates a phantom extension.
- **Groups**
  The groups to which the user belongs. These can be edited through the Groups menu.

3.3.1 Details

This menu is accessed by clicking on **View Details** in the **Phone User List**. If shows specific settings for an individual telephone user.



Phone User

The following settings are shown in this panel:

- **Name:** *Default = Blank.*
  Use this field to enter the extension user's full name. The recommended format is *<first name><space><last name>*. When set, the **Name** is used for display by phones during calls and within these menus, otherwise ***ExtnXXX*** is shown. Only alphanumeric characters and spaces are supported in this field. Do not use punctuation characters such as #, ?, /, -, _ , ^, > and ,. The entry in this field should not start with a space or number. The name is also used by the auto attendant **Dial by Name** function.

- **Extension:** *Information field, not editable.*
  This is the extension number of the user.

- **Email:** *Default = Blank.*
  When the user has a new message they can be emailed with an alert or a copy of the message. This is called voicemail email. Use this field to enter their email address in the format ***name@domain***. This option requires the system to have been configured with SMTP server settings.

- **Calling Line Identity:** *Default = Blank.*
  This setting in only available on ***PBX*** mode systems. Where supported by the line provider, the value is sent on outgoing calls. This setting is not used with analog or SIP trunks.

    - Changing the calling party number may not be supported by the line provider or may be an additional chargeable service. It will also be subject to restrictions on what numbers can be used. It is normally a requirement that the calling party number used must be a valid number for return calls to the same trunk. Use of an invalid number may cause the call to be dropped or the number to be replaced by a default value.

- **Call Forwarding:** *Default = No.*
  If selected, the user is able to forward calls to external numbers.

- **Give System Access:** *Default = No.*
  This setting can only be changed when logged in using the ***BusinessPartner*** account. It is greyed out for other accounts.

    - If changed to ***Yes***, a Basic Edition Web Manager service user account for the user is automatically created in the **Service Users** 116 menu. The name used for the user's account is ***ExtnXX*** where *XX* is the extension number and the default password is ***password***. These can be changed through the **Service Users** 116 menu if necessary.

    - If the control is already set to ***Yes***, it indicated that a service user account for Basic Edition Web Manager and associated with the extension already exists.

- Changing the setting to ***No*** will delete the associated service user account.

Voicemail
The following settings are shown in this panel:

- **Automatic VMS Cover:** *Default = Assigned.*
  If ***Assigned***, voicemail is used to answer calls to the user that have rung for the **VMS Cover Ring** time. This setting is ignored for any extension configured as a loudspeaker paging extension.
- **Voicemail Code:** *Default = Blank. Range = Blank or 1 to 15 digits.*
  This code is used to control access to the user's mailbox to collect messages. The mailbox user can change the code themselves after they enter their mailbox.
- **Send to Email:** *Default = Off.*
  This setting is used if an email address for the user has been set above and the system is configured with voicemail email operation. It sets whether the user receives an email when they have a new voicemail message.
  - ***Off***
    Switches off the use of email for new message alerts.
  - ***Copy***
    Send an email to the user's email address with the voicemail message attached. The method leaves the original message in the user's voicemail mailbox.
  - ***Forward***
    Send an email to the user's email address with the voicemail message attached. This method deletes the original message from the user's voicemail mailbox
  - ***Alert***
    Send an email alert about the new message but do not attach the message to the email.

Details
The following settings are shown in this panel:

- **List in Directory:** *Default = Off*
  If selected, the user is included in the directory of users displayed on phones.
- **Calls Barred:** *Default = Off.*
  If selected, the extension user cannot make any outgoing external calls except to numbers in the **Emergency Numbers List** and any **Allowed Lists** of which they are a member.
- **Language:**
  The language entered here will affect the language of prompts displayed on the user's extension and the prompts played to the user when they access their voicemail mailbox. Possible languages are:
  - ***Arabic***, ***Cantonese***, ***Danish***, ***Dutch***, ***English (UK)***, ***English (US)***, ***Finnish***, ***French***, ***French (Canada)***, ***German***, ***Italian***, ***Korean***, ***Mandarin***, ***Norwegian***, ***Polish***, ***Portuguese***, ***Portuguese (Brazil)***, ***Russian***, ***Spanish***, ***Spanish (Argentina)***, ***Spanish (Latin)***, ***Spanish (Mexico)***, ***Swedish***, ***Turkish***.

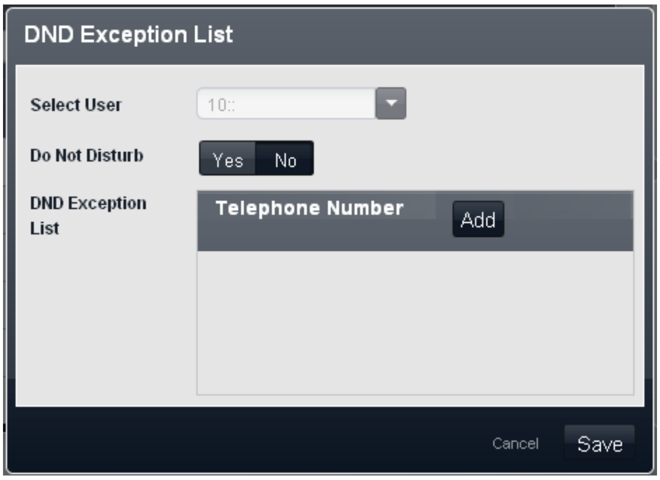
3.3.2 DND Exception List

This menu allows you to see and change a user's current do not disturb status and to edit their do not disturb exception numbers. Users can switch do not disturb on/off using a programmed key on their phone or an option in their phone's menus.

Do not disturb prevents the user from receiving calls.

- Only calls from numbers in the user's Do Not Disturb Exceptions list are treated normally.
- The user is not included in hunt group and page calls.
- Calls direct to the user's extension number hear busy tone or are diverted to voicemail if enabled.
- All the users call forwarding, follow me and call coverage settings are ignored.

This menu is accessed by selecting **User** in the menu bar, selecting a user and clicking **View Details** and then clicking on edit icon in the **Do Not Disturb Exceptions** panel.



- **Select User**
  Select the user whose current do not disturb settings are displayed.
- **Do Not Disturb:** *Default = Off*
  When checked the user's extension is considered busy, except for calls coming from sources listed in their Do Not Disturb Exception List. When a user has do not disturb in use, their normal extension will give alternate dial tone when off hook. Users with DND on are indicated as 'busy' on any BLF indicators set to that user.
- **DND Exception List:** *Default = Blank.*
  This is the list of telephone numbers that are still allowed when the user has do not disturb enabled. For example this could be an assistant or an expected phone call. Internal extension numbers or external telephone numbers can be entered. If you wish to add a range of numbers, you can either enter each number separately or make use of the wildcards ***N*** (single digit) and ***X*** (multiple digits)in the number. For example, to allow all numbers from 7325551000 to 7325551099, the DND Exception number can be entered as either ***73255510XX*** or ***73255510N***. Note that this list is only applied to direct calls to the user.
  - Calls to a hunt group of which the user is a member do not use the Do Not Disturb Exceptions list.

3.3.3 Button Programming

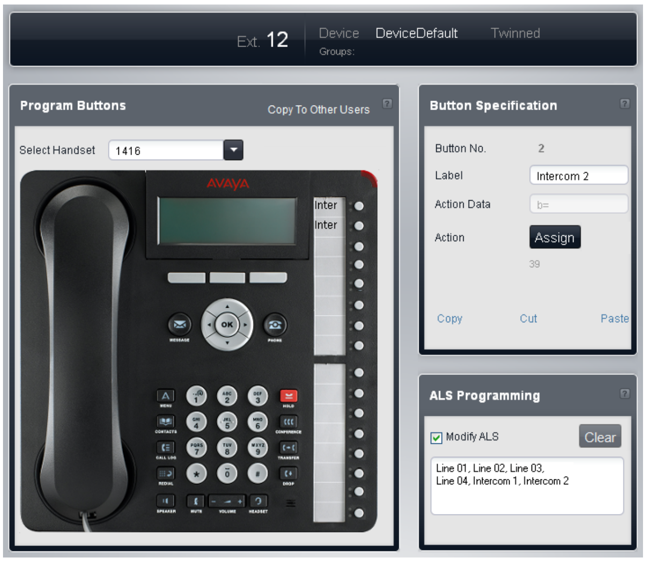
Most Avaya phones have buttons to which functions can be assigned. For some phones, additional buttons can also be added by attaching a button module to the phone.

Default Buttons and Button Numbering

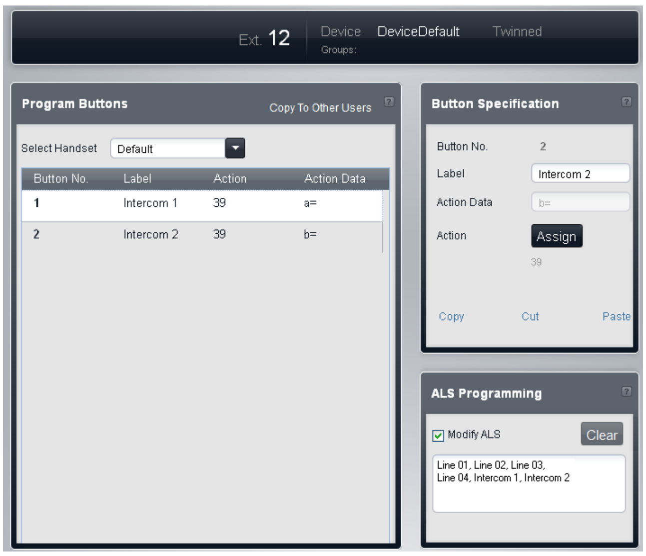
The default button assignment depends on whether the system's **Mode** 65 is set as ***Key*** or ***PBX***.

- **Key Mode**
  - **01-02: Intercom Buttons**
    The first two buttons are used as **Intercom 1** and **Intercom 2** buttons for internal calls. This function is automatically assigned to the buttons by the system and cannot be overridden by the system administrator or extension user.
  - **03+: Line Buttons**
    Buttons 03 and upwards up to the system's **Number of Lines** setting are used as line appearance buttons for external calls. These cannot be overridden by the extension user.
  - **Other Buttons**
    Any additional buttons can be used for additional functions. These buttons can be programmed by the system administrator and, for some functions, the extension user.
  - **Button Numbering**
    All buttons are numbered from 01, from left to right, starting from the bottom row up.
- **PBX Mode**
  - **01-03 (ETR 01-02): Call Appearance Buttons**
    The first three buttons (two only on ETR phones) are used call appearance buttons for making and answering calls. They can be used for both internal and external calls. This function is automatically assigned to the buttons by the system and cannot be overridden by the system administrator or extension user.
  - **Other Buttons**
    Any additional buttons can be used for additional functions. These buttons can be programmed by the system administrator and, for some functions, the extension user.
  - **Button Numbering**
    All buttons are numbered from 01, from left to right, starting from the bottom row up. However for 1400, 9400 and 9500 Series telephones, buttons are numbered from 01 from left to right, starting from the top row downwards.

The menu can operate in either of two ways, depending on whether the phone type is known or not. If the phone type is known, for 1400 Series phones a picture of the phone is used. If the phone type is not known and for non-1400 Series phones, a list of buttons is used.



Button programming menu in picture mode.



Button programming menu in list mode.

Program Buttons

This table displays the list of features programmed on each of the user's buttons.

- **Select Handset**
  If the type of telephone connected to the user's extension is known, it is shown here and the menu displays a picture of the phone buttons. If the telephone type is not known, the field display ***Default*** and a list of the current button settings. You can use the selector to choose whether you want to program the buttons using the picture or list mode.

- **Button No**
  The button to which the feature is programmed. The position of the button will vary depending on the type of phone.

- **Label**
  If the phone displays text labels next to each button, you can enter the text that should be displayed. To enter the label, click on the label space after having selected the action for the button.

- **Action:**
  This is the action performed by the button when pressed. To select the action place your cursor in the box, right click and select **Assign a Feature** from the drop menu. This will display a menu from which you can select the feature required.

- **Action Data**
  For some actions, when selecting the action you are asked to enter action data.

- **Copy to Other Users**
  Displays a list of users and allows you to select which the users to which you want the current users buttons copied.

Button Specification

This panel displays the setting of the currently selected button. To change the currently selected button, click on a button on the telephone picture or click on a row in the list of buttons.

To change the buttons action, click on the **Assign Feature** button. For a summary of the possible features, see **Programming Features** 33, **System Features** 33 and **Line Assignment** 33.

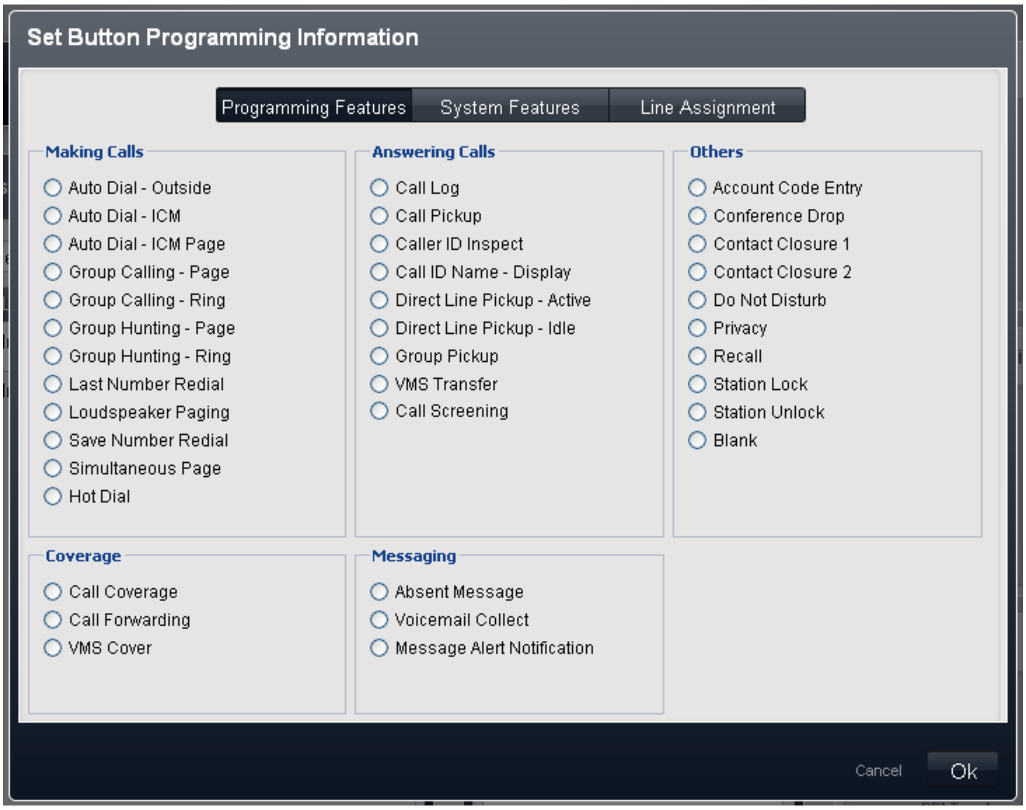
ALS Programming

Automatic line selection is used to select which available line is used when the extension goes off hook to make a call without the user first pressing a specific line or intercom button, for example if the user just lifts the handset or presses the speaker button. By default all analog line buttons (lowest to highest) and the two intercom buttons are used in that order.

- **Modify ALS Programming:** *Default = Off.*
  If **Modify ALS Programming** is selected, the order of line selection is displayed and can be edited.

3.3.3.1 Programming Features

This menu allows a range of individual functions to be assigned to the button.



Making Calls

- **Auto Dial - Outside** 221: *Action Data = Telephone number to dial.*
  A button set to this feature dials the stored number using the first available line appearance in the user's automatic line selection setting.

- **Auto Dial - ICM** 220: *Action Data = User extension number.*
  A button set to this function can be used to make an intercom call to the configured extension. It will also indicate when that user is idle or active.

- **Auto Dial - ICM Page** 220: *Action Data =User extension number.*
  A button set to this function can be used to page the configured extension.

- **Group Calling - Page** 223: *Action Data = Calling group 1 to 4*
  A button set to this function can be used to make a page call to the available members of the configured calling group.

- **Group Calling - Ring** 223: *Action Data = Calling group 1 to 4.*
  A button set to this function can be used to make a call to the available members of the configured calling group.

- **Group Hunting - Page** 227: *Action Data = Hunt group 1 to 6.*
  A button set to this function can be used to make a page call to the available members of the configured hunt group.

- **Group Hunting - Ring** 227: *Action Data = Hunt group 1 to 6.*
  A button set to this function can be used to make a call to the available members of the configured hunt group.

- **Last Number Redial** 227: *Action Data = None.*
  A button set to this function redials the last outgoing external number dialed by the user.

- **Loudspeaker Paging** 227: *Action Data = None*
  A button set to this functions makes a page call to the system's designated loudspeaker extension port.

- **Save Number Redial** 228: *Action Data = None.*
  A button set to this function allows the user to save the number dialed during a call and to redial that number when idle. This can be used when the number dialed does not answer.

- **Simultaneous Page** 228: *Action Data = None.*
  A button set to this function allows the user to make a page call to both the loudspeaker extension and the extensions in first calling group, 71.

- **Hot Dial** 226: *Action Data = None.*
  A button set to this function allows the user to turns hot dialing on or off. When on, the extension user is able to begin dialing without going off-hook. For ETR extensions hot dial is off by default. For DS and TCM digital stations, hot dial is on by default and cannot be changed.

Answering Calls

- **Call Log** 223: *Action Data = None.*
  A button set to this function allows the user to access the system call log. The user must also be one of the 3 extensions configured for **Log All Caller ID Calls for Users**.

- **Call Pickup** 223: *Action Data = Extension number.*
  A button set to this function performs a call pickup from the target extension. If the target has parked calls, a parked call is retrieved in preference to any ring call at the target. Extension users can park calls by transferring the call their own extension number. Parked calls will recall after 3 minutes.

- **Caller ID Inspect** 223: *Action Data = None.*
  When off hook on a call, pressing this button allows the user to then press another active line appearance or intercom button to view caller number information for that call.

- **Call ID Name - Display** 223: *Action Data = None.*
  On some phones, after the call is answered the call display is not able to show both the caller ID name and number. This function allows the user on such phones to toggle between the name and the number. If the user has this feature enabled, removing this button will turn the feature off.

- **Call Screening** 224: *Action Data = None.*
  A button set to this function is used to enable or disable call screening. While enabled, when a caller is presented to the user's voicemail mailbox, if the user's phone is idle they will hear through the phone's handsfree speaker the caller leaving the message and can select to answer or ignore the call.

- **Direct Line Pickup - Active** 220: *Action Data = None.*
  A button set to this function allows the user to pickup a ringing, held or connected call on the specified line. Users can also dial intercom ***68LL*** where ***LL*** is the line number.

- **Direct Line Pickup - Idle** 227: *Action Data = None.*
  A button set to this function allows the user to seize and make a call using the specified line if that line is idle. Users can also dial intercom **8*LL*** where ***LL*** is the line number.

- **Group Pickup** 228: *Action Data = Pickup Group number 1 to 4.*
  A button set to this function allows the user to pickup the longest ringing call at the specified group.

- **VMS Transfer** 229: *Action Data = None.*
  A button set to this function allows the user to transfer a call directly into the voicemail mailbox of another user.

Other

- **Account Code Entry** 220: *Action Data = None.*
  A button set to this feature allows the user to enter a voluntary account code to be associated with the current call or with the call made after entry of the account code. Not supported by POTS phones.

- **Conference Drop** 226: *Action Data = None.*
  A button set to this function acts as a call drop button. On Avaya digital stations, a list of conference parties is displayed from which the user can select which call to drop. On ETR phones, the last added external party is dropped.

- **Contact Closure 1** 226/**Contact Closure 2** 226: *Action Data = None.*
  A button set to this function Allows the user to activate the phone system's contact closure 1 or contact closure 2 switch. The user must also be a member of the appropriate **Contact Closure Group**. While the contact is on, the button lamp is green at the user's extension and red at any other users configured for the same contact closure. The duration and type of closure is configured in the **Contact Closure Group** settings.

- **Do Not Disturb** 226: *Action Data = None.*
  A button set to this function allows the user to redirect all call to them while still being able to make calls. Incoming calls follow voicemail coverage if on, else they receive busy. Do not disturb overrides call forwarding. If the user has this feature enabled, removing this button will turn the feature off.

- **Privacy** 228: *Action Data = None.*
  A button set to this function allows the user to switch call privacy on or off during a call. When on, other users with line appearances for the same line are not able to join the call using that button. If the user has this feature enabled, removing this button will turn the feature off.

- **Recall** 228: *Action Data = None.*
  A button set to this function allows the user to send a recall or hook flash signal.

- **Station Lock** 229: *Action Data = None.*
  A button set to this function allows the user to lock their extension by entering a 4 digit code. When locked, the extension can only be used to make emergency calls and dial marked speed dials. To unlock the phone the same 4 digit code must be used.

- **Station Unlock** 229: *Action Data = None.*
A button set to this function allows the system administrator extensions (the first two extensions in the system) to unlock any extension without knowing the 4 digit code that was used to lock the extension.

- **911-View/Emergency View** 220: *Action Data = None*
A button set to this function indicates when a call has been made using a number in the system's Emergency Numbers 58 list. Pressing the button displays a list of such calls.

- **Blank**
When selected, this option removes all programming from the button.

Coverage

- **Call Coverage** 222: *Action Data = XX-YY where if XX is the source extension and YY is the destination extension.*
A button set to this function allows the user to turn call coverage on or off. If the user has this feature enabled, removing this button will turn the feature off.

- **Call Forwarding** 222: *Action Data = XX-YY where if XX is the source extension and YY is the destination extension.*
A button set to this function allows the user to turn call forwarding on or off. If the user has this feature enabled, removing this button will turn the feature off.

- **VMS Cover** 229: *Action Data = None.*
A button set to this function allows the user to turn voicemail coverage of their calls on or off.

Messaging

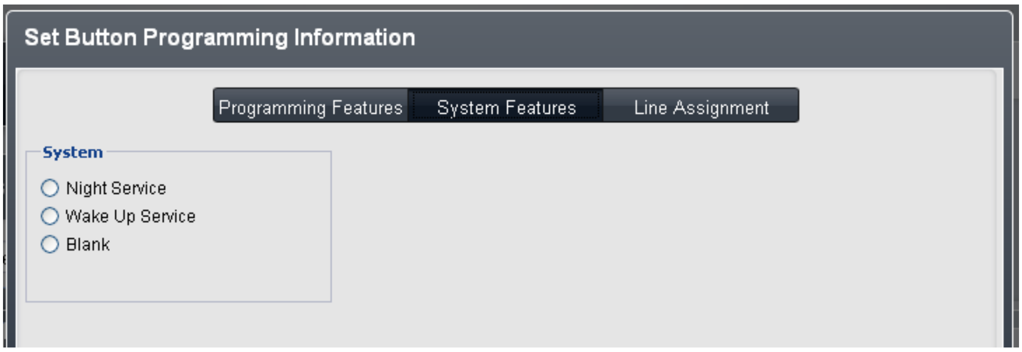
- **Absent Text** 220: *Action Data = None*
A button set to this function allows the user to set or clear an absence text message. When set, the message is displayed on their extension and also on other extensions when they call the user. If the user has this feature enabled, removing this button will turn the feature off.

- **Voicemail Collect** 227: *Action Data = None.*
A button set to this function allows the user to access the voicemail to collect messages.

- **Message Alert Notification** 227: Action Data =
A button set to this function allows the user to inspect the current state of another user's message waiting lamp. It can only be used in conjunction with other users for which this user has **Auto Dial - ICM** buttons configured.

3.3.3.2 System Features

This tab and its button functions are only for the first extension in the system. These features are linked to the usage of the **System Password** as they affect the operation of the phone system for all users and trunks.



- **Night Service:** *Action Data = None.*
A button set to this function allows the user to switch night service on or off. The **System Password**, if set, is required to use this feature. If the user has this feature enabled, removing this button will turn the feature off.

- **Wake Up Service:** *Action Data = None.*
It allows the user to set an alarm call to occur another extension in the next 24-hours. When the alarm occurs, if the call is answered the targeted user will hear music on hold if available, otherwise they will hear repeated double tones. If the call is not answered another attempt is made 5 minutes later, however only 2 attempts are made. Only one alarm can be set against each user at any time. Setting another alarm will override any existing alarm.

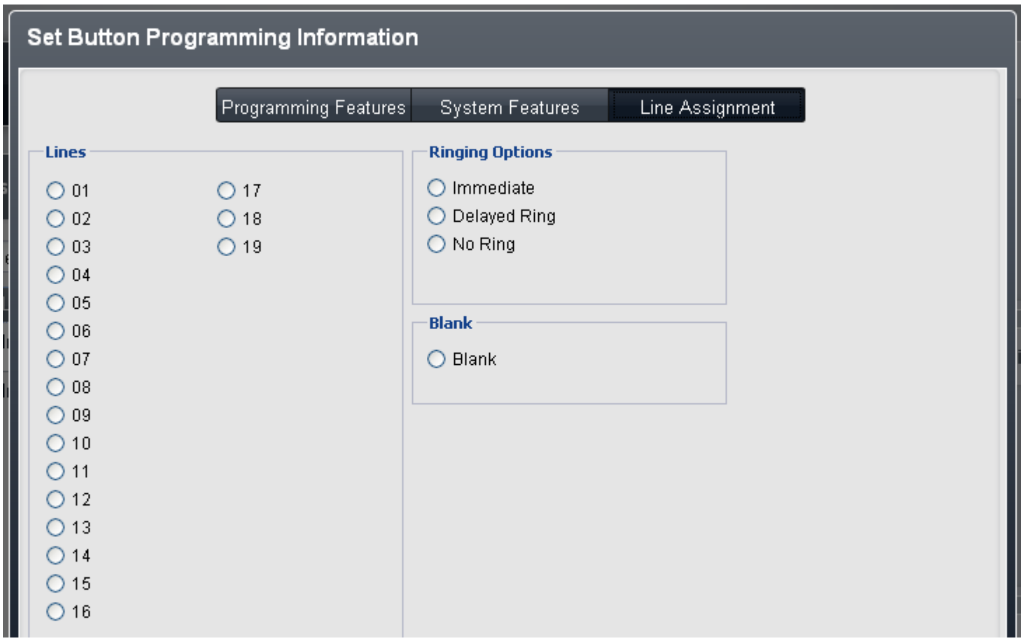
- **Blank**
When selected, this option removes all programming from the button.

3.3.3.3 Line Assignment

This menu enables you customize lines by setting the programmable button as a line appearance button to make and answer calls on a particular line.

For systems operating in ***PBX*** mode, buttons can also be selected for ARS selector 59 group numbers. Those can be used to make calls but not to receive calls. When pressed, an available line in the ARS selector group is seized.

- Note that for systems running in ***Key*** mode, a number of each users programmable buttons are automatically configured as line appearance buttons according to the system **Number of Lines** setting. If the system setting **Number of Lines** is changed, it may overwrite all or some of the current button programming.



- **Lines**
  Select the line with which the button will be associated. For systems operating in ***Key*** mode, the ARS Selector group numbers are also listed.

- **Ringing Options:**
  Select whether the phone should provide audible alerting when a call is waiting to be answered on the line. Not used for buttons assigned to ARS Selectors.

  - ***Immediate***
    Provide audible alerting as normal.

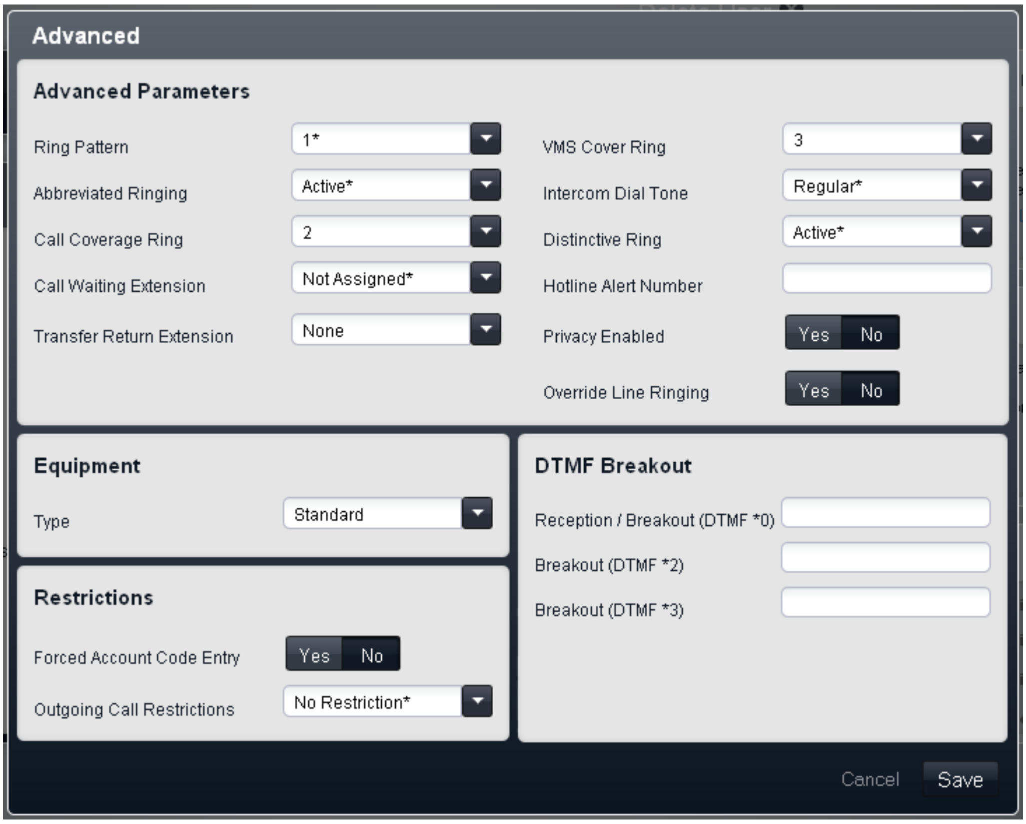
  - ***Delayed Ring***
    Only provide audible alerting after approximately 15 seconds.

  - ***No Ring***
    Do not provide any audible alerting.

- **Blank**
  When selected, this option removes all programming from the button.

3.3.4 Advanced

This menu shows additional user settings. The menu is accessed by clicking on the Advanced button in the user **Details** menu.



Advanced Parameters

The following settings are shown in this panel:

- **Ring Pattern:** *Default = 1.*
  Selects the ring pattern that should be used for the call when alerting on a user extension. The available patterns depend on the phone type.

- **Abbreviated Ringing:** *Default = Active.*
  When active on an ETR or a Avaya digital station, if a user is already connected to a call, any additional call will give just a single quiet ring. Note that for additional calls alerting on line appearance buttons, the ***Immediate***, ***Delayed Ring*** or ***No Ring*** settings of the button still apply.

- **Call Coverage Ring:** *Default = 2 (10 seconds).*
  Programmable buttons set to **Call Coverage** can be used to switch call coverage on or off for a user. When on, calls that ring unanswered for this number of rings are redirected to alert on a covering extension. Ensure that this setting is set lower than the users **VMS Cover Ring** if using **Automatic VMS Cover**.

- **Call Waiting Extension:** *Default = Not Assigned.*
  If **Assigned**, on an analog extension, when the user is on a call, an additional call will cause a tone to be heard as part of the existing call.

- **Transfer Return Extension:** *Default = None.*
  Set the destination for transferred calls that ring unanswered for longer than the **Transfer Return Ring** [43] setting. Note that if a door phone or paging extension is selected, the call will continue ringing at the transfer destination rather than returning.

- **VMS Cover Ring:** *Default = 3 (15 seconds). Range = 0 to 9.*
  For a user with **Automatic VMS Cover** enabled, this value sets how long a call alerts the user's extension before it is redirected to voicemail.

  - The option ***0*** for immediate voicemail is available. 0 is the only value usable for phantom extensions. If selected it has the following effects.

    - For a call that would have otherwise have alerted at the extension, the call now goes immediately to voicemail.

    - If the extension has call forwarding set, the forwarded call will continue ringing at the forwarding target rather than going to voicemail.

- If the extension is the target for another extension's call forwarding, the call will go immediately to the forwarding extension's voicemail.

- **Intercom Dial Tone:** *Default = Regular.*
  This setting allows selection of which dial tone is used for intercom (internal) calls. ***Regular*** matches the dial tone used by the phone system. ***Machine*** matches the normal CO dial tone.

- **Distinctive Ringing:** *Default = Active.*
  This setting is used for analog extensions only. If active, the phone will use, if supported, different ring patterns to indicate internal, external and recall calls.

- **Hotline Alert Number:** *Default = Blank.*
  If a number is entered here, when the extension goes off-hook by simply lifting the handset or pressing a speaker button (rather than first selecting a line or intercom button), this number is called.

- **Privacy Enabled:** *Default = Off.*
  If off, when connected to an external call on a particular line, other users with a line appearance for that line are able to join that call. If on, other user cannot join calls. A user can switch privacy on/off using a programmable button set to the **Privacy** feature.

- **Override Line Ringing:** *Default = Off.*
  For each line, a ring pattern setting can be applied to be used with incoming calls. They are overridden if the user's **Override Line Ringing** setting is enabled. BST phones always override line ringing regardless of this setting.

DTMF Breakout

These numbers are used to allow caller's to select to be transferred to another extension instead of leaving a message.

- **Reception / Breakout (DTMF 0):** *Default = Blank.*
  Sets the internal number to which a caller is transferred if they press ***0*** whilst listening to the mailbox greeting.

- **Breakout (DTMF 2):** *Default = Blank.*
  Sets the internal number to which a caller is transferred if they press ***2*** whilst listening to the mailbox greeting.

- **Breakout (DTMF 3):** *Default = Blank.*
  Sets the internal number to which a caller is transferred if they press ***3*** whilst listening to the mailbox greeting.

Equipment

This panel shows the type of telephone device connected to the user's extension port.

  - **! WARNING - Reboot Required**
    Changing this setting requires the system to be rebooted 12 for the change to take effect. Rebooting the system will end all calls currently in progress.

- **Type:**
  The possible types of extension device are:

  - ***Loudspeaker Paging***
    Select this option for an extension connected to a paging amplifier. Only one extension of this type is supported on the system.

  - ***Door Phone 1 / Door Phone 2***
    Select this option for an extension connected to a door phone. The system can support two such devices.

  - ***Fax Machine***
    Select this option for an extension connected to a fax machine.

  - ***Standard***
    Select this option for a standard telephone extension.

  - ***Phantom***
    This option is automatically selected for users who do not have a matching physical extension. Phantom users can still be used for a range of functions such as voicemail. The setting cannot be changed except by installing the appropriate extension hardware.

Restrictions

- **Forced Account Code Entry:** *Default = Off.*
  For each user, if this setting is selected, that user is required to enter an account code from the **Account Codes** list when making an external call. This can only be overridden by use of the **Password** 65 to make a call.

- **Outgoing Call Restrictions:** *Default = No Restriction.*
  For each user, this field sets the type of outgoing external calls that the user can normally make. Any restrictions applied do not apply to numbers in the **Emergency Numbers List** and to numbers in any **Allowed Lists** of which the user is a member.

  - ***No Restrictions***
    The user can make outgoing external calls. The **Allowed Lists** and **Disallowed Lists** of which the user is a member still apply.

  - ***Inside only***
    The user can only make internal calls.

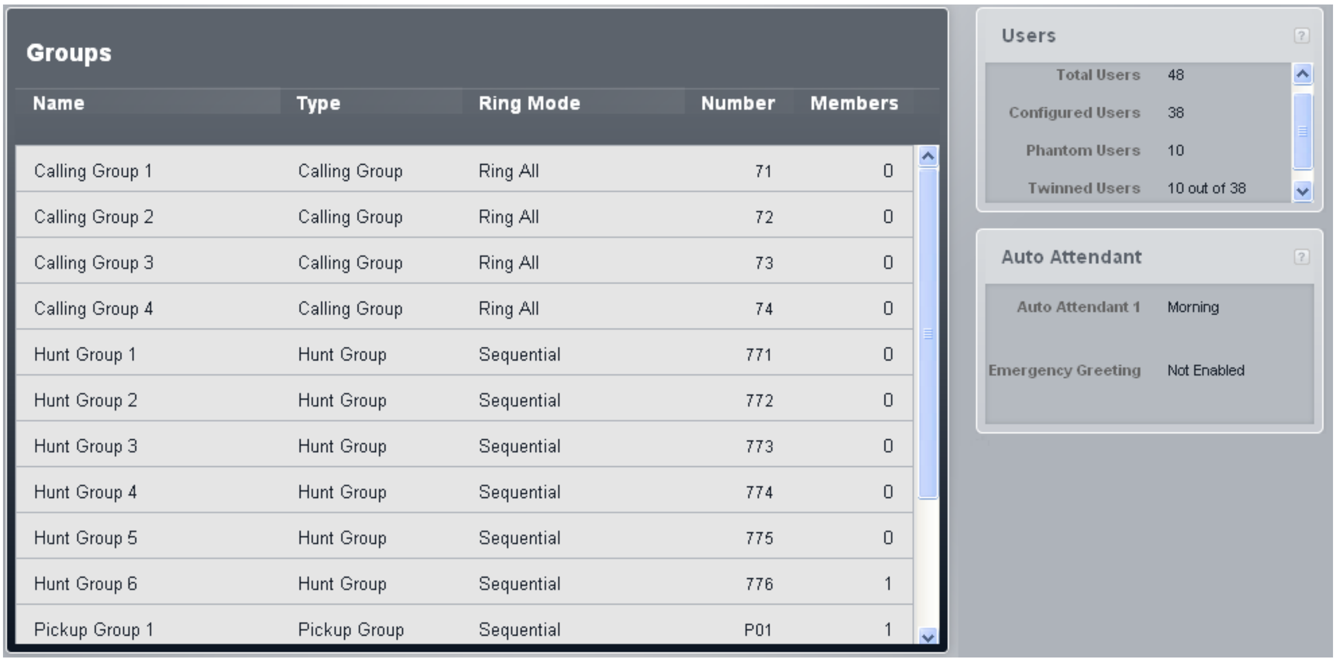
  - ***Local only***
    The user can only make outgoing external calls to numbers matching local numbers.

3.4 Incoming Call Management

3.4.1 Groups

A group is a collection of users used for a special purpose. For example, a group can be used as the destination for incoming calls on an external trunk.

The system has a fixed set of groups. You cannot add or remove groups. However you can edit the the user membership of all the groups. A user can be a member of several or even all groups.



Groups

- **Name**
  The names of the groups are fixed.

- **Type**
  This indicates the main purpose of the group.

  - ***Calling Group*** 167
    This type of group can be set as the **Coverage Destination** of a trunk or trunk channel. It can be also the destination of calls routed using DID call mapping or call-by-call settings. Calling Group 1 is also used by the **Simultaneous Page** function (****70***).

  - ***Hunt Group*** 167
    This type of group can be set as the **Coverage Destination** of a trunk or trunk channel. It can also be the destination of calls routed using DID call mapping or call-by-call settings.

  - ***Night Service Group*** 167
    When the system is put into night service, this group overrides the **Coverage Destination** of all trunks.

  - ***Operator Group*** 167
    This option is only available for systems with their **Mode** set to ***PBX***. By default the group contains the first extension on the system. For PRI and BRI trunks, it is fixed incoming destination for calls unless DID Mapping is applied to the call. It can also be selected as the destination for incoming SIP calls.

  - ***Pickup Group*** 167
    Users can be pickup calls currently alerting any member of a pickup group. They do not need to be a member of the group.

- **Ring Mode**
  This settings indicates the order in which the idle group members are alerted when there is a call to the group. For full details see Group Call Distribution 169.

  - ***Ring all***
    This type of group alerts all the idle groups members at the same time until answered.

  - ***Rotary***
    This type of group alerts each idle group member in extension number order for 15 seconds each until answered. It starts with the available member after the last member to answer a group call.

- **Number**
  The use of a number associated with a group depends on the group type:

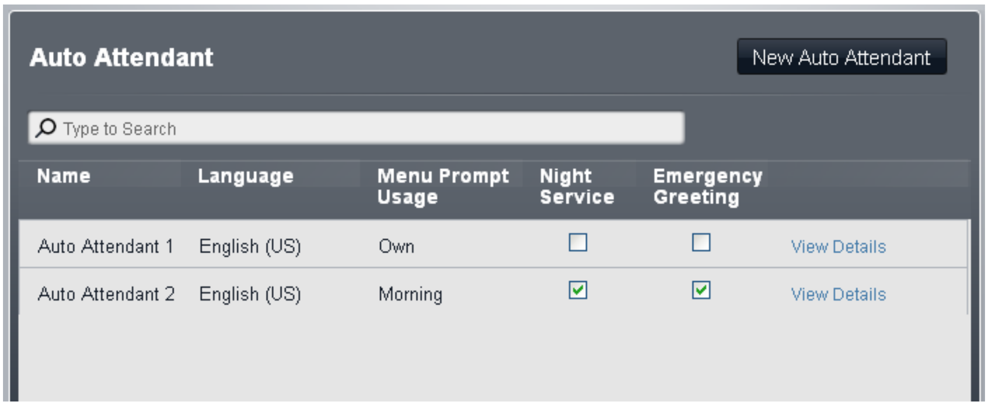
    - Calling, operator and hunt groups can be called internally using their extension number. The groups number can also be used as the target for call transfers. * can be used in front of the group extension number for a page call to the group.

    - Pickup groups numbers are used to answer a call currently alerting any member of the group. For example, to pickup a call alerting members of pickup group 1, dial 661.

- **Members**
  This indicates the number of users who are currently set as members of the group.

3.4.2 Auto Attendants

This menu is accessed by selecting **Incoming Call Management** in the menu bar and clicking **Auto Attendant**. The menu displays a list of the auto attendants that have been configured for use. It can be used to adjust their settings and to add additional auto attendants.



Auto Attendant

The settings shown for each auto attendant are shown below. These can be changed by double-clicking on the auto attendant.

- **Language:** *Default = Match system language.*
  This settings controls the language used for auto attendant action prompts. The options are ***Arabic***, ***Brazilian Portuguese***, ***Canadian French***, ***Cantonese***, ***Danish***, ***Dutch***, ***Finnish***, ***French***, ***German***, ***Italian***, ***Korean***, ***Mandarin***, ***Norwegian***, ***Portuguese***, ***Russian***, ***Spanish***, ***Spanish (Argentinian)***, ***Spanish (Latin)***, ***Spanish (Mexican)***, ***Swedish***, ***Taiwanese***, ***UK English***, ***US English***.

- **Menu Prompt Usage:** *Default = Each menu uses own prompt.*
  Each time profile option used by an auto attendant can have its own set of actions and therefore may require a separate actions prompt to be played after the appropriate greeting prompt. The settings ***Each menu uses own prompt*** does that. Alternately one of the menu options can be selected as the menu option prompt played at all times of day.

- **Night Service:** *Default = On.*
  If selected, when the system is in night service, the auto attendant will switch to using its out of hours greetings and menu actions. If not selected, when the system is in night service, the auto attendant will continue using the greetings and menu options as determined by its own time profile settings.

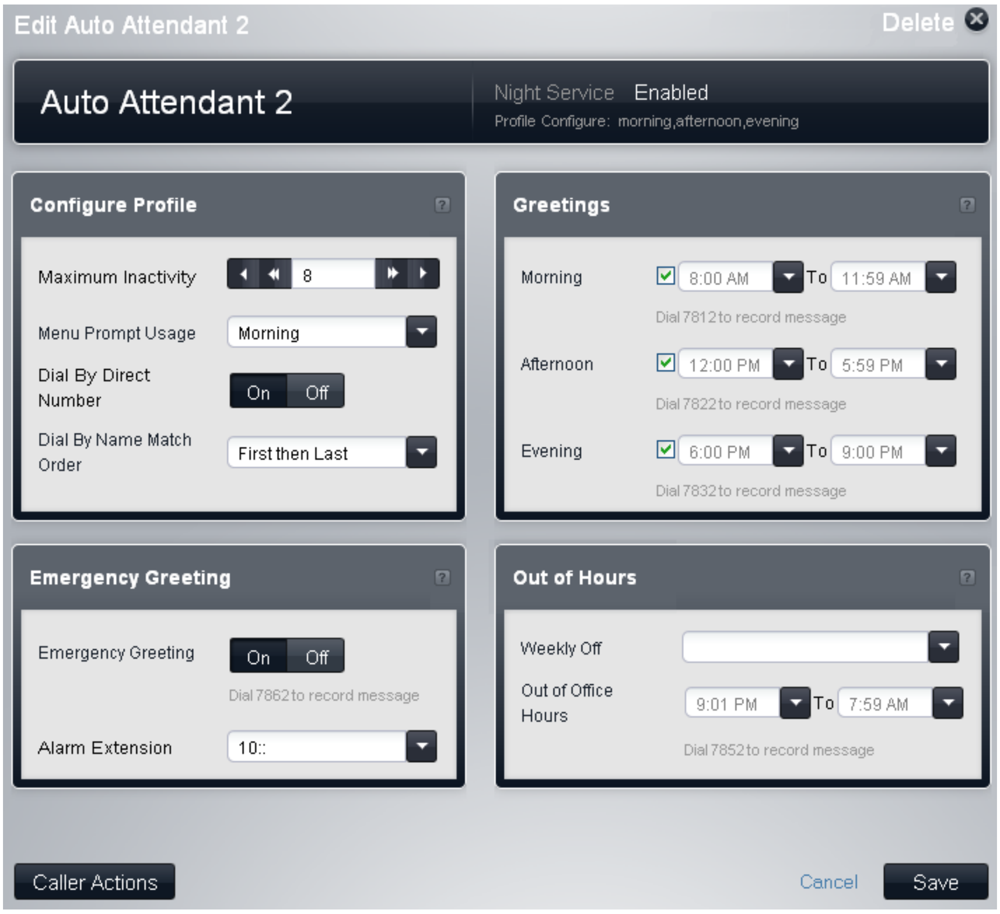
- **Emergency Service:** *Default = Off*.
  This field indicates when the auto attendants emergency greeting option has been enabled by someone using the auto attendant. The field can also be used to disable the emergency greeting without having to go through the auto attendant menu.

Additional settings can be viewed and edited by clicking on the **View Details** link.

3.4.2.1 Auto Attendant Details

This menu is accessed by clicking on the **View Details** link next to an auto attendant in the list of auto attendants 48. It displays the detailed setting for the particular auto attendant.

To return to the list of auto attendants, click on **<< View All Auto Attendants** just below the menu bar. The **Caller Actions** 51 button can be used to display and configure the options used by the auto attendant in response to caller key presses.



3.4.2.1.1 Configure Profile

These are the general settings for the auto attendant.

- **Maximum Inactivity:** *Default = 8 seconds. Range = 1 to 20 seconds.*
  This field sets how long after playing the prompts the auto attendant should wait for a valid key press. If exceeded, the caller is transferred to the operator (the first extension in the system).

- **Menu Prompt Usage:** *Default = Each menu uses own prompt.*
  Each time profile option used by an auto attendant can have its own set of actions and therefore may require a separate actions prompt to be played after the appropriate greeting prompt. The settings ***Each menu uses own prompt*** does that. Alternately one of the menu options can be selected as the menu option prompt played at all times of day.

- **Dial By Direct Number:** *Default = On for the first default auto attendant. Off for other auto attendants.*
  This setting affects the operation of any key presses in the auto attendant menu set to use the ***Dial By Number*** action.

  - If selected, the key press for the action is included in any following digits dialed by the caller for extension matching. For example, if 2 is set in the actions to ***Dial by Number***, a caller can dial 20 for extension 20.

  - If not selected, the key press for the action is not included in any following digits dialed by the caller for extension matching. For example, if 2 is set in the actions to ***Dial by Number***, a caller must dial 2 and then 20 for extension 20.

- **Dial by Name Match Order:** *Default = Last then First.*
  Determines the name order used for keys set to the ***Dial by Name*** action. The options are ***First then Last*** or ***Last then First***.

3.4.2.1.2 Emergency Greeting

Using the ***Transfer to Greeting*** action from an auto attendant, an emergency greeting can be recorded and then enabled or disabled. When enabled, the emergency greeting is played to callers in advanced of any other auto attendant greeting. The internal number to dial to record the greeting is indicated below the setting. Use of this feature requires the system password to be set.

- **Emergency Service:** *Default = Off*.
  This field indicates when the auto attendants emergency greeting option has been enabled by someone using the auto attendant. The field can also be used to disable the emergency greeting without having to go through the auto attendant menu.

- **Alarm Extension:** *Default = 10.*
  When the auto attendant's **Emergency Greeting** option is enabled, in addition to callers to the auto attendant hearing the emergency greeting, a warning message is also displayed on the extension indicated by this field. Typically this would be the extension of someone who is able to disable the emergency greeting when it is no longer required.

3.4.2.1.3 Greetings

The auto attendant can provide different greetings and menu actions at different times of day. These fields are used to set the operating hours for morning, afternoon and evening options.

- If any of the periods overlap, the settings of the first of the overlapping periods are used.
- Outside any of these periods when they are enabled, the out of hours settings of the auto attendant are applied.
- The settings in the **Night Service** panel below also indicate specific days on which the out of hours settings override the morning, afternoon and evening settings.
- If the auto attendant has the **Night Service** 48 option enabled, the attendants out of hours settings are used when the system is put into night service.

- **Morning**
  When selected, the morning greeting and caller actions are used during the times indicated. The internal number to dial to record the greeting for this period is shown below the setting.
- **Afternoon**
  When selected, the afternoon greeting and caller actions are used during the times indicated. The internal number to dial to record the greeting for this period is shown below the setting.
- **Evening**
  When selected, the evening greeting and caller actions are used during the times indicated. The internal number to dial to record the greeting for this period is shown below the setting.

3.4.2.1.4 Out of Hours

Normally the auto attendant's out of hours settings are used at any times when the other time profiles (morning, evening and afternoon) are not valid. This set of settings allows you to define specific days and times on those days when the out of hours settings override all the other. This can be used to automatically apply the out of hours settings all day on specific days.

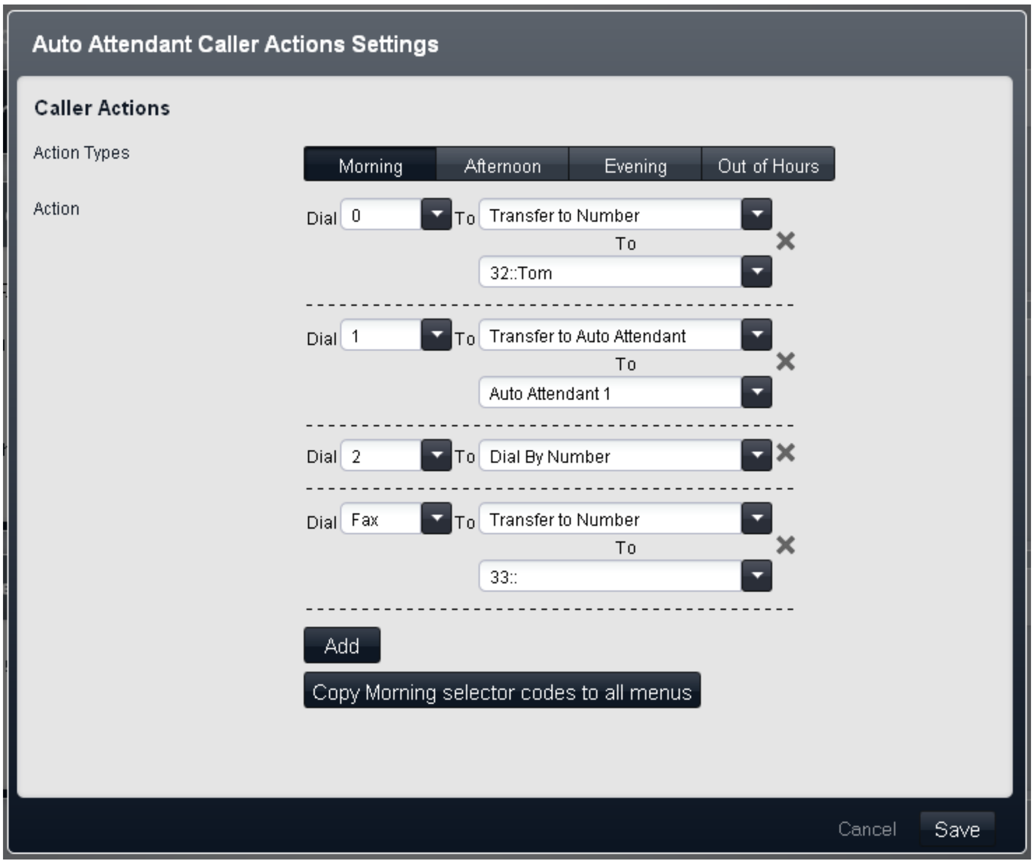
The out of hours settings can also be set to override the other time profiles when the system is in night service. However that method is turned on and off manual using a night service button. The two methods can be combined, but the automatic settings below will still be applied when night service is switched off.

- **Weekly Off**
  This drop down list is used to select the days on which the **Out of Office Hours** set below should be used to override the other auto attendant time settings.
- **Out of office hours**
  These fields set the times used for automatic out of hours operation on the days selected above. The internal number to dial to record the greeting for this period is shown below the setting.

3.4.2.2 Caller Actions

This menu is accessed by clicking on the **Caller Actions** button on the auto attendant details menu. It displays the key presses that have associated actions.

Separate sets of actions can be configured for use during the morning, afternoon and evening periods defined in the auto attendant detail settings. An additional out of hours set can be defined for use at other time (out of hours).



- **Dial**
  The standard telephone dial pad keys, ***0*** to ***9*** plus * and #. The option ***Fax*** can also be selected. This can be used with a ***Transfer to Number*** action to redirect fax calls.

- **To**
  The following actions can be assigned to a key.

  - ***No Action***
    The corresponding key takes no action.

  - ***Dial by Name***
    The caller is asked to dial the name of the user they require and then press #. The recorded mailbox names of matching users are then played back for the caller to make a selection. The name order used is set by the **Dial by Name Match Order** setting. Users without a recorded name prompt or not set to **List in Directory** are not included. Users can record their name by accessing their mailbox and dialing *05.

  - ***Dial By Number***
    This option allows the caller to dial the extension number of the user they require. No destination is set for this option. The attendant's **Dial By Direct Number** setting determines how the digits dialed with this action are used.

  - ***Transfer to Auto Attendant***
    This option transfers the caller to another indicated auto attendant. This will skip the greeting menu of that auto attendant, playing just the current menu options greeting instead.

  - ***Transfer to Emergency Greeting***
    This option transfers the caller to a set of prompts for recording the emergency greeting and for selecting whether the emergency greeting is enabled or not.

    - If a system password 65 has been set, the caller is asked to enter that password before they can continue.

    - When the emergency greeting is enabled, it is played to other auto attendant callers before any other auto attendant greeting.

    - When the emergency greeting is enabled, a warning is displayed on the auto attendant's **Alarm Extension**.

- ***Transfer to Number***
  Transfer the call to the extension or group set with the action.

- ***Replay Menu Greeting***
  Repeat the prompt listing the current menu options.

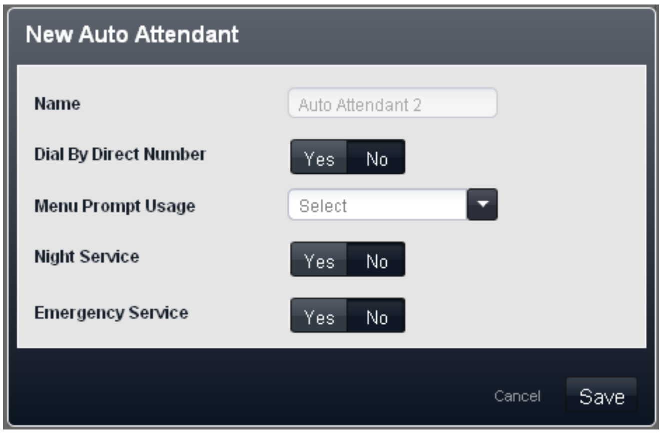
- **To**
  Some actions require a destination to be specified:

  - For the **Transfer to Number** action. The drop down list can be used to select from the available extension and groups configured on the phone system. This list contains an option to collect voicemail.

  - For **Transfer to Auto Attendant** allows selection of the target auto attendant. The option **76: Modem** can be used to select the V32 modem supported by the first analog trunk. This can be used for basic remote access for maintenance.

3.4.2.3 New Auto Attendant

The following menu options are displayed when the **New Auto Attendant** option is clicked on the list of existing auto attendants 48.

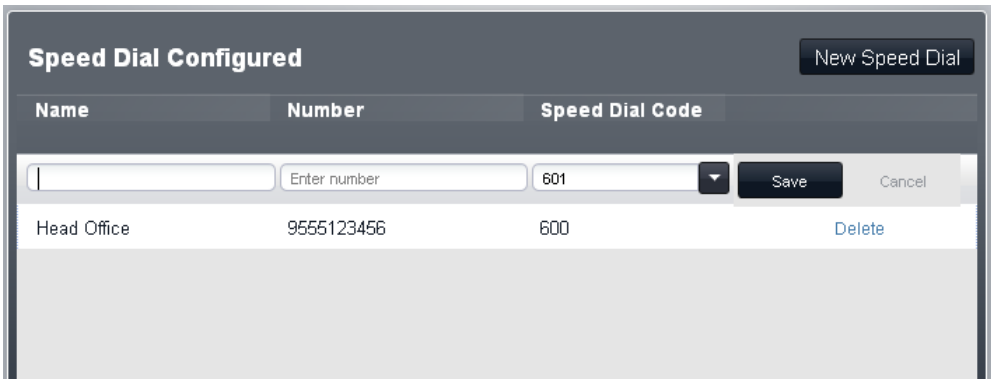


- **Dial By Direct Number:** *Default = On for the first default auto attendant. Off for other auto attendants.*
  This setting affects the operation of any key presses in the auto attendant menu set to use the ***Dial By Number*** action.
  - If selected, the key press for the action is included in any following digits dialed by the caller for extension matching. For example, if 2 is set in the actions to ***Dial by Number***, a caller can dial 20 for extension 20.
  - If not selected, the key press for the action is not included in any following digits dialed by the caller for extension matching. For example, if 2 is set in the actions to ***Dial by Number***, a caller must dial 2 and then 20 for extension 20.
- **Menu Prompt Usage:** *Default = Each menu uses own prompt.*
  Each time profile option used by an auto attendant can have its own set of actions and therefore may require a separate actions prompt to be played after the appropriate greeting prompt. The settings ***Each menu uses own prompt*** does that. Alternately one of the menu options can be selected as the menu option prompt played at all times of day.
- **Night Service:** *Default = On.*
  If selected, when the system is in night service, the auto attendant will switch to using its out of hours greetings and menu actions. If not selected, when the system is in night service, the auto attendant will continue using the greetings and menu options as determined by its own time profile settings.
- **Emergency Service:** *Default = Off.*
  This field indicates when the auto attendants emergency greeting option has been enabled by someone using the auto attendant. The field can also be used to disable the emergency greeting without having to go through the auto attendant menu.

3.5 Outgoing Call Management

3.5.1 Speed Dial

This menu allows you to configure names and numbers that can be accessed by users dialing the associated speed dial code, 600 to 699. The menu is accessed by selecting **Outgoing Call Management** from the menu and clicking on **Speed Dial**.



Speed Dials Configured

The menu displays the settings of the existing system speed dials.

- **Name**
  This is the name that will be associated with the speed dial.
- **Number**
  This is the external number that will be dialed by the telephone system when the speed dial code is dialed by an extension user.
  - Speed dials beginning with * are called 'marked speed dials' and are treated differently. A user can use a marked speed dial even if the number is in one of the user's assigned disallowed number lists. Marked speed dials can also be used when an extension is locked. When dialed, the * is not included. If a * is required to be dialed, the speed dial should be start with **.
  - For PBX mode systems, if the system is configured to use an **Outside Line** prefix for outgoing external calls, that prefix should be included in external speed dial numbers.
- **Speed Dial Code**
  Select a number between 600 and 699. Each number can only appear once in the list. This is the short form substitute number for often-used long numbers.

Adding a Speed Dial

To add a new speed dial, click New Speed Dial. Enter the required values for the speed dial and then click **Save**.

Editing a Speed Dial

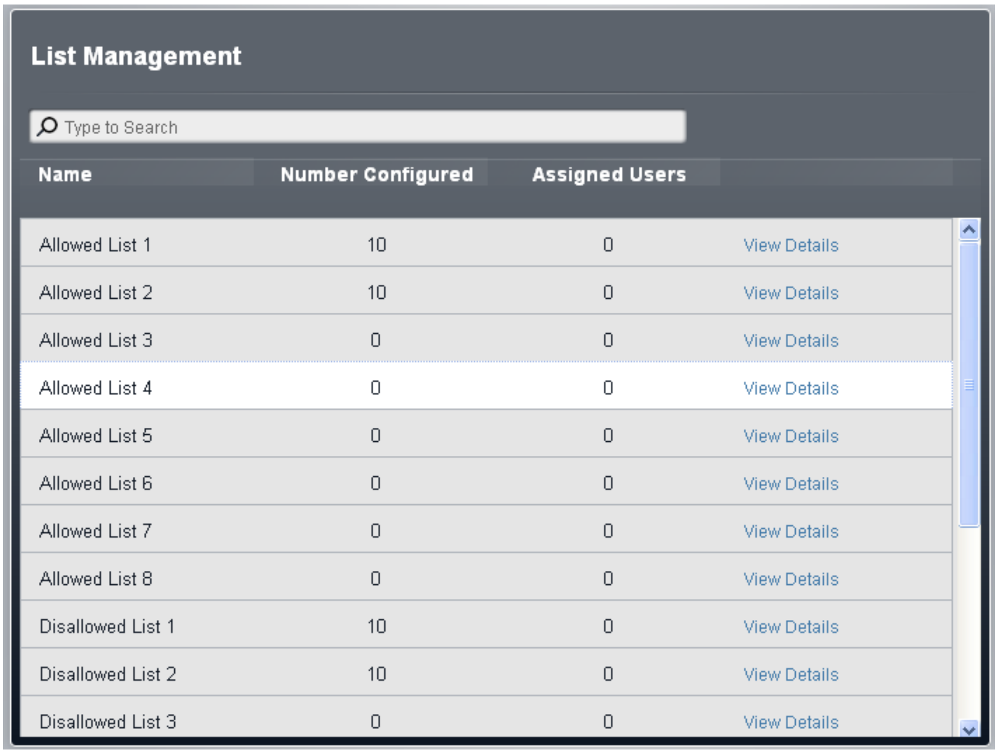
Any existing speed dial can be edited by double-clicking on the speed dial entry in the list.

3.5.2 Calling List

The set of lists is accessed by selecting **Outgoing Call Management** and clicking on **Calling List**. It can also be accessed from the **Phone User List** 33 by clicking on the edit icon shown in the **Outgoing Calls** panel.

The system uses the lists to control the numbers that users can or cannot dial.

- **Allowed List**
  Allowed lists are used to enter numbers or types of numbers that users associated with the list can dial even if they are restricted from dialing other numbers. Up to 8 such lists, each containing 10 numbers, can be configured.
- **Disallowed List**
  Disallowed lists are used to enter numbers or types of numbers that users associated with the list cannot dial. Up to 10 such lists can be configured. Up to 8 such lists, each containing 10 numbers, can be configured.
- **Emergency Number List**
  This list is used to enter numbers that all users can dial at any time regardless of any other settings that might restrict them from dialing numbers for outgoing calls. Up to 10 numbers can be configured in this list.
- **Account Codes**
  Up to 99 account codes can be entered. In addition selected users can be configured to have to enter an account code whenever they make an outgoing external call.


List Management

Type to Search

Name	Number Configured	Assigned Users	
Allowed List 1	10	0	View Details
Allowed List 2	10	0	View Details
Allowed List 3	0	0	View Details
Allowed List 4	0	0	View Details
Allowed List 5	0	0	View Details
Allowed List 6	0	0	View Details
Allowed List 7	0	0	View Details
Allowed List 8	0	0	View Details
Disallowed List 1	10	0	View Details
Disallowed List 2	10	0	View Details
Disallowed List 3	0	0	View Details

To view the contents of a list and the users who are assigned to the list click on **View Details**.

To Edit a Numbers List

1. From the menu bar, click on **User**.
2. The **Outgoing Calls** panel next to the list of users gives a summary of the currently configured lists. Click on the edit icon.
3. From the **List Management** table, select the **View Details** link of the list that you want to edit.
   - The **Telephone Numbers in List** panel displays the allowed numbers. Edit the numbers as required.
   - The **Assign Users to List** panel is used to set which users are assigned to the list.
4. When completed click **Save**.
5. To access another list click on **<< Previous List** or **Next List >>**. Alternatively click on **<< Back** to return to the table of all the lists.

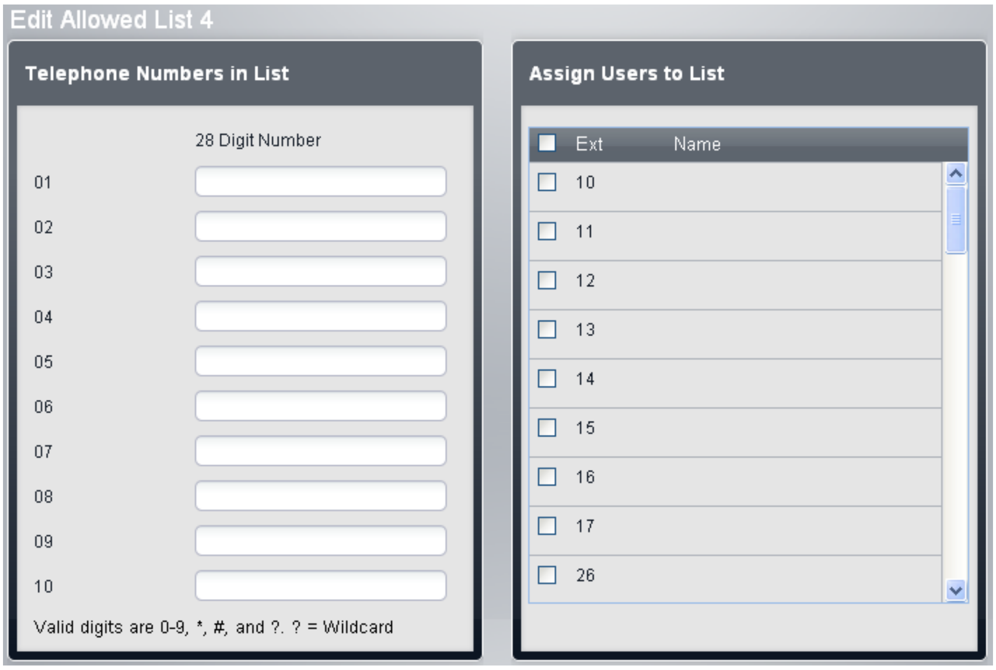
3.5.2.1 Allowed List

Each allowed list contains external telephone numbers that members of the list are allowed to dial. The allowed lists to which a user is assigned override any disallowed lists to which they are also assigned. The numbers in the user's assigned allowed lists also override the **Calls Barred** and **Outgoing Call Restrictions** 43 settings that may be applied to a user.

There are eight lists, each containing up to 10 numbers. Each number can use the telephone dialing digits 0 to 9, *, # and can be up to 28 digits long. You can also use the ? character as a single digit wildcard.

To Edit an Allowed Numbers List

1. From the menu bar, click on **User**.
2. The **Outgoing Calls** panel next to the list of users gives a summary of the currently configured lists. Click on the edit icon.
3. From the **List Management** table, select the **View Details** link of the list that you want to edit.



- The **Telephone Numbers in List** panel displays the allowed numbers. Edit the numbers as required.
- The **Assign Users to List** panel is used to set which users are assigned to the list.

4. When completed click **Save**.
5. To access another list click on **<< Previous List** or **Next List >>**. Alternatively click on **<< Back** to return to the table of all the lists.

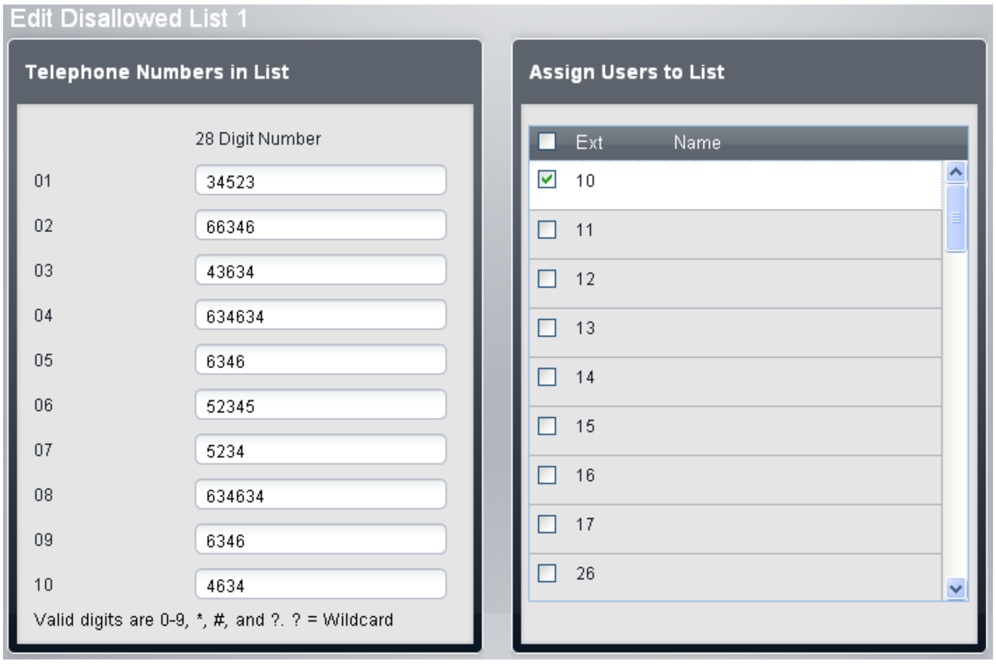
3.5.2.2 Disallowed List

Each disallowed list contains external telephone numbers that users who are assigned to the list cannot dial. There are eight lists, each containing up to 10 numbers. Each number can use the telephone dialing digits 0 to 9, *, # and can be up to 28 digits long. You can also use the ? character as a single digit wildcard.

Numbers in the disallowed lists of which a user is a member are overridden if they also appear in the allowed numbers lists, emergency number list of which the user is a member and also by marked system speed dials.

To Edit an Disallowed Numbers List

1. From the menu bar, click on **User**.
2. The **Outgoing Calls** panel next to the list of users gives a summary of the currently configured lists. Click on the edit icon.
3. From the **List Management** table, select the **View Details** link of the list that you want to edit.



- The **Telephone Numbers in List** panel displays the allowed numbers. Edit the numbers as required.
- The **Assign Users to List** panel is used to set which users are assigned to the list.

4. When completed click **Save**.
5. To access another list click on **<< Previous List** or **Next List >>**. Alternatively click on **<< Back** to return to the table of all the lists.

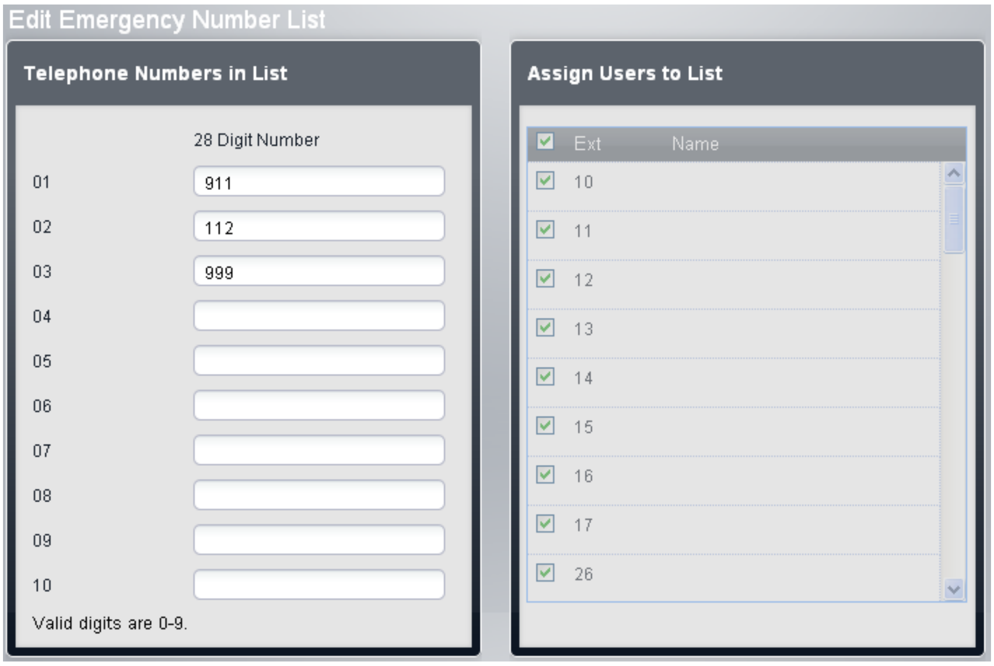
3.5.2.3 Emergency Numbers

You can enter 10 emergency phone numbers into this list. This list is applied to all users and overrides any dialing restrictions that may also be applied to the users. Use of numbers in the list can be viewed using a 911-View/E-View 220 button.

By default the normal emergency numbers for the system locale are automatically added and should not be removed.

To Edit the Emergency Numbers List

1. From the menu bar, click on **User**.
2. The **Outgoing Calls** panel next to the list of users gives a summary of the currently configured lists. Click on the edit icon.
3. From the **List Management** table, select the **View Details** link of the list that you want to edit.



- The **Telephone Numbers in List** panel displays the allowed numbers. Edit the numbers as required.
- The **Assign Users to List** panel is used to set which users are assigned to the list.

4. When completed click **Save**.
5. To access another list click on **<< Previous List** or **Next List >>**. Alternatively click on **<< Back** to return to the table of all the lists.

3.5.2.4 Account Codes

Extensions can be required to enter a valid account code when they make an outgoing external call. The **Account Code Entries** list contains the account codes that are accepted as being valid and the selected users who are required to enter one of these codes, ie. the users who are set to **Forced Account Code Entry**.

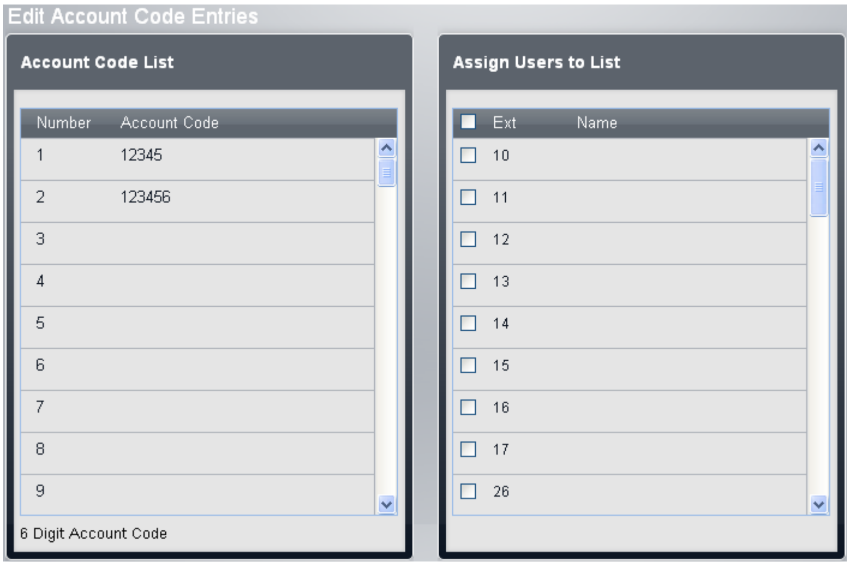
Account codes are commonly used to control cost allocation and out-going call restriction. The account code used on a call is included in the call information output by the system SMDR call log. Users can enter an account code during a call using an **Account Code Entry** button. Once a user has entered an account code with a call, only that user can change that calls account code by entering another one.

Once a call has been completed using an account code, the account code information is removed from the user's call information. This means that redial functions will not re-enter the account code.

All users (except analog phones) can also enter voluntary account codes at any time during a call by using an **Account Code Entry** button. Voluntary account codes are recorded in the same way as forced account codes but are not validated.

To Edit the Account Codes List

1. From the menu bar, click on **User**.
2. The **Outgoing Calls** panel next to the list of users gives a summary of the currently configured lists. Click on the edit icon.
3. From the **List Management** table, select the **View Details** link of the list that you want to edit.



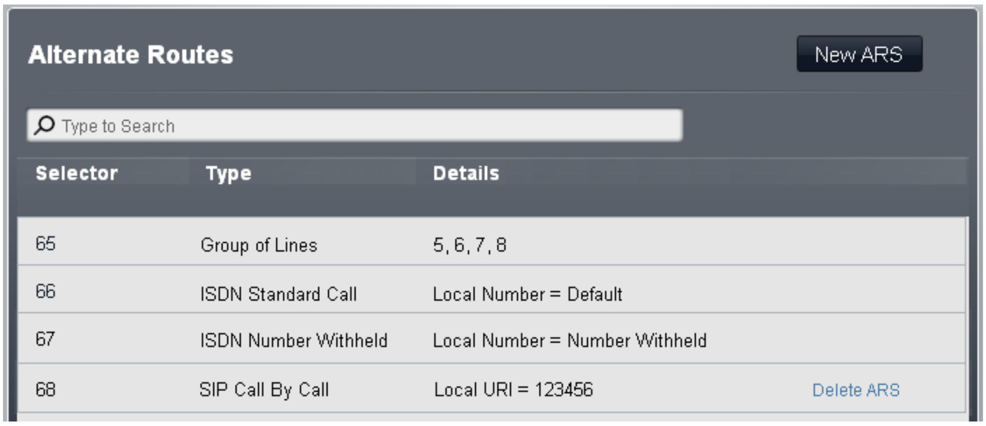
- The **Account Code List** panel displays the account codes. Edit these as required.
- The **Telephone Numbers in List** panel displays the allowed numbers. Edit the numbers as required.
- The **Assign Users to List** panel is used to set which users are assigned to the list.

4. When completed click **Save**.
5. To access another list click on **<< Previous List** or **Next List >>**. Alternatively click on **<< Back** to return to the table of all the lists.

3.5.3 Alternate Route Selection

Alternate route selection is used by systems that are operating in ***PBX*** mode. It is used to group trunks and then associate different types of outgoing numbers with those trunk groups. Alternate route selectors are grouping of trunks, trunks channels or trunk settings. Once a set of ARS entries has been created, they can be associate with the different dialing prefixes 64.

This menu is accessed by clicking **Outgoing Call Management** in the menu bar, selecting **Alternate Route Selection**.



This table shows the existing ARS entries. A particular entry can be edited by double clicking on the entry, adjusting the settings and then clicking **Save**. New entries can be added using the **New ARS** button. Existing entries other than the defaults (65, 66 and 67) can be deleted using the **Delete ARS** link.

- **Selector**
  This must be a number in the range 65 to 99. Selectors 65, 66 and 67 are default entries that cannot be removed. The selector number is used in the Dial Numbers table to associate the ARS with external dialing prefixes that can use it. It can also be used as a line appearance button to allow users to specifically select the ARS group for an outgoing call.

  - ***65: Group of Lines***
    This ARS selector cannot be deleted. By default it contains all analog lines in the system when the system was installed or defaulted. However, it can be edited to change the lines included including adding non-analog lines. This selector and 66 below are used as the default for all classes of external call.

  - ***66: ISDN Standard Call - Local Number = Default***
    This ARS selector cannot be deleted. Calls routed to this selector will use an available ISDN trunk channel with the calling party information set to match the user's **User Calling Line Identity** if set or otherwise blank (to be set by the trunk provider). This selector and 65 above are used as the default for all classes of external call.

  - ***67: ISDN Number Withheld - Local Number = Withheld***
    This ARS selector cannot be deleted. Calls routed to this selector will use an available ISDN trunk channel with the calling party information set to withheld.

- **Type**
  The ARS Selector group can be used for the following functions:

  - ***Group of Lines***
    This type of selector is used to create a group of lines. The lines are selected using the **Select Lines** table below. For a call routed to this selector, an available line from that group is used.

  - ***ISDN Local Number***
    This type of selector is used to set an outgoing local number on an ISDN call. For a call routed to this ARS selector, an available ISDN channel is used with the calling party element of the Q.931 setup set to match the local number specified.

    - Changing the calling party number may not be supported by the line provider or may be an additional chargeable service. It will also be subject to restrictions on what numbers can be used. It is normally a requirement that the calling party number used must be a valid number for return calls to the same trunk. Use of an invalid number may cause the call to be dropped or the number to be replaced by a default value.

    - The default ARS Selector entry 66 is set to ***Local number=default***. It uses the user's **User CLI** if set.

  - ***ISDN Standard Call***
    This type of selector is used to select an available ISDN channel for the call.

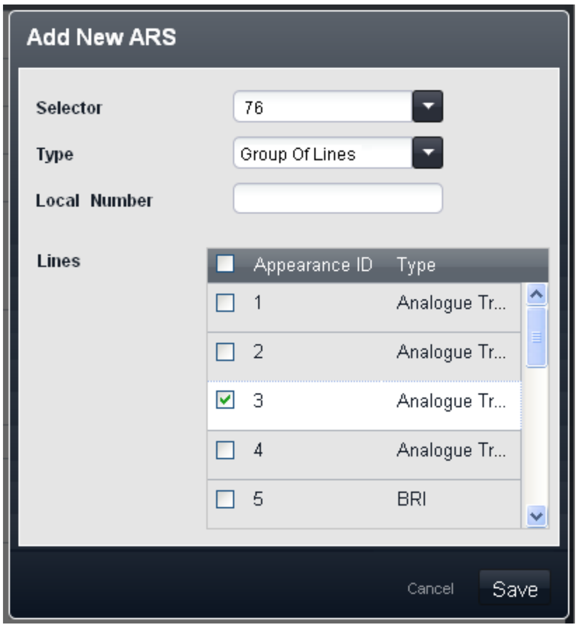
  - ***ISDN Number Withheld***
    This type of selector is used to withhold any outgoing local number information on an ISDN call. For a call routed to this ARS selector, an available ISDN channel is used with the calling party element of the Q.931 setup set to withheld.

  - ***SIP Call-by-Call***
    These entries appear when entries are created in a SIP trunk's **Call-by-Call** 98 table. They cannot be edited through the ARS Selectors form. By having an associated ARS Selector number, the entry can be selected as the destination for specific out going calls.

- **Details**
  This field show either the lines currently selected for use with the ARS selector or the local number setting for the calling party number.

3.5.3.1 New ARS

This menu is accessed by clicking **Outgoing Call Management** in the menu bar, select **Alternate Route Selection** and then select **New ARS**.



Add New ARS

Adjust the settings as required and click **Save**.

- **Selector**
  This must be a number in the range 65 to 99. Selectors 65, 66 and 67 are default entries that cannot be removed. The selector number is used in the Dial Numbers table to associate the ARS with external dialing prefixes that can use it. It can also be used as a line appearance button to allow users to specifically select the ARS group for an outgoing call.
  - ***65: Group of Lines***
    This ARS selector cannot be deleted. By default it contains all analog lines in the system when the system was installed or defaulted. However, it can be edited to change the lines included including adding non-analog lines. This selector and 66 below are used as the default for all classes of external call.
  - ***66: ISDN Standard Call - Local Number = Default***
    This ARS selector cannot be deleted. Calls routed to this selector will use an available ISDN trunk channel with the calling party information set to match the user's **User Calling Line Identity** if set or otherwise blank (to be set by the trunk provider). This selector and 65 above are used as the default for all classes of external call.
  - ***67: ISDN Number Withheld - Local Number = Withheld***
    This ARS selector cannot be deleted. Calls routed to this selector will use an available ISDN trunk channel with the calling party information set to withheld.
- **Type**
  The ARS Selector group can be used for the following functions:
  - ***Group of Lines***
    This type of selector is used to create a group of lines. The lines are selected using the **Select Lines** table below. For a call routed to this selector, an available line from that group is used.
  - ***ISDN Local Number***
    This type of selector is used to set an outgoing local number on an ISDN call. For a call routed to this ARS selector, an available ISDN channel is used with the calling party element of the Q.931 setup set to match the local number specified.
    - Changing the calling party number may not be supported by the line provider or may be an additional chargeable service. It will also be subject to restrictions on what numbers can be used. It is normally a requirement that the calling party number used must be a valid number for return calls to the same trunk. Use of an invalid number may cause the call to be dropped or the number to be replaced by a default value.
    - The default ARS Selector entry 66 is set to ***Local number=default***. It uses the user's **User CLI** if set.
  - ***ISDN Standard Call***
    This type of selector is used to select an available ISDN channel for the call.

- ***ISDN Number Withheld***
  This type of selector is used to withhold any outgoing local number information on an ISDN call. For a call routed to this ARS selector, an available ISDN channel is used with the calling party element of the Q.931 setup set to withheld.

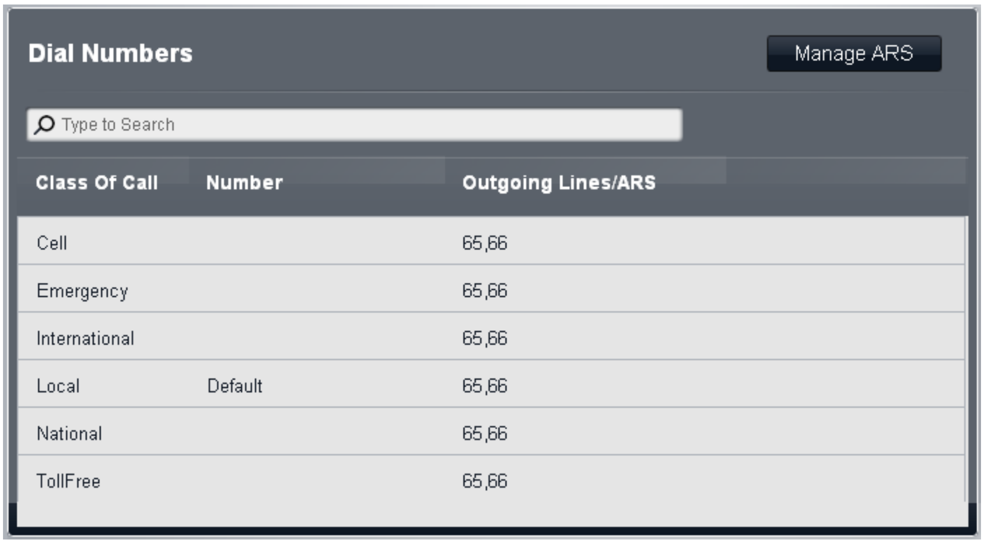
- ***SIP Call-by-Call***
  These entries appear when entries are created in a SIP trunk's **Call-by-Call** 98 table. They cannot be edited through the ARS Selectors form. By having an associated ARS Selector number, the entry can be selected as the destination for specific out going calls.

- **Details**
  This field show either the lines currently selected for use with the ARS selector or the local number setting for the calling party number.

3.5.4 Dial Numbers

Alternate route selection is used by systems that are operating in ***PBX*** mode. It is used to group trunks and then associate different types of outgoing numbers with those trunk groups. This menu is used to group different dialing prefixes by the type of call that they represent. Each of these groups can then be associated with the trunks that should be used for those calls.

This menu is accessed by clicking **Outgoing Call Management** in the menu bar, and selecting **Dial Numbers**.



- **Class of Call**
  The available classes are ***Local***, ***National***, ***International***, ***Emergency***, ***Cell*** and ***Toll Free***. For each, you can define the numbers the dialing prefixes that match that call type and the ARS selector groups to which matching calls should be routed.

- **Number**
  For each class of call, this field is used to define the dialing prefix (up to 5 digits) expected for the call to match the class. Multiple prefix numbers can be entered, each separated by a comma.

  - To edit the numbers, double click on the entry. The fields for **Number** and for **Outgoing Lines/ARS** become adjustable. Edit the values as required and click **Save**.

    - Do not include the **Outside Line** prefix digit configured in the system settings.

    - If a match occurs in more than one class, the most exact match is used, ie. the one with the most digits. If multiple matches still exist, the match that occurs first in the table is used.

    - Numbers cannot be set for the **Local** class. This class is used for any calls that do not match any other class. However the ARS selectors used by this class can be changed.

- **Outgoing Lines/ARS**
  This field indicates the ARS selectors currently associated with the Class of Call. These contain the trunks that are used by the Class of Call and are set using the Select Outgoing ARS table.

  - To edit the numbers, double click on the entry. The fields for **Number** and for **Outgoing Lines/ARS** become adjustable. Edit the values as required and click **Save**.

- **Manage ARS**
  This button can be used to access the menu for creating, deleting and editing the ARS trunk groups59.

3.6 System

3.6.1 Switch

This menu is accessed by selecting **System** in the menu bar and clicking on **Switch**.

3.6.1.1 System Parameters

The following settings are shown in this panel:

- **System Name**
  A name used to identify the system. This is typically used to identify the configuration by the location or customer's company name. Some features require the system to have a name. This field is case sensitive. Do not use <, >, |, \0, :, *, ?, . or /.

- **Mode:**
  The system can operate in either ***Key*** or ***PBX*** mode. Changing the mode requires the IP Office system to be restarted and will overwrite all existing button programming. For more details see Setting the System Mode 135.

  - **Key**
    The **Number of Lines** setting is used to automatically assign line appearance buttons on all extensions with programmable buttons. To make external calls the user should select an available line appearance button. Outbound call routing is determined by which line appearance button the user selects before dialing or by the user's automatic line selection settings.

  - **PBX**
    No line appearances are automatically assigned to programmable buttons. The **Outside Line** setting is used to set the dialing prefix that indicates that the call is an external one for which an available line should be seized. The **Alternate Route Selection** 59 settings are used to determine which lines are used for each outgoing call. Line appearance buttons can also still be configured for making and answering external calls.

- **Voicemail Mode:** *Default = Intuity Mode. Software level = 8.0+.*
  Embedded voicemail can use either ***IP Office Mode*** or ***Intuity Mode*** key presses for mailbox functions. End users should be provided with the appropriate mailbox user guide for the mode selected. Pre-Release 8.0 systems use ***IP Office Mode*** only.

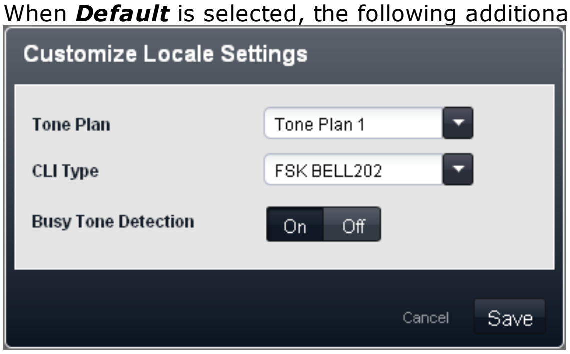
- **IP Address**
  This field sets the address of the PC allowed to send files to the system's memory card.

- **Country:**
  This option sets a range of country specific telephony settings. The system language can be changed from the Country setting using the separate **Language** setting below.

  - **! WARNING - Reboot Required**
    Changing this setting requires the system to be rebooted 121 for the change to take effect. Rebooting the system will end all calls currently in progress.

  - The supported countries are ***Argentina***, ***Australia***, ***Bahrain***, ***Belgium***, ***Brazil***, ***Canada***, ***Chile***, ***China***, ***Customize***, ***Denmark***, ***Egypt***, ***Finland***, ***France***, ***Germany***, ***Greece***, ***Hong Kong***, ***Hungary***, ***Iceland***, ***India***, ***Italy***, ***Korea***, ***Kuwait***, ***Mexico***, ***Netherlands***, ***New Zealand***, ***Norway***, ***Oman***, ***Pakistan***, ***Peru***, ***Poland***, ***Portugal***, ***Qatar***, ***Russia***, ***Saudi Arabia***, ***Singapore***, ***South Africa***, ***Spain***, ***Sweden***, ***Switzerland***, ***Taiwan***, ***Turkey***, ***United Arab Emirates***, ***United States***, ***Venezuela***.

  - When ***Default*** is selected, the following additional fields are available:



    - **Tone Plan:** *Default = Tone Plan 1*
      Select a tone plan to be used for different ringing signals such as dial tone and ring tone.

    - **CLI Type:** *Default = FSK V23*
      Set the method for passing caller ID information to analog extensions. The options are ***DTMF***, ***FSK Bell 202*** or ***FSK V23***.

    - **Busy Tone Detection:** *Default = Off*
      Enable or disable the use of busy tone detection for call clearing.

- **Language**
  This field sets the language used for voicemail prompts and phone displays if the language is available. Possible languages are:

  - ***Arabic***, ***Cantonese***, ***Danish***, ***Dutch***, ***English (UK)***, ***English (US)***, ***Finnish***, ***French***, ***French (Canada)***, ***German***, ***Italian***, ***Korean***, ***Mandarin***, ***Norwegian***, ***Polish***, ***Portuguese***, ***Portuguese (Brazil)***, ***Russian***, ***Spanish***, ***Spanish (Argentina)***, ***Spanish (Latin)***, ***Spanish (Mexico)***, ***Swedish***, ***Turkish***.

  - **! WARNING - Reboot Required**
    Changing this setting requires the system to be rebooted 121 for the change to take effect. Rebooting the system will end all calls currently in progress.

  - For each user, their language settings can be changed using the user's **Language** setting. This affects the language used on their phone's display and for mailbox access prompts.

  - For each auto attendant, the system language setting can be overridden by the auto attendant's own **Language** 48 setting.

- **Password:** *Default = Blank. Range = 4 digits.*
  This is a four digit code used to restrict access to some functions. Once set, the system password must be used to override station lock, forced account, numbers in the disallowed calls list and night service restrictions to make a call. The system password is also requested when a user switches the phone system into or out of night service mode or tries to access an auto attendant's emergency greeting settings.

  - For M-Series and T-Series phones, the system password, if set, is also used to control access to phone based administration from the first two extensions in the system.

- **Number of Lines:** *Default = 5 or the number of analog trunks present when the system is first started.*
  This option is only available for systems with their **Mode** *(see above)* set to ***Key***. For phones with programmable buttons, those buttons can be configured as line appearance buttons that each match a particular incoming line. This setting controls how many of buttons on every user's phone are automatically allocated as line appearance buttons. The assignment is done starting from button 03 upwards in order of the lines available.

- **! Warning**
  If the **Number of Lines** value is changed, all existing line appearance buttons and automatic line selection settings are overwritten. The existing functions on other programmable buttons are also overwritten if they are in the range of buttons now specified for lines. Therefore it is recommended that this setting is only changed when a system is first installed.

* **Outside Line:** *Default = Depend on system locale, see below.*
  This option is only available for systems with their **Mode** (see above) set to ***PBX***. It sets the digit which, when dialed, indicates that the call is intended to be external. Routing of any additional digits is then determined through the **Alternate Route Selection** 59 settings.

  - **! WARNING - Reboot Required**
    Changing this setting requires the system to be rebooted 12 for the change to take effect. Rebooting the system will end all calls currently in progress.

  - ***9 (Operator is 0)***
    The prefix 9 is used for external calls. The digit 0 is used for calls to the operator extension (the first extension in the system). This is the default setting for systems with the **Country** setting ***United States***.

  - ***None***
    No prefix is used for external calls. Any dialing that does not match an internal dial plan number is assumed to be an external call. This is the default setting for systems with the **Country** setting other then ***Germany*** or ***United States***. The digit 0 is used for calls to the operator extension (the first extension in the system).

  - ***0 (Operator is 9)***
    The prefix 0 is used for external calls. The digit 9 is used for calls to the operator extension (the first extension in the system). This is the default setting for systems with the **Country** setting ***Germany***.

* **Log All Caller ID Calls for Users:** *Default = None selected.*
  All extensions have a call log of their last 30 calls (incoming answered and missed). The user can access this using a programmable button set to **Call Log** or their phone's Call Log or History button if it has one. In addition, up to 3 extensions can be configured to have access to the call log of the last 400 calls (incoming answered and missed) for the whole system. These fields are used to select those users. Only calls that include caller ID are included. The **!** character on the phone display indicates that there are unviewed call details in the call log.

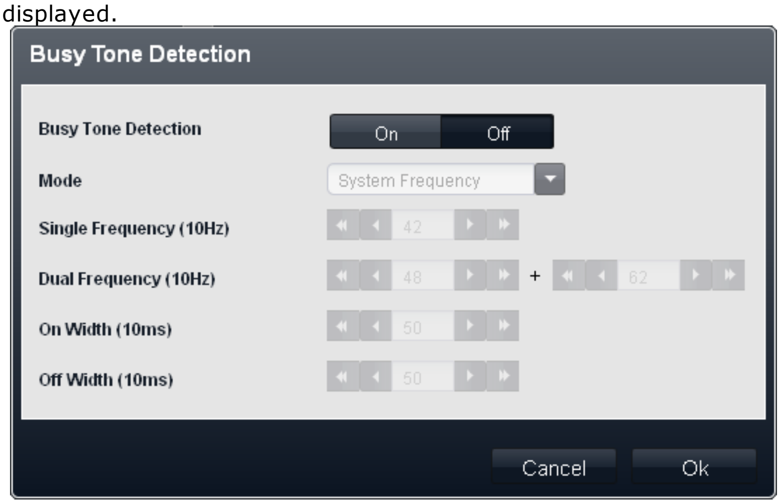
* **Allow Unsupervised Analog Trunk Disconnect:** *Default = No.*
  When using analog trunks, various methods are used for trunk supervision, ie. to detect when the far end of the trunk has disconnected and so disconnect the local end of the call. Depending on the locale, the system uses Disconnect Clear signalling and or Busy Tone Detection. This setting should only be enabled if it is known that the analog trunks do not provide disconnect clear signalling or reliable busy tone. When enabled:

  - Disconnect clear signalling detection is turned off. Busy tone detection remains on.

  - Unsupervised transfers and trunk-to-trunk transfers of analog trunk calls are not allowed.

  - A wider range of busy tones which may signal that the caller has disconnected are used to disconnect calls connected to voicemail.

  - When this setting is changed to ***No***, the configuration settings for **Busy Tone Detection** are displayed.



  - **Mode:** *Default = System Frequency*
    If set to ***System Frequency***, the settings used are the default settings for the system locales. To change the settings, select either ***Single Frequency*** or ***Dual Frequency*** to match the line providers requirements.

- **Single Frequency**
  If the **Mode** is set to ***Single Frequency***, set the frequency.

- **Dual Frequency**
  If the **Mode** is set to ***Dual Frequency***, set the frequencies.

3.6.1.2 Network Settings

The following settings are shown in this panel.

  - **! WARNING - Reboot Required**
    Changing any of these settings requires the system to be rebooted 121 for the changes to take effect. Rebooting the system will end all calls currently in progress.

- **Receive IP Address Via DHCP Server:** *Default = Yes.*
  This setting controls whether the system acts as a DHCP client or uses a fixed IP address.

  - If enabled, the system acts as a DHCP client and requests IP address details for its LAN port when the system is started.

    - If it receives a response, the address details it has been given by the DHCP server are shown in the field below but cannot be adjusted.

    - If it does not receive a response, it default to using the address 192.168.42.1. It is still a DHCP client and will request an address again when it is next restarted.

  - If not enabled, the system uses the IP address values set in the fields below.

- **System IP Address:** *Default = 192.168.42.1*
  Enter the IP address that the telephone system should use if **Receive IP Address Via DHCP Server** is not selected. If **Receive IP Address Via DHCP Server** is selected, this field is greyed out but does display the IP address that the system is currently using.

- **Subnet Mask:** *Default = 255.255.255.0*
  Enter the Sub-Net Mask that the telephone system should use if **Receive IP Address Via DHCP Server** is not selected. If **Receive IP Address Via DHCP Server** is selected, this field is greyed out but does display the IP address that the system is currently using.

- **Default Gateway:** *Default = 0.0.0.0*
  Enter the **Default Gateway** that the telephone system should use if **Receive IP Address Via DHCP Server** is not selected. If **Receive IP Address Via DHCP Server** is selected, this field is greyed out but does display the IP address that the system is currently using.

- **DNS Settings**
  This option is greyed out on systems configured to use DHCP as the DHCP server will provide DNS information. For systems not using DHCP, clicking DNS Settings displays the **DNS Settings** menu.

  - **DNS Server IP Address:** *Default = 0.0.0.0*
    This field sets the address for the primary DNS server that the system should use to try to resolve domain names to IP addresses.

  - **Backup DNS Server IP Address:** *Default = 0.0.0.0*
    This field sets the address for the secondary DNS server that the system should use if there is no response from the primary DNS server.

3.6.1.3 System Details

The following settings are shown in this panel:

- **Companding Law**
  The system is automatically defaulted to A-Law or U-Law by the type of SD Feature Key dongle inserted into the unit. Typically U-Law is used in North American locales, A-Law is used in most other locales. U-Law is also called Mu-Law or µ-Law. For some installations, it may be necessary to change this setting if advised by the external line provider.

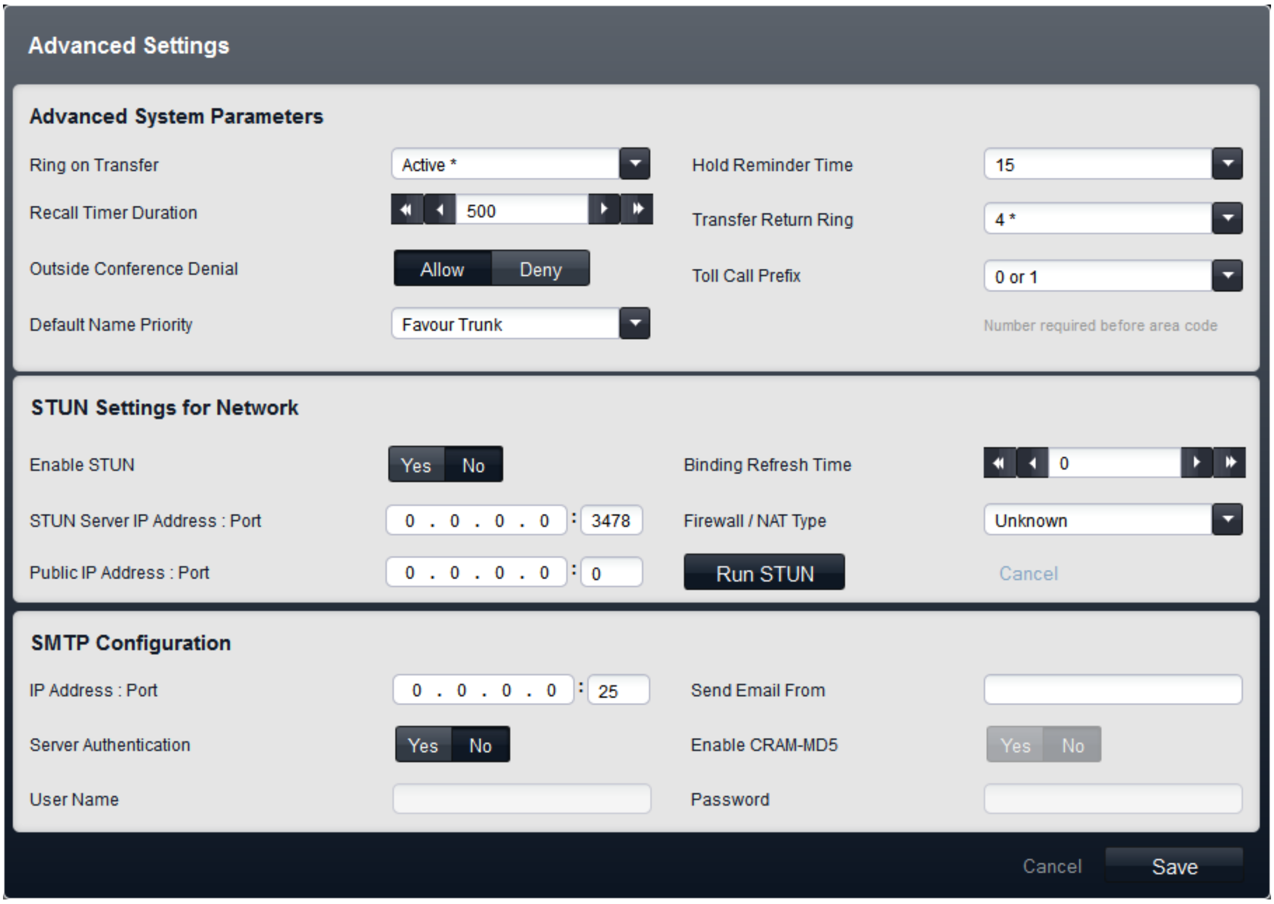
  - ETR6 cards are not supported for systems running in A-Law mode.

- **Automatic DST:** *Default = On.*
  When selected, the telephone system will automatically apply daylight saving time (DST) adjustments to its internal clock. This feature should only be used for systems in a North American locale.

- **Enable Network Time Synchronization:** *Default = On.*
  When selected, the system will use the time included in the ICLID on incoming calls as its system time. Note that this feature uses the first analog trunk on the card installed in slot 1 of the system control unit.

3.6.1.4 Advanced

This menu is accessed by clicking the **Advanced** button on the **System** menu.



Advanced System Parameters

The following settings are shown in this panel:

- **Ring on Transfer:** *Default = Active.*
  If selected, callers being transferred hear ringing during the transfer process. If not selected, the caller will hear music on hold.

- **Hold Reminder Time:** *Default = 60 seconds. Range = 0 (Off) to 180 seconds*.
  This setting controls how long calls remain on hold before recalling to the user who held the call. Note that the recall only occurs if the user has no other connected call. Recalled calls will continue ringing and do not follow forwards or go to voicemail.

- **Recall Timer Duration:** *Default = 500. Range = 25 to 800 milliseconds.*
  This is the flash pulse width used for analog trunks and T1 trunks.

  - **! WARNING - Reboot Required**
    Changing this setting requires the system to be rebooted [12] for the change to take effect. Rebooting the system will end all calls currently in progress.

- **Transfer Return Ring:** *Default = 4 (20 seconds), Range 1 to 180 seconds.*
  Sets the delay after which any call transferred by a user that remains unanswered, should return to the user. A return call will continue ringing and does not follow any forwards or go to voicemail. Transfer return will occur if the user has an available call appearance button. Transfer return is not applied if the transfer is to a hunt group.

- **Outside Conference Denial:** *Default = Allowed.*
When set to the ***Allowed***, more than one outside line can be added to a conference. When set to the ***Disallowed***, a second outside line can not be added to a conference. This feature does not change based on the type of outside line. The intent of this feature is to minimize toll fraud. For example, if set to disallowed, this would prevent someone from accepting an outside call at an extension, conferencing in another outside party, and then walking away allowing the two parties to converse.

- **Toll Call Prefix:** *Default = 0 or 1 Required Before Area Code.*
Allows selection between ***0 or 1 Required Before Area Code*** or ***Area Code and Number Only***.

  - **! WARNING - Reboot Required**
  Changing this setting requires the system to be rebooted 121 for the change to take effect. Rebooting the system will end all calls currently in progress.

- **Default Name Priority:** *Default = Favour Trunk. Software level = 8.0+.*
For SIP trunks, the caller name displayed on an extension can either be that supplied by the trunk or one obtained by checking for a number match in the system speed dials. This setting determines which method is used by default. For each SIP line, this setting can be overridden by the line's own **Name Priority** setting if required.

  - ***Favour Trunk***
  Display the name provided by the trunk. For example, the trunk may be configured to provide the calling number or the name of the caller. The system should display the caller information as it is provided by the trunk.

  - ***Favour Directory***
  Search for a number match in the system speed dials. The first match is used and overrides the name provided by the SIP line. If no match is found, the name provided by the line is used.

STUN Settings for Network

These settings are used if SIP trunks are added to the phone system's configuration using the **SIP Trunk Administration** 98 menu. These settings are necessary to allow SIP connections from the network on which the phone system is attached to reach the public network on which the SIP provider is located.

The following fields can be completed either manually or the phone system can attempt to automatically discover the appropriate values. To complete the fields automatically, only the **STUN Server IP Address** is required. STUN operation is then tested by clicking **Run STUN**. If successful the remaining fields are filled with the results.

  - **! WARNING - Reboot Required**
  Changing any of these settings requires the system to be rebooted 121 for the changes to take effect. Rebooting the system will end all calls currently in progress.

- **Enable STUN:** *Default = Off*
This field is used to select whether STUN is used or not.

- **STUN Server IP Address:** *Default = Blank*
This is the IP address of the line providers SIP STUN server. The phone system will send basic SIP messages to this destination and from data inserted into the replies can try to determine the type ITSP NAT changes being applied by any firewall between it and the ITSP.

- **STUN Port:** *Default = 3478*
Defines the port to which STUN requests are sent if STUN is used.

- **Firewall/NAT Type:** *Default = Unknown*
The settings here reflect different types of network firewalls.

  - ***Blocking Firewall***
  Allow outgoing TFTP WRQ. Typically this will be the case. It has been observed that the Avaya corporate firewall permits outgoing TFTP RRQ.

  - ***Symmetric Firewall***
  SIP packets are unchanged but ports need to be opened and kept open with keep-alives. If this type of NAT is detected or manually selected, a warning 'Communication is not possible unless the STUN server is supported on same IP address as the ITSP will be displayed as part of the manager validation.

  - ***Open Internet***
  No action required. If this mode is selected, STUN lookups are not performed.

  - ***Symmetric NAT***
  A symmetric NAT is one where all requests from the same internal IP address and port, to a specific destination IP address and port, are mapped to the same external IP address and port. If the same host sends a packet with the same source address and port, but to a different destination, a different mapping is used. Furthermore, only the external host that receives a packet can send a UDP packet back to the internal host. SIP Packets need to be mapped but STUN will not provide the correct information unless the IP address on the STUN server is the same as the ITSP Host. If this type of NAT/Firewall is detected or manually selected, a warning 'Communication is not possible unless the STUN server is supported on same IP address as the ITSP' will be displayed as part of the manager validation.

- ***Full Cone NAT***
  A full cone NAT is one where all requests from the same internal IP address and port are mapped to the same external IP address and port. Furthermore, any external host can send a packet to the internal host, by sending a packet to the mapped external address. SIP packets need to be mapped to NAT address and Port; any Host in the internet can call in on the open port, that is the local info in the SDP will apply to multiple ITSP Hosts.

- ***Restricted Cone NAT***
  A restricted cone NAT is one where all requests from the same internal IP address and port are mapped to the same external IP address and port. Unlike a full cone NAT, an external host (with IP address X) can send a packet to the internal host only if the internal host had previously sent a packet to IP address X. SIP packets needs to be mapped. Responses from hosts are restricted to those that a packet has been sent to. So if multiple ITSP hosts are to be supported, a keep alive will need to be sent to each host. If this type of NAT/Firewall is detected or manually selected, no warning will be displayed for this type of NAT.

- ***Port Restricted Cone NAT***
  A port restricted cone NAT is like a restricted cone NAT, but the restriction includes port numbers. Specifically, an external host can send a packet, with source IP address X and source port P, to the internal host only if the internal host had previously sent a packet to IP address X and port P. SIP packets needs to be mapped. Keep-alives must be sent to all ports that will be the source of a packet for each ITSP host IP address. If this type of NAT/Firewall is detected or manually selected, no warning will be displayed for this type of NAT. However, some Port Restricted have been found to be more symmetric in behavior, creating a separate binding for each opened Port, if this is the case the manager will display NATs a warning 'Communication is not possible unless the STUN server is supported on same IP address as the ITSP' as part of the manager validation.

- ***Unknown***
  Use this setting if the other settings are unsuitable

- ***Static Port Block***
  Use the RTP port range 49152 to 53246.

- **Binding Refresh Time (seconds):** *Default = 0 (Never). Range = 0 to 3600 seconds.*
  Having established which TCP/UDP port number to use, either through automatic or manual configuration, the phone system can send recurring 'SIP Options requests' to the remote proxy terminating the trunk. Those requests will keep the port open through the firewall. Requests are sent every x seconds as configured by this field. If a binding refresh time has not been set you may experience problems receiving inbound SIP calls as they are unable to get through the Firewall. In these circumstances make sure that this value has been configured.

- **Public IP Address:** *Default = 0.0.0.0*
  This value is either entered manually or discovered by the Run STUN process. If no address is set, the phone system IP address is used.

- **Public Port:** *Default = 0*
  This value is either entered manually or discovered by the Run STUN process.

- **Run STUN**
  This button tests STUN operation between the phone system and the STUN Server IP Address set above. If successful the results are used to automatically fill the remaining fields with the discovered values. Before using **Run STUN** the SIP trunk must be configured.

SMTP Server Configuration

Email can be used to provide users with an alert when they have a new voicemail message. This feature is called voicemail email. This requires the system to be configured with details of an SMTP email server account which is used to forward the messages to the user's email address.

- **! WARNING - Reboot Required**
  Changing any of these settings requires the system to be rebooted 121 for the changes to take effect. Rebooting the system will end all calls currently in progress.

- **FQDN/IP Address:** *Default = 0.0.0.0*
  This field sets the IP address of the SMTP server being used to forward emails.

- **Port:** *Default = 25. Range = 0 to 65534.*
  This field sets the destination port on the SMTP server.

- **Send Email From:** *Default = Blank*
  This field sets the sender address to be used for emails from the system. Depending of the authentication requirements of the SMTP server, this may need to be a valid email address hosted by that server. Otherwise the SMTP email server may need to be configured to allow SMTP relay of this address.

- **Server Authentication:** *Default = On*
  This field should be selected if the SMTP server being used requires authentication to allow the sending of emails. When selected, the **User Name** and **Password** fields become available.

- **Enable CRAM-MD5:** *Default = Off.*
  This field should be selected if the SMTP server uses CRAM-MD5.

- **Use STARTTLS:** *Default = Off. (Release 9.0.3).*
  Select this field to enable TLS/SSL encryption. Encryption allows voicemail-to-email integration with hosted email providers that only permit SMTP over a secure transport.

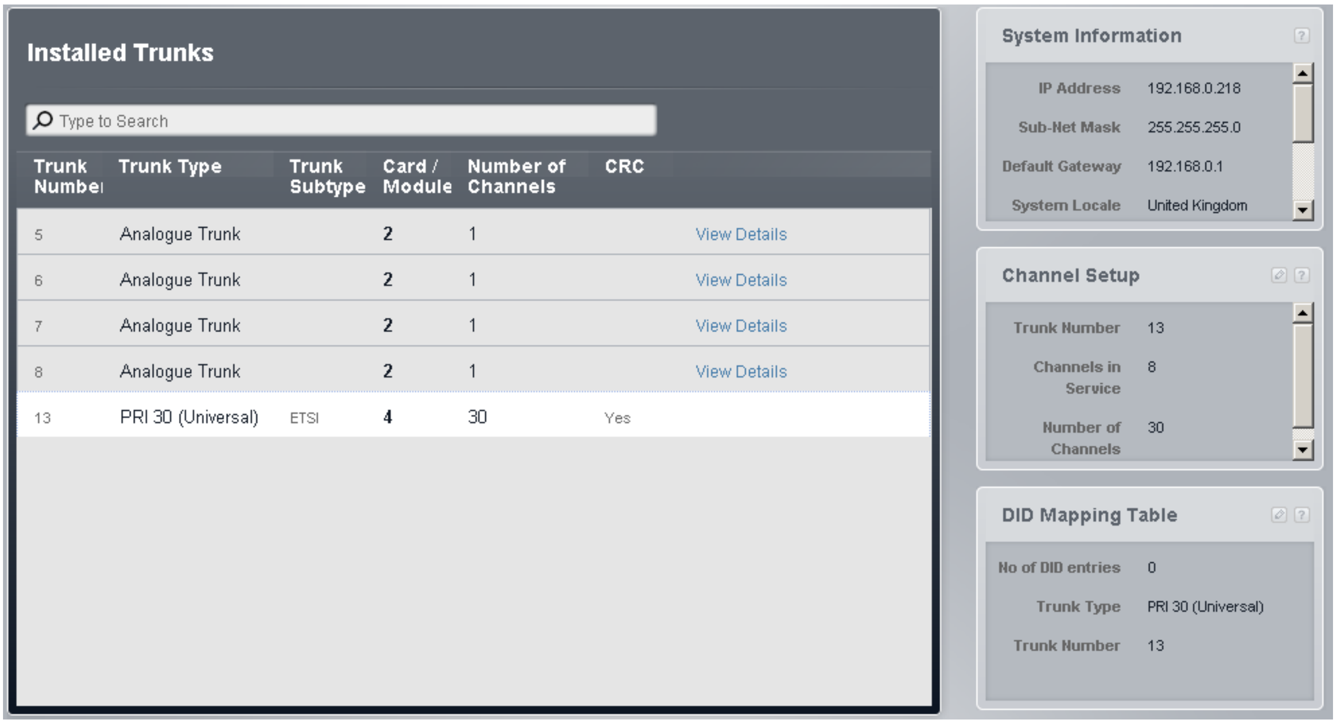
- **User Name:** *Default = Blank*
  This field sets the user name to be used for SMTP server authentication.

- **Password:** *Default = Blank*
  This field sets the password to be used for SMTP server authentication.

3.6.2 Trunks

This menu is accessed by selecting **System** and then **Trunks** from the menu bar. This table list the trunks installed in the system. It does not include SIP trunks. For most trunk types, various trunk settings can be accessed by clicking the **View Details** option next to the trunk entry.

For trunks that support multiple channels, the individual channel settings can be accessed by selecting the trunk in the table and then clicking on the edit icon in the **Channel Setup** panel on the right. Similarly, for trunks that support DID or ICLID.



Installed Trunks

- **Trunk Number:** *Information only, not editable.*
- **Trunk Type:** *Not Editable*
  This value indicates the type of trunk. The menu fields and sub-menus will vary depending on the **Line Type**.
- **Trunk Subtype**
  The option is used with non-analog trunks. There are cases where the trunk sub-type can be changed.
  - For trunks provided by IP500 BRI trunk cards, the trunk subtype is fixed as ***ETSI***.
  - For trunks provided by IP500 PRI-U trunk cards, the trunk sub-type depends on the system's region.
    - In North American locales, the **Trunk Type** is shown as ***PRI 24 (Universal)*** and the **Trunk Subtype** can be set to either ***PRI*** or ***T1***.
    - In non-North American locales, the **Trunk Type** is shown as ***PRI 30 (Universal)*** and the **Trunk Subtype** is fixed as ***ETSI***.
- **Card/Module**
  Indicates the card slot or expansion module being used for the trunk device providing the line. 1 to 4 match the slots on the front of the phone system from left to right. Expansion modules are numbered from 6 upwards.
- **Number of Channels**
  This field is shown only for ETSI PRI trunks. The setting can be changed by double clicking on the entry, selecting the value required and then clicking **Save**. For other types of PRI trunk, the service status of channels can be accessed through channel setup.
- **CRC**
  This field is shown only for ETSI PRI trunks. The setting can be changed by double clicking on the entry, selecting the value required and then clicking **Save**.
- **View Details**
  This option is used to access additional trunk details. The range of details depend on the trunk type. This option is not shown for ETSI PRI trunks.

Displaying Trunk Details
Additional settings are available for most trunk types. To view those settings, click on the **View Details** option next to the trunk.

Changing the Trunk Type
The trunks provided by an IP500 PRI-U card installed in a North American system can operate in either ***T1*** or ***PRI*** mode. To change the current mode, double click on the trunk entry in the list and select the required trunk subtype. The change must then be saved and the system rebooted.

Setting Up Trunk Channels
Non-analog trunks support multiple channels. The number of channels depend on the trunk type and its sub-type. Each channel can have its own call routing settings.

1. Click on the trunk in the trunk list. A summary of the trunk's channels is shown in the **Channel Setup** box on the right.
2. Click on the edit icon in the **Channel Setup** box.

Setting Up DID Mapping
Incoming calls on a trunk channel can include addition information that the system can use for call routing. This allows the call to override the settings of the particular trunk channel on which it arrived and is done through the trunk's DID Mapping table.

1. Click on the trunk in the trunk list. A summary of the trunk's DID entries is shown in the **DID Mapping Table** box on the right.
2. Click on the edit icon in the **DID Mapping Table** box.

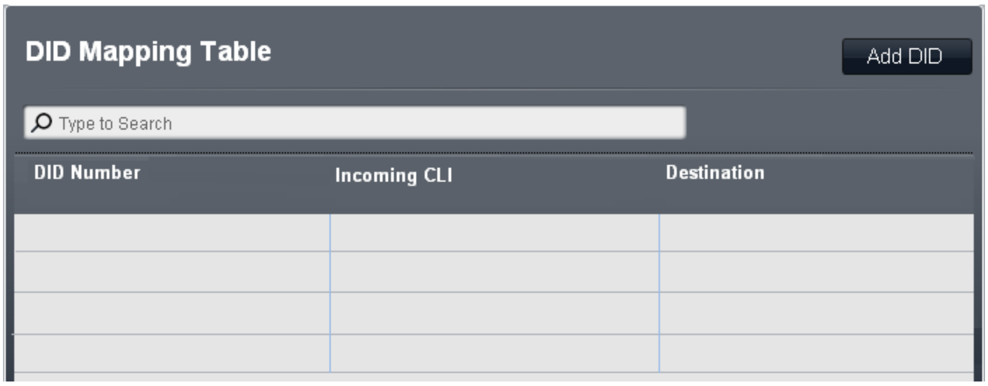
DID Mapping Table

This menu is accessed by selecting a trunk in the trunk's list and then clicking the edit icon in the **DID Mapping Table** box on the right.

DID call mapping can be used on all trunk types except analog and SIP (SIP trunks use can use Call-by-Call routing which is similar). When used, incoming calls that include DID and or ICLID digits are checked for a match in the trunks DID mapping table. If a match is found, it is used. This overrides the **Coverage Destination** settings of the trunk channel on which the call was received. In addition, calls routed by DID mapping are not affected by the phone system being put into night service.

- For systems running in Key mode, DID call mapping can be used in addition to the Coverage Destination settings of a trunk channel and only overrides the trunk channel's settings if a match is found.
- For systems running in PBX mode, DID call mapping overrides all trunk channel call routing settings. A default entry, that matches any calls for which there is no other specific match, is added to the DID call mapping settings of every trunk. The default entry cannot be deleted and has the **Operator** group as its fixed destination.

- **Default DID Routing**
  If the system is in ***Key*** system mode and no match is found, the call is routed to the first extension in the system. If the system is in ***PBX*** mode and no match is found, the call is routed to the Operator group.



- **DID Number**
  If the incoming DID of a call on the trunk matches the DID set here, it will be routed to this destination. The system supports up to 4 digits DID (additional digits after the first 4 are ignored). Leave blank if only CLI matching is required.
- **Incoming CLI**
  If the incoming caller number on the trunk matches the Incoming CLI set here, it will be routed to this destination. Leave blank if only DID matching is required.
- **Destination**
  When this field is selected, the drop down list allows selection of the destination for matching calls. The options differ depending on whether the system's **Mode** [65] is set to ***Key*** or ***PBX***.
  - **Extension**
    Route incoming calls to a particular extension.
  - **Phantom Extension**
    A phantom extension can be selected as the destination for calls.
  - **Hunt Group**
    Incoming calls can be routed to one of the 6 rotary hunt groups.
  - **Calling Group**
    For systems with their **Mode** set to ***PBX***, incoming calls can be routed to one of these 4 ring all groups.
  - **Operator Group**
    For systems with their **Mode** set to ***PBX***, incoming calls are routed to the **Operator Group**.
  - **Voicemail**
    Route incoming calls to the systems voicemail to collect messages. This requires the caller to know the mailbox number and passcode.
  - **76: Modem**
    The option ***76: Modem*** can be selected to route the call to the systems built in V32 modem function. This is only intended for basic access by system maintainers.
  - **Auto Attendant**
    Any configured voicemail auto attendants can be selected as the call destination.

Channel Setup

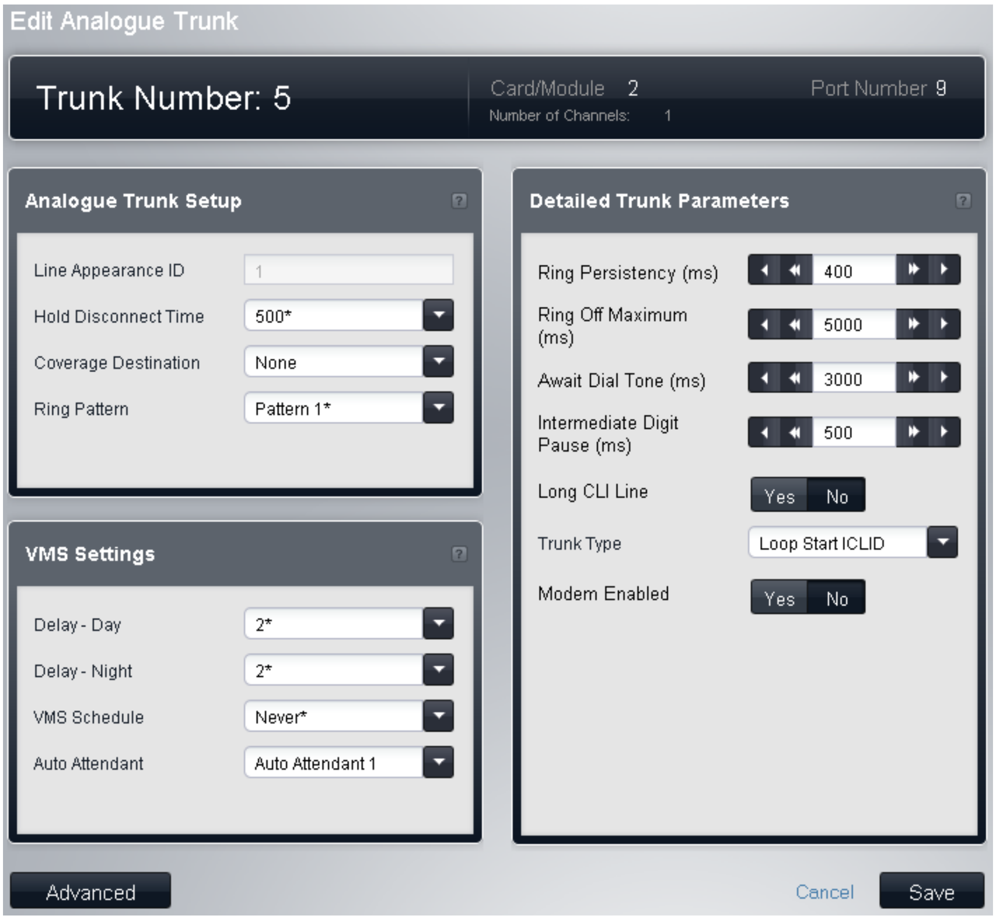
The menu is accessed by selecting the trunk in the trunks list and then clicking on the edit icon in the **Channel Setup** box on the right.

The menu that appears and the options in the menu will depend on the trunk type.

- **PRI Channel Setup** 83
- **T1 Channel Setup** 92
- **PRI (ETSI) Channel Setup** 90
- **BRI Trunk Channel Setup** 81

3.6.2.1 Analog Trunks Details

This menu is accessed by clicking the **View Details** link adjacent to a trunk in the **Installed Trunks** 73 table.



Analog Trunk Setup

- **ID/Line Appearance ID:** *Default = Auto-assigned*
  This number is used to uniquely identify the telephone line or channel. The number can be assigned to programmable buttons on extensions to allow the users to make and answer calls on that line or channel.

- **Hold Disconnect Time:** *Default = 500ms.*
  Also known as Disconnect Clear or Reliable Disconnect. This is a method used by the analog line provider to signal that the call has ended.
  - **! WARNING - Reboot Required**
    Changing this setting requires the system to be rebooted 12 for the change to take effect. Rebooting the system will end all calls currently in progress.

- **Coverage Destination:** *Default = None.*
  This option sets where incoming calls should alert in addition to alerting on those extension that have a line appearance button programmed for the line. When the phone system is in night service mode, calls alert at the members of the **Night Service** group.

  - **None**
    If set to ***None***, incoming calls will only alert on user extensions with line appearance buttons that match the line's **Appearance ID**.
  - **Extension**
    Route incoming calls to a particular extension.
  - **Phantom Extension**
    A phantom extension can be selected as the destination for calls.
  - **Hunt Group**
    Incoming calls can be routed to one of the 6 rotary hunt groups.
  - **Calling Group**
    For systems with their **Mode** set to ***PBX***, incoming calls can be routed to one of these 4 ring all groups.
  - **Operator Group**
    For systems with their **Mode** set to ***PBX***, incoming calls are routed to the **Operator Group**.

- **Voicemail**
  Route incoming calls to the systems voicemail to collect messages. This requires the caller to know the mailbox number and passcode.

- **Ring Pattern:** *Default = 1.*
  Selects the ring pattern that should be used for calls when alerting on an extension. Calls forwarded, sent to call coverage or to a hunt group will always use the line ring pattern. Calls direct to an extension will use the line ringing pattern unless the user has **Override Line Ringing** set. Not used for calls presented to the user as a member of the Operator group. This feature is also not used for BST phones.

VMS Settings

These settings are used to control when and how quickly the system will use one of its auto attendants to answer a currently unanswered call.

- **! WARNING - Reboot Required**
  Changing any of these settings requires the system to be rebooted 12 for the changes to take effect. Rebooting the system will end all calls currently in progress.

- **Delay - Day:** *Default = 2. Range = 0 to 6 (number of rings).*
  Set the number of rings before an unanswered call should be redirected the selected auto attendant when the system is not running in night service mode and the **VMS Schedule** is set to ***Always*** or ***Days Only***.

- **Delay - Night:** *Default = 2. Range = 0 to 6 (number of rings).*
  Sets the number of rings before an unanswered call should be redirected to the selected auto attendant when the system is running in night service mode and the **VMS Schedule** is set to ***Always*** or ***Night Only***.

- **Schedule:** *Default = Never.*
  This option determines when the **VMS Delay** settings above should be used and unanswered calls redirected to the selected auto attendant. The options are:

  - ***Always***
    Redirect calls when the system is in both day and night service modes.

  - ***Day Only***
    Redirect calls only when the system is not in night service.

  - ***Night Only***
    Redirect calls only when the system is in night service.

  - ***Never***
    Do not redirect calls.

- **Auto Attendant:** *Default = Auto Attendant 1.*
  This field allows selection of which auto attendant is used.

Detailed Trunk Parameters

- **! WARNING - Reboot Required**
  Changing any of these settings requires the system to be rebooted 12 for the changes to take effect. Rebooting the system will end all calls currently in progress.

- **Ring Persistency:** *Default = 400ms. Range = 0 to 2550ms.*
  The minimum duration of signal required to be recognized.

- **Ring Off Maximum:** *Default = 6000ms. Range = 0 to 25500ms.*
  The time before signaling is regarded as ended.

- **Await Dial Tone:** *Default = 3000ms. Range = 0 to 25500ms.*
  Sets how long the system should wait before dialing out.

- **Intermediate Digit Pause:** *Default = 500ms. Range = 0 to 2550ms.*
  Pause between digits transmitted to the line.

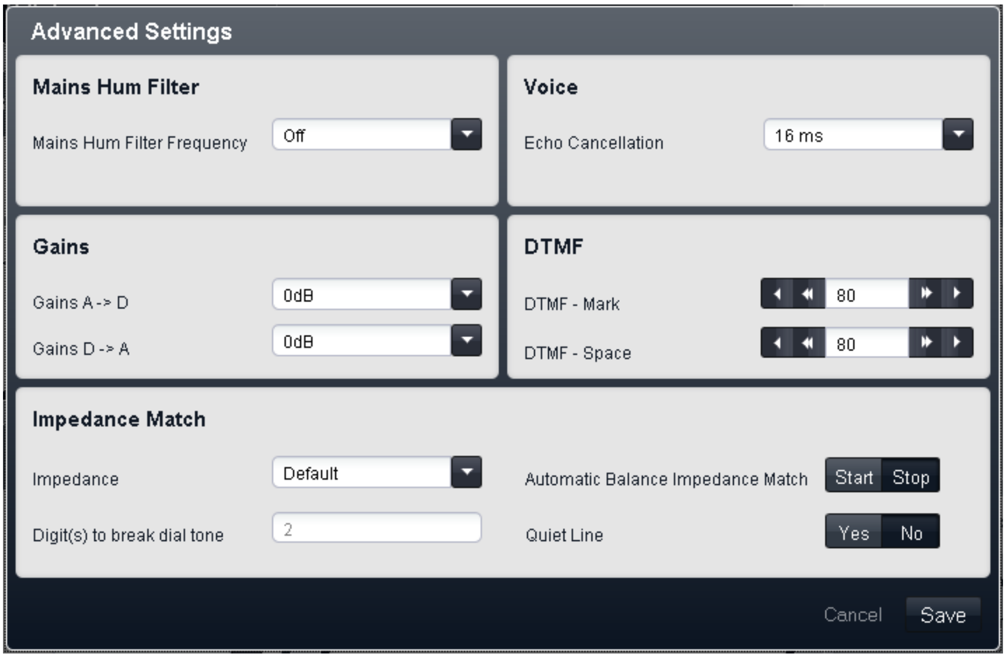
- **Long CLI Line:** *Default = Off*
  The CLI signal on some long analog lines can become degraded and is not then correctly detected. If you are sure that CLI is being provided but not detected, selecting this option may resolve the problem.

- **Trunk Type:** *Default = Loop Start ICLID*
  Indicates whether the trunk receives incoming caller ID information or not. If caller ID information is not provided, select ***Loop Start***. If caller ID information is received, select ***Loop Start ICLID***.

- **Modem Enabled:** *Default = Off*
  The first analog trunk can be set to modem operation (V32 with V42 error correction). This allows the trunk to answer incoming modem calls and be used for system maintenance. When on, the trunk can only be used for analog modem calls. The short code *9000* can be used to toggle this setting. The modem can be accessed via an auto attendant or DID/SIP URI by selecting 76 as the destination.

Advanced

This menu is accessed by clicking the **Advanced** button on the analog trunk settings 77 menu.



- **! WARNING - Reboot Required**
  Changing any of these settings requires the system to be rebooted 12 for the changes to take effect. Rebooting the system will end all calls currently in progress.

Mains Hum Filter

- **Mains Hum Filter:** *Default = Off.*
  If mains hum interference on the lines is detected or suspected, this settings can be used to attempt to remove that interference. The options are ***Off***, ***50Hz*** or ***60Hz***.

Voice

- **Echo Cancellation:** *Default = 16ms.*
  Allows settings of ***Off***, ***8***, ***16***, ***32***, ***64*** and ***128*** milliseconds. The echo cancellation should only be adjusted as high as required to remove echo problems. Setting it to a higher value than necessary can cause other distortions.

- **Echo Reduction:** *Default = On. (ATM4Uv2 card only)*
  Used when impedance matching is not required but echo reduction is. Options are On or Off.

Gains

These settings should not be adjusted without guidance from the line provider.

- **A -> D:** *Default = 0dB. Range = -10.0dB to +6.0dB in 0.5dB steps.*
  Sets the analog to digital gain.

- **D -> A:** *Default = 0dB. Range = -10.0dB to +6.0dB in 0.5dB steps.*
  Sets the digital to analog gain.

DTMF

- **DTMF Mark:** *Default = 80 (80ms). Range = 0 to 255.*
  Interval when DTMF signal is kept active during transmission of DTMF signals.

- **DTMF Space:** *Default = 80 (80ms). Range = 0 to 255.*
  Interval of silence between DTMF signal transmissions.

Impedance Match

- **Impedance:** *Default = Default*
  Set the impedance used for the line. The settings vary depending on the system's **Country** setting. These options are only available for ***Bahrain***, ***Egypt***, ***Kuwait***, ***Morocco***, ***Oman***, ***Pakistan***, ***Qatar***, ***Saudi Arabia***, ***South Africa***, ***Turkey***, ***United Arab Emirates*** *and* ***United States***. For Release 8.1+ they are also available for ***Canada***.

- **Automatic:** *Default = Yes. (ATM4Uv2 card only)*
  When set to ***Yes***, the impedance is set to the default for the system Locale. When set to ***No***, the **Impedance** value can be manually set.

- **Digits to break dial tone:** *Default = 2. Range = Up to 3 digits.*
  During impedance testing, once the system has seized a line, it dials this digit or digits to the line. In some cases it may be necessary to use a different digit or digits. For example, if analog trunk go via another PBX system or Centrex, it will be necessary to use the external trunk dialing prefix of the remote system plus another digit, for example 92.

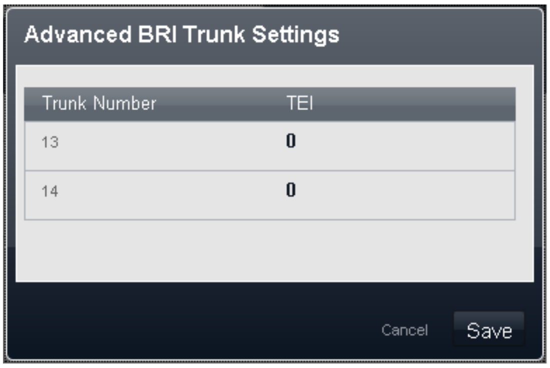
- **Automatic Balance Impedance Match:**
  These controls can be used to test the impedance of a line and to then display the best match resulting from the test. Testing should be performed with the line connected but the phone system otherwise idle. To start testing click **Start**. The phone system will then send a series of signals to the line and monitor the response, repeating this at each possible impedance setting. Testing can be stopped at any time by clicking **Stop**. When testing is complete, Manager displays the best match and asks whether that match should be used for the line. If ***Yes*** is selected, Manager asks whether the match should be applied to all other analog lines provided by the same analog trunk card or module. To conform with the Receive Objective Loudness Rating at distances greater than 2.7km from the central office, on the analogue trunks a receive gain of 1.5 db needs to be added.

- **Quiet Line:** *Default = Off*
  This setting may be required to compensate for signal loss on long lines.

3.6.2.2 BRI Trunk Details

This menu is accessed by clicking the **View Details** link adjacent to a trunk in the **Installed Trunks** 73 table.

- **! WARNING - Reboot Required**
  Changing any of these settings requires the system to be rebooted 12 for the changes to take effect. Rebooting the system will end all calls currently in progress.

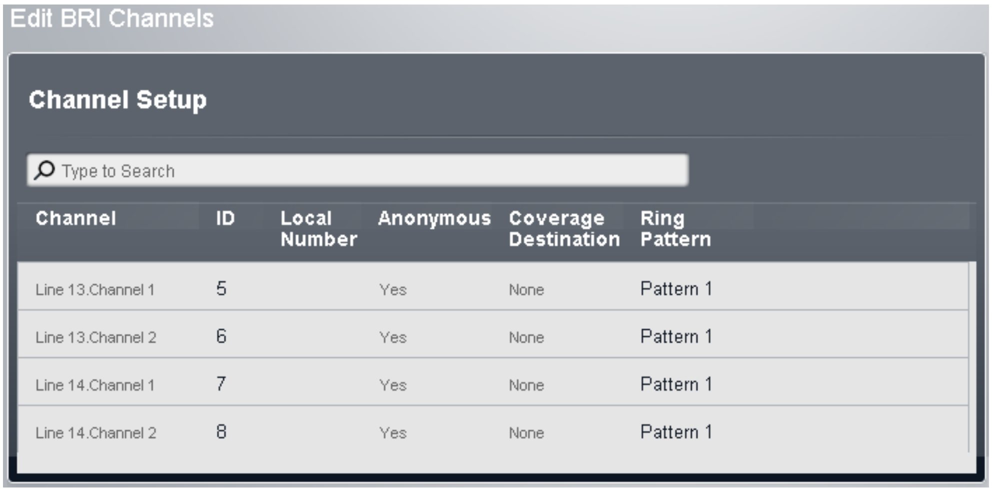


- **Trunk Number**
  The BRI line number. For information only

- **TEI:** *Default = 0*
  This is the Terminal Equipment Identifier number associated with the line. It is used to identify each device connected to a particular ISDN line. For Point-to-Point lines this is 0. It can also be 0 on a Point to Multipoint line, however if multiple devices are sharing a Point-to-Multipoint line it should be set to 127 which results in the exchange allocating the TEIs to be used by each device.

BRI Channel Setup

This menu is accessed by selecting the trunk in the **Installed Trunks** 73 table and then clicking on the edit icon in the **Channel Setup** panel on the right.



- **! WARNING - Reboot Required**
  Changing any of these settings requires the system to be rebooted 12 for the changes to take effect. Rebooting the system will end all calls currently in progress.

Channel Setup

- **No:** *For information only, not editable.*

- **ID/Line Appearance ID:** *Default = Auto-assigned*
  This number is used to uniquely identify the telephone line or channel. The number can be assigned to programmable buttons on extensions to allow the users to make and answer calls on that line or channel.

- **Local Number**
  Information only. Use to store an associated number for test calls to the line.

- **Anonymous:** *Default = No*
  If selected, withhold sending caller ID information on outgoing calls. For system's with their **Mode** set to ***PBX***, this setting can be overridden by the ARS selector used to route the outgoing call.

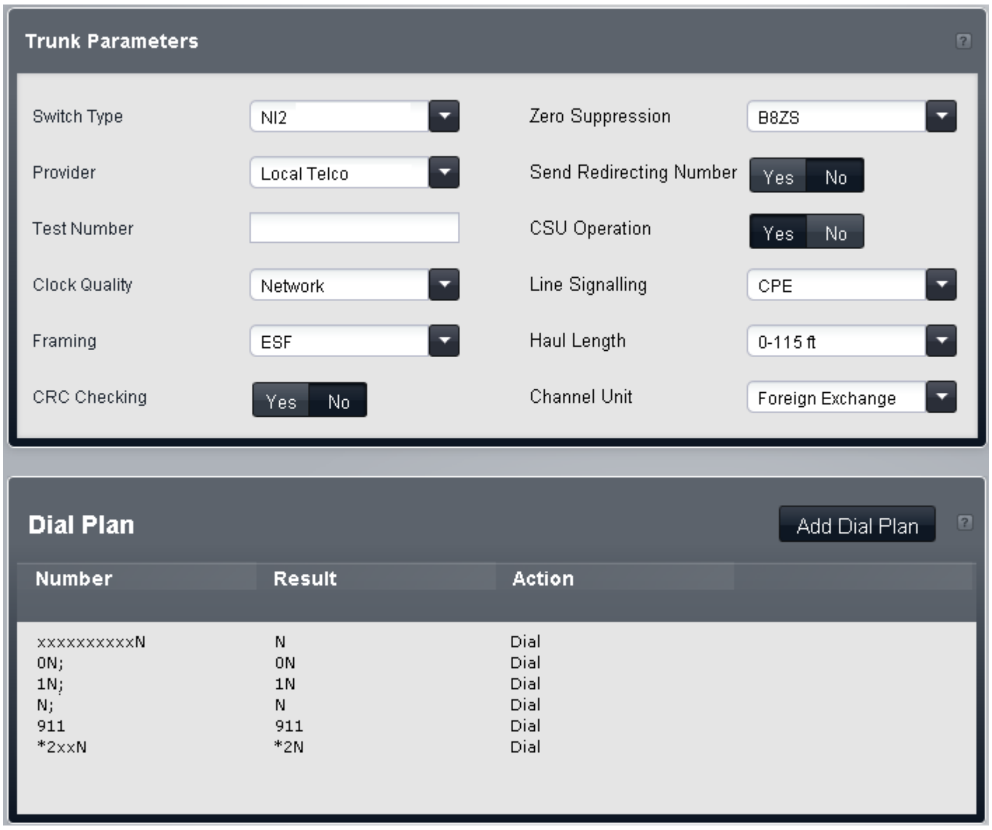
- **Coverage Destination:** *Default = None.*
  This option sets where incoming calls should alert in addition to alerting on those extension that have a line appearance button programmed for the line. When the phone system is in night service mode, calls alert at the members of the **Night Service** group.

  - **None**
    If set to ***None***, incoming calls will only alert on user extensions with line appearance buttons that match the line's **Appearance ID**.
  - **Extension**
    Route incoming calls to a particular extension.
  - **Phantom Extension**
    A phantom extension can be selected as the destination for calls.
  - **Hunt Group**
    Incoming calls can be routed to one of the 6 rotary hunt groups.
  - **Calling Group**
    For systems with their **Mode** set to ***PBX***, incoming calls can be routed to one of these 4 ring all groups.
  - **Operator Group**
    For systems with their **Mode** set to ***PBX***, incoming calls are routed to the **Operator Group**.
  - **Voicemail**
    Route incoming calls to the systems voicemail to collect messages. This requires the caller to know the mailbox number and passcode.

- **Ring Pattern:** *Default = 1.*
  Selects the ring pattern that should be used for calls when alerting on an extension. Calls forwarded, sent to call coverage or to a hunt group will always use the line ring pattern. Calls direct to an extension will use the line ringing pattern unless the user has **Override Line Ringing** set. Not used for calls presented to the user as a member of the Operator group. This feature is also not used for BST phones.

3.6.2.3 PRI Trunk Details

This menu is accessed by clicking the **View Details** link adjacent to a trunk in the **Installed Trunks** 73 table.



- **! WARNING - Reboot Required**
  Changing any of these settings requires the system to be rebooted 121 for the changes to take effect. Rebooting the system will end all calls currently in progress.

Trunk Parameters

- **Switch Type:** *Default = NI2*
  Options ***4ESS***, ***5ESS***, ***DMS100*** and ***NI2***.
- **Provider:** *Default = Local Telco*
  Select the PSTN service provider (***AT&T***, ***Sprint***, ***WorldCom*** or ***Local Telco***). When set to AT&T, an additional AT & T Provider Setup menu can be accessed using a button below the menu.
- **Test Number:**
  Used to remember the external telephone number of this line to assist with loop-back testing. For information only.
- **Send Redirecting Number:** *Default = Off*
- **Clock Quality:** *Default = Network*
  Leave as ***Network*** unless advised otherwise by Avaya.
- **Framing:** *Default = ESF*
  Selects the type of signal framing used (***ESF*** or ***D4***).
- **CRC Checking:** *Default = On*
  Turns CRC on or off.
- **Zero Suppression:** *Default = B8ZS*
  Selects the method of zero suppression used (***B8ZS*** or ***AMI ZCS***).
- **CSU Operation:**
  Tick this field to enable the T1 line to respond to loop-back requests from the line.
- **Line Signaling:** *Default = CPE*
  The field can be set to either ***CPE*** (Customer Premises Equipment) or ***CO*** (Central Office). This field should normally be left at its default of CPE. The setting CO is normally only used in lab back-to-back testing.
- **Haul Length:** *Default = 0-115 feet*
  Sets the line length to a specific distance.

- **Channel Unit:** Default = Foreign Exchange
  This field should be set to match the channel signaling equipment provided by the Central Office. The options are ***Foreign Exchange***, ***Special Access*** or ***Normal***.

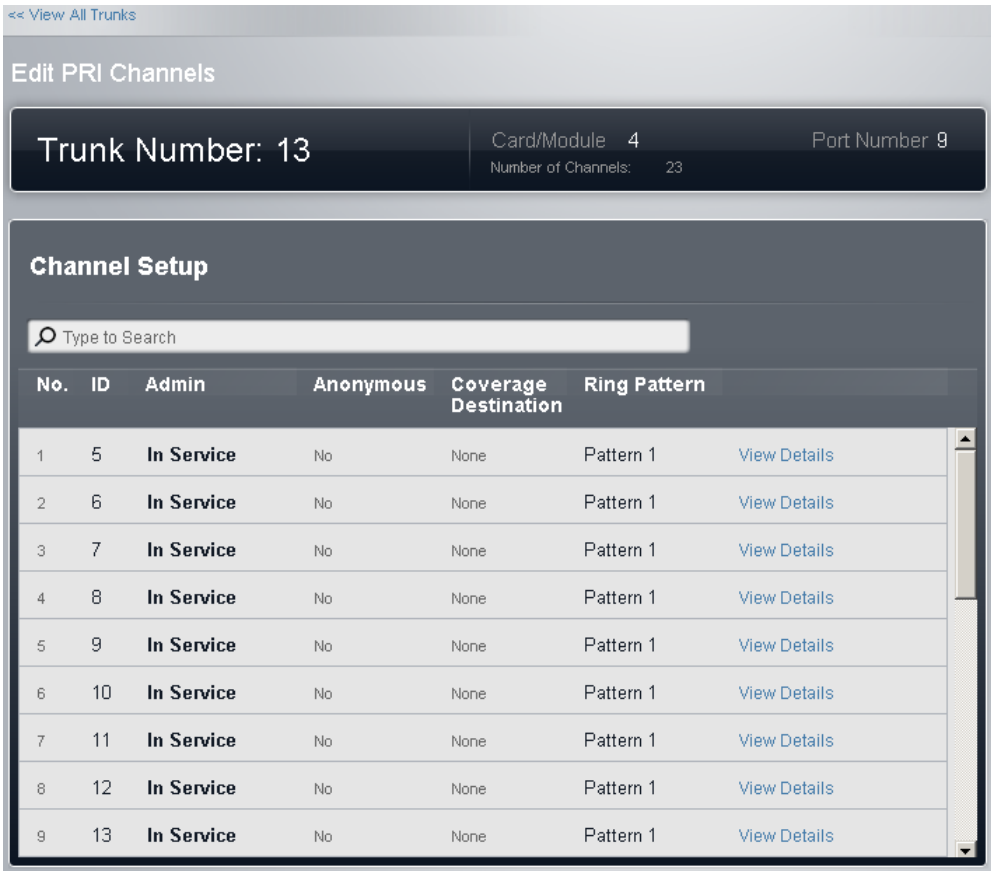
Dial Plan

The dial plan is used to apply number translations to the digits received by the line for output to the line provider and to indicate any special service required from the line provider, for example to withhold the call ID. The default dial plan is as shown below.

Dialled Number	Result	Action
xxxxxxxxxxN	N	Dial
0N;	0N	Dial
1N;	1N	Dial
N;	N	Dial
911	911	Dial
***2xxN**	*2N	Dial
***3xxN**	*3N	Dial
***xxN**	*N	Dial
***65**		Explicitly not Anonymous
***67**		Call Anonymously

PRI Channels

This menu is accessed by selecting the trunk in the **Installed Trunks** 73 table and then clicking on the edit icon in the **Channel Setup** panel on the right.



- **! WARNING - Reboot Required**
  Changing any of these settings requires the system to be rebooted 121 for the changes to take effect. Rebooting the system will end all calls currently in progress.

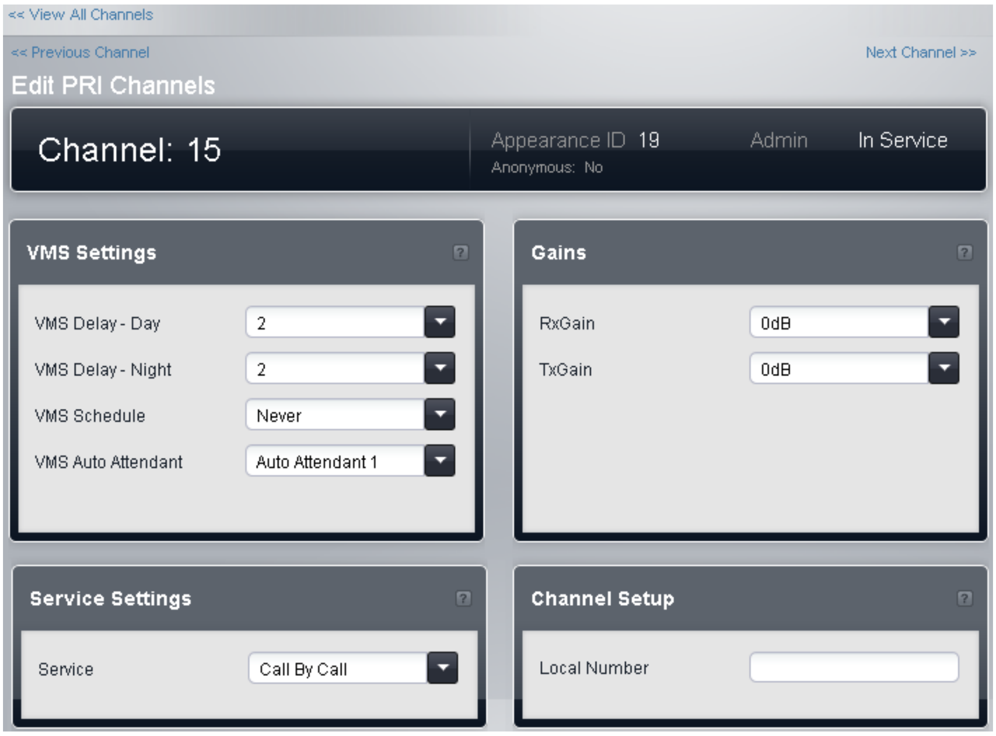
- **No:** *For information only, not editable.*

- **ID/Line Appearance ID:** *Default = Auto-assigned*
  This number is used to uniquely identify the telephone line or channel. The number can be assigned to programmable buttons on extensions to allow the users to make and answer calls on that line or channel.

- **Admin:** *Default = In Service*
  Options are ***In Service***, ***DID Only***, ***Maintenance*** and ***Out of Service***.
- **Anonymous:** *Default = No*
  If selected, withhold sending caller ID information on outgoing calls. For system's with their **Mode** set to ***PBX***, this setting can be overridden by the ARS selector used to route the outgoing call.
- **Coverage Destination:** *Default = None.*
  This option sets where incoming calls should alert in addition to alerting on those extension that have a line appearance button programmed for the line. When the phone system is in night service mode, calls alert at the members of the **Night Service** group.
  - **None**
    If set to ***None***, incoming calls will only alert on user extensions with line appearance buttons that match the line's **Appearance ID**.
  - **Extension**
    Route incoming calls to a particular extension.
  - **Phantom Extension**
    A phantom extension can be selected as the destination for calls.
  - **Hunt Group**
    Incoming calls can be routed to one of the 6 rotary hunt groups.
  - **Calling Group**
    For systems with their **Mode** set to ***PBX***, incoming calls can be routed to one of these 4 ring all groups.
  - **Operator Group**
    For systems with their **Mode** set to ***PBX***, incoming calls are routed to the **Operator Group**.
  - **Voicemail**
    Route incoming calls to the systems voicemail to collect messages. This requires the caller to know the mailbox number and passcode.
- **Ring Pattern:** *Default = 1.*
  Selects the ring pattern that should be used for calls when alerting on an extension. Calls forwarded, sent to call coverage or to a hunt group will always use the line ring pattern. Calls direct to an extension will use the line ringing pattern unless the user has **Override Line Ringing** set. Not used for calls presented to the user as a member of the Operator group. This feature is also not used for BST phones.

PRI Channel Setup

This menu is accessed by selecting the trunk in the **Installed Trunks** 73 table and then clicking on the edit icon in the **Channel Setup** panel on the right. Select the required channel and then click on **View Details**.



VMS Settings

These settings are used to control when and how quickly the system will use one of its auto attendants to answer a currently unanswered call.

- **! WARNING - Reboot Required**
  Changing any of these settings requires the system to be rebooted 121 for the changes to take effect. Rebooting the system will end all calls currently in progress.

- **Delay - Day:** *Default = 2. Range = 0 to 6 (number of rings).*
  Set the number of rings before an unanswered call should be redirected the selected auto attendant when the system is not running in night service mode and the **VMS Schedule** is set to ***Always*** or ***Days Only***.

- **Delay - Night:** *Default = 2. Range = 0 to 6 (number of rings).*
  Sets the number of rings before an unanswered call should be redirected to the selected auto attendant when the system is running in night service mode and the **VMS Schedule** is set to ***Always*** or ***Night Only***.

- **Schedule:** *Default = Never.*
  This option determines when the **VMS Delay** settings above should be used and unanswered calls redirected to the selected auto attendant. The options are:

  - ***Always***
    Redirect calls when the system is in both day and night service modes.

  - ***Day Only***
    Redirect calls only when the system is not in night service.

  - ***Night Only***
    Redirect calls only when the system is in night service.

  - ***Never***
    Do not redirect calls.

- **Auto Attendant:** *Default = Auto Attendant 1.*
  This field allows selection of which auto attendant is used.

Gains

These settings are used to adjust the signal received and sent by the system.

- **Tx Gain:** *Default = 0dB*
  Sets the transmit gain applied to the outgoing signal sent from the system.

- **Rx Gain:** *Default = 0dB*
  Sets the receive gain applied to the incoming signal received by the system.

Service Settings

This panel is usable for trunks where the **Provider** 83 is set to ***AT&T***. It allows selection of the service provided on the channel.

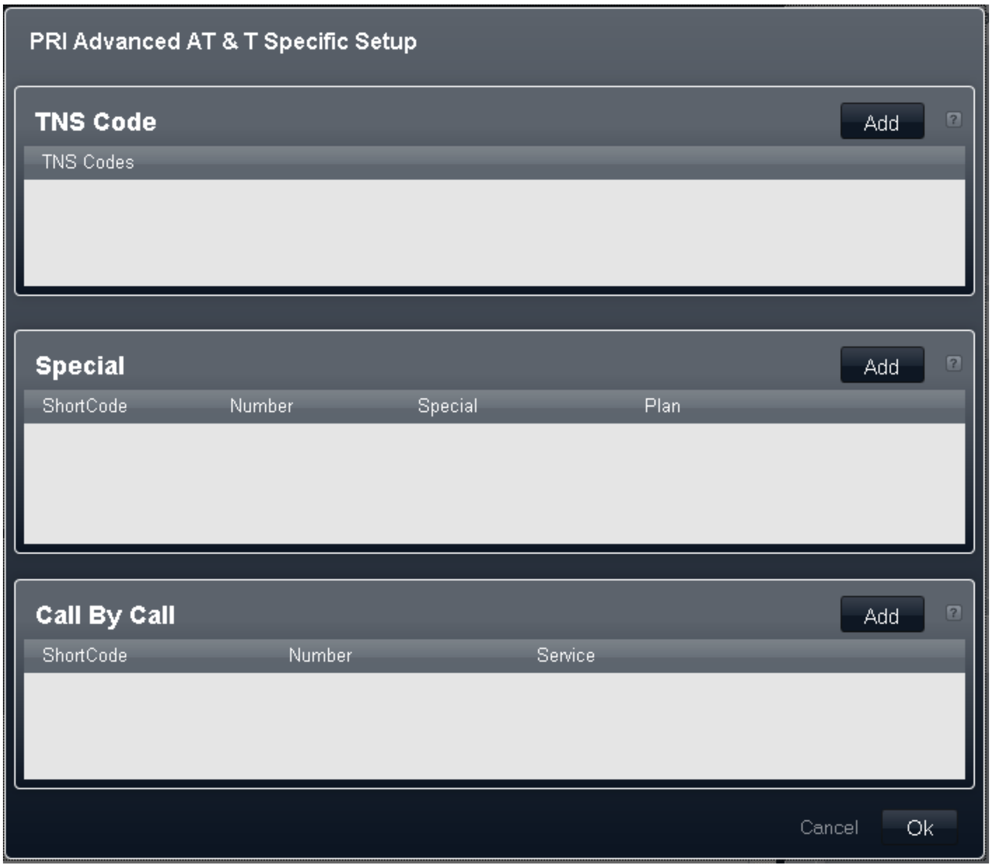
- **Service**
  The service required by the call from ***SDN (inc GSDN)***, ***MegaCom800***, ***MegaCom***, ***Wats***, ***Accunet***, ***ILDS***, ***I800***, ***ETN***, ***Private Line*** or ***AT&T Multiquest***. The option ***Call by Call*** sets the channel to use the service that matches the dialed number, as set in the trunk's **Call by Call** settings.

Channel Setup

- **Local Number**
  Information only. Use to store an associated number for test calls to the line.

PRI Advanced AT&T Specific Setup

These settings are only available for a PRI trunk which has its **Provider** 83 setting set to ***AT&T***. This menu is accessed by clicking the **View Details** link adjacent to a trunk in the **Installed Trunks** 73 table and then clicking the **AT&T Provider Setup** button at the bottom of the screen.



TNS Code

- **TNS Codes**
  This table is used to set the TNS (Transit Network Selection) information element for 4ESS and 5ESS exchanges. It is also used to set fields in the NSF information element. These are prefixes for alternative long distance carriers. When a number dialed matches an entry in the table, that pattern is stripped from the number before being sent out. For example, if the pattern 10XXX is added to this tab, when 10288 is dialed, the 10 is removed and 288 is placed in the calls TNS and NSF information fields.

Special

- **Short code:**
  The number dialed by the user.
- **Number:**
  The number to be dialed to line.
- **Special:** *Default = No Operator*
  The available options are ***No Operator***, ***Local Operator*** or ***Presubscribed Operator***.
- **Plan:** *Default = National*
  The available options are ***National*** or ***International***.

An example set of settings would be:

Short Code	Number	Special	Plan
011N	N	No Operator	International
010N	N	Local Operator	International
01N	N	Local Operator	National
00N	N	Presubscribed Operator	National
0N	N	Presubscribed Operator	National
1N	1N	No operator	National

Call By Call

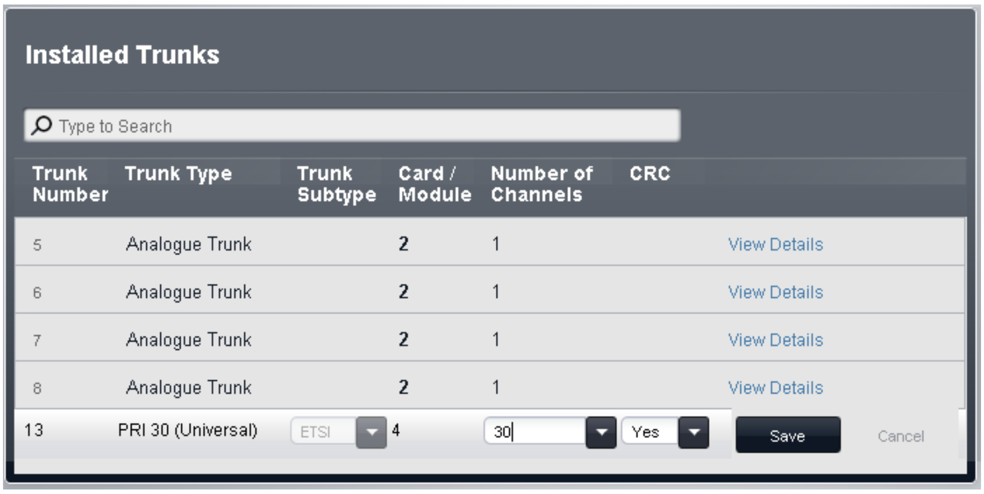
Settings in this tab are only used when calls are routed via a channel which has its **Service** set to ***Call by Call***. It allows short codes to be created to route calls to a different services according to the number dialed. Call By Call reduces the costs and maximizes the use of facilities. Call By Call chooses the optimal service for a particular call by including the Bearer capability in the routing decision. This is particularly useful when there are limited resources.

- **Short Code:**
  The number dialed.
- **Number:**
  The number to be dialed to line.
- **Service:** *Default = AT&T*
  The service required by the call from ***SDN (inc GSDN)***, ***MegaCom800***, ***MegaCom***, ***Wats***, ***Accunet***, ***ILDS***, ***I800***, ***ETN***, ***Private Line*** or ***AT&T Multiquest***.

3.6.2.4 PRI Trunk (ESTI)

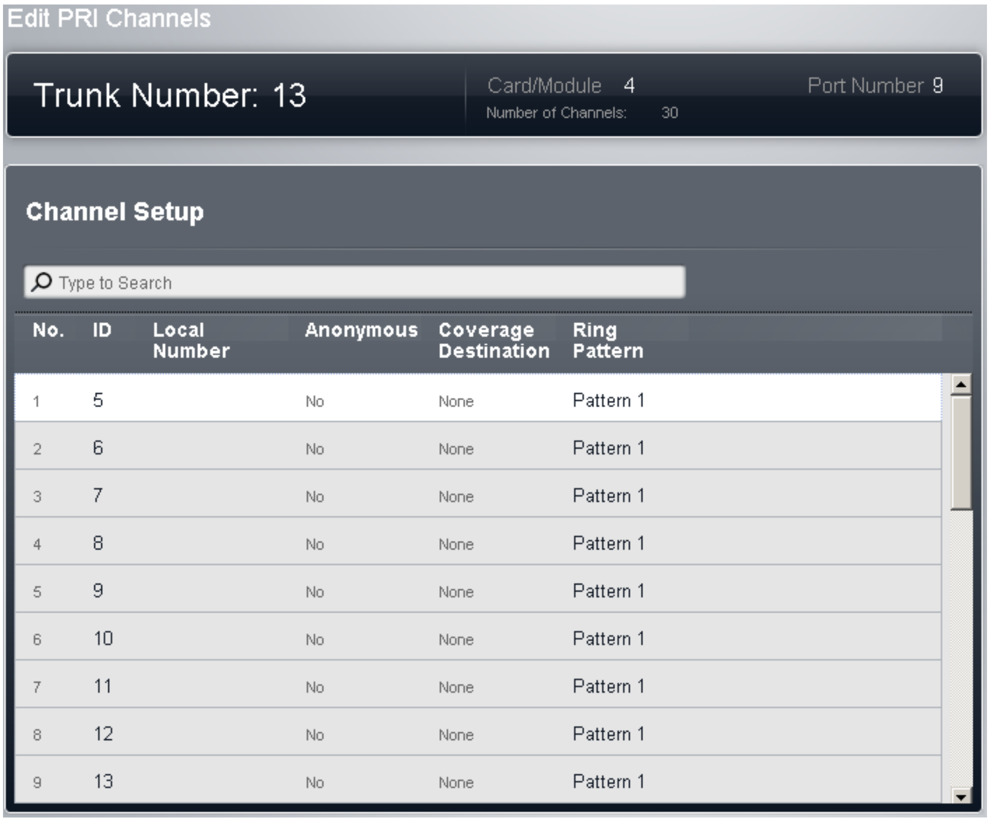
The settings for an ESTI PRI trunk can be changed by double clicking on the trunk entry in the **Installed Trunks** 73 table. Adjust the values are required and then click on **Save**.

- **! WARNING - Reboot Required**
  Changing any of these settings requires the system to be rebooted 12 for the changes to take effect. Rebooting the system will end all calls currently in progress.



Channel Setup

This menu is accessed by selecting the trunk in the **Installed Trunks** 73 table and then clicking on the edit icon in the **Channel Setup** panel on the right.



- **! WARNING - Reboot Required**
  Changing any of these settings requires the system to be rebooted 12 for the changes to take effect. Rebooting the system will end all calls currently in progress.

- **No:** *For information only, not editable.*
- **ID/Line Appearance ID:** *Default = Auto-assigned*
  This number is used to uniquely identify the telephone line or channel. The number can be assigned to programmable buttons on extensions to allow the users to make and answer calls on that line or channel.

- **Local Number**
  Information only. Use to store an associated number for test calls to the line.

- **Anonymous:** *Default = No*
  If selected, withhold sending caller ID information on outgoing calls. For system's with their **Mode** set to ***PBX***, this setting can be overridden by the ARS selector used to route the outgoing call.

- **Coverage Destination:** *Default = None.*
  This option sets where incoming calls should alert in addition to alerting on those extension that have a line appearance button programmed for the line. When the phone system is in night service mode, calls alert at the members of the **Night Service** group.

  - **None**
    If set to ***None***, incoming calls will only alert on user extensions with line appearance buttons that match the line's **Appearance ID**.

  - **Extension**
    Route incoming calls to a particular extension.

  - **Phantom Extension**
    A phantom extension can be selected as the destination for calls.

  - **Hunt Group**
    Incoming calls can be routed to one of the 6 rotary hunt groups.

  - **Calling Group**
    For systems with their **Mode** set to ***PBX***, incoming calls can be routed to one of these 4 ring all groups.

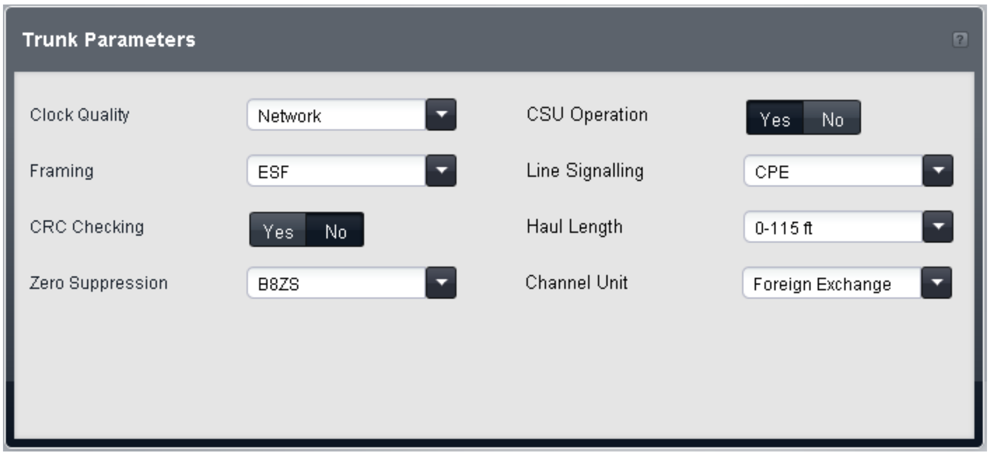
  - **Operator Group**
    For systems with their **Mode** set to ***PBX***, incoming calls are routed to the **Operator Group**.

  - **Voicemail**
    Route incoming calls to the systems voicemail to collect messages. This requires the caller to know the mailbox number and passcode.

- **Ring Pattern:** *Default = 1.*
  Selects the ring pattern that should be used for calls when alerting on an extension. Calls forwarded, sent to call coverage or to a hunt group will always use the line ring pattern. Calls direct to an extension will use the line ringing pattern unless the user has **Override Line Ringing** set. Not used for calls presented to the user as a member of the Operator group. This feature is also not used for BST phones.

3.6.2.5 T1 Trunk Details

This menu is accessed by clicking the **View Details** link adjacent to a trunk in the **Installed Trunks** 73 table.

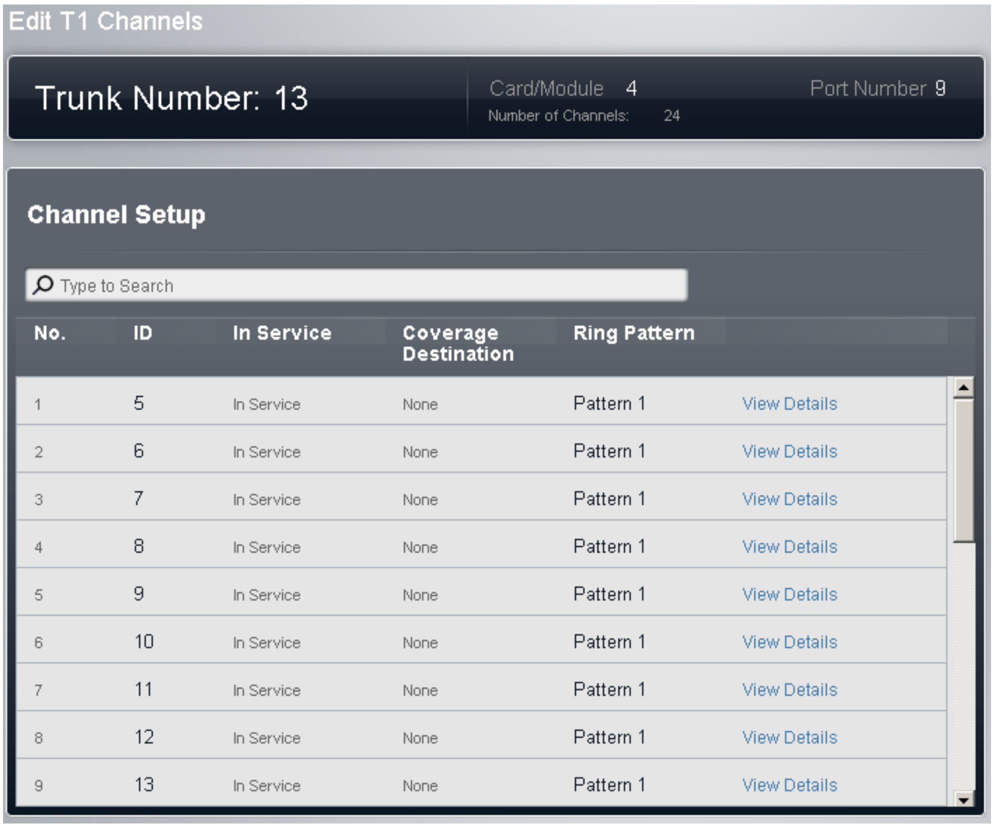


- **! WARNING - Reboot Required**
  Changing any of these settings requires the system to be rebooted 121 for the changes to take effect. Rebooting the system will end all calls currently in progress.

- **Clock Quality:** *Default = Network*
  Leave as ***Network*** unless advised otherwise by Avaya.
- **Framing:** *Default = ESF*
  Selects the type of signal framing used (***ESF*** or ***D4***).
- **CRC Checking:** *Default = On*
  Turns CRC on or off.
- **Zero Suppression:** *Default = B8ZS*
  Selects the method of zero suppression used (***B8ZS*** or ***AMI ZCS***).
- **CSU Operation:**
  Tick this field to enable the T1 line to respond to loop-back requests from the line.
- **Line Signaling:** *Default = CPE*
  The field can be set to either ***CPE*** (Customer Premises Equipment) or ***CO*** (Central Office). This field should normally be left at its default of CPE. The setting CO is normally only used in lab back-to-back testing.
- **Haul Length:** *Default = 0-115 feet*
  Sets the line length to a specific distance.
- **Channel Unit:** Default = Foreign Exchange
  This field should be set to match the channel signaling equipment provided by the Central Office. The options are ***Foreign Exchange***, ***Special Access*** or ***Normal***.

T1 Channels

This menu is accessed by selecting the trunk in the **Installed Trunks** 73 table and then clicking on the edit icon in the **Channel Setup** panel on the right.



- **! WARNING - Reboot Required**
  Changing any of these settings requires the system to be rebooted [121] for the changes to take effect. Rebooting the system will end all calls currently in progress.

- **No:** *For information only, not editable.*

- **ID/Line Appearance ID:** *Default = Auto-assigned*
  This number is used to uniquely identify the telephone line or channel. The number can be assigned to programmable buttons on extensions to allow the users to make and answer calls on that line or channel.

- **In Service:** *Default = Out of Service.*
  The setting can be used to sets whether the trunk channel is in use.

- **Coverage Destination:** *Default = None.*
  This option sets where incoming calls should alert in addition to alerting on those extension that have a line appearance button programmed for the line. When the phone system is in night service mode, calls alert at the members of the **Night Service** group.

  - **None**
    If set to ***None***, incoming calls will only alert on user extensions with line appearance buttons that match the line's **Appearance ID**.

  - **Extension**
    Route incoming calls to a particular extension.

  - **Phantom Extension**
    A phantom extension can be selected as the destination for calls.

  - **Hunt Group**
    Incoming calls can be routed to one of the 6 rotary hunt groups.

  - **Calling Group**
    For systems with their **Mode** set to ***PBX***, incoming calls can be routed to one of these 4 ring all groups.

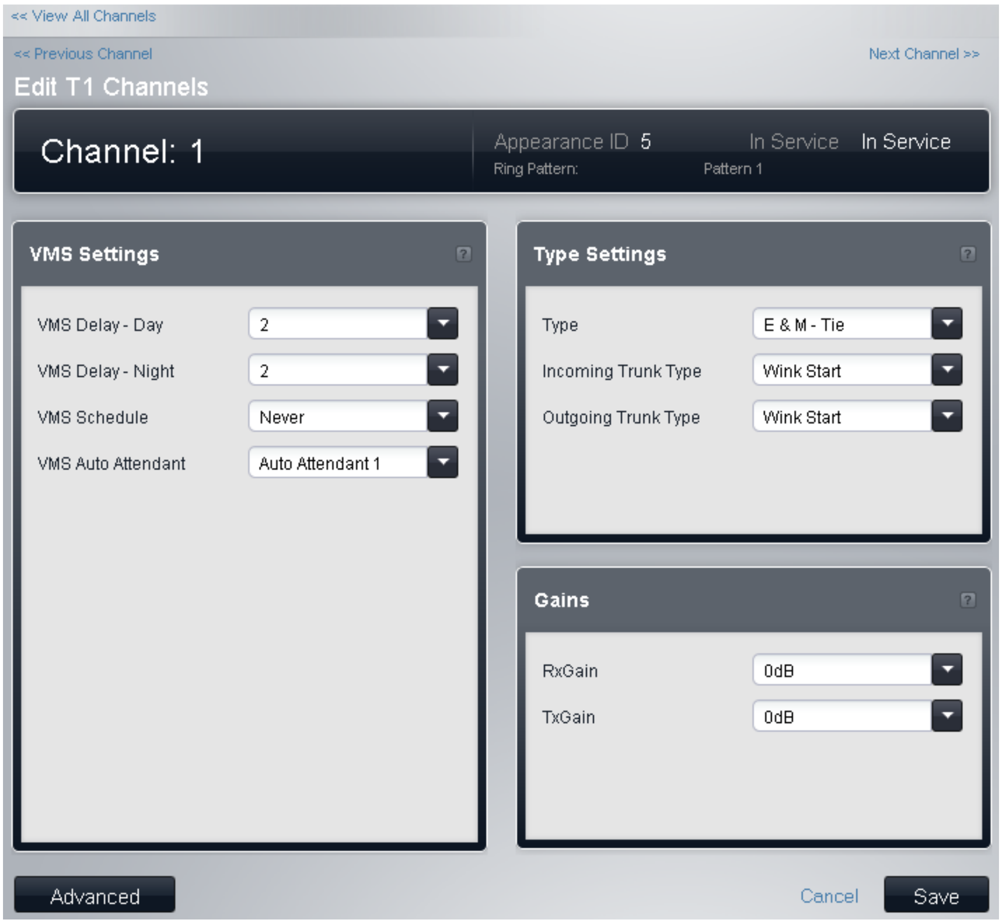
  - **Operator Group**
    For systems with their **Mode** set to ***PBX***, incoming calls are routed to the **Operator Group**.

  - **Voicemail**
    Route incoming calls to the systems voicemail to collect messages. This requires the caller to know the mailbox number and passcode.

* **Ring Pattern:** *Default = 1.*
  Selects the ring pattern that should be used for calls when alerting on an extension. Calls forwarded, sent to call coverage or to a hunt group will always use the line ring pattern. Calls direct to an extension will use the line ringing pattern unless the user has **Override Line Ringing** set. Not used for calls presented to the user as a member of the Operator group. This feature is also not used for BST phones.

T1 Channel Setup

This menu is accessed by selecting the trunk in the **Installed Trunks** [73] table and then clicking on the edit icon in the **Channel Setup** panel on the right. Select the required channel and then click on **View Details**.



- **! WARNING - Reboot Required**
  Changing any of these settings requires the system to be rebooted [12] for the changes to take effect. Rebooting the system will end all calls currently in progress.

VMS Settings

These settings are used to control when and how quickly the system will use one of its auto attendants to answer a currently unanswered call.

- **! WARNING - Reboot Required**
  Changing any of these settings requires the system to be rebooted [12] for the changes to take effect. Rebooting the system will end all calls currently in progress.

- **Delay - Day:** *Default = 2. Range = 0 to 6 (number of rings).*
  Set the number of rings before an unanswered call should be redirected the selected auto attendant when the system is not running in night service mode and the **VMS Schedule** is set to ***Always*** or ***Days Only***.

- **Delay - Night:** *Default = 2. Range = 0 to 6 (number of rings).*
  Sets the number of rings before an unanswered call should be redirected to the selected auto attendant when the system is running in night service mode and the **VMS Schedule** is set to ***Always*** or ***Night Only***.

- **Schedule:** *Default = Never.*
  This option determines when the **VMS Delay** settings above should be used and unanswered calls redirected to the selected auto attendant. The options are:

  - ***Always***
    Redirect calls when the system is in both day and night service modes.

  - ***Day Only***
    Redirect calls only when the system is not in night service.

  - ***Night Only***
    Redirect calls only when the system is in night service.

    - ***Never***
      Do not redirect calls.

  - **Auto Attendant:** *Default = Auto Attendant 1.*
    This field allows selection of which auto attendant is used.

Type Settings

  - **No:** *For information only, not editable.*

  - **ID/Line Appearance ID:** *Default = Auto-assigned*
    This number is used to uniquely identify the telephone line or channel. The number can be assigned to programmable buttons on extensions to allow the users to make and answer calls on that line or channel.

- **Type:** *Default = Out of Service*
  The T1 emulates the following connections (***Ground Start***, ***Loop Start***, ***E & M - TIE***, ***E & M - DID***, ***E & M Switched 56K***, ***Direct Inward Dial***, ***Clear Channel 64K*** or ***Out of Service***). Trunks set to ***E & M - DID*** will only accept incoming calls. If ***E&M - TIE*** is selected and the **Outgoing Trunk Type** is set to ***Automatic***, no secondary dial tone is provided for outgoing calls on this channel.

- **Incoming Trunk Type:** *Default = Wink-Start*
  Used for E&M types only. The handshake method for incoming calls (***Automatic***, ***Immediate***, ***Delay Dial*** or ***Wink-Start***).

- **Outgoing Trunk Type:** *Default = Wink-Start*
  Used for E&M types only. The handshake method for outgoing calls (***Automatic***, ***Immediate***, ***Delay Dial*** or ***Wink-Start***).

Gains

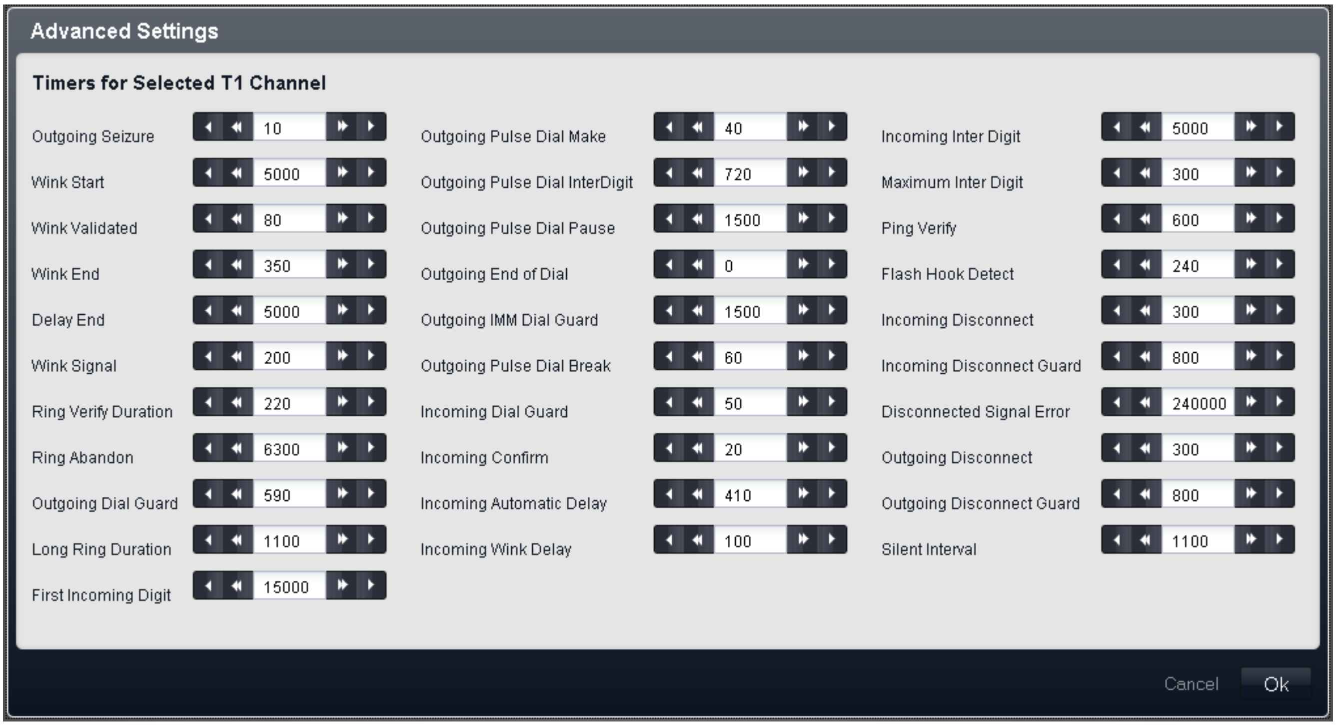
These settings are used to adjust the signal received and sent by the system.

  - **Tx Gain:** *Default = 0dB*
    Sets the transmit gain applied to the outgoing signal sent from the system.

  - **Rx Gain:** *Default = 0dB*
    Sets the receive gain applied to the incoming signal received by the system.

T1 Advanced Channel Setup

This menu is accessed by clicking the **Advanced** button on the channel setup menu. Only adjust these values under guidance from the line provider.

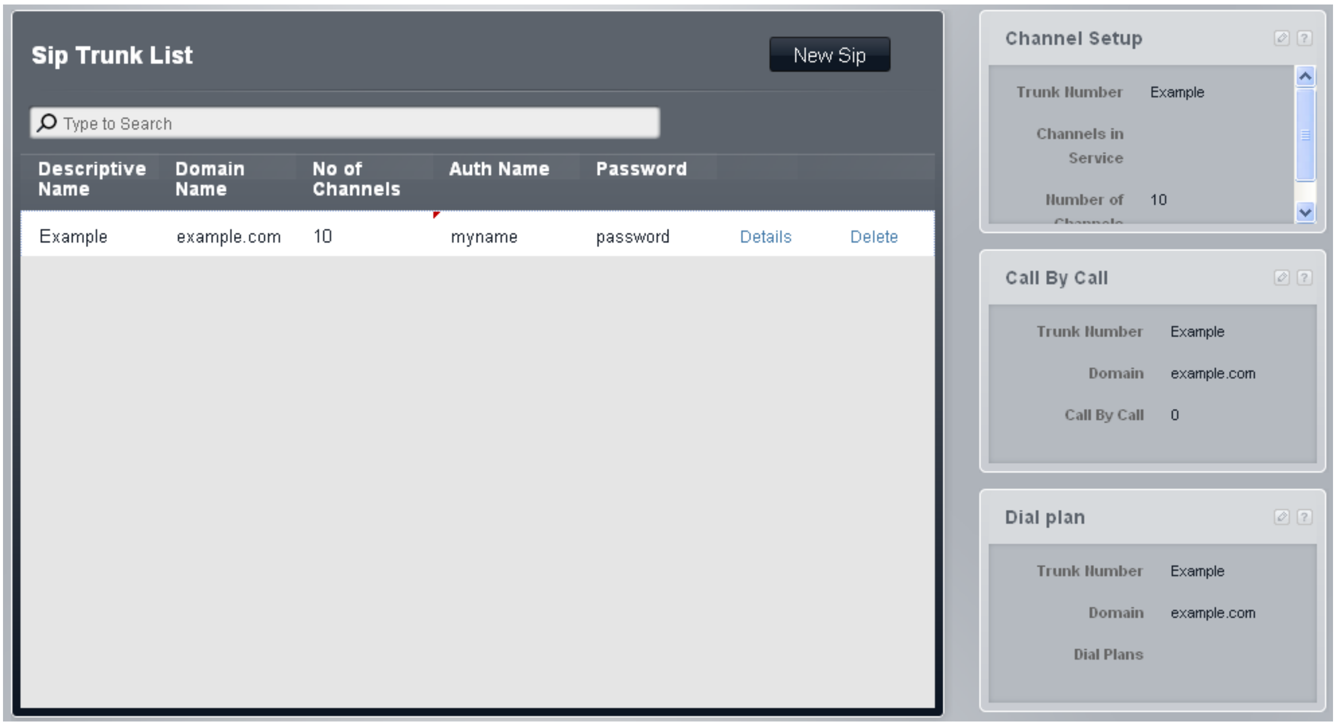


- **! WARNING - Reboot Required**
  Changing any of these settings requires the system to be rebooted 121 for the changes to take effect. Rebooting the system will end all calls currently in progress.

3.6.3 SIP Trunks

This menu is used to add SIP trunks to the phone system configuration. The menu is accessed by selecting **System** in the menu bar and clicking **SIP Trunks**.

- **! WARNING - Reboot Required**
  Adding or deleting any SIP trunk requires the system to be rebooted 12 for the changes to take effect. Rebooting the system will end all calls currently in progress.



SIP Trunk Pre-Requisites

Before adding any SIP trunks, the system must be configured to support SIP operation:

- **SIP Trunk Channel Licenses**
  The system can support 3 simultaneous SIP calls without needing licenses. Additional simultaneous calls, up to 20 in total, require the addition of licenses to the configuration.
  - **VCM Channels**
    Note that for SIP calls the system also requires VCM channels. For a system those are provided by installing IP500 Combination base cards. Each of these cards (up to 2) provides 10 VCM channels.
- **STUN Settings**
  The system's STUN settings need to be configured to allow it to connect to the Internet for SIP calls. This is done through the **STUN Settings for Network** panel of the system's **Advanced** 69 settings menu.

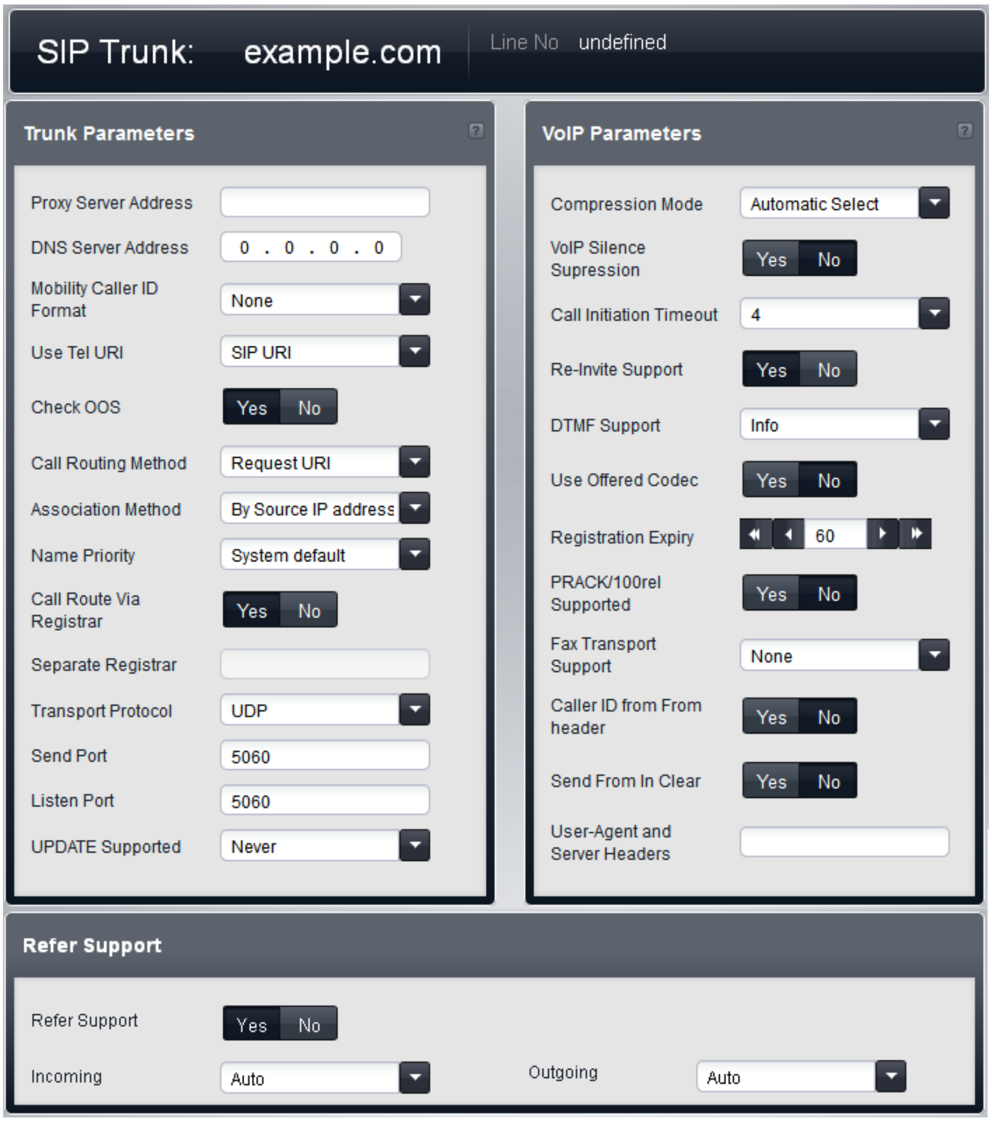
SIP Trunk List

- **! WARNING - Reboot Required**
  Changing any of these settings requires the system to be rebooted 12 for the changes to take effect. Rebooting the system will end all calls currently in progress.

- **Descriptive Name**
  A name for the trunk. This is affect the trunks operation.
- **Domain Name:** *Default = Blank*
  Each SIP Trunk configuration has a unique ITSP Domain name needed by SIP end points in order to register with the IP Office. This is a string which may be directly resolved to an IP Address, or may require DNS lookup to resolve the domain name to the Service provider's address. If this field is left blank, registration is against the LAN IP address.
- **Number of Channels:** *Default = 10*
  Number of trunk channels between 1 and 24.
- **Authentication Name:** *Default = Blank.*
  This value is provided by the SIP ITSP.
- **Password:** *Default = Blank.*
  This value is provided by the SIP ITSP.
- **Details**
  Clicking on **Details** 99 will display the additional settings for the selected SIP trunk.

3.6.3.1 SIP Trunk Details

This menu is accessed by selecting **System** and then **SIP Trunks** from the menu bar. Select the trunk required and click on **Details**.



3.6.3.1.1 SIP Trunk Details

- **Proxy Server Address**
  In exceptional circumstances, the IP Address of the proxy server may be explicitly identified as either a different IP Address, or a different domain address resolvable by DNS.
  - **! WARNING - Reboot Required**
    Changing this setting requires the system to be rebooted 12 for the change to take effect. Rebooting the system will end all calls currently in progress.

- **DNS Server Address**
  If the proxy server address is set to a named server, the address of the DNS server used for name resolution should be entered here.

- **Mobility Caller ID Format**
  This option corresponds to the standard "draft-ietf-sip-privacy-04". The options are ***None***, ***Remote Party ID***, ***P Asserted ID*** or ***Diversion Header***.

- **Use Tel URI:** *Default = SIP URI.*
  Select the format of numbering to be used in the From field on outgoing calls. The options are ***SIP URI*** or ***Tel URI***. Tel URI uses the format TEL: +1-425-555-4567. SIP URI uses the format *name@example.com*).

- **Check OOS:** *Default = On. Software level = 8.0+.*
  When enabled, the system will regularly check if the trunk is in service. Checking that SIP trunks are in service ensures that outgoing calls are not delayed waiting for response on a SIP trunk that is not currently usable. Depending on the trunk's **Transport Protocol**, the trunks current service status is checked using the following methods:

* For all trunks, regular OPTIONS messages are sent. If no reply is received, the trunk is taken out of service.
* For TCP trunks, if the TCP connection is disconnected the trunk will be taken out of service.
* For trunks using DNS, if the IP address is not resolved or the DNS resolution has expired, the trunk is taken out of service.

- **Call Routing Method:** *Default = Request URI. Software level = 8.0+.*
  This field allows selection of which part of the incoming SIP information should be used for the incoming number. The options are to match either the ***Request URI*** or the ***To Header*** element provided with the incoming call.
- **Association Method:** *Default = By Source IP address. Software level = 8.0+.*
  This field sets the method by which a SIP line is associated with an incoming SIP request. The search for a line match for an incoming request is done against each line until a match occurs. If no match occurs, the request is ignored. This method allow multiple SIP lines with the same address settings. This may be necessary for scenarios where it may be required to support multiple SIP lines to the same ITSP. For example when the same ITSP supports different call plans on separate lines or where all outgoing SIP lines are routed from the system via an additional on-site system.
  - ***By Source IP Address***
    This option uses the source IP address and port of the incoming request for association. The match is against the configured remote end of the SIP line, using either an IP address/port or the resolution of a fully qualified domain name. This matches the method used by pre-8.0 systems.
  - ***"From" header hostpart against ITSP domain***
    This option uses the host part of the From header in the incoming SIP request for association. The match is against the line's **Domain Name**.
  - ***R-URI hostpart against ITSP domain***
    This option uses the host part of the Request-URI header in the incoming SIP request for association. The match is against the line's **Domain Name**.
  - ***"To" header hostpart against ITSP domain***
    This option uses the host part of the To header in the incoming SIP request for association. The match is against the line's **Domain Name**.
  - ***"From" header hostpart against DNS-resolved ITSP domain***
    This option uses the host part of the FROM header in the incoming SIP request for association. The match is found by comparing the FROM header against a list of IP addresses resulting from resolution of the line's **Domain Name** or, if set, the **Proxy Server Address**.
  - ***"Via" header hostpart against DNS-resolved ITSP domain***
    This option uses the host part of the VIA header in the incoming SIP request for association. The match is found by comparing the VIA header against a list of IP addresses resulting from resolution of the line's **Domain Name** or, if set, the line's **Proxy Server Address**.
  - ***"From" header hostpart against ITSP proxy***
    This option uses the host part of the "From" header in the incoming SIP request for association. The match is against the line's **Proxy Server Address**.
  - ***"To" header hostpart against ITSP proxy***
    This option uses the host part of the From header in the incoming SIP request for association. The match is against the line's **Proxy Server Address**.
  - ***R-URI hostpart against ITSP proxy***
    This option uses the host part of the Request-URI in the incoming SIP request for association. The match is against the line's **Proxy Server Address**.
- **Name Priority:** *Default = Favour Trunk.*
  For SIP trunks, the caller name displayed on an extension can either be that supplied by the trunk or one obtained by checking for a number match in the system speed dials. This setting determines which method is used by the line. Select one of the following options:
  - **System Default**
    Use the system's **Default Name Priority** 65 setting, the default being ***Favour Trunk***.
    - ***Favour Trunk***
      Display the name provided by the trunk. For example, the trunk may be configured to provide the calling number or the name of the caller. The system should display the caller information as it is provided by the trunk.
    - ***Favour Directory***
      Search for a number match in the system speed dials. The first match is used and overrides the name provided by the SIP line. If no match is found, the name provided by the line is used.

- **Calls Route Via Registrar:** *Default = On*
  Normally SIP REGISTER requests and INVITE requests use the same server destination. This option should only be deselected when the service provider does not expect REGISTER requests to go to the same destination as the INVITE requests. You should only set this under specific instruction from the service provider.

- **Separate Registrar**
  This field is available when **Calls Route Via Registrar** is deselected. It is used to enter the address of the SIP server that should be used for registration. You should only set this under specific instruction from the service provider.

- **Transport Protocol:** *Default = Both TCP & UDP.*
  Both TCP and UDP SIP end points are supported. This field can be used to restrict the IP Office to just TCP or UDP if required.

- **Send Port:** *Default = 5060.*
  The port to use for outgoing call support.

- **Listen Port:** *Default = 5060.*
  The port to use for incoming call support.

- **UPDATE Supported:** *Default = Never. Software level = 8.0+.*
  The SIP UPDATE method (RFC 3311) allows a client to update parameters of a session (such as the set of media streams and their codecs) but has no impact on the state of a dialog. It is similar to re-INVITE, but can be sent before the initial INVITE has completed. This allows it to update session parameters within early dialogs.

3.6.3.1.2 VoIP Parameters

- **Compression Mode:** *Default = Automatic Selection*
  This defines the type of compression which is to be used for calls on this line.

- **VOIP Silence Suppression:** *Default = Off*
  When selected, this option will detect periods of silence on any call over the line and will not send any data during those silent periods.

- **Call Initiation Timeout:** *Default = 4 seconds.*
  Sets how long to wait for successful connection before treating the line as busy.

- **RE-Invite Supported:** *Default = Off.*
  When enabled, Re-Invite can be used during a session to change the characteristics of the session, for example when the target of an incoming call or a transfer does not support the codec originally negotiated on the trunk. Requires the ITSP to also support Re-Invite.

- **DTMF Support:** *Default = RFC2833*
  This setting is used to select the method by which DTMF key presses are signaled to the remote end. The supported options are ***In Band***, ***RFC2833*** or ***Info***.

- **Use Offerer's Codec:** *Default = Off.*
  Normally for SIP calls, the answerer's codec preference is used. This option can be used to override that behavior and use the codec preferences offered by the caller.

- **Registration Expiry:** *Default = 60 minutes.*
  This setting defines how often registration with the SIP ITSP is renewed following any previous registration.
  - **! WARNING - Reboot Required**
    Changing this setting requires the system to be rebooted [12] for the change to take effect. Rebooting the system will end all calls currently in progress.

- **PRACK/100rel Supported:** *Default = Off. Software level = 8.0*
  This option sets whether Provisional Reliable Acknowledgement (PRACK) and 100rel are enabled. 100rel allows SDP negotiation to be completed while the call is in ringing state and allows further media changes for announcements or progress tones before a call is actually answered. PRACK, as defined in RFC 3262, provides a mechanism to ensure the delivery of provisional responses such as announcement messages. Provisional responses provide information on the status of the call request that is still in progress.
  - Example: When a call to a mobile or cell phone is in the process of being connected, there may be a delay while the cell phone is located. Provisional information allow features such as an announcement *"please wait while we attempt to reach the subscriber"* to be played while the call setup is still in progress.

- **Fax Transport Support:** *Default = Off. Software level = 8.0+*
  This option is only available if **Re-Invite Supported** is selected. When enabled, the system performs fax tone detection on calls routed via the line and, if fax tone is detected, renegotiates the call codec as configured below. The SIP line provider must support the selected fax method and **Re-Invite**.
  - ***None***
    Select this option if fax is not supported by the line provider.
  - ***G711***
    G711 is used for the sending and receiving of faxes.

- ***T38***
  T38 is used for the sending and receiving of faxes.

- ***T38 Fallback***
  T38 is used for the sending and receiving of faxes. On outgoing fax calls, if the called destination does not support T38, a re-invite it sent for fax transport using G711.

- **Caller ID from From Header:** *Default = Off. Software Level = 8.1.*
  Incoming calls can include caller ID information in both the From field and in the PAI fields. When this option is selected, the caller ID information in the From field is used rather than that in the PAI fields.

- **Send From In Clear:** *Default = Off. Software Level = 8.1.*
  When selected, the user ID of the caller is included on the From field. This applies even if the caller has selected to be or is configured to be anonymous, though their anonymous state is honored in other fields used to display the caller identity.

- **User-Agent and Server Headers:** *Default = Blank (Use system type and software level). Software Level = 8.1.*
  The value set in this field is used as the User-Agent and Server value included in SIP request headers made by this line. Setting a unique value can be useful in call diagnostics when the system has multiple SIP trunks.

3.6.3.1.3 Refer Support

- **Refer Support:** *Default = On.*
  REFER is the method used by many SIP devices, including SIP trunks, to transfer calls. These settings can be used to control whether REFER is used as the method to transfer calls on this SIP trunk to another call on the same trunk. If supported, once the transfer has been completed, the IP Office system is no longer involved in the call. If not supported, the transfer may still be completed but the call will continue to be routed via the IP Office.

- **Incoming:** *Default = Auto*
  Select whether REFER can or should be used when an attempt to transfer an incoming call on the trunk results in an outgoing call on another channel on the same trunk. The options are:

  - ***Always***
    Always use REFER for call transfers that use this trunk for both legs of the transfer. If REFER is not supported, the call transfer attempt is stopped.

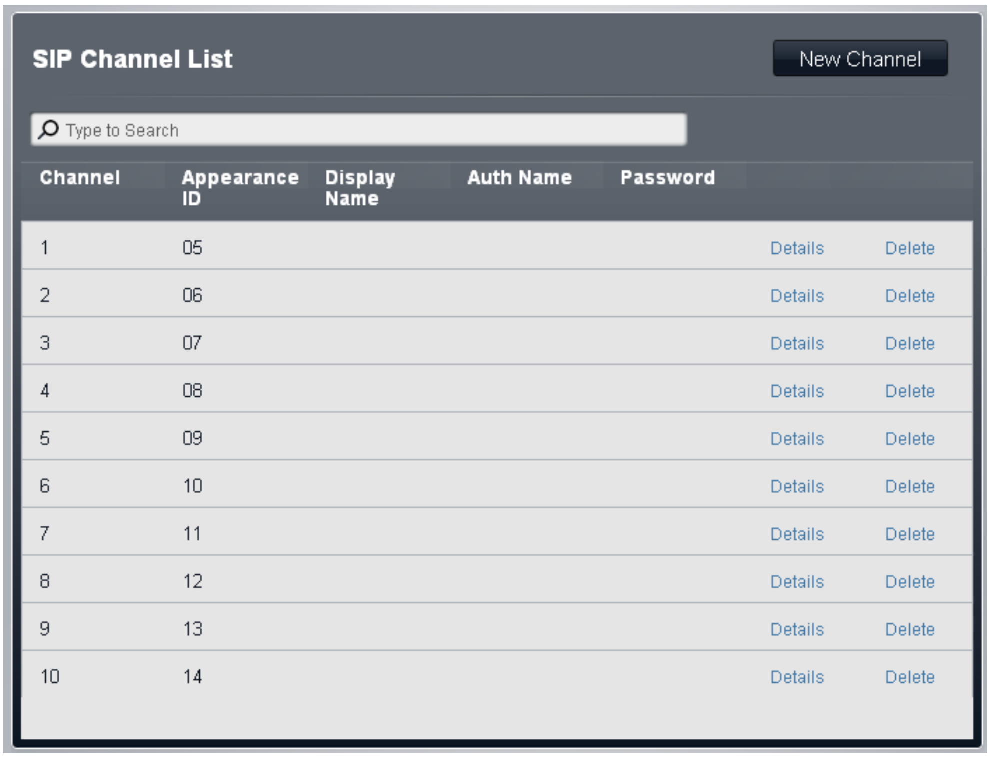
  - ***Auto***
    Request to use REFER if possible for call transfers that use this trunk for both legs of the transfer. If REFER is not supported, transfer the call via the system as for the ***Never*** setting below.

  - ***Never***
    Do not use REFER for call transfers that use this trunk for both legs of the transfer. The transfer can be completed but will use 2 channels on the trunk.

- **Outgoing:** *Default = Auto*
  Select whether REFER can or should be used when attempt to transfer an outgoing call on the trunk results in an incoming call on another channel on the same trunk. This uses system resources and may incur costs for the duration of the transferred call. The options available are the same as for the **Incoming** setting.

3.6.3.2 SIP Channel List

This menu is accessed by selecting **System** and then **SIP Trunks** from the menu bar. Select the trunk required and click on **Details**. Click on the edit icon in the **Channel Setup** panel.



- **Channel**
  Channel number, cannot be edited.

- **Appearance ID**
  Appearance ID numbers can be used to associate each channel a **Line Appearance** button on phones that support button programming. That button can then be used to make and answer calls using the channel. The line appearance ID for each channel is automatically assigned to those channels that have their **Direction** set as ***Bothways***.

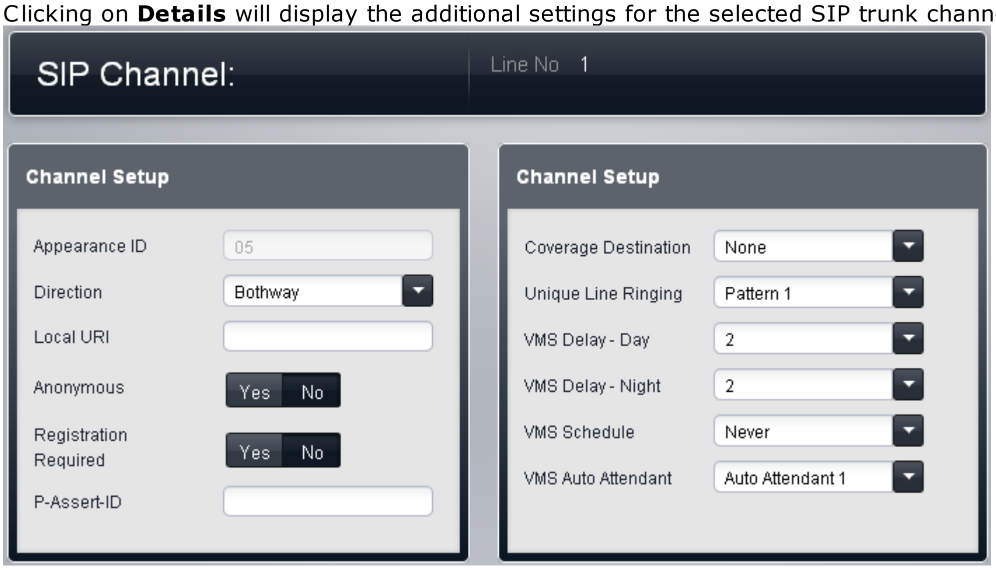
- **Display Name:** *Default = Use Authentication Name*
  This field sets the 'Name' value for SIP calls.

- **Authentication Name**
  When making a call, some service providers will often send an authentication challenge to validate the call before it is connected. This challenge requires the INVITE is re-submitted with Authentication data, including a network account name (provided by the service provider during installation). The network account name is the "Auth name". It can be blank, in which case the **Local URI** is used.

- **Password:** *Default = Blank.*
  This value is provided by the SIP ITSP.

  - **! WARNING - Reboot Required**
    Changing this setting requires the system to be rebooted 12 for the change to take effect. Rebooting the system will end all calls currently in progress.

- **Details**
  Clicking on **Details** will display the additional settings for the selected SIP trunk channel.



  - **Appearance ID**
    Appearance ID numbers can be used to associate each channel a **Line Appearance** button on phones that support button programming. That button can then be used to make and answer calls using the channel. The line appearance ID for each channel is automatically assigned to those channels that have their **Direction** set as ***Bothways***.

  - **Direction:** *Default = Bothways*
    Sets the allowed operation mode of the line. For systems running in Key mode, a line can be set to either ***Bothway*** (incoming and outgoing) or ***Incoming Call by Call*** (incoming only). For a system running in PBX mode, a line can be set to either ***Bothway*** (incoming and outgoing) or ***Call by Call*** (incoming and outgoing).

    - **! WARNING - Reboot Required**
      Changing this setting requires the system to be rebooted 12 for the change to take effect. Rebooting the system will end all calls currently in progress.

    - ***Bothway***
      When set to ***Bothway***, incoming calls are presented to line appearance buttons matching the channels **Appearance ID** and to the channels **Coverage Destination** if set. For Key mode systems, outgoing calls are routed to the channel by pressing the matching line appearance button selection or by automatic line selection. In addition, on PBX mode systems, outgoing calls can be routed to the channel by including the line appearance in the **ARS Selector** 59 that matches the dialed digits.

    - ***Incoming Call by Call***
      For systems running in ***Key*** mode, when set to ***Incoming Call by Call***, incoming calls are routed using the **Call by Call** table. The **Appearance ID**, **Coverage Destination** and **Ring Pattern** fields are greyed-out as those settings are not applied. The trunk channel is not used for outgoing calls.

    - ***Call by Call***
      For systems running in ***PBX*** mode, when set to ***Call by Call***, incoming calls are routed using the **Call by Call** table. The **Appearance ID**, **Coverage Destination** and **Ring Pattern** fields are greyed-out as those settings are not applied. In PBX mode, call by call entries can be used given ARS selector numbers (see below) which allow the trunk channel to also be used for outgoing calls.

  - **Local URI:**
    The user part of the SIP URI. This specifies the contents of the FROM field when making a call (sending an INVITE).

    - **! WARNING - Reboot Required**
      Changing this setting requires the system to be rebooted 12 for the change to take effect. Rebooting the system will end all calls currently in progress.

  - **Anonymous:**
    Withhold the calling parties information.

    - **! WARNING - Reboot Required**
      Changing this setting requires the system to be rebooted 12 for the change to take effect. Rebooting the system will end all calls currently in progress.

  - **Registration Required**
    When selected, each local URI with unique Authentication credentials will register independently.

- **P-Assert-ID**
  If this field is configured, the channel can be used in *SIPConnect Option 1* model for separating Public and Private PSTN identity (Sipconnect technical recommendation v 10, section 12.1.1). You can only use Explicit CLI configurations over SIP if using Option1 model for identity. In this case, calls over this channel will always have a fixed P-Assert-ID, but the From field may vary.

- **Coverage Destination:** *Default = None.*
  This option sets where incoming calls should alert in addition to alerting on those extension that have a line appearance button programmed for the line. When the phone system is in night service mode, calls alert at the members of the **Night Service** group.

  - **None**
    If set to ***None***, incoming calls will only alert on user extensions with line appearance buttons that match the line's **Appearance ID**.

  - **Extension**
    Route incoming calls to a particular extension.

  - **Phantom Extension**
    A phantom extension can be selected as the destination for calls.

  - **Hunt Group**
    Incoming calls can be routed to one of the 6 rotary hunt groups.

  - **Calling Group**
    For systems with their **Mode** set to ***PBX***, incoming calls can be routed to one of these 4 ring all groups.

  - **Operator Group**
    For systems with their **Mode** set to ***PBX***, incoming calls are routed to the **Operator Group**.

  - **Voicemail**
    Route incoming calls to the systems voicemail to collect messages. This requires the caller to know the mailbox number and passcode.

  - **! WARNING - Reboot Required**
    Changing this setting requires the system to be rebooted 12 for the change to take effect. Rebooting the system will end all calls currently in progress.

  - The **Coverage Destination** is not used for SIP trunks with their direction set to ***Incoming Call by Call***.

- **Ring Pattern:** *Default = 1.*
  Selects the ring pattern that should be used for calls when alerting on an extension. Calls forwarded, sent to call coverage or to a hunt group will always use the line ring pattern. Calls direct to an extension will use the line ringing pattern unless the user has **Override Line Ringing** set. Not used for calls presented to the user as a member of the Operator group. This feature is also not used for BST phones.

  - **! WARNING - Reboot Required**
    Changing this setting requires the system to be rebooted 12 for the change to take effect. Rebooting the system will end all calls currently in progress.

- **Delay - Day:** *Default = 2. Range = 0 to 6 (number of rings).*
  Set the number of rings before an unanswered call should be redirected the selected auto attendant when the system is not running in night service mode and the **VMS Schedule** is set to ***Always*** or ***Days Only***.

- **Delay - Night:** *Default = 2. Range = 0 to 6 (number of rings).*
  Sets the number of rings before an unanswered call should be redirected to the selected auto attendant when the system is running in night service mode and the **VMS Schedule** is set to ***Always*** or ***Night Only***.

- **Schedule:** *Default = Never.*
  This option determines when the **VMS Delay** settings above should be used and unanswered calls redirected to the selected auto attendant. The options are:

  - ***Always***
    Redirect calls when the system is in both day and night service modes.

  - ***Day Only***
    Redirect calls only when the system is not in night service.

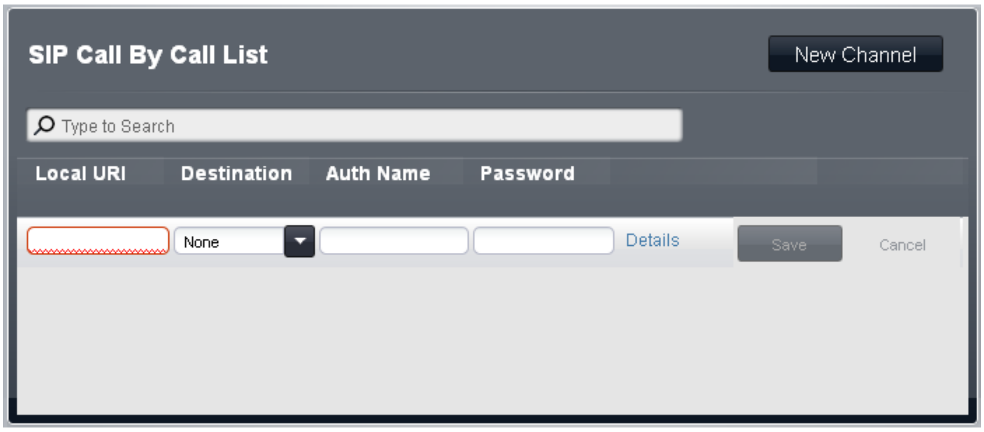
  - ***Night Only***
    Redirect calls only when the system is in night service.

  - ***Never***
    Do not redirect calls.

- **Auto Attendant:** *Default = Auto Attendant 1.*
  This field allows selection of which auto attendant is used.

3.6.3.3 Call by Call Settings

This menu is accessed by selecting **System** and then **SIP Trunks** from the menu bar. Select the trunk required and click on **Details**. Click on the edit icon in the **Call by Call** panel.



SIP Call By Call List

These settings are used to match calls received on SIP trunks channels set to ***Incoming Call-by-Call***. For systems operating in ***Key System*** mode, the default entry is used for all calls for which there is no other match and is fixed to route those calls to the **Operator Group**.

- **ARS**
  This setting is only shown for ***PBX*** mode systems. For those systems, each call-by-call entry can be assigned to an ARS Selector 59 number. That selector number can then be used as the destination for outgoing calls.

- **Local URI:**
  The user part of the SIP URI. This specifies the contents of the FROM field when making a call (sending an INVITE).

- **Destination**
  Where incoming calls with matching digits should be routed. The drop-down list contains the extensions and groups on the IP Office system.

  - **Extension**
    Route incoming calls to a particular extension.

  - **Phantom Extension**
    A phantom extension can be selected as the destination for calls.

  - **Calling Group**
    For systems with their **Mode** set to ***PBX***, incoming calls can be routed to one of these 4 ring all groups.

  - **Operator Group**
    For systems with their **Mode** set to ***PBX***, incoming calls are routed to the **Operator Group**.

  - **Voicemail**
    Route incoming calls to the systems voicemail to collect messages. This requires the caller to know the mailbox number and passcode.

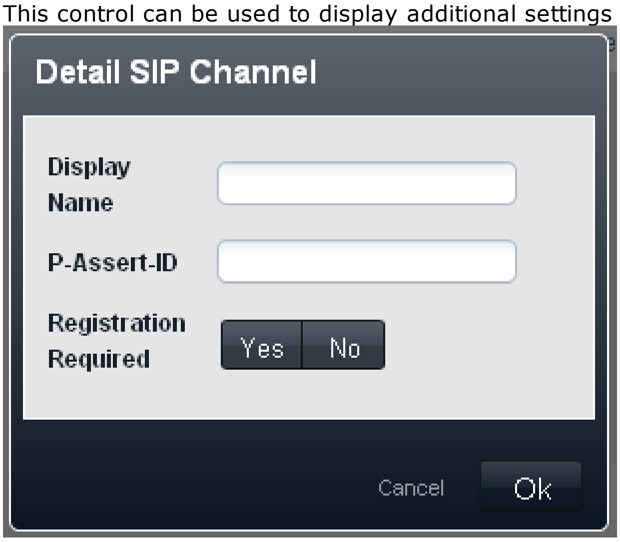
  - **76: Modem**
    The option ***76: Modem*** can be selected to route the call to the systems built in V32 modem function. This is only intended for basic access by system maintainers.

  - **Auto Attendant**
    Any configured voicemail auto attendants can be selected as the call destination.

- **Authentication Name**
  When making a call, some service providers will often send an authentication challenge to validate the call before it is connected. This challenge requires the INVITE is re-submitted with Authentication data, including a network account name (provided by the service provider during installation). The network account name is the "Auth name". It can be blank, in which case the **Local URI** is used.

- **Password:** *Default = Blank.*
  This value is provided by the SIP ITSP.

- **Details**
  This control can be used to display additional settings associated with the call by call entry.

  - **Display Name:** *Default = Use Authentication Name*
    This field sets the 'Name' value for SIP calls using this URI.

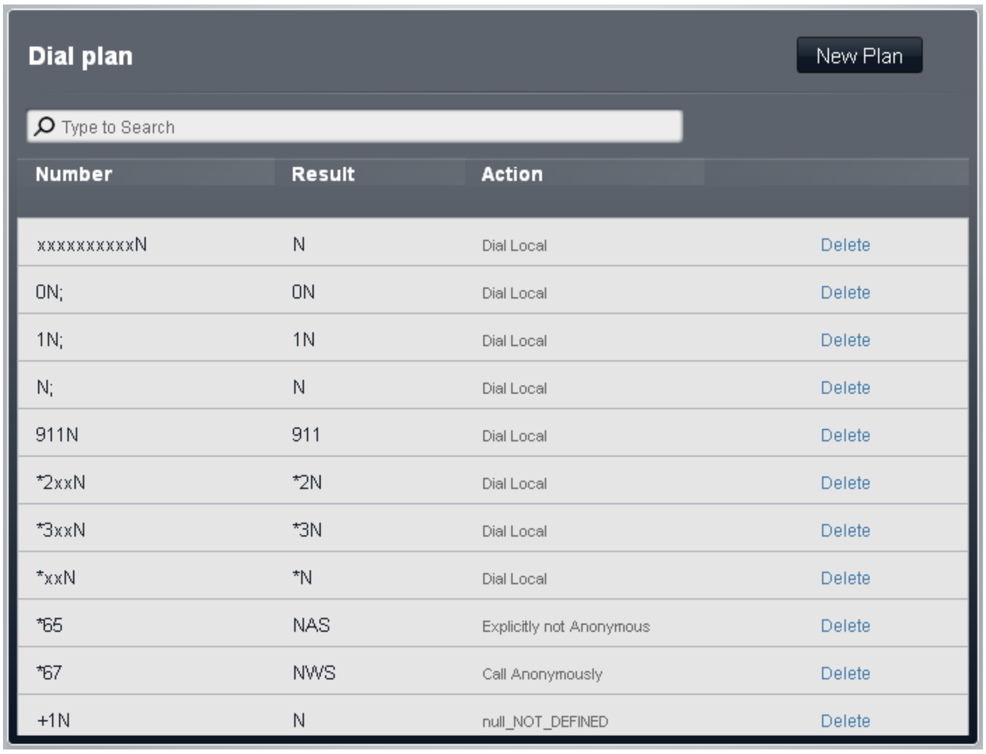
  - **P-Assert-ID**
    If this field is configured, the channel can be used in *SIPConnect Option 1* model for separating Public and Private PSTN identity (Sipconnect technical recommendation v 10, section 12.1.1). You can only use Explicit CLI configurations over SIP if using Option1 model for identity. In this case, calls over this channel will always have a fixed P-Assert-ID, but the From field may vary.

  - **Registration Required**
    When selected, each local URI with unique Authentication credentials will register independently.

3.6.3.4 Dial Plan Settings

This menu is accessed by selecting **System** and then **SIP Trunks** from the menu bar. Select the trunk required and click on **Details**. Click on the edit icon in the **Dial Plan** panel.

The dial plan is used to apply number translations to the digits received by the line for output to the line provider and to indicate any special service required from the line provider, for example to withhold the call ID. The default dial plan is as shown below.



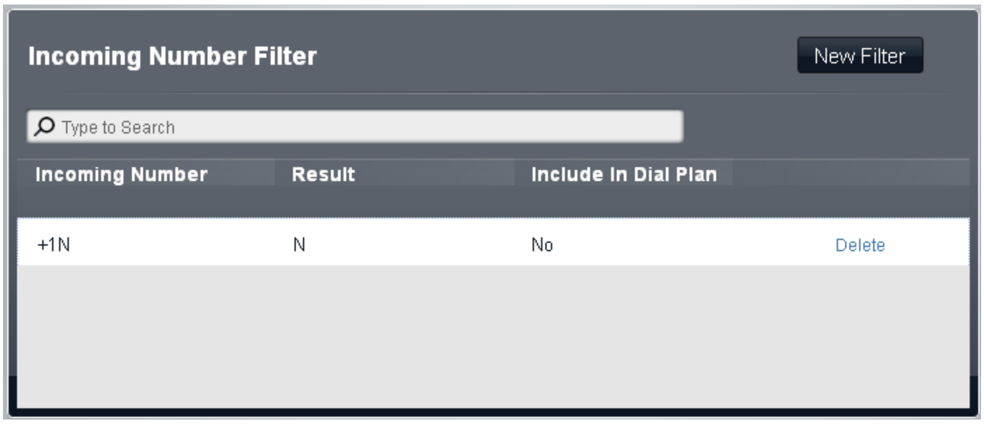
The following are the default entries used in a dial plan for North American locales.

Dialled Number	Result	Action
xxxxxxxxxxN	N	Dial Local
0N;	0N	Dial Local
1N;	1N	Dial Local
N;	N	Dial Local
911	911	Dial Local
***2xxN**	*2N	Dial Local
***3xxN**	*3N	Dial Local
***xxN**	*N	Dial Local
***65**		Explicitly not Anonymous
***67**		Call Anonymously

3.6.3.5 Incoming Number Filter

The default incoming number filter simply converts international USA numbers received into local 10 digit numbers. However, it is also useful for mapping PC calls (from skype, google, windows etc) into a dialable number plan. One nice way to use this is to map PC calls into numbers in area code "555".

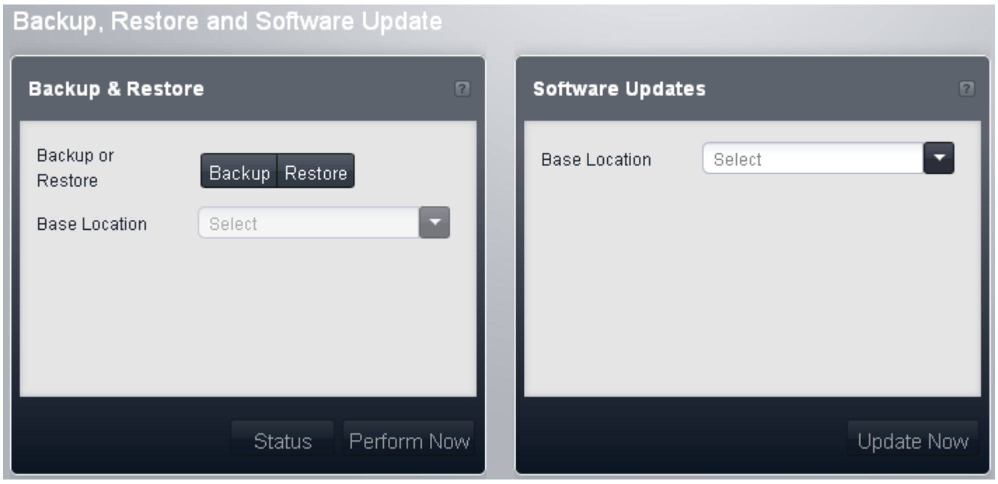
This menu is accessed by selecting **System** and then **SIP Trunks** from the menu bar. Select the trunk required and click on the edit icon in the **Dial Plan** panel. Then click on the edit icon in the **Incoming Number Filter** panel.



- **Incoming Number**
  Used to match the incoming number received.
- **Result**
  The replacement for the incoming number.
- **Include in Dial Plan**
  When you select include in dial plan, the system will automatically substitute the number you dial for outgoing calls as well.

3.6.4 Backup and Update

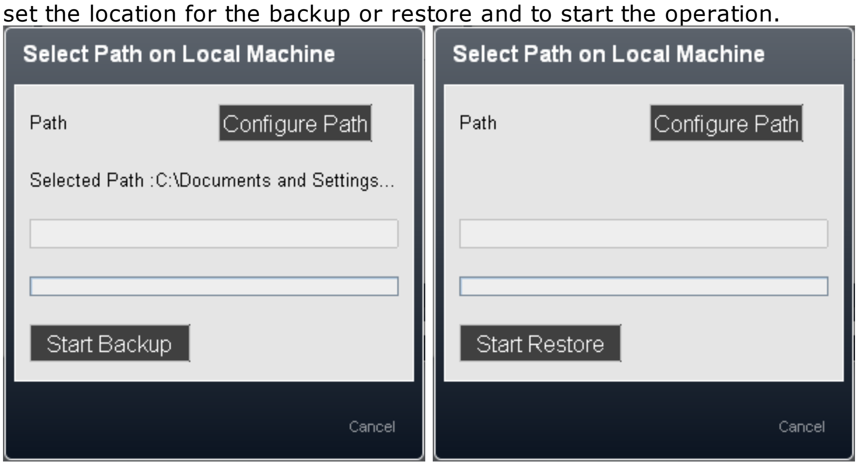
This menu is accessed by selecting **System** in the menu bar and clicking **Backup and Restore**.



3.6.4.1 Backup & Restore

This menu allows you to perform a backup or a restore of all the files in the **/System/Primary** folder on the System SD card. It does not include the files from embedded voicemail mailboxes and auto attendants.

- **Backup or Restore**
  Select the button for the type of action that you want to perform. Following a **Restore** you will need to restart the system for the restored files to take effect.
- **Base Location**
  Set the location to which you want to backup or from which you want to restore a previous backup. The options are ***SD Card*** or ***Local Machine***.
  - ***SD Card***
    Backup or restore between the ***/System/Primary*** and ***/System/Backup*** folders on the System SD card.
  - ***Local Machine***
    Backup to or restore from the computer from which you are accessing web based management. This requires the PC and browser to support Java. Once the backup or restore is started, you will be prompted for the file path for the operation. Note that the full backup set is approximately 550MB in size and takes at least 60 minutes or more depending on the speed of the link between the browsing computer and the IP Office system. A backup consists of a ***payload.xml*** file and a **/System** sub-folder added to the path specified when the operation is run.
- **Status**
  Clicking this button will briefly display a message detailing the status of any backup or restore operations currently in progress.
- **Perform Now**
  Clicking this button will start the selected operation. If the **Base Location** is set to ***Local Machine***, you may be prompted to confirm whether you want to allow Java to run, select **Yes**, you are then shown a menu to set the location for the backup or restore and to start the operation.



- **Configure Path**
  Click this button to select the folder into which you want to store the backup or from which you want to restore a previous backup. A backup consists of a ***payload.xml*** file and a **/System** sub-folder in the folder specified.

  - When backing up, if an existing payload.xml file exists, the system will compare its files with those listed in the file and will not do a backup if all files match. In this case, if there is some mismatch, the existing backup is overwritten.

- **Start Backup / Start Restore**
  Click this button to start the backup or restore operation. Once the operation has started, the status of the operation is displayed.

3.6.4.2 Software Updates

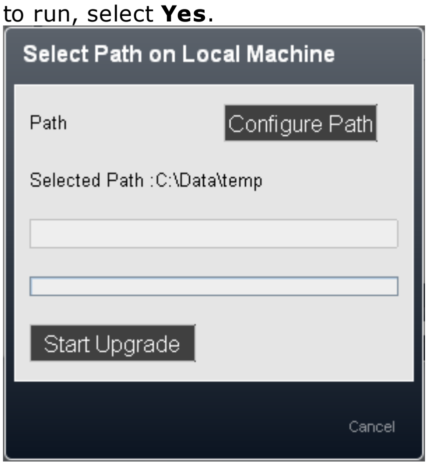
Software updates in the form of specially prepared file set files may be made available by Avaya. These can be used to update the software being used by a system.

- **! WARNING**
  You must read all IP Office Technical Bulletins relating to the IP Office software release to which you are upgrading the system before you upgrade. The Technical Bulletin may contain addition warnings or steps that must be performed before or after upgrading.

- **! WARNING**
  Before running an update, ensure you have taken a backup of the system.

- **Base Location**
  Set the location from which you want to upgrade. The only option currently supported is ***Local Machine***, ie. the computer from which you are accessing web based management.

- **Update Now**
  Clicking this button will start the operation. You may be prompted to confirm whether you want to allow Java to run, select **Yes**.



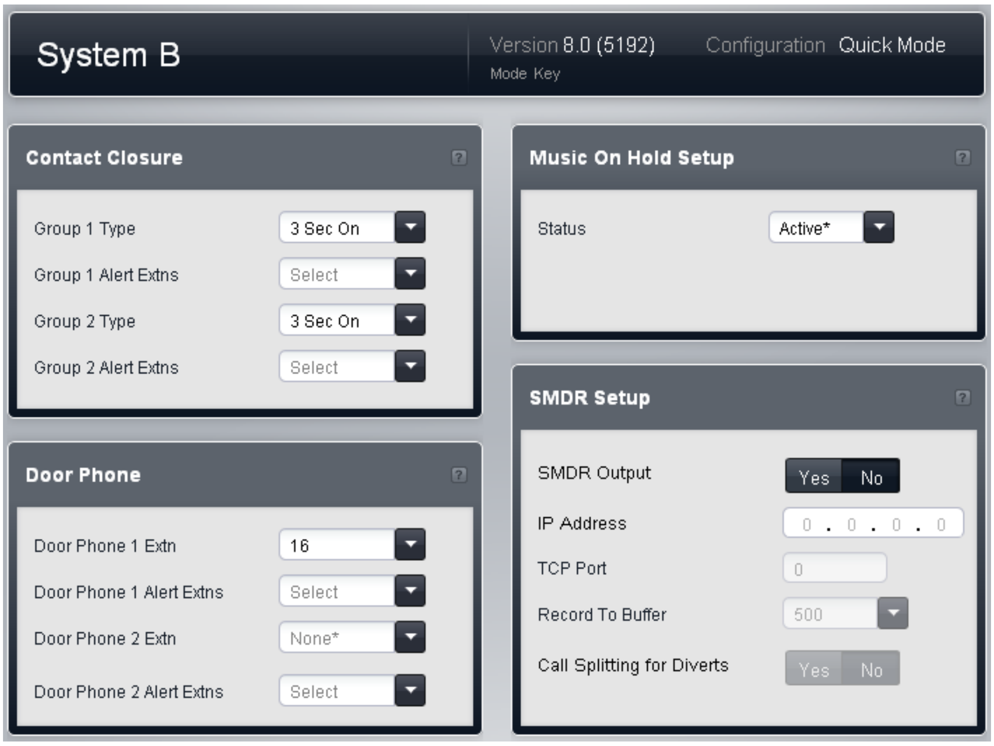
  - **Configure Path**
    Click this button to select the folder containing the files and folders to be used for the upgrade. The location should include a file ***UpPackInv.xml***.

    - **Example:** If IP Office Manager is installed on the PC in its default location, then the path from which to upgrade using the files installed with IP Office Manager is ***C:\Program Files\IP Office\Manager\MemoryCards***.

  - **Start Upgrade**
    Once the path has been set, clicking this button will start the upgrade operation. Once the operation has completed, you must login again and restart the system 121 for the new files to be used.

3.6.5 Auxiliary Equipment

This menu is accessed by selecting **System** in the menu bar and clicking **Auxiliary Equipment**. The menu is used to configure the operation of a range of additional equipment that may be connected to the system.



3.6.5.1 Contact Closure

The phone system has two ports which can be connected to external relay systems, for example systems used to open doors. These connect to the Ext O/P socket on the rear of the system's control unit. Refer to the installation manual for details.

Once a external relay system is connected, you can configure which extension users are able to activate the relay ports and the type of activation (open, close, pulse).

There are two separate menus, one for **Contact Closure Group 1** and one for **Contact Closure Group 2**. Each has the same range of settings.

- **Contact Closure Type:** *Default = 3 Seconds On.*
  Sets how long the closure is activated when a user presses a contact closure button. The options are ***1 Second On***, ***3 Seconds On***, ***5 Seconds On*** and ***Toggle***. (change the contact between open or closed).

- **Extensions to be enabled:** *Default = None.*
  This table is used to select which user extensions are able to activate the contact closure by dialing feature codes at their extension or using programmable buttons set to the **Contact Closure** feature.

3.6.5.2 Door Phone

Up to two analog extension ports can be configured as door phones. When either door phone is taken off-hook, the other extensions configured below are alerted and can answer the door phone user. This option is typically used to connect a phone in a public area to a receptionist or similar.

There are two sets of settings, one for **Door Phone 1** and one for **Door Phone 2**. Each has the same range of settings.

- **Door Phone Extns:** *Default = None.*
  Use the drop down list to select the extension to which the door phone is connected. The **Equipment** 43 setting of the user is automatically set to ***Door Phone 1*** or ***Door Phone 2*** by this.

  - **! WARNING - Reboot Required**
    Changing this setting requires the system to be rebooted 12 for the change to take effect. Rebooting the system will end all calls currently in progress.

- **Door Phone Alert Extns:** *Default = None.*
  This table is used to select which extensions are alerted and can answer when the door phone is taken off hook.

3.6.5.3 Music on Hold Setup

The phone system supports an external music on hold source. This connects to the **Audio** port on the rear of the system's control unit. You can configure whether the input to this port is played to callers when they are put on hold.

The music on hold input can also be played to callers being transferred rather than ringing tone. That behaviour is controlled by the system's **Ring on Transfer** 69 setting.

- **Status:** *Default = Active.*
  If enabled, the system will use the external music source connected to the phone system. If not enabled, the system provides a double beep tone repeated every 5 seconds.

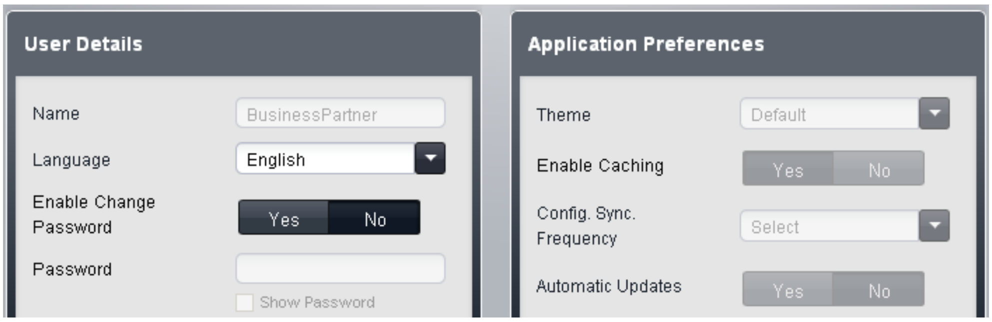
3.6.5.4 SMDR Setup

The phone system can log call details at the end of each call. These SMDR records (Station Message Detail Recording) can be sent to a specified IP address where they can be collected and processed by 3rd party call logging software.

- **SMDR output:** *Default = Off*
  This control can be used to switch the output of SMDR on or off.
- **IP Address:** *Default = 0.0.0.0 (Listen).*
  The destination IP address for SMDR records.
- **TCP Port:** *Default = 0.*
  The destination IP port for SMDR records.
- **Record to Buffer:** *Default = 500. Range = 10 to 3000.*
  The phone system can buffer up to 3000 SMDR records if it detects a communications failure with destination address. When the buffer is full, each new record overwrites the oldest record.
- **Call Splitting for Diverts:** *Default = Off.*
  When enabled, for calls forwarded off-switch using an external trunk, the SMDR produces separate initial call and forwarded call records. This applies for calls forwarded by forward unconditional, forward on no answer or forward on busy. The two sets of records will have the same Call ID. The call time fields of the forward call record are reset from the moment of forwarding on the external trunk.

3.6.6 User Preferences

These settings relate to the service user account that you are using to administer the system configuration.



3.6.6.1 User Details

These settings relate to the account which you have used to login to the Basic Edition Web Manager menus. You can use the fields to change the name and password.

- **Name**
  This field is not changeable when logged in using that account. It shows the name set for the login account of up to 31 characters. For service users other than ***Administrator*** and ***BusinessPartner***, the name can be changed by the ***BusinessPartner*** account user through the **Service Users** [116] menu. The name for the ***Administrator*** and ***BusinessPartner*** users cannot be changed.

- **Language**
  Select the preferred language that you want used for the menus.

- **Enable Change Password**
  When set to **Yes**, you can use the **Password** field to enter a new password.

- **Password**
  Enter the new password that you want to use for logging in to Basic Edition Web Manager. You can set a password of up to 31 characters. The ***BusinessPartner*** account user can also change the password for accounts, other than ***Administrator***, through the **Service Users** [116] menu.

3.6.6.2 Application Preferences

These settings affect how the Basic Edition Web Manager menus operate. They do not affect the configuration of the system.

- **Theme:** *Default = Default.*
  The theme selected changes the look and feel of the Basic Edition Web Manager menus.

- **Enable Caching:** *Default = Yes.*
  When a Basic Edition Web Manager menu is first displayed, the setting values to show for the menu are requested from the system. The Basic Edition Web Manager menu can then temporarily store (cache) those values.

  - If **Enable Caching** is enabled, when the menu is next displayed, the cached values are used rather than being requested from the system again.

  - If **Enable Caching** is disabled, when a menu is next displayed, the values for setting in the menu are requested again from the system.

- **Config. Sync. Frequency:** *Default = 60 seconds.*
  Changes to the system configuration can occur from other sources while you have the configuration open in Basic Edition Web Manager. For example a user may change their do not disturb numbers through the menu on their phone. This setting controls how often the Basic Edition Web Manager menus should check if changes have occurred.

  - If **Config. Sync.** is enabled, the settings on the system are rechecked using the set frequency. If they have changed, the action performed depends on the **Automatic Updates** setting below.

  - If **Config. Sync.** is disabled, the current settings on the system are not rechecked at regular intervals.

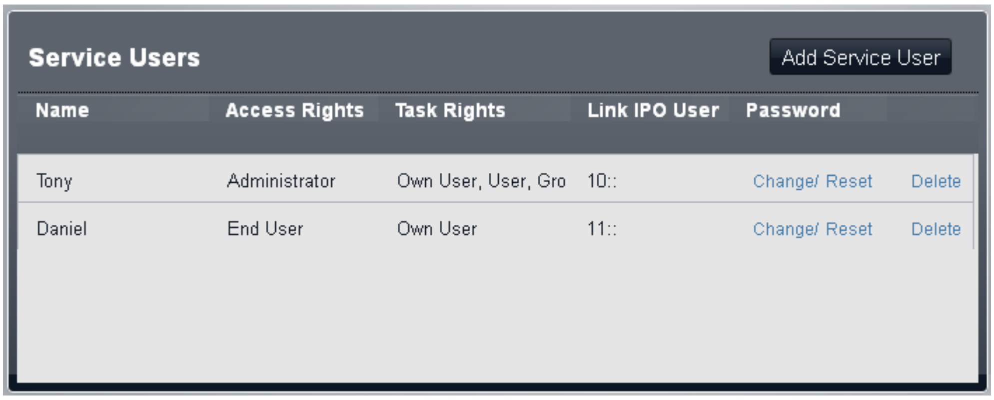
- **Automatic Updates:** *Default = Yes.*
  Changes to the system configuration can occur from other sources while you have the configuration open in Basic Edition Web Manager. For example a user may change their do not disturb numbers through the menu on their phone. If **Automatic Updates** is enabled, when changes are made elsewhere while you are logged into Basic Edition Web Manager, the settings shown in Basic Edition Web Manager are updated. If **Automatic Updates** is disabled, when changes occur elsewhere you will be altered and prompted to select an action.

3.6.6.3 Service Users

This menu is only usable by a user logged in using the ***BusinessPartner*** name and password. The menu allows them to create and configure additional accounts for use with Basic Edition Web Manager.

This menu is access by selecting **System** in the menu and clicking on **User Preferences**. Then click the edit icon in **Role Based Rights** panel.

Service Users



- **Name**
  The service user name. This is used as part of the Basic Edition Web Manager login.

- **Access Rights**
  This field sets what actions the service user can perform.

  - ***Administrator***
    Service users with this setting can adjust the settings on any menus that they can access.

  - ***End User***
    Service users with this setting can view the settings on any menus that they can access but can only adjust their own user settings.

- **Task Rights**
  This field is used to select which menus in the Basic Edition Web Manager interface the service user can access: ***Own User***, ***User***, ***Group***, ***Auto Attendant***, ***System***, ***Dashboard (Home)***, ***Trunks***, ***SIP Trunks***, ***Backup***, ***Restore***, ***Upgrade***, ***Registration***, ***Licenses***, ***Auxiliary Equipment***, ***User Preferences***, ***System Status*** and ***Automatic Route Selection***.

- **Link Phone User**
  This field is used to associate the service user account with an extension on the system.

- **Password**
  This column provides controls to either change or reset the service user's password.

3.6.7 License

This menu is used to enter and display the licenses that the system has. The menu is access by selecting **System** in the menu bar and clicking on **Licenses**.

The set of PLDS licenses for a system are supplied in a non-editable XML file. This file is uploaded to the system. To add or delete a PLDS license requires replacement of the whole file. The license file is unique to the PLDS Host ID which is based on the feature key serial number of the System SD card installed in the system prefixed with 11. This host ID is shown in an information panel to the right of the license list.

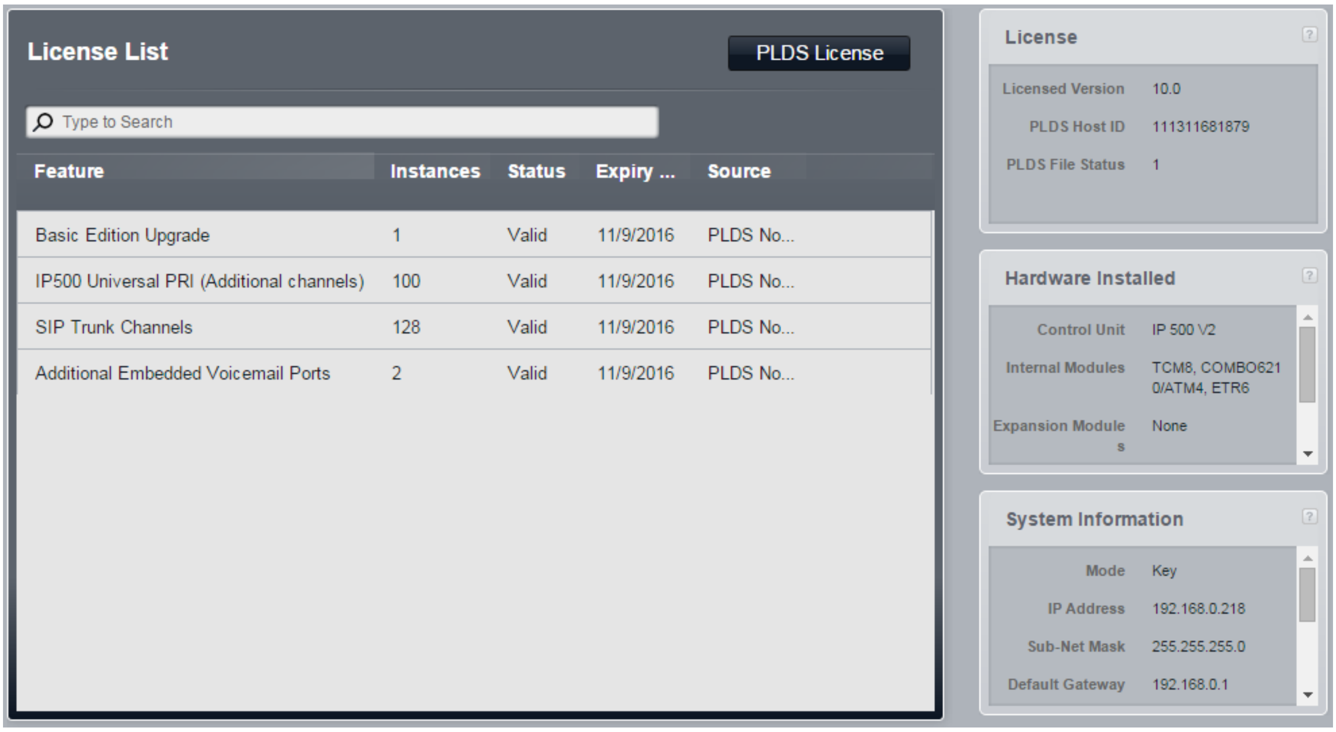
Licenses

Licenses are required for some features. The license keys are entered into the system configuration and are based on the unique Feature Key number of the System SD card installed in the system and the feature being enabled.

- **SIP Trunk Channel Licenses**
  The system can support 3 simultaneous SIP calls without needing licenses. Additional simultaneous calls, up to 20 in total, require the addition of licenses to the configuration.
  - **VCM Channels**
    Note that for SIP calls the system also requires VCM channels. For a system those are provided by installing IP500 Combination base cards. Each of these cards (up to 2) provides 10 VCM channels.
- **IP500 PRI Channel Licenses**
  The IP500 PRI 1 trunk daughter card supports the use of its first 8 channels unlicensed. Use of additional channels require licenses to be added to the configuration. The maximum number of channels depends on the current **Line Sub-Type** setting of the PRI trunk.
- **Embedded Voicemail Additional Ports**
  Unlicensed, the embedded voicemail provided by the system supports 2 simultaneous connections and 15 hours of storage. This can be expanded up to 6 channels by the addition of licenses, each of which enables an additional two channels. Each license also enables an additional 5 hours of storage.

License List

For each license key in the PLDS XML file, the following information is displayed:



- **Feature:** *Information field, not editable.*
  The name of the feature licensed.
- **Instances:** *Information field, not editable.*
  This field typically indicates how many items are enabled by the license. The meaning of this varies depending on the feature being licensed.
- **Status:**
  This field shows the status of the license.
  - ***Unknown*** is shown for newly entered licenses until the configuration is sent to the phone system and then reloaded again.
  - ***Valid*** is shown if the license key matches the System SD card's Feature Key number.

  - ***Invalid*** is shown if the license key does not match the SD card serial number.
  - ***Dormant*** is shown if the license key is valid but is conditional on another license that is not present.
  - ***Obsolete*** is shown if the license key is valid but the license is no longer used by the version of software installed in the phone system.
- **Expiry Date:** *Information field, not editable.*
  Some licenses have an expiry date, for example trial licenses. This field will indicate that date.
- **Source:** *Information field, not editable.*
  Indicates the type of license.

3.6.8 System Shutdown

The system should always be shutdown using the following process before it is switched off. This ensure that system actions such a file writes are completed before the power is removed. It also ensure that the current configuration in the system's memory is backed up to its System SD card.

The shut down can be either indefinite or for a set period of time after which the system will automatically reboot. This command performs a indefinite system shutdown. To restart the system power will need to be removed and then reapplied. To performed a timed shutdown refer to Shutting Down the System 233.

! WARNINGS

- A shutdown must always be used to switch off the system. Simply removing the power cord or switching off the power input may cause errors.
- This is not a polite shutdown, any calls and services in operation will be stopped. Once shutdown, the system cannot be used to make or receive calls until restarted.
- The shutdown process takes up to a minute to complete. When shutdown, the CPU LED and the IP500 base card LEDs 1 and 9 (if trunk daughter card fitted) will flash red rapidly. The memory card LEDs on the rear of the control unit are extinguished. Do not remove power from the system or remove any of the memory cards until the system is in the this state.
- To restart a system when shutdown indefinitely, or to restart a system before the timed restart, switch power to the system off and on again.

3.7 Monitoring

This menu give access to a series of commands for system maintenance. These are:

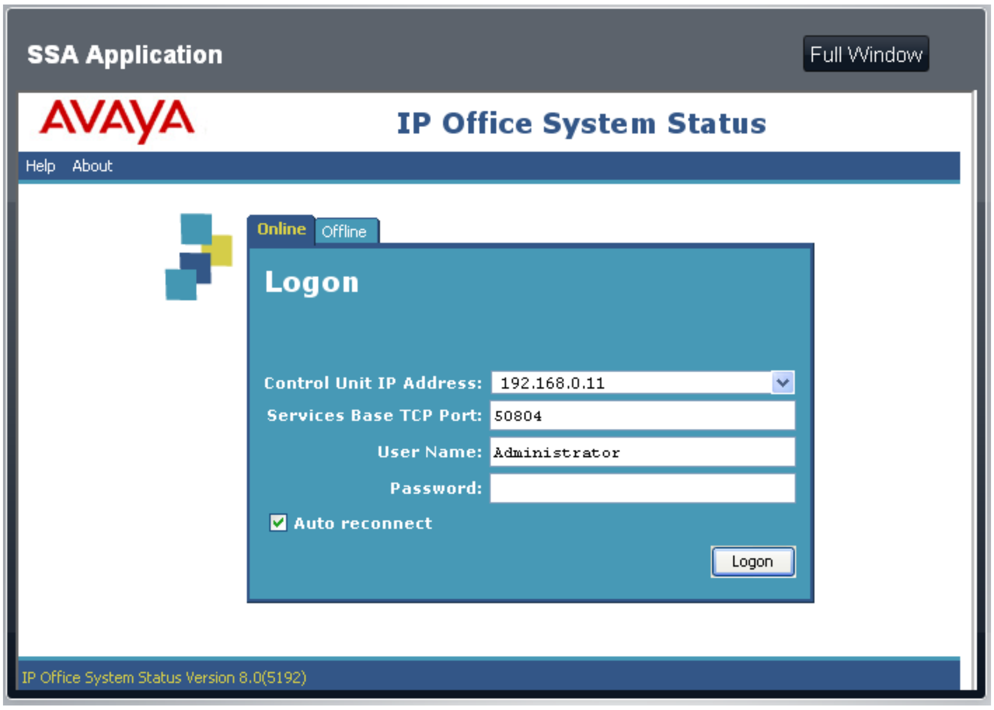
- **System Status** 121
- **Upload Configuration** 121
- **Erase Security Settings** 121
- **Erase Configuration** 121
- **Memory Card Start** 121
- **Memory Card Stop** 121
- **Copy to Optional SD** 121

The additional system maintenance commands, **Reboot** 124 and **System Shutdown** 120 are also available from other menus.

3.7.1 System Status

This menu is accessed by selecting **Monitoring** from the menu bar and clicking on **System Status**. IP Office System Status is a separate application from Basic Edition Web Manager. However it can be launched from within the Basic Edition Web Manager pages and uses the same login information. Not supported with Google Chrome.

- If using Internet Explorer 8, when logging in you may see a warning asking *"Do you want to view only the webpage content that was delivered sercurely?"*. Select ***No***. If you select ***Yes***, the System Status page within web management will be blank.



To Run IP Office System Status in a Full Window

The **Full Window** button can be used to start IP Office System Status in a separate browser window.

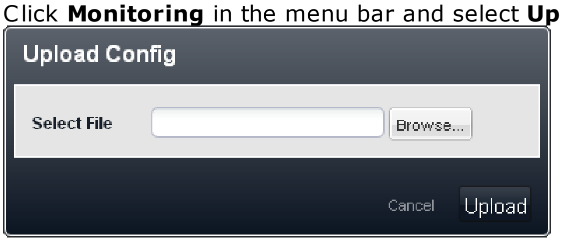
To Run IP Office System Status as a Separate Application

1. Start you web browser.
2. Enter the IP address of the system in the format *http://<IP Address>*, for example ***http://192.168.42.1***.
3. The web page shown displays a number of links, select the ***System Status*** link.

3.7.2 Upload Configuration

A configuration file can be uploaded to the system. Configuration files can be created using IP Office Manager.

1. Click **Monitoring** in the menu bar and select **Upload Config**.

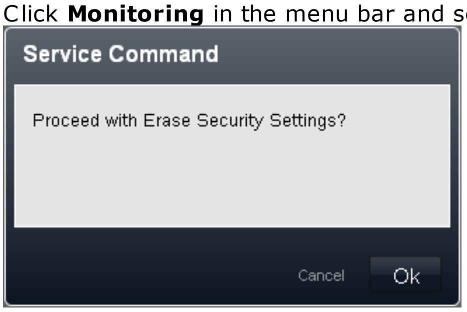


2. Use the **Browse...**. button to locate and select the configuration file to be uploaded.
3. Click **Upload**.

3.7.3 Erase Security Settings

This command will return all security settings back to their defaults, including deleting any additional service users that have been created through Role Based Rights.

1. Click **Monitoring** in the menu bar and select **Erase Security Settings**.



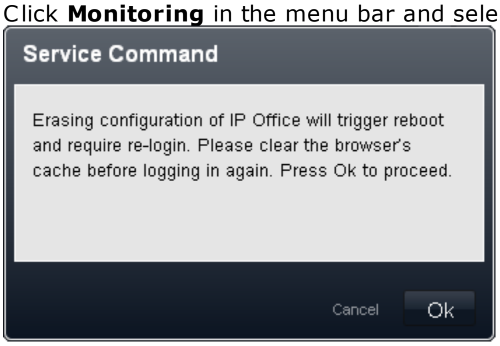
2. Click **OK**.

3.7.4 Erase Configuration

This command returns the configuration settings of the system back to their default values. This action does not affect the system's security settings or audit trail record. This command can also be performed from either of the first two extensions in the system using the **Restart -Defaults** command, see Phone Based Administration 25.

- **! WARNING**
  This command will reset all settings, including erasing all licenses. This command should only be used if you have a copy of the system configuration.

1. Click **Monitoring** in the menu bar and select **Erase Configuration**.



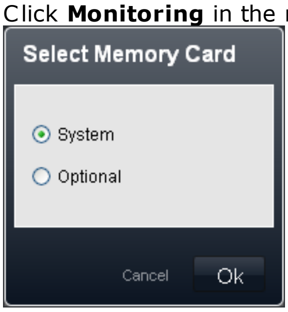
2. Click **OK**.

3.7.5 Memory Card Start

If a memory card has been stopped/shutdown, it needs to be restarted for the system to recognize and use the card.

If the card has been removed from the system's control unit, it is automatically restarted when it is reinserted into the control unit. This process can be used if the shutdown card has remained in the control unit after being stopped.

1. Click **Monitoring** in the menu bar and select **Memory Card Start**.



2. Click **OK**.

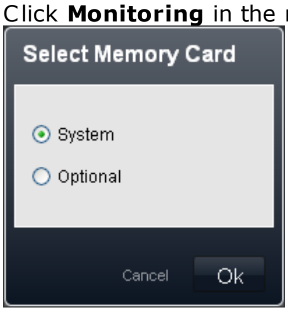
3.7.6 Memory Card Stop

Rather than shutting down the whole system, it is possible to just shut down the system's memory card. Once shutdown the card can be removed from the system to perform action such as load additional files onto the card or copy files from the card.

Shutting down the System SD card will halt voicemail services including user mailboxes and auto attendants. In addition, because the System SD card is used to validate licenses, licensed features will only operate for a further 2 hours before also halting.

A memory card that has been shutdown can be restarted using IP Office System Status or Basic Edition Web Manager. If the card has been removed from the system's control unit, it is automatically restarted when it is reinserted into the control unit.

1. Click **Monitoring** in the menu bar and select **Memory Card Stop**.



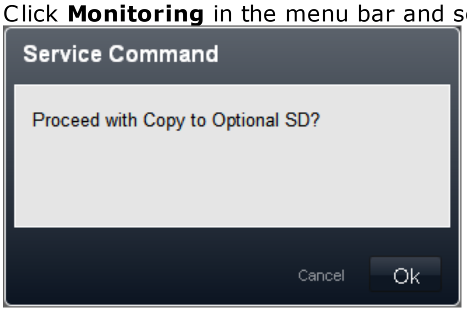
2. Click **OK**.

3.7.7 Copy to Optional SD

If memory cards are present in both the **System SD** and **Optional SD** card slots , you can copy **all** the contents of the **System SD** card to the **Optional SD** card. Depending on the number of files, this process can take up to 30 minutes to complete.

New files added after the process is started or file changes made after the process is started may not be included. Therefore it is recommended that this command is used during an idle period if available, for example outside normal business hours.

1. Click **Monitoring** in the menu bar and select **Copy System Card**.

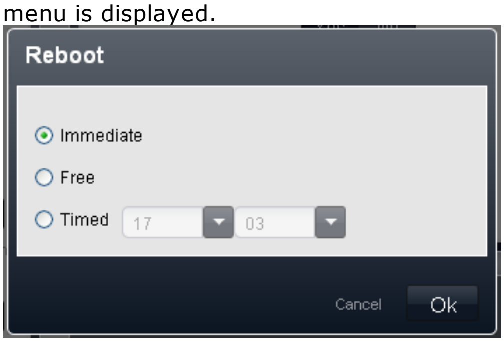


2. Click **OK**.

3.7.8 Reboot

Some changes require the system to be rebooted before the changes come into effect. Rebooting the system will end all calls currently in progress.

1. Click **Reboot**. The command is located in the top right of the Basic Edition Web Manager screen. The reboot menu is displayed.



2. Select the type of reboot required.

   - ***Immediate***
     If this option is selected, the reboot is started once **OK** is clicked. Any existing calls are ended without any warning.
   - ***Free***
     If this option is selected, after **OK** is clicked, the system will wait until there are no calls in progress before starting the reboot process.
   - ***Timed***
     If this option is selected, a time can also be selected for when the reboot should occur.

3. Click **OK**.

3.8 Tools

This menu provides access to a number of miscellaneous options.

3.8.1 About

This menu shows the version of the core software being run by the IP Office system. This is also shown on the **Home** 31 menu.

3.8.2 On-Boarding

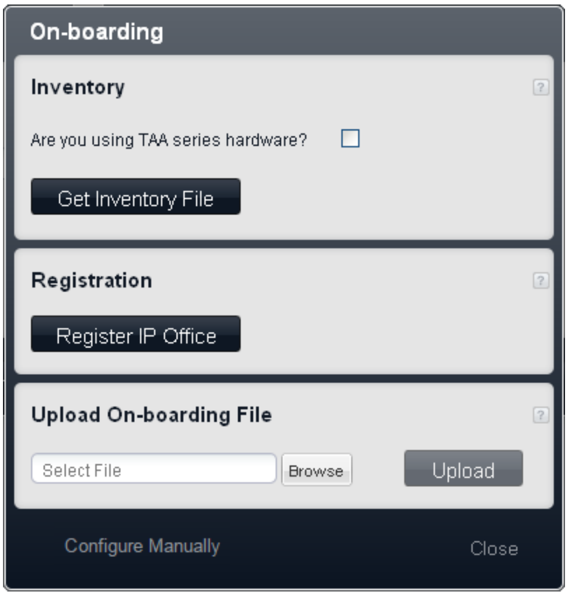
On-boarding is a process through which you can register the IP Office system for remote support and download a file that configures an SSL VPN service for the system. That service can then be used for remote system maintenance via an Avaya VPN Gateway (AVG) server.

- For information about how to configure and administer SSL VPN services, see the Avaya IP Office SSL VPN Solutions Guide. You can download the guide from http://support.avaya.com.

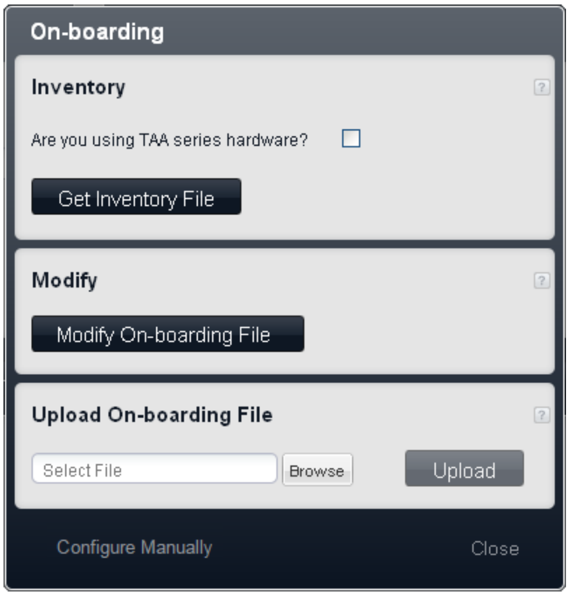
To go through the process of on-boarding, use this menu to:

1. Use **Get Inventory File** to download an inventory.xml file from the system.
2. Use **Register IP Office** to register the system with the the Avaya Global Registration Tool (GRT) website.
3. Once registered, download an on-boarding file for the system from the Avaya Global Registration Tool. This file contains the settings required to establish an SSL VPN connection between the IP Office system and an Avaya VPN Gateway (AVG) server.

4. Use the **Upload On-boarding File** section to upload the on-boarding file to the system. Note that the options in the menu vary depending on whether the system has been on-boarded already or not.



Unregistered System



Registered System

- **Inventory**
  Generate an inventory of the IP Office system. When you register the IP Office system for remote support, the inventory file is required as part of the registration and is uploaded to the Avaya Global Registration Tool (GRT) where the inventory data is populated in the Avaya Customer Support (ACS) database.
  - **Are you using TAA series hardware?**
    Systems purchased under US Federal Acquisition Regulations (FAR) must comply with the requirements of the Trade America Act (TAA). For various items of IP Office hardware there are TAA compatible variants. Select this option if the IP Office system includes TAA hardware. This is usually indicated by TAA appearing on the label on the back of the system control unit.
  - **Get Inventory File**
    Clicking this button downloads an ***inventory.xml*** file to the computer you are using to access web management.
- **Registration**
  This menu is section is shown for unregistered systems. You can register the IP Office system on the Avaya Global Registration Tool (GRT) web site and enable remote support. After you enable remote support for the system, you can download an on-boarding file from the GRT web site and import it into your IP Office system using the Upload On-boarding File section below.
  - **Register IP Office**
    Clicking this button opens a new browser window and loads the Avaya Global Registration Tool (GRT) web site.
- **Modify**
  This menu section is shown for registered systems.
  - **Modify On-boarding File**
    When you click **Modify On-boarding File**, you are prompted to download the on-boarding file from the Avaya web site
- **Upload On-boarding File**
  Once a system has been registered on the Avaya Global Registration Tool (GRT) web site you can download an on-boarding file for that system from the site. That file then needs to be uploaded to the system to allow remote support.
  - **Select File/Browse**
    Use this field and the **Browse** button to enter the file path for the on-boarding file.
  - **Upload**
    Once you have selected the on-boarding file, clicking this button will upload the file to the system.

3.9 Information Panels

The panels, up to 3, on the right display summary information for different aspects of the system configuration and vary depend on the current menu selected.

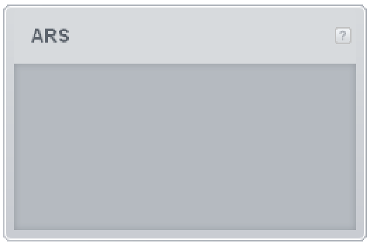
- **Accessing Panel Help**
  Within each panel you can click the help icon to display a summary of the panel and to access further help information.

- **Editing Panel Settings**
  Some panels also include an edit icon. This can be used to access the appropriate configuration settings.

The various information panels that are displayed are:

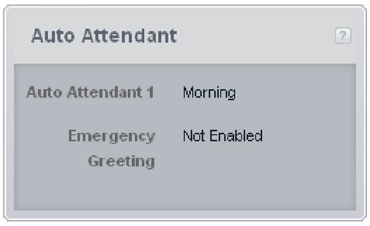
• ARS 126	• Features Configured 128	• SIP Trunks 131
• Auto Attendant 126	• Groups 128	• Speed Dial Setup 131
• Call by Call 126	• Hardware Installed 129	• System Information 130
• Channel Setup 127	• Incoming Calls 129	• Role Based Rights 130
• Dial Plan 127	• Incoming Number Filter 129	• Trunks in Service 131
• DID Mapping Table 127	• License 129	• Users 131
• Do Not Disturb Exceptions 127	• Outgoing Calls 130	

3.9.1 ARS



This panel is used to display a summary of the system's Alternate Route Selection settings.

3.9.2 Auto Attendant



This panel gives a summary of the auto attendant services (up to 9) currently configured.

- **Auto Attendant**
  For each configured auto attendant, this item lists the current service being provided by the auto attendant. Each auto attendant can be configured with different greeting and options for morning, afternoon, evening and out of hours periods.

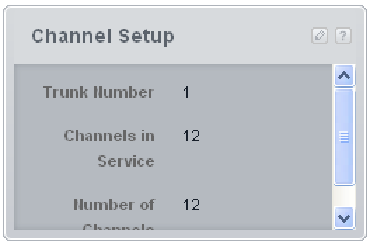
- **Emergency Greeting**
  For each auto attendant, this item lists whether the auto attendant is currently set to play its emergency greeting or not.

3.9.3 Call by Call

This panel shows a summary of the call by call settings configured for the currently selected SIP trunk in the SIP Trunks 98 menu.

The edit icon can be used to access the Call by Call Settings 107 menu in order to edit the settings.

3.9.4 Channel Setup

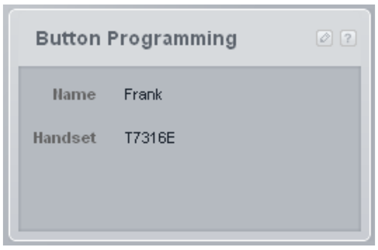


This panel give a summary of the configured channels of the currently selected BRI or PRI trunk in the Trunks 73 menu. It is also used for the currently selected SIP trunk in the SIP Trunks 98 menu.

The edit icon can be used to access the channel setup menu for the selected trunk. The menu that appears and the options in the menu will depend on the trunk type:

- **PRI Channel Setup**
- **T1 Channel Setup**
- **PRI (ETSI) Channel Setup**
- **BRI Trunk Channel Setup**
- **SIP Trunk Channel Setup** 103

3.9.5 Default Button Programming

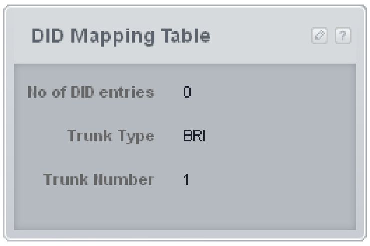


This panel shows a summary of the user's current programmable button settings. Programmable buttons are supported on Avaya phones. They are not supported on traditional analog phones.

The panel is shown on the User 33 menu. The information changes to match the currently selected user in that menu. The edit icon can be used to access the Button Programming menu to edit the user's settings.

- **Name**
  The name of the extension user.
- **Handset**
  If a phone is currently connected to the user's extension, the type of phone is known is indicated here.

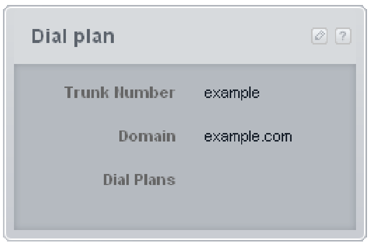
3.9.6 DID Mapping Table



This panel give a summary of the configured DID mapping for the currently selected BRI or PRI trunk in the Trunks 73 menu.

The edit icon can be used to access the DID Mapping menu for the selected trunk.

3.9.7 Dial Plan



This panel shows a summary of the dial plans configured for the currently selected SIP trunk in the SIP Trunks 98 menu.

The edit icon can be used to access the Dial Plan Settings 109 menu in order to edit the settings.

3.9.8 Do Not Disturb Exceptions

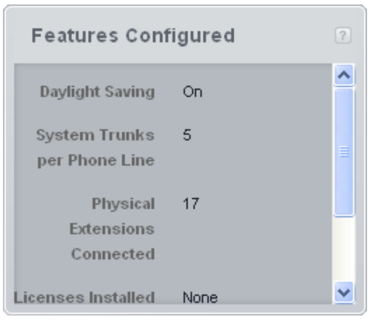
This panel shows a summary of the user's do not disturb settings. The panel is shown on the User 33 menu. The information changes to match the currently selected user in that menu.

The edit icon can be used to access the DND Exception List 36 menu to edit the user's settings.

- **DND Enabled**
  This item indicates whether do not disturb is currently enabled. DND can be enabled through editing the user configuration or through functions on the user's phone.

- **DND Exceptions**
  This item lists the numbers, calls from which, are able to still call the user when they have do not disturb enabled.

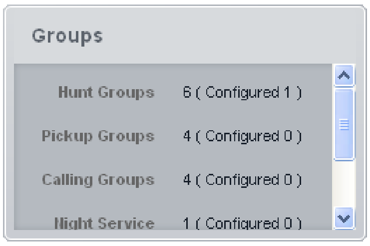
3.9.9 Features Configured



This panel summarizes some key features of the current system configuration.

- **Daylight Saving**
  This item indicates whether automatic daylight saving time is enabled. This option is only supported for systems in North American countries.
- **System Trunks per Phone Line**
- **Physical Extensions Connected**
  This item lists the number of actual extensions connected. Note that for some extension types, for example analog extensions, the system assumes that an extension is connected if a physical port for connection is installed in the system's control unit.
- **Licenses Installed**
  This item list the number of licenses [118] added to the system configuration.
- **Trunks per Phone**
  For system's running in Key mode, a number of trunks are automatically assigned to programmable buttons on each user's extension if possible. This is controlled by the **Number of Lines** setting on the System [65] menu.
- **Night Service**
  This item indicates whether the system is currently in night service or not. In night service, the incoming call routing applied to calls is changed.

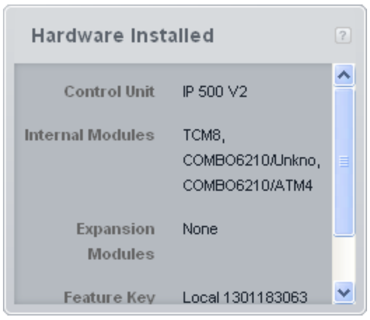
3.9.10 Groups



This panel gives a summary of the hunt groups that have been configured, that it the groups that contain users as their members. The number and type of hunt groups supported by the system is fixed. The groups can be edited through the Groups [46] menu.

- ***Calling Group*** [167]
  This type of group can be set as the **Coverage Destination** of a trunk or trunk channel. It can be also the destination of calls routed using DID call mapping or call-by-call settings. Calling Group 1 is also used by the **Simultaneous Page** function (***70**).
- ***Hunt Group*** [167]
  This type of group can be set as the **Coverage Destination** of a trunk or trunk channel. It can also be the destination of calls routed using DID call mapping or call-by-call settings.
- ***Night Service Group*** [167]
  When the system is put into night service, this group overrides the **Coverage Destination** of all trunks.
- ***Operator Group*** [167]
  This option is only available for systems with their **Mode** set to ***PBX***. By default the group contains the first extension on the system. For PRI and BRI trunks, it is fixed incoming destination for calls unless DID Mapping is applied to the call. It can also be selected as the destination for incoming SIP calls.
- ***Pickup Group*** [167]
  Users can be pickup calls currently alerting any member of a pickup group. They do not need to be a member of the group.

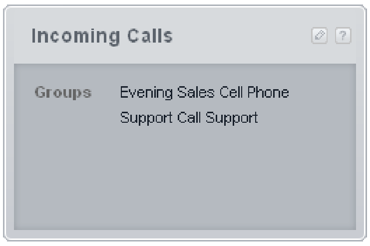
3.9.11 Hardware Installed



This panel shows a summary of the key system hardware. The panel is shown on the Home 31 page. It is also shown on the Alternate Route Selection 59 and System 65 menus.

- **Control Unit**
  This is the type of system control unit from which the configuration has been received.
- **Internal Modules**
  This item lists the internal cards installed in the control unit. IP500 V2 control unit have 4 card slots. Each can be fitted with a IP500 base card which typically provides connections for up to 8 extensions. In most cases each base card can also be fitted with a trunk card which provides support for physical trunk connections to the base card.
- **Expansion Modules**
  This item lists any external expansion modules attached to the control unit. These are used to provide ports for the connection of additional extension and trunks. An IP500 V2 control unit has ports for connecting up to 8 external expansion modules as long as the systems number of supported extensions is not exceeded.
- **Feature Key**
  This item shows the feature key number of the feature key dongle installed in the control unit. This number is used to check and validate any licenses 118 added to the system configuration. For IP500 V2 control units, the feature key dongle is the System SD card installed in the back of the control unit. The feature key number is also printed on the card label.
- **Serial Number**
  This item is the unique serial number of the control unit.

3.9.12 Incoming Calls



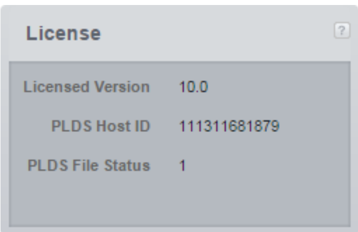
This panel shows a summary of the incoming call routing. The panel is shown on the Calling Lists 55 menu.

3.9.13 Incoming Number Filter

This panel shows a summary of the incoming number filters configured for the currently selected SIP trunk in the SIP Trunks 98 menu.

The edit icon can be used to access the Incoming Number Filters 98 menu in order to edit the settings.

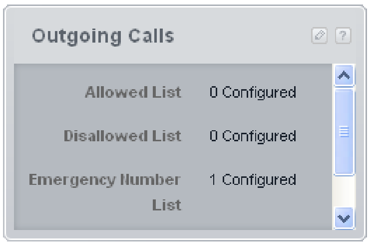
3.9.14 License



This panel shows a summary of the key licensing information:

- **Licenses Version**
  This is the version of software for which the system is currently licensed.
- **PLDS Host ID**
  This is the host ID used to validate the licenses. This is based on the feature key number of the system SD card prefixed with 11.
- **PLDS File Status**
  Indicates if a valid PLDS license file is present.

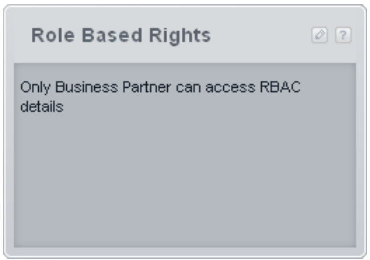
3.9.15 Outgoing Calls



This panel a summary of the configuration of the lists that are used to control which numbers users can dial when making outgoing calls. The edit icon can be used to access the List Management 55 menu to edit the settings.

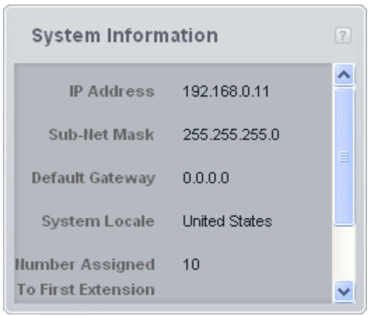
- **Allowed List**
  Allowed lists are used to enter numbers or types of numbers that users associated with the list can dial even if they are restricted from dialing other numbers. Up to 8 such lists, each containing 10 numbers, can be configured.
- **Disallowed List**
  Disallowed lists are used to enter numbers or types of numbers that users associated with the list cannot dial. Up to 10 such lists can be configured. Up to 8 such lists, each containing 10 numbers, can be configured.
- **Emergency Number List**
  This list is used to enter numbers that all users can dial at any time regardless of any other settings that might restrict them from dialing numbers for outgoing calls. Up to 10 numbers can be configured in this list.
- **Account Codes**
  Up to 99 account codes can be entered. In addition selected users can be configured to have to enter an account code whenever they make an outgoing external call.

3.9.16 Role Based Rights



This panel is used to access the security settings for accounts to use web based management. Clicking the edit icon accesses the Service Users 116 menu.

3.9.17 System Information



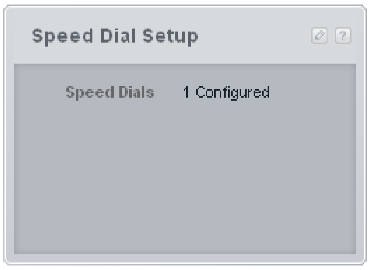
This panel provides a quick summary of the key system hardware. Most of the settings shown in this panel can be altered through the System | System 65 menu if required.

- **IP Address**
  This is the IP address assigned to the LAN port on the back of the control unit. The address is normally obtained by the system using DHCP or it can be set manually. Otherwise the address defaults to 192.168.42.1.
- **Sub-Net Mask**
  This is the sub-net mask associated with the IP address above.
- **Default Gateway**
  This is the default gateway address associated with the IP address above.
- **System Locale**
  This is the Country setting of the system. It affects a range of feature settings including many that are not user configurable and related to trunk operation. Therefore it is important to ensure that this setting is correct for the system's actual location.
- **Number Assigned to First Extension**
  This is the extension number of the first extension port in the system. That is the port that is on the top-left of the first internal card slot in the control unit (regardless of whether a card is installed in that port or not). For further details of the extension numbering options refer to Dial Plan 14.
- **Total Number of Extensions**
  This item shows the number of extensions supported by the system. Systems using 2 digit extension numbering support 48 extensions. Systems using 3 digit extension numbering support 100 extensions.

3.9.18 SIP Trunks

This panel displays a summary of the SIP trunks in the system's configuration. The edit icon can be clicked to access the SIP Trunks 98 menu.

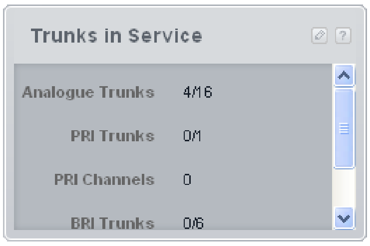
3.9.19 Speed Dial Setup



This panel shows a summary of the current system speed dials configured. The edit icon can be used to access the Speed Dial 54 menu in order to edit system speed dials.

- **Speed Dials**
  This item shows the number of system speed dial numbers that have been configured. Up to 100 system speed dials can be configured.

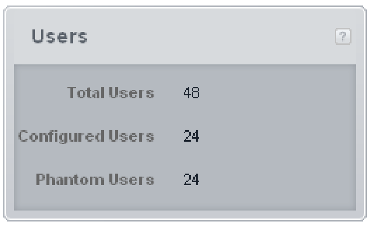
3.9.20 Trunks in Service



This panel shows a summary of the trunks present and the maximum number of trunks that could be added. SIP trunks are not included.

- **Analogue Trunks**
  This items shows the number of actual analogue trunk ports installed in the system and the maximum number of such trunk ports that could be installed.
- **PRI Trunks**
  This item shows the number of PRI trunk ports installed in the system and the maximum number of such trunk ports that could be installed. Note that both PRI and BRI trunks cannot be installed in the same system.
- **PRI Channels**
  Each PRI trunk can support multiple channels. The number of channels is shown here. Note that the maximum number of channels is dependent on the particular configuration of the PRI trunk.
- **BRI Trunks**
  This items shows the number of actual BRI trunk ports installed in the system and the maximum number of such trunk ports that could be installed. Note that both PRI and BRI trunks cannot be installed in the same system.
- **BRI Channels**
  Each BRI trunk can support up to 2 channels. This items shows the number of BRI channels configured and the maximum number that could be configured.

3.9.21 Users



This panel gives a summary of the number of users supported by the system. The panel is shown on the Groups 46 menu and the Calling Lists 55 menu.

- **Total Users**
  This item shows the number of extensions supported by the system. Systems using 2 digit extension numbering support 48 extensions. Systems using 3 digit extension numbering support 100 extensions.
- **Configured Users**
  This item shows the number of actual physical extension ports present in the system.
- **Phantom Users**
  This items shows the number of users present in the configuration who are not matched by physical extension ports to which a phone can be connected. These users can still be configured and used for some functions.

Chapter 4.
Initial Configuration

4. Initial Configuration

This section covers the recommended initial configuration actions for a new system. Many of the settings are set by default, however they should still be checked.

Initial Configuration Processes

- **! Warning**
  The processes marked **(!)** in this list are ones which, if the setting is changed, require the system to be restarted in order for the new setting to take effect. Changing them may also cause other settings to reset back to default values. These are additional reasons why these settings should be sets as part of initial system configuration.

1. **Set the System Mode** 135 **(!)**
   The system can operate in either ***PBX*** or ***Key*** mode.
2. **Set the Country** 136 **(!)**
   The correct country setting sets a range of internal settings, especially relating to the operation of trunks, that are otherwise not adjustable through the configuration.
3. **Set the Default Language** 137 **(!)**
   The system's language for phone displays and voicemail prompts default to the best match to the country setting. However it should still be checked.
4. **Set the Number of Lines** 138
   This option is used for ***Key*** mode systems. If changed it will overwrite existing button programming.
5. **Set the Outside Line Prefix** 143 **(!)**
   This option is used for ***PBX*** mode systems. A prefix is not required but 0 or 9 can be used if required.
6. **Adding Licenses** 139
   The use of and capacity of some features requires licenses added to the configuration.
7. **Change the Network Settings** 140 **(!)**
   By default, if connected to a customer network the system requests IP address settings as a DHCP client.
8. **Set the Emergency Numbers** 141
   The correct emergency numbers for the country must be set to ensure that they are excluded from any outgoing call restrictions that may be setup later.
9. **Select Music on Hold** 143
10. **Adjust Automatic Line Selection** 144
    For users on a Key mode system, if the user simply goes off-hook to make a call, the system needs to use automatic line selection to determine which of the user's available line or intercom buttons is used for the call.

4.1 Setting the System Mode (PBX or Key)

The system can operate in either of two modes; ***PBX*** or ***Key***. The selected mode affects the system's outgoing call routing [17] and incoming call routing [18] settings.

Default Setting

The default setting for the system's **Mode** is determined by the type of SD card installed in the system.

- **IP Office U-Law**
  A system fitted with this type of card will default to U-Law telephony and ***Key*** mode operation. Intended for North American locales. System's running in this mode are referred to as IP Office Basic Edition systems.
- **IP Office A-Law**
  A system fitted with this type of card will default to A-Law telephony and ***PBX*** mode operation. Intended for locales outside North America. System's running in this mode are referred to as IP Office Basic Edition systems.
- **IP Office Partner Mode**
  A system fitted with this type of card will default to U-Law telephony and ***Key*** mode operation. Supported only in North American locales. System's running in this mode are referred to as IP Office Basic Edition - PARTNER Mode systems.
- **IP Office Norstar Mode**
  A system fitted with this type of card will default to A-Law telephony and ***Key*** mode operation. Supported only in Middle East and North African locales. System's running in this mode are referred to as IP Office Basic Edition - Norstar Mode systems.

Changing the System Mode

- **! WARNING - Reboot Required**
  Changing this setting requires the system to be rebooted [12] for the change to take effect. Rebooting the system will end all calls currently in progress.
- In addition, any existing button programming is removed and all buttons are defaulted according to the requirements of the selected mode.

1. Click on **System** in the menu bar and then click on **Switch**.
2. Change the currently selected **Mode** to the required setting; ***PBX*** or ***Key***.
   - **Key**
     The **Number of Lines** setting is used to automatically assign line appearance buttons on all extensions with programmable buttons. To make external calls the user should select an available line appearance button. Outbound call routing is determined by which line appearance button the user selects before dialing or by the user's automatic line selection settings.
   - **PBX**
     No line appearances are automatically assigned to programmable buttons. The **Outside Line** setting is used to set the dialing prefix that indicates that the call is an external one for which an available line should be seized. The **Alternate Route Selection** [59] settings are used to determine which lines are used for each outgoing call. Line appearance buttons can also still be configured for making and answering external calls.
3. Click **Save**.

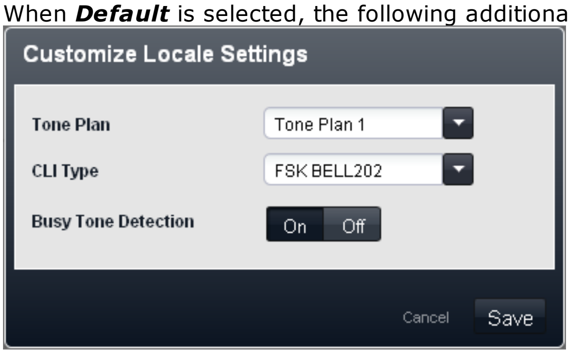
4.2 Setting the System Country

The system's country setting must be correctly set. It is used to adjust the system operation to match the requirements of telephone service providers and users in that country. Not setting the country correctly may cause problems.

- **! WARNING - Reboot Required**
  Changing this setting requires the system to be rebooted 12 for the change to take effect. Rebooting the system will end all calls currently in progress.

Setting the System Country

1. Click on **System** in the menu bar and then click on **Switch**.
2. The **Country** field is used to select the country.
   - The supported countries are ***Argentina***, ***Australia***, ***Bahrain***, ***Belgium***, ***Brazil***, ***Canada***, ***Chile***, ***China***, ***Customize***, ***Denmark***, ***Egypt***, ***Finland***, ***France***, ***Germany***, ***Greece***, ***Hong Kong***, ***Hungary***, ***Iceland***, ***India***, ***Italy***, ***Korea***, ***Kuwait***, ***Mexico***, ***Netherlands***, ***New Zealand***, ***Norway***, ***Oman***, ***Pakistan***, ***Peru***, ***Poland***, ***Portugal***, ***Qatar***, ***Russia***, ***Saudi Arabia***, ***Singapore***, ***South Africa***, ***Spain***, ***Sweden***, ***Switzerland***, ***Taiwan***, ***Turkey***, ***United Arab Emirates***, ***United States***, ***Venezuela***.
   - When ***Default*** is selected, the following additional fields are available:



     - **Tone Plan:** *Default = Tone Plan 1*
       Select a tone plan to be used for different ringing signals such as dial tone and ring tone.
     - **CLI Type:** *Default = FSK V23*
       Set the method for passing caller ID information to analog extensions. The options are ***DTMF***, ***FSK Bell 202*** or ***FSK V23***.
     - **Busy Tone Detection:** *Default = Off*
       Enable or disable the use of busy tone detection for call clearing.
3. Click **Save**.

4.3 Setting the System Language

Changing the system's country setting 136 also automatically changes the systems language to the best match. The language is used as follows:

- The messages and menus displayed on phones will be changed to match the language if possible.
- The language used by the systems voicemail services is changed to match the system language if possible.
- For each user, their own language settings can be changed using the user's language 198 setting. This affects the language used on their phone's display and for mailbox access prompts.
- For each auto attendant, the system language setting can be overridden by the auto attendant's own language 182 setting.

Warning: Installed and Available Languages

By default not all languages are included in the voicemail/auto attendant prompt files on the system. If the language required is not present, either ***UK English*** or ***US English*** is used. Additional languages can be loaded using IP Office Manager, they cannot be loaded using Basic Edition Web Manager. The languages present by default depend on the type of System SD card used by the system:

- **IP Office A-Law SD Card:** UK English, French and Spanish.
- **IP Office U-Law SD Card:** US English, Candian French and Latin Spanish.
- **PARTNER SD Card:** UK English, French and Spanish.
- **Norstar SD Card:** UK English, French, Arabic.

Setting the System Language

- **! WARNING - Reboot Required**
  Changing this setting requires the system to be rebooted 121 for the change to take effect. Rebooting the system will end all calls currently in progress.

1. Click on **System** in the menu bar and then click on **Switch**.
2. The **Language** field is used to select the system language. Possible languages are:
   - ***Arabic***, ***Cantonese***, ***Danish***, ***Dutch***, ***English (UK)***, ***English (US)***, ***Finnish***, ***French***, ***French (Canada)***, ***German***, ***Italian***, ***Korean***, ***Mandarin***, ***Norwegian***, ***Polish***, ***Portuguese***, ***Portuguese (Brazil)***, ***Russian***, ***Spanish***, ***Spanish (Argentina)***, ***Spanish (Latin)***, ***Spanish (Mexico)***, ***Swedish***, ***Turkish***.
3. Click **Save**.

4.4 Setting the Number of Lines

For systems with their **Mode** 135 set to ***Key***, when the system's **Number of Lines** setting is changed, the following other changes to the configuration occur:

- The number of line appearance buttons set on all user extensions is reset to match the Number of Lines values. The buttons are assigned from button 03 upwards and will overwrite any existing buttons that are set to become line appearance buttons.
- The user's automatic line selection settings are reset to match the number of lines.

When a system is first installed, the **Number of Lines** setting is automatically set to match the number of analog trunks present in the system. This means that all analog lines are automatically added as line appearances and added to the automatic line selection settings of users. If no analog trunks are present when the system is installed, the setting defaults to the first 5 lines.

Changing the Number of Lines Setting

- **! Warning**
  If the **Number of Lines** value is changed, all existing line appearance buttons and automatic line selection settings are overwritten. The existing functions on other programmable buttons are also overwritten if they are in the range of buttons now specified for lines. Therefore it is recommended that this setting is only changed when a system is first installed.

1. Click on **System** in the menu bar and then click on **Switch**.
2. In the **System Parameters** panel, change the **Number of Lines** setting to the required value.
3. Click **Save**.

4.5 Adding Licenses

Licenses are required for some features. The license keys are entered into the system configuration and are based on the unique Feature Key number of the System SD card installed in the system and the feature being enabled.

- **SIP Trunk Channel Licenses**
  The system can support 3 simultaneous SIP calls without needing licenses. Additional simultaneous calls, up to 20 in total, require the addition of licenses to the configuration.
  - **VCM Channels**
    Note that for SIP calls the system also requires VCM channels. For a system those are provided by installing IP500 Combination base cards. Each of these cards (up to 2) provides 10 VCM channels.
- **IP500 PRI Channel Licenses**
  The IP500 PRI 1 trunk daughter card supports the use of its first 8 channels unlicensed. Use of additional channels require licenses to be added to the configuration. The maximum number of channels depends on the current **Line Sub-Type** setting of the PRI trunk.
- **Embedded Voicemail Additional Ports**
  Unlicensed, the embedded voicemail provided by the system supports 2 simultaneous connections and 15 hours of storage. This can be expanded up to 6 channels by the addition of licenses, each of which enables an additional two channels. Each license also enables an additional 5 hours of storage.

Checking the System Feature Key Number/PLDS Host ID

The set of PLDS licenses for a system are supplied in a non-editable XML file. This file is uploaded to the system. To add or delete a PLDS license requires replacement of the whole file. The license file is unique to the PLDS Host ID which is based on the feature key serial number of the System SD card installed in the system prefixed with 11. This host ID is shown in an information panel to the right of the license list.

Adding Licenses

First check that the licence information that you have been supplied has been issued against the correct PLDS host ID. That number should match the Feature Key number of the System SD card installed in the system plus a prefix of 11. Licenses issued against another ID number will be invalid.

To upload the license file using Basic Edition Web Manager:

1. Select **System | License**.
2. The field **PLDS Host ID** shows the serial number of the System SD card fitted to the system prefixed with 11. Check that this number matches the one against which the licenses have been issued.
3. Click **PLDS License**.
4. Select **Send To IP Office** and click **OK**.
5. Browse to the XML license file that you have for the system and click **Upload**. Click **OK**.
6. The file is uploaded to the system and read. The menu should list the licenses and show their status as **Valid**.

4.6 Changing Network Settings

IP connection to the system is done using the **LAN** port on the back of the system's control unit. During installation, it uses the LAN port to request an IP address from any DHCP server. If there is a DHCP server on the customer's network, that server will give the system an IP address.

If the system was not able to get an address using DHCP when it was first started, it will use the default address ***192.168.42.1/255.255.255.0*** for the LAN port. However, the system is still defaulted as a DHCP client and so will request an address again if it is restarted. Therefore if the system has been started before being connected to the customer's network, it can still be connected and restarted in order to obtain an address from the network.

The **WAN** port on the back of the system's control unit is not normally used. Its only use if as a fallback method to connect a PC in order to configure the system, see PC Connection8.

Changing the System's Network Settings

1. Click on **System** in the menu bar.

2. The network address settings for the system's LAN port are shown in the **Network Settings** panel:

   - **Receive IP Address Via DHCP Server:** *Default = Yes.*
     This setting controls whether the system acts as a DHCP client or uses a fixed IP address.

     - If enabled, the system acts as a DHCP client and requests IP address details for its LAN port when the system is started.

       - If it receives a response, the address details it has been given by the DHCP server are shown in the field below but cannot be adjusted.

       - If it does not receive a response, it default to using the address 192.168.42.1. It is still a DHCP client and will request an address again when it is next restarted.

     - If not enabled, the system uses the IP address values set in the fields below.

   - **System IP Address:** *Default = 192.168.42.1*
     Enter the IP address that the telephone system should use if **Receive IP Address Via DHCP Server** is not selected. If **Receive IP Address Via DHCP Server** is selected, this field is greyed out but does display the IP address that the system is currently using.

   - **Subnet Mask:** *Default = 255.255.255.0*
     Enter the Sub-Net Mask that the telephone system should use if **Receive IP Address Via DHCP Server** is not selected. If **Receive IP Address Via DHCP Server** is selected, this field is greyed out but does display the IP address that the system is currently using.

   - **Default Gateway:** *Default = 0.0.0.0*
     Enter the **Default Gateway** that the telephone system should use if **Receive IP Address Via DHCP Server** is not selected. If **Receive IP Address Via DHCP Server** is selected, this field is greyed out but does display the IP address that the system is currently using.

3. Once the settings are set as required, click **Save**.

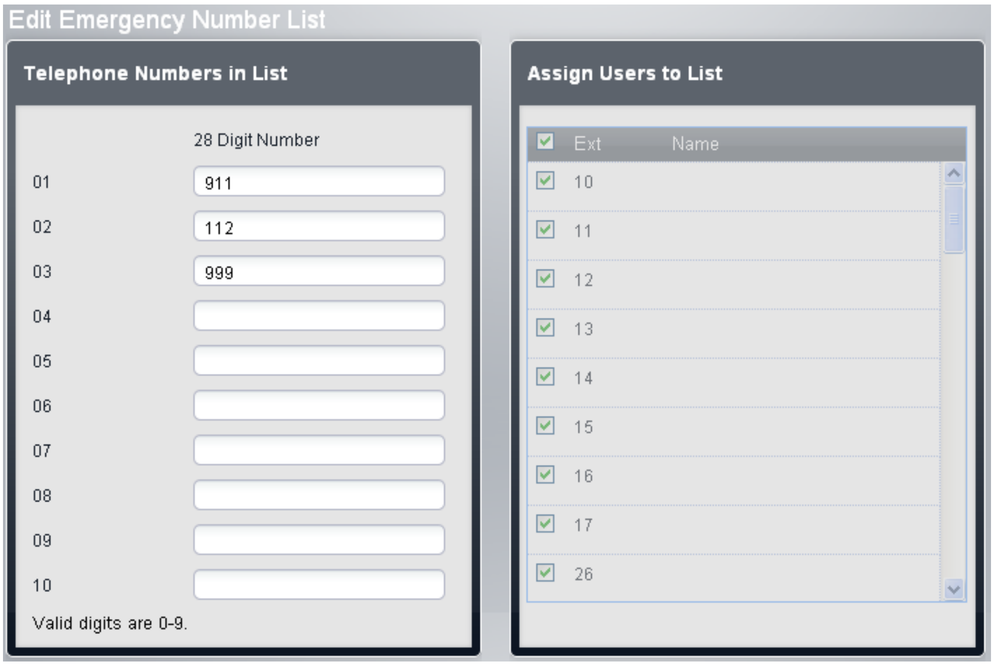
4.7 Setting the Emergency Numbers

You can enter 10 emergency phone numbers into this list. This list is applied to all users and overrides any dialing restrictions that may also be applied to the users. Use of numbers in the list can be viewed using a 911-View/E-View220 button.

By default the normal emergency numbers for the system locale are automatically added and should not be removed.

To Edit the Emergency Numbers List

1. From the menu bar, click on **User**.
2. The **Outgoing Calls** panel next to the list of users gives a summary of the currently configured lists. Click on the edit icon.
3. From the **List Management** table, select the **View Details** link of the list that you want to edit.



- The **Telephone Numbers in List** panel displays the allowed numbers. Edit the numbers as required.
- The **Assign Users to List** panel is used to set which users are assigned to the list.

4. When completed click **Save**.
5. To access another list click on **<< Previous List** or **Next List >>**. Alternatively click on **<< Back** to return to the table of all the lists.

4.8 Setting the Outside Line Prefix

This option is only used with systems set to PBX mode. It sets the digit which, when dialed at the start of a number, indicates that the call is intended to be external. The options are to use ***0***, ***9*** or no prefix.

Note that the setting also changes the digits used for calls to the first extension on the system. Normally, in addition to the extension's extension number, the number 0 can be used to call that extension. If the number 0 is set as the outside line prefix, the number 9 is used for the first extension.

Setting the System's Outside Line Prefix

1. Click on **System** in the menu bar.
2. In the **System Parameters**, set the **Outside Line** setting to the required option.
   - ***9 (Operator is 0)***
     The prefix 9 is used for external calls. The digit 0 is used for calls to the operator extension (the first extension in the system). This is the default setting for systems with the **Country** setting ***United States***.
   - ***None***
     No prefix is used for external calls. Any dialing that does not match an internal dial plan number is assumed to be an external call. This is the default setting for systems with the **Country** setting other then ***Germany*** or ***United States***. The digit 0 is used for calls to the operator extension (the first extension in the system).
   - ***0 (Operator is 9)***
     The prefix 0 is used for external calls. The digit 9 is used for calls to the operator extension (the first extension in the system). This is the default setting for systems with the **Country** setting ***Germany***.
3. Click **Save**.

4.9 Music on Hold

The phone system supports an external music on hold source. This connects to the **Audio** port on the rear of the system's control unit. You can configure whether the input to this port is played to callers when they are put on hold.

The music on hold input can also be played to callers being transferred rather than ringing tone. That behaviour is controlled by the system's Ring on Transfer 69 setting.

The port is a 3.5mm stereo jack socket suitable for use with the most standard audio leads and connection to the 'headphone' output socket of most audio systems. The use of a 'headphone' socket allows simple volume adjustment. Connection via a 'Line Out' socket may require additional equipment in order to adjust the volume level.

Enabling Music on Hold

1. Click **System** in the menu bar and then click **Auxiliary Equipment**.
2. In the **Music on Hold** panel, select the required option.
   - ***On***
     This is the default. If enabled, the system will use the external music source connected to the phone system for its music on hold.
   - ***Off***
     If not enabled, the system provides a double beep tone repeated every 5 seconds.
3. Click **Save**.

Using Music on Hold for Call Transfers

Calls being transferred normally hear ringing while the transfer process is in progress. This can be changed to hearing the system's music on hold source.

1. Click on **System** in the menu bar and then click on **Switch**.
2. Click on the **Advanced** button.
3. The **Ring on Transfer** setting controls whether callers hear ringing or music on hold while being transferred.
4. Click **Save**.

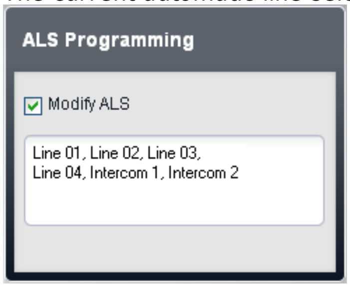
4.10 Automatic Line Selection

For systems running in **Key** mode, when a user makes a call, they can indicate if it is an external or internal call by first pressing a line appearance button or an intercom button respectively. If the user just lifts the handset without first pressing a button, the system uses the user's automatic line selection settings (ALS) to determine which available button to seize for the call.

By default, each extensions' automatic line selection list contains line buttons in sequence from line 1 up to the system's **Number of Lines** setting and then the intercom or call appearance buttons.

Manually Editing a User's Automatic Line Selection Setting

1. Click on **User** in the menu bar.
2. Highlight the required user by clicking on them.
3. Click on the edit icon in the **Button Programming** panel on the right.
4. The current automatic line selection settings are shown in the **ALS Programming** panel.



5. Select **Modify ALS**.
6. In the text box, enter the sequence of line and intercom buttons that should be use for automatic line selection. Separate each entry with a comma.
   - For a line button, enter ***Line XX*** where ***XX*** is replaced by the line number.
   - For an intercom button, enter ***Intercom Y*** when ***Y*** is replaced by the intercom button number.
7. Click **Save**.

Chapter 5.
Setting the Date and Time

5. Setting the Date and Time

The system has a clock which it uses to maintain its date and time. However it needs a source to set the initial date and time. This can either be done manually or using the information provided with certain types of telephone call.

5.1 Manually Setting the Date

The first two extensions on the system can be used to set the system date using phone based administration 25. The exact method will depend on the type of phone. For full details refer to the Phone Based Administration manual.

1408/1416/9404/9408/9504/9508 Phone

1. At either of the first two extensions on the system, press **Admin**.
2. Use the up or down arrow buttons to scroll the display to **System Administration**. When highlighted, press **Select**.
3. Use the up or down arrow buttons to scroll the display to **System Parameters**. When highlighted, press **Select**.
4. Use the up or down arrow buttons to scroll the display to **System Date**. When highlighted, press **Select**.
5. Enter the new system date in the format *MMDDYY* using the dial pad. For example 120409 is the date December 4th 2009.
6. Exit programming by pressing **PHONE**.

M7324/M7324N/M7310/M7310N/T7316/T7316E Phone

1. At either of the first two extensions on the system, press **Feature **config** (that is **Feature **266344**).
2. If the system has a system password set, it is requested. Enter the password.
3. The phone displays **System Admin**.
4. Dial **#101**. The phone will display **System Date**.
5. Enter the new system date in the format *MMDDYY* using the dial pad. For example 120409 would be used to enter the date December 4th 2009.
6. To exit phone based administration at any time, press the **Release** button.

ETR 18D/ETR 34D Phone

1. At either or the first two extensions on the system, press **Feature 0 0** followed by two presses of the first intercom or call appearance button.
2. **System Administration** is shown on the display.
3. Dial **#101**. The phone will display **System Date**.
4. Enter the new system date in the format *MMDDYY* using the dial pad. For example 120409 would be used to enter the date December 4th 2009.
5. Exit programming by pressing **Feature 00**. You can also exit programming mode by lifting the handset, then placing it back in the cradle.

5.2 Manually Setting the Time

The first two extensions on the system can be used to set the system time using phone based administration 25. The exact method will depend on the type of phone. For full details refer to the Phone Based Administration manual.

1408/1416/9404/9408/9504/9508 Phone

1. At either of the first two extensions on the system, press **Admin**.
2. Use the up or down arrow buttons to scroll the display to **System Administration**. When highlighted, press **Select**.
3. Use the up or down arrow buttons to scroll the display to **System Parameters**. When highlighted, press **Select**.
4. Use the up or down arrow buttons to scroll the display to **System Time**. When highlighted, press **Select**.
5. Enter the new system time in the format *HHMM* using the dial pad. The 24-hour clock is used, for example 4:22pm is entered as 1622.
6. Exit programming by pressing **PHONE**.

M7324/M7324N/M7310/M7310N/T7316/T7316E Phone

1. At either of the first two extensions on the system, press **Feature **config** (that is **Feature **266344**).
2. If the system has a system password set, it is requested. Enter the password.
3. The phone displays **System Admin**.
4. Dial **#101**. The phone will display **System Date**.
5. Enter the new system time in the format *HHMM* using the dial pad. The 24-hour clock is used, for example 4:22pm is entered as 1622.
6. To exit phone based administration at any time, press the **Release** button.

ETR 18D/ETR 34D Phone

1. At either or the first two extensions on the system, press **Feature 0 0** followed by two presses of the first intercom or call appearance button.
2. **System Administration** is shown on the display.
3. Dial **#101**. The phone will display **System Date**.
4. Enter the new system time in the format *HHMM* using the dial pad. The 24-hour clock is used, for example 4:22pm is entered as 1622.
5. Exit programming by pressing **Feature 00**. You can also exit programming mode by lifting the handset, then placing it back in the cradle.

5.3 Using Daylight Saving Time

This feature automatically updates the system clock for annual daylight savings time changes. This feature is only supported in North American locales.

1. Click on **System** in the menu bar and then click on **Switch**.
2. Click the **Advanced** button.
3. In the **System Details** panel, set the **Automatic DST** option as required. When enabled, the telephone system will automatically apply daylight saving time (DST) adjustments to its internal clock. This feature should only be used for systems in a North American locale.
4. Click **Save**.

5.4 Using Network Time Synchronization

Network time synchronization lets you synchronize the telephone system time and date with the network time information that your service provider includes on calls that include caller ID. If network time synchronization is not selected the system time has to be set manually 147.

- If network time synchronization is enabled and your service provider sends Caller ID from another time zone, the system clock will not match your local time.
- Note that this feature uses the first analog trunk on the card installed in slot 1 of the system control unit.

1. Click on **System** in the menu bar and then click on **Switch**.
2. Click the **Advanced** button.
3. In the **System Details** panel, set the **Enable Network Time Synchronization** option as required. When enabled, the system attempt to obtain the time .
4. Click **Save**.

Chapter 6.
Incoming Call Routing

6. Incoming Call Routing

The options for routing incoming calls depend on whether the system is set to ***PBX*** or ***Key*** mode.

Key Mode

For an incoming external calls on a line, the following options control where the call is presented:

- **Line Appearance Buttons**
  The call will alert on any line appearance buttons that matches the line. Each line has a line number which can be assigned to line appearance buttons on users' phones. Users can answer the call by pressing the alerting line appearance button on their phone.
  - **Number of Lines**
    By default, all analog lines in the system are assigned to line appearance buttons when the system is installed. Lines are assigned for all users starting from button 03 upwards in order of line numbering.
  - **Line Assignment**
    Through individual user button programming, any programmable button can be configured as a line appearance for a particular line.
- **Coverage Destination**
  The **Coverage Destination** setting of each line can be used to select whether an incoming call on that line is also presented to one of the following options in addition to alerting on any matching line appearances. For PRI and BRI trunks, it is not possible to know on which of the trunk's channels incoming calls will arrive. Therefore in most cases, the coverage destination and other settings of each line on the trunk should be set to the same values.
  - **Coverage Extension**
    The call alerts on an intercom button of a selected line coverage extension. The user's call coverage, VMS coverage and call forwarding settings are applied to the call. Any extension can be used as the destination including a phantom extension.
  - **Hunt Group**
    The call is presented, in sequence, to each of the available members of a selected hunt group until answered. Any of the 6 rotary hunt groups can be used as the destination.
- **Auto Attendant Coverage**
  Each trunk or trunk channel can be configured to send unanswered calls to an auto attendant after a set delay (which can be set to 0 for immediate answer). This can be set to operate when the system is in day and or night service. This is done using the **VMS Schedule**, **VMS Delay - Day**, **VMS Delay - Night** and **VMS Auto Attendant** settings of each line.

The following methods can be used to override the normal call routing detailed above:

- **DID Call Mapping**
  For BRI, ETSI PRI and PRI trunks, if the incoming call matches a configured DID and or ICLID number, the **Coverage Destination** setting for the DID/ICLID match is used rather than the line's **Coverage Destination**. DID can also be used on some types of T1 trunk.
- **SIP Call by Call Table**
  For SIP trunks, if the incoming call matches a configured URI, it is presented to the extension or group specified in the SIP line's **Call by Call Table**.
- **Night Service**
  Switching on night service overrides the routing of calls to Coverage Destinations. Instead the calls change to alerting the users who are members of the Night Service group. The settings for auto attendant coverage (VMS Schedule) can also be varied depending on whether the system is in night service or not.

PBX Mode

In PBX mode, a new group, the **Operator Group**, is used as a the default destination for call. This group contains the first extension on the system.

- For analog trunks trunks, the trunk's **Coverage Destination** is defaulted to the ***Operator Group*** but can be changed if required.
- For PRI and BRI trunks all incoming call routing is done by DID Call Mapping. Each DID table has a non-removable default route which is used for any calls that do not match any other specific DID entry. The destination for this default entry is the Operator Group.
- SIP trunks are defaulted to call by call operation, again with a default call by call destination of the ***Operator Group***.

The following new destinations for incoming calls are available:

- **Operator Group**
  This group is the default destination for all incoming calls. The group contains the first extension on the system but can be edited to contain other extensions.
- **Calling Groups**
  In ***Key*** mode these 4 groups are only used internally. In ***PBX*** mode these groups are also available as a destination for trunk calls in the **Coverage Destination** selections, DID Call Mapping and SIP Call by Call tables. A calling group can also be selected as the destination for an auto-attendant transfer.

Night Service Mode

In both modes, when the system is put into night service, all incoming calls except those to specific DID call mapping or SIP call by call destinations, are rerouted to alert the users who are members of the night service group.

6.1 Programming Line Appearance Buttons

Each trunk channel in the system is assigned an appearance ID, starting from 01 upwards. You cannot edit or change the IDs but they are displayed in the trunk settings.

Default Line Buttons

For systems running in Key mode, all extension users are assigned a number of line appearance buttons by default. Usually one for every analog trunk. This is controlled by the system's **Number of Lines** 138 setting.

Programming a Line Button

1. Click on **User** in the menu bar.
2. Highlight the required user by clicking on them.
3. Click on the edit icon in the **Button Programming** panel on the right.
4. The current button programming for the user is displayed.
5. In the **Program Buttons** panel click on the button that you want to program. The current settings of that button are displayed in the **Button Specification** panel.
6. Click on the **Assign Feature** button.
7. Select **Line Assignment**.
8. Select the line appearance ID that you want the button to match.
9. Select the ringing option.
   - ***Immediate***
     Provide audible alerting as normal.
   - ***Delayed Ring***
     Only provide audible alerting after approximately 15 seconds.
   - ***No Ring***
     Do not provide any audible alerting.
10. Click **OK**.

6.2 Setting a Trunk's Coverage Destinations

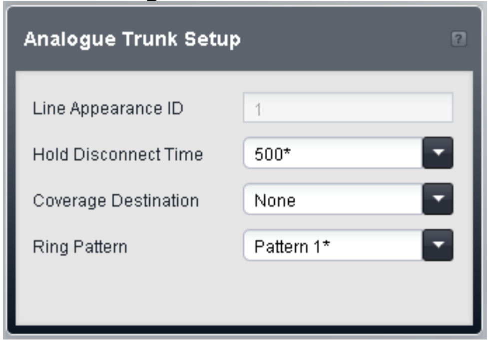
In addition to alerting on a kine appearance button associated with the trunk or trunk channels line number, a coverage destination can be assigned to the trunk or trunk channel.

- **None**
  If set to ***None***, incoming calls will only alert on user extensions with line appearance buttons that match the line's **Appearance ID**.
- **Extension**
  Route incoming calls to a particular extension.
- **Phantom Extension**
  A phantom extension can be selected as the destination for calls.
- **Hunt Group**
  Incoming calls can be routed to one of the 6 rotary hunt groups.
- **Calling Group**
  For systems with their **Mode** set to ***PBX***, incoming calls can be routed to one of these 4 ring all groups.
- **Operator Group**
  For systems with their **Mode** set to ***PBX***, incoming calls are routed to the **Operator Group**.
- **Voicemail**
  Route incoming calls to the systems voicemail to collect messages. This requires the caller to know the mailbox number and passcode.

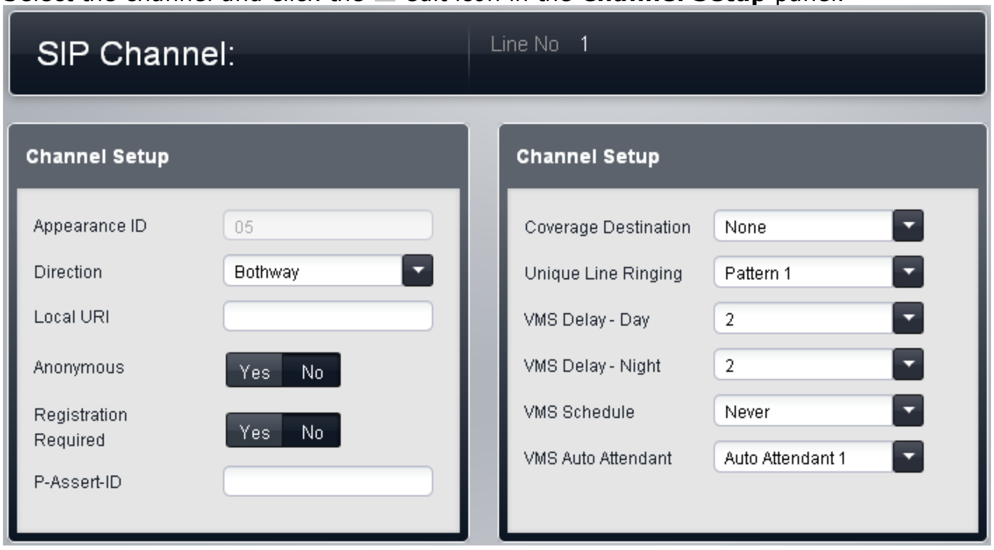
Setting a Trunk's Coverage Destinations

1. Click on **System** in the menu bar and then click on **Trunks**.

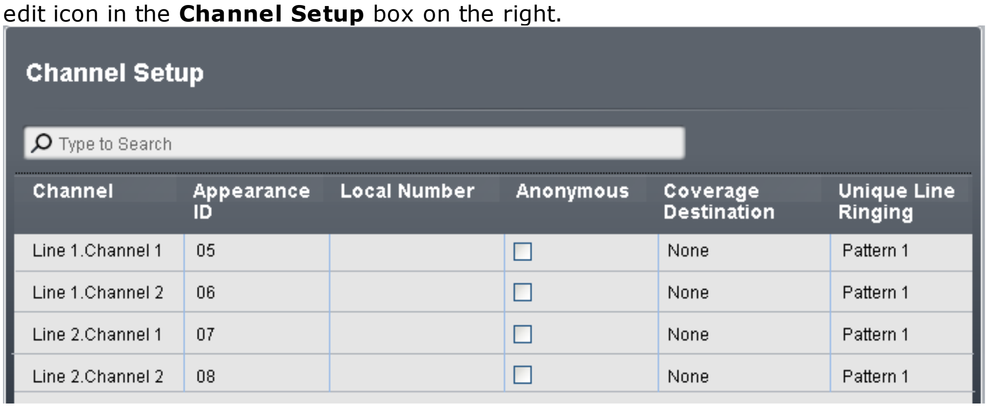
   - **Analog Trunks**
     Click on **System** in the menu bar and then click on **Trunks**. Select the trunk and click on **View Details**. The **Coverage Destination** is shown in the **Analog Trunk Setup** panel.



   - **SIP Trunks**
     Click on **System** in the menu bar and then click on **SIP Trunks**. Select the trunk and click on Details. Select the channel and click the edit icon in the **Channel Setup** panel.



   - **Other Trunks Types**
     Click on **System** in the menu bar and then click on **Trunks**. Select the trunk and then click on the edit icon in the **Channel Setup** box on the right.



2. Click on **Coverage Destination** and select the required destination. The options available will depend on the trunk type and the operating mode of the system.

   - **None**
     If set to ***None***, incoming calls will only alert on user extensions with line appearance buttons that match the line's **Appearance ID**.

   - **Extension**
     Route incoming calls to a particular extension.

   - **Phantom Extension**
     A phantom extension can be selected as the destination for calls.

- **Hunt Group**
  Incoming calls can be routed to one of the 6 rotary hunt groups.
- **Calling Group**
  For systems with their **Mode** set to ***PBX***, incoming calls can be routed to one of these 4 ring all groups.
- **Operator Group**
  For systems with their **Mode** set to ***PBX***, incoming calls are routed to the **Operator Group**.
- **Voicemail**
  Route incoming calls to the systems voicemail to collect messages. This requires the caller to know the mailbox number and passcode.

3. Click **Save**.

6.3 Enabling Auto Attendant Coverage

For each trunk, an auto attendant can be selected to automatically answer any call not answered at the trunk or trunk channels **Coverage Destination** 153. Use of auto attendant coverage can selected for day and or night service. Also the delay before the auto attendant answer the call can be adjusted for day and night service.

This option is not used for calls routed to their destination by DID Call Mapping or SIP Call-by-Call settings.

Configuring Auto Attendant Call Coverage

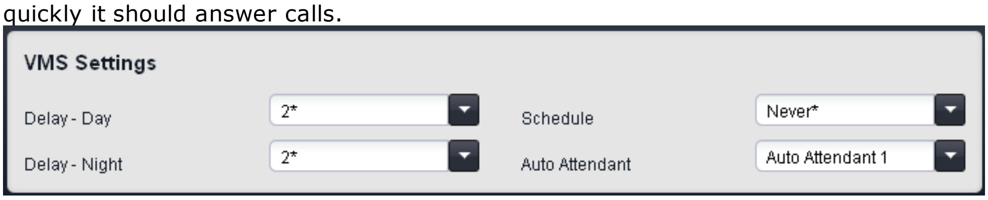
1. The method for accessing a trunk's VMS Settings depends on the trunk type.

   - **Analog Trunks**
     Select **System** in the menu bar and click **Trunks**. Select the trunk, click **View Details** and then **Advanced**. The trunk's VMS Settings are part of the menu now displayed.

   - **PRI Trunks**
     Select **System** in the menu bar and click **Trunks**. Select the trunk and then click the edit icon in the **Channel Setup** panel. Select the channels and click **View Details**.

   - **SIP Trunks**
     Select System in the menu bar and click SIP Trunks. Select the trunk and then click the edit icon in the **Channel Setup** panel. Select the channels and click **Details**.

2. Adjust the **VMS Settings** to set which auto attendant should be used, when it should be used and how quickly it should answer calls.



   - **Delay - Day:** *Default = 2. Range = 0 to 6 (number of rings).*
     Set the number of rings before an unanswered call should be redirected the selected auto attendant when the system is not running in night service mode and the **VMS Schedule** is set to ***Always*** or ***Days Only***.

   - **Delay - Night:** *Default = 2. Range = 0 to 6 (number of rings).*
     Sets the number of rings before an unanswered call should be redirected to the selected auto attendant when the system is running in night service mode and the **VMS Schedule** is set to ***Always*** or ***Night Only***.

   - **Schedule:** *Default = Never.*
     This option determines when the **VMS Delay** settings above should be used and unanswered calls redirected to the selected auto attendant. The options are:

     - ***Always***
       Redirect calls when the system is in both day and night service modes.

     - ***Day Only***
       Redirect calls only when the system is not in night service.

     - ***Night Only***
       Redirect calls only when the system is in night service.

     - ***Never***
       Do not redirect calls.

   - **Auto Attendant:** *Default = Auto Attendant 1.*
     This field allows selection of which auto attendant is used.

3. Click **Save**.

6.4 Using DID Call Mapping

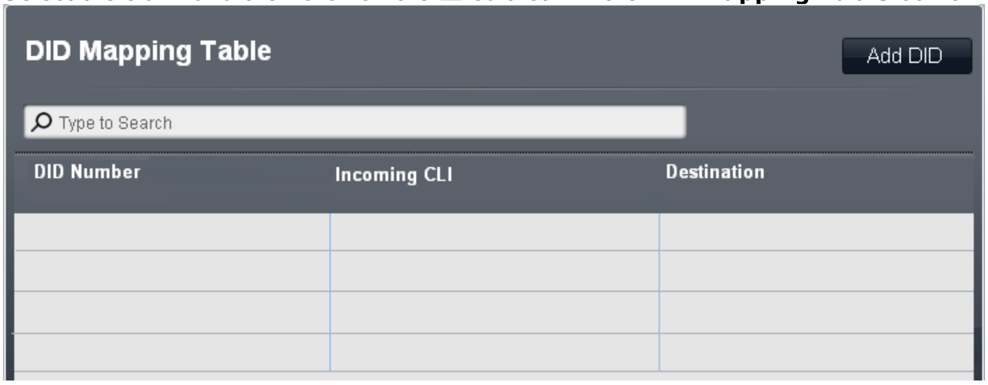
Incoming calls that include DID digits are routed to a destination on the system by checking for a match in the trunks DID mapping table. This call routing overrides the **Coverage Destination** settings of the trunk channel on which the call was received. In addition, calls routed by DID mapping are not affected by the phone system being put into night service 210.

In addition to alerting on a kine appearance button associated with the trunk or trunk channels line number, a coverage destination can be assigned to the trunk or trunk channel.

- **Extension**
  Route incoming calls to a particular extension.
- **Phantom Extension**
  A phantom extension can be selected as the destination for calls.
- **Hunt Group**
  Incoming calls can be routed to one of the 6 rotary hunt groups.
- **Calling Group**
  For systems with their **Mode** set to ***PBX***, incoming calls can be routed to one of these 4 ring all groups.
- **Operator Group**
  For systems with their **Mode** set to ***PBX***, incoming calls are routed to the **Operator Group**.
- **Voicemail**
  Route incoming calls to the systems voicemail to collect messages. This requires the caller to know the mailbox number and passcode.
- **76: Modem**
  The option ***76: Modem*** can be selected to route the call to the systems built in V32 modem function. This is only intended for basic access by system maintainers.
- **Auto Attendant**
  Any configured voicemail auto attendants can be selected as the call destination.

Setting a Trunk's Coverage Destinations

1. Click on **System** in the menu bar and then click on **Trunks**.
2. Select the trunk and then click on the edit icon in the **DID Mapping Table** box on the right.



3. Click on **New DID Number**. The options available will depend on the trunk type and the operating mode of the system.

   - **DID Number**
     If the incoming DID of a call on the trunk matches the DID set here, it will be routed to this destination. The system supports up to 4 digits DID (additional digits after the first 4 are ignored). Leave blank if only CLI matching is required.
   - **Incoming CLI**
     If the incoming caller number on the trunk matches the Incoming CLI set here, it will be routed to this destination. Leave blank if only DID matching is required.
   - **Destination**
     When this field is selected, the drop down list allows selection of the destination for matching calls. The options differ depending on whether the system's **Mode**[65] is set to ***Key*** or ***PBX***.
   - **Extension**
     Route incoming calls to a particular extension.
   - **Phantom Extension**
     A phantom extension can be selected as the destination for calls.
   - **Hunt Group**
     Incoming calls can be routed to one of the 6 rotary hunt groups.
   - **Calling Group**
     For systems with their **Mode** set to ***PBX***, incoming calls can be routed to one of these 4 ring all groups.
   - **Operator Group**
     For systems with their **Mode** set to ***PBX***, incoming calls are routed to the **Operator Group**.
   - **Voicemail**
     Route incoming calls to the systems voicemail to collect messages. This requires the caller to know the mailbox number and passcode.
   - **76: Modem**
     The option ***76: Modem*** can be selected to route the call to the systems built in V32 modem function. This is only intended for basic access by system maintainers.
   - **Auto Attendant**
     Any configured voicemail auto attendants can be selected as the call destination.

4. Click **Save**.

Chapter 7.
PBX Mode Outgoing Call Routing

7. PBX Mode Outgoing Call Routing

Each phone is configured with 3 call appearance buttons (2 only on ETR phones). These can be used to make both internal and external calls. The dialing of an external call can be indicated by the dialing starting with a specific prefix (9 or 0) if required, otherwise any number not matching an internal extension or function is automatically assumed to be external.

The line used for an outgoing external call is determined by a set of **Alternate Route Selection** (ARS) settings:

- ARS Selectors are created. These are groups of lines or selectors for specific functions using any available ISDN line.
- Different classes of call (sets of external number prefixes) are then mapped to those ARS Selectors. The system supports classes for **Emergency**, **National**, **International**, **Cell** and **Toll Free** sets of prefixes. An additional **Local** class is used for any calls that do not match one of the other classes.

When a user dials an external number, it is matched to a selector and uses the function and one of the lines specified by that selector. For SIP trunks set to call by call mode, each call by call entry also has an ARS selector settings which allows it to also be used for outgoing calls.

Line appearances can still be used to make and answer calls on a particular line but are not added by default. They can also be used to select a particular ARS selector for an outgoing call.

Default Alternate Route Selection

The following ARS selectors exist by default. They cannot be deleted though the lines associated with ARS 65 can be edited. By default all the classes of external call are set to use the first available trunk or trunk channel in ARS 65 and 66, that is the first available analog trunk or ISDN trunk channel.

- ***65: Group of Lines***
  This ARS selector cannot be deleted. By default it contains all analog lines in the system when the system was installed or defaulted. However, it can be edited to change the lines included including adding non-analog lines. This selector and 66 below are used as the default for all classes of external call.
- ***66: ISDN Standard Call - Local Number = Default***
  This ARS selector cannot be deleted. Calls routed to this selector will use an available ISDN trunk channel with the calling party information set to match the user's **User Calling Line Identity** if set or otherwise blank (to be set by the trunk provider). This selector and 65 above are used as the default for all classes of external call.
- ***67: ISDN Number Withheld - Local Number = Withheld***
  This ARS selector cannot be deleted. Calls routed to this selector will use an available ISDN trunk channel with the calling party information set to withheld.

External Dialing Restrictions

A number of methods to restrict which users can make external calls and which numbers they can call are still applied before alternate route selection is used to seize a particular trunk or trunk channel and dial the call.

- **Allowed Number Lists** 188 / **Disallowed Number Lists** 186
  These lists are used in define numbers that can or cannot be dialed. Users are then associated with the different lists.

Allowed Numbers	Each allowed list contains external telephone numbers that members of the list are allowed to dial. The allowed lists to which a user is assigned override any disallowed lists to which they are also assigned. The numbers in the user's assigned allowed lists also override the **Calls Barred** and **Outgoing Call Restrictions** 43 settings that may be applied to a user. There are eight lists, each containing up to 10 numbers. Each number can use the telephone dialing digits 0 to 9, *, # and can be up to 28 digits long. You can also use the ? character as a single digit wildcard.
Disallowed Numbers	Each disallowed list contains external telephone numbers that users who are assigned to the list cannot dial. There are eight lists, each containing up to 10 numbers. Each number can use the telephone dialing digits 0 to 9, *, # and can be up to 28 digits long. You can also use the ? character as a single digit wildcard. Numbers in the disallowed lists of which a user is a member are overridden if they also appear in the allowed numbers lists, emergency number list of which the user is a member and also by marked system speed dials.
Emergency Numbers	You can enter 10 emergency phone numbers into this list. This list is applied to all users and overrides any dialing restrictions that may also be applied to the users. Use of numbers in the list can be viewed using a 911-View/E-View 220 button.

- **Account Codes** 190
  Each user can be configured to need to enter a valid account code whenever they make an external call.

- **Outgoing Call Restrictions** 185
  For each user, the type of external calls that the user is able to make can be configured.
- **Marked Speed Dials** 191
  When a user uses a stored system speed dial number, the actual number dialed is subject to all the call barring methods as if the user had dialed the number directly. However system speed dials set as 'marked speed dials' override any call restrictions.
- **Night Service** 210
  When the system is set to night service, any users in the **Night Service Group** need to enter the system password when making an external call.

7.1 Setting the Outside Line Prefix

This option is only used with systems set to PBX mode. It sets the digit which, when dialed at the start of a number, indicates that the call is intended to be external. The options are to use ***0***, ***9*** or no prefix.

Note that the setting also changes the digits used for calls to the first extension on the system. Normally, in addition to the extension's extension number, the number 0 can be used to call that extension. If the number 0 is set as the outside line prefix, the number 9 is used for the first extension.

Setting the System's Outside Line Prefix

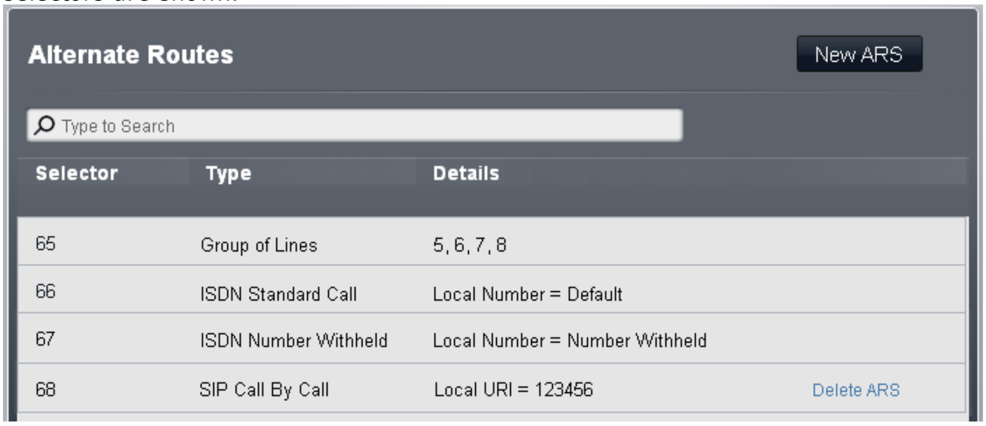
1. Click on **System** in the menu bar.
2. In the **System Parameters**, set the **Outside Line** setting to the required option.
   - ***9 (Operator is 0)***
     The prefix 9 is used for external calls. The digit 0 is used for calls to the operator extension (the first extension in the system). This is the default setting for systems with the **Country** setting ***United States***.
   - ***None***
     No prefix is used for external calls. Any dialing that does not match an internal dial plan number is assumed to be an external call. This is the default setting for systems with the **Country** setting other then ***Germany*** or ***United States***. The digit 0 is used for calls to the operator extension (the first extension in the system).
   - ***0 (Operator is 9)***
     The prefix 0 is used for external calls. The digit 9 is used for calls to the operator extension (the first extension in the system). This is the default setting for systems with the **Country** setting ***Germany***.
3. Click **Save**.

7.2 Editing the ARS Selectors

ARS selectors are grouping of trunks, trunks channels or trunk settings. Once a set of ARS entries has been created, they can be associate with the different class of call 164 dialing prefixes.

Editing the ARS Selectors

1. Click **Outgoing Call Management** in the menu bar. Select **Alternate Route Selection**. The existing ARS selectors are shown.



- **To Delete an Entry**
  Click the Delete ARS link to the right of the entry. Note that the entries 65, 66 and 67 are default entries and cannot be deleted.

- **To Edit an Entry**
  Double click on the entry. The fields for the selector's **Type** and **Details** settings can now be edited. When the selector is set as required click **Save**. For details of the settings see below.

- **To Add a New Entry**
  Click on the **New ARS** button. When the selector is set as required click **Save**. For details of the settings see below.

ARS Selector Settings

- **Selector**
  This must be a number in the range 65 to 99. Selectors 65, 66 and 67 are default entries that cannot be removed. The selector number is used in the Dial Numbers table to associate the ARS with external dialing prefixes that can use it. It can also be used as a line appearance button to allow users to specifically select the ARS group for an outgoing call.

  - ***65: Group of Lines***
    This ARS selector cannot be deleted. By default it contains all analog lines in the system when the system was installed or defaulted. However, it can be edited to change the lines included including adding non-analog lines. This selector and 66 below are used as the default for all classes of external call.

  - ***66: ISDN Standard Call - Local Number = Default***
    This ARS selector cannot be deleted. Calls routed to this selector will use an available ISDN trunk channel with the calling party information set to match the user's **User Calling Line Identity** if set or otherwise blank (to be set by the trunk provider). This selector and 65 above are used as the default for all classes of external call.

  - ***67: ISDN Number Withheld - Local Number = Withheld***
    This ARS selector cannot be deleted. Calls routed to this selector will use an available ISDN trunk channel with the calling party information set to withheld.

- **Type**
  The ARS Selector group can be used for the following functions:

  - ***Group of Lines***
    This type of selector is used to create a group of lines. The lines are selected using the **Select Lines** table below. For a call routed to this selector, an available line from that group is used.

  - ***ISDN Local Number***
    This type of selector is used to set an outgoing local number on an ISDN call. For a call routed to this ARS selector, an available ISDN channel is used with the calling party element of the Q.931 setup set to match the local number specified.

    - Changing the calling party number may not be supported by the line provider or may be an additional chargeable service. It will also be subject to restrictions on what numbers can be used. It is normally a requirement that the calling party number used must be a valid number for return calls to the same trunk. Use of an invalid number may cause the call to be dropped or the number to be replaced by a default value.

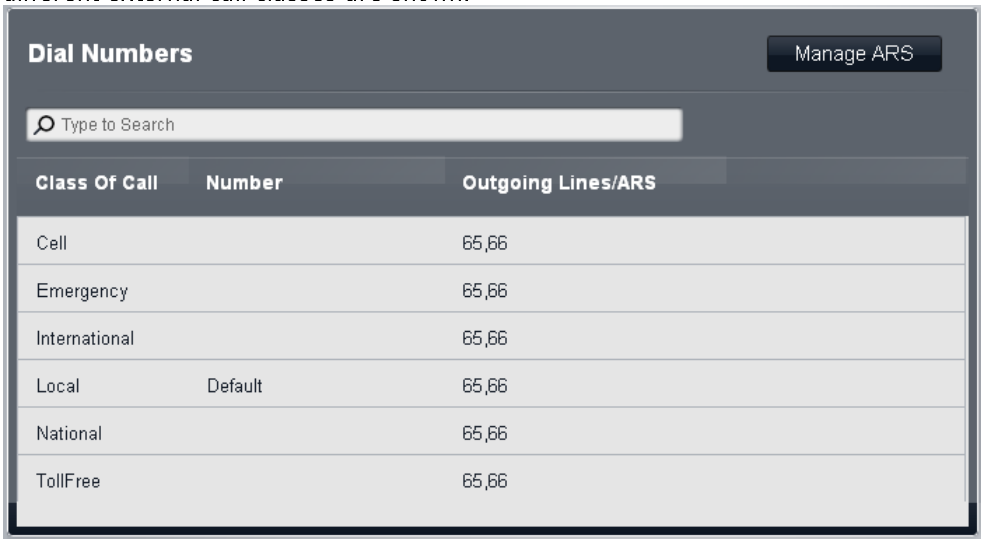
    - The default ARS Selector entry 66 is set to ***Local number=default***. It uses the user's **User CLI** if set.
  - ***ISDN Standard Call***
    This type of selector is used to select an available ISDN channel for the call.
  - ***ISDN Number Withheld***
    This type of selector is used to withhold any outgoing local number information on an ISDN call. For a call routed to this ARS selector, an available ISDN channel is used with the calling party element of the Q.931 setup set to withheld.
  - ***SIP Call-by-Call***
    These entries appear when entries are created in a SIP trunk's **Call-by-Call** 98 table. They cannot be edited through the ARS Selectors form. By having an associated ARS Selector number, the entry can be selected as the destination for specific out going calls.
- **Details**
  This field show either the lines currently selected for use with the ARS selector or the local number setting for the calling party number.

7.3 Editing the External Call Classes

The available external call classes are **National**, **International**, **Emergency**, **Cell** and **Toll Free**. For each, you can define the numbers the dialing prefixes that match that call type and the ARS selector groups to which matching calls should be routed. The **Local** class is used as the default for any calls that do not match another class.

Editing the Class of Call Entries

1. Click **Outgoing Call Management** in the menu bar and select **Dial Numbers**. The settings for the different external call classes are shown.



2. To edit a particular class, double click on the entry. The setting for the **Number** and the **Outgoing Lines/ARS** fields become editable.

   - **Number**
     For each class of call, this field is used to define the dialing prefix (up to 5 digits) expected for the call to match the class. Multiple prefix numbers can be entered, each separated by a comma.
     - To edit the numbers, double click on the entry. The fields for **Number** and for **Outgoing Lines/ARS** become adjustable. Edit the values as required and click **Save**.
       - Do not include the **Outside Line** prefix digit configured in the system settings.
       - If a match occurs in more than one class, the most exact match is used, ie. the one with the most digits. If multiple matches still exist, the match that occurs first in the table is used.
       - Numbers cannot be set for the **Local** class. This class is used for any calls that do not match any other class. However the ARS selectors used by this class can be changed.
   - **Outgoing Lines/ARS**
     This field indicates the ARS selectors currently associated with the Class of Call. These contain the trunks that are used by the Class of Call and are set using the Select Outgoing ARS table.
     - To edit the numbers, double click on the entry. The fields for **Number** and for **Outgoing Lines/ARS** become adjustable. Edit the values as required and click **Save**.

3. Click **Save**.

Chapter 8. Groups

8. Groups

A group is a collection of users used for a special purpose. For example, a group can be used as the destination for incoming calls on an external trunk.

The system has a fixed set of groups. You cannot add or remove groups. However you can edit the the user membership of all the groups. A user can be a member of several or even all groups.

While call to a group alert at user extensions, they will not go to user voicemail. They will follow user forwarding if set to forward to another extension. They will not follow external forwarding.

The following different types of groups are used:

- ***Calling Group*** 167
  This type of group can be set as the **Coverage Destination** of a trunk or trunk channel. It can be also the destination of calls routed using DID call mapping or call-by-call settings. Calling Group 1 is also used by the **Simultaneous Page** function (****70***).

- ***Hunt Group*** 167
  This type of group can be set as the **Coverage Destination** of a trunk or trunk channel. It can also be the destination of calls routed using DID call mapping or call-by-call settings.

- ***Night Service Group*** 167
  When the system is put into night service, this group overrides the **Coverage Destination** of all trunks.

- ***Operator Group*** 167
  This option is only available for systems with their **Mode** set to ***PBX***. By default the group contains the first extension on the system. For PRI and BRI trunks, it is fixed incoming destination for calls unless DID Mapping is applied to the call. It can also be selected as the destination for incoming SIP calls.

- ***Pickup Group*** 167
  Users can be pickup calls currently alerting any member of a pickup group. They do not need to be a member of the group.

8.1 Groups

8.1.1 Calling Groups

You can call, transfer and forward calls to the group by dialing the calling group number 71 to 74 respectively. The simultaneous page function, accessed by dialing ***70***, pages the extension connected to a loudspeaker paging device and the users who are members of Calling Group 1.

These groups are ***Ring All*** groups, that is a call to the group will alert all the available group members (those not already on a call) until the call is answered.

For ***Key*** mode systems, these groups are only used internally. For ***PBX*** mode systems, these groups can be selected as destinations for incoming calls within trunk settings.

8.1.2 Hunt Groups

All systems have 6 hunt groups. You can call, transfer and forward calls to the group by dialing the calling group number 771 to 774 respectively.

These groups are ***Rotary*** groups, that is a call to the group will alert the first the available group members after the last one to answer a group call, for approximately 15 seconds, and then the next member and the next in extension number order until the call is answered.

This type of group can be set as the **Coverage Destination** for incoming calls on all trunk types in ***Key*** mode and PRI, T1 and SIP trunk channels on ***PBX*** mode. It can also be set as the destination for calls routed using DID call mapping or call-by-call settings.

8.1.3 Night Service Group

The members of the night service group are used as the destination for most external calls when the system is in night service 210.

- When the system is put into night service, the **Coverage Destinations** of all trunks and trunk channels are overridden. Instead all calls to which the **Coverage Destination** would have been applied are present to the users in the night service group. The only exception are calls routed to specific destinations by DID call mapping or Call-by-Call settings.
- The group is a ***Ring All*** groups, that is a call to the group will alert all the available group members (those not already on a call) until the call is answered.
- Any ring delays on trunks are ignored. Instead calls redirected to the **Night Service Group** members alert immediately.
- If a system password is set, users in the **Night Service Group** need to enter that number when dialing external calls. The only exception is if dialing numbers in the **Emergency Numbers** list and marked system speed dials. No additional outgoing call restrictions are applied to any users not in the **Night Service Group**.

8.1.4 Operator Group

This group is only available in ***PBX*** 135 mode. The group has one default member, the first extension on the system, however other extensions can be added if required.

This group is used as the default destination for incoming calls on all types of trunk. In addition you can call, transfer and forward calls to the group by dialing the operator group number 75.

The group is a ***Ring All*** group, that is a call to the group will alert all the available group members (those not already on a call) until the call is answered.

8.1.5 Pickup Groups

The system has 4 pickup groups. A call alerting any user who is a member of a pickup group can be answered by another user by dialing the pickup group number 661 to 664 respectively. The user doing the call pickup does not need to be a member of the group themselves.

8.2 Displaying the Groups

1. Click **Incoming Call Management** in the menu bar and then click **Groups**.
2. The list of groups is displayed.

Name	Type	Ring Mode	Number	Members
Calling Group 1	Calling Group	Ring All	71	0
Calling Group 2	Calling Group	Ring All	72	0
Calling Group 3	Calling Group	Ring All	73	0
Calling Group 4	Calling Group	Ring All	74	0
Hunt Group 1	Hunt Group	Sequential	771	0
Hunt Group 2	Hunt Group	Sequential	772	0
Hunt Group 3	Hunt Group	Sequential	773	0
Hunt Group 4	Hunt Group	Sequential	774	0
Hunt Group 5	Hunt Group	Sequential	775	0
Hunt Group 6	Hunt Group	Sequential	776	1
Pickup Group 1	Pickup Group	Sequential	P01	1

- **Name**
  The names of the groups are fixed.
- **Type**
  This indicates the main purpose of the group.
  - ***Calling Group*** 167
    This type of group can be set as the **Coverage Destination** of a trunk or trunk channel. It can be also the destination of calls routed using DID call mapping or call-by-call settings. Calling Group 1 is also used by the **Simultaneous Page** function (****70***).
  - ***Hunt Group*** 167
    This type of group can be set as the **Coverage Destination** of a trunk or trunk channel. It can also be the destination of calls routed using DID call mapping or call-by-call settings.
  - ***Night Service Group*** 167
    When the system is put into night service, this group overrides the **Coverage Destination** of all trunks.
  - ***Operator Group*** 167
    This option is only available for systems with their **Mode** set to ***PBX***. By default the group contains the first extension on the system. For PRI and BRI trunks, it is fixed incoming destination for calls unless DID Mapping is applied to the call. It can also be selected as the destination for incoming SIP calls.
  - ***Pickup Group*** 167
    Users can be pickup calls currently alerting any member of a pickup group. They do not need to be a member of the group.
- **Ring Mode**
  This settings indicates the order in which the idle group members are alerted when there is a call to the group. For full details see Group Call Distribution 169.
  - ***Ring all***
    This type of group alerts all the idle groups members at the same time until answered.
  - ***Rotary***
    This type of group alerts each idle group member in extension number order for 15 seconds each until answered. It starts with the available member after the last member to answer a group call.
- **Number**
  The use of a number associated with a group depends on the group type:
  - Calling, operator and hunt groups can be called internally using their extension number. The groups number can also be used as the target for call transfers. * can be used in front of the group extension number for a page call to the group.
  - Pickup groups numbers are used to answer a call currently alerting any member of the group. For example, to pickup a call alerting members of pickup group 1, dial 661.
- **Members**
  This indicates the number of users who are currently set as members of the group.

8.3 Group Call Distribution

A line can be configured to present its incoming calls to one of the 6 hunt groups (rotary) or 4 calling groups (ring all).

Rotary Ringing

This mode of ringing is used by the 6 Hunt Groups. The groups can be rung directly or used as the coverage destination for incoming calls on trunks and trunk channels.

A call to a hunt group first alerts the first available member (not on another call) of the group after the one who last answered a group call. If after approximately 15 seconds the call has not been answered, it changes to alerting the next available member of the hunt group in extension number order. When alerting the currently targeted member of the group:

- If the user does not have a line appearance for the line, it alerts on an intercom or call appearance button.
- If the user has a line appearance for the line, the call alerts on the line appearance button with the standard indication of a ringing call for me (slow green flash). The line's ringing options are overridden and the line always rings immediately.
- Any other users, including other group members, with the same line appearance will see the line button indicate a ringing call but not for me (slow red flash).
- A call targeted to a group does not go to voicemail. Instead it will alert the group until answered or abandoned.
- Any extension in the system can answer the call using the line appearance of the line or using one of the pickup features (active line pickup, call pickup, group call pickup).

If the currently targeted user has forwarding to another extension set, the call is forwarded. Forwarding to an external number is not used, the call will alert the original extension instead.

If coverage is active at the targeted hunt group extension, it is not followed and alerts the normal number of rings before hunting on to the next hunt group extension.

Ring All

This mode of ring is used by the 4 Calling Groups and the Operator Group. The groups can be rung directly or used as the coverage destination for incoming calls on trunks and trunk channels.

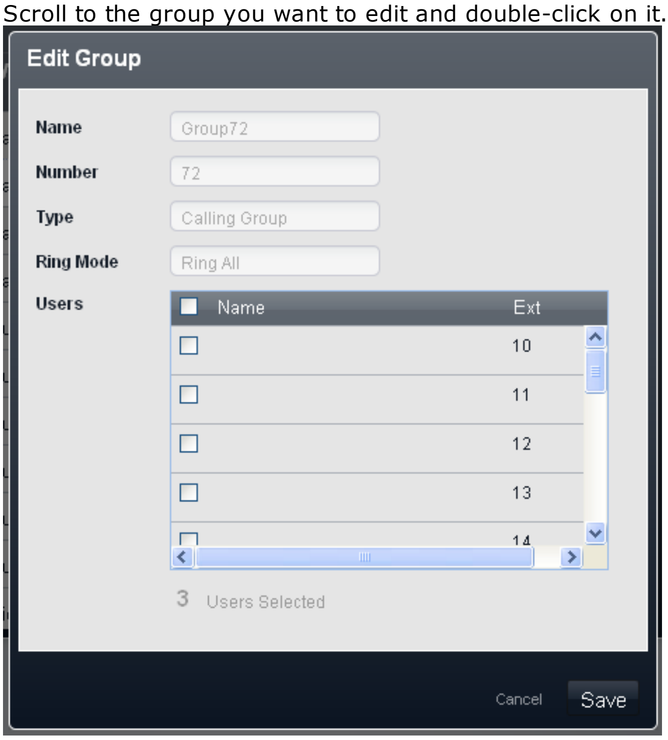
A call to the hunt group alerts all members who are idle (not on another call). The call alerts in the same way as for a rotary group but does not hunt if not answered after 15 seconds, instead it continues to alert the same members until answered.

8.4 Editing Group Membership

The following process can be used to edit the list of users who are members of a particular group.

Editing Group Membership

1. Click **Incoming Call Management** in the menu bar and then click **Groups**.
2. The list of groups is displayed. The Members column indicates the number of users who are currently a member of each group.
3. Scroll to the group you want to edit and double-click on it. The groups details are displayed.



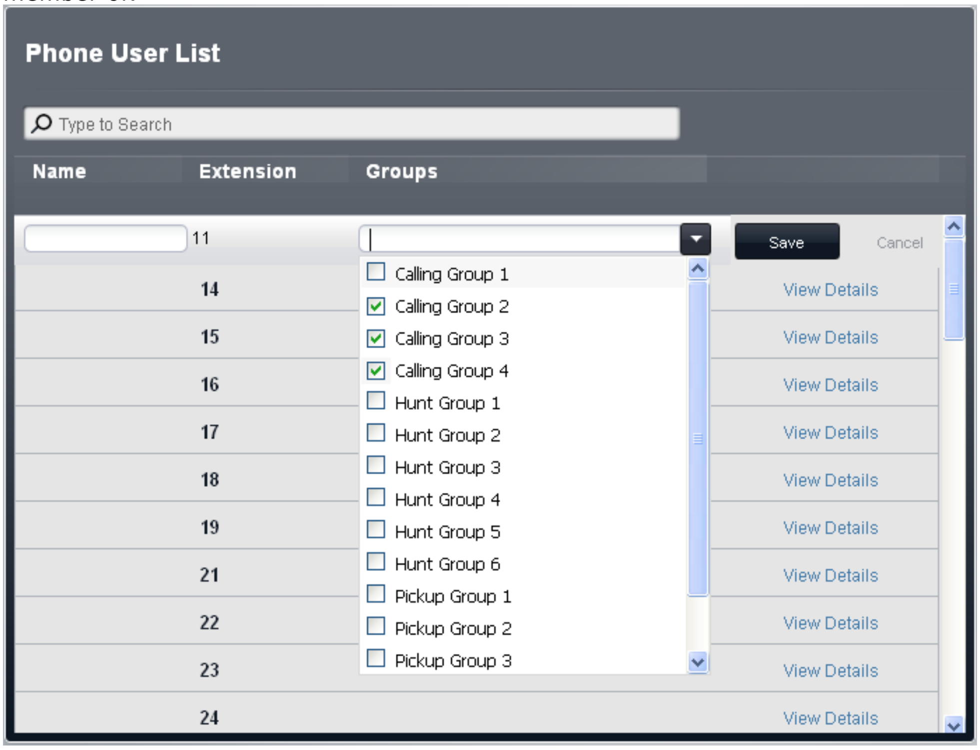
4. Use the list of users to select which users should be members of the group.
5. Click Save.

8.5 Changing a User's Group Memberships

You can edit the groups of which a user is a member through the user list.

Changing a User's Group Memberships

1. Click **User** in the menu bar.
2. The list of users is displayed. The groups column lists the user's current group memberships.
3. Scroll the list to the user who you want to edit and double click on the row.
4. Click on the field in the groups column. The drop-down list allows you to select which groups the user is a member of.



5. Set the user's group memberships and click **Save**.

Chapter 9. Auto Attendant Configuration

9. Auto Attendant Configuration

The system supports up to 9 auto attendants. These can be used to answer incoming calls on selected lines. Alternatively an auto attendant can be used to cover calls on a line that are unanswered by the extension or group that should answer.

When an auto attendant answers a call, it prompts the caller to make a selection by pressing the telephone key after having heard the list of options available. Options can include being transferred to another number, dialing the number of the extension they require, being transferred to another auto attendant, etc. Possible actions that can be assigned to each key are:

- ***Dial by Name***
  The caller is asked to dial the name of the user they require and then press #. The recorded mailbox names of matching users are then played back for the caller to make a selection. The name order used is set by the **Dial by Name Match Order** setting. Users without a recorded name prompt or not set to **List in Directory** are not included. Users can record their name by accessing their mailbox and dialing *05.

- ***Dial By Number***
  This option allows the caller to dial the extension number of the user they require. No destination is set for this option. The attendant's **Dial By Direct Number** setting determines how the digits dialed with this action are used.

- ***Transfer to Auto Attendant***
  This option transfers the caller to another indicated auto attendant. This will skip the greeting menu of that auto attendant, playing just the current menu options greeting instead.

- ***Transfer to Emergency Greeting***
  This option transfers the caller to a set of prompts for recording the emergency greeting and for selecting whether the emergency greeting is enabled or not.

  - If a system password 65 has been set, the caller is asked to enter that password before they can continue.

  - When the emergency greeting is enabled, it is played to other auto attendant callers before any other auto attendant greeting.

  - When the emergency greeting is enabled, a warning is displayed on the auto attendant's **Alarm Extension**.

- ***Transfer to Number***
  Transfer the call to the extension or group set with the action.

- ***Replay Menu Greeting***
  Repeat the prompt listing the current menu options.

When Do Calls Go to an Auto Attendant?

The system supports the configuration of up to 9 auto attendant services to answer and redirect calls. If an auto attendant has been configured, it can be used to answer calls on external lines as follows:

- **Unanswered Calls Coverage**
  Each trunk or trunk channel is configured with a coverage destination for its incoming calls. It can also be configured with a selected auto attendant that will be used to answer unanswered calls at the trunk or trunk channels coverage destination. When the auto attendant is used and the delay before it answers unanswered call can be adjusted.

- **Immediate Call Coverage**
  The delay used for unanswered call coverage can be set to immediate. When that is done, the auto attendant will answer before the trunk or trunk channels configured coverage destination.

- **Answering Specific External Calls**
  On trunks that support the setting of separate DID or call-by-call destinations, an auto attendant can be selected as the destination for matching calls. This allows auto attendants to be used to answer incoming calls to specific numbers separate from the trunk channels own coverage destination settings. This option is not supported on analog trunks.

- **Manual Answering**
  Each auto attendant has an internal number assigned to it. That number can be used by extension users to transfer calls to the auto attendant.

9.1 Licensing

The system's voicemail services, both user mailboxes and call auto attendants, operate without requiring any licenses. However the number of simultaneous calls and the message storage capacity is controlled by the number of voicemail licenses added to the system's configuration.

- **Number of Simultaneous Calls**
  Without any voicemail licenses, the number of simultaneous calls using the voicemail services is limited to 2. Each voicemail license added enables an additional 2 simultaneous calls, up to a maximum of 6.

- **Voicemail Storage Capacity**
  The user mailboxes are also limited to a maximum of 15 hours storage for messages, greeting and prompts. However, each additional voicemail license added also enables an additional 5 hours of storage up to a maximum of 25 hours.

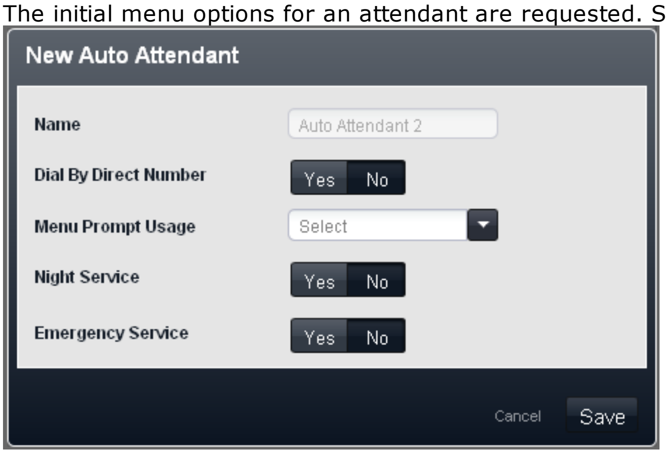
9.2 Adding an Auto Attendant

Auto attendant 1 is present in the system configuration by default and cannot be removed. However you can add up to 8 more auto attendants.

Before adding the auto attendant, you need to have worked out the following details:

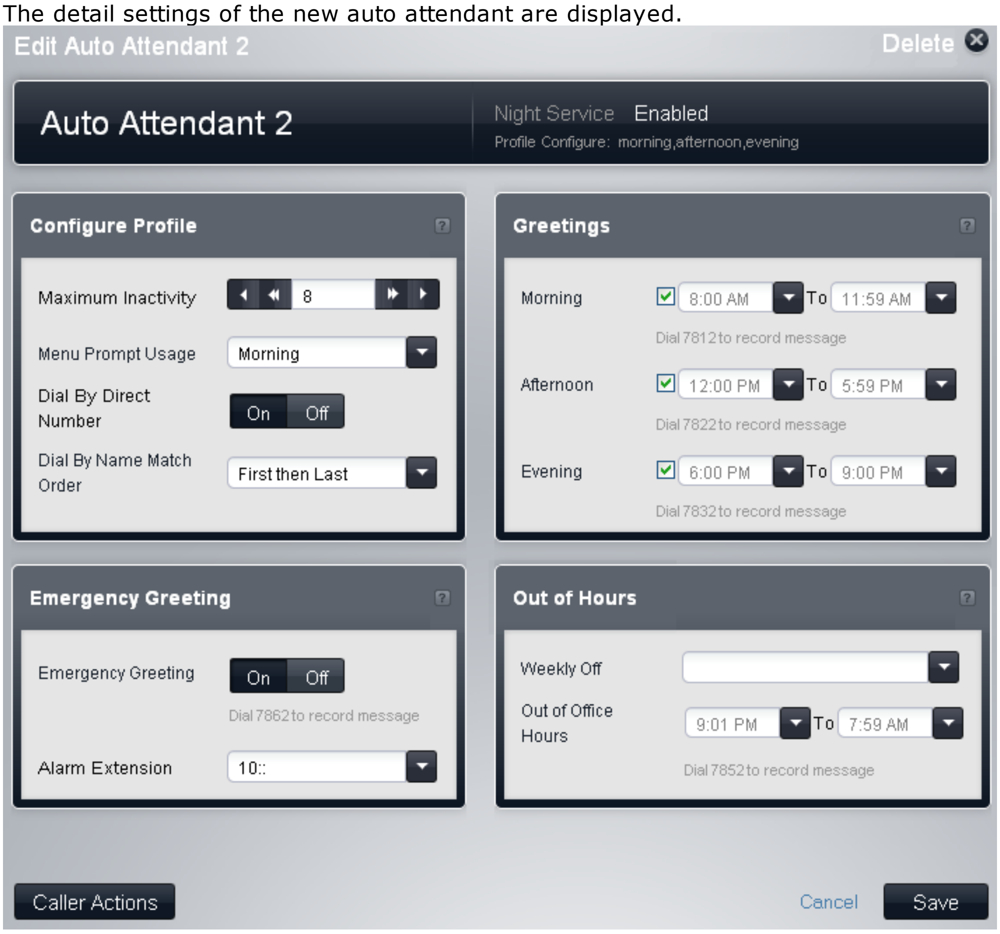
- The times of day during which the different possible greetings (morning, afternoon, evening and out of hours) should be used.
- The actions that the auto attendant should offer to callers during each of those time periods.

1. Click **Incoming Call Management** in the menu bar and then click **Auto Attendant**.
2. The list of existing auto attendants is displayed. Click **New Auto Attendant**.
3. The initial menu options for an attendant are requested. Set these as required and then click **Save**



   - **Dial By Direct Number:** *Default = On for the first default auto attendant. Off for other auto attendants.* This setting affects the operation of any key presses in the auto attendant menu set to use the ***Dial By Number*** action.
     - If selected, the key press for the action is included in any following digits dialed by the caller for extension matching. For example, if 2 is set in the actions to ***Dial by Number***, a caller can dial 20 for extension 20.
     - If not selected, the key press for the action is not included in any following digits dialed by the caller for extension matching. For example, if 2 is set in the actions to ***Dial by Number***, a caller must dial 2 and then 20 for extension 20.
   - **Menu Prompt Usage:** *Default = Each menu uses own prompt.* Each time profile option used by an auto attendant can have its own set of actions and therefore may require a separate actions prompt to be played after the appropriate greeting prompt. The settings ***Each menu uses own prompt*** does that. Alternately one of the menu options can be selected as the menu option prompt played at all times of day.
   - **Night Service:** *Default = On.* If selected, when the system is in night service, the auto attendant will switch to using its out of hours greetings and menu actions. If not selected, when the system is in night service, the auto attendant will continue using the greetings and menu options as determined by its own time profile settings.
   - **Emergency Service:** *Default = Off*. This field indicates when the auto attendants emergency greeting option has been enabled by someone using the auto attendant. The field can also be used to disable the emergency greeting without having to go through the auto attendant menu.
4. After clicking **Save**, a summary menu is displayed. Click **More Settings >>**.

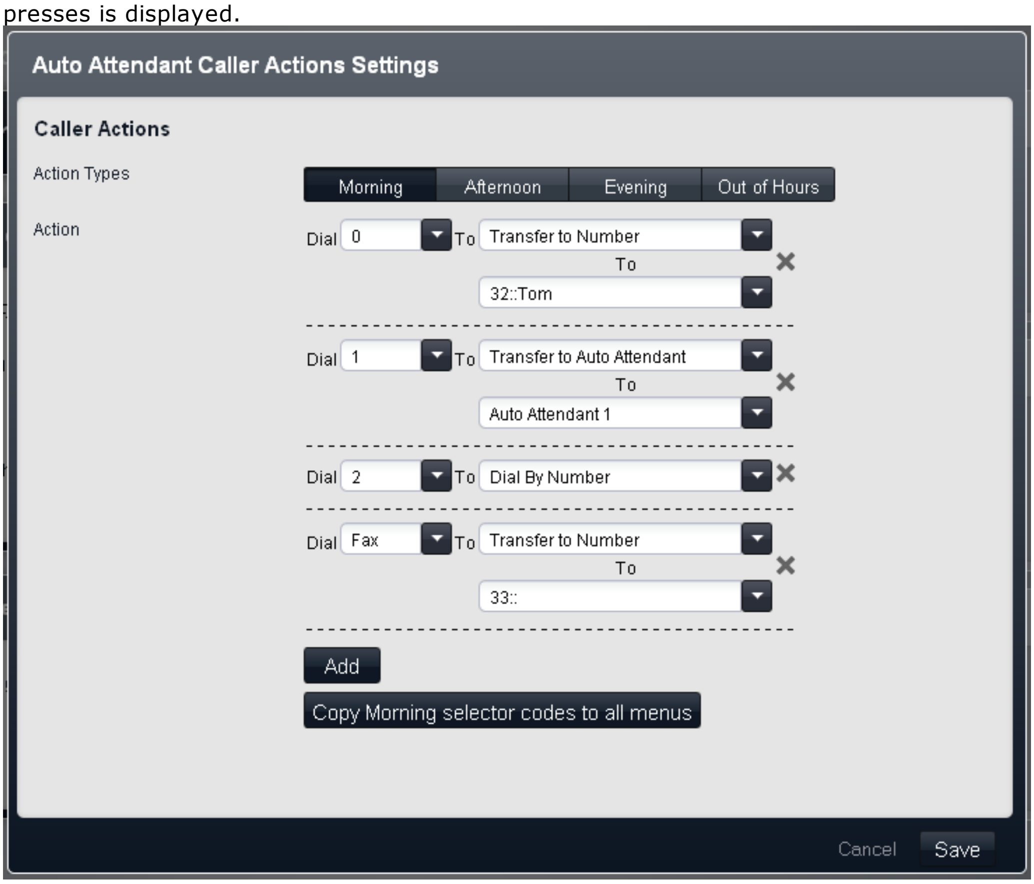
5. The detail settings of the new auto attendant are displayed.



6. Adjust the settings as required for when the auto attendant should operate in its morning, afternoon, evening and out of hours modes. Note the numbers shown in the menu. They are internal numbers that can be dialed to record the auto attendant prompts 179.

- If the greetings time periods overlap, the greeting and actions used is the first one that is valid in the order morning, afternoon or evening. For call outside those times, the out of office settings are used.

7. Click **Caller Actions**. The menu for setting which keys presses callers can use and the action for those presses is displayed.



8. You can configure different sets of key presses for each time period. If you want them all to be the same, you can set up one time period and then use the copy button to copy the same button settings to all time periods.
   - To remove a key press, click on the **X** delete icon.
   - To add a key press, click **Add** and select the key press that the action should be used in response to.
   - You can add a **Fax** option. The system uses that as the destination for any calls where the auto attendant detects incoming fax tone rather than a caller. For this action select ***Transfer to Number*** and set the destination as an extension that has had its equipment type also set to ***Fax***.
9. Ensure that you have made notes of the settings as you need to include the options in the prompts recorded for the auto attendant.
10. Click **Save**. The auto attendant settings are displayed again.
11. Click **Save** again. The list of existing auto attendants is displayed.
12. You should now record greetings 179 for the auto attendant.
13. Test operation by dialing the internal access number 179 for the auto attendant that you have just created.
14. Once you are satisfied with the operation of the auto attendant, you can use it as the destination for external 180 lines or for line coverage.

9.3 Accessing an Attendant Internally

Each auto attendant is automatically assigned an internal number which can be used to access the auto attendant. This can be used for testing the auto attendant. It can also be used as the destination for call transfers and call forwarding.

Attendant	1	2	3	4	5	6	7	8	9
Auto Attendant Access	**7801**	**7802**	**7803**	**7804**	**7805**	**7806**	**7807**	**7808**	**7809**

9.4 Recording Prompts

The system uses a series of internal numbers to allow the recording of auto attendant prompts.

Dialing the appropriate number shown in the table below allows recording and playback of the matching auto attendant prompt. The numbers are also displayed in the web manager auto attendant menus.

It is important to understand that callers to an auto attendant hear more than one prompt

1. If the auto attendant's **Emergency Greeting** setting is enabled, the recorded emergency greeting if played.
2. Depending on the time profile being used, the morning, afternoon, evening or out of hours greeting is then played.
3. Finally the morning, afternoon, evening or out of hours menu is played.

	Auto Attendant								
Greeting Prompts	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**
Morning Greeting	7811	7812	7813	7814	7815	7816	7817	7818	7819
Afternoon Greeting	7821	7822	7823	7824	7825	7826	7827	7828	7829
Evening Greeting	7831	7832	7833	7834	7835	7836	7837	7838	7839
Out of Hours Greeting	7851	7852	7853	7854	7855	7856	7857	7858	7859
Emergency Greeting	7861	7862	7863	7864	7865	7866	7867	7868	7869
Action Prompts									
Morning Menu	7841	7842	7843	7844	7845	7846	7847	7848	7849
Afternoon Menu	7871	7872	7873	7874	7875	7876	7877	7878	7879
Evening Menu	7881	7882	7883	7884	7885	7886	7887	7888	7889
Out of Hours Menu	7891	7892	7893	7894	7895	7896	7897	7898	7899
Auto Attendant Access	**7801**	**7802**	**7803**	**7804**	**7805**	**7806**	**7807**	**7808**	**7809**

The auto attendant access numbers shown above allow internal access to an auto attendant. Calls can be transferred to these numbers.

9.5 Answering External Calls with an Attendant

When Do Calls Go to an Auto Attendant?

The system supports the configuration of up to 9 auto attendant services to answer and redirect calls. If an auto attendant has been configured, it can be used to answer calls on external lines as follows:

- **Unanswered Calls Coverage**
  Each trunk or trunk channel is configured with a coverage destination for its incoming calls. It can also be configured with a selected auto attendant that will be used to answer unanswered calls at the trunk or trunk channels coverage destination. When the auto attendant is used and the delay before it answers unanswered call can be adjusted.

- **Immediate Call Coverage**
  The delay used for unanswered call coverage can be set to immediate. When that is done, the auto attendant will answer before the trunk or trunk channels configured coverage destination.

- **Answering Specific External Calls**
  On trunks that support the setting of separate DID or call-by-call destinations, an auto attendant can be selected as the destination for matching calls. This allows auto attendants to be used to answer incoming calls to specific numbers separate from the trunk channels own coverage destination settings. This option is not supported on analog trunks.

- **Manual Answering**
  Each auto attendant has an internal number assigned to it. That number can be used by extension users to transfer calls to the auto attendant.

Configuring Auto Attendant Call Coverage

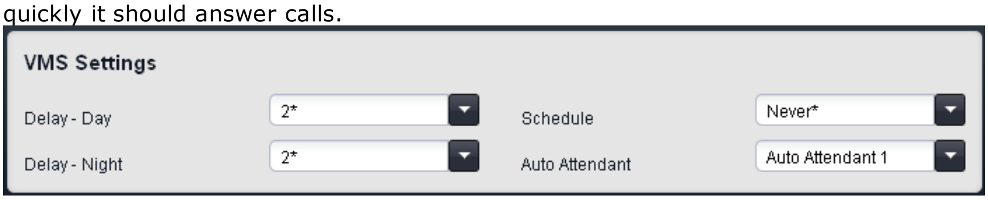
1. The method for accessing a trunk's VMS Settings depends on the trunk type.

   - **Analog Trunks**
     Select **System** in the menu bar and click **Trunks**. Select the trunk, click **View Details** and then **Advanced**. The trunk's VMS Settings are part of the menu now displayed.

   - **PRI Trunks**
     Select **System** in the menu bar and click **Trunks**. Select the trunk and then click the edit icon in the **Channel Setup** panel. Select the channels and click **View Details**.

   - **SIP Trunks**
     Select System in the menu bar and click SIP Trunks. Select the trunk and then click the edit icon in the **Channel Setup** panel. Select the channels and click **Details**.

2. Adjust the **VMS Settings** to set which auto attendant should be used, when it should be used and how quickly it should answer calls.



   - **Delay - Day:** *Default = 2. Range = 0 to 6 (number of rings).*
     Set the number of rings before an unanswered call should be redirected the selected auto attendant when the system is not running in night service mode and the **VMS Schedule** is set to ***Always*** or ***Days Only***.

   - **Delay - Night:** *Default = 2. Range = 0 to 6 (number of rings).*
     Sets the number of rings before an unanswered call should be redirected to the selected auto attendant when the system is running in night service mode and the **VMS Schedule** is set to ***Always*** or ***Night Only***.

   - **Schedule:** *Default = Never.*
     This option determines when the **VMS Delay** settings above should be used and unanswered calls redirected to the selected auto attendant. The options are:

     - ***Always***
       Redirect calls when the system is in both day and night service modes.

     - ***Day Only***
       Redirect calls only when the system is not in night service.

     - ***Night Only***
       Redirect calls only when the system is in night service.

     - ***Never***
       Do not redirect calls.

   - **Auto Attendant:** *Default = Auto Attendant 1.*
     This field allows selection of which auto attendant is used.

3. Click **Save**.

9.6 Changing an Auto Attendants Language

By default each auto attendant uses the same language prompts as the system language setting. However, the language used by a particular auto attendant can be changed.

Warning: Installed and Available Languages

By default not all languages are included in the voicemail/auto attendant prompt files on the system. If the language required is not present, either ***UK English*** or ***US English*** is used. Additional languages can be loaded using IP Office Manager, they cannot be loaded using Basic Edition Web Manager. The languages present by default depend on the type of System SD card used by the system:

- **IP Office A-Law SD Card:** UK English, French and Spanish.
- **IP Office U-Law SD Card:** US English, Candian French and Latin Spanish.
- **PARTNER SD Card:** UK English, French and Spanish.
- **Norstar SD Card:** UK English, French, Arabic.

Changing the Auto Attendant Language

1. Click **Incoming Call Management** in the menu bar and then click **Auto Attendant**.
2. Double click on the details of the required auto attendant.
3. In the **Language** column, select the required language. Possible languages are:
   - ***Arabic***, ***Cantonese***, ***Danish***, ***Dutch***, ***English (UK)***, ***English (US)***, ***Finnish***, ***French***, ***French (Canada)***, ***German***, ***Italian***, ***Korean***, ***Mandarin***, ***Norwegian***, ***Polish***, ***Portuguese***, ***Portuguese (Brazil)***, ***Russian***, ***Spanish***, ***Spanish (Argentina)***, ***Spanish (Latin)***, ***Spanish (Mexico)***, ***Swedish***, ***Turkish***.
4. Click **Save**.

Chapter 10. Restricting Outgoing Calls

10. Restricting Outgoing Calls

The system provides a number of methods to restrict which external numbers a user can or cannot dial.

- **Allowed Number Lists** 188 / **Disallowed Number Lists** 186
  These lists are used in define numbers that can or cannot be dialed. Users are then associated with the different lists.

Allowed Numbers	Each allowed list contains external telephone numbers that members of the list are allowed to dial. The allowed lists to which a user is assigned override any disallowed lists to which they are also assigned. The numbers in the user's assigned allowed lists also override the **Calls Barred** and **Outgoing Call Restrictions** 43 settings that may be applied to a user. There are eight lists, each containing up to 10 numbers. Each number can use the telephone dialing digits 0 to 9, *, # and can be up to 28 digits long. You can also use the ? character as a single digit wildcard.
Disallowed Numbers	Each disallowed list contains external telephone numbers that users who are assigned to the list cannot dial. There are eight lists, each containing up to 10 numbers. Each number can use the telephone dialing digits 0 to 9, *, # and can be up to 28 digits long. You can also use the ? character as a single digit wildcard. Numbers in the disallowed lists of which a user is a member are overridden if they also appear in the allowed numbers lists, emergency number list of which the user is a member and also by marked system speed dials.
Emergency Numbers	You can enter 10 emergency phone numbers into this list. This list is applied to all users and overrides any dialing restrictions that may also be applied to the users. Use of numbers in the list can be viewed using a 911-View/E-View 220 button.

- **Account Codes** 190
  Each user can be configured to need to enter a valid account code whenever they make an external call.
- **Outgoing Call Restrictions** 185
  For each user, the type of external calls that the user is able to make can be configured.
- **Marked Speed Dials** 191
  When a user uses a stored system speed dial number, the actual number dialed is subject to all the call barring methods as if the user had dialed the number directly. However system speed dials set as 'marked speed dials' override any call restrictions.
- **Night Service** 210
  When the system is set to night service, any users in the **Night Service Group** need to enter the system password when making an external call.

10.1 Barring a User from Making External Calls

You can restrict a user from make any outgoing external calls except to numbers in the **Emergency Numbers List** and any **Allowed Lists** of which they are a member. Th

Barring a User from Making External Calls

1. Click on **User** in the menu bar.
2. Scroll the list of users to the required user and click on **View Details**.
3. Change the **Calls Barred** setting to ***Yes*** to restrict the caller's external calls.
4. Click **Save**.

10.2 Restricting a User's External Calls

Restricting a User's External Calls

1. Click on **User** in the menu bar.
2. Scroll the list of users to the required user and click on **View Details**.
3. Click **Advanced**.
4. Select the required setting for the user's Outgoing Call Restrictions setting.
   - **Outgoing Call Restrictions:** *Default = No Restriction.*
     For each user, this field sets the type of outgoing external calls that the user can normally make. Any restrictions applied do not apply to numbers in the **Emergency Numbers List** and to numbers in any **Allowed Lists** of which the user is a member.
     - ***No Restrictions***
       The user can make outgoing external calls. The **Allowed Lists** and **Disallowed Lists** of which the user is a member still apply.
     - ***Inside only***
       The user can only make internal calls.
     - ***Local only***
       The user can only make outgoing external calls to numbers matching local numbers.
5. Click **Save**.

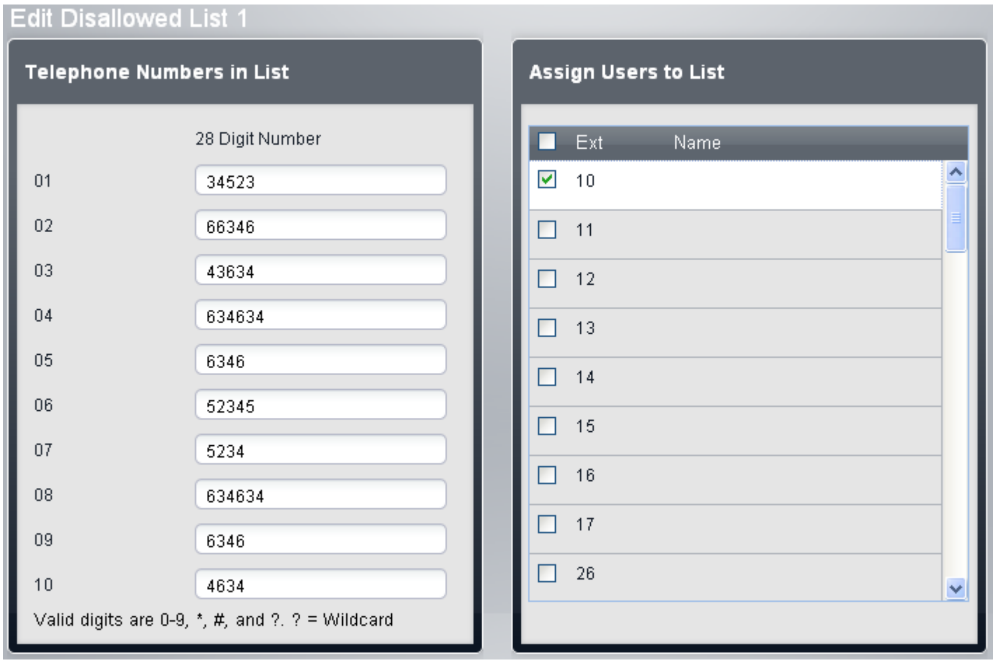
10.3 Setting Up Disallowed Numbers

Each disallowed list contains external telephone numbers that users who are assigned to the list cannot dial. There are eight lists, each containing up to 10 numbers. Each number can use the telephone dialing digits 0 to 9, *, # and can be up to 28 digits long. You can also use the ? character as a single digit wildcard.

Numbers in the disallowed lists of which a user is a member are overridden if they also appear in the allowed numbers lists, emergency number list of which the user is a member and also by marked system speed dials.

To Edit an Disallowed Numbers List

1. From the menu bar, click on **User**.
2. The **Outgoing Calls** panel next to the list of users gives a summary of the currently configured lists. Click on the edit icon.
3. From the **List Management** table, select the **View Details** link of the list that you want to edit.



- The **Telephone Numbers in List** panel displays the allowed numbers. Edit the numbers as required.
- The **Assign Users to List** panel is used to set which users are assigned to the list.

4. When completed click **Save**.
5. To access another list click on **<< Previous List** or **Next List >>**. Alternatively click on **<< Back** to return to the table of all the lists.

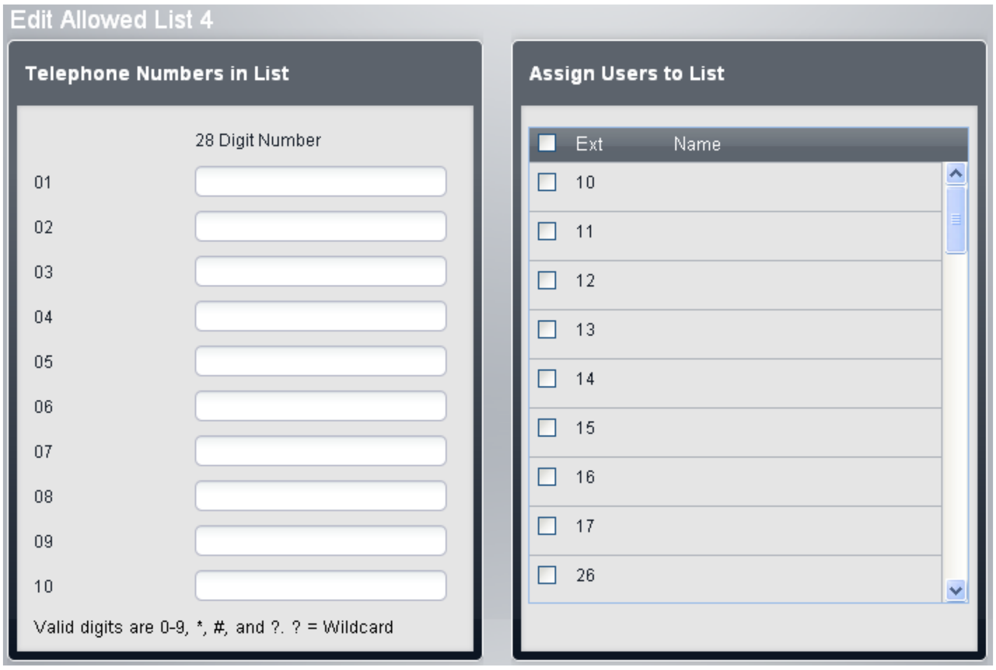
10.4 Setting Up Allowed Numbers

Each allowed list contains external telephone numbers that members of the list are allowed to dial. The allowed lists to which a user is assigned override any disallowed lists to which they are also assigned. The numbers in the user's assigned allowed lists also override the **Calls Barred** and **Outgoing Call Restrictions** 43 settings that may be applied to a user.

There are eight lists, each containing up to 10 numbers. Each number can use the telephone dialing digits 0 to 9, *, # and can be up to 28 digits long. You can also use the ? character as a single digit wildcard.

To Edit an Allowed Numbers List

1. From the menu bar, click on **User**.
2. The **Outgoing Calls** panel next to the list of users gives a summary of the currently configured lists. Click on the edit icon.
3. From the **List Management** table, select the **View Details** link of the list that you want to edit.



   - The **Telephone Numbers in List** panel displays the allowed numbers. Edit the numbers as required.
   - The **Assign Users to List** panel is used to set which users are assigned to the list.

4. When completed click **Save**.
5. To access another list click on **<< Previous List** or **Next List >>**. Alternatively click on **<< Back** to return to the table of all the lists.

10.5 Using Account Codes

Extensions can be required to enter a valid account code when they make an outgoing external call. The **Account Code Entries** list contains the account codes that are accepted as being valid and the selected users who are required to enter one of these codes, ie. the users who are set to **Forced Account Code Entry**.

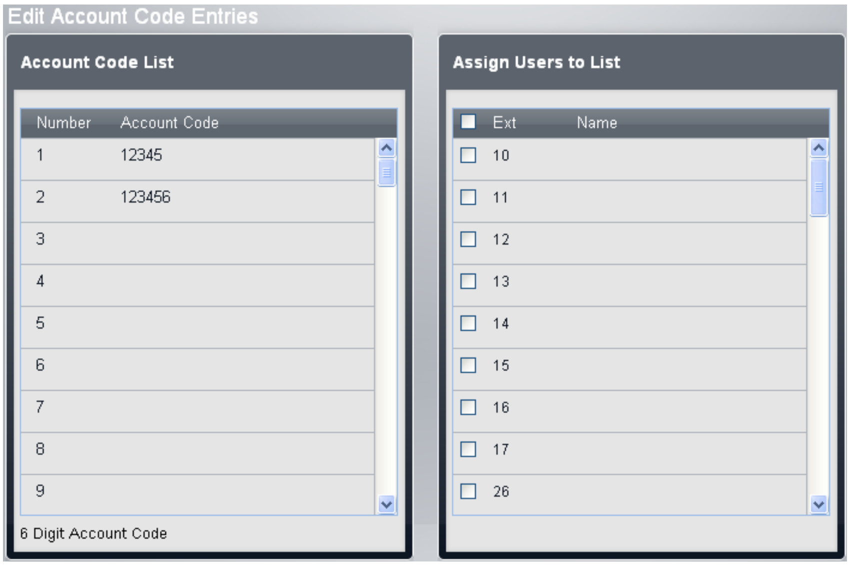
Account codes are commonly used to control cost allocation and out-going call restriction. The account code used on a call is included in the call information output by the system SMDR call log. Users can enter an account code during a call using an **Account Code Entry** button. Once a user has entered an account code with a call, only that user can change that calls account code by entering another one.

Once a call has been completed using an account code, the account code information is removed from the user's call information. This means that redial functions will not re-enter the account code.

All users (except analog phones) can also enter voluntary account codes at any time during a call by using an **Account Code Entry** button. Voluntary account codes are recorded in the same way as forced account codes but are not validated.

To Edit the Account Codes List

1. From the menu bar, click on **User**.
2. The **Outgoing Calls** panel next to the list of users gives a summary of the currently configured lists. Click on the edit icon.
3. From the **List Management** table, select the **View Details** link of the list that you want to edit.



   - The **Account Code List** panel displays the account codes. Edit these as required.
   - The **Telephone Numbers in List** panel displays the allowed numbers. Edit the numbers as required.
   - The **Assign Users to List** panel is used to set which users are assigned to the list.
4. When completed click **Save**.
5. To access another list click on **<< Previous List** or **Next List >>**. Alternatively click on **<< Back** to return to the table of all the lists.

Setting a User to Forced Account Code Entry

The user's forced account code entry setting can be changed directly. Doing so will also change their assigned or unassigned status in the account codes list.

1. Click on **User** in the menu bar.
2. Scroll the list of users to the required user and click on **View Details**.
3. Click **Advanced**.
4. Select the required setting for the user's **Forced Account Code Entry** setting.
   - **Forced Account Code Entry:** *Default = Off.*
     For each user, if this setting is selected, that user is required to enter an account code from the **Account Codes** list when making an external call. This can only be overridden by use of the **Password** 65 to make a call.
5. Click **Save**.

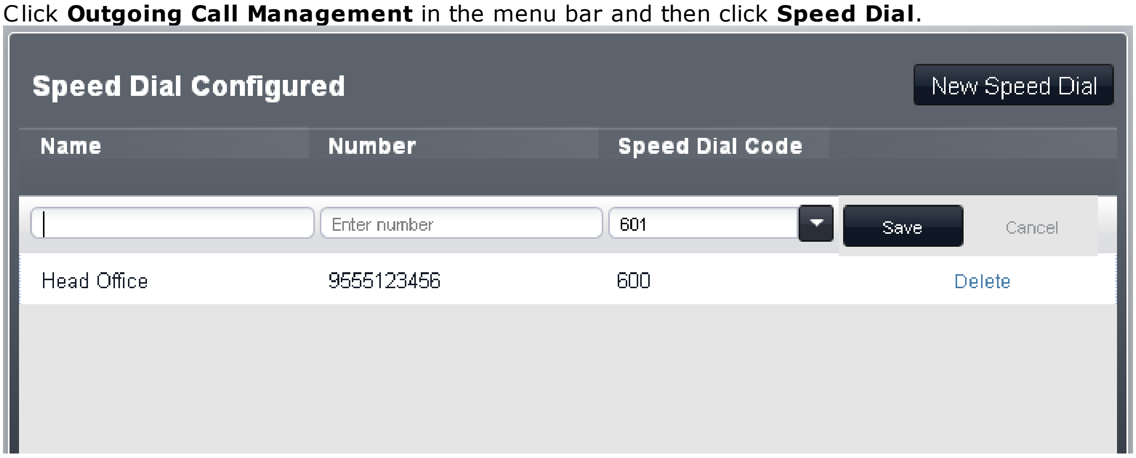
10.6 Using Marked Speed Dials

Some system speed dials can be set as being 'marked speed dials'. These can be used to override restrictions that would otherwise be applied if the user dialed that number directly or used the same number in an unmarked speed dial.

- Users can use a marked speed dial even if the number it dials is in one of the their assigned disallowed number lists.
- Users can use a marked speed dials even when the extension from which they are dialing is locked.
- Users in the night service group can use a marked speed dial when the system is in night service without having to enter the system password.

Creating Marked Speed Dials

1. Click **Outgoing Call Management** in the menu bar and then click **Speed Dial**.



2. Depending on whether you want to add or edit a speed dial:
   - To add a new speed dial click **New Speed Dial**.
   - To edit an existing speed dial, double click on the speed dials existing details.
3. Check the speed dial name and number are correct. To make the speed dial a marked speed dial, add a * to the front of the number. When the speed dial is used, the * is not included in the dialed number. If a * is required in the dial number, the speed dial should be start with **.
4. Click **Save**.

Chapter 11.
Voicemail Configuration

11. Voicemail Configuration

All systems support user voicemail as standard. A mailbox is automatically created for every user and its use is enabled.

This section covers those settings that can be controlled and adjusted through the system configuration. There are additional settings that can be adjusted by the mailbox user through the menu prompts provided by the voicemail mailbox when collecting messages. For details of those options refer to the appropriate mailbox user guide.

When Do Calls Go To a User's Mailbox?

If a user has voicemail enabled, calls directed to ring at that user's extension go to the user's voicemail after having rung them for the time set by the user's **Voicemail Coverage Rings** setting (approximately 15 seconds by default). For incoming external calls, this will apply if the user is set as the line's **Coverage Destination**.

- The above does not apply for calls altering just on a line appearance button that the user has assigned or alerting the user as part of a hunt group.

11.1 Licensing

The system's voicemail services, both user mailboxes and call auto attendants, operate without requiring any licenses. However the number of simultaneous calls and the message storage capacity is controlled by the number of voicemail licenses added to the system's configuration.

- **Number of Simultaneous Calls**
  Without any voicemail licenses, the number of simultaneous calls using the voicemail services is limited to 2. Each voicemail license added enables an additional 2 simultaneous calls, up to a maximum of 6.

- **Voicemail Storage Capacity**
  The user mailboxes are also limited to a maximum of 15 hours storage for messages, greeting and prompts. However, each additional voicemail license added also enables an additional 5 hours of storage up to a maximum of 25 hours.

11.2 Setting the Mailbox Mode

The user mailbox controls can operate in either of two ways; IP Office mode or Intuity mode. The selected mode affects the keys presses used for different actions when the mailbox is accessed. The system's mailbox mode setting affects all user mailboxes.

Separate mailbox user guides are available for each mode.

Changing the Mailbox Mode

1. Click on **System** in the menu bar and then click on **Switch**.
2. The **Voicemail Mode** setting is shown in the **System Parameters** panel.
3. Select the mode required and click **Save**.

11.3 Switching Voicemail On/Off

The user can turn the use of voicemail to answer their calls on or off. Depending on the user's phone, they can do this using options in the phones own menus. If this is not possible, you can provided their phone with a VMS Cover button which they can then use to turn voicemail on or off.

Programming a Voicemail Cover Button for a User

1. Click on **User** in the menu bar.
2. Highlight the required user by clicking on them.
3. Click on the edit icon in the **Button Programming** panel on the right.
4. The current button programming for the user is displayed.
5. In the **Program Buttons** panel click on the button that you want to program. The current settings of that button are displayed in the **Button Specification** panel.
6. Click on the **Assign Feature** button.
7. Select **Programming Features**.
8. From the **Coverage** panel select **VMS Cover**.
9. Click **OK**.

Turning a User's Voicemail On or Off

If required, you can turn a user's use of voicemail on or off through the system configuration.

1. Click on **User** in the menu bar.
2. Scroll the list of users to the required user and click on **View Details**.
3. In the **Voicemail** panel, use the Automatic VMS Cover setting to switch voicemail use on or off.
4. Click **Save**.

11.4 Setting the Voicemail Answer Delay

The delay before voicemail answers any unanswered call alerting the user can be adjusted. The normal default is approximately 15 seconds. Setting a delay does not enable or disable the use of voicemail, that is done using the **Automatic VMS Cover** 196 setting.

Changing a User's Mailbox Answer Delay

1. Click on **User** in the menu bar.
2. Scroll the list of users to the required user and click on **View Details**.
3. Click **Advanced**.
4. In the **Advanced Parameters** panel, use the **VMS Cover Ring** setting to adjust the delay before unanswered calls are answered by voicemail.
   - **VMS Cover Ring:** *Default = 3 (15 seconds). Range = 0 to 9.*
     For a user with **Automatic VMS Cover** enabled, this value sets how long a call alerts the user's extension before it is redirected to voicemail.
     - The option ***0*** for immediate voicemail is available. 0 is the only value usable for phantom extensions. If selected it has the following effects.
       - For a call that would have otherwise have alerted at the extension, the call now goes immediately to voicemail.
       - If the extension has call forwarding set, the forwarded call will continue ringing at the forwarding target rather than going to voicemail.
       - If the extension is the target for another extension's call forwarding, the call will go immediately to the forwarding extension's voicemail.
5. Click **Save**.

11.5 Setting a Voicemail Code

Each user mailbox can be given a voicemail code that must be entered when accessing the mailbox to collect messages and change mailbox settings. The user can set and change their own voicemail code through the mailbox prompts. However, if necessary, you can also set and change the voicemail code through the system configuration.

Changing a User's Voicemail Code

1. Click on **User** in the menu bar.
2. Scroll the list of users to the required user and click on **View Details**.
3. In the **Voicemail** panel, enter the required code in the **Voicemail Code** field.
4. Click **Save**.

11.6 Mailbox Breakout Settings

Callers hearing a user's mailbox greeting can be allowed to dial a key sequence in order to be transferred to another number rather than leave a message. For example, they can be prompted to "Dial 0 to be transferred to the Sales team." This is called mailbox breakout.

For each user, you can configure 3 breakout numbers, triggered by caller's dialing the appropriate digits during the mailbox greeting. Those digits are 0, 2 or 3. The destination numbers for the transfer must be internal.

- **Important**
  If you change a user's mailbox breakout settings, remember to inform the user so that they can change their recorded mailbox greeting

Changing a User's Mailbox Breakout Settings

1. Click on **User** in the menu bar.
2. Scroll the list of users to the required user and click on **View Details**.
3. Click **Advanced**.
4. In the **DTMF Breakout** panel, set the numbers that should be used for each of the mailbox breakout options.
5. Click **Save**.

11.7 Mailbox Language

By default the mailbox services provided to a user use the language set for the whole system 137. However, a different language can be selected for a user. If the voicemail service has prompts for that language available, it will then use them.

Not that changing a user's language may also affect the language used for menus and information displayed on their phone.

Warning: Installed and Available Languages

By default not all languages are included in the voicemail/auto attendant prompt files on the system. If the language required is not present, either ***UK English*** or ***US English*** is used. Additional languages can be loaded using IP Office Manager, they cannot be loaded using Basic Edition Web Manager. The languages present by default depend on the type of System SD card used by the system:

- **IP Office A-Law SD Card:** UK English, French and Spanish.
- **IP Office U-Law SD Card:** US English, Candian French and Latin Spanish.
- **PARTNER SD Card:** UK English, French and Spanish.
- **Norstar SD Card:** UK English, French, Arabic.

Changing a User's Language Preference

1. Click on **User** in the menu bar.
2. Scroll the list of users to the required user and click on **View Details**.
3. In the **Details** panel, use the **Language** control to select the preferred language required by the user. Possible languages are:
   - ***Arabic***, ***Cantonese***, ***Danish***, ***Dutch***, ***English (UK)***, ***English (US)***, ***Finnish***, ***French***, ***French (Canada)***, ***German***, ***Italian***, ***Korean***, ***Mandarin***, ***Norwegian***, ***Polish***, ***Portuguese***, ***Portuguese (Brazil)***, ***Russian***, ***Spanish***, ***Spanish (Argentina)***, ***Spanish (Latin)***, ***Spanish (Mexico)***, ***Swedish***, ***Turkish***.
4. Click **Save**.

11.8 Using Voicemail Transfer

Users can transfer calls direct to another user's mailbox. In some cases they can do this using standard controls provided by their phone, in which case refer to the appropriate telephone user guide.

For users whose phone does not provide its own control for voicemail transfer, you may be able to add a programmable button set to the voicemail transfer. After pressing the button, the user's current call is put on hold and the user can then enter the target extension number to indicate the mailbox required.

Programming a Voicemail Transfer Button

1. Click on **User** in the menu bar.
2. Highlight the required user by clicking on them.
3. Click on the edit icon in the **Button Programming** panel on the right.
4. The current button programming for the user is displayed.
5. In the **Program Buttons** panel click on the button that you want to program. The current settings of that button are displayed in the **Button Specification** panel.
6. Click on the **Assign Feature** button.
7. Select **Programming Features**.
8. From the **Answering Calls** panel select **VMS Transfer**.
9. Click **OK**.

11.9 Using Voicemail Email

Voicemail email allows a mailbox user to receive an email when they have a new voicemail message. That email can be a simple alert or it can include a copy of the voicemail message.

The use of voicemail email require the system to be configured with SMTP server settings and email addresses for users. Once setup, users are able to access email settings through their voicemail mailboxes prompts and select the voicemail email mode that they want used for their new messages:

- ***Off***
  Switches off the use of email for new message alerts.
- ***Copy***
  Send an email to the user's email address with the voicemail message attached. The method leaves the original message in the user's voicemail mailbox.
- ***Forward***
  Send an email to the user's email address with the voicemail message attached. This method deletes the original message from the user's voicemail mailbox
- ***Alert***
  Send an email alert about the new message but do not attach the message to the email.

- **! WARNING**
  The size of voicemail attachments is approximately 1MB per minute. The use of attachments will impose additional traffic on the email server and large attachments may be blocked.

Configuring Voicemail Email

The following steps are required to allow users to use voicemail email.

1. Configure the Email Server Settings 201.
2. Set User Email Addresses 202.

11.9.1 Configuring the Email Server Settings

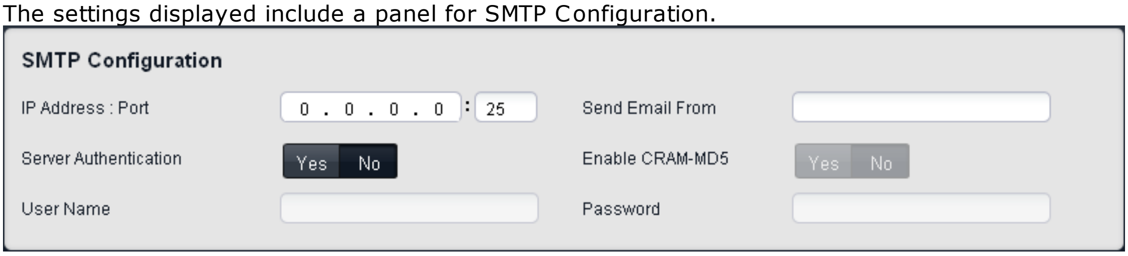
Voicemail email requires the system to be able to access an SMTP email server. It then sends email messages to user email addresses via this server.

This process requires the system to be given its own email address. In most cases, that will need to match the details of an email account configured on the SMTP server.

If the user email addresses are not hosted on the same email server, the email server will need to relay the messages from the system to the user email. Most email servers are not configured to relay or forward email messages by default. They will typically only forward emails if the sender's email address settings match a mailbox account on the server or has been specifically configured as an address to be relayed.

Configuring the Email Server Settings

1. Click on **System** in the menu bar and then click on **Switch**.
2. Click on **Advanced**.
3. The settings displayed include a panel for SMTP Configuration.



- **FQDN/IP Address:** *Default = 0.0.0.0*
  This field sets the IP address of the SMTP server being used to forward emails.
- **Port:** *Default = 25. Range = 0 to 65534.*
  This field sets the destination port on the SMTP server.
- **Send Email From:** *Default = Blank*
  This field sets the sender address to be used for emails from the system. Depending of the authentication requirements of the SMTP server, this may need to be a valid email address hosted by that server. Otherwise the SMTP email server may need to be configured to allow SMTP relay of this address.
- **Server Authentication:** *Default = On*
  This field should be selected if the SMTP server being used requires authentication to allow the sending of emails. When selected, the **User Name** and **Password** fields become available.
- **Enable CRAM-MD5:** *Default = Off.*
  This field should be selected if the SMTP server uses CRAM-MD5.
- **Use STARTTLS:** *Default = Off. (Release 9.0.3).*
  Select this field to enable TLS/SSL encryption. Encryption allows voicemail-to-email integration with hosted email providers that only permit SMTP over a secure transport.
- **User Name:** *Default = Blank*
  This field sets the user name to be used for SMTP server authentication.
- **Password:** *Default = Blank*
  This field sets the password to be used for SMTP server authentication.

4. Click **Save**.

11.9.2 Setting User Email Addresses

In order to use voicemail email features, an email address for the user must be entered into the system configuration. The user cannot do this themselves. However, once an address is set, the user is able to access voicemail email settings through their voicemail mailbox.

- If the email address is not one matching an account hosted by the email server to which the system initially sends emails, then the email server may need to additional configuration to allow email forwarding or relay by the email account being used by the system.

Setting the User Email Address

1. Click on **User** in the menu bar.
2. Scroll to the required user and click on **View Details**.
3. In the **Phone User** panel, enter the user's email address in the **Email** field.
4. Click **Save**.

11.9.3 Changing a User's Voicemail Email Mode

Once a user has an email address set in the system configuration, that user is able to switch voicemail email for their new voicemail messages on or off. They do this through the prompts provided when they access their mailbox. Refer to the appropriate mailbox user guides. However, through the system configuration you can see and, if required, change the setting for a user.

Changing a User's Voicemail Email Mode Setting

1. Click on **User** in the menu bar.
2. Scroll to the required user and click on **View Details**.
3. In the **Voicemail** panel, use the **Send to Email** control to select the required mode for the user's voicemail email.
   - ***Off***
     Switches off the use of email for new message alerts.
   - ***Copy***
     Send an email to the user's email address with the voicemail message attached. The method leaves the original message in the user's voicemail mailbox.
   - ***Forward***
     Send an email to the user's email address with the voicemail message attached. This method deletes the original message from the user's voicemail mailbox
   - ***Alert***
     Send an email alert about the new message but do not attach the message to the email.
4. Click **Save**.

Chapter 12.
Configuring Users and Extensions

12. Configuring Users and Extensions

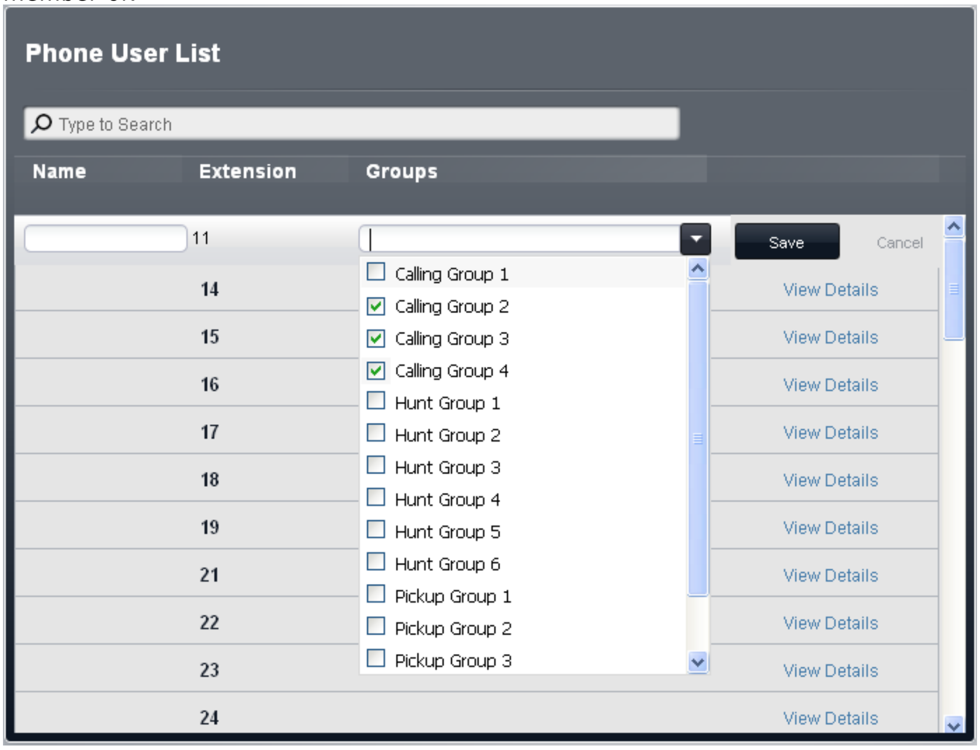
The system automatically creates a user entry in its configuration for every possible extension port that it could support. It does this regardless of whether physical equipment for the port is installed or not. User entries in the configuration that are not matched by a physical extension port are treated as phantom users 23. Therefore you do not need to add or delete users from the configuration

12.1 Changing a User's Group Memberships

You can edit the groups of which a user is a member through the user list.

Changing a User's Group Memberships

1. Click **User** in the menu bar.
2. The list of users is displayed. The groups column lists the user's current group memberships.
3. Scroll the list to the user who you want to edit and double click on the row.
4. Click on the field in the groups column. The drop-down list allows you to select which groups the user is a member of.



5. Set the user's group memberships and click **Save**.

12.2 Loudspeaker Paging

A loudspeaker paging device can be connected to an analog extension port on the system. Only one such connection is supported. That device can then be used as the paging target for several preset function.

Setting the Loudspeaker Extension

1. Click on **User** in the menu bar.
2. Scroll the list of users to the required user and click on **View Details**.
3. Click **Advanced**.
4. In the **Equipment** panel, use the **Type** drop-down list to select **Loudspeaker Paging**.
5. Click **Save**.

Paging the Loudspeaker Extension

The loudspeaker paging extension can be paged dialing 70 when making an internal call.

Making a Simultaneous Page

In addition to paging the loudspeaker paging device, users can also page the loudspeaker paging device and the members of **Calling Group 1** at the same time. This is done by dialing *70 when making an internal call

12.3 Door Phone Operation

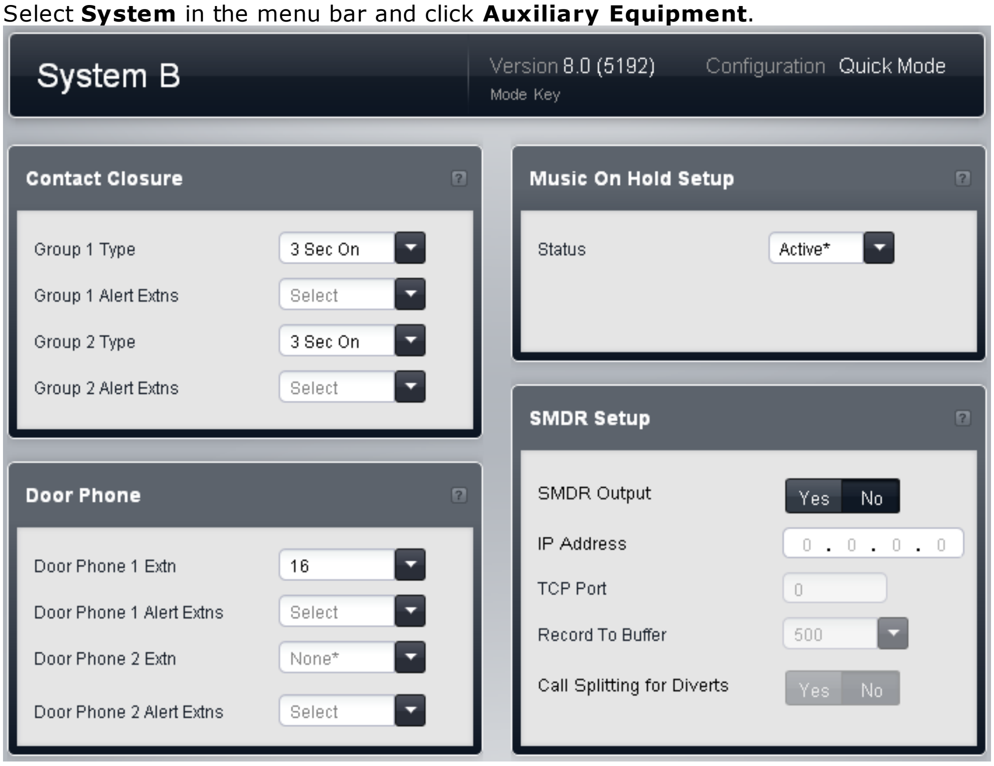
Up to two analog extension ports can be configured as door phones. When either door phone is taken off-hook, the other extensions configured below are alerted and can answer the door phone user. This option is typically used to connect a phone in a public area to a receptionist or similar.

Setting the Door Phone Extensions

1. Click on **User** in the menu bar.
2. Scroll the list of users to the required user and click on **View Details**.
3. Click **Advanced**.
4. In the **Equipment** panel, use the **Type** drop-down list to select **Door Phone 1** or **Door Phone 2**.
5. Click **Save**.

Setting the Door Phone Alert Extensions

1. Select **System** in the menu bar and click **Auxiliary Equipment**.



2. In the **Door Phone** panel, use the **Door Phone 1 Extn** or **Door Phone 2 Extn** drop downs to select the extension that is being used as a door phone.
3. Use the matching **Door Phone 1 Alert Extns** or **Door Phone 2 Alert Extns** drop down to select the extensions that should be alerted when the door phone goes off hook.
4. Click **Save**.

12.4 Using the System Password

This 4 digit password, if set, is used for several functions:

- The **Night Service** button user must enter the system password in order to switch night service on.
- The users in the Night Service group must enter the system password in order to make an external call to any numbers not in the Emergency Numbers list or marked speed dial numbers.
- Override station lock.
- Override forced account code.
- Change an auto attendant's emergency greeting setting.

Setting the System Password

1. Click on **System** in the menu bar and then click on **Switch**.
2. In the **Password** field, enter the system password. This must be a four digit number.
3. Click **Save**.

12.5 One Touch Transfer

One touch transfer operation can be used with a number of different button types. With a call currently connected, the user can start the transfer process by pressing a button pre configured for the destination rather than having to first press **TRANSFER**.

The button types that support this operation are listed below. Buttons programmed for voice or page calls can be used.

- **Auto Dial ICM**
- **Auto Dial ICM - Page**
- **Group Calling - Ring**
- **Group Calling - Page**
- **Group Hunting - Ring**
- **Group Hunting - Page**
- **Simultaneous Page**

1. With a currently connected call, the user starts the transfer by pressing the button programmed for the transfer destination.
2. The system seizes an intercom button using the user's automatic line selection 144 setting. If no intercom buttons are available, the button press is ignored.
3. When an intercom button is seized, the system puts the connected call on hold pending transfer and makes the voice or page call to the transfer destination.
4. The user can switch between calls using the appropriate intercom and or line appearance for each call.
    - If the transfer destination is busy then the transfer cannot be completed. The user should press the appropriate appearance button for the held call to reconnect to the caller.
5. The user can complete the transfer by going on hook (replacing the handset, pressing **SPEAKER** or pressing **HEADSET** depending on how they were handling the call being transferred) or pressing **TRANSFER** or selecting the Complete soft key on the display.

- Calls transferred using one touch transfer are still subject to voicemail coverage or transfer return in the same way as normal transferred calls.
- Using this feature and trying to complete a transfer to a door-phone, or a loudspeaker paging extension, is not allowed. The transfer attempt is dropped and the original call remains on hold.

12.6 Page and Direct Calls

There are a number of ways to make page calls.

To Page a Group

For a call to a group of extensions, putting a * in front of the extension number of the group being called makes the call a page call. The group type is ignored, instead, all users who are members of the group, have phones that support auto answer and are currently not on another call, hear a single beep and can then hear you. However, you cannot hear the group members.

To Page the Loudspeaker Extension

A loudspeaker paging device can be connected to an analog extension port on the system. The equipment **Type** of that extension is then set to ***Loudspeaker Paging***. That device can then be accessed by any user by dialing 70 when making an internal call.

To Make a Simultaneous Page

In addition to loudspeaker paging *(see above)*, users can also page the loudspeaker paging device and the members of **Calling Group 1** at the same time by dialing *70 when making an internal call.

To Make a Direct Call to an Extension

For an internal call to an individual extension, putting ***** in front of the extension number of the user being called makes the call into a direct voice call. If supported by the phone being called, the call is automatically answered after the called users hears 3 beeps. If the phone called does not support auto answer, the call is turned into a normal call.

Unlike a page call, the called user can also speak without having to take any further action if their phone has a handsfree microphone. Otherwise they need to pickup the handset to be heard.

If the user called is already on a call when you attempt a direct voice call to them, you call is turned into a normal waiting call.

Chapter 13. Night Service

13. Night Service

Night service can be used to redirect incoming calls outside normal business hours. When night service is enabled:

- When the system is put into night service, the **Coverage Destinations** of all trunks and trunk channels are overridden. Instead all calls to which the **Coverage Destination** would have been applied are present to the users in the night service group. The only exception are calls routed to specific destinations by DID call mapping or Call-by-Call settings.
- The group is a ***Ring All*** groups, that is a call to the group will alert all the available group members (those not already on a call) until the call is answered.
- Any ring delays on trunks are ignored. Instead calls redirected to the **Night Service Group** members alert immediately.
- If a system password is set, users in the **Night Service Group** need to enter that number when dialing external calls. The only exception is if dialing numbers in the **Emergency Numbers** list and marked system speed dials. No additional outgoing call restrictions are applied to any users not in the **Night Service Group**.
- The auto attendant coverage options for lines change to their night service settings. This may alter whether auto attendant coverage is used for the line or not and the delay applied before coverage is applied.
- Those auto attendants with their own Night Service setting enabled, switch to using their out of hours settings regardless of their own time settings.

How is Night Service Controlled

The phone connected to the first extension on the system (the top-left most port on the system's control unit) is given a **Night Service** button. The extensionuser can then use that button to manually put the system into night service when required.

Configuring Night Service

1. The following steps are necessary to use night service:
   a. Add a Night Service Button 211.
   b. Add Users to the Night Service Group 212.
2. The following step is optional but recommended when using night service:
   a. Set a System Password 212.
3. The following steps can also be used with night service:
   a. Adjusting trunk auto attendant coverage 213.
   b. Overriding auto attendant time settings 214.

13.1 Adding a Night Service Button

The **Night Service** button feature is used on a button on the first extension on the system to turn night service on and off.

- You can only program a **Night Service** button on the first extension port on the system.
- This function is only supported on a button that includes LED/LCD indicator. The indicator is lit when the function is enabled.
- If the user has this feature enabled, removing this button will turn the feature off.
- Dialing restrictions for extensions not in the Night Service Group remain the same as during normal daytime operation.
- If you reassign the Night Service Button, it is removed from the button where it was previously assigned.
- Night service uses the System Password, if set, as follows:
    - You must enter the password when turning Night Service on or off.
    - Calls to numbers other than marked system dials or in the Emergency Numbers 58 list requires entry of the system password.
- If you have a voice messaging system, VMS Hunt Schedule determines when outside calls should ring voicemail. The status of the Night Service Button tells the voice messaging system to operate in day or night mode.
- The Night Service Button returns to the status (on/off) it was in immediately after a power failure or a system reset.
- Night Service is unavailable on T1 lines with Direct Inward Dialing (DID)..

Programming a Night Service Button

1. Click on **User** in the menu bar.
2. Highlight the required user by clicking on them.
3. Click on the edit icon in the **Button Programming** panel on the right.
4. The current button programming for the user is displayed.
5. In the **Program Buttons** panel click on the button that you want to program. The current settings of that button are displayed in the **Button Specification** panel.
6. Click on the **Assign Feature** button.
7. Select **System Features**.
8. Select **Night Service**.
9. Click **OK**.

13.2 Editing the Night Service Group

The members of the night service group are used as the destination for most external calls when the system is in night service 210. The night service group is a collective group, that is all members are alerted at the same time.

- When the system is put into night service, the **Coverage Destinations** of all trunks and trunk channels are overridden. Instead all calls to which the **Coverage Destination** would have been applied are present to the users in the night service group. The only exception are calls routed to specific destinations by DID call mapping or Call-by-Call settings.
- The group is a ***Ring All*** groups, that is a call to the group will alert all the available group members (those not already on a call) until the call is answered.
- Any ring delays on trunks are ignored. Instead calls redirected to the **Night Service Group** members alert immediately.
- If a system password is set, users in the **Night Service Group** need to enter that number when dialing external calls. The only exception is if dialing numbers in the **Emergency Numbers** list and marked system speed dials. No additional outgoing call restrictions are applied to any users not in the **Night Service Group**.

Editing the Night Service Group

1. Click **Incoming Call Management** in the menu bar and then click **Groups**.
2. The list of groups supported by the system is displayed.
3. Scroll the list to the **Night Service Group** and double click on it.
4. The **Edit Group** menu is shown for the group.
5. Select the users who should be members of the group.
6. Click **Save**.

Changing a User's Group Memberships

1. Click **User** in the menu bar.
2. Scroll the list to display the user whose group membership you want to adjust.
3. Double click on the user.
4. Click on the box in the **Groups** column. The drop-down lists all the groups and shows the groups for which the user is a selected member. Select the **Night Service Group**.
5. Click **Save**.

13.3 Setting the System Password

The system password is used for a number of functions. In conjunction with night service operation, if the system password is set, it restricts the external numbers that members of the night service group can dial.

Note however that the password also affects a other functions if set.

- The **Night Service** button user must enter the system password in order to switch night service on.
- The users in the Night Service group must enter the system password in order to make an external call to any numbers not in the Emergency Numbers list or marked speed dial numbers.
- Override station lock.
- Override forced account code.
- Change an auto attendant's emergency greeting setting.

Setting the System Password

1. Click on **System** in the menu bar and then click on **Switch**.
2. In the **Password** field, enter the system password. This must be a four digit number.
3. Click **Save**.

13.4 Adjusting Trunk Auto Attendant Coverage

An auto attendants can be used to answer incoming calls on a trunk or trunk channel. Through the trunk or trunk channels own settings, you can configure which auto attendant is used. You can alter whether the auto attendant is used during day service, night service or both. You can also adjust the delay used to answer calls during day service and during night service.

Note that these settings are not used for calls on the trunk or trunk channel that have been routed to a destination using DID call mapping or Call-by-Call settings.

Changing the Auto Attendant Coverage Settings of a Line

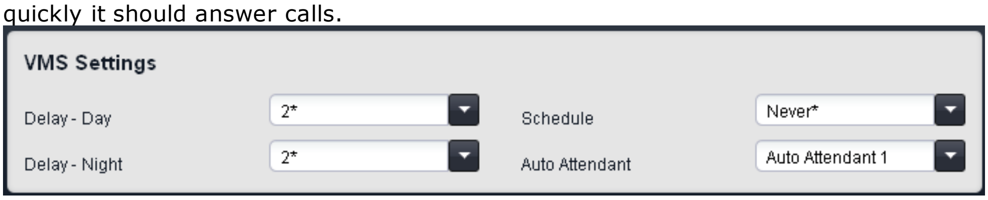
1. The method for accessing a trunk's VMS Settings depends on the trunk type.

   - **Analog Trunks**
     Select **System** in the menu bar and click **Trunks**. Select the trunk, click **View Details** and then **Advanced**. The trunk's VMS Settings are part of the menu now displayed.

   - **PRI Trunks**
     Select **System** in the menu bar and click **Trunks**. Select the trunk and then click the edit icon in the **Channel Setup** panel. Select the channels and click **View Details**.

   - **SIP Trunks**
     Select System in the menu bar and click SIP Trunks. Select the trunk and then click the edit icon in the **Channel Setup** panel. Select the channels and click **Details**.

2. Adjust the **VMS Settings** to set which auto attendant should be used, when it should be used and how quickly it should answer calls.



   - **Delay - Day:** *Default = 2. Range = 0 to 6 (number of rings).*
     Set the number of rings before an unanswered call should be redirected the selected auto attendant when the system is not running in night service mode and the **VMS Schedule** is set to ***Always*** or ***Days Only***.

   - **Delay - Night:** *Default = 2. Range = 0 to 6 (number of rings).*
     Sets the number of rings before an unanswered call should be redirected to the selected auto attendant when the system is running in night service mode and the **VMS Schedule** is set to ***Always*** or ***Night Only***.

   - **Schedule:** *Default = Never.*
     This option determines when the **VMS Delay** settings above should be used and unanswered calls redirected to the selected auto attendant. The options are:

     - ***Always***
       Redirect calls when the system is in both day and night service modes.

     - ***Day Only***
       Redirect calls only when the system is not in night service.

     - ***Night Only***
       Redirect calls only when the system is in night service.

     - ***Never***
       Do not redirect calls.

   - **Auto Attendant:** *Default = Auto Attendant 1.*
     This field allows selection of which auto attendant is used.

3. Click **Save**.

13.5 Overriding Auto Attendant Time Settings

Each auto attendant can have its own time settings for morning, afternoon, evening and out of hours. These control which greetings the auto attendant uses and the options it provides to callers at those different times.

Each auto attendant also has a **Night Service** setting. This is used as follows:

- Those auto attendants with their **Night Service** option enabled will immediately switch to using their out of hours settings when the system is put into night service.
- Those auto attendants with their **Night Service** option disabled will continue to follow their own time settings regardless of whether the system is in night service of not.

Checking and Changing an Auto Attendants Night Service Setting

1. Click **Incoming Call Management** in the menu bar and then click **Auto Attendant**.
2. The list of auto attendants is displayed.
3. To change the setting for a particular auto attendant in the list, double-click on the auto attendant. Change the **Night Service** setting and click **Save**.

Chapter 14. Button Programming

14. Button Programming

Most Avaya phones have buttons to which functions can be assigned. For some phones, additional buttons can also be added by attaching a button module to the phone.

Default Buttons and Button Numbering

The default button assignment depends on whether the system's **Mode** 65 is set as ***Key*** or ***PBX***.

- **Key Mode**
  - **01-02: Intercom Buttons**
    The first two buttons are used as **Intercom 1** and **Intercom 2** buttons for internal calls. This function is automatically assigned to the buttons by the system and cannot be overridden by the system administrator or extension user.
  - **03+: Line Buttons**
    Buttons 03 and upwards up to the system's **Number of Lines** setting are used as line appearance buttons for external calls. These cannot be overridden by the extension user.
  - **Other Buttons**
    Any additional buttons can be used for additional functions. These buttons can be programmed by the system administrator and, for some functions, the extension user.
  - **Button Numbering**
    All buttons are numbered from 01, from left to right, starting from the bottom row up.
- **PBX Mode**
  - **01-03 (ETR 01-02): Call Appearance Buttons**
    The first three buttons (two only on ETR phones) are used call appearance buttons for making and answering calls. They can be used for both internal and external calls. This function is automatically assigned to the buttons by the system and cannot be overridden by the system administrator or extension user.
  - **Other Buttons**
    Any additional buttons can be used for additional functions. These buttons can be programmed by the system administrator and, for some functions, the extension user.
  - **Button Numbering**
    All buttons are numbered from 01, from left to right, starting from the bottom row up. However for 1400, 9400 and 9500 Series telephones, buttons are numbered from 01 from left to right, starting from the top row downwards.

Programming a User's Buttons

1. From the menu bar, click **User**.
2. In the **Phone User List**, locate the user who you want to edit and click on the **View Details** link adjacent to their details.
3. On the right is a panel labeled **Button Programming**. Click on the edit icon in the panel.
4. The **Program Buttons** panel can work in either of two modes. To use it in photo mode, ensure that the **Select Handset** option is set to the type of telephone the user has. To use it in list mode, set the **Select Handset** option to ***Default***.
5. Select the button you want to program:
   - In photo mode, click on the button in the telephone picture.
   - In list mode, click on the button in the list of buttons.
6. The **Button Specification** panel shows the current settings of the button. You can use this panel to change the button's settings or to change its function.
   a. To change the buttons function, click **Assign Feature**. From the menu that is displayed, select the required function. Some functions require additional information, a menu to enter that information displayed when then is the case.
      - Intercom buttons are automatically programmed by the telephone system and cannot be overridden.
      - To remove the existing function from a button select ***Blank***.
   b. Click **OK**.
7. After making any changes, select any other button that you want to program.
8. When you have completed programming the user's buttons, click **Save**.
9. Click **<< Back** to return to the user list and program the buttons for other users.

14.1 Button Programming Functions

Function	Description	LED
911-View/Emergency View 220	A button set to this function indicates when a call has been made using a number in the system's Emergency Numbers 58 list. Pressing the button displays a list of such calls.	Yes
Absent Message 220	A button set to this function allows the user to set or clear an absent message for display on their phone. When set, the absent message is also displayed on other extensions when they call the user.	Yes
Account Code Entry 220	A button set to this function allows the user to enter an account code prior to making a call or during a call.	Yes
Active Line Pickup 220	A button set to this function allows the user to answer a call on a particular line. It can be used if the call is ringing, held or already answered by another extension.	–
Auto Dial - Intercom 220	A button set to this function allows the user to make a call to another specified extension. The button lamp will also indicate when that other extension is in use.	–
Auto Dial - Other 221	A button set to this function allows the user to make a call using a number stored by the button. The number can be an internal number, an external number, an account code or any other number. The button can then be used when a number of that type needs to be dialed.	–
Call Coverage 222	A button set to this function allows the user to switch call coverage for their extension on or off. The button settings include the extension number of the extension providing coverage. When on, calls to the user that ring unanswered for the user's number of coverage rings then also start ringing at the call coverage extensions.	–
Caller ID Log 223	A button set to this function allows the user to view the phone system's call log of all caller IDs of calls received by the system. To use the button the user must be one of the three extensions configured for call ID logging.	Yes
Call Forwarding 222	A button set to this function allows the user to redirect all their calls to another number. Extensions with Remote Call Forwarding enabled can also forward calls externally by specifying a personal speed dial as the destination.	–
Call Pickup 223	A button set to this function allows the user to pickup a call alerting at a specified extension. Separate buttons can be created for each extension for which call pickup is required.	–
Caller ID Inspect 223	A button set to this function allows the user to see the caller ID of a call on another line without interrupting the current call to which they are connected.	Yes
Caller ID Name Display 223	A button set to this function allows the user to swap the display of caller ID name and number information on their extension.	Yes
Calling Group 223	A button set to this function allows the user to call or page the calling group represented by the button.	–
Call Screening 224	This function is used to enable or disable call screening. While enabled, when a caller is presented to the user's voicemail mailbox, if the user's phone is idle they will hear through the phone's handsfree speaker the caller leaving the message and can select to answer or ignore the call.	Yes
Conference Drop 226	A button set to this function allows the user to drop a call from a conference.	–
Contact Closure 1 226	A button set to this function allows the user to operate the system's contact closure 1 connection. The user must be a member of the contact closure group.	–
Contact Closure 2 226	A button set to this function allows the user to operate the system's contact closure 2 connection. The user must be a member of the contact closure group.	–
Do Not Disturb 226	A button set to this function allows the user to set the extension's do not disturb on or off.	Yes

Function	Description	LED
Hot Dial 226	A button set to this function allows the user to dial a number without first going off hook or pressing the **SPEAKER** button. Automatic line selection is used to select a line.	Yes
Hunt Group 227	A button set to this function allows the user to call or page the hunt group represented by the button.	–
Idle Line Pickup 227	A button set to this function allows the user to seize a line if that line is idle. This allows the user to access line for which they do not have a line appearance button on their extension.	–
Last Number Redial 227	A button set to this function allows the user to redial the last external number dialed.	–
Loudspeaker Paging 227	A button set to this function allows the user to redial the last external number dialed.	–
Message Alert Notification 227	A button set to this function allows a user to see the current state of other user's message waiting lamps. It can only be used in conjunction with other users for which this user has **Auto Dial - Intercom** buttons configured.	–
Night Service Button 210	A night service button is used to switch night service 210 on/off.	Yes
Pickup Group 228	A button set to this function allows the user to answer a call being presented to any extension that is a member of the pickup group configured for the button.	–
Privacy 228	A button set to this function allows the user to turn privacy on or off. When on, other extensions are not able to bridge into the user's calls.	–
Recall 228	A button set to this function allows the user to send a recall or hook flash signal.	–
Save Number Redial 228	A button set to this function allows the user to save the number dialed during a call and to redial that number when idle. This can be used when the number dialed does not answer.	–
Simultaneous Page 228	A button set to this function allows the user to save the number dialed during a call and to redial that number when idle. This can be used when the number dialed does not answer.	–
Station Lock 229	A button set to this function allows the user to lock their extension from being used to make calls. After they press the button, they are prompted to enter a four digit code after which the extension is locked. If the extension is already locked, pressing the button prompts for reentry of the four digit code to unlock the extension.	–
Station Unlock 229	This function can only be used by the first two extensions in the system. A button set to this function allows the user to unlock any extension without needing to know the code that was used to lock that extension. When the button is pressed, the user is prompted to enter the number of the locked extension.	–
VMS Cover 229	A button set to this function allows the user to switch use of voicemail coverage for their extension on or off.	–
VMS Mailbox Transfer 229	A button set to this function allows the user to transfer their current call to another extension's mailbox. After pressing the button, the current call is put on hold and the user can then enter the target extension number to indicate the mailbox required.	–
Wake Up Service 230	A **Wake Up Service** button can be assigned for the first extension in the system only. Using this button, the extension user can set wake up calls within the next 24-hour period for any other extension.	Yes

- Some functions are unique. That is, if already assigned to a button, assigning the function to another button automatically clears the setting from the existing button.
- Some functions are only supported on buttons that include lights to indicate status. If programmed onto a button without lights, the function may not work.

14.2 Manager Buttons

In some cases, the names used for the programming of button features in the web manager application differ from those used in the phone based administration menus. The table below matches the names used by each interface.

There are also some functions available through web manager that are not accessible through phone based administration.

Phone	Manager	Phone	Manager
Absent Message	Absent Text	**Hunt Group**	Group Hunting - Page
Account Code Entry	Account Code Entry		Group Hunting - Ring
Auto Dial - Other	Auto Dial - Outside	**Pickup Group**	Group Pickup
Auto Dial - Intercom	Auto Dial - ICM	**Hot Dial**	Hot Dial
	Auto Dial - ICM Page	**Last Number Redial**	Last Number Redial
Call Coverage	Call Coverage	**Loudspeaker Page**	Loudspeaker Paging
Caller ID Name Display	Call ID Name - Display	**Night Service**	Night Service
Caller ID Log	Call Log	*Not available*	Outgoing Call Restriction
Call Pickup	Call Pickup	**Privacy**	Privacy
Caller ID Inspect	Caller ID Inspect	**Recall**	Recall
Conference Drop	Conference Drop	**Saved Number Redial**	Save Number Redial
Contact Closure 1	Contact Closure 1	**Simultaneous Page**	Simultaneous Page
Contact Closure 2	Contact Closure 2	**Station Lock**	Station Lock
Active Line Pickup	Direct Line Pickup - Active	**Station Unlock**	Station Unlock
Idle Line Pickup	Direct Line Pickup - Idle	**VMS Cover**	VMS Cover
Do Not Disturb	Do Not Disturb	**Voice Mailbox Transfer**	VMS Transfer
Calling Group	Group Calling - Page	**Hunt Group 777**	Voicemail Collect
	Group Calling - Ring	**Wake Up Service**	Wake Up Service
		Not available	911 View/Emergency View

14.3 911-View/Emergency Call View

A button set to this function indicates when a call has been made using a number in the system's Emergency Numbers 58 list. Pressing the button displays a list of such calls.

- The button flashes when an emergency call is in progress. It is on when there are previous emergency calls in the call log.
- Pressing the button displays details of any emergency call is progress (up to 10). The **History** option displays details of previous emergency calls (up to 30) and allows deletion of previous call details.
- If an extension already has a button set to this function, creating another button with this function will automatically clear the setting from the existing button.
- This function is only supported on a button that includes LED/LCD indicator. The indicator is lit when the function is enabled.
- Not supported on ETR6 and 1403 phones. Not supported on BST phones.

14.4 Absent Message

A button set to this function allows the user to set or clear an absent message for display on their phone. When set, the absent message is also displayed on other extensions when they call the user.

- If an extension already has a button set to this function, creating another button with this function will automatically clear the setting from the existing button.
- This function is only supported on a button that includes LED/LCD indicator. The indicator is lit when the function is enabled.
- If the user has this feature enabled, removing this button will turn the feature off.
- The button can also be used to check the absent message setting of other users. When pressed, pressing the **Auto Dial - Intercom** button of another user will display that users current absent message setting (alternately select **Insp** and dial the user's extension number).
- Not supported on ETR6 and 1403 phones. Not supported on BST phones without a display and soft keys.

14.5 Account Code Entry

A button set to this function allows the user to enter an account code prior to making a call or during a call.

- If an extension already has a button set to this function, creating another button with this function will automatically clear the setting from the existing button.
- This function is only supported on a button that includes LED/LCD indicator. The indicator is lit when the function is enabled.
- Once a user has associated an account code with a call, only that user can change the account code by entering another one.
- To access this function without a programmable button:
  - On DS and ETR phones, press **FEATURE** and then dial **12**.
  - On BST phones, press **FEATURE** and dial **900**.

14.6 Active Line Pickup

A button set to this function allows the user to answer a call on a particular line. It can be used if the call is ringing, held or already answered by another extension.

- If an extension already has a button set to this function, creating another button with this function will automatically clear the setting from the existing button.
- To access this function without a programmable button, press an intercom or call appearance button and then **68** and the line number.

14.7 Auto Dial - Intercom

A button set to this function allows the user to make a call to another specified extension. The button lamp will also indicate when that other extension is in use.

- This type of button can be used for one touch transfer operation.

14.8 Auto Dial - Other

A button set to this function allows the user to make a call using a number stored by the button. The number can be an internal number, an external number, an account code or any other number. The button can then be used when a number of that type needs to be dialed.

14.9 Call Coverage

A button set to this function allows the user to switch call coverage for their extension on or off. The button settings include the extension number of the extension providing coverage. When on, calls to the user that ring unanswered for the user's number of coverage rings then also start ringing at the call coverage extensions.

- When on, a call to the extension that ring unanswered for the extension's Call Coverage Rings setting will also start alerting on the covering extension specified by the button.
- If an extension already has a button set to this function, creating another button with this function will automatically clear the setting from the existing button.
- This function does not require a button that includes lights. However, if set on a button with lights, the green LED will match whether the function is on or off.
- If the user has this feature enabled, removing this button will turn the feature off.
- Programming the destination and/or the originator onto the call coverage button is optional.
- To access this function without a programmable button:
  - On DS and ETR phones, press **FEATURE** and then dial **20**.
  - On BST phones, press **FEATURE** and dial **932**.

14.10 Call Forwarding

A button set to this function allows the user to redirect all their calls to another number. Extensions with Remote Call Forwarding enabled can also forward calls externally by specifying a personal speed dial as the destination.

- If an extension already has a button set to this function, creating another button with this function will automatically clear the setting from the existing button.
- If the user has this feature enabled, removing this button will turn the feature off.
- For analog lines and T1 lines without DID, the extension must be the Line Coverage Extension for that line.
- You can forward outside, intercom, transferred and voice signaled calls.
- You cannot forward group calls, calls to doorphone alert extensions, coverage calls, transfer-return calls and night service calls.
- The system will only forward calls on lines that have reliable disconnect. For these lines, Hold Disconnect Time must be set to a value other than 00 (No Detection).
- The extension must have an available line to forward the call to an outside number.
- The system uses the extension’s automatic line selection setting to determine which line to use for the outgoing call.
- Extension's with Remote Call Forwarding enabled can also forward calls externally by specifying a personal speed dial (80 to 99) as the destination.
- Programming the destination and/or the originator onto the call coverage button is optional.
- Extensions configured as doorphone extension or loudspeaker device will ignore any forwarding set on the extension.
- Do not disturb overrides call forwarding.
- To access this function without a programmable button:
  - On DS and ETR phones, press **FEATURE** and then dial **11**. To switch forwarding off enter the extension's own number as the destination.
  - On BST phones, press **FEATURE** and dial **4**.

14.11 Call Pickup

A button set to this function allows the user to pickup a call alerting at a specified extension. Separate buttons can be created for each extension for which call pickup is required.

- If an extension already has a button set to this function and target, creating another button with this function to the same target will automatically clear the setting from the existing button.
- To access this function without a programmable button, press an intercom or call appearance button and dial **6** followed by the extension number.

14.12 Caller ID Inspect

A button set to this function allows the user to see the caller ID of a call on another line without interrupting the current call to which they are connected.

- If an extension already has a button set to this function, creating another button with this function will automatically clear the setting from the existing button.
- This function is only supported on a button that includes LED/LCD indicator. The indicator is lit when the function is enabled.
- To access this function without a programmable button:
  - On DS and ETR phones, press **FEATURE** and then dial **17**.
  - On BST phones, press **FEATURE** and dial ***0**.

14.13 Caller ID Log

A button set to this function allows the user to view the phone system's call log of all caller IDs of calls received by the system. To use the button the user must be one of the three extensions configured for call ID logging.

- If an extension already has a button set to this function, creating another button with this function will automatically clear the setting from the existing button.
- This function is only supported on a button that includes LED/LCD indicator. The indicator is lit when the function is enabled.
- To access this function without a programmable button, press **FEATURE** and then dial **23**.
- To access this function without a programmable button:
  - On DS and ETR phones, press **FEATURE** and dial **23**.
  - On BST phones, press **FEATURE** and dial **812**.

14.14 Caller ID Name Display

A button set to this function allows the user to swap the display of caller ID name and number information on their extension.

On some phones, after the call is answered the call display is not able to show both the caller ID name and number. This function allows the user on such phones to toggle between the name and the number.

- If an extension already has a button set to this function, creating another button with this function will automatically clear the setting from the existing button.
- This function is only supported on a button that includes LED/LCD indicator. The indicator is lit when the function is enabled.
- If the user has this feature enabled, removing this button will turn the feature off.
- To access this function without a programmable button:
  - On DS and ETR phones, press **FEATURE** and dial **16**.
  - On BST phones, press **FEATURE** and dial **933**.

14.15 Calling Group

A button set to this function allows the user to call or page the calling group represented by the button.

14.16 Call Screening

This function is used to enable or disable call screening. While enabled, when a caller is presented to the user's voicemail mailbox, if the user's phone is idle they will hear through the phone's handsfree speaker the caller leaving the message and can select to answer or ignore the call.

- This feature is supported on ETR6D, ETR18D, ETR34D, 1408, 1416, 9500 Series, M7310, M7310N, M7208, M7208N, M7324, M7324N, T7208, T7316 and T7316E phones.
- Call screening is only applied as follows:
  - It is only applied to calls that have audible alerted at the user's extension before going to voicemail. This requires the user to have both voicemail coverage and call screening enabled and the phone's ringer not set to silent. However it is not applied if the user transfers the call to voicemail.
  - It is only applied if the user's phone is idle, that is not on a call or with a call held pending transfer or conference.
  - Calls that ring the user, are then rerouted (for example follow a forward on busy setting) and then return to the user's mailbox are screened.
- While a call is being screened, the phone can be used to either answer or ignore the screened call. Auto answer options are ignored.
  - **Answering a screened call:**
    While a call is being screened, it can be answered by pressing the **Answer** soft key. On ETR phones, pressing the **MIC/HFAI** button will answer the call. Pressing the call appearance or line button on which the call is indicated will also answer the call.
    - When answered:
      - The phone's microphone is unmuted and a normal call between the user and the caller now exists.
      - The voicemail recording stops but that portion of the call already recorded is left as a new message in the user's mailbox.
  - **Ignoring a screened call:**
    While a call is being screened, it can be ignored by pressing the **Ignore** soft key if displayed. On 1400 and 9500 Series phones, pressing the **SPEAKER** button will ignore the call. On ETR phones, pressing the **SPKR** button will ignore the call. On M-Series and T-Series phones, pressing the **Release** key will ignore the call. On all phones, the **Call Screening** button can be press to both turn off call screening and to ignore the currently screened call.
    - When ignored:
      - The call continues to be recorded until the caller hangs up or transfers out of the mailbox.
      - The user's phone returns to idle with call screening still enabled. However any other call that has already gone to voicemail is not screened.
- While a call is being screened:
  - The mailbox greeting played and the caller can be heard on the phone's speakerphone. The caller cannot hear the user.
  - The user is regarded as being active on a call. They will not be presented with hunt group calls and additional personal calls use abbreviated ringing.
  - For 1400/9500 Series phones, if the phone's default audio path is set to headset or the phone is idle on headset, then the screened call is heard through the headset.
  - Any additional calls that go to the user's mailbox when they are already screening a call, remain at the mailbox and are not screened even if the existing call being screened is ended.
  - Making or answering another call while listening to a screened call is treated as ignoring the screened call.
  - Another user bridging into a screen call answers the call.
  - Phone based administration cannot be accessed and the hold, transfer and conference buttons are ignored.
  - The screened caller using DTMF breakout ends the call screening.
- Enabling do not disturb overrides call screening except for calls from numbers in the user's do not disturb exceptions list.
- Locking the phone overrides call screening.
- Manual call recording cannot be applied to a call being screened.
- While a call is being screened, it uses one of the available voicemail channels. If no voicemail channels are available, call screening does not occur.

- If an extension already has a button set to this function, creating another button with this function will automatically clear the setting from the existing button.
- This function is only supported on a button that includes LED/LCD indicator. The indicator is lit when the function is enabled.
- If the user has this feature enabled, removing this button will turn the feature off.
  - Any call currently being screened will continue being screened but further calls will not receive screening.
- Direct Line Pickup can be used to answer a call that is being screened.
- While listening to call screening, you can press an appearance button to make, answer or join another call. When you do this, the screened call is ignored and the new call is connected. However, on ETR phones the new call is connected as listen-only (microphone off and speaker on). In order to speak on the call the user needs to lift the handset or touch the **Mic/HFAI** button.

14.17 Conference Drop

A button set to this function allows the user to drop a call from a conference.

- If an extension already has a button set to this function, creating another button with this function will automatically clear the setting from the existing button.
- On 1400 Series phones a list of conference parties is displayed from which the user can select which call to drop.
- On ETR phones, the last added external party is dropped.
- To access this function without a programmable button:
  - On DS and ETR phones, press **FEATURE** and dial **06**.
  - On BST phones, press **FEATURE** and dial **934**.

14.18 Contact Closure 1

A button set to this function allows the user to operate the system's contact closure 1 connection. The user must be a member of the contact closure group.

- If an extension already has a button set to this function, creating another button with this function will automatically clear the setting from the existing button.
- To access this function without a programmable button:
  - On DS and ETR phones, press **FEATURE** and dial **41**.
  - On BST phones, press **FEATURE** and dial **9*41**.

14.19 Contact Closure 2

A button set to this function allows the user to operate the system's contact closure 2 connection. The user must be a member of the contact closure group.

- If an extension already has a button set to this function, creating another button with this function will automatically clear the setting from the existing button.
- To access this function without a programmable button:
  - On DS and ETR phones, press **FEATURE** and dial **42**.
  - On BST phones, press **FEATURE** and dial **9*42**.

14.20 Do Not Disturb

Do not disturb prevents the user from receiving calls.

- Only calls from numbers in the user's Do Not Disturb Exceptions list are treated normally.
- The user is not included in hunt group and page calls.
- Calls direct to the user's extension number hear busy tone or are diverted to voicemail if enabled.
- All the users call forwarding, follow me and call coverage settings are ignored.
- This function is only supported on a button that includes LED/LCD indicator. The indicator is lit when the function is enabled.
- If the user has this feature enabled, removing this button will turn the feature off.
- To access this function without a programmable button:
  - On DS and ETR phones, press **FEATURE** and dial **01**.
  - On BST phones, press **FEATURE** and dial **85** (on) or **#85** (off).

14.21 Hot Dial

- This option is not used by DS and BST phones. Those phone have hot dialing always switched on.
- If an extension already has a button set to this function, creating another button with this function will automatically clear the setting from the existing button.
- This function is only supported on a button that includes LED/LCD indicator. The indicator is lit when the function is enabled.

- To access this function without a programmable button, press **FEATURE** and then dial **26**.

14.22 Hunt Group

A button set to this function allows the user to call or page the hunt group represented by the button.

- If an extension already has a button set to this function and target, creating another button with this function to the same target will automatically clear the setting from the existing button.
- To access this function without a programmable button, press an intercom or call appearance button and then dial **77** and the hunt group number (1 to 6). The additional number **777** can be used for access to voicemail to collect messages.
- A page call that is auto-answered by the first available extension in the hunt group can be selected by adding a * in front of the hunt group number.
- This type of button can be used for one touch transfer operation.

14.23 Idle Line Pickup

A button set to this function allows the user to seize a line if that line is idle. This allows the user to access line for which they do not have a line appearance button on their extension.

- If an extension already has a button set to this function, creating another button with this function will automatically clear the setting from the existing button.
- To access this function without a programmable button, press a call appearance button and then dial **8** followed by the two digit line number.

14.24 Last Number Redial

A button set to this function allows the user to redial the last external number dialed.

- If an extension already has a button set to this function, creating another button with this function will automatically clear the setting from the existing button.

14.25 Loudspeaker Page

A button set to this function allows the user to make a page call to the extension configured as being connected to the loudspeaker equipment.

- If an extension already has a button set to this function, creating another button with this function will automatically clear the setting from the existing button.
- To access this function without a programmable button, press an intercom or call appearance button and dial **70**.

14.26 Message Alert Notification

A button set to this function allows a user to see the current state of other user's message waiting lamps. It can only be used in conjunction with other users for which this user has **Auto Dial - Intercom** buttons configured.

- If an extension already has a button set to this function, creating another button with this function will automatically clear the setting from the existing button.

14.27 Night Service

The **Night Service** button feature is used on a button on the first extension on the system to turn night service on and off.

- You can only program a **Night Service** button on the first extension port on the system.
- This function is only supported on a button that includes LED/LCD indicator. The indicator is lit when the function is enabled.
- If the user has this feature enabled, removing this button will turn the feature off.
- Dialing restrictions for extensions not in the Night Service Group remain the same as during normal daytime operation.
- If you reassign the Night Service Button, it is removed from the button where it was previously assigned.
- Night service uses the System Password, if set, as follows:

- You must enter the password when turning Night Service on or off.
- Calls to numbers other than marked system dials or in the Emergency Numbers 58 list requires entry of the system password.

- If you have a voice messaging system, VMS Hunt Schedule determines when outside calls should ring voicemail. The status of the Night Service Button tells the voice messaging system to operate in day or night mode.
- The Night Service Button returns to the status (on/off) it was in immediately after a power failure or a system reset.
- Night Service is unavailable on T1 lines with Direct Inward Dialing (DID)..

14.28 Pickup Group

A button set to this function allows the user to answer a call being presented to any extension that is a member of the pickup group configured for the button.

- If an extension already has a button set to this function and target, creating another button with this function to the same target will automatically clear the setting from the existing button.
- To access this function without a programmable button, press an intercom or call appearance button and dial **661** to **664** for the group (1 to 4) from which to pickup the call.
- When there are multiple calls ringing the members of a pickup group, the longest ringing call is picked up.

14.29 Privacy

A button set to this function allows the user to turn privacy on or off. When on, other extensions are not able to bridge into the user's calls.

- If an extension already has a button set to this function, creating another button with this function will automatically clear the setting from the existing button.
- This function does not require a button that includes lights. However, if set on a button with lights, the green LED will match whether the function is on or off.
- If the user has this feature enabled, removing this button will turn the feature off.
- To access this function without a programmable button:
  - On DS and ETR phones, press **FEATURE** and then dial **07**.
  - On BST phones, press **FEATURE** and dial **83**.

14.30 Recall

A button set to this function allows the user to send a recall or hook flash signal.

- If an extension already has a button set to this function, creating another button with this function will automatically clear the setting from the existing button.

14.31 Saved Number Redial

A button set to this function allows the user to save the number dialed during a call and to redial that number when idle. This can be used when the number dialed does not answer.

- If an extension already has a button set to this function, creating another button with this function will automatically clear the setting from the existing button.
- For analog telephone devices, when using last number redial, saved number redial, system speed dial and personal speed dial features no dial tones for the digits dialed are played to the caller.

14.32 Simultaneous Page

A button set to this function allows the user to make a page call to both the loudspeaker extension and the extensions in first calling group, 71.

- If an extension already has a button set to this function, creating another button with this function will automatically clear the setting from the existing button.
- To access this function without a programmable button, press an intercom or call appearance button and dial ***70**.

- This type of button can be used for one touch transfer operation.

14.33 Station Lock

A button set to this function allows the user to lock their extension from being used to make calls. After they press the button, they are prompted to enter a four digit code after which the extension is locked. If the extension is already locked, pressing the button prompts for reentry of the four digit code to unlock the extension.

- Any locked extension can be unlocked from either of the first two extensions in the system without needing the four digit locking code using a Station Unlock button.
- If an extension already has a button set to this function, creating another button with this function will automatically clear the setting from the existing button.
- To access this function without a programmable button:
  - On DS and ETR phones, press **FEATURE** and then dial **21**.
  - On BST phones, press **FEATURE** and dial **936**.

14.34 Station Unlock

This function can only be used by the first two extensions in the system. A button set to this function allows the user to unlock any extension without needing to know the code that was used to lock that extension. When the button is pressed, the user is prompted to enter the number of the locked extension.

- If an extension already has a button set to this function, creating another button with this function will automatically clear the setting from the existing button.
- To access this function without a programmable button:
  - On DS and ETR phones, press **FEATURE** and then dial **22**.
  - On BST phones, press **FEATURE** and dial **937**.

14.35 VMS Cover

A button set to this function allows the user to switch use of voicemail coverage for their extension on or off.

When on, calls to the extension are redirected to the extension's mailbox when they ring unanswered for the extension's **VMS Coverage Rings** setting. When off, calls to the extension continue to ring at the extension until answered or the caller hangs up.

If the feature is programmed onto a button with LEDs/LCD, it will indicate when the feature is active.

- If an extension already has a button set to this function, creating another button with this function will automatically clear the setting from the existing button.
- To access this function without a programmable button:
  - On DS and ETR phones, press **FEATURE** and then dial **15**.
  - On BST phones, press **FEATURE** and dial **984**.

14.36 Voice Mailbox Transfer

A button set to this function allows the user to transfer their current call to another extension's mailbox. After pressing the button, the current call is put on hold and the user can then enter the target extension number to indicate the mailbox required.

- If an extension already has a button set to this function, creating another button with this function will automatically clear the setting from the existing button.

14.37 Wake Up Service

A **Wake Up Service** button can be assigned for the first extension in the system only. Using this button, the extension user can set wake up calls within the next 24-hour period for any other extension.

- If an extension already has a button set to this function, creating another button with this function will automatically clear the setting from the existing button.
- This function is only supported on a button that includes LED/LCD indicator. The indicator is lit when the function is enabled.
- Removing the wake up service button from the extension does not remove any existing wake up service alarms that have been set.

How the Wake Up Service Operates

Using the button, the extension can set a wake up call by specifying the target extension and the time.

- When the scheduled time is reach, the system will make an intercom call to the target extension. The call is indicated as a **Wake Up Call** in the display. The wake up call will alert for approximately 30 seconds.
- Wake up calls ignore settings such as Do Not Disturb, forwarding, call coverage and coverage to voicemail.
- If the extension user is on a call:
  - For an analog extension, the wake up call is treated as unanswered.
  - For other extensions, the wake up call will alert with just an abbreviated ring.
- When a user answers a wake up call, they hear music on hold if available, otherwise they will hear repeated double tones.
- Once a wake up call is answered, it is treated as being completed and no further call attempts are made.
- If the wake up call is not answered or the extension is busy, the wake up call is rescheduled for 5 minutes later.
- Only 2 attempts are made to send a wakeup call. If neither is answered the wake up call is cleared.
- If a wake up call is already scheduled for an extension, setting up a new wake up call to that extension will erase the existing wakeup call.
- Wake up calls are shown in the SMDR output with the name "Wake Up Call".

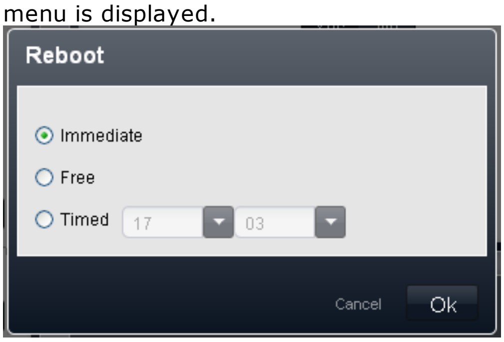
Chapter 15. Maintenance

15. Maintenance

15.1 Rebooting the System

Some changes require the system to be rebooted before the changes come into effect. Rebooting the system will end all calls currently in progress.

1. Click **Reboot**. The command is located in the top right of the Basic Edition Web Manager screen. The reboot menu is displayed.



2. Select the type of reboot required.

   - ***Immediate***
     If this option is selected, the reboot is started once **OK** is clicked. Any existing calls are ended without any warning.

   - ***Free***
     If this option is selected, after **OK** is clicked, the system will wait until there are no calls in progress before starting the reboot process.

   - ***Timed***
     If this option is selected, a time can also be selected for when the reboot should occur.

3. Click **OK**.

15.2 Shutting Down the System

The system should always be shutdown using the following process before it is switched off. This ensure that system actions such a file writes are completed before the power is removed. It also ensure that the current configuration in the system's memory is backed up to its System SD card.

The shut down can be either indefinite or for a set period of time after which the system will automatically reboot.

! WARNINGS

- A shutdown must always be used to switch off the system. Simply removing the power cord or switching off the power input may cause errors.
- This is not a polite shutdown, any calls and services in operation will be stopped. Once shutdown, the system cannot be used to make or receive calls until restarted.
- The shutdown process takes up to a minute to complete. When shutdown, the CPU LED and the IP500 base card LEDs 1 and 9 (if trunk daughter card fitted) will flash red rapidly. The memory card LEDs on the rear of the control unit are extinguished. Do not remove power from the system or remove any of the memory cards until the system is in the this state.
- To restart a system when shutdown indefinitely, or to restart a system before the timed restart, switch power to the system off and on again.

Shutting Down the System using Basic Edition Web Manager

This method performs a indefinite system shutdown. To restart the system power will need to be removed and then reapplied.

1. Click **Monitoring** in the menu bar and select **System Shutdown**.

Shutting Down the System using IP Office System Status

1. Click **Monitoring** in the menu bar and then click on **System Status**.
   - Alternatively, open a new browser or browser window and enter the address ***https://<IP Address>:8433/ssa/index.html*** using the same *<IP Address>* as used for Basic Edition Web Manager of the system. This option provides IP Office System Status a larger window for displaying information.
2. In the IP Office System Status **Logon** window, enter the same IP address, user name and password as you have used for Basic Edition Web Manager access to the system.
3. Click **Logon**.
4. In the navigation tree on the left, click on **System**.
5. At the bottom of the screen, click **Shutdown System**.
6. Select how long you want the system shutdown. If you want to shutdown and then switch off the system's power, select **Indefinite**.
7. Click **OK**.
8. You will be asked to confirm the action. Click **Yes**.
9. The system will start its shutdown process. This will disconnect the IP Office System Status application and Basic Edition Web Manager.
10. When the shutdown is complete, LEDs 1 and 9 on each card in the front of the control unit will flash red. If shutdown indefinitely, power to the system can now be safely switched off.

System Shutdown Using the Control Unit AUX Button

When the **AUX** button is pressed for between more than 5 seconds, the IP500 V2 control unit will shutdown with the restart timer set to 10 minutes.

15.3 Backing Up System Files

The system's System SD card hold two copies of most files. Those loaded by the system when it starts are stored in a ***/System/Primary*** folder. An additional set of files are stored in a ***/System/Backup*** folder on the same card.

Using IP Office System Status, you can have the system copy the contents of the **/Primary** folder to the **/Backup** folder. This may be useful if you anticipate making a number of configuration changes but want to keep a backup copy of the existing configuration and other files.

This process does not backup the prompts, messages and greetings used by the system's voicemail mailboxes and auto attendants. They are stored in the card's ***/lvmail*** and ***/dynamic/lvmail*** folders.

The backup files can be restored 234 using IP Office System Status.

Backup System Files using IP Office System Status

1. Click **Monitoring** in the menu bar and then click on **System Status**.
    - Alternatively, open a new browser or browser window and enter the address ***https://<IP Address>:8433/ssa/index.html*** using the same *<IP Address>* as used for Basic Edition Web Manager of the system. This option provides IP Office System Status a larger window for displaying information.
2. In the IP Office System Status **Logon** window, enter the same IP address, user name and password as you have used for Basic Edition Web Manager access to the system.
3. Click **Logon**.
4. In the navigation tree on the left, click on **System**.
5. At the bottom of the screen, click **Backup System Files**.
6. You will be asked to confirm the action. Click **Yes**.
7. While the backup is in progress, the **Backup System Files** and **Restore System Files** buttons are greyed out. Do not perform any other system actions until the backup is completed.

15.4 Restoring System Files

The system's System SD card hold two copies of most files. Those loaded by the system when it starts are stored in a ***/System/Primary*** folder. An additional set of files are stored in a ***/System/Backup*** folder on the same card.

Using IP Office System Status, you can have the system copy the contents of the **/Backup** folder to the **/Primary** folder. In order for the system to then use the restored files, it needs to be shutdown, switched off and then switched on again.

- **! WARNING**
  This process will overwrite the current files that the systems loaded when it started. They cannot be restored unless you have another copy of the files.

Restoring System Files using IP Office System Status

1. Click **Monitoring** in the menu bar and then click on **System Status**.
    - Alternatively, open a new browser or browser window and enter the address ***https://<IP Address>:8433/ssa/index.html*** using the same *<IP Address>* as used for Basic Edition Web Manager of the system. This option provides IP Office System Status a larger window for displaying information.
2. In the IP Office System Status **Logon** window, enter the same IP address, user name and password as you have used for Basic Edition Web Manager access to the system.
3. Click **Logon**.
4. In the navigation tree on the left, click on **System**.
5. At the bottom of the screen, click **Restore System Files**.
6. You will be asked to confirm the action. Click **Yes**.
7. While the restore is in progress, the **Backup System Files** and **Restore System Files** buttons are greyed out. Do not perform any other system actions until the restore is completed.
8. Once the file restore is complete, use **Shutdown System** to shutdown the system indefinitely.
9. Switch power to the system off and then on again.

15.5 Shutting Down a Memory Card

Rather than shutting down the whole system, it is possible to just shut down the system's memory card. Once shutdown the card can be removed from the system to perform action such as load additional files onto the card or copy files from the card.

Shutting down the System SD card will halt voicemail services including user mailboxes and auto attendants. In addition, because the System SD card is used to validate licenses, licensed features will only operate for a further 2 hours before also halting.

A memory card that has been shutdown can be restarted using IP Office System Status or Basic Edition Web Manager. If the card has been removed from the system's control unit, it is automatically restarted when it is reinserted into the control unit.

Memory Card Shutdown Up using Basic Edition Web Manager

1. Click **Monitoring** in the menu bar and select **Memory Card Stop**.

Memory Card Shutdown using IP Office System Status

1. Click **Monitoring** in the menu bar and then click on **System Status**.
    - Alternatively, open a new browser or browser window and enter the address ***https://<IP Address>:8433/ssa/index.html*** using the same *<IP Address>* as used for Basic Edition Web Manager of the system. This option provides IP Office System Status a larger window for displaying information.
2. In the IP Office System Status **Logon** window, enter the same IP address, user name and password as you have used for Basic Edition Web Manager access to the system.
3. Click **Logon**.
4. In the navigation tree on the left, click on the **+** symbol next to **System**.
5. Click on the **+** symbol next to **Memory Cards**.
6. Click on the memory card, **System SD** or **Optional SD**, that you want to shutdown.
7. At the bottom of the screen, click **Shutdown**.
8. You will be asked to confirm the action. Click **Yes**.
9. Do not remove the memory card from the system until the adjacent **System SD** or **Optional SD** LED is extinguished.

15.6 Starting a Memory Card

If a memory card has been stopped/shutdown, it needs to be restarted for the system to recognize and use the card.

If the card has been removed from the system's control unit, it is automatically restarted when it is reinserted into the control unit. This process can be used if the shutdown card has remained in the control unit after being stopped.

Memory Card Start Up using Basic Edition Web Manager

1. Click **Monitoring** in the menu bar and select **Memory Card Start**.

Memory Card Start Up using IP Office System Status

1. Click **Monitoring** in the menu bar and then click on **System Status**.
    - Alternatively, open a new browser or browser window and enter the address ***https://<IP Address>:8433/ssa/index.html*** using the same *<IP Address>* as used for Basic Edition Web Manager of the system. This option provides IP Office System Status a larger window for displaying information.
2. In the IP Office System Status **Logon** window, enter the same IP address, user name and password as you have used for Basic Edition Web Manager access to the system.
3. Click **Logon**.
4. In the navigation tree on the left, click on the **+** symbol next to **System**.
5. Click on the **+** symbol next to **Memory Cards**.
6. Click on the memory card, **System SD** or **Optional SD**, that you want to start.
7. At the bottom of the screen, click **Start Up**.

15.7 Copying the System Card

If memory cards are present in both the **System SD** and **Optional SD** card slots , you can copy **all** the contents of the **System SD** card to the **Optional SD** card. Depending on the number of files, this process can take up to 30 minutes to complete.

New files added after the process is started or file changes made after the process is started may not be included. Therefore it is recommended that this command is used during an idle period if available, for example outside normal business hours.

Copying the System SD Card using Basic Edition Web Manager

1. Click **Monitoring** in the menu bar and select **Memory Card Copy**.

Copying the System SD Card using IP Office System Status

1. Click **Monitoring** in the menu bar and then click on **System Status**.
   - Alternatively, open a new browser or browser window and enter the address ***https://<IP Address>:8433/ssa/index.html*** using the same *<IP Address>* as used for Basic Edition Web Manager of the system. This option provides IP Office System Status a larger window for displaying information.
2. In the IP Office System Status **Logon** window, enter the same IP address, user name and password as you have used for Basic Edition Web Manager access to the system.
3. Click **Logon**.
4. In the navigation tree on the left, click on the **+** symbol next to **System**.
5. Click on the **+** symbol next to **Memory Cards**.
6. Click on **System SD**.
7. At the bottom of the screen, click **Copy System Card**.
8. You will be asked to confirm the action. Click **Yes**.
9. While the backup is in progress, the **Shutdown**, **Start Up** and **Copy System Card** buttons are greyed out. Do not perform any other system actions until the copying is completed.

Chapter 16.
Other System Administration Tools

16. Other System Administration Tools

These table list the functions and configuration settings accessible from the system administrator tools. It does not cover administration of their own settings by users which can also be done using phone based administration or Basic Edition Web Manager. Note that the names of some features do vary depending on which tool is being used for administration.

System Maintenance Functions

	Function	IP Office Manager	Basic Edition Web Manager	Phone Based Admin	IP Office System Status
General	**System Discovery**	Yes	–	–	–
	Add/Display VM Locales	Yes	–	–	–
	Launch System Status	Yes	Yes	–	–
	Onboarding (Global Registration)	–	Yes	–	–
	View TFTP Log	Yes	–	–	–
	Turn SSL VPN Service On/Off	No	No	Yes	–
Offline Configuration	**Create Configuration File**	Yes	–	–	–
	Save Configuration as File	Yes	–	–	–
	Load Configuration from File	Yes	Yes	–	–
Default	**Erase Configuration (Default)**	Yes	Yes	Yes	–
	Default Security Settings	–	Yes	–	–
Reboot	**Reboot Warning**	Yes	–	Yes	–
	Reboot - Immediate	Yes	Yes	Yes	–
	Reboot - When Free	Yes	Yes	–	–
	Reboot - At Set Time	Yes	Yes	–	–
System Shutdown	**System Shutdown - Indefinite**	Yes	Yes	Yes	Yes
	System Shutdown - Timed	Yes	–	–	Yes
System Upgrade	**Remote Software Upgrade**	Yes	Yes	–	–
	System Upgrade from Optional SD Card	–	–	Yes	–
	Switch to Standard Mode	Yes	–	–	–
SD Card Management	**Format SD Card**	Yes	–	–	Yes
	Recreate SD Card	Yes	–	–	–
	Shutdown Memory Card	Yes	Yes	Yes	Yes
	Startup Memory Card	Yes	Yes	Yes	Yes
	Embedded File Management	Yes	–	–	–
Administrator	**Set Administrator Password**	Yes	Yes	Yes	–
	Create Additional Administrators	–	Yes	–	–
	Enable User Admin for User	–	Yes	–	–
Backup/Restore	**Clear Backup Alarm**	–	–	Yes	–
	Backup to System SD	–	Yes	Yes	Yes
	Backup to PC	–	Yes	–	–
	Restore from System SD	–	Yes	Yes	Yes
	Restore from PC	–	Yes	–	–
	System Copy to Optional SD Card	–	Yes	Yes	Yes
Extension Settings	**2 Digit/3 Digit Numbering**	Yes	–	Yes	–
	Renumber Extension	Yes	–	–	–
	Copy Extension Settings	–	–	Yes	–
Display System Details	**Software Level**	Yes	Yes	Yes	–
	IP Address	Yes	Yes	Yes	–
	Feature Key Number	Yes	Yes	Yes	–

	Function	IP Office Manager	Basic Edition Web Manager	Phone Based Admin	IP Office System Status
	Installed Hardware	Yes	Yes	–	–
Trunk Templates	**Analog Trunk Templates**	Yes	–	–	–
	SIP Trunk Templates	Yes	–	–	–

System Settings

	Function	IP Office Manager	Basic Edition Web Manager	Phone Based Admin
System Parameters	**System Name**	Yes	Yes	–
	System Mode	Yes	Yes	Yes
	Voicemail Mode	Yes	Yes	–
	Country	Yes	Yes	Yes
	Receive IP Address Via DHCP Server	Yes	Yes	–
	IP Address	Yes	Yes	–
	Sub-Net Mask	Yes	Yes	–
	Default Gateway	Yes	Yes	–
	DNS Server IP Address	–	Yes	–
	Backup DNS Server IP Address	–	Yes	–
	Automatic Daylight Saving Time	Yes	Yes	Yes
	Set System Date	–	–	Yes
	Set System Time	–	–	Yes
	Language	Yes	Yes	Yes
	Number of Lines	Yes	Yes	Yes
	Outside Line Prefix	Yes	Yes	Yes
	System Password	Yes	Yes	Yes
	Log Caller ID Extensions	Yes		Yes
	Unsupervised Analog Trunk Disconnect Handling	Yes	Yes	Yes
System Speed Dials	**System Speed Dials**	Yes	Yes	Yes
	Import	Yes	–	–
	Export	Yes	–	–
Licenses	**Licenses**	Yes	Yes	–
	Import	Yes	–	–
	Export	Yes	–	–
Advanced	**Enable Network Time Synchronization**	Yes	Yes	Yes
	Hold Reminder Time	Yes	Yes	Yes
	Transfer Return Ring	Yes	Yes	Yes
	Outside Conference Denial	Yes	Yes	Yes
	Default Name Priority	Yes	–	–
	Ring on Transfer	Yes	Yes	Yes
	Recall Timer Duration	Yes	Yes	Yes
	Toll Call Prefix	Yes	Yes	Yes
	Companding Law	Yes	Yes	–
STUN Settings for Network	**Enable STUN**	Yes	Yes	–
	STUN Server IP Address	Yes	Yes	–
	STUN Port	Yes	Yes	–
	Firewall/NAT Type	Yes	Yes	–
	Binding Refresh Time (seconds)	Yes	Yes	–

	Function	IP Office Manager	Basic Edition Web Manager	Phone Based Admin
	Public IP Address	Yes	Yes	–
	Public Port	Yes	Yes	–
	Run STUN	Yes	Yes	–
SMTP Server Configuration	**IP Address**	Yes	Yes	–
	Port	Yes	Yes	–
	Email From Address	Yes	Yes	–
	Server Requires Authentication	Yes	Yes	–
	User Name	Yes	Yes	–
	Password	Yes	Yes	–
	Use Challenge Response Authentication	Yes	Yes	–
Busy Tone Detection	**Mode**	Yes	Yes	–
	Single Frequency	Yes	Yes	–
	Dual Frequency	Yes	Yes	–
	On Width	Yes	Yes	–
	Off Width	Yes	Yes	–

User Settings

	Function	IP Office Manager	Basic Edition Web Manager	Phone Based Admin
User Settings	**Name**	Yes	Yes	Yes
	Language	Yes	Yes	Yes
	Exclude from Directory\List in Directory	Yes	Yes	–
	User CLI	Yes	Yes	–
	Outgoing Call Bar	Yes	Yes	Yes
	Call Forwarding	Yes	Yes	Yes
	List Membership	Yes	Yes	Yes
	Group Membership	Yes	Yes	Yes
	Display Extension Port Location	Yes	Yes	–
User Advanced Parameters	**Ring Pattern**	Yes	Yes	Yes
	Abbreviated Ringing	Yes	Yes	Yes
	Call Coverage Ring	Yes	Yes	Yes
	Call Waiting Extension	Yes	Yes	Yes
	Automatic VMS Cover	Yes	Yes	Yes
	Transfer Return Extension	Yes	Yes	Yes
	VMS Cover Rings	Yes	Yes	Yes
	Intercom Dial Tone	Yes	Yes	Yes
	Distinctive Ringing	Yes	Yes	Yes
	Hotline Alert Number	Yes	Yes	Yes
	Privacy Enabled	Yes	Yes	Yes
	Override Line Ringing	Yes	Yes	Yes
User Voicemail Settings	**Clear Voicemail Code**	Yes	Yes	Yes
	Set Voicemail Code	Yes	Yes	–
	Voicemail Email Address	Yes	Yes	–
	DTMF Breakout	Yes	Yes	–
	Voicemail Email Mode	Yes	Yes	–
Equipment Type	**Loudspeaker Paging**	Yes	Yes	Yes
	Door Phone 1 / Door Phone 2	Yes	Yes	Yes
	Fax Machine	Yes	Yes	Yes
	Standard	Yes	Yes	Yes
User Restrictions	**Forced Account Code Entry**	Yes	Yes	Yes
	Outgoing Call Restriction	Yes	Yes	Yes
DND Exceptions List	**Turn Do Not Disturb On/Off**	Yes	Yes	–
	Do Not Disturb Exception List	Yes	Yes	Yes
Speed Dials	**Personal Speed Dials**	–	–	Yes

Button Programming

	Function	IP Office Manager	Basic Edition Web Manager	Phone Based Admin
Button Programming	**Button Programming**	Yes	Yes	Yes
	Copy Feature Buttons	Yes	Yes	–
	Print Labels	Yes	–	–
	Label Button	Yes	Yes	–
	Modify ALS Programming	Yes	Yes	–
	Clear ALS	Yes	Yes	–
	Print Label for this Extension	Yes	–	–
Button Features	**Absent Text**	Yes	Yes	Yes
	Account Code Entry	Yes	Yes	Yes
	Auto Dial - Outside	Yes	Yes	Yes
	Auto Dial - ICM	Yes	Yes	Yes
	Auto Dial - ICM Page	Yes	Yes	Yes
	Call Coverage	Yes	Yes	Yes
	Call Forwarding	Yes	Yes	Yes
	Call Log	Yes	Yes	Yes
	Call Pickup	Yes	Yes	Yes
	Caller ID Inspect	Yes	Yes	Yes
	Call ID Name - Display	Yes	Yes	Yes
	Call Screening	Yes	Yes	Yes
	Conference Drop	Yes	Yes	Yes
	Contact Closure 1/Contact Closure 2	Yes	Yes	Yes
	Direct Line Pickup - Active	Yes	Yes	Yes
	Direct Line Pickup - Idle	Yes	Yes	Yes
	Do Not Disturb	Yes	Yes	Yes
	Group Calling - Page	Yes	Yes	Yes
	Group Calling - Ring	Yes	Yes	Yes
	Group Hunting - Page	Yes	Yes	Yes
	Group Hunting - Ring	Yes	Yes	Yes
	Group Pickup	Yes	Yes	Yes
	Hot Dial	Yes	Yes	Yes
	Last Number Redial	Yes	Yes	Yes
	Lines	Yes	Yes	Yes
	" Ringing Options	Yes	Yes	Yes
	Loudspeaker Paging	Yes	Yes	Yes
	Message Alert Notification	Yes	Yes	Yes
	Night Service	Yes	Yes	Yes
	Privacy	Yes	Yes	Yes
	Recall	Yes	Yes	Yes
	Save Number Redial	Yes	Yes	Yes
	Simultaneous Page	Yes	Yes	Yes
	Station Lock	Yes	Yes	Yes
	Station Unlock	Yes	Yes	Yes
	VMS Cover	Yes	Yes	Yes
	VMS Transfer	Yes	Yes	Yes
	Voicemail Collect	Yes	Yes	–
	Wake Up Service	Yes	Yes	Yes

List and Group Management

	Function	IP Office Manager	Basic Edition Web Manager	Phone Based Admin
Lists	**Allowed Lists**	Yes	Yes	Yes
	Disallowed Lists	Yes	Yes	Yes
	Emergency Number List	Yes	Yes	Yes
	Account Codes	Yes	Yes	Yes
Groups	**Hunt Groups**	Yes	Yes	Yes
	Pickup Groups	Yes	Yes	Yes
	Calling Groups	Yes	Yes	Yes
	Night Service Group	Yes	Yes	Yes
	Operator Group	Yes	Yes	Yes

PBX Outgoing Call Routing

	Function	IP Office Manager	Basic Edition Web Manager	Phone Based Admin
ARS Selectors	**Selector**	Yes	Yes	Yes
	Type	Yes	Yes	Yes
	Details/Lines	Yes	Yes	Yes
Dial Numbers	**Class of Call**	Yes	Yes	Yes
	Number	Yes	Yes	Yes
	Outgoing Lines/ARS	Yes	Yes	Yes

Auxiliary Equipment

	Function	IP Office Manager	Basic Edition Web Manager	Phone Based Admin
Door Phone Extensions 1 and 2	**Assign Extension**	Yes	Yes	Yes
	Extensions to be alerted	Yes	Yes	Yes
Music on Hold	**Status**	Yes	Yes	Yes
SMDR	**SMDR output**	Yes	Yes	–
	IP Address	Yes	Yes	–
	TCP Port	Yes	Yes	–
	Record to Buffer	Yes	Yes	–
	Call Splitting for Diverts	Yes	Yes	–
Contact Closure 1 and 2	**Contact Closure Type**	Yes	Yes	Yes
	Extensions to be enabled	Yes	Yes	Yes

Auto Attendant

	Function	IP Office Manager	Basic Edition Web Manager	Phone Based Admin
Greeting Times	**Morning**	Yes	Yes	–
	Afternoon	Yes	Yes	–
	Evening	Yes	Yes	–
	Out of Hours	Yes	Yes	–
Profiles	**Maximum Inactivity**	Yes	Yes	–
	Menu Prompt	Yes	Yes	–
	Direct Dial By Number	Yes	Yes	–
	Follow Night Service	Yes	Yes	–
	Dial by Name Match Order	Yes	Yes	–
	Language	Yes	Yes	–
	Out of Hours	Yes	Yes	–
	Weekly Off	–	Yes	–
	Emergency Greeting	Yes	Yes	–
	Alarm Extension	Yes	Yes	–
Actions	**Morning**	Yes	Yes	–
	Afternoon	Yes	Yes	–
	Evening	Yes	Yes	–
	Out of Hours	Yes	Yes	–
	Key	Yes	Yes	–
	Action	Yes	Yes	–
	Destination	Yes	Yes	–

SIP Trunk Settings

	Function	IP Office Manager	Basic Edition Web Manager	Phone Based Admin
SIP Trunk List	**Name**	Yes	Yes	–
	Domain Name	Yes	Yes	–
	No of Channels	Yes	Yes	–
	Authentication Name	Yes	Yes	–
	Password	Yes	Yes	–
	Transport Protocol	Yes	Yes	–
	Send Port	Yes	Yes	–
	Listen Port	Yes	Yes	–
Trunk Parameters	**Proxy Server Address**	Yes	Yes	–
	DNS Server Address	Yes	Yes	–
	Mobility Caller ID Format	Yes	Yes	–
	Use Tel URI	Yes	Yes	–
	Check OOS	Yes	Yes	–
	Call Routing Method	Yes	Yes	–
	Association Method	Yes	Yes	–
	Call Route Via Register	Yes	Yes	–
	Name Priority	Yes	Yes	–
	User-Agent and Server Headers	Yes	Yes	–
	UPDATE Supported	Yes	Yes	–
	Separate Register	Yes	Yes	–
VoIP Parameters	**Compression Mode**	Yes	Yes	–
	VoIP Silence Suppression	Yes	Yes	–
	Call Initiation Timeout	Yes	Yes	–
	Re-Invite Support	Yes	Yes	–
	DTMF Support	Yes	Yes	–
	Use Offered Codec	Yes	Yes	–
	Registration Expiry	Yes	Yes	–
	PRACK/100rel Supported	Yes	Yes	–
	Fax Transport Support	Yes	Yes	–
	Caller ID from From Header	Yes	Yes	
	Send From In Clear	Yes	Yes	
Refer Support	**Refer Support**	Yes	Yes	–
	Incoming	Yes	Yes	–
	Outgoing	Yes	Yes	–
Channel List	**Appearance ID**	Yes	Yes	–
	Display Name	Yes	Yes	–
	Authentication Name	Yes	Yes	–
	Password	Yes	Yes	–
Channel Setup	**Direction**	Yes	Yes	–
	Local URI	Yes	Yes	–
	Anonymous	Yes	Yes	–
	Registration Required	Yes	Yes	–
	P-Asserted-ID	Yes	Yes	–
	Coverage Destination	Yes	Yes	–
	Unique Line Ringing	Yes	Yes	–
	VMS Delay - Day	Yes	Yes	–

	Function	IP Office Manager	Basic Edition Web Manager	Phone Based Admin
	VMS Delay - Night	Yes	Yes	–
	VMS Schedule	Yes	Yes	–
	VMS Auto Attendant	Yes	Yes	–
Call by Call List	**Local URI**	Yes	Yes	–
	Destination	Yes	Yes	–
	Authentication Name	Yes	Yes	–
	Password	Yes	Yes	–
	Display Name	Yes	Yes	–
	P-Assert-ID	Yes	Yes	–
	Registration Required	Yes	Yes	–
Dial Plan	**Number**	Yes	Yes	–
	Result	Yes	Yes	–
	Action	Yes	Yes	–
Incoming Number Filter	**Incoming Number**	Yes	Yes	–
	Result	Yes	Yes	–
	Include in Dial Plan	Yes	Yes	–

DID Mapping Table

	Function	IP Office Manager	Basic Edition Web Manager	Phone Based Admin
DDI	**DID Number**	Yes	Yes	–
	Incoming CLI	Yes	Yes	–
	Destination	Yes	Yes	–

Analog Trunk

	Function	IP Office Manager	Basic Edition Web Manager	Phone Based Admin
Analog Trunk Setup	**Appearance ID**	Yes	Yes	–
	Hold Disconnect Time	Yes	Yes	Yes
	Coverage Destination	Yes	Yes	–
	Ring Pattern	Yes	Yes	–
VMS Settings	**Delay - Day**	Yes	Yes	–
	Delay - Night	Yes	Yes	–
	VMS Schedule	Yes	Yes	–
	Auto Attendant	Yes	Yes	–
Detailed Trunk Parameters	**Ring Persistency**	Yes	Yes	–
	Ring Off Maximum	Yes	Yes	–
	Await Dial Tone	Yes	Yes	–
	Intermediate Digit Pause	Yes	Yes	–
	Long CLI Line	Yes	Yes	–
	Trunk Type	Yes	Yes	–
	Modem Enabled	Yes	Yes	–
Advanced Settings	**Mains Hum Filter**	Yes	Yes	–
	Echo Cancellation	Yes	Yes	–
Gains	**Gains A > D**	Yes	Yes	–
	Gains D > A	Yes	Yes	–
DTMF	**DTMF - Mark**	Yes	Yes	–
	DTMF - Space	Yes	Yes	–
Impedance Match	**Impedance**	Yes	Yes	–
	Digits to break dial tone	Yes	Yes	–
	Automatic Balance Impedance Match	Yes	Yes	–
	Quiet Line	Yes	Yes	–

BRI Trunk

	Function	IP Office Manager	Basic Edition Web Manager	Phone Based Admin
Trunk	**TEI**	Yes	Yes	–
Channel Setup	**Appearance ID**	Yes	Yes	–
	Local Number	Yes	Yes	–
	Anonymous	Yes	Yes	–
	Coverage Destination	Yes	Yes	–
	Ring Pattern	Yes	Yes	–

PRI Trunk Settings

	Function	IP Office Manager	Basic Edition Web Manager	Phone Based Admin
Trunk Parameters	**Switch Type**	Yes	Yes	–
	Provider	Yes	Yes	–
	Test Number	Yes	Yes	–
	Clock Quality	Yes	Yes	–
	Framing	Yes	Yes	–
	CRC Checking	Yes	Yes	–
	Zero Suppression	Yes	Yes	–
	Send Redirecting Number	Yes	Yes	–
	CSU Operation	Yes	Yes	–
	Line Signalling	Yes	Yes	–
	Haul Length	Yes	Yes	–
	Channel Unit	Yes	Yes	–
Dial Plan	**Number**	Yes	Yes	–
	Result	Yes	Yes	–
	Action	Yes	Yes	–
PRI Channels	**Appearance ID**	Yes	Yes	–
	Admin	Yes	Yes	–
	Local Number	Yes	Yes	–
	Anonymous	Yes	Yes	–
	Coverage Destination	Yes	Yes	–
	Ring Pattern	Yes	Yes	–
VMS Settings	**VMS Delay - Day**	Yes	Yes	–
	VMS Delay Night	Yes	Yes	–
	VMS Schedule	Yes	Yes	–
	VMS Auto Attendant	Yes	Yes	–
Service Settings	**Service**	–	Yes	–
Gains	**Rx Gain**	Yes	Yes	–
	Tx Gain	Yes	Yes	–

AT&T Specific Setup

	Function	IP Office Manager	Basic Edition Web Manager	Phone Based Admin
TNS Code	**TNS Codes**	Yes	Yes	–
Special	**Short Code**	Yes	Yes	–
	Number	Yes	Yes	–
	Special	Yes	Yes	–
	Plan	Yes	Yes	–
Call by Call	**Short Code**	Yes	Yes	–
	Number	Yes	Yes	–
	Service	Yes	Yes	–

ETSI PRI Trunk Settings

	Function	IP Office Manager	Basic Edition Web Manager	Phone Based Admin
Trunk	**Trunk Subtype**	Yes	Yes	–
	Number of Channels	Yes	Yes	–
	CRC	Yes	Yes	–
Channel Settings	**Appearance ID**	Yes	Yes	–
	Local Number	Yes	–	–
	Anonymous	Yes	Yes	–
	Coverage Destination	Yes	Yes	–
	Ring Pattern	Yes	Yes	–

T1 Trunk Settings

	Function	IP Office Manager	Basic Edition Web Manager	Phone Based Admin
Trunk Parameters	**Clock Quality**	Yes	Yes	–
	Framing	Yes	Yes	–
	CRC Checking	Yes	Yes	–
	Zero Suppression	Yes	Yes	–
	CSU Operation	Yes	Yes	–
	Line Signalling	Yes	Yes	–
	Haul Length	Yes	Yes	–
	Channel Unit	Yes	Yes	–
Channel Settings	**Appearance ID**	Yes	Yes	–
	In Service	Yes	Yes	–
	Coverage Destination	Yes	Yes	–
	Unique Line Ringing	Yes	Yes	–
VMS Settings	**VMS Delay - Day**	Yes	Yes	–
	VMS Delay - Night	Yes	Yes	–
	VMS Schedule	Yes	Yes	–
	VMS Auto Attendant	Yes	Yes	–
Type Settings	**Type**	Yes	Yes	–
	Incoming Trunk Type	Yes	Yes	–
	Outgoing Trunk Type	Yes	Yes	–
Gains	**Rx Gain**	Yes	Yes	–
	Tx Gain	Yes	Yes	–
Timers	**Timers**	Yes	Yes	–

Chapter 17.
SMDR Call Logging

17. SMDR Call Logging

The control unit is able to send SMDR (Station Message Detail Reporting) records to an IP address and port. Each SMDR record contains call information in a comma-separated format (CSV) format, that is variable-width fields with each field separated by commas. The 3rd party call logging software running at the specified address collects the records and uses the data to provide call information.

Normally an SMDR record is output for each call between two parties when the call has been completed. In some scenarios, for example transfers and conferences, separate SMDR records may be output for each part of the call. See SMDR Examples 258.

Configuring the SMDR Destination

1. Click **System** in the menu bar and then click **Auxiliary Equipment**.
2. The **SMDR Setup** panel is used to set the destination for any SMDR records output by the system.
   - **SMDR output:** *Default = Off*
     This control can be used to switch the output of SMDR on or off.
   - **IP Address:** *Default = 0.0.0.0 (Listen).*
     The destination IP address for SMDR records.
   - **TCP Port:** *Default = 0.*
     The destination IP port for SMDR records.
   - **Record to Buffer:** *Default = 500. Range = 10 to 3000.*
     The phone system can buffer up to 3000 SMDR records if it detects a communications failure with destination address. When the buffer is full, each new record overwrites the oldest record.
   - **Call Splitting for Diverts:** *Default = Off.*
     When enabled, for calls forwarded off-switch using an external trunk, the SMDR produces separate initial call and forwarded call records. This applies for calls forwarded by forward unconditional, forward on no answer or forward on busy. The two sets of records will have the same Call ID. The call time fields of the forward call record are reset from the moment of forwarding on the external trunk.
3. Click **Save**.

SMDR Records

- An SMDR record is generated for each call between two devices on the IP Office system. Devices include extensions, trunk lines (or channels on a trunk), voicemail channels, conference channels and IP Office tones.
- Calls that are not presented to another device do not generate an SMDR record. For example internal users dialing short code that simply changes a configuration setting.
- The SMDR record is generated when the call ends. Therefore the order of the SMDR records output does not match the call start times.
- Each record contains a call ID which is increased by 1 for each subsequent call.
- When a call moves from one device to another, an SMDR record is output for the first part of the call and an additional SMDR record will be generated for the subsequent part of the call.
- Each of these records will have the same Call ID.
- Each record for a call indicates in the Continuation field if there will be further records for the same call.
- Wake up calls produce an SMDR record even if the intended extension was busy at the time of the call. Party1 is shown as **Wakeup Call**.

Call Times

- Each SMDR record can include values for ringing time, connected time, held time and parked time. The total duration of an SMDR record is the sum of those values.
- The time when a call is not in any one of the states above, for example when one party to the call has disconnected, is not measured and included in SMDR records.
- Where announcements are being used, the connected time for a call begins either when the call is answered or the first announcement begins.
- All times are rounded up to the nearest second.
- Each SMDR record has a Call Start time taken from the system clock time. For calls being transferred or subject to call splitting, each of the multiple SMDR records will have the same Call Start time as the original call.

17.1 SMDR Fields

The SMDR output contains the following fields. Note that time values are rounded up to the nearest second.

1. **Call Start**
Call start time in the format YYYY/MM/DD HH:MM:SS. For all transferred call segment this is the time the call was initiated, so each segment of the call has the same call start time.

2. **Connected Time**
Duration of the connected part of the call in HH:MM:SS format. This does not include ringing, held and parked time. A lost or failed call will have a duration of 00:00:00. The total duration of a record is calculated as *Connected Time + Ring Time + Hold Time + Park Time.*

3. **Ring Time**
Duration of the ring part of the call in seconds.
    - For inbound calls this represents the interval between the call arriving at the switch and it being answered, not the time it rang at an individual extension.
    - For outbound calls, this indicates the interval between the call being initiated and being answered at the remote end if supported by the trunk type. Analog trunks are not able to detect remote answer and therefore cannot provide a ring duration for outbound calls.

4. **Caller**
The callers' number. If the call was originated at an extension, this will be that extension number. If the call originated externally, this will be the CLI of the caller if available, otherwise blank.

5. **Direction**
Direction of the call – ***I*** for Inbound, ***O*** for outbound. Internal calls are represented as ***O*** for outbound. This field can be used in conjunction with **Is_Internal** below to determine if the call is internal, external outbound or external inbound.

6. **Called Number**
This is the number called by the IP Office. For a call that is transferred this field shows the original called number, not the number of the party who transferred the call.
    - **Internal calls:** The extension, group or short code called.
    - **Inbound calls:** The DDI dialed by the caller if available.
    - **Outbound calls:** The dialed digits.
    - **Voice Mail:** Calls to a user's own voicemail mailbox.

7. **Dialled Number**
For internal calls and outbound calls, this is identical to the **Called Number** above. For inbound calls, this is the DDI of the incoming caller.

8. **Account**
The last account code attached to the call. Note: IP Office account codes may contain alphanumeric characters.

9. **Is Internal**
0 or ***1***, denoting whether both parties on the call are internal or external (***1*** being an internal call). Calls to SCN destinations are indicated as internal.

Direction	Is Internal	Call Type
I	**0**	Incoming external call.
O	**1**	Internal call.
O	**0**	Outgoing external call.

10. **Call ID**
This is a number starting from 1,000,000 and incremented by 1 for each unique call. If the call has generates several SMDR records, each record will have the same Call ID. Note that the Call ID used is restarted from 1,000,000 is the IP Office is restarted.

11. **Continuation**
1 if there is a further record for this call id, ***0*** otherwise.

12. **Party1Device**
The device 1 number. This is usually the call initiator though in some scenarios such as conferences this may vary. If an extension/hunt group is involved in the call its details will have priority over a trunk, this includes remote SCN destinations.

Type	Party Device	Party Name
Internal Number	**E**<*extension number*>	<name>
Voicemail	**V**<*9500 + channel number*>	**VM Channel** <*channel number*>

Type	Party Device	Party Name
Conference	**V***<1><conference number>+<channel number>*	**CO Channel** *<conference number.channel number>*
Line	**T***<9000+line number>*	**Line** *<line number>.<channel if applicable>*
Other	**V***<8000+device number>*	**U***<device class> <device number>***.***<device channel>*
Unknown/Tone	**V8000**	**U1 0.0**

13. **Party1Name**
    The name of the device – for an extension or agent, this is the user name.

14. **Party2Device**
    The other party for the SMDR record of this call segment. See **Party1Device** above.

15. **Party2Name**
    The other party for the SMDR record of this call segment. See **Party1Name** above.

16. **Hold Time**
    The amount of time in seconds the call has been held during this call segment.

17. **Park Time**
    The amount of time in seconds the call has been parked during this call segment.

18. **AuthValid**
    This field is used for authorization codes. This field shows ***1*** for valid authorization or ***0*** for invalid authorization.

19. **AuthCode**
    This field shows either the authorization code used or ***n/a*** if no authorization code was used.

20. **User Charged**
    This and the following fields are used for ISDN Advice of Charge (AoC). The user to which the call charge has been assigned. This is not necessarily the user involved in the call.

21. **Call Charge**
    The total call charge calculated using the line cost per unit and user markup.

22. **Currency**
    The currency. This is a system wide setting set in the IP Office configuration.

23. **Amount at Last User Change**
    The current AoC amount at user change.

24. **Call Units**
    The total call units.

25. **Units at Last User Change**
    The current AoC units at user change.

26. **Cost per Unit**
    This value is set in the IP Office configuration against each line on which Advice of Charge signaling is set. The values are 1/10,000th of a currency unit. For example if the call cost per unit is £1.07, a value of 10700 should be set on the line.

27. **Mark Up**
    Indicates the mark up value set in the IP Office configuration for the user to which the call is being charged. The field is in units of 1/100th, for example an entry of 100 is a markup factor of 1.

28. **External Targeting Cause**
    This field indicates who or what caused the external call and a reason code. For example ***U FU*** indicates that the external call was caused by the Forward Unconditional setting of a User.

Targeted by		Reason Code	
HG	Hunt Group.	**fb**	Forward on Busy.
U	User.	**fu**	Forward unconditional.
LINE	Line.	**fnr**	Forward on No Response.
AA	Auto Attendant.	**fdnd**	Forward on DND.
ICR	Incoming Call Route.	**CfP**	Conference proposal (consultation) call.
RAS	Remote Access Service.	**Cfd**	Conferenced.
		XfP	Transfer proposal (consultation) call.
		Xfd	Transferred call.

29. **External Targeter Id**
The associated name of the targeter indicated in the External Targeting Cause field. For hunt groups and users this will be their name in the IP Office configuration. For an Incoming Call Route this will be the Tag if set, otherwise ***ICR***.

30. **External Targeted Number**
This field is used for forwarded calls to an external line. It shows the external number called by the IP Office as a result of the off switch targeting where as other called fields give the original number dialed.

17.2 SMDR Examples

The following are examples of IP Office SMDR records for common call scenarios.

Basic Examples

Lost incoming Call

In this record, the Call duration is zero and the Continuation field is 0, indicating that the call was never connected. The Ring Time shows that it rang for 9 seconds before ending.

```
2008/06/28 09:28:41,00:00:00,9,8004206,I,4324,4324,,0,1000014155,0,E4324,Joe Bloggs,T9161,LINE 5.1,0,0
```

Call Answered by Voicemail

In this example, 15 has made a call to 11. However the Party2Device and Party2Name show that the call was answered by voicemail.

```
2008/10/20 06:43:58,00:00:10,21,15,O,11,11,,I,28,0,E15,Extn15,V9051,VM Channel 1,0,0
```

Call Transferred to Voicemail

In this example, the Continuation field in the first record tells us that it wasn't the end of the call. The matching Call ID identifies the second record as part of the same call. The change in Party 1 details between the two records show that the call was transferred to voicemail.

```
2008/06/28 09:30:57,00:00:13,7,01707392200,I,299999,299999,,0,1000014160,1,E4750,John Smith,T9002,LINE 1.2,11,0
2008/06/28 09:30:57,00:00:21,0,01707392200,I,299999,299999,,0,1000014160,0,V9502,VM Channel 2,T9002,LINE 1.2,0,0
```

External Call

The Is Internal field being 0 shows this to be a external call. The Direction field as I shows that it was an incoming call. The Ring Time was 7 seconds and the total Connected Time was 5 seconds.

```
2008/08/01 15:14:19,00:00:05,7,01707299900,I,23,390664,,0,1000013,0,E23,Extn23,T9001,Line 1.2,0,0,,,,,,,,,,,,,
```

Internal call

The Is Internal field being 1 shows this to be a internal call. The Ring Time was 4 seconds and the total Connected Time was 44 seconds.

```
2008/06/26 10:27:44,00:00:44,4,4688,O,4207,4207,,1,1000013898,0,E4688,Joe Bloggs,E4207,John Smith,0,0
```

Outgoing Call

The combination of the Direction field being outbound and the Is Internal field be 0 show that this was a outgoing external call. The line (and in this case channel) used are indicated by the Party2 Name and being a digital channel the Ring Time before the call was answered is also shown.

```
2008/06/28 08:55:02,00:08:51,9,4797,O,08000123456,08000123456,,0,1000014129,0,E4797,Joe Bloggs,T9001,LINE 1.1,0,0
```

Voicemail Call

The two records below show calls to voicemail. The first shows the Dialed Number as*17, the default short code for voicemail access. The second shows the Dialed Number as VoiceMail, indicating some other method such as the Message key on a phone was used to initiate the call.

```
2008/06/28 09:06:03,00:00:19,0,4966,O,*17,*17[1],,1,1000014131,0,E4966,John Smith,V9501,VM Channel 1,0,0
2008/06/28 09:06:03,00:00:19,0,4966,O,VoiceMail,VoiceMail,,1,1000014134,0,E4966,John Smith,V9501,VM Channel 1,0,0
```

Parked Call

In this example the first record has a Park Time showing that the call was parked. The Continuation field indicates that the call did not end this way and there are further records. The second record has the same Call ID and shows a change in the Party2Name [4], indicating that party unparked the call. Note also that both records share the same call start time.

```
2008/10/20 07:18:31,0:00:12,3,215,O,210,210,,1,38,1,E15,Extn15,E10,Extn10,0,7
2008/10/20 07:18:31,0:00:10,0,215,O,210,210,,1,38,0,E15,Extn15,E11,Extn11,0,0
```

Incoming call with Account Code

In this example, at some stage as the call was made or during the call, an Account Code has been entered. In this specific case it is a text account code which can be selected and entered by the user using IP Office Phone Manager.

```
2008/06/28 11:29:12,00:00:02,2,5002,I,1924,1924,Support,0,1000014169,0,E1924,Extn1924,T9620,LINE 8.20,0,0
```

Conference Using Conference Button

In this example, an extension user answers a call and then brings in another user by using the Conference button on their phone. Again we see records for the initial call, the conference proposal call and then for the 3 parties in the conference that is created.

```
2008/07/09 15:05:41,00:00:04,3,13,O,11,11,,1,1000009,1,E13,Extn13,E11,Extn11,0,0
2008/07/09 15:05:26,00:00:09,3,17,O,13,13,,1,1000008,1,E17,Extn17,E13,Extn13,10,0
2008/07/09 15:05:41,00:00:08,0,,O,,,,1,1000009,0,E11,Extn11,V11001,CO Channel 100.1,0,0
2008/07/09 15:05:50,00:00:10,0,13,O,11,11,,1,1000010,0,E13,Extn13,V11002,CO Channel 100.2,0,0
2008/07/09 15:05:26,00:00:10,0,17,O,13,13,,1,1000008,0,E17,Extn17,V11003,CO Channel 100.3,0,0
```

Adding a Party to a Conference

This example is a variant on that above. Having started a conference, extension 13 adds another party.

```
2008/07/09 15:08:31,00:00:03,3,13,O,11,11,,1,1000014,1,E13,Extn13,E11,Extn11,0,0
2008/07/09 15:08:02,00:00:22,6,17,O,13,13,,1,1000013,1,E17,Extn17,E13,Extn13,9,0
2008/07/09 15:08:45,00:00:02,4,13,O,403,13,,0,1000016,1,E13,Extn13,E403,Libby Franks,0,0
2008/07/09 15:08:02,00:00:24,0,17,O,13,13,,1,1000013,0,E17,Extn17,V11003,CO Channel 100.3,0,0
2008/07/09 15:08:39,00:00:17,0,13,O,11,11,,1,1000015,0,E13,Extn13,V11002,CO Channel 100.2,8,0
2008/07/09 15:08:31,00:00:26,0,,O,,,,1,1000014,0,E11,Extn11,V11001,CO Channel 100.1,0,0
2008/07/09 15:08:45,00:00:12,0,,O,403,403,,0,1000016,0,E403,Libby Franks,V11004,CO Channel 100.4,0,0
```

Transfer

In this example 2126 has called 2102. The record (1) for this has the Continuation set a 1 indicating that it has further records. In the following record (3) with the same Call ID it can be seen that the Party 2 Device and Party 2 Name fields have changed, indicating that the call is now connected to a different device, in this example 2121. We can infer the blind transfer from the intermediate record (2) which shows a call of zero Connected Time between the original call destination 2102 and the final destination 2121.

```
2008/07/09 17:51,00:00:38,18,2126,O,2102,2102,,1,1000019,1,E2126,Extn2126,E2102,Extn2102,19,0
2008/07/09 17:52,00:00:00,7,2102,O,2121,2121,,1,1000020,0,E2102,Extn2102,E2121,Extn2121,0,0
2008/07/09 17:51,00:00:39,16,2126,O,2102,2102,,1,1000019,0,E2126,Extn2126,E2121,Extn2121,0,0
```

In this second example, extension 22 answers an external call and then transfers it to extension 23. Again the two legs of the external call have the same time/date stamp and same call ID.

```
2008/08/01 15:23:37,00:00:04,7,01707299900,I,4001,390664,,0,1000019,1,E22,Extn22,T9001,Line 1.1,6,0,,,,,,,,,,,,,
2008/08/01 15:23:46,00:00:00,3,22,O,23,23,,1,1000020,0,E22,Extn22,E23,Extn23,0,0,,,,,,,,,,,,,
2008/08/01 15:23:37,00:00:04,4,01707299900,I,4001,390664,,0,1000019,0,E23,Extn23,T9001,Line 1.1,0,0,,,,,,,,,,,,,
```

Busy/Number Unavailable Tone

In this example 2122 calls 2123 who is set to DND without voicemail. This results in 2122 receiving busy tone.

The record shows a call with a Connected Time of 0. The Call Number field shows 2123 as the call target but the Party 2 Device and Party 2 Name fields show that the connection is to a virtual device.

```
2008/07/09 17:59,00:00:00,0,2122,O,2123,2123,,1,1000033,0,E2122,Extn2122,V8000,U1 0.0,0,0
```

Call Pickup

The first record shows a call from 2122 to 2124 with a Connected Time of zero but a Ring Time of 8. The Continuation field indicates that the call has further records.

The second record has the same Call ID but the Party 2 Device and Party 2 Name details show that the call has been answered by 2121.

```
2008/07/09 18:00,00:00:00,8,2122,O,2124,2124,,1,1000038,1,E2122,Extn2122,E2124,Extn2124,0,0
2008/07/09 18:00,00:00:38,1,2122,O,2124,2124,,1,1000038,0,E2122,Extn2122,E2121,Extn2121,0,0
```

Park and Unpark

Parking and unparking of a call at the same extension is simply shown by the Park Time field of the eventual SMDR record. Similarly calls held and unheld at the same extension are shown by the Held Time field of the eventual SMDR record for the call. The records below however show a call parked at one extension and then unparked at another.

The records show a call from 17 to 13. 13 then parks the call shown by the Park Time. The call is unparked by 11, hence the first record is indicated as continued in its Continuation field. The matching Call ID indicates the subsequent record for the call.

```
2008/07/09 16:39:11,00:00:00,2,17,O,13,13,,1,1000052,1,E17,Extn17,E13,Extn13,0,4
2008/07/09 16:39:11,00:00:02,0,17,O,13,13,,1,1000052,0,E207,Extn17,E11,Extn11,0,0
```

Outgoing External Call

The External Targeting Cause indicates that the external call was caused by a user. The lack of specific reason implies that it was most likely dialed. The External Targeter ID is the user name in this example

```
… 16:23:06,00:00:04,5,13,O,9416,9416,,0,1000035,0,E13,Extn13,T9005,Line 5.1,0,0,,,Extn13,,,,,,,,U,Extn13,,
```

Rerouted External Call

In this example an incoming external call has been rerouted back off switch, shown by the Party 1 fields and the Party 2 fields being external line details. The External Targeter Cause shows that rerouting of the incoming call was done by an incoming call route (ICR). The External Targeter ID in this case is the Tag set on the incoming call route. The External Targeted Number is the actual external number call.

```
… 08:14:27,00:00:03,5,392200,I,9416,200,,0,1000073,0,T9005,Line 5.1,T9005,Line 5.2,0,0,,,,0000.00,,0000.00,0,0,618,0.01,
   ICR,Main ICR,416,
```

Transferred Manually

In this example the internal user transfers a call to an external number. The External Targeting Cause in the first record indicates that this external call is the result of a user (U) transfer proposal (XfP) call. The Continuation field indicates that another record with the same Call ID will be output.

The additional records are output after the transferred call is completed. The first relates to the initial call prior. The second is the transferred call with the External Targeting Cause now indicating user (U) transferred (Xfd).

```
… 16:33:19,00:00:05,3,13,O,9416,9416,,0,1000044,1,E13,Extn13,T9005,Line 5.1,0,0,,,,,,,,,,,U XfP,Extn17,,
… 16:33:09,00:00:02,2,17,O,13,13,,1,1000043,0,E17,Extn17,E13,Extn13,11,0,,,,,,,,,,,,,,
… 16:33:19,00:00:04,0,17,O,9416,9416,,0,1000044,0,E17,Extn17,T9005,Line 5.1,0,0,,,Extn17,,,,,,,,U Xfd,Extn13,,
```

External Conference Party

This is similar to internal conferencing (see examples above) but the conference setup and progress records include External Targeting Cause codes for user (U) conference proposal (CfP) and user (U) conferenced (Cfd).

```
… 16:48:58,00:00:02,2,13,O,9416,9416,,0,1000066,1,E13,Extn13,T9005,Line 5.1,0,0,,,,,,,,,,,U CfP,Extn13,,
… 16:48:37,00:00:04,3,13,O,17,17,,1,1000064,1,E13,Extn13,E17,Extn17,7,0,,,,,,,,,,,,,,
… 16:49:04,00:00:08,0,13,O,9416,9416,,1,1000067,0,E13,Extn13,V11002,CO Channel 100.2,0,0,,,,,,,,,,,,,,
… 16:48:37,00:00:13,0,,O,,,,1,1000064,0,E207,Extn17,V11003,CO Channel 100.3,0,0,,,,,,,,,,,,,,
… 16:48:58,00:00:13,0,,O,9416,9416,,0,1000066,0,V11001,CO Channel 100.1,T9005,Line 5.1,0,0,,,Extn13,,,,,,,,U Cfd,Extn13,
```

Two Outgoing External Calls Transferred Together

This scenario shows an outgoing call which is then transferred to another outgoing call.

```
2009/02/19 11:13:26,00:00:06,0,13,O,9403,9403,,0,1000012,1,E13,Extn13,T9001,Line 1.0,8,0,n/a,0,,,,,,,,U,Extn13,,
2009/02/19 11:13:36,00:00:02,0,13,O,8404,8404,,0,1000013,0,E13,Extn13,T9002,Line 2.0,0,0,n/a,0,,,,,,,,U XfP,Extn13,,
2009/02/19 11:13:26,00:00:11,0,8404,I,404,,,0,1000012,0,T9002,Line 2.0,T9001,Line 1.0,0,0,n/a,0,,,,,,,,LINE Xfd,
   0.1038.0 13 Alog Trunk:2,,
```

Chapter 18.
Document History

18. Document History

7th July 2016	**03b**	Update for IP Office Release 10.0. First source for help.
2nd August 2016	**03c**	Updated list of supported browsers for web management. [110276] Update to include StartTLS field setting. [110569] Update to hunt group modes. [109441]
24th October 2016	**03d**	Restructure to account for PDF based help.
1st December 2017	**03e**	Sync of version numbering shown on document and set in source.
15th March 2018	**04a**	Removed RAM/NVRAM config memory warning. Delay is less than a few seconds now.
20th April 2020	**05a**	IP Office 11.1.0.1: Addition of 911-View/Emergency View 220 button feature.

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