IP Office Basic Edition

Basic Edition Manager R11.1

- Issue 10a – (Thursday, July 23, 2020)

Contents

Chapter 1. Telephony Features

1. Telephony Features

This section covers details of the feature configurable for an IP Office Basic Edition system using IP Office Manager. It is an operating mode of IP Office that support up to 32 analogue trunks and 100 users (100 if using a 3 digit dial plan, 48 is using a 2 digit dial plan).

IP Office Basic Edition

IP Office Basic Edition mode is the default mode assumed by a IP500v2 control unit fitted with a new IP Office A-Law or IP Office U-Law SD card. In addition to analogue trunks, SIP trunks and digital (BRI or PRI) trunks are also supported.

Quick mode itself also operates in either of two system modes; behaving as either a key system or a PBX system 6. Systems with an Mu-Law SD card default to Key system operation, those with an A-Law SD card default to PBX system operation. However this setting can be changed within the system configuration if required.

Quick mode systems can be changed to IP Office standard mode operation if required. This is done by selecting using IP Office Manager (**File | Advanced | Switch to Standard Mode** 142).

IP Office Basic Edition - PARTNER Mode

IP Office Basic Edition - PARTNER Mode is the default mode assumed by a IP500v2 control unit fitted with a new IP Office PARTNER SD card. In addition to analogue trunks, SIP trunks and digital PRI trunks are also supported.

Partner mode can operate as either a key system or a PBX system 6. By default it uses Key system operation, however this setting can be changed within the system configuration if required.

IP Office Basic Edition - Norstar Mode

IP Office Basic Edition - Norstar Mode mode is the default mode assumed by a IP500v2 control unit fitted with a new IP Office Norstar SD card. In addition to analogue trunks, SIP trunks and digital (BRI or E1 PRI) trunks are also supported.

Norstar mode can operate as either a key system or a PBX system 6. By default it uses key system operation, however this setting can be changed within the system configuration if required.

1.1 What's New in This Release

The following changes have been made for the IP Office Release 11.1.0.1 software:

- **911-View/Emergency View Button** 119
  A button set to this function indicates when a call has been made using a number in the system's Emergency Number List 41. Pressing the button displays a list of such calls.

1.2 Key System or PBX System

The operating mode of a system can be changed. Two modes are supported; key mode and PBX mode. The selected mode affects a number of controls, mainly around the making of outgoing calls and the routing of incoming calls.

1.2.1 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 setting is used to determine which button, if available, gets used.
- If an external number is added to contacts, when dialing, it rings on the first available line from the ALS. If there is no line added to the ALS, the call is not made. If it is an internal number it dials on the intercom.

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 assumed to be external.

The line used for an outgoing external call is determined by configuration settings. ARS Selectors are created which can be groups of lines or specific functions using any available ISDN lines. Different external number prefixes are then mapped to those ARS Selectors. 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.

All calls go through intercom. This is true for internal and external calls (ARS). If you enable a prefix for external numbers, then you must be manually modify the number stored in contacts. If the number was not added before the prefix option in the Manager was selected, you have to add the number with the selected prefix.

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 / Disallowed Number Lists**
  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 regardless of any other call barring. The users allowed lists override any disallowed lists 41 of which they are also member and the user's **Outgoing Call Bar** 46 and **Outgoing Call Restrictions** 53 settings. 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 members of the list are not allowed to dial. 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 43. 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.
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 119 button.

- **Account Codes**
  Each user can be configured to need to enter a valid account code whenever they make an external call.
- **Outgoing Call Restrictions**
  For each user, the type of external calls that the user is able to make can be configured.
- **Marked Speed Dials**
  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** 16
  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.

1.2.2 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** 52
      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 line 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** 16
  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 16, 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.

1.2.2.1 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** IP Office Release 6.1+ supports phantom extensions 17. One of these can be selected as the destination for calls.	✓	✓	✓	✓	✓	✓	✓	–	–	–	✓	✓
• **Hunt Group** Incoming calls can be routed to one of the 6 rotary hunt groups 57.	✓	✓	✓	✓	✓	✓	✓	–	–	–	✓	✓
• **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 **System Mode** 38 set to ***PBX System***, incoming calls are routed to the **Operator Group** 57.	–	–	–	–	–	–	✓*	–	–	–	–	✓
• **Calling Group** For systems with their **System Mode** 38 set to ***PBX System***, incoming calls can be routed to one of the 4 collective calling groups 57.	–	–	–	–	–	–	✓	–	–	–	✓	✓

* = Default destination.

1.2.2.2 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** IP Office Release 6.1+ supports phantom extensions 17. One of these can be selected as the destination for calls.	–	✓	✓	✓	✓	✓	–	✓	✓	✓	✓	✓
• **Hunt Group** Incoming calls can be routed to one of the 6 rotary hunt groups 57.	–	✓	✓	✓	✓	✓	–	✓	✓	✓	✓	✓
• **Voicemail** Route incoming calls to the systems voicemail to collect messages. This requires the caller to know the mailbox number and passcode.	–	✓	✓	✓	✓	✓	–	✓	✓	✓	✓	✓
• **76: Modem** For Release 6.1+, the option ***76: Modem*** can be selected to route the call to the systems built in V32 modem 18 function. This is intended for basic configuration access by system maintainers.	–	✓	✓	✓	✓	✓	–	✓	✓	✓	✓	✓
• **Auto Attendant** For Release 6.1+, any of the configured voicemail auto attendants can be selected as the call destination.	–	✓	✓	✓	✓	✓	–	✓	✓	✓	✓	✓
• **Operator Group** For systems with their **System Mode** 38 set to ***PBX System***, incoming calls are routed to the **Operator Group** 57.	–	–	–	–	–	–	–	✓*	✓*	✓*	✓*	✓
• **Calling Group** For systems with their **System Mode** 38 set to ***PBX System***, incoming calls can be routed to one of the 4 collective calling groups 57.	–	–	–	–	–	–	–	✓	✓	✓	✓	✓

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

1.3 Dial Plan

Extension Numbering

The system can be configured to use either a 2 digit or 3 digit dial plan for user extensions.

- 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 17.
- 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** 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 18. 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** 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** extension.

Auto Attendant Numbers

Dialing the appropriate number shown in the table below allows recording and playback of the matching auto attendant prompt. It is important to remember that callers always hear two prompts, a greeting prompt and then a menu prompt. In addition, they may also first hear the emergency greeting if it has been activated.

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

1.4 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.

1.5 Voicemail Operation

All Basic Edition systems include voicemail as standard. By default up to 2 calls can simultaneously use voicemail services. By adding licenses, this can be increased up to 6 simultaneous calls.

When Do Calls Go To a User's Mailbox?

If a user has voicemail enabled (**VMS Cover** set to ***Enabled*** *(the default)*), calls directed to ring at that user's extension go to the user's voicemail after having rung 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.
- There are number of methods for a user to switch their VMS Cover setting on or off (through their mailbox, through the phone menus, or using a **VMS Cover** 128 button)
- A **VMS Transfer** 128 button can be configured to let a user transfer calls directly to the mailbox of other users.

When Do Calls Go to an Auto Attendant?

The Basic Edition voicemail 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 as follows:

- **Immediate Auto Attendant Service**
  One of the auto attendants can be specified as the **Coverage Destination** for a particular line. The call is presented immediately to that auto attendant.
- **Delayed/Optional Auto Attendant Service**
  The **VMS Schedule** setting of each line can be used to set whether unanswered calls should go to a selected auto attendant. The settings can be enabled for day service, night service 16, both or never (the default). The delay used before going to the auto attendant is set by the line's **VMS Delay - Day** and **VMS Delay - Night** settings as appropriate.

1.6 Night Service

Use this feature to program a button on the first extension on the system to turn night service on and off. When night service is on, all lines assigned to the telephones of the users in the night service group 57 ring immediately, regardless of their normal line ringing settings.

Night service is useful if you want phones to ring after regular business hours. For example, although Shipping Department workers do not answer calls directly during the day, you want them to answer incoming calls after hours.

- You must program a Night Service Button on the first extension 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 38 as follows if set:
    - You must enter the password when turning Night Service on or off.
    - Calls to numbers other than marked system dials or in the Emergency Phone Number List 41 require 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 138.
- Night Service is unavailable on T1 lines with Direct Inward Dialing (DID).

1.7 Phantom Extensions

For Release 6.1 and higher, extension users are created in the system configuration for all users regardless of whether they are matched by physical extension ports available. 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.

1.8 One Touch Transfer

Release 6.1 and higher supports one touch transfer operation 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 48 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.

1.9 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.

Remote access requires entry of the user name and password used for IP Office Manager as the connection name and password.

1.10 SIP Trunks

IP Office Basic Edition 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 Basic Edition system supports 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.

Chapter 2.
The Manager Application

2. The Manager Application

IP Office Manager is a Windows PC application used to configure Avaya IP Office telephone systems. This document covers the use of Manager with Basic Edition systems to load, edit and save the configuration of those systems. The **Configuration Settings** 37 section covers details of the individual configuration settings accessible using IP Office Manager.

! Important: IP Office Manager is an Offline Editor

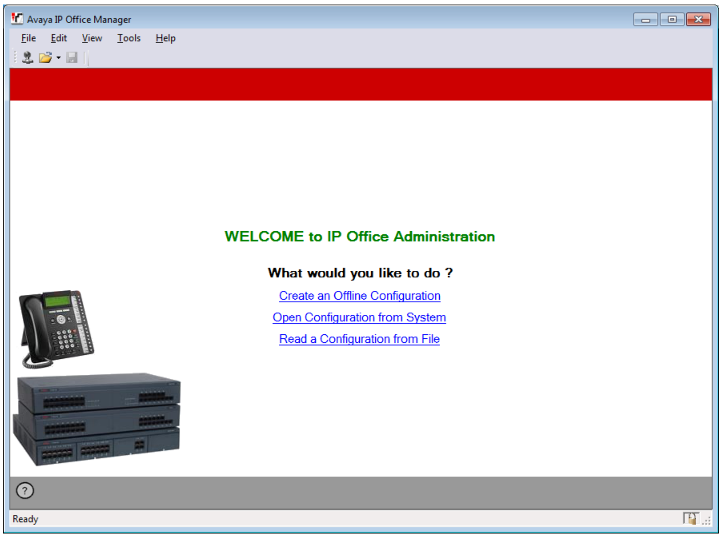
When a system configuration is loaded into Manager, it is a configuration file copied to the Manager PC. Any changes made to that configuration have no effect on the system until the copy is saved back to the system from the Manager PC.

Manager Modes

The menus and options displayed by Manager vary depending on the actions you are performing. Manager can run in the following modes.

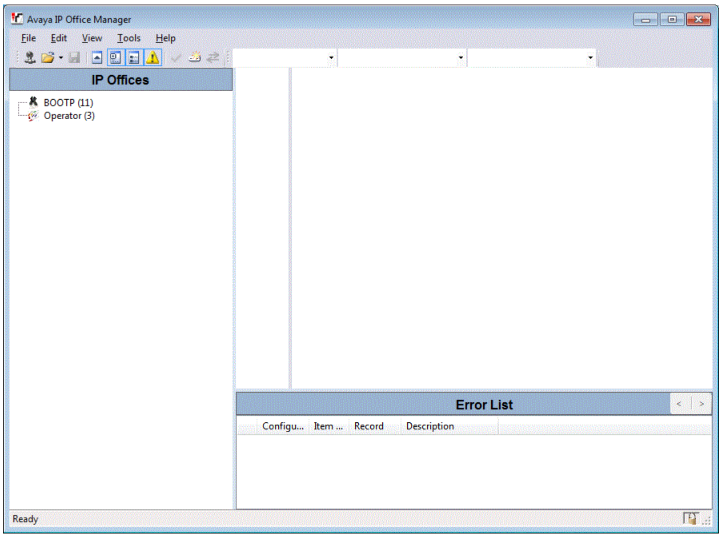
Simplified View 31

This is Managers default mode when no IP Office configuration has been opened.



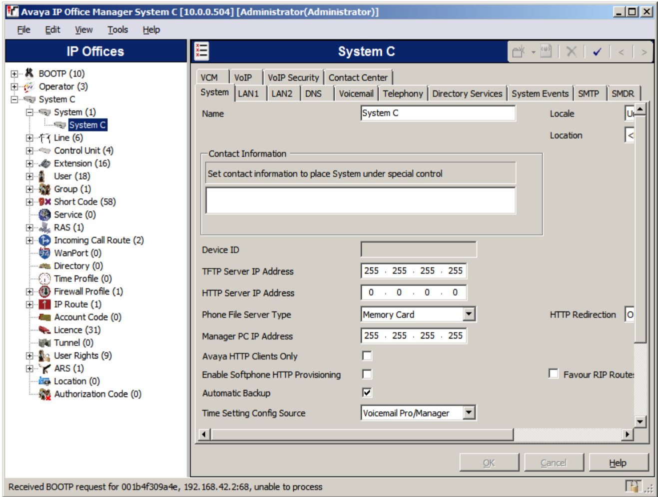
Advanced View 32

This mode can be selected instead of the Simplified view when no configuration is loaded. It is not normally used for Basic Edition systems and so is not covered by this document.



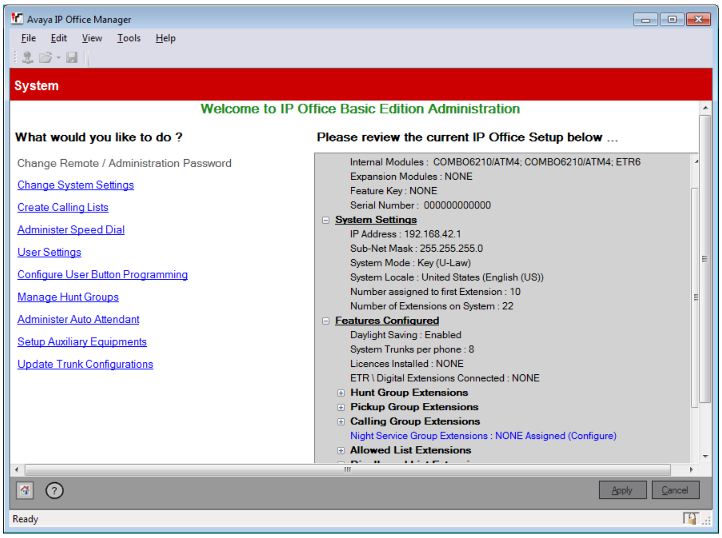
IP Office Configuration Mode 32

When the configuration from an IP Office system running in IP Office Standard mode is opened in Manager, Manager displays options for that mode. This mode is not covered by this document.



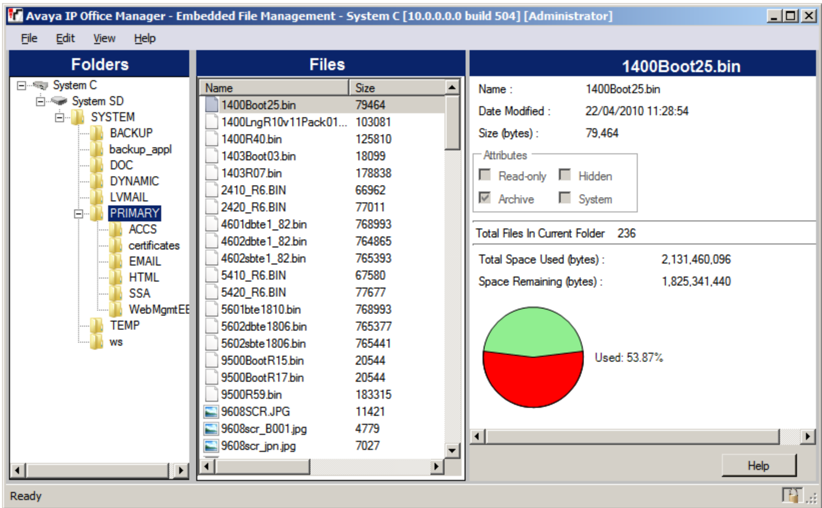
Basic Edition Configuration Settings 37

When the configuration from an IP Office system running Basic Edition is opened in Manager, Manager switches to this mode.



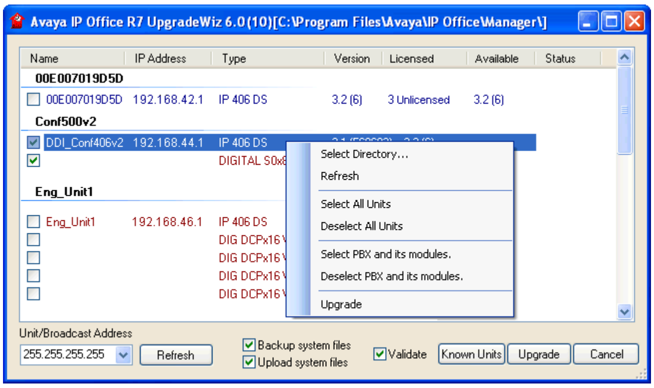
Embedded File Management 143

For systems with a memory card installed, Manager can be used to view and manage the files stored on the card. This is accessed through the **File | Advanced | Embedded File Management.** 143.

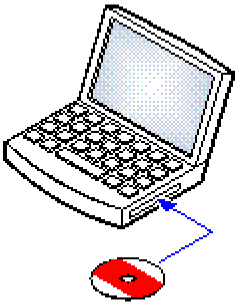


Upgrade Wizard 140

The Upgrade Wizard is a component of Manager used to upgrade the firmware run by the control unit and expansion modules within an IP Office system.



2.1 Installing Manager



The IP Office Administration suite consists of a number of applications for IP Office installers and maintainers.

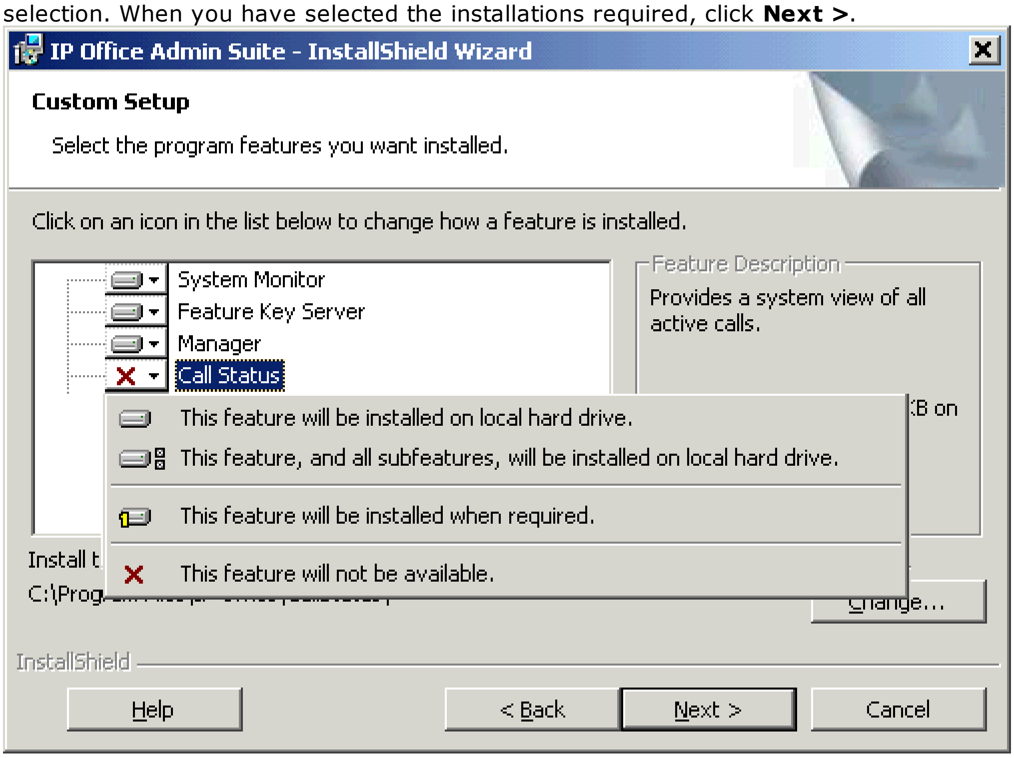
- ☐ **System Monitor** - *Install* ✓
- ☐ **Manager** - *Install* ✓
- ☐ **System Status Application** - *Install* ✓
- ☐ **Call Status** - *Optional*
  This software is not supported with IP Office Release 7.0 systems. It is provided only for the maintenance of older systems.

Requirements

- ☐ **IP Office Applications DVD**
  Alternatively the IP Office Administrator Applications suite can be downloaded from Avaya's support website (*http://support.avaya.com*).
- ☐ **Windows PC Requirements**
  The specifications below are the minimum requirements for IP Office Manager. If other applications are to be installed on the PC their individual requirements should also be meet.
  - **Standard Manager:** Core i3 CPU, 4GB RAM, 32/64-bit OS
  - **Server Edition Manager:** Core i5 CPU, 6GB RAM , 32/64-bit OS
  - **Server Edition Select Manager:** Core i5 CPU, 8GB RAM, 64-bit OS
  - IP Office Manager is only supported on Windows set to 100% font size display.

Installing the IP Office Admin Applications

1. Using the **Add or Remove Programs** option in the Windows Control Panel, check that the PC does not already have a version of the IP Office Admin suite installed.
    - If 'yes' and the suite is a pre-IP Office 3.2 version, remove the existing IP Office Admin suite via Add/Remove Programs.
    - If the existing suite is IP Office 3.2 or higher, it is possible to upgrade without removing the previous installation. However, if the system already has a USB Feature Key, the key should be removed prior to upgrading and then reinserted and the PC restarted.
2. Insert the IP Office Administrator Applications DVD. Select the option for the IP Office Administration Suit. A folder window will display the installation files for the administration suite.
3. Right-click on **setup.exe**. and select **Run as administrator**.
4. Select the language you want to use for the installation process. This does not affect the language used by Manager when running. Click **Next >**.
5. Select who should be able to run the Admin Suite applications. Click **Next >**.
6. If required select the destination to which the applications should be installed. We recommend that you accept the default destination. Click **Next >**.
7. The next screen is used to select which applications in the suite should be installed. Clicking on each will display a description of the application. Click on the ▾ next to each application to change the installation selection. When you have selected the installations required, click **Next >**.



8. Ensure that at minimum **System Monitor** and **Manager** are selected. Click **Next >**.
9. Click **Install**.
10. Installation of Windows .Net2 components may be required. If menus for this appear, follow the prompts to install .Net.
11. If requested, reboot the PC.

2.2 Starting Manager

1. Select **Start | Programs | IP Office | Manager**.

   - If the PC has firewall software installed, you may be prompted as to whether you want to allow this program to access the network. Select **Yes** or **OK**.

2. When the Manager application starts, it briefly displays a splash screen. It will then perform several possible actions: and then presents the welcome screen.

3. By default the application will automatically scans the local network for an IP Office system. This behavior can be disabled in the Manager application preferences 133 in which case the default welcome page is displayed (see Simplified View 31).

   a. If only one system is found and it is current Administrator account password is ***password***, Manager will automatically load and display the configuration from that system.

   b. If only one system is found but its administrator password is not set to ***password***, the menu for entering the valid name and password is displayed.

   c. If several systems are found, the **Select IP Office** menu is displayed. Use this menu to select which system to load. For details of adjusting the **Select IP Office** menu see Setting the Discover Address 27.

   d. If no system is found or an invalid name and password are used, then the Manager simplified view 31 **Welcome** menu is displayed.

2.3 Setting the Discovery Addresses

By default, when or **File | Open configuration** is selected, Manager's **Select IP Office** menu appears. It performs a UDP broadcast to the address 255.255.255.255. This broadcast will only locate IP Office systems that are on the same network subnet as the PC running IP Office Manager.

The process above is called discovery. A UDP broadcast will not be routed to other networks and subnets. Therefore to find IP Office systems not located on the same subnet as the Manager PC, the following other options are supported.

- **Specific Addressing**
  The Unit/Broadcast Address shown on the Select IP Office menu can be changed to the specific IP address of the required system. A single address is routable and so can be used to discover an IP Office system on another subnet.
- **TCP Discovery Address Ranges**
  IP Office 3.2+ systems support discovery by TCP as well as UDP. To support this, a set of TCP addresses and address ranges can be specified for use by the Select IP Office discovery process.
- **Known Units Discovery**
  The IP Office 4.0 Q2 2007 maintenance release adds supports for a system whereby IP Office Manager can write the details of systems it discovers to a file. The list of systems in that file can then be used for access to those systems. See Known Units Discovery 28.
- **DNS Lookup**
  IP Office Manager 6.2 can be configured to locate IP Office systems using DNS name lookup. This requires the IP Office systems on a customer network to be added as names on the customer's DNS server and the Manager PC to be configured to use that server for DNS name resolution. The use of DNS is configured through **File | Preferences | Discovery** 135.

Changing the Initial Discovery Settings
The Discovery tab of the Manager Preferences menu can be used to set the UDP and TCP addresses used by the discovery process run by the Select IP Office menu.

1. Select **File | Preferences** menu.
2. Select the **Discovery** tab.
3. Under **UDP Discovery** you can enter the default UDP broadcast address to be used by the discovery process.
4. In the **IP Search Criteria** box you can enter IP addresses and IP address ranges for TCP discovery. Addresses should be separated by semi-colons, ranges by - dashes.

2.4 Known IP Office Discovery

The Manager **Select IP Office** menu normally displays IP Office systems discovered by Manager using either UDP broadcast and or TCP requests (see Setting the Discovery Addresses 27). Manager can be configured to also record details of discovered units and then display a list of those previously discovered ('known') IP Office systems.

Configuring Manager for Known System Discovery

Use of known systems discovery is not enabled by default. The IP Office Manager must be configured for the feature with a file location to which it can store and retrieve known system details.

1. Select **File | Preferences**.
2. Select the **Directories** tab.
3. In the **Known Units File** field, enter the directory path and file name for a CSV file into which Manager can write details of the IP Office systems it discovers. If the file specified does not exist it will be created by Manager.
4. Click **OK**.

Using Known System Discovery

1. When the **Select IP Office** screen is displayed click on **Known Units**.
2. The screen displays the list of IP Office systems previously discovered and stored in the CSV file.
    - To select an IP Office control unit, highlight the row containing unit data and click **OK**. The selected unit will appear in the **Select IP Office** window.
    - To filter displayed units, type the first few characters of the unit name in the **Filter** field. Any unit whose name does not match the filter will be temporarily hidden.
    - Each discovery appends data to the known unit list. It is possible that details of some entries in the list may be out of date. Right clicking on the leftmost (grey) column of any row will bring up a floating menu offering the options of **Refresh** and **Delete**.
    - A new entry may be manually added without having to access the system first through normal discovery. Enter the IP address of the new system in the IP Address column of the blank row shown with a * and select **Refresh** from the floating menu. This will update the Known Units file with data relating to the unit with the specified address.
    - Select **Cancel** to return to the **Select IP Office** menu.

Note:

- The key used by the Known Systems CSV file is the IP address. The file cannot contain entries for separate systems that use the same IP address for access.
- The file can be made read only. In that case any attempts using Manager to update the file will be ignored.

2.5 Saving the Configuration

The current configuration settings open within Manager can be sent to the IP Office system.

1. The first steps of this process depend on whether you are sending a configuration received from the IP Office system or sending one opened offline/created new.
   - **A Configuration Opened from an IP Office**
     Click in the main toolbar or select **File | Save Configuration** from the menu bar.
   - **A Configuration Created Offline or Opened from a PC File**
     Select **File | Offline | Send Config** from the menu bar.
2. The **Send Configuration** window is displayed.
   - **Configuration Reboot Mode**
     If Manager thinks the changes made to the configuration settings are mergeable, it will select Merge by default, otherwise it will select Immediate.
     - **Merge**
       Send the configuration settings without rebooting the IP Office. This option is selected by default if the changes made since the configuration was loaded into Manager as mergeable, do not select this option otherwise.
     - **Immediate**
       Send the configuration and then reboot the IP Office.
     - **When Free**
       Send the configuration and reboot the IP Office when there are no calls in progress. This mode can be combined with the **Call Barring** options.
     - **Timed**
       The same as When Free but waits for a specific time after which it then wait for there to be no calls in progress. The time is specified by the **Reboot Time**. This mode can be combined with the **Call Barring** options.
   - **Reboot Time**
     This setting is used when the reboot mode **Timed** is selected. It sets the time for the IP Office reboot. If the time is after midnight, the IP Office's normal daily backup is canceled.
   - **Call Barring**
     These settings can be used when the reboot mode When Free is selected. They bar the sending or receiving of any new calls.
3. Click **OK**. A Service User name and password may be requested.
   - If the service user name or password used do not have a match on the IP Office, ***"Access Denied"*** is displayed.
   - The message ***Failed to save the configuration data. (Internal error)*** may indicate that the system has booted using software other than that in its System SD card's primary folder.

2.6 Saving a Configuration to a PC File

The IP Office configuration settings shown within Manager can be saved to a .cfg file on the Manager PC. These files can be used as backups.

Automatically Saving Configuration Copies

By default, Manager creates a file copy of the configuration before it is sent to the IP Office system. This copy is stored in Manager's Working Directory 134 using the IP Office's system name and .cfg. This behavior is controlled by the **Backup File on Send (File | Preferences | Security)** 136 option.

Saving a Configuration Received from an IP Office

1. Select **File | Save Configuration as** from the menu bar.

Saving a Configuration opened on the PC

1. Click in the main toolbar or select **File | Save Configuration** from the menu bar.

2.7 Loading a PC File

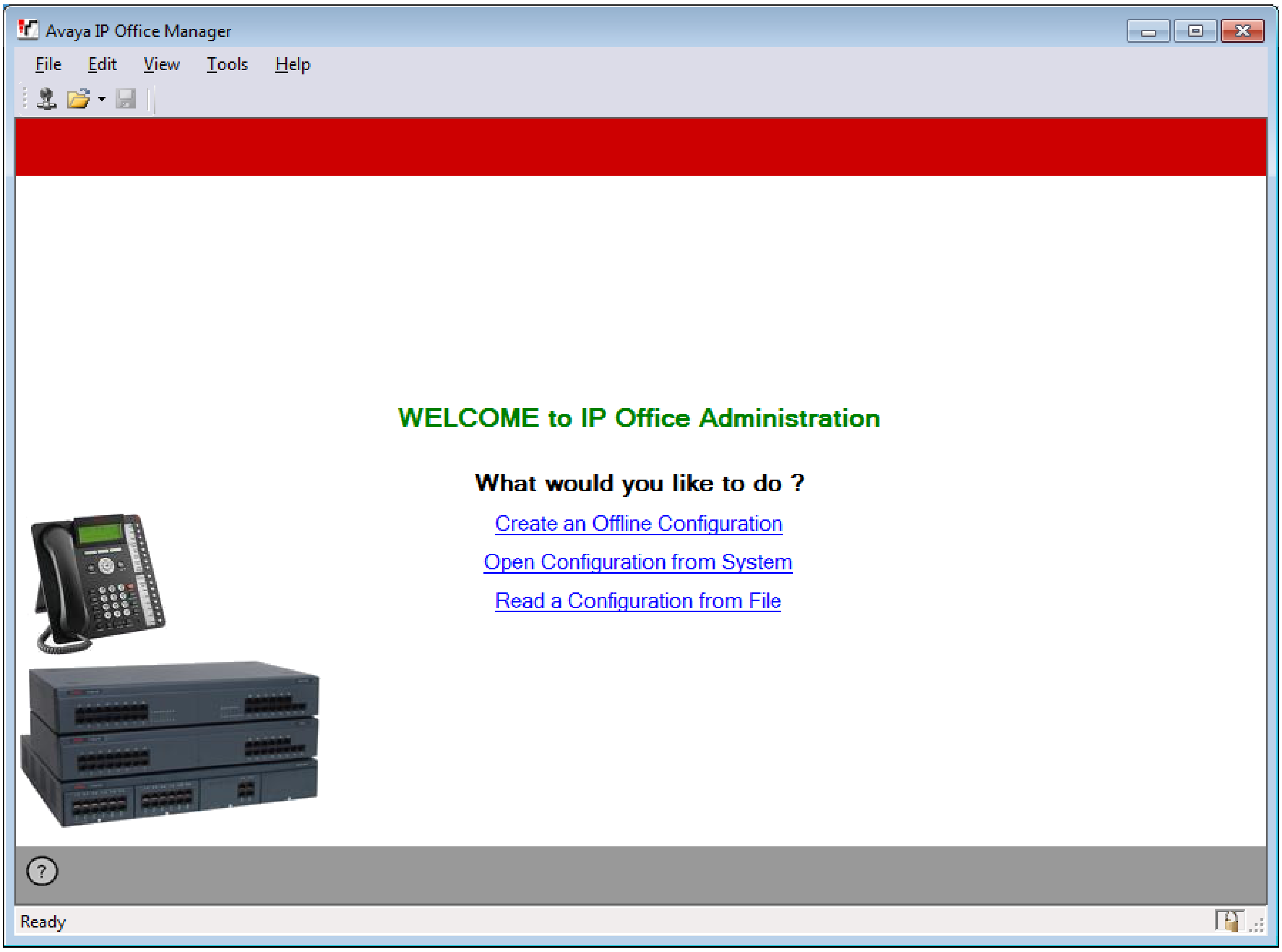
Configuration files previously saved onto the PC can be reloaded into Manager. Select **File | Offline | Open File** or from the default simplified view select Read a Configuration from File.

In order to send that configuration to a system the **File | Offline | Send Config** command must be used.

- A configuration created offline should only ever be loaded to a system with the matching hardware configuration. Doing otherwise may cause system faults.

2.8 Simplified View

This is the default view displayed by Manager when it doesn't have a system configuration loaded.



The screen provides three main actions:

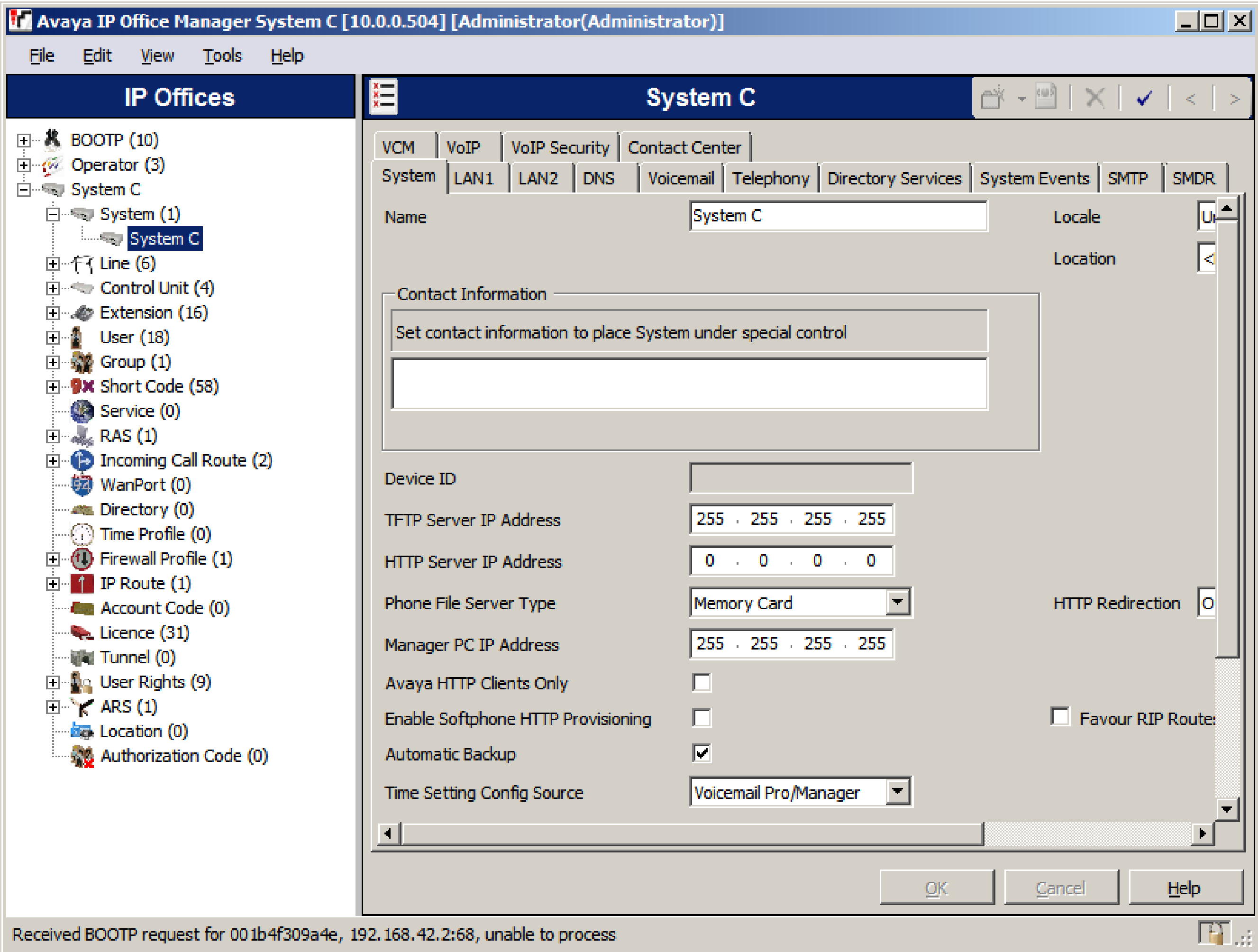
- **Create an Offline Configuration** 35
  Create an Basic Edition configuration by selecting from a menu of hardware options. That configuration can then be saved as a file on the PC or uploaded to a system.
- **Open Configuration from System** 26
  Restarts the process used by Manager to locate an IP Office system and load its configuration.
- **Read a Configuration from File** 30
  Load a configuration that has been saved as a file on the PC.

Manager can be switched from simplified view to advanced view by selecting **View | Advanced View**. The advanced view is not normally used with Basic Edition systems and so is not covered by this manual.

Whether Manager uses simplified view or advanced view when it has no configuration loaded is set by the Manager preferences 133 setting **Set Simplified View as default**.

2.9 Advanced View

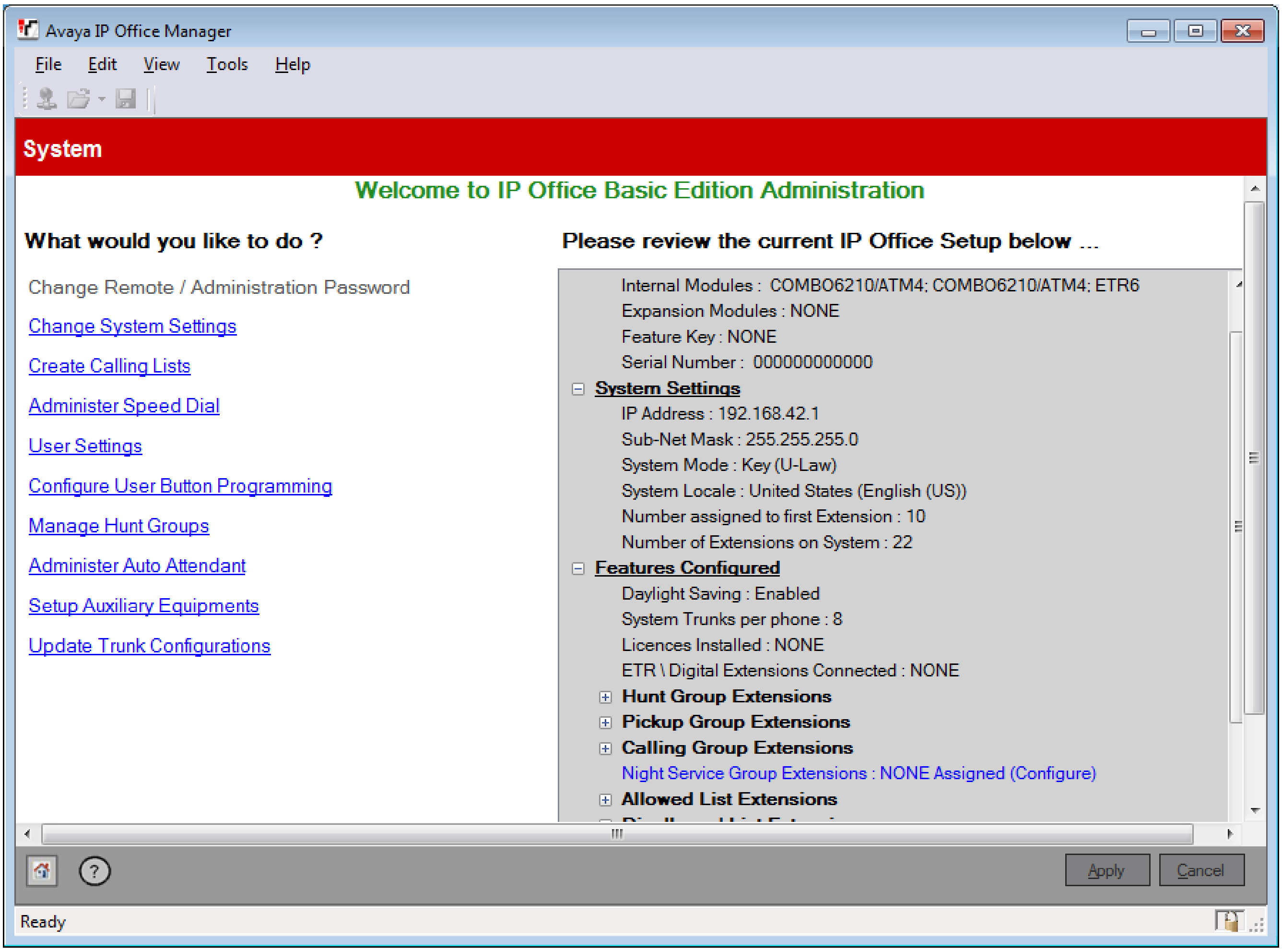
This view is used by IP Office Manager for the administration of standard IP Office systems. It is not used for the administration of Basic Edition systems.



If Manager is running in this mode, you can return to simplified view by selecting **View | Simplified View**. Alternatively you can use the advanced view to load a configuration. The Manager will automatically return to simplified view mode when an Basic Edition configuration is loaded (as long as **Default to Standard Mode** is not selected in preferences).

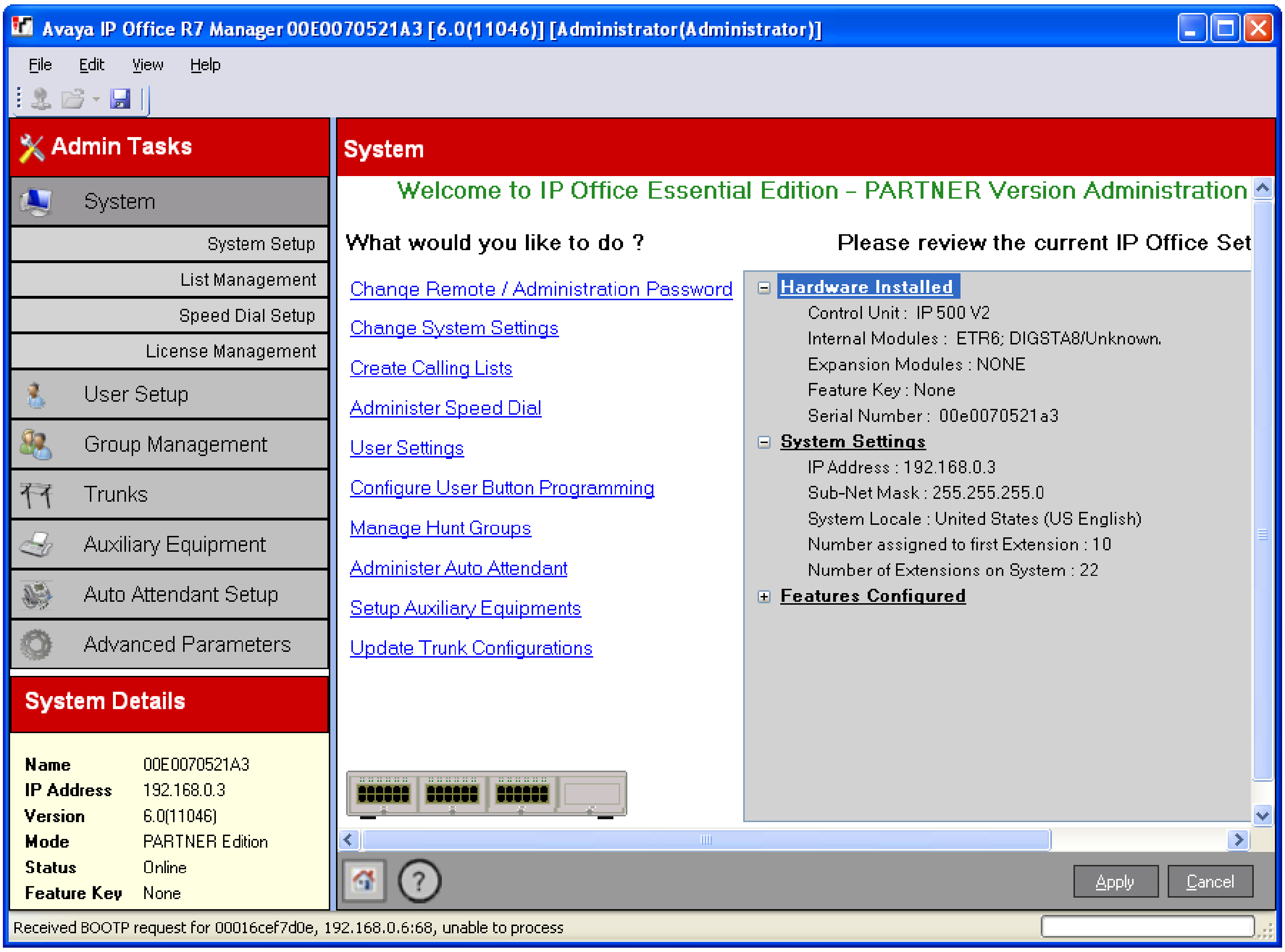
2.10 The System Page

This is the default or home page when a configuration has been loaded into IP Office Manager. It displays a summary of the system and a list of links for common configuration tasks.



2.11 The Admin Tasks List

The Admin Tasks list is hidden by default but can be displayed by deselecting **View | Hide Admin Tasks**. When displayed the list provides a set of links to access all of the Basic Edition system configuration menus.



2.12 Creating a Configuration File

Manager can be used to create a configuration file for a system.

- A configuration created offline should only ever be loaded to a system with the matching hardware configuration. Doing otherwise may cause system faults.
- Once the system has been installed, changing the order or combination of cards will require the system configuration to be defaulted.

1. Close any current configuration open in Manager.
2. Select **Create an Offline Configuration** or **File | Offline | Create New Config.**
3. Ensure that the **Configuration** setting is set to the required IP Office Basic Edition mode.
4. Select the **System Mode** required. The options are ***Key*** or ***PBX***. For more details see **Key System or PBX System** 6.
   - **Key System**
     The **Number of Lines** setting (see below) 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 48 settings.
   - **PBX System**
     No line appearances are automatically assigned. The **Outside Line** setting (see below) 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 **Outbound Call Handling** 99 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.
5. Set the **Locale** to match the default locale and language that should be used for the system. This will also affect the extension and daughter cards available for selection in the following steps. Changing the locale will cause any existing hardware selections to be cleared.
6. Select the extension number length that should be used.
7. In the **Select Extension and Daughter Cards** section, select the cards that match those in the system to which the configuration will be loaded.
   - Ensure that these match the actual physical positions of the cards that are or will be installed in the system. If the arrangement of cards needs to be changed at a later date, it may require the whole configuration to be deleted.
   - For system administration through the first two extensions, the card in slot 1 must support Avaya digital phones.
   - For IP Office Quick systems, the ETR6 extension card is only selectable for systems with a ***United States***, ***Canada*** or ***Mexico*** locale.
     BRI trunk cards are not selectable for systems with a ***United States***, ***Canada*** or ***Mexico*** locale.
8. Use the ***Expansion Modules*** box to select the expansion module if one exists.
9. When the hardware selection is as required, click **OK**.
10. The configuration is now created and loaded into Manager for editing.
11. Once this configuration has been edited as required it can be saved on the PC or sent to a system.
    a. **To Save a Configuration File on the PC**
       Use **File | Save Configuration**.
    b. **To Send the Configuration to a System**
       If the system which you want to use the configuration is available, use **File | Offline | Send Configuration** 137 to send the configuration to it.
       - **! WARNING:** This action will cause the system to reboot and will disconnect all current calls and service.
       - Ensure that you have a copy of the systems existing configuration before overwriting it with the off-line configuration.
       - After sending the configuration, you should receive the configuration back from the system and note any new validation errors shown by Manager. For example, if using Embedded Voicemail, some sets of prompt languages may need to be updated to match the new configurations locale setting using the **Add/Display VM Locales** 145 option.

Chapter 3.
Configuration Settings

3. Configuration Settings

This section details the configuration settings accessible through IP Office Manager.

3.1 Remote/Administrator Password

This menu is accessed from the **System** 33 page by selecting **Remote / Administrator Password**.
This menu cannot be accessed from the **Admin Tasks** 34.

New systems use default security settings with the user name ***Administrator*** and password ***Administrator***. This is the password used by IP Office Manager for access to a system. As a minimum, you should change the **Remote/Administrator Password**. Failure to do so will leave the system potentially insecure.

This password is also used for connection to the system by the ***Administrator*** account to System Status Application, System Monitor and IP Office Web Manager.

This command is greyed out and disabled when editing an off-line configuration.

Changing the Remote / Administration Password

1. From the Manager home page, click **Change Remote/Administrator Password**.
2. The **Change Password** menu is displayed.
3. Enter the new password, confirm it and click **OK**.
4. Click **Apply** in the system page or click on the icon.
5. In the **Send Configuration** menu click **OK**.
6. The user name and password will be requested. Enter **Administrator** and the old password.

3.2 System Settings

This menu is accessed from the **System** 33 page by selecting **Change System Settings**.	
This menu is accessed from the **Admin Tasks** 34 list by selecting **System**.	

This window displays a summary of the hardware components installed in the phone system. It also enables configuration of system-specific settings.

Installed Hardware

This section displays a list of the hardware components (control unit and its base cards) for trunks and extensions that are installed in the telephone system. These values are for information only and cannot be edited.

System Parameters

This section is used to configure the following system settings.

- **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 /.
- **System Mode:**
  Basic Edition systems can operate in either ***Key System*** or ***PBX System*** mode. For more details see Key System or PBX System 6. Changing the mode requires the IP Office system to be restarted and will overwrite button programming.
  - **Key System**
    The **Number of Lines** setting (see below) 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 48 settings.
  - **PBX System**
    No line appearances are automatically assigned. The **Outside Line** setting (see below) 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 **Outbound Call Handling** 99 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.
  - **Add/Display VM Locales:** *Software level = Release 8.0+.*
    For new IP500 V2 SD cards and cards recreated using IP Office Manager, the following Embedded Voicemail languages set are placed onto cards by default. Using this option displays the list of languages that can be uploaded from IP Office Manager. Those languages already present or not supported are greyed out. If a locale is selected for the system, a user, a short code or an incoming call route which is not present on the SD card, IP Office Manager will display an error. This command can be used to upload the required language prompts to correct the error.
    - **IP Office A-Law/Norstar SD Cards:** UK English.
    - **IP Office U-Law/PARTNER SD Cards:** US English.
- **File Writer IP Address:**
  This field set the address of the PC allowed to send files to the System SD card installed in the system using HTTP or TFTP methods other than embedded file management. On systems with an Avaya memory card, this field sets the address of the PC allowed to send files to the memory card using HTTP or TFTP methods other than embedded file management. An address of 255.255.255.255 allows access from any address. If Embedded File Management 149 is used, this address is overwritten by the address of the PC using embedded file management (unless set to 255.255.255.255).
- **Country:**
  This option sets a range of country specific telephony settings. It also sets the default language used on phone displays and for voicemail prompts. If the setting is changed it will cause the settings of all users and auto attendants to change to match. The system language can be changed using the separate **Language** setting below.
  - On Norstar systems, when **Default** is selected, the following additional fields are available.
    On Quick systems, when **Customize** 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.

- **! WARNING**
  Changing the system language requires the system to be rebooted when the changes are sent back to the system.
- For each user, their language settings can be changed using the user's **Language** 46 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** 105 setting.

- **Language**
  The default system language is normally set by the system's **Country** selection above (indicated in brackets after the country name). However, this field can be used to change the system language if required. When used, it sets the language used for voicemail prompts and phone displays if the language is available. The language settings can also be set separately for each user 46 and for each auto attendant 105 service.
  - For Basic Edition Quick systems, the options 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***.
  - For Basic Edition Partner systems, the options are ***French (Canada)***, ***Spanish (Latin)***, ***English (US)***.
  - For Basic Edition Norstar systems, the options are ***Arabic***, ***French***, ***English (UK)***.

- **Receive IP Address Via DHCP Server:** *Default = On*
  When selected, the telephone system acts as a DHCP client and will obtain its IP address details by making DHCP requests when started. If not selected, the telephone system uses the IP address set in the fields below.
  - **IP Address (LAN1):** *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.
  - **Sub-Net Mask (LAN1):** *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.
  - **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.

- **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.

- **Automatic Daylight Saving Time:** *Default = On.* (Not applicable to Norstar systems)
  When selected, the telephone system will automatically apply daylight saving time adjustments to its internal clock. This feature should only be used for systems in a North American locale.

- **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 **System Mode** *(see above)* set to ***Key System***. 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 this value is changed, all existing line appearance buttons and automatic line selection 48 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.

- **Outside Line:** *Default = Depend on system locale, see below.*
  This option is only available for systems with their **System Mode** (see above) set to ***PBX System***. 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 **Outbound Call Handling** 99 settings.
  - ***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 12 is assumed to be an external call. This is the default setting for systems with the **Country** setting other then ***Germany*** or ***United States***.

- ***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***.

- **System 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, disallowed calls list and night service outward 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.

- **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 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.

- **Unsupervised Analog Trunk Disconnect Handling:** *Default = Off (On Norstar systems, On for Default locale).*
  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 IP Office uses Disconnect Clear signalling and or Busy Tone Detection 113. 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.

3.2.1 List Management

This menu is accessed from the **System** 33 page by selecting **Create Calling Lists**.	
This menu is accessed from the **Admin Tasks** 34 list by selecting **System | List Management**.	

Calling lists control the numbers user can or cannot dial. You can also indicate which lists a user belongs to through the **User Setup** 46 menu.

After highlighting the item you want to move, use the **Add** or **Remove** buttons to move users to and from the *Selected Users* list . The different types of Calling list are:

List Type	Description
Allowed Lists 41	Sets numbers that associated users can dial even when call restrictions are applied. 8 lists of 10 numbers.
Disallowed Lists 41	Sets numbers that associated users cannot dial. 8 lists of 10 numbers.
Emergency Number List 41	Sets up to 10 numbers that override all dialing restrictions at all times.
Account Code Entries 42	Sets up to 99 accounts codes and which users are required to enter an account code when making external calls.

3.2.1.1 Allowed Lists

This menu is accessed from the **System** 33 page by selecting **Create Calling Lists | Allowed Lists**.	
This menu is accessed from the **Admin Tasks** 34 list by selecting **System | List Management | Allowed Lists**.	

Each allowed list contains external telephone numbers that members of the list are allowed to dial regardless of any other call barring. The users allowed lists override any disallowed lists 41 of which they are also member and the user's **Outgoing Call Bar** 46 and **Outgoing Call Restrictions** 53 settings.

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.

3.2.1.2 Disallowed Lists

This menu is accessed from the **System** 33 page by selecting **Create Calling Lists | Disallowed Lists**.	
This menu is accessed from the **Admin Tasks** 34 list by selecting **System | List Management**.	

Each disallowed list contains external telephone numbers that users who are members of the list are not allowed to dial.

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 43.

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.

3.2.1.3 Emergency Number List

This menu is accessed from the **System** 33 page by selecting **Create Calling Lists | Emergency Number Lists**.	
This menu is accessed from the **Admin Tasks** 34 list by selecting **System | List Management**.	

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 119 button.

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

3.2.1.4 Account Code Entries

This menu is accessed from the **System** [33] page by selecting **Create Calling Lists | Account Code Entries**.	
This menu is accessed from the **Admin Tasks** [34] list by selecting **System | List Management**.	

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 call log. Users can enter an account code during a call using an **Account Code Entry** [49] 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.

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**.

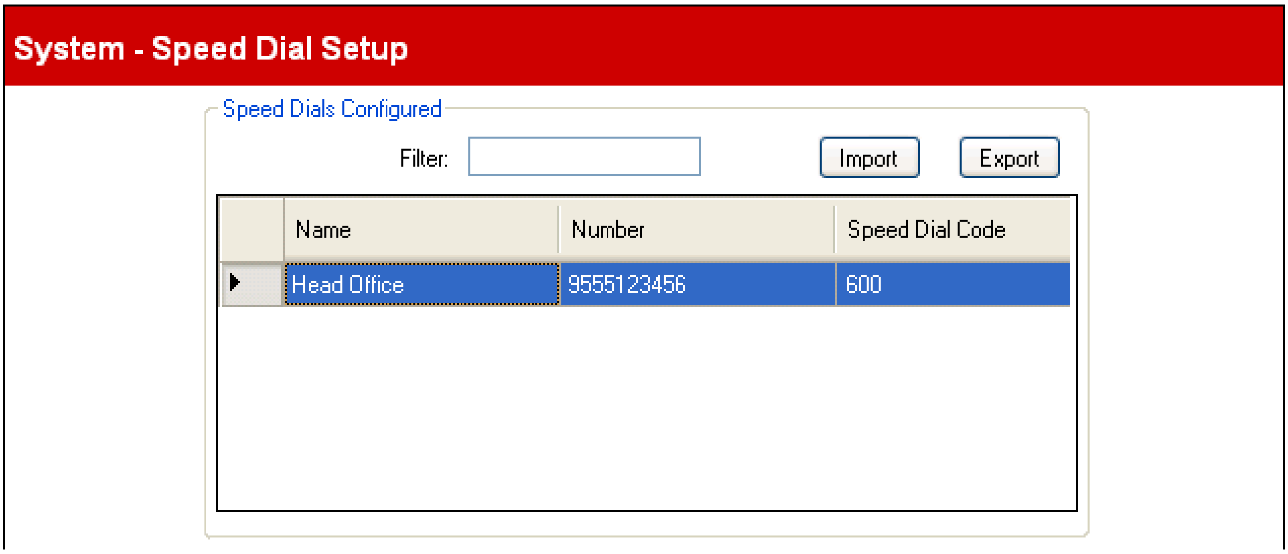
All users (except analog phones) can also enter voluntary account codes at any time during a call by using an Account Code Entry button. Voluntary account codes are recorded in the same way as forced account codes but are not validated.

Using the **Assign Users to List** menu to add or remove users from the **Selected User** list will enable/disable the **Forced Account Code Entry** [53] setting for the appropriate users.

3.2.2 Speed Dial Setup

This menu is accessed from the **System** 33 page by selecting **Administer Speed Dial**.
This menu is accessed from the **Admin Tasks** 34 list by selecting **System | Speed Dial Setup**.

This menu allows you to configure names and numbers that can be accessed by dialing the associated speed dial code, 600 to 699.



- 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.

Speed Dials Configured

- **Filter**
  This option allows you to show only speed dial entries where the name, number or speed dial code matches the filter value entered. If there are no matches the whole set of speed dial entries is displayed.

- **Import**
  Allows you to import a CSV text file of speed dials. Each line of the file should contain a name, number and speed dial code, each separated by a comma. If an entry being imported matches an existing name it will overwrite the existing entry. If an entry being imported matches an existing speed dial code, it will be assigned an unused speed dial code.

```
Head Office, 555123456, 600
Acme, 555654321, 601
```

- **Export**
  This control allows you to export a CSV text file of speed dials. You can then edit the file using a text editor.

  - **Comma Separated Variable text Files (.csv)**
    These are plain text files. In addition to being exported from Manager these files can be created and edited using programs such as WordPad. Manager imports and exports CSV files using UTF-8 character encoding which uses a double byte to support characters with diacritic marks such as **ä**. Other applications, such as Excel, may, depending on the user PC settings, use different encoding which will cause such characters to be removed or corrupted. Care should be taken to ensure that any tool used to create or edit the CSV supports all the characters expected and uses UTF-8 format.

    - **Exporting from Manager to Excel**
      Do not double-click on the file exported from Manager. Start Excel and use **File | Open** to select the file. Excel will recognize that the file uses UTF-8 encoding and start its text file importation wizard. Follow the wizard instructions and select comma as the field delimiter.

- **Speed Dial Entries**
  For each speed dial entry in the menu, the following values are used:

  - **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 a disallowed list of which the user is a member. 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** 38 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.

3.2.3 License Management

This menu cannot be accessed from the **System** 33 page.
This menu is accessed from the **Admin Tasks** 34 list by selecting **System | License Management**.

This menu is used to enter licenses required for additional telephone system features. For example, licenses are used to enable additional voicemail ports. Each license is uniquely based on the feature being licensed and the serial number of the SD card plugged into the system control unit.

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.

- **PLDS Host ID:**
  This is the unique key against which the licenses are validated.
  - The PLDS host ID used for the license file must match that shown by the 12 digit **PLDS ID** printed on the SD card label. Older cards have an 10 digit feature key serial number shown by **FK** or **FK SN**. For those cards add the prefix ***11*** to obtain the card's PLDS ID.
- **PLDS File Status:**
  Indicates whether a valid PLDS XML file has been uploaded to the system.

License Settings

For each license key entered, the following information is displayed:

- **Type:** *Information field, not editable.*
  Shows the name of the feature 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 SD card serial 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.
- **Quantity:** *Information field, not editable.*
  This field indicates how many items are enabled by the license. The meaning of this will vary depending on the feature being licensed.
- **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 licenses and how it was obtained by the system.

3.3 User Setup

This menu is accessed from the **System** 33 page by selecting **User Settings**.	
This menu is accessed from the **Admin Tasks** 34 list by selecting **User Setup**.	

This menu allows configuration of extension user settings. Note that # before an extension number indicates a phantom user, i.e. one not matched by an actual extension. Phantom users 17 can still be used for mailbox services and other features.

Configure User List

This list shows the current settings of all the extension users. The list is scrollable and sortable. The current group and list settings are show in the list and for the currently selected user can be edited in the **Membership Assignment** table below the list.

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

- **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 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.

- **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.
  - If the upgraded set of prompts for the language selected are not available on the system, IP Office Manager will display a warning. The **Add/Display VM Locales** 145 command can be used to upload the prompts from IP Office Manager to the system.

- **Exclude from Directory:** *Default = Off*
  If selected, the user is not included in the directory of users displayed on phones.

- **User CLI:** *Default = Blank.*
  This setting in only available on ***PBX System*** 6 mode systems. Where supported by the line provider, this CLI will be 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.

- **Outgoing Call Bar:** *Default = Off.*
  If selected, the extension user cannot make any outgoing external calls except to numbers in the **Emergency Number List** 41 and any **Allowed Lists** 41 of which they are a member.

- **List Memberships:** *Information field, not editable.*
  This field shows a summary of the **Allowed Lists** 41 (**AL**) and **Disallowed Lists** 41 (**DL**) to which the user belongs. If the user is selected, these can be edited in the **Membership Assignment** table below.

- **Group Memberships:** *Information field, not editable.*
  This field shows a summary of the hunt groups, pickup groups and calling groups to which the user belongs. If the user is selected, these can be edited in the **Membership Assignment** table below.

Membership Assignment

This section allows the calling list and group memberships of the currently selected user to be edited. The **Type** option is used to select either **List** or **Group** memberships.

- **List** 41
  If **List** is selected, the list of lists that exist and the lists of which the user is a member are displayed.

- **Group** 57
  If **Group** is selected, the list of groups that exist and the groups of which the user is a member are displayed.

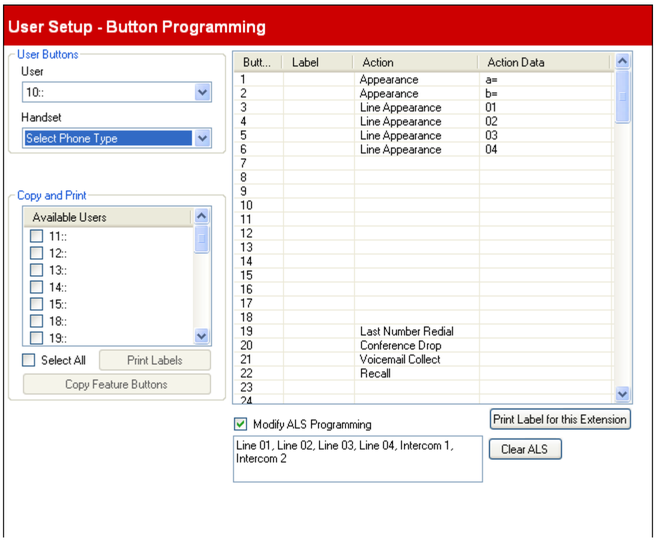
3.3.1 Button Programming

This menu is accessed from the **System** 33 page by selecting **Configure User Button Programming**.
This menu is accessed from the **Admin Tasks** 34 list by selecting **User Setup | Button Programming**.

Most Avaya phones have programmable buttons to which a variety of functions can be assigned. This menu can be used to edit the button settings. It can also be used to adjust the automatic line selection 48 order used by the phone.

- Note that for systems running in ***Key System*** 38 mode, a number of each users programmable buttons are automatically configured as line appearance button according to the system **Number of Lines** 38 setting. If the system setting **Number of Lines** 38 is changed, it may overwrite all or some of the current button programming.

The menu can operate in either of two ways, depending on whether the phone type is known or not. See the **Handset** setting.



Non-Graphical Mode
(Unknown phone type)

Graphical Mode
(Known phone type)

User Buttons

- **User**
  This drop down list is used to select the extension user whose programmed buttons are displayed for editing.

- **Handset**
  When a configuration is loaded from the telephone system at Manager 21 start-up, if the type of phone currently plugged into the extension port is recognized, the menu switches to graphical mode and displays a picture of the phone. If the phone type is not known, the menu can either be used in non-graphical mode or a phone type can be selected from the drop down list to switch to graphical mode.

  - For some phones the system can only detect the likely type. For example, ***M7100***, ***M7100N***, ***T7100*** or ***ACU*** telephones are all initially listed as being an ***M7100***. The drop-down list allows selection of the exact model if required.

Copy and Print

This section of the menu allows you to copy the current user's button program settings to other extension users.

- **Available Users**
  Select the users to which you want apply either of the actions below.

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

- **Print Labels**
  If you have the DESI label printing application installed on the computer, this control offers a list of connected printers and transfers the information required to print labels to the selected machine.

Button Programming (Non-Graphic Mode)

This table is shown in the non-graphic mode of operation. It displays the list of features programmed on each of the possible buttons.

- **Button**
  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 displays a menu of available button features. See Programming Features 49 in the next section.

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

Button Programming (Graphic Mode)

When a image of the telephone type is displayed, button programming is performed by clicking on the image.

- **To select the button function:**
  Right-click on the button and select Assign a Feature from the drop down menu. This displays a menu of available button features. See Programming Features 49 in the next section. If the feature requires action data, the menu prompts you to select or enter the data after selecting the feature.

- **To change the button label:**
  By default the button label is set automatically to match the selected button feature. To change the label, click on the label area next to the button.

Other Controls

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

- **Print Label for this Extension**
  If you have the DESI label printing application installed on the computer, this control transfers the information required to print labels for the current user.

3.3.1.1 Programming Features

This Set Button Programming Information menu allows a range of individual functions to be assigned to the button.

Making Calls

- **Auto Dial - Outside** 120: *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 48 setting.
- **Auto Dial - ICM** 119: *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** 119: *Action Data =User extension number.*
  A button set to this function can be used to page the configured extension.
- **Group Calling - Page** 122: *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 57.
- **Group Calling - Ring** 122: *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** 125: *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 57.
- **Group Hunting - Ring** 125: *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** 125: *Action Data = None.*
  A button set to this function redials the last outgoing external number dialed by the user.
- **Loud Speaker Paging** 125: *Action Data = None*
  A button set to this functions makes a page call to the system's designated loudspeaker extension port.
- **Save Number Redial** 127: *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** 127: *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** 125: *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. (Not applicable to Norstar systems.)

Answering Calls

- **Call Log** 122: *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** 38.
- **Call Pickup** 122: *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** 122: *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** 122: *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** 123: *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** 119: *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**[125]: *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 ***8LL*** where ***LL*** is the line number.

- **Pickup Group**[126]: *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**[128]: *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**[119]: *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**[124]: *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. (Not applicable to Norstar systems.)

- **Contact Closure 1**[124]/**Contact Closure 2**[124]: *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**[104]. 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**[124]: *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**[126]: *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**[127]: *Action Data = None.*
A button set to this function allows the user to send a recall or hook flash signal.

- **Station Lock**[127]: *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**[127]: *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:**[119] *Action Data = None.*
A button set to this function indicates when a call has been made using a number in the system's Emergency Number List[41]. Pressing the button displays a list of such calls.

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

Coverage

- **Call Coverage**[121]: *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**[121]: *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**[128]: *Action Data = None.*
A button set to this function allows the user to turn voicemail coverage of their calls on or off.

Messaging

- **Absent Message**[119]: *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** 125**:** *Action Data = None.*
  A button set to this function allows the user to access the voicemail to collect messages.
- **Message Alert Notification** 126**:** 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.

3.3.1.2 System Programming Features

The System Programming Feature 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** 37 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. When night service is on, use and behavior of VMS on some trunks may change depending on the trunk configuration. Also when night service is on, users in the night service group 57 must first use the **System Password** to make outgoing external calls other than emergency calls. If the user has this feature enabled, removing this button will turn the feature off.

- **Wake Up Service:** *Action Data = None, Software level = 6.1*
  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 hear repeated 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.

3.3.1.3 Line Assignment

The Line Assignment tab 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 System 38 mode, buttons can also be selected for ARS selector group 100 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 System*** 38 mode, a number of each users programmable buttons are automatically configured as line appearance button according to the system **Number of Lines** 38 setting. If the system setting **Number of Lines** 38 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 System*** 38 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 three rings (15 seconds).

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

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

3.3.2 Advanced Settings

This menu cannot be accessed from the **System** 33 page.
This menu is accessed from the **Admin Tasks** 34 list by selecting **User Setup | Advanced Settings**.

The Advanced Settings menu is used to configure user settings.

- **User Selection** - **Select User**
  This drop down list is used to select the user whose settings are displayed for editing.

- **Base Card # / Expansion Module #**
  This value indicates the control unit base card or external expansion module of the user's extension port. The 4 possible base cards are numbered 1 to 4 from left to right when facing the control unit. The type of base card port is also indicated: ***BP*** indicates an analog phone extension port, ***BD*** indicates a digital (DS or TCM) port.

- **Port**
  This value indicates the port number on the control unit base card or the external expansion module.

Advanced Parameters

- **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). On Norstar systems, default = 4 (20 seconds).*
  Programmable buttons set to **Call Coverage** 121 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.

- **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.

- **Transfer Return Extension:** *Default = None, Software level = 6.1+.*
  Set the destination for transferred calls that ring unanswered for longer than the **Transfer Return Ring** 110 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.*
  If **Automatic VMS Cover** above is assigned, this value sets how long a call alerts the user's extension before it is redirected to voicemail.

  - For Release 6.1+, 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. On Norstar systems, default = On*
  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** 49 feature.

- **Override Line Ringing:** *Default = Off. Software level = 6.1+.*
  For each line, unique line ringing settings 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.

Voicemail Settings

The Automatic VMS Cover and VMS Cover Ring settings above control whether and when voicemail is used to answer calls. The settings below control other aspects of voicemail operation for the user.

- **Voicemail Code:** *Default = Blank. Range = Blank or 1 to 15 digits.*
  This code is used to control access to the mailbox to collect messages. The mailbox user can change the code after they enter the mailbox by dialing *04.

- **Voicemail Email:** *Default = Blank.*
  When the user has a new message they can be emailed with an alert or a copy of the message, see **Voicemail Email Mode** below. 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 110.

- **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 *0/0):** *Default = Blank.*
    Sets the number to which a caller is transferred if they press ***0*** (Intuity mailbox mode) or ****0*** (IP Office mailbox mode) while listening to the mailbox greeting.

  - **Breakout (DTMF 2):** *Default = Blank.*
    Sets the number to which a caller is transferred if they press ***2*** (Intuity mailbox mode) or ****2*** (IP Office mailbox mode) while listening to the mailbox greeting.

  - **Breakout (DTMF 3):** *Default = Blank.*
    Sets the number to which a caller is transferred if they press ***3*** (Intuity mailbox mode) or ****3*** (IP Office mailbox mode) while listening to the mailbox greeting.

- **Voicemail Email:** *Default = Off.*
  This setting is used if an email address for the user has been set above and the system is configured with SMTP server settings 110. It sets whether the user receives an email when they have a new voicemail message and the type of 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 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 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.

Equipment Type

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

- ***Door Phone 1 / Door Phone 2***
  Select this option for an extension connected to a door phone. The phone system can support two such devices. The setting is linked to the **Assign Extension** setting on the **Door Phone 1** 103 and **Door Phone 2** 103 menus which set which users are alerted when the door phone goes off hook.

- ***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 17 can still be used for a range of functions such as voicemail. The setting cannot be changed.

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 Code Entries** 42 list when making an external call. This can only be overridden by use of the **System Password** 37 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 Number List** 41 and to numbers in any **Allowed Lists** 41 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.

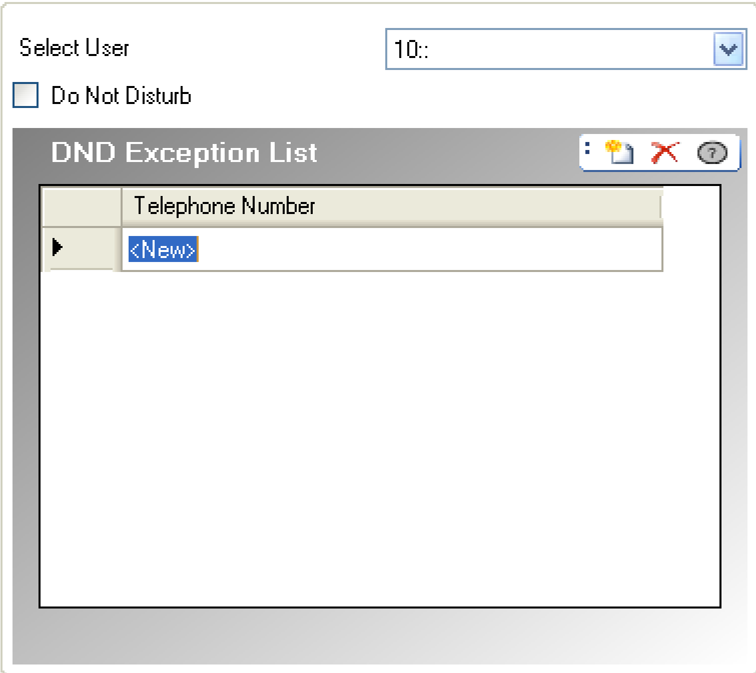
  - ***Local only***
    The user can only make outgoing external calls to numbers matching local numbers.

3.3.3 DND Exception List

This menu cannot be accessed from the **System** 33 page.
This menu is accessed from the **Admin Tasks** 34 list by selecting **User Setup | Advanced Settings**.

For Release 7.0, IP Office Manager can be used to see and edit users' do not disturb settings. Users themselves can switch do not disturb on/off using a programmed key 49 on their phone or an option in their phone's menus.

Do not disturb prevents the user from receiving hunt group and page calls. Direct callers hear busy tone or are diverted to voicemail if available. It overrides any call forwarding, follow me and call coverage settings. A set of exception numbers can be added to list numbers from which the user still wants to be able to receive calls when they have do not disturb enabled.



- **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.

3.4 Group Management

This menu is accessed from the **System** 33 page by selecting **Manage Hunt Groups**.
This menu is accessed from the **Admin Tasks** 34 list by selecting **Group Management**.

A hunt group is a collection of users accessible through a single directory number. Calls to that hunt group can be answered by any available member of the group. The order in which calls are presented can be adjusted by selecting different group types and adjusting the order in which group members are listed.

The **Group Management** menu is used to configure which extensions are members of the different available groups. You can also indicate which groups a user uses through the **User Setup** 46 menu.

Group Category	Number	Ring Mode	Description
Hunt Groups	6	Rotary	Hunt groups are usable as the coverage destination for incoming external calls. Six hunt groups may be configured. Each extension can be a member of several hunt groups. For each external line, one of the hunt groups can be selected as the line's **Coverage Destination**.
Pickup Groups	4	Rotary	Users can be configured to pickup a call currently alerting any member of a pickup group. Four pickup groups can be configured.
Calling Groups	4	Ring All	Users can call or transfer calls to a calling group. Four calling groups can be configured. Calling Group 1 is used by the **Simultaneous Page** function.
Night Service Group	1	Ring All	When the phone system is set to night service mode, incoming external calls other than those routed by DDI are rerouted to the users in the night service group.
Operator Group	1	Ring All	This option is only available for systems with their **System Mode** set to ***PBX System***. By default the group contains the first extension on the system and is used as the default destination for DID calls. It can also be selected as the destination for incoming SIP calls. For PRI and BRI trunks it is fixed incoming destination for calls unless DID Mapping is applied to the call. (Not applicable to Partner systems.)

Hunt Groups Configured

The groups available on a system are not adjustable. This list is used to display the groups available and select which group is currently editable in the table below.

- **Name:** *Information only, not editable.*
- **Number:** *Information only, not editable.*
- **Ring Mode:** *Information only, not editable.*
  The ring mode of a group sets the order in which members of the group are used.
  - ***Rotary***
    The available group members are alerted one at a time in extension number sequence starting from the last rung member. Ringing calls are picked up in oldest first order.
  - ***Ring All***
    All the available group members are alerted at the same time.

Assign Users to Group

This table is used to select which extension users are members of the currently selected group.

Group Call Distribution

A line can be configured to present its incoming calls to one of the 6 hunt groups. The incoming calls hunts from one hunt group extension to the next using the same hunting algorithm as used for an intercom call to that hunt group extension number. The call rings with the outside call ringing pattern and the display shows caller ID information if any.

If the hunt group extension that is chosen to ring as part of the selection algorithm has a line appearance for the line, then the call alerts on the line appearance with the standard slow flashing green LED indicative of a ringing call for me. Line ringing options are overridden and the line always rings immediately. Any other extensions in the hunt group with the line appearance, that have not been selected as part of the algorithm, will show the slow flashing red LED indicative of a ringing call but not for me. In addition, any other extensions in the system with the line appearance but not part of the hunt group, will show the slow flashing red LED indication.

If the hunt group extension that is chosen to ring as part of the selection algorithm does not have a line appearance for the line, then the call alerts on an intercom button.

When the hunt group extension that is ringing answers the call, the green LED goes steady (red off) and all other extensions in the system with the line appearance transition to the green off/steady red LED indication.

After three rings the call shall hunt to the next available extension in the hunt group using the hunt algorithm. When the call hunts the previously alerting extension stops alerting and returns to the idle condition. If the call was ringing on a line appearance, the line appearance state changes to slow red flashing indicating that the call is ringing elsewhere. If the call had been ringing on an ICOM appearance, the intercom button appearance is idled.

At any time while the call is hunting from extension to extension, any extension in the system can answer the call by either touching the line appearance of the line, or using one of the pickup features (active line pickup, call pickup, group call pickup).

An outside call that hunts never goes to voicemail and will hunt until answered or abandoned.

Outside calls ringing into a hunt group to a targeted extension are eligible for internal forwarding that might be active at the targeted hunt group extension. The call will be forwarded to another extension and if unanswered, continues hunting away from the forward-to extension to the next hunt group extension. If forwarding to an outside number is active at the targeted hunt group extension, then the call is never forwarded and alerts the target normally.

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.

3.5 Trunks

This menu is accessed from the **System** 33 page by selecting **Update Trunk Configurations**.
This menu is accessed from the **Admin Tasks** 34 list by selecting **Trunks**.

The **Trunks** menu displays a list of the **Installed Trunks** (excluding SIP trunks 87). When you are setting up Trunk Channels, a **Back** option is displayed at the bottom of the ***Advanced Settings*** screen. It returns you to the previous menu so that you can select another trunk line.

⚠ During initial Trunks set-up it is advisable to click **Apply** and save your changes before continuing with another trunk or pressing **Back** in the ***Advanced Settings*** screen. This is because if **Cancel** is subsequently used, you will lose **all** changes since your last click of **Apply** in the current session, thus losing any setting already made for other trunks.

3.5.1 Analog Trunks

This menu is accessed from the **System** 33 page by selecting **Update Trunk Configurations**.
This menu is accessed from the **Admin Tasks** 34 list by selecting **Trunks**.

If a trunk with the **Line Type** of ***Analog Trunk*** is selected in the list of installed trunks, its settings are displayed below the list of installed trunks. IP Office Manager can be used to apply an existing trunk template 63 to an analog trunk.

- **Installed Trunks**
  This table displays information about the trunk cards installed in the phone system. Selecting a trunk in the list displays its trunk settings below the list.
  - **Line Number:** *Information only, not editable.*
  - **Line Type:** *Not Editable*
    This value indicates the type of trunk. The menu fields and sub-menus will vary depending on the **Line Type**.
  - **Line Subtype**
    This option is not used for analog trunks.
  - **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**
    The number of channels supported by a trunk depends on the **Line Type** and **Line Subtype**. Each channels can be used for a separate external call (incoming or outgoing) and can be represented by a line appearance button.
    - For analog trunks, each trunk supports just one call (one channel).
- **Advanced Setup**
  This hot link option calls up a further window that is used to display and edit additional settings for the selected trunk and its trunk channels.

Analog Trunk Setup

- **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.
- **Coverage Destination:** *Default = None. System Mode = Key System*
  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** 57 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**
  IP Office Release 6.1+ supports phantom extensions 17. One of these can be selected as the destination for calls.

- **Hunt Group**
  Incoming calls can be routed to one of the 6 rotary hunt groups 57.

- **Calling Group**
  For systems with their **System Mode** 38 set to ***PBX System***, incoming calls can be routed to one of the 4 collective calling groups 57.

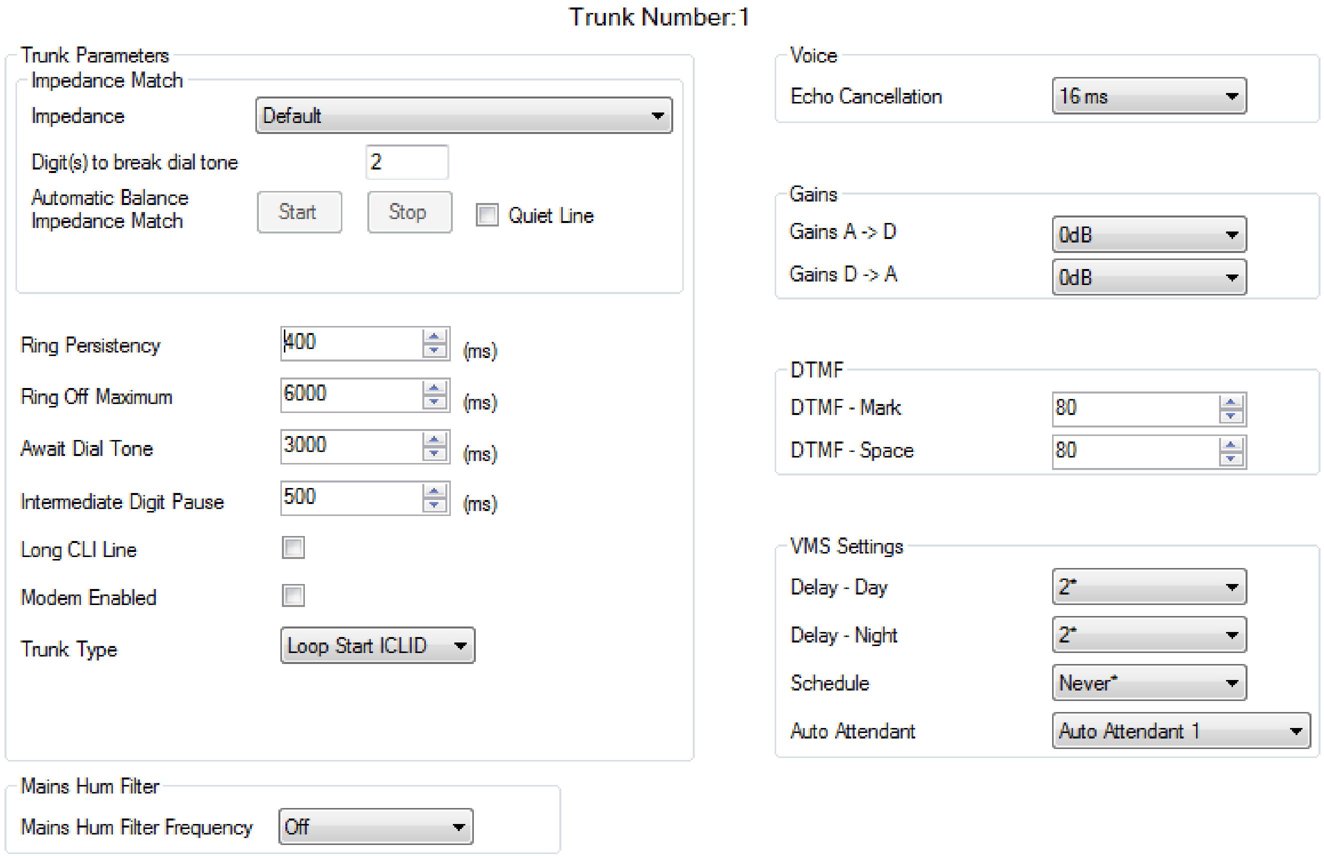
- **Operator Group**
  For systems with their **System Mode** 38 set to ***PBX System***, incoming calls are routed to the **Operator Group** 57.

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

- **Unique Line Ringing:** *Default = 1. Software level = 6.1+.*
  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.

3.5.1.1 Analog Advanced Setup

This menu is accessed from the **System** 33 page by selecting **Update Trunk Configurations**.
This menu is accessed from the **Admin Tasks** 34 list by selecting **Trunks**.



Trunk Parameters

- **Impedance Match**
  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 8.0+ they are also available for ***Canada***.

  - **Impedance:** *Default = Default*
    Set the impedance used for the line. The settings vary depending on the system's **Country** setting.

- **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.

  - **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.

- **Modem Enabled:** *Default = Off*
  The first analog trunk can be set to modem operation 18 (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. For Release 6.1 and higher, the modem feature can be accessed via an auto attendant or DID/SIP URI by selecting 76 as the destination.

- **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***.

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.

VMS Settings

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

- **VMS Delay - Night:** *Default = 2. Range = 0 to 6 (number of rings).*
  Sets the number of rings before an unanswered call should be redirected to an 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 an auto attendant. The options are:

  - ***Always***
    Redirect calls when the system is in both day and night service 16 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. Software Level = 6.1+.*
  This field allows selection of which auto attendant is used by this line.

3.5.1.2 Analog Trunk Templates

IP Office Manager can be used to import trunk settings from a template. If you have multiple system using the same provider, this may simplify configuration and maintenance of the systems.

- Trunk templates are used by different types of IP Office system. Those template settings not supported by an Basic Edition system are ignored.

Importing Templates

Templates must be placed in the correct Manager **\Templates** sub-folder. This can be done using the following command:

1. Select **Tools | Import Templates in Manager**.
2. Browse to the current folder containing the templates that you want to import and select that folder.
3. Click **OK**.
4. Any template files in the folder will be copied to the correct Manager sub-folder.

Copying a Trunk Template

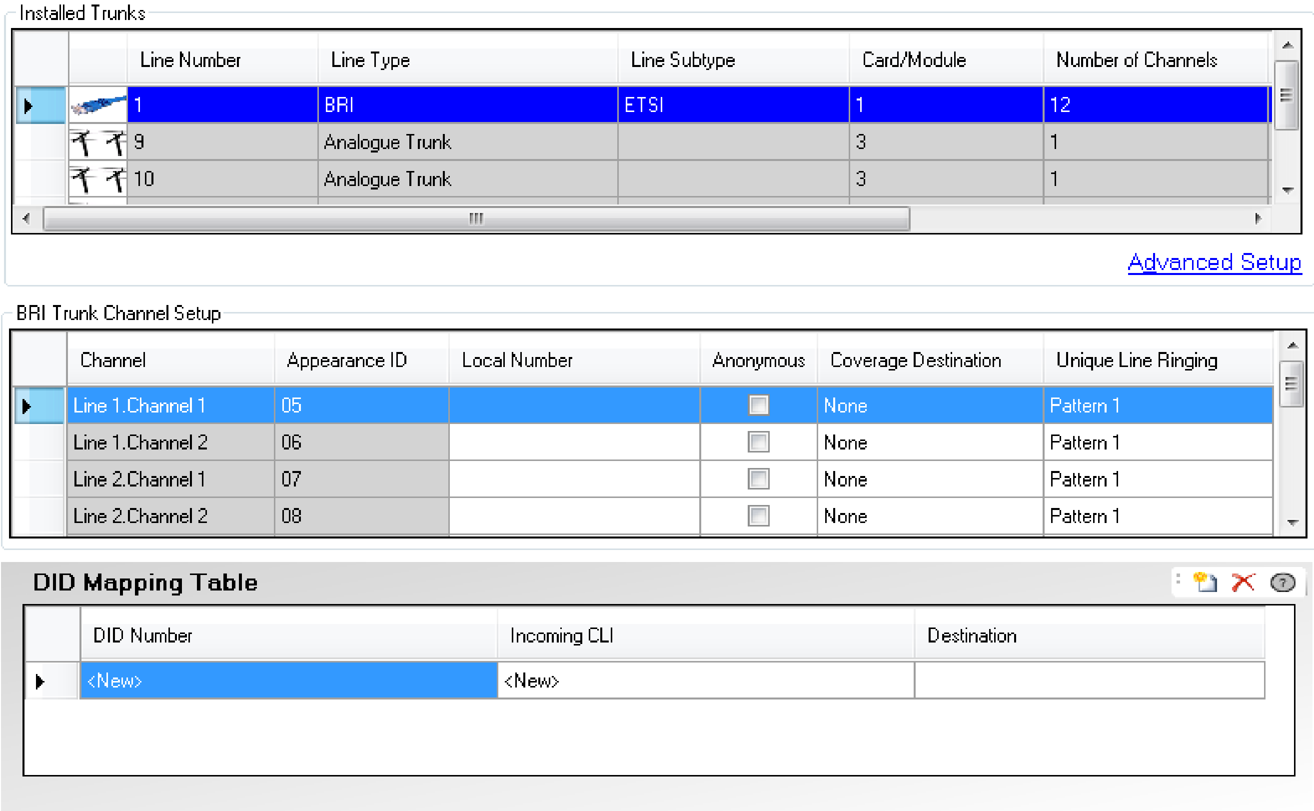
1. Select **Update Trunk Configurations** or in Admin Tasks, select **Trunks**.
2. Click on the button at the left hand of an analog trunk to select it. Then right click and select **Copy Settings from Template**.
3. Use the menu to select the template required.
4. Select the trunks to which you want the template applied.
5. Click **Copy Settings**.

3.5.2 BRI Trunk

This menu is accessed from the **System** 33 page by selecting **Update Trunk Configurations**.
This menu is accessed from the **Admin Tasks** 34 list by selecting **Trunks**.

BRI trunks are not available in North American locales.

- **ETSI PRI/BRI Trunks**
  In ***PBX System*** mode, all incoming call routing is done using the trunk's DID Mapping Table. The table includes a default non-editable entry that routes any calls for which there is no other match to the **Operator Group**.



BRI Trunk in Key Mode System

- **Installed Trunks**
  This table displays information about the trunk cards installed in the phone system. Selecting a trunk in the list displays its trunk settings below the list.
  - **Line Number:** *Information only, not editable.*
  - **Line Type:** *Not Editable*
    This value indicates the type of trunk. The menu fields and sub-menus will vary depending on the **Line Type**.
  - **Line Subtype**
    This option is fixed to ***ETSI*** for BRI trunks.
  - **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**
    The number of channels supported by a trunk depends on the **Line Type** and **Line Subtype**. Each channels can be used for a separate external call (incoming or outgoing) and can be represented by a line appearance button.
    - For a BRI card, 2 channels are supported for each physical connector (2 or 4) provided by the BRI trunk card.
  - **Send original calling party for forwarded calls:** *Default = Off. (Release 9.0.3)*
    Use the original calling party ID when forwarding calls.
  - **Originator number for forwarded calls:** *Default = blank. (Release 9.0.3)*
    The number used as the calling party ID when forwarding calls or routing twinned calls. This field is grayed out when the **Send original calling party for forwarded calls** setting is enabled.

- **Advanced Setup** 68
  This hot link option calls up a further window that is used to display and edit additional settings for the selected trunk and its trunk channels.

BRI Trunk Channel Setup

- **Channel:** *For information only, not editable. Not that this indicates the maximum number of channels, not the number of licensed channels usable.*
- **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 any associated number for test calls to the line.
  - **Anonymous:** *Default = Off*
    If selected, withhold sending caller ID information on outgoing calls.
- **Coverage Destination:** *Default = None. System Mode = Key System*
  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** 57 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**
    IP Office Release 6.1+ supports phantom extensions 17. One of these can be selected as the destination for calls.
  - **Hunt Group**
    Incoming calls can be routed to one of the 6 rotary hunt groups 57.
  - **Calling Group**
    For systems with their **System Mode** 38 set to ***PBX System***, incoming calls can be routed to one of the 4 collective calling groups 57.
  - **Operator Group**
    For systems with their **System Mode** 38 set to ***PBX System***, incoming calls are routed to the **Operator Group** 57.
  - **Voicemail**
    Route incoming calls to the systems voicemail to collect messages. This requires the caller to know the mailbox number and passcode.
- **Unique Line Ringing:** *Default = 1. Software level = 6.1+.*
  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.

DID Mapping Table

This table is used to set the destination for incoming calls that include DID digits. These are routed by matching the DID and ICLID information received with the call to an entry in the table. This overrides the **Coverage Destination** settings of the channel on which the call was received. Calls routing by DID mapping are not affected by the phone system being put into night service.

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.

The table is applied to all channels.

- **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 **System Mode** 38 is set to ***Key System*** or ***PBX System***.

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

  - **Phantom Extension**
    IP Office Release 6.1+ supports phantom extensions 17. One of these can be selected as the destination for calls.

  - **Hunt Group**
    Incoming calls can be routed to one of the 6 rotary hunt groups 57.

  - **Calling Group**
    For systems with their **System Mode** 38 set to ***PBX System***, incoming calls can be routed to one of the 4 collective calling groups 57.

  - **Operator Group**
    For systems with their **System Mode** 38 set to ***PBX System***, incoming calls are routed to the **Operator Group** 57.

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

  - **76: Modem**
    For Release 6.1+, the option ***76: Modem*** can be selected to route the call to the systems built in V32 modem 18 function. This is intended for basic configuration access by system maintainers.

  - **Auto Attendant**
    For Release 6.1+, any of the configured voicemail auto attendants can be selected as the call destination.

3.5.2.1 BRI Advanced Setup

This menu is accessed from the **System** 33 page by selecting **Update Trunk Configurations**.
This menu is accessed from the **Admin Tasks** 34 list by selecting **Trunks**.

- **Line 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.

3.5.3 PRI Trunks

PRI trunks can be set to a number of different line subtypes.

For Quick systems: ***PRI***, ***T1*** and ***ETSI***.

For Partner systems: ***PRI*** *or* ***T1***.

For Norstar systems: ***ETSI***.

- **PRI** 74
  Available for Canada, Mexico and United States. Supports up to 23 channels.
- **T1** 81
  Available for Canada, Mexico and United States. Supports up to 24 channels.
- **ETSI** 70
  Available for countries other than Canada, Mexico and United States. Supports up to 30 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.

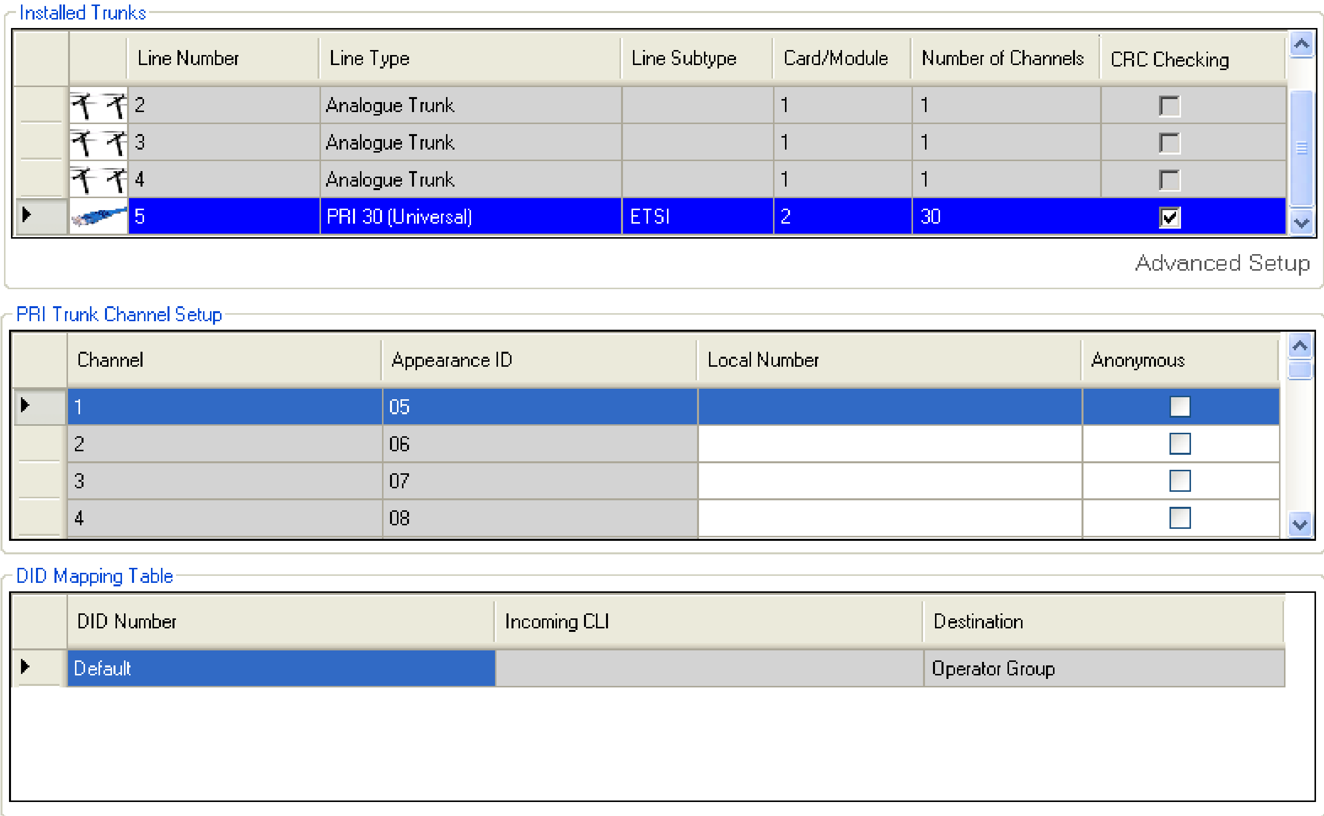
3.5.3.1 ETSI PRI Trunk

This menu is accessed from the **System** 33 page by selecting **Update Trunk Configurations**.
This menu is accessed from the **Admin Tasks** 34 list by selecting **Trunks**.

If a PRI trunk with the **Line Subtype** of ***ETSI*** is selected in the list of installed trunks, its settings are displayed below the list of installed trunks.

- **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.

- **ETSI PRI/BRI Trunks**
  In ***PBX System*** mode, all incoming call routing is done using the trunk's DID Mapping Table. The table includes a default non-editable entry that routes any calls for which there is no other match to the **Operator Group**.



- **Installed Trunks**
  This table displays information about the trunk cards installed in the phone system. Selecting a trunk in the list displays its trunk settings below the list.
  - **Line Number:** *Information only, not editable.*
  - **Line Type:** *Not Editable*
    This value indicates the type of trunk. The menu fields and sub-menus will vary depending on the **Line Type**.
  - **Line Subtype**
    For non-North American locales, the **Line Subtype** of PRI trunks is ***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**
    The number of channels supported by a trunk depends on the **Line Type** and **Line Subtype**. Each channels can be used for a separate external call (incoming or outgoing) and can be represented by a line appearance button.
    - For a PRI card (***ETSI***), up to 30 channels are supported. The number of channels should be set to match the number supported by the line provider.
  - **CRC Checking:** *Default = On*
    This setting is only used with ETSI E1 PRI trunks. Switches CRC on or off.

- **Send original calling party for forwarded calls:** *Default = Off. (Release 9.0.3)*
  Use the original calling party ID when forwarding calls.

- **Originator number for forwarded calls:** *Default = blank. (Release 9.0.3)*
  The number used as the calling party ID when forwarding calls or routing twinned calls. This field is grayed out when the **Send original calling party for forwarded calls** setting is enabled.

- **Advanced Setup**
  This option is not used for ***ETSI*** trunks.

PRI Trunk Channel Setup

- **Channel:** *For information only, not editable. Not that this indicates the maximum number of channels, not the number of licensed channels usable.*
- **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 any associated number for test calls to the line.
  - **Anonymous:** *Default = Off*
    If selected, withhold sending caller ID information on outgoing calls. For PBX Mode systems this may also be invoked or overridden by the ARS selector used to route the call.
- **Coverage Destination:** *Default = None. System Mode = Key System*
  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** 57 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**
    IP Office Release 6.1+ supports phantom extensions 17. One of these can be selected as the destination for calls.
  - **Hunt Group**
    Incoming calls can be routed to one of the 6 rotary hunt groups 57.
  - **Calling Group**
    For systems with their **System Mode** 38 set to ***PBX System***, incoming calls can be routed to one of the 4 collective calling groups 57.
  - **Operator Group**
    For systems with their **System Mode** 38 set to ***PBX System***, incoming calls are routed to the **Operator Group** 57.
  - **Voicemail**
    Route incoming calls to the systems voicemail to collect messages. This requires the caller to know the mailbox number and passcode.
- **Unique Line Ringing:** *Default = 1. Software level = 6.1+.*
  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.

DID Mapping Table

This table is used to set the destination for incoming calls that include DID digits. These are routed by matching the DID and ICLID information received with the call to an entry in the table. This overrides the **Coverage Destination** settings of the channel on which the call was received. Calls routing by DID mapping are not affected by the phone system being put into night service.

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.

The table is applied to all channels.

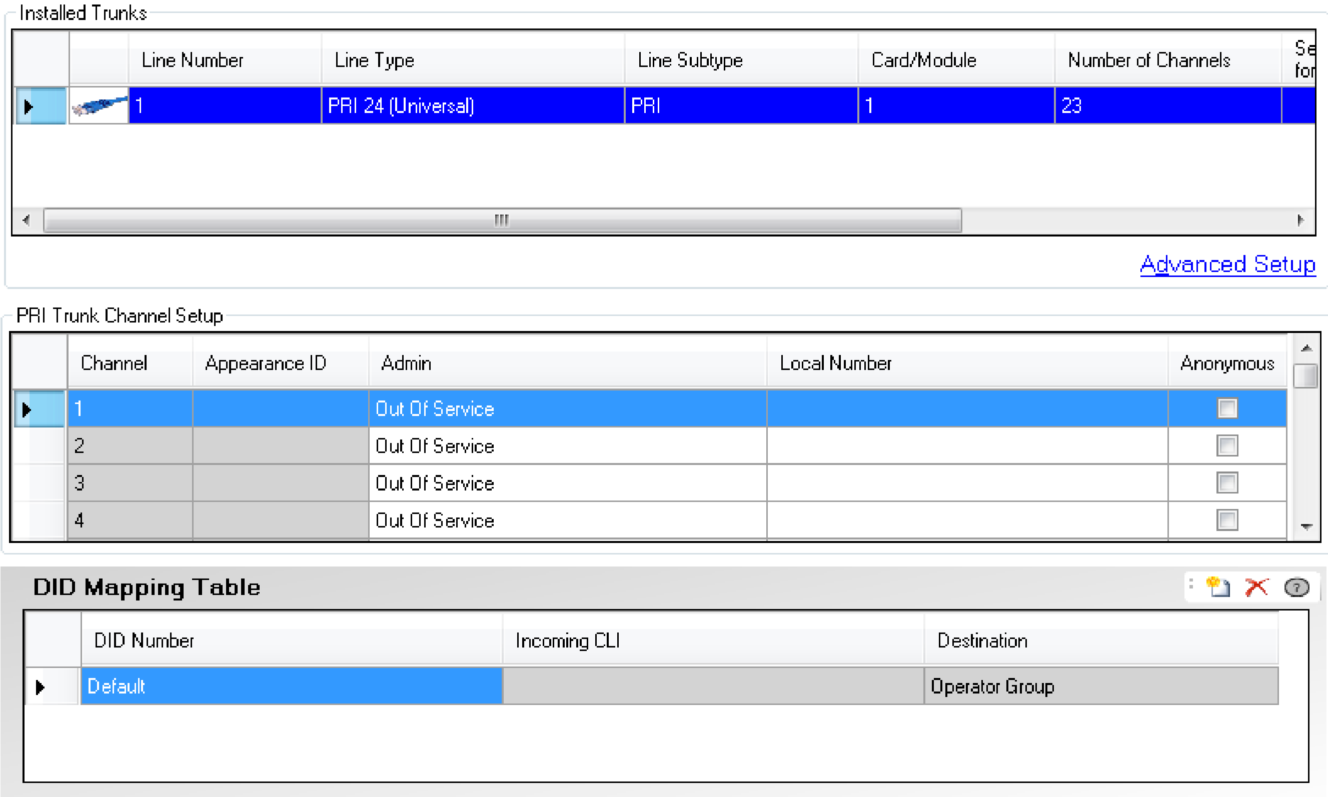
- **DID Number**
  If the incoming DID of a call on the trunk matches the DID set here, it will be routed to this destination. The system supports up to 4 digits DID (additional digits after the first 4 are ignored). Leave blank if only CLI matching is required.
- **Incoming CLI**
  If the incoming caller 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 **System Mode** 38 is set to ***Key System*** or ***PBX System***.
  - **Extension**
    Route incoming calls to a particular extension.
  - **Phantom Extension**
    IP Office Release 6.1+ supports phantom extensions 17. One of these can be selected as the destination for calls.
  - **Hunt Group**
    Incoming calls can be routed to one of the 6 rotary hunt groups 57.
  - **Calling Group**
    For systems with their **System Mode** 38 set to ***PBX System***, incoming calls can be routed to one of the 4 collective calling groups 57.
  - **Operator Group**
    For systems with their **System Mode** 38 set to ***PBX System***, incoming calls are routed to the **Operator Group** 57.
  - **Voicemail**
    Route incoming calls to the systems voicemail to collect messages. This requires the caller to know the mailbox number and passcode.
  - **76: Modem**
    For Release 6.1+, the option ***76: Modem*** can be selected to route the call to the systems built in V32 modem 18 function. This is intended for basic configuration access by system maintainers.
  - **Auto Attendant**
    For Release 6.1+, any of the configured voicemail auto attendants can be selected as the call destination.

3.5.3.2 PRI Trunks

This menu is accessed from the **System** 33 page by selecting **Update Trunk Configurations**.
This menu is accessed from the **Admin Tasks** 34 list by selecting **Trunks**.

If a PRI trunk with the **Line Subtype** of ***PRI*** is selected in the list of installed trunks, its settings are displayed below the list of installed trunks. Clicking on **Advanced Setup** 77 when a PRI line type is selected, accesses a menu of additional settings for the trunk and settings for the trunks individual 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.



- **Installed Trunks**
  This table displays information about the trunk cards installed in the phone system. Selecting a trunk in the list displays its trunk settings below the list.

  - **Line Number:** *Information only, not editable.*
  - **Line Type:** *Not Editable*
    This value indicates the type of trunk. The menu fields and sub-menus will vary depending on the **Line Type**.
  - **Line Subtype**
    For North American locales, the **Line Subtype** of PRI trunks is set to either ***PRI*** or ***T1***. The setting used should match the service supported by the line provider.
  - **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**
    The number of channels supported by a trunk depends on the **Line Type** and **Line Subtype**. Each channels can be used for a separate external call (incoming or outgoing) and can be represented by a line appearance button.
    - For a PRI card, the number of channels depends on the **Line Subtype**. For a ***PRI*** trunk, 23 channels are supported, for a ***T1*** trunk, 24 channels are supported.
  - **Send original calling party for forwarded calls:** *Default = Off. (Release 9.0.3)*
    Use the original calling party ID when forwarding calls.
  - **Originator number for forwarded calls:** *Default = blank. (Release 9.0.3)*
    The number used as the calling party ID when forwarding calls or routing twinned calls. This field is grayed out when the **Send original calling party for forwarded calls** setting is enabled.

- **Advanced Setup**
  This is used to access features that should only be adjusted to match the requirements of the line provider.

PRI Trunk Channel Setup

- **Admin:** *Default = Out of Service*
  Options are ***In Service***, ***DID Only***, ***Maintenance*** and ***Out of Service***.

- **Channel:** *For information only, not editable. Not that this indicates the maximum number of channels, not the number of licensed channels usable.*

- **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 any associated number for test calls to the line.

  - **Anonymous:** *Default = Off*
    If selected, withhold sending caller ID information on outgoing calls.

- **Coverage Destination:** *Default = None. System Mode = Key System*
  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** 57 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**
    IP Office Release 6.1+ supports phantom extensions 17. One of these can be selected as the destination for calls.

  - **Hunt Group**
    Incoming calls can be routed to one of the 6 rotary hunt groups 57.

  - **Calling Group**
    For systems with their **System Mode** 38 set to ***PBX System***, incoming calls can be routed to one of the 4 collective calling groups 57.

  - **Operator Group**
    For systems with their **System Mode** 38 set to ***PBX System***, incoming calls are routed to the **Operator Group** 57.

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

- **Unique Line Ringing:** *Default = 1. Software level = 6.1+.*
  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.

DID Mapping Table

This table is used to set the destination for incoming calls that include DID digits. These are routed by matching the DID and ICLID information received with the call to an entry in the table. This overrides the **Coverage Destination** settings of the channel on which the call was received. Calls routing by DID mapping are not affected by the phone system being put into night service.

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.

The table is applied to all channels.

- **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 **System Mode** 38 is set to ***Key System*** or ***PBX System***.

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

  - **Phantom Extension**
    IP Office Release 6.1+ supports phantom extensions 17. One of these can be selected as the destination for calls.

  - **Hunt Group**
    Incoming calls can be routed to one of the 6 rotary hunt groups 57.

  - **Calling Group**
    For systems with their **System Mode** 38 set to ***PBX System***, incoming calls can be routed to one of the 4 collective calling groups 57.

  - **Operator Group**
    For systems with their **System Mode** 38 set to ***PBX System***, incoming calls are routed to the **Operator Group** 57.

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

  - **76: Modem**
    For Release 6.1+, the option ***76: Modem*** can be selected to route the call to the systems built in V32 modem 18 function. This is intended for basic configuration access by system maintainers.

  - **Auto Attendant**
    For Release 6.1+, any of the configured voicemail auto attendants can be selected as the call destination.

3.5.3.2.1 Details

This menu is accessed from the **System** 33 page by selecting **Update Trunk Configurations**.
This menu is accessed from the **Admin Tasks** 34 list by selecting **Trunks**.

This menu allows setting of advanced trunk settings that normally do not need to be changed. The **Channel Setup** 78 option give access to a menu for configuring individual channels.

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 79 menu can be accessed from 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***.

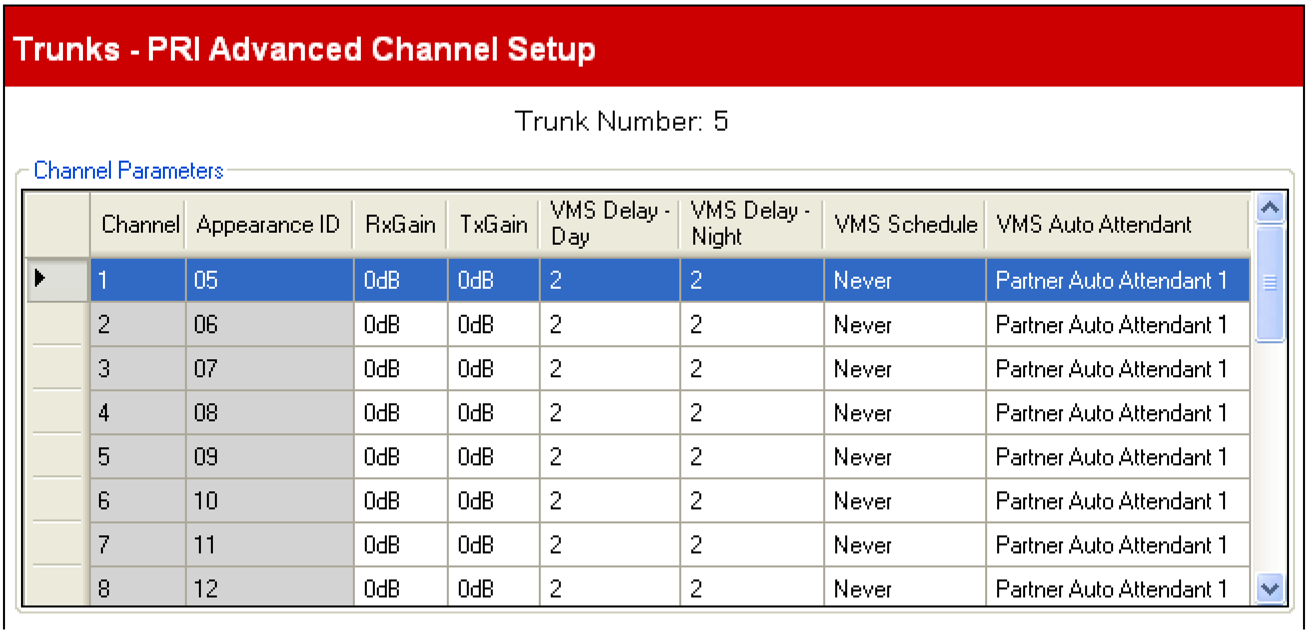
Dial Plan

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

This menu is accessed from the **System** 33 page by selecting **Update Trunk Configurations**.
This menu is accessed from the **Admin Tasks** 34 list by selecting **Trunks**.

This menu allows the adjustment of settings for each channel of the PRI trunk.



Channel Parameters

- **Channel:** *For information only, not editable. Not that this indicates the maximum number of channels, not the number of licensed channels usable.*

- **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.

- **Tx Gain:** *Default = 0dB*
  The transmit gain in dB.

- **Rx Gain:** *Default = 0dB*
  The receive gain in dB.

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

- **VMS Delay - Night:** *Default = 2. Range = 0 to 6 (number of rings).*
  Sets the number of rings before an unanswered call should be redirected to an 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 an auto attendant. The options are:

  - ***Always***
    Redirect calls when the system is in both day and night service 16 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.

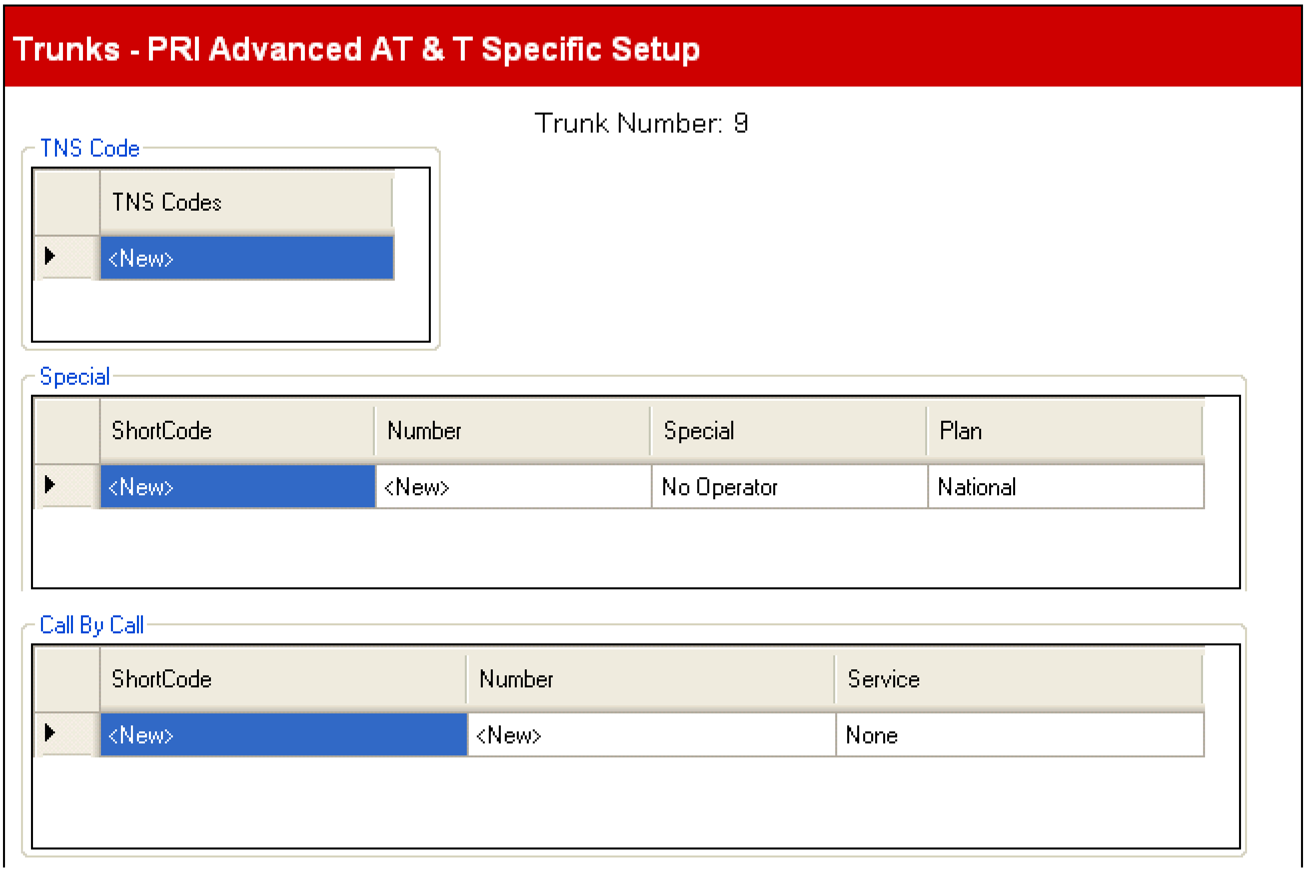
  - ***Never***
    Do not redirect calls.

- **Auto Attendant:** *Default = Auto Attendant 1. Software Level = 6.1+.*
  This field allows selection of which auto attendant is used by this line.

3.5.3.2.3 PRI Advanced AT&T Specific Setup

This menu is accessed from the **System** 33 page by selecting **Update Trunk Configurations**.
This menu is accessed from the **Admin Tasks** 34 list by selecting **Trunks**.

These settings are only available for a PRI trunk where the **Provider** 77 has been set to ***AT&T***.



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 which results from the application of the rules specified in the User or System Short code tables and the Network Selection table and the Call-by-call table to 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.

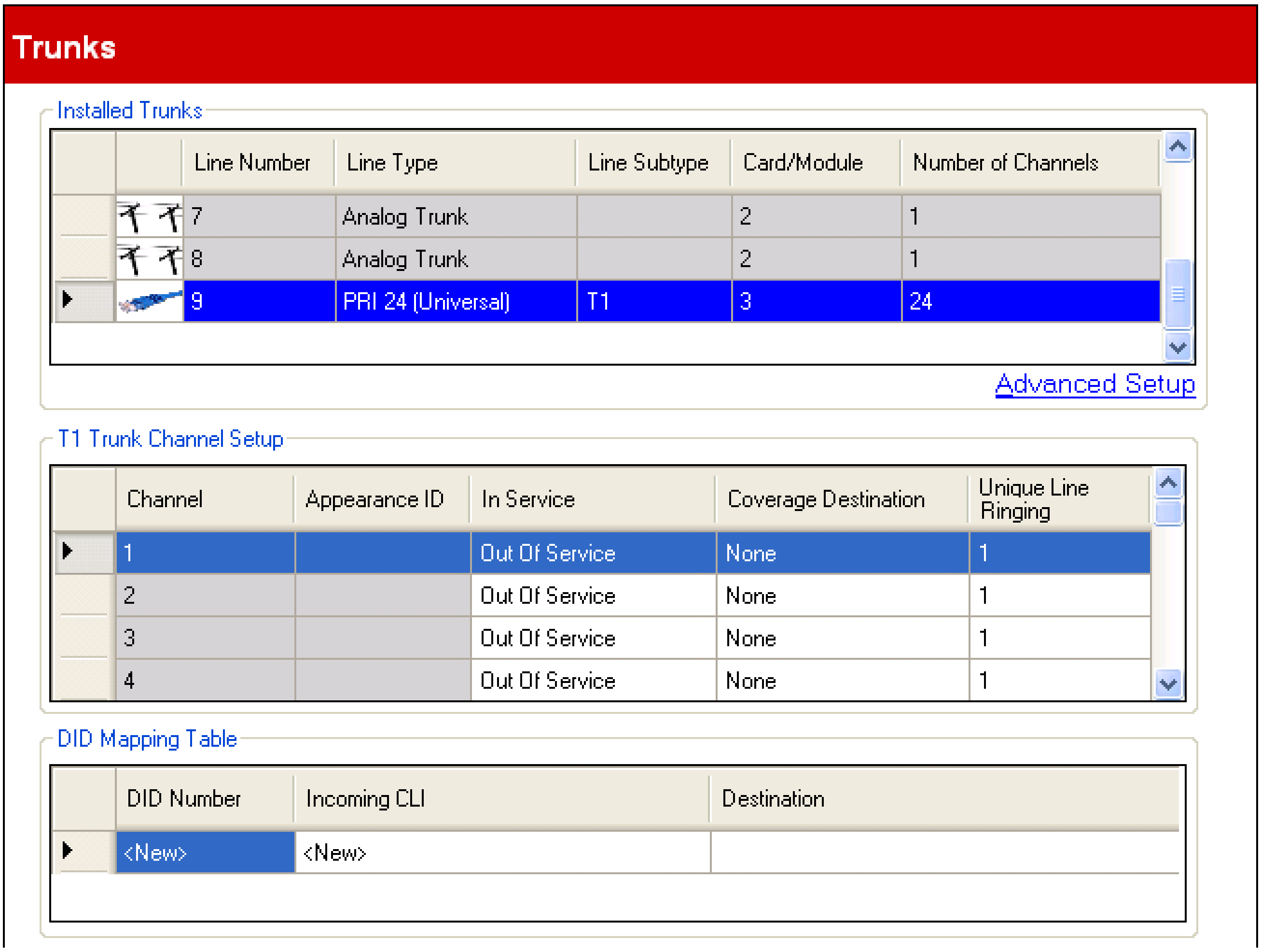
- **Short Code:**
  The number dialed.
- **Number:**
  The number to be dialed to line.
- **Service:** *Default = AT&T*
  The service required by the call from ***SDN (inc GSDN)***, ***MegaCom800***, ***MegaCom***, ***Wats***, ***Accunet***, ***ILDS***, ***I800***, ***ETN***, ***Private Line*** or ***AT&T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

3.5.3.3 T1 Trunks

This menu is accessed from the **System** 33 page by selecting **Update Trunk Configurations**.
This menu is accessed from the **Admin Tasks** 34 list by selecting **Trunks**.

If a PRI trunk with the **Line Subtype** of ***T1*** is selected in the list of installed trunks, its settings are displayed below the list of installed trunks. Clicking on **Advanced Setup** 84 accesses a menu of additional settings for the trunk and settings for the trunk's individual 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.



- **Installed Trunks**
  This table displays information about the trunk cards installed in the phone system. Selecting a trunk in the list displays its trunk settings below the list.

  - **Line Number:** *Information only, not editable.*
  - **Line Type:** *Not Editable*
    This value indicates the type of trunk. The menu fields and sub-menus will vary depending on the **Line Type**.
  - **Line Subtype**
    For North American locales, the **Line Subtype** of PRI trunks is set to either ***PRI*** or ***T1***. The setting used should match the service supported by the line provider.
  - **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**
    The number of channels supported by a trunk depends on the **Line Type** and **Line Subtype**. Each channels can be used for a separate external call (incoming or outgoing) and can be represented by a line appearance button.

- For a PRI card, the number of channels depends on the **Line Subtype**. For a ***PRI*** trunk, 23 channels are supported, for a ***T1*** trunk, 24 channels are supported.

- **Advanced Setup**
This is used to access features that should only be adjusted to match the requirements of the line provider.

T1 Trunk Channel Setup

This table is used to set which trunk channels are available for use.

- **Channel:** *For information only, not editable. Not that this indicates the maximum number of channels, not the number of licensed channels usable.*

- **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.*
Selects whether the trunk channel is in use.

- **Coverage Destination:** *Default = None. System Mode = Key System*
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** 57 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**
  IP Office Release 6.1+ supports phantom extensions 17. One of these can be selected as the destination for calls.

  - **Hunt Group**
  Incoming calls can be routed to one of the 6 rotary hunt groups 57.

  - **Calling Group**
  For systems with their **System Mode** 38 set to ***PBX System***, incoming calls can be routed to one of the 4 collective calling groups 57.

  - **Operator Group**
  For systems with their **System Mode** 38 set to ***PBX System***, incoming calls are routed to the **Operator Group** 57.

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

- **Unique Line Ringing:** *Default = 1. Software level = 6.1+.*
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.

DID Mapping Table

This table is used to set the destination for incoming calls that include DID digits. These are routed by matching the DID and ICLID information received with the call to an entry in the table. This overrides the **Coverage Destination** settings of the channel on which the call was received. Calls routing by DID mapping are not affected by the phone system being put into night service.

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.

The table is applied to all channels.

- **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 **System Mode** 38 is set to ***Key System*** or ***PBX System***.

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

  - **Phantom Extension**
    IP Office Release 6.1+ supports phantom extensions 17. One of these can be selected as the destination for calls.

  - **Hunt Group**
    Incoming calls can be routed to one of the 6 rotary hunt groups 57.

  - **Calling Group**
    For systems with their **System Mode** 38 set to ***PBX System***, incoming calls can be routed to one of the 4 collective calling groups 57.

  - **Operator Group**
    For systems with their **System Mode** 38 set to ***PBX System***, incoming calls are routed to the **Operator Group** 57.

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

  - **76: Modem**
    For Release 6.1+, the option ***76: Modem*** can be selected to route the call to the systems built in V32 modem 18 function. This is intended for basic configuration access by system maintainers.

  - **Auto Attendant**
    For Release 6.1+, any of the configured voicemail auto attendants can be selected as the call destination.

This menu is accessed from the **System** 33 page by selecting **Update Trunk Configurations**.
This menu is accessed from the **Admin Tasks** 34 list by selecting **Trunks**.

This menu allows setting of advanced T1 trunk settings that normally do not need to be changed. The **Channel Setup** 85 option give access to a menu for configuring individual channels.

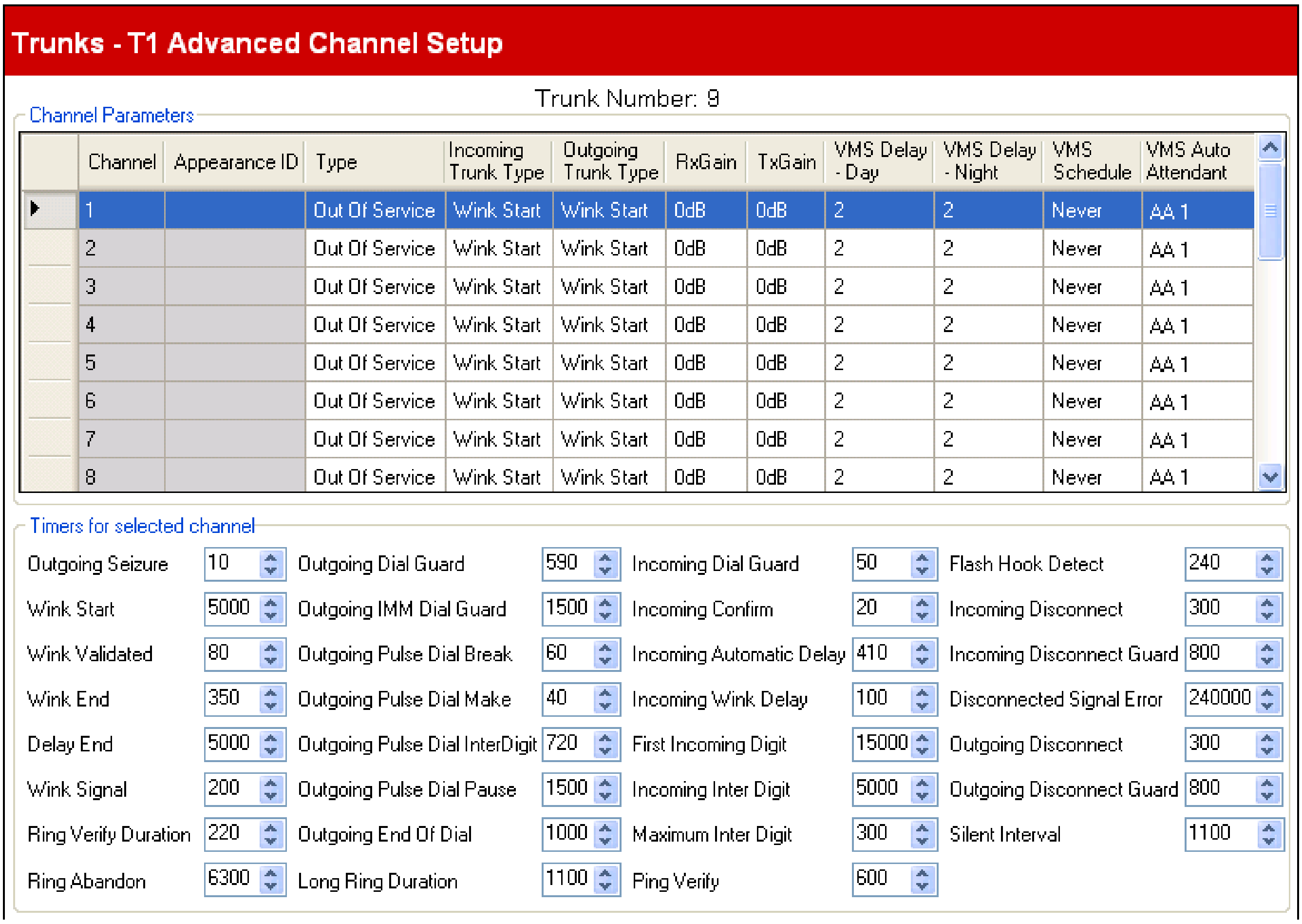
Trunk Parameters

- **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***.

3.5.3.3.2 T1 Advanced Channel Setup

This menu is accessed from the **System** 33 page by selecting **Update Trunk Configurations**.	
This menu is accessed from the **Admin Tasks** 34 list by selecting **Trunks**.	

This menu allows the adjustment of settings for each channel of the T1 trunk.



Channel Parameters

- **Channel:** *For information only, not editable. Not that this indicates the maximum number of channels, not the number of licensed channels usable.*

- **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.

- **Dial Type:** *Default = DTMF Dial*
  Select the dialing method required (***DTMF Dial*** or ***Pulse Dial***).

- **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***).
- **Tx Gain:** *Default = 0dB*
  The transmit gain in dB.

- **Rx Gain:** *Default = 0dB*
  The receive gain in dB.

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

- **VMS Delay - Night:** *Default = 2. Range = 0 to 6 (number of rings).*
  Sets the number of rings before an unanswered call should be redirected to an 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 an auto attendant. The options are:

  - ***Always***
    Redirect calls when the system is in both day and night service 16 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. Software Level = 6.1+.*
  This field allows selection of which auto attendant is used by this line.

Timers for selected channel

Only adjust these values under guidance from the line provider.

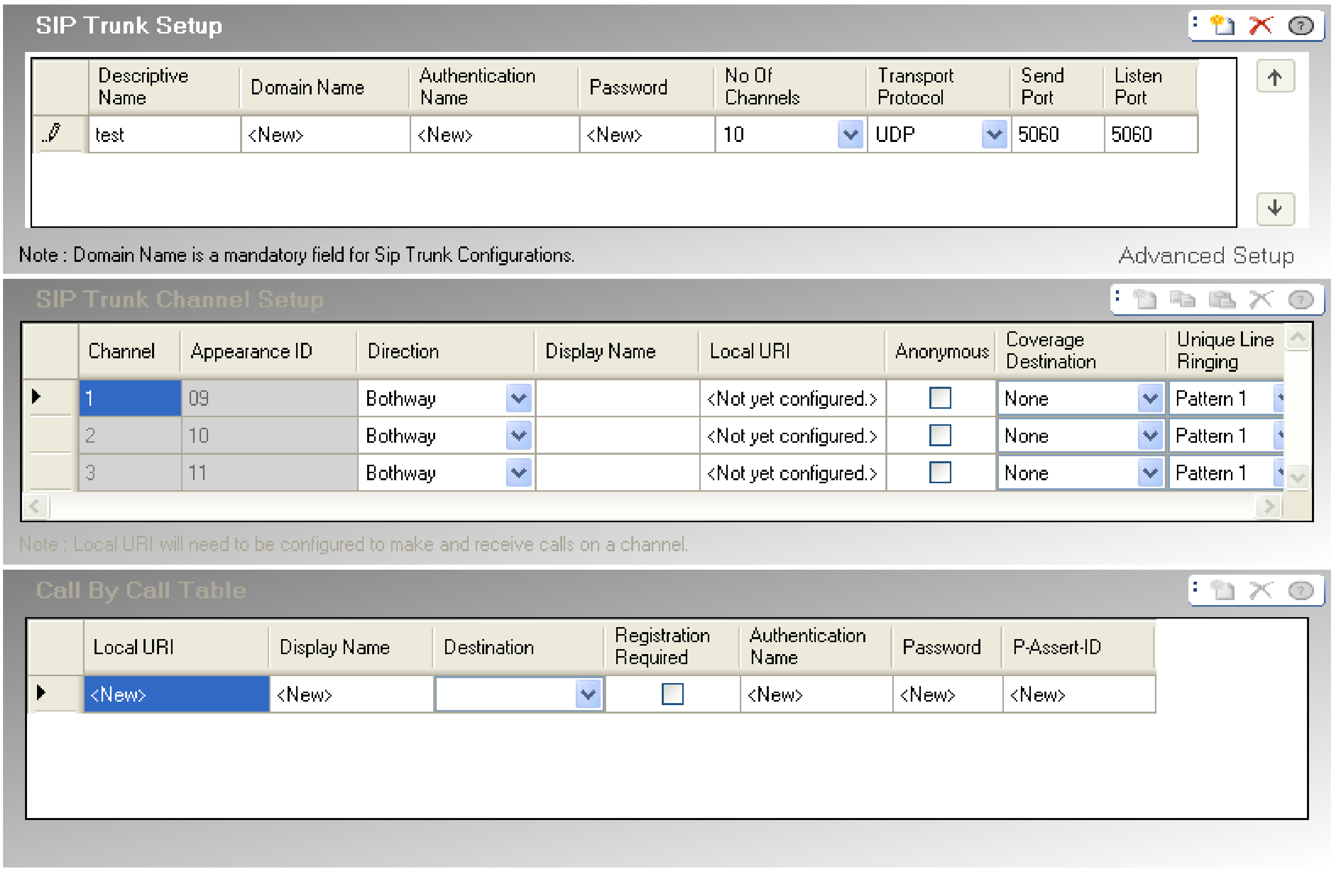
3.5.4 SIP Trunk Administration

This menu cannot be accessed from the **System** 33 page.
This menu is accessed from the **Admin Tasks** 34 list by selecting **Trunks | SIP Trunk Administration**.

This menu is used to add SIP trunks to the phone system configuration. Before adding any SIP trunks, the system must be configured for general SIP operation through the **STUN Settings for the Network** section of the **Advanced Parameters** 110 settings.

- **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 45 to the configuration.

  - **VCM Channels**
    Note that for SIP calls the system also requires VCM channels. For a Basic Edition system those are provided by installing IP500 Combination base cards. Each of these cards provides 10 VCM channels.



SIP Trunk Setup

- **Descriptive Name**
  The name of the trunk
- **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.
- **Authentication Name:** *Default = Blank.*
  This value is provided by the SIP ITSP.
- **Password:** *Default = Blank.*
  This value is provided by the SIP ITSP.
- **Number of Channels:** *Default = 10*
  Number of trunk channels between 1 and 24
- **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.
- **Advanced Setup**
  Clicking on **Advanced Setup** 61 when an SIP Trunk line type is selected in the list and a domain name has been entered, accesses a menu of additional settings.

SIP Trunk Channel Setup

- **Channel**
  Set by the system. Shown for information only.

- **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).

  - ***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 48. In addition, on PBX mode systems, outgoing calls can be routed to the channel by including the line appearance in the **ARS Selector** 99 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 **Unique Line Ringing** 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 **Unique Line Ringing** 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.

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

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

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

- **Coverage Destination:** *Default = None. System Mode = Key System*
  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** 57 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**
    IP Office Release 6.1+ supports phantom extensions 17. One of these can be selected as the destination for calls.

  - **Hunt Group**
    Incoming calls can be routed to one of the 6 rotary hunt groups 57.

  - **Calling Group**
    For systems with their **System Mode** 38 set to ***PBX System***, incoming calls can be routed to one of the 4 collective calling groups 57.

  - **Operator Group**
    For systems with their **System Mode** 38 set to ***PBX System***, incoming calls are routed to the **Operator Group** 57.

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

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

- **Unique Line Ringing:** *Default = 1. Software level = 6.1+.*
  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.

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

- **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.

- **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.

Call by Call Table

These settings are used to match calls received on SIP trunks channels set to ***Incoming Call-by-Call*** above. 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 System*** 6 mode systems. For those systems, each call-by-call entry can be assigned to an ARS Selector 100 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).

- **Display Name:** *Default = Use Authentication Name*
  This field sets the 'Name' value for SIP calls using this URI. The value can either be entered manually or the options ***Use Authentication Name*** or ***Use Internal Data*** selected.

- **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**
    IP Office Release 6.1+ supports phantom extensions 17. One of these can be selected as the destination for calls.

  - **Calling Group**
    For systems with their **System Mode** 38 set to ***PBX System***, incoming calls can be routed to one of the 4 collective calling groups 57.

  - **Operator Group**
    For systems with their **System Mode** 38 set to ***PBX System***, incoming calls are routed to the **Operator Group** 57.

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

  - **76: Modem**
    For Release 6.1+, the option ***76: Modem*** can be selected to route the call to the systems built in V32 modem 18 function. This is intended for basic configuration access by system maintainers.

  - **Auto Attendant**
    For Release 6.1+, any of the configured voicemail auto attendants can be selected as the call destination.

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

- **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.

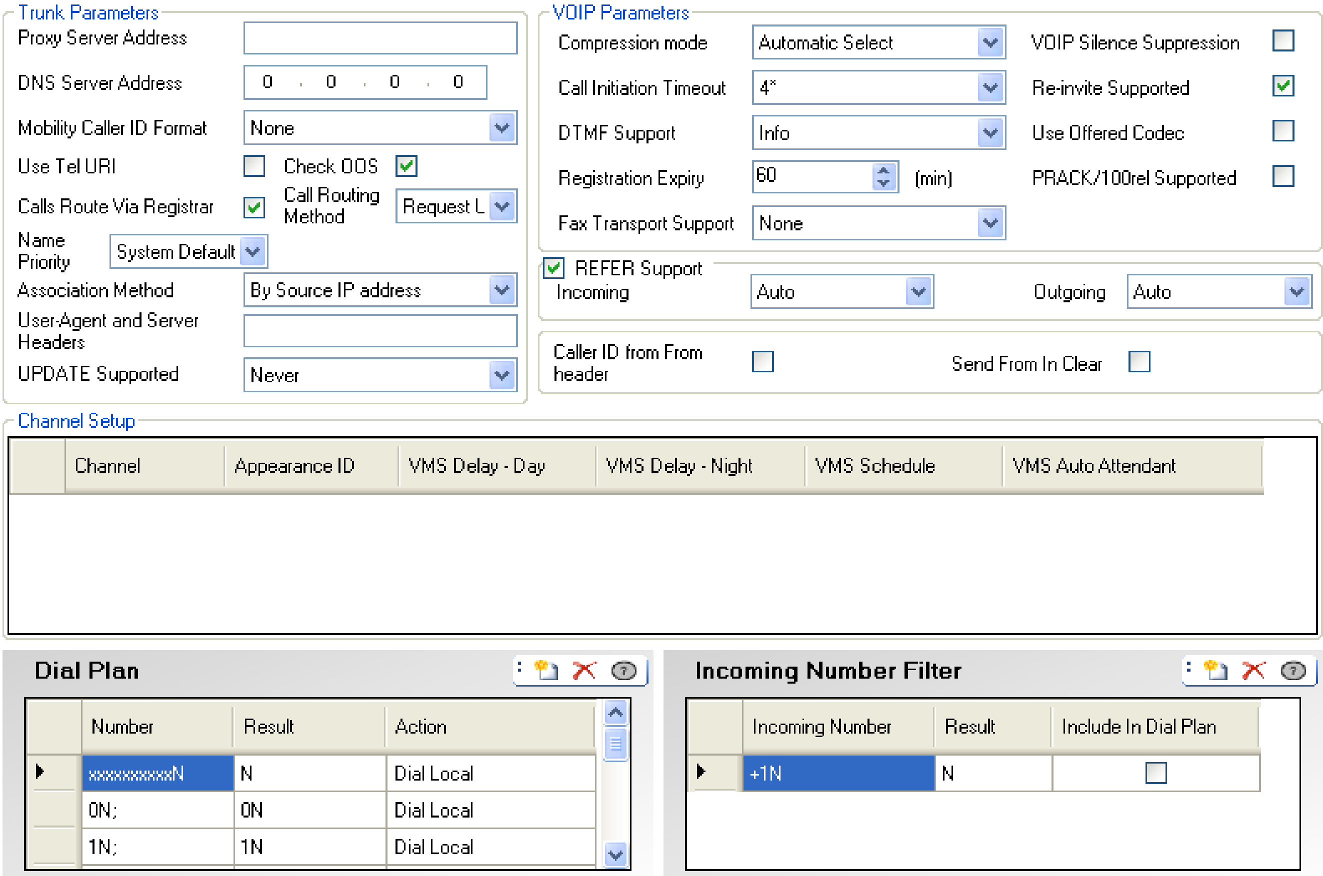
- **Password:** *Default = Blank.*
  This value is provided by the SIP ITSP.

- **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.

3.5.4.1 SIP Trunk Advanced

This menu cannot be accessed from the **System** 33 page.
This menu is accessed from the **Admin Tasks** 34 list by selecting **Trunks | SIP Trunk Administration | Advanced Setup**.

These settings are used for configuration of individual SIP channels and more advanced SIP trunk settings.



Trunk Parameters

- **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.

- **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 = Off.*
  Use Tel URI format (for example TEL: +1-425-555-4567) rather than SIP URI format (for example *name@example.com*). This affects the ***From*** field of outgoing calls.

- **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.

- **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.

- **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.

- **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 the line. Select one of the following options:

  - **System Default**
    Use the system's **Default Name Priority** setting (Advanced Parameters110).

  - ***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.

- **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**.

- **User-Agent and Server Headers:** *Default = Blank (Use system type and software level). Software Level = 8.1 FP1.*
  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.

- **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.

VOIP Parameters

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

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

- **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***.

- **Registration Expiry:** *Default = 60 minutes.*
  This setting defines how often registration with the SIP ITSP is renewed following any previous registration.

- **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.

- **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.

- **Use Offered 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.

- **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.

- **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.

- **REFER Support:** *Default = On, Software level = 7.0+*
  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.

- **Caller ID from From Header:** *Default = Off. Software Level = 8.1 FP1.*
  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 FP1.*
  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.

Channel Setup

- **Channel**
  Channel number, cannot be edited
- **Appearance**
  Each channel can be accessed through pressing a Line Appearance to make calls, answer calls or conference. Lamps on the button reflect whether the channel is in use.
- **VMS Delay - Day:** *Default = 2. Range = 0 to 6 (number of rings).*
  Set the number of rings before an unanswered call should be redirected to an auto attendant when the system is not running in night service mode and the **VMS Schedule** is set to ***Always*** or ***Days Only***.
- **VMS Delay - Night:** *Default = 2. Range = 0 to 6 (number of rings).*
  Sets the number of rings before an unanswered call should be redirected to an 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 an auto attendant. The options are:
  - ***Always***
    Redirect calls when the system is in both day and night service 16 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. Software Level = 6.1+.*
  This field allows selection of which auto attendant is used by this line.

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 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

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"

- **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.

3.5.4.2 SIP Templates

IP Office Manager can be used to import trunk settings from a template. If you have multiple system using the same provider, this may simplify configuration and maintenance of the systems.

- Trunk templates are used by different types of IP Office system. Those template settings not supported by an Basic Edition system are ignored.

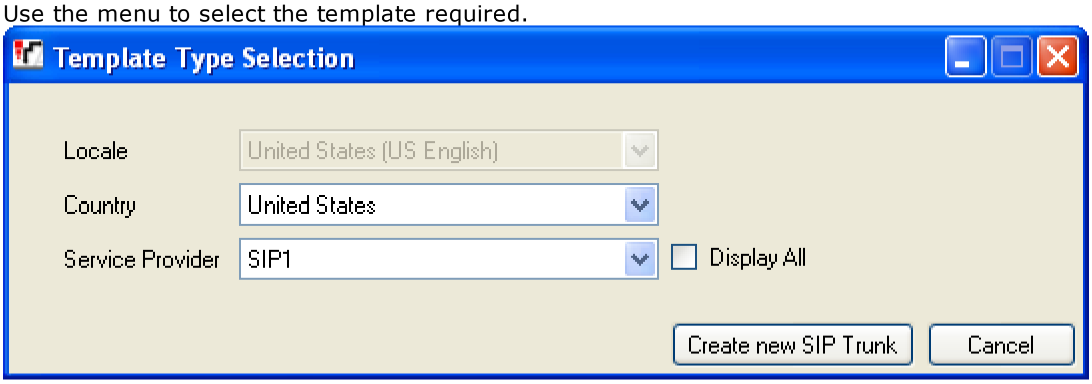
Importing Templates

Templates must be placed in the correct Manager **\Templates** sub-folder. This can be done using the following command:

1. Select **Tools | Import Templates in Manager**.
2. Browse to the current folder containing the templates that you want to import and select that folder.
3. Click **OK**.
4. Any template files in the folder will be copied to the correct Manager sub-folder.

Loading a SIP Trunk Template

1. Place the supplied template into the Manager application's **Template** sub-folder (by default ***C:\Program Files (x86)\Avaya\IP Office\Manager\Templates***).
2. In Admin Tasks, select **Trunks | SIP Trunk Administration**.
3. Click on the button at the left hand of a trunk to select it. Then right-click and select **New SIP Trunk from Template**. Alternatively click on the **New SIP Trunk from Template** icon top-right.
4. Use the menu to select the template required.



5. Select **Create New SIP Trunk**.

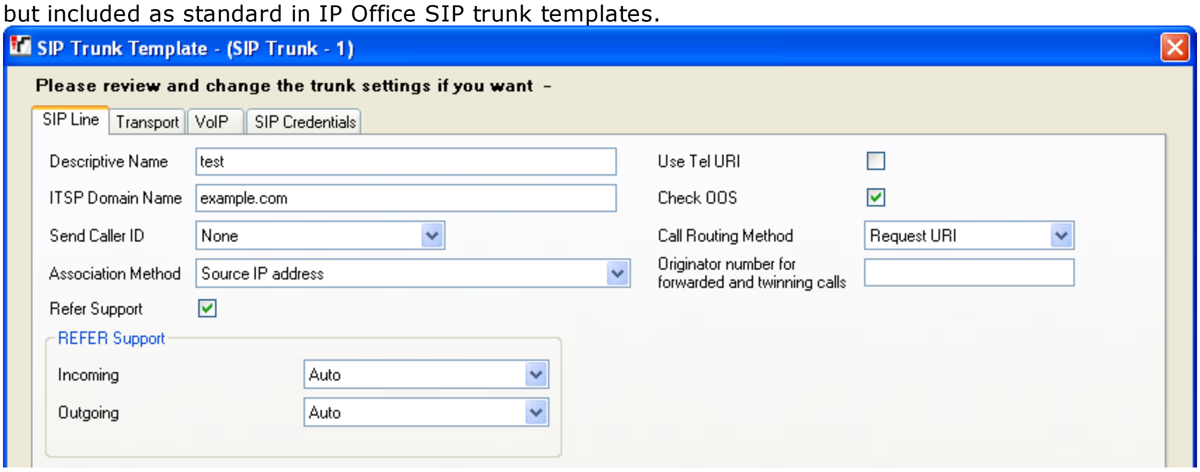
Saving a SIP Trunk Template

This topic is hidden at the request of the DevConnect team though its pointless as this is already public knowledge through the technical bulletins.

The functions to export a selected trunk's setting to a template are not enabled by default. To enable template exporting, using the Windows Registry Editor, change the value of **TemplateProvisioning** registry key (**HKEY_CURRENT_USER | Software | Avaya | IP400 | Manager**) to ***1***. Restart IP Office Manager.

1. In Admin Tasks, select **Trunks | SIP Trunk Administration**.
2. In the **SIP Trunk Setup** table select the row representing the trunk you want to export as a template.
3. Click on the button at the left hand of a trunk to select it. Then right-click and select **Generate SIP Trunk Template**. Alternatively, click on the **Generate SIP Trunk Template** icon top-right.

4. The SIP trunks settings are shown. Note that this includes settings not supported by Basic Edition systems but included as standard in IP Office SIP trunk templates.



5. Click on **Export**.
6. Select the SIP provider's country and enter a name.

7. Click on **Generate Template**.
8. Click on **OK** to save the template.

3.5.5 Outbound Call Handling

This menu cannot be accessed from the **System** 33 page.
This menu is accessed from the **Admin Tasks** 34 list by selecting **Trunks | Outbound Call Handling**.

This menu is used by systems with their **System Mode** set to ***PBX System***. For more details refer to **Key System or PBX System** 6. It is used to determine which line should be used to route an outgoing call when the user dials a number beginning with the system's **Outside Line** 38 prefix.

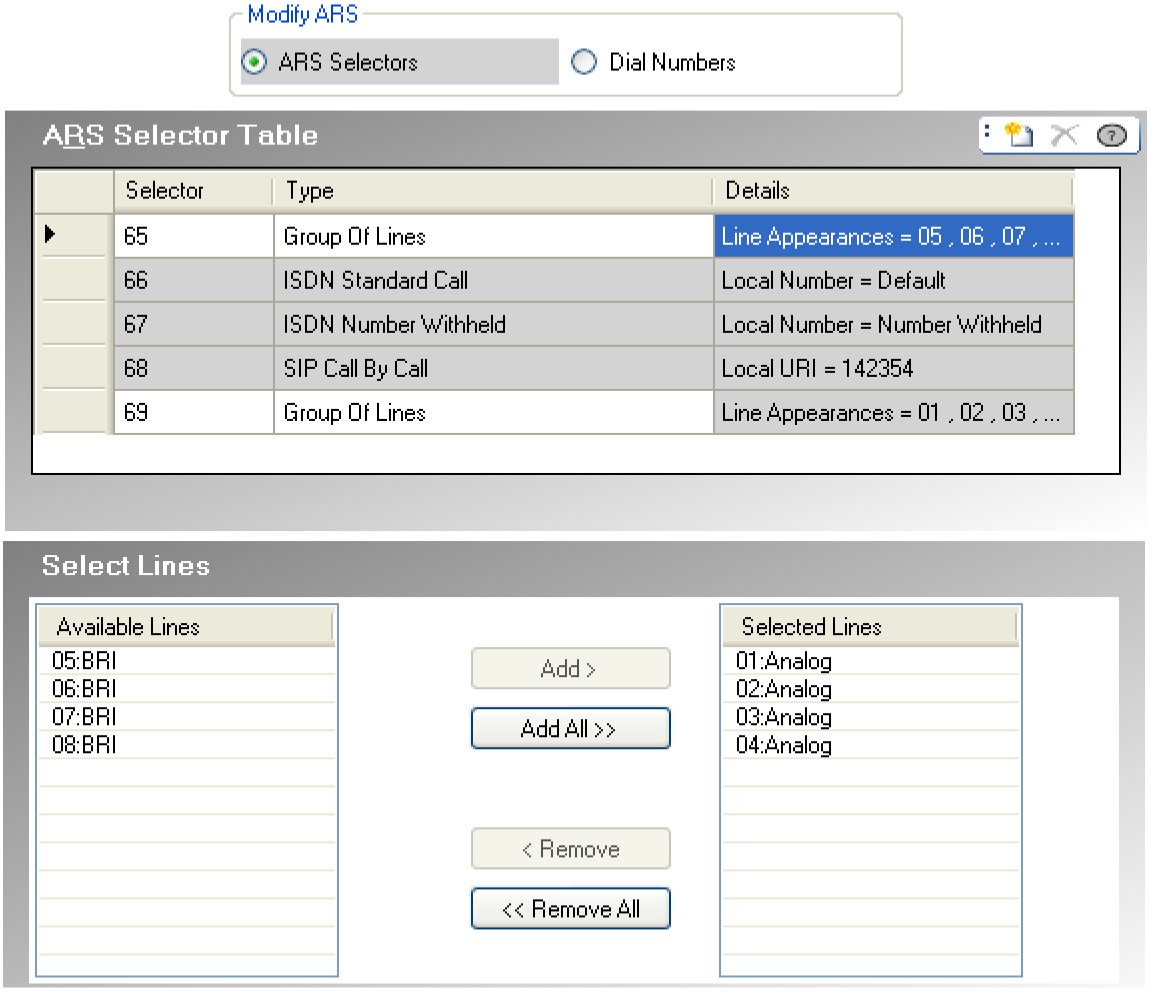
The call routing is done in two parts:

- The **ARS Selectors** 100 table is used to create groups of lines, each group with an ARS Selector number. The same line can be in more than one group.
- The **Dial Numbers** 102 table is used to match the number dialed by a user to a required ARS Select group number. When a match is found, an available line in that ARS Select group is seized for the call.

3.5.5.1 ARS Selectors

This menu cannot be accessed from the **System** 33 page.
This menu is accessed from the **Admin Tasks** 34 list by selecting **Trunks | Outbound Call Handling | ARS Selectors**.

ARS selectors are used to create groups of available lines. These can then be specified as the groups of line to be used for different types of external calls in the **Dial Numbers Table** 102. An available trunk in an ARS selector group can seized by dialing ***8*** followed by the ARS selector group number or using a line appearance 52 button configured for the ARS selector group number.



- **ARS Selector Table**
  This table is used to edit and add ARS selectors. A selector number can be dialed to seize a matching line by dialing ***8XX*** where ***XX*** is the selector number required. Selector numbers can also be assigned to line appearances 52 to make outgoing calls.

  - **Selector**
    This must be a number in the range 65 to 99. Selectors 65, 66 and 67 are used by default entries.

    - ***65: Group of Lines***
      This entry cannot be deleted. By default it contains all analog lines in the system, however it can be edited to change the lines included.

    - ***66: ISDN Standard Call - Local Number = Default***
      This entry cannot be deleted. Calls routed to this entry will use an available ISDN line with the calling party information set to match the user's **User CLI** if set or otherwise blank (to be set by the provider).

    - ***67: ISDN Number Withheld - Local Number = Withheld***
      This entry cannot be deleted. Calls routed to this entry will use an available ISDN line 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** 87 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.

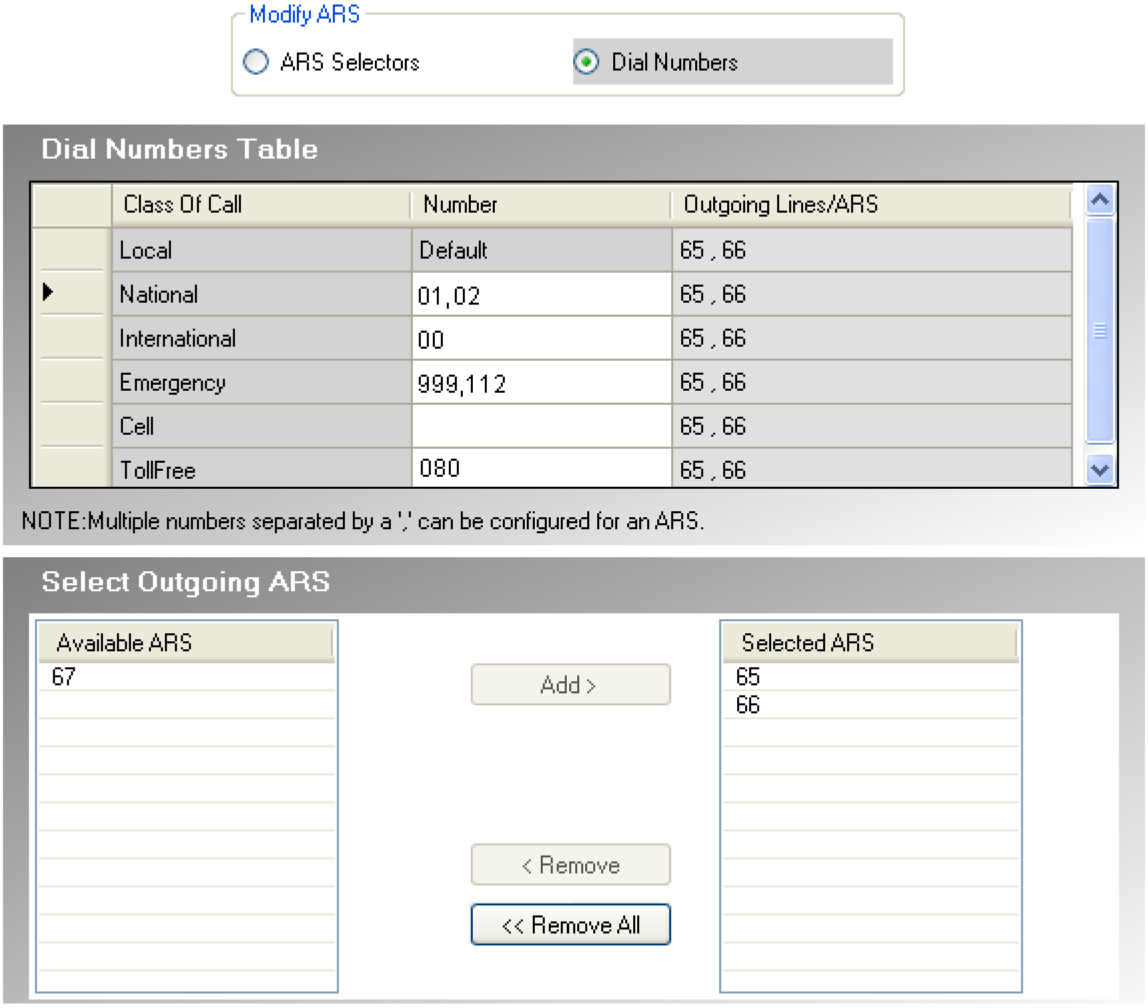
- **Details**
  This field show either the lines currently selected for use with the ARS selector or the local number setting for the calling party number.

- **Select Lines**
  This table is used to add or remove lines from the currently selected ARS selector if its **Type** is set to ***Group of Lines***.

3.5.5.2 Dial Numbers

This menu cannot be accessed from the **System** 33 page.
This menu is accessed from the **Admin Tasks** 34 list by selecting **Trunks | Outbound Call Handling | Dial Numbers**.

The **Dial Numbers Table** is used to match dialing prefixes to the group of trunks defined in the **ARS Selector** 100 table.



- **Dial Numbers Table**
  - **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.
    - Do not include the **Outside Line** 38 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.
- **Select Outgoing ARS**
  This table is used to associate ARS selectors configured on the **ARS Selector** 100 table with the currently selected **Class of Call** in the **Dial Numbers Table**. Multiple ARS selectors can be selected and the same ARS selector can be associated with more than one Class of Call.

3.6 Auxiliary Equipment

This menu is accessed from the **System** 33 page by selecting **Setup Auxiliary Equipments**.
This menu is accessed from the **Admin Tasks** 34 list by selecting **Auxiliary Equipment**.

These menus are used to configure the operation of a range of additional features provided by the telephone system.

Equipment	Description
Door Phone	If a handset has been configured as being a door phone, you can specify which extension is alerted when that door phone is used. Two door phones can be configured into the system. See **Door Phone** 103 for further detail.
Contact Closure	The phone system has two ports which can be connected to external relay systems, for example systems used to open doors. You can configure which users are able to activate those ports and the type of activation. See **Contact Closure** 104 for further detail.
Music on Hold	The phone system supports an external music on hold source. This is played to callers when they are put on hold. The source of the music is connected to the phone system by the system maintainer.
SMDR	The phone system can log call details at the end of each call. These SMDR records 153 (Station Message Detail Recording) can be output and sent to a specified IP address where they can be collected and processed by 3rd party call logging software.

3.6.1 Door Phone

This menu is accessed from the **System** 33 page by selecting **Setup Auxiliary Equipments**.
This menu is accessed from the **Admin Tasks** 34 list by selecting **Auxiliary Equipment**.

If a handset has been configured as being a door phone, you can specify which extension is alerted when that door phone is used. Two door phones can be configured into the system.

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

- **Assign Extension:** *Default = None.*
  Use the drop down list to select the extension to which the door phone is connected. The extension **Equipment Type** (**User Setup | Advanced Settings** 53) is set to ***Door Phone 1*** or ***Door Phone 2*** as appropriate.

- **Extensions to be alerted:** *Default = None.*
  This table is used to select which extensions are alerted and can answer when the door phone is used.

3.6.2 Music on Hold

This menu is accessed from the **System** 33 page by selecting **Setup Auxiliary Equipments**.
This menu is accessed from the **Admin Tasks** 34 list by selecting **Auxiliary Equipment**.

The phone system supports an external music on hold source. This is played to callers when they are put on hold. The source of the music is connected to the phone system by the system maintainer.

The phone systems music on hold source can also be used for callers being transferred instead of using ringing tone. This depends on the **Ring on Transfer** 110 setting.

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

3.6.3 SMDR

This menu is accessed from the **System** 33 page by selecting **Setup Auxiliary Equipments**.
This menu is accessed from the **Admin Tasks** 34 list by selecting **Auxiliary Equipment**.

The phone system can log call details at the end of each call. These SMDR records 153 (Station Message Detail Recording) can be output and 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.

3.6.4 Contact Closure Group

This menu is accessed from the **System** 33 page by selecting **Setup Auxiliary Equipments**.
This menu is accessed from the **Admin Tasks** 34 list by selecting **Auxiliary Equipment**.

The phone system has two ports which can be connected to external relay systems, for example systems used to open doors. You can configure which users are able to activate those ports and the type of activation.

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** 49 feature.

3.7 Auto Attendant Setup

This menu is accessed from the **System** 33 page by selecting **Administer Auto Attendant**.
This menu is accessed from the **Admin Tasks** 34 list by selecting **Auto Attendant Setup**.

This menu is used to configure the auto attendant facilities provided by the phone system. Up to 9 auto attendants are supported. The **Auto Attendant** field drop down list is used to select which auto attendant is being configured.

When Do Calls Go to an Auto Attendant?

The Basic Edition voicemail 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 as follows:

- **Immediate Auto Attendant Service**
  One of the auto attendants can be specified as the **Coverage Destination** for a particular line. The call is presented immediately to that auto attendant.
- **Delayed/Optional Auto Attendant Service**
  The **VMS Schedule** setting of each line can be used to set whether unanswered calls should go to a selected auto attendant. The settings can be enabled for day service, night service 16, both or never (the default). The delay used before going to the auto attendant is set by the line's **VMS Delay - Day** and **VMS Delay - Night** settings as appropriate.

Greeting Times

The auto attendant can provide different greetings at different times of the day. The greeting is always followed by the separate menu options greeting. These fields are used to set the time periods during which each greetings is used.

If the time periods overlap, the greeting used is the first one that is valid for the time period in the order morning, afternoon or evening. For call outside a configured time period or when the system is set to night service, the out of office hours greeting is used.

- **Morning:** *Default = 08:00 to 11:59*
  Set the operation times for the morning greetings.
- **Afternoon:** *Default = 12:00 to 17:59*
  Set the operation times for the afternoon greetings.
- **Evening:** *Default = 18:00 to 21:00*
  Set the operation times for the evening greetings.

Configure Profiles

These are the general settings for the auto attendant.

- **Auto Attendant:**
  This drop down list is used to select which auto attendant is being configured. For Release 6.1+, up to 9 auto attendants are supported.
  - The Add button can be used to create a new auto attendant. Up to 9 auto attendants are supported.
  - The **Delete** button removes the currently selected auto attendant. It does not erase any greetings that have been recorded for the auto attendant. The first auto attendant cannot be removed.
- **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:** *Default = Each menu uses its own prompt, Software level = 6.1*
  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 its own prompt*** does that. Alternately one of the menu options can be selected as the menu option prompt played at all times of day.
- **Direct Dial By 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.
- **Follow Night Service:** *Default = On, Software level = 6.1+*
  When selected, while 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 use the greetings and menu options as determined by its time profile settings.
- **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***.
- **Language:** *Default = Match system language, Software level = 6.1+.*
  This settings controls the language used for auto attendant action prompts. If not set the system Language 38 setting is used.
- **Morning**
  When this **Profile** is selected, the morning greeting and morning menu options are used during the times indicated in **Greeting Times** for morning. The **Record Greeting** number indicates the internal number that can be dialed to hear the current greeting and record a new greeting.
- **Afternoon**
  When this **Profile** is selected, the afternoon greeting and afternoon menu options are used during the times indicated in **Greeting Times** for afternoon. The **Record Greeting** number indicates the internal number that can be dialed to hear the current greeting and record a new greeting.
- **Evening**
  When this **Profile** is selected, the evening greetings and evening menu options are used during the times indicated in **Greeting Times** for evening. The **Record Greeting** number indicates the internal number that can be dialed to hear the current greeting and record a new greeting.
- **Out of office hours**
  For times not covered by the morning, afternoon or evening settings, the Out of Hours greetings and Out of Hours menu options are used. If Follow Night Service is selected, this also applies when the system is put into night service. The **Record Greeting** number indicates the internal number that can be dialed to hear the current greeting and record a new greeting.
- **Menu options**
  This greeting should details the options available to callers after hearing the auto attendant greeting. The **Record Greeting** number indicates the internal number that can be dialed to hear the current greeting and record a new greeting. Note that number changes depending on whether you are selecting the menu options for the morning, afternoon, evening or out of hours. The Menu Prompt setting (see above) controls whether separate menu prompts are used for each time state or not.

- **Emergency Greeting:** *Software level = 6.1*
  Using the ***Transfer to Emergency Greeting*** action from an auto attendant, an emergency greeting can be recorded and activated or deactivated. When active, the emergency greeting is played to callers in advanced of any other auto attendant greeting. The **Record Greeting** number indicates the internal number that can be dialed to hear the current greeting and record a new greeting. Use of this feature requires the system password to be set. When an emergency greeting message is active, a message to that effect is also displayed on the **Alarm Extension** *(see below)*.

- **Alarm Extension:** *Default = 10, Software level = 6.1*
  When the auto attendant's **Emergency Greeting** is active *(see above)*, a warning message is also displayed on the extension indicated by this field.

Setup Auto Attendant Actions

This table allows you to assign which key presses have associated actions for the auto attendant.

- **Type:** *Software level = 6.1*
  The auto attendant menu options can be varied according to the time of day. These radio buttons are used to select which set of actions are current displayed and editable within IP Office Manager.
  - The button below the key actions table can be used to copy the current set of actions for the selected **Type** to all the other types in the current auto attendant. This overrides all the existing settings for actions and destinations.
- **Key**
  The standard telephone dial pad keys, ***0*** to ***9*** plus *, # and ***Fax***.
- **Action**
  The following actions can be assigned to each key.
  - ***No Action***
    The corresponding key takes no action.
  - ***Dial by Name***
    Callers are asked to dial the name of the user they require and then press #. The recorded mailbox name prompts 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 set to **Exclude from Directory** are not included. Users can record their name by accessing their mailbox and dialing *05.
  - ***Dial By Number***
    This option allows callers with DTMF phones to dial the extension number of the user they require. No destination is set for this option. The **Direct Dial-By-Number** setting above determines how the digits dialed with this action are used.
  - ***Transfer to Auto Attendant:*** *Software level = 6.1+*
    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:*** *Software level = 6.1+*
    This option transfers the caller to a set of prompts for recording the emergency greeting and for selecting whether the emergency greeting is active or not.
    - If a system password has been set, the caller is asked to enter that password before they can continue.
    - When the emergency greeting is active, it is played to other auto attendant callers before any other auto attendant greeting.
    - When the emergency greeting is active, a warning is displayed on the auto attendant's **Alarm Extension**.
  - ***Transfer to Number***
    Transfer the call to the extension or group selected in the **Destination** field.
  - ***Replay Menu Greeting***
    Repeat the current menu options greeting.
- **Destination**
  Sets the destination 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.

Recording Auto Attendant Greetings

Dialing the appropriate number shown in the table below allows recording and playback of the matching auto attendant prompt. It is important to remember that callers always hear two prompts, a greeting prompt and then a menu prompt. In addition, they may also first hear the emergency greeting if it has been activated.

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

3.8 Advanced Parameters

This menu cannot be accessed from the **System** [33] page.
This menu is accessed from the **Admin Tasks** [34] list by selecting **Advanced**.

Advanced System Settings

- **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.

- **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.

- **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.

- **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.

- **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.

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

- **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***.

- **Companding Law** (Quick systems only)
  The IP Office system is defaulted to A-Law or U-Law by the SD Feature Key dongle inserted into the unit or by the locale selected when creating an off-line configuration. 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.

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

STUN Settings for Network

These settings are used if SIP trunks are added to the phone system's configuration using the **SIP Trunk Administration** 87 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.

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

- **STUN Server 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.

- **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.

- **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.

  - **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.

- **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.

SMTP Server Configuration

Email can be used to provide users with an alert when they have a new voicemail message. 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. The user email addresses are set through the user's Advanced Settings 53.

- **Server 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.
- **Email From Address:** *Default = Blank*
  This field sets the sender address to be used with mailed alarms. 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 support SMTP relay.
- **Server Requires 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.
- **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
- **Use Challenge Response Authentication:** *Default = Off.*
  This field should be selected if the SMTP uses CRAM-MD5.

Busy Tone Detection

The use of busy tone detection to clear calls is normally set to match the requirements of the system locale. These settings can be used to change the settings if required for a particular site.

- **! WARNING**
  Changes to these settings will require the system to restart when the changes are saved.
  - **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.
  - **On Width:**
    Set the width used for on signal detection.
  - **Off Width:**
    Set the width used for off signal detection.

Chapter 4.
Button Programming

4. 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.

Normally buttons are numbered from 01, from left to right, starting from the bottom row up. However for 1400 and 9500 Series telephones, this changes if the **System Mode** 38 is set to ***PBX System*** in which case buttons are numbered from 01 from left to right, starting from the top row downwards.

Default Buttons

The default button assignment depends on whether the system is a ***Key System*** or ***PBX System***.

- **PBX System**
  - **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 the functions listed in Button Programming Functions 116. These buttons can be programmed by the system administrator and, for some functions, the extension user.
- **Key System**
  - **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.
  - **03-07: Line Buttons**
    Buttons 03 and upwards up to the number of lines assigned to the extension are used as line appearance buttons for external calls. These can only be programmed by a system administrator using the Number of Lines 38 and Line Assignment 52 functions. They cannot be overridden by the extension user.
  - **Other Buttons**
    Any additional buttons can be used for the functions listed in Button Programming Functions 116. These buttons can be programmed by the system administrator and, for some functions, the extension user.

4.1 Button Programming Functions

Function	Description	LED
911-View/Emergency View 119	A button set to this function indicates when a call has been made using a number in the system's Emergency Number List 41. Pressing the button displays a list of such calls.	Yes
Absent Message 119	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 119	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 119	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 119	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 120	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 121	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 122	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 121	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 122	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 122	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 122	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 122	A button set to this function allows the user to call or page the calling group represented by the button.	–
Call Screening 123	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 124	A button set to this function allows the user to drop a call from a conference.	–
Contact Closure 1 124	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 2 124	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 124	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 125	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 48 is used to select a line.	Yes
Hunt Group 125	A button set to this function allows the user to call or page the hunt group represented by the button.	–
Idle Line Pickup 125	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 125	A button set to this function allows the user to redial the last external number dialed.	–
Loudspeaker Paging 125	A button set to this function allows the user to redial the last external number dialed.	–
Message Alert Notification 126	For IP Office Release 7.0, 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** 119 buttons configured.	–
Night Service Button 16	A night service button is used to switch night service on/off.	Yes
Pickup Group 126	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 126	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 127	A button set to this function allows the user to send a recall or hook flash signal.	–
Save Number Redial 127	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 127	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 127	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 127	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 128	A button set to this function allows the user to switch use of voicemail coverage for their extension on or off.	–
VMS Mailbox Transfer 128	A button set to this function allows the user to transfer their current call to an 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 129	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, ie. if already assigned to a button, assigning the function to another button will automatically clear 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.

4.2 Manager Buttons

In some cases, the names used for the programming of button features in the 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 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

4.3 911-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 Number List 41. 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 automatically clears 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.

4.4 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 automatically clears 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.
- For IP Office Release 7.0, the button can also be used to check the absent message setting of other users. When pressed, pressing the **Auto Dial - Intercom** 119 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.

4.5 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 automatically clears 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**.

4.6 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 automatically clears 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.

4.7 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 18 operation.

4.8 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.

4.9 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 automatically clears 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**.

4.10 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 automatically clears 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 48 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**.

4.11 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.

4.12 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 automatically clears 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**.

4.13 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 automatically clears 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 dial **23**.
  - On BST phones, press **FEATURE** and dial **812**.

4.14 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 automatically clears 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**.

4.15 Calling Group

A button set to this function allows the user to call or page the calling group represented by the button.

4.16 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, 1608, 1616, 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 automatically clears 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.

4.17 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 automatically clears 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**.

4.18 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 automatically clears 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**.

4.19 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 automatically clears 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**.

4.20 Do Not Disturb

Use this feature to be able to press a programmed button to prevent incoming calls for the extension from ringing (LEDs/LCD still flash). When Do Not Disturb is on, external callers hear ringing while internal callers hear a busy signal. You should use Do Not Disturb only if someone answers external calls for your extension when you do not answer them.

You can configure do not disturb exceptions. These are numbers that are still able to call even when do not disturb is on.

- 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, press **FEATURE** and then dial **01**.
- Do not disturb overrides call forwarding.
- 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).

4.21 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 automatically clears 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**.

4.22 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 18 operation.

4.23 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 automatically clears 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.

4.24 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 automatically clears 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.

4.25 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 automatically clears the setting from the existing button.
- To access this function without a programmable button, press an intercom or call appearance button and dial **70**.

4.26 Message Alert Notification

For IP Office Release 7.0, 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** 119 buttons configured.

- If an extension already has a button set to this function, creating another button with this function automatically clears the setting from the existing button.

4.27 Night Service

Use this feature to program a button on the first extension on the system to turn night service on and off. When night service is on, all lines assigned to the telephones of the users in the night service group 57 ring immediately, regardless of their normal line ringing settings.

Night service is useful if you want phones to ring after regular business hours. For example, although Shipping Department workers do not answer calls directly during the day, you want them to answer incoming calls after hours.

- You must program a Night Service Button on the first extension 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 38 as follows if set:
  - You must enter the password when turning Night Service on or off.
  - Calls to numbers other than marked system dials or in the Emergency Phone Number List 41 require 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 138.
- Night Service is unavailable on T1 lines with Direct Inward Dialing (DID).

4.28 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.

4.29 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 automatically clears 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**.

4.30 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 automatically clears the setting from the existing button.

4.31 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 automatically clears 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.

4.32 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 automatically clears 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 18 operation.

4.33 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 127 button.
- If an extension already has a button set to this function, creating another button with this function automatically clears 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**.

4.34 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 automatically clears 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**.

4.35 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 53 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 automatically clears 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**.

4.36 Voice Mailbox Transfer

A button set to this function allows the user to transfer their current call to an 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 automatically clears the setting from the existing button.

4.37 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 automatically clears 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 hear a repeated double tone.
- 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 5.
Manager Menu Commands

5. Manager Menu Commands

The commands available through the Manager menu bar, change according to the mode in which Manager is running. Commands may also be grayed out if not usable. The following sections outline the functions of each command. The **Edit** and **Help** menus are not included.

Simplified View

These menu options are available when there is no configuration loaded in the Manager application.

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Embedded File Management

These menu options are available when IP Office Manager is switched to Embedded File Management 143 mode.

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5.1 File Menu

5.1.1 Open Configuration

This command displays the Select IP Office menu used to receive an IP Office systems configuration settings. See Starting Manager 26.

5.1.2 Close Configuration

This command closes the currently loaded configuration without saving it.

5.1.3 Save Configuration

The **File | Save** command saves the amended configuration. If the configuration was received from an IP Office system, the **Send Config** 29 menu is displayed. If the configuration file was opened from a file on the PC 30 or created from new 35, the file is saved as a file on the PC.

5.1.4 Save Configuration As

The **File | Save As** command allows you to save a configuration a file on the Manager computer. The command displays the **Save File As** menu box. You can enter the new file name, including the drive and directory.

Configurations saved onto the PC in this way can be reopened 30.

5.1.5 Preferences

This command displays a menu for configuring various aspects of Manager's operation. The menu is divided into a number of tabs.

Note that some of the preferences settings are not applicable when managing an Basic Edition system.

5.1.5.1 Preferences

This tab is accessed through **File | Preferences** and then selecting the **Preferences** sub-tab.

- **Edit Services Base TCP Port:** *Default = On.*
  This field shows or hides the Service Base TCP Port setting.
  - **Service Base TCP Port:** *Default = 50804*
    TCP access to the configuration and security settings on a system requires IP Office Manager to send its requests to a specific port. This setting allows the TCP Base Port used to be set to match the TCP Base Port setting of the system.
  - **Service Base HTTP Port:** *Default = 80*
    HTTP access to the configuration and security settings on a system requires IP Office Manager to send its requests to a specific port. This setting allows the HTTP Base Port used to be set to match the HTTP Base Port setting of the system.
- **Enable Time Server:** *Default = On.*
  This setting allows Manager to respond to time requests from IP Office systems.
- **Enable BootP and TFTP Servers:** *Default = On.*
  This setting allows Manager to respond to BOOTP request from IP Office systems for which it also has a matching BOOTP entry. It also allows the IP Office to respond to TFTP requests for files.
- **Auto Connect on start up:** *Default = On*
  If on, when Manager is started it will automatically launch the **Select IP Office** menu and display any discover IP Offices. If only one IP Office is discover, Manager will automatically attempt to login using the default name and password and if this fails display the login request for that IP Office instead.
- **Set Simplified View as Default:** *Default = On*
  If on, the Manager will start in simplified view 31 if no configuration is loaded.
- **Default to Standard mode:** *Default = Off*
  If on, when a configuration for a new or defaulted system running in IP Office Basic Edition mode is loaded, Manager will automatically convert the configuration to IP Office Essential Edition. Sending the configuration back to the system will restart it in IP Office Essential Edition. Only select this option if the only systems you expect to install are IP Office Essential Edition systems.
  - This setting does not affect existing systems with non-default configurations.
- **Use Remote Access**
  This setting is not used with IP Office Basic Edition systems.
- **SE Central Access**
  This setting is not used with IP Office Basic Edition systems.
- **SE Central Access port**
  This setting is not used with IP Office Basic Edition systems.

5.1.5.2 Directories

This tab is accessed through **File | Preferences** and then selecting the **Directories** sub-tab.

These fields set the default location where Manager will look for and save files. This tab is also accessed by the **File | Change Working Directory** command.

- **Working Directory (.cfg files)**
  Sets the directory into which Manager saves .cfg files. By default this is the Manager application's program directory.
- **Binary Directory (.bin files)**
  Sets the directory in which the Manager upgrade wizard, HTTP, TFTP and BOOTP functions look for firmware files requested by phones, expansion module, control units and other hardware components. That includes .bin file, .scr files and .txt files. By default this is the Manager application's program directory.
  - Note that in the Upgrade Wizard 140, right-clicking and selecting **Change Directory** also changes this setting.
- **Known Units File:** *Software level = 4.0 Q2 2007 maintenance release+.*
  Sets the file and directory into which Manager can record details of the IP Office systems it has discovered. Once a file location has been specified, a **Known Units** 28 button becomes available on the discovery menu used for loading IP Office configuration. Pressing that button displays the known units file as a list from which the required IP Office system can be selected. It also allows sorting of the list and entries to be removed.

5.1.5.3 Discovery

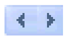
This tab is accessed through **File | Preferences** and then selecting the **Discovery** sub-tab.

These settings affect the **Select IP Office** menu used by Manager to discovery IP Office systems. By default IP Office 3.2 systems respond to both UDP and TCP discovery. Pre-3.2 IP Office systems only support UDP discovery.

- **TCP Discovery:** *Default = On*
  This setting controls whether Manager uses TCP to discover IP Office systems. Only IP Office 3.2 and higher systems can respond to TCP discovery. The addresses used for TCP discovery are set through the **IP Search Criteria** field below.
  - **NIC IP/NIC Subnet**
    This area is for information only. It shows the IP address settings of the LAN network interface cards (NIC) in the PC running Manager. Double-click on a particular NIC to add the address range it is part of to the IP Search Criteria. Note that if the address of any of the Manager PCs NIC cards is changed, the Manager application should be closed and restarted.
- **IP Search Criteria**
  This tab is used to enter TCP addresses to be used for the discovery process. Individual addresses can be entered separated by semi-colons, for example 135.164.180.170; 135.164.180.175. Address ranges can be specified using dashes, for example 135.64.180.170 - 135.64.180.175.
- **UDP Discovery:** *Default = On*
  This settings controls whether Manager uses UDP to discover IP Office systems. Pre-3.2 IP Office systems only respond to UDP discovery. By default IP Office 3.2 and higher systems also respond to UDP discovery but that can be disabled through the IP Office system's security settings.
  - **Enter Broadcast IP Address:** *Default = 255.255.255.255*
    The broadcast IP address range that Manager should used during UDP discovery. Since UDP broadcast is not routable, it will not locate IP Office systems that are on different subnets from the Manager PC unless a specific address is entered.
- **Use DNS:** *Software level = IP Office Manager 6.2+.*
  Selecting this option allows IP Office Manager to use DNS name (or IP address) lookup to locate an IP Office system. Note that this overrides the use of the TCP Discovery and UDP Discovery options above. This option requires the IP Office IP address to be assigned as a name on the users DNS server. When selected, the **Unit/Discovery Address** field on the **Select IP Office** menu is replaced by a **Enter Unit DNS Name or IP Address** field.
- **SCN Discovery**
  If enabled, when discovering IP Office systems, the list of discovered systems will group IP Offices in the same Small Community Network and allow them to be loaded as a single configuration. Basic Edition systems are not supported in a Small Community Network.

5.1.5.4 Visual Preferences

This tab is accessed through **File | Preferences** and then selecting the **Visual Preferences** sub-tab.

- **Icon size**
  Sets the size for the icons in the navigation pane between ***Small***, ***Medium*** or ***Large***.
- **Multiline Tabs:** *Default = Off.*
  In the details pane, for entry types with more than two tabs, Manager can either use  buttons to scroll the tabs horizontally or arrange the tabs into multiple rows. This setting allows selection of which method Manager uses.

5.1.5.5 Security

This tab is accessed through **File | Preferences** and then selecting the **Security** sub-tab.

Controls the various security settings of Manager.

- **Request Login on Save:** *Default = On*
  By default a valid user name and password is required to receive a configuration from an IP Office and also to send that same configuration back to the IP Office. Deselecting this setting allows Manager to send the configuration back without having to renter user name and password details. This does not apply to a configuration that has been saved on PC and then reopened. This setting can only be changed when a configuration has been opened using a user name and password with Administrator rights or security administration rights.

- **Close Configuration/Security Settings After Send:** *Default = On.*
  When selected, the open configuration file or security settings are closed after being sent back to the IP Office system.

- **Save Configuration File After Load:** *Default = On.*
  When selected, a copy of the configuration is saved to Manager's working directory 134. The file is named using the IP Office's system name and the suffix ***.cfg***. This setting can only be changed when a configuration has been opened using a user name and password with Administrator rights or security administration rights.

- **Backup Files on Send:** *Default = On.*
  If selected, whenever a copy of a configuration is sent to an IP Office system, a backup copy is saved in IP Office Manager's working directory 134. The file is saved using the system name, date and a version number followed by the **Backup File Extension** as set below. This setting can only be changed when a configuration has been opened using a user name and password with Administrator rights or security administration rights.

  - **Backup File Extension:** *Default = .BAK*
    Sets the file extension to use for backup copies of system configurations generated by the **Backup Files on Send** option above.

  - **Number of Backup Files to keep:** *Default = Unlimited, Software level = 4.2+.*
    This option allows the number of backup files kept for each system to be limited. If set to a value other then **Unlimited**, when that limit would be exceeded, the file with the oldest backup file is deleted.

- **Enable Application Idle Timer (5 minutes):** *Default = Off, Software level = 4.1+.*
  When enabled, no keyboard or mouse activity for 5 minutes will cause the Manager to grey out the application and re-request the current service user password. This setting can only be changed when a configuration has been opened using a user name and password with Administrator rights or security administration rights.

- **Secure Communications:** *Default = Off, Software level = 4.1+.*
  When selected, any service communication from Manager to IP Office uses the TLS protocol. This will use the ports set for secure configuration and secure security access. It also requires the configuration and or security service within the IP Office's security configuration settings to have been set to support secure access. Depending on the level of that secure access selected, it may be necessary for the Manager Certificate Checks below to be configured to match those expected by the IP Office configuration and or security service.

  - When **Secure Communications** is set to ***On***, a padlock icon is displayed at all times in the lower right Manager status field.

5.1.5.6 Validation

This tab is accessed through **File | Preferences** and then selecting the **Validation** sub-tab.

By default Manager validates the whole configuration when it is loaded and individual fields whenever they are edited. This tab allows selection of when automatic validation should be applied to configuration files loaded into Manager.

- **Validate configuration on open**
  Automatically validate configuration files when they are opened in Manager.

- **Validate configuration on edit**
  Validate the whole configuration when **OK** is clicked after editing a record. For large configurations, disabling this option removes the delay caused by validating the configuration after every edit.

- **Prompt for configuration validation on save or send**
  If selected, when saving or sending a configuration, a prompt is displayed asking whether the configuration should be validated. If validation is selected and errors are found, the send or save process is canceled. This option is disabled if **Validate configuration on edit** is selected.

5.1.6 Offline

5.1.6.1 Create New Config

This command starts a menu that allows you to create an offline configuration 35 by specifying the system locales, the type of IP Office control unit and expansion modules and the trunk cards fitted.

The same action is performed by the icon in the Main Toolbar.

5.1.6.2 Open File

This command allows a configuration file stored on PC to be opened in Manager.

5.1.6.3 Send Config

This command is used to send an offline configuration to an IP Office system.

- After sending the configuration, you should receive the configuration back from the system and note any new validation errors shown by Manager. For example, if using Embedded Voicemail, some sets of prompt languages may need to be updated to match the new configurations locale setting using the **Add/Display VM Locales** 145 option.

5.1.6.4 Receive Config

This command displays the **Select IP Office** menu used to receive an IP Office systems configuration settings.

Once the configuration has been received, you are prompted to save it on the PC.

5.1.7 Advanced

5.1.7.1 Erase Configuration

This command returns the configuration settings of an IP Office system back to their default values. It also resets the user name and password for configuration access back to ***Administrator*** and ***Administrator***.

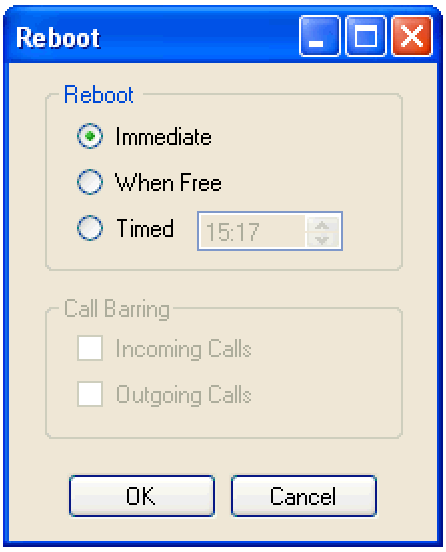
When this command is used, the **Select IP Office** menu is displayed. Once an IP Office system is selected, a valid user name and password are required to complete the action.

This command can also be performed from either of the first two extensions in the system using the **Restart - Defaults** command.

5.1.7.2 Reboot

When this command is used, the **Select IP Office** menu is displayed. Once an IP Office system is selected, a valid user name and password are required. The type of reboot can then be selected.

An immediate reboot can also be performed from either of the first two extensions in the system using the **Reset - Save All** command.



- **Reboot**
  If Manager thinks the changes made to the configuration settings are mergeable, it will select Merge by default, otherwise it will select ***Immediate***.

  - ***Immediate***
    Send the configuration and then reboot the IP Office.

  - ***When Free***
    Send the configuration and reboot the IP Office when there are no calls in progress. This mode can be combined with the **Call Barring** options.

  - ***Timed***
    The same as When Free but waits for a specific time after which it then wait for there to be no calls in progress. The time is specified by the **Reboot Time**. This mode can be combined with the **Call Barring** options.

- **Reboot Time**
  This setting is used when the reboot mode **Timed** is selected. It sets the time for the IP Office reboot. If the time is after midnight, the IP Office's normal daily backup is canceled.

- **Call Barring**
  These settings can be used when the reboot mode When Free is selected. They bar the sending or receiving of any new calls.

5.1.7.3 System Shutdown

This command can be used to shutdown systems with IP Office Release 6 or higher software. The shut down can be either indefinite or for a set period of time after which the IP Office will 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 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.

An indefinite shutdown can also be performed from either of the first two extensions in the system using the **Shutdown - Save All** command.

1. Once you have selected the IP Office system from the **Select IP Office** menu, the **System Shutdown Mode** menu is displayed.
2. Select the type of shutdown required. If **Indefinite** is used, the system can only be restarted by having its power switched off and then on again. If a **Timed** shutdown is selected, the IP Office will reboot after the set time has elapsed.

5.1.7.4 Upgrade

This command starts the **Upgrade Wizard** tool. This tool is used to compare the software level of the control unit and expansion modules within IP Office systems against the software level of the .bin binary files Manager has available. The Upgrade Wizard can then be used to select which units to upgrade.

- **WARNING**
  Incorrect use of the upgrade command can halt IP Office operation and render units in the system unusable. You must refer to the IP Office Technical Bulletins for a specific release for full details of performing software upgrades to that release. For example:
  - When upgrading a system from a pre-8.0 release, the systems security settings should first be defaulted. This allows the security settings to then be altered during the upgrade to support IP Office Web Manager.
- Performing any other actions on a system during an upgrade or closing the upgrade wizard and Manager during an upgrade may render systems unusable.
- During an upgrade the IP Office system may restrict calls and services. It will reboot and disconnect all current calls and services.

The list area shows details of IP Office systems found by the Upgrade Wizard and the software currently held by that system. The **Version** column details the current software each unit in the systems is running whilst the **Available** column shows the version of .bin file Manager has available for that type of unit (a – indicates no file available).

- The check boxes are used to select which units should be upgraded. Upgrading will require entry of a valid name and password for the selected IP Office system.
- The **Validate** option should remain selected wherever possible. When selected, the upgrade process is divided as follows: transfer new software, confirm transfer, delete old software, restart with new software. If **Validate** is not selected, the old software is deleted before the new software is transferred.
- The **Backup system files** option will cause the IP500 V2 to backup its memory card files as part of the upgrade.
- The **Upload system files** option will upload various files:
  - It copies the binary files used by the IP Office control unit and external expansion modules.
  - It copies the firmware files used by phones supported by the system.
  - For systems configured to running in IP Office Basic Edition, IP Office Basic Edition - PARTNER Mode or IP Office Basic Edition - Norstar Mode mode, the files for IP Office Web Manager are copied.
  - For systems configured to run Embedded Voicemail, the Embedded Voicemail prompts for those supported languages set as the system locale, user locales, incoming call route locales and short code locales are upgraded. In addition the English language prompts are upgraded as follows: **IP Office A-Law/Norstar SD Cards** - UK English, **IP Office U-Law/PARTNER SD Cards** - US English.

Searching for Particular Systems

The default address used by the Upgrade Wizard is the address shown in the Manager title bar, which is selected through File | Preferences 135. If the unit required is not found, the address used can be changed.

1. Enter or select the required address in the **Unit/Broadcast Address** field.
2. Click **Refresh** to perform a new search.

Changing the .bin File Directory Used

The directory in which the Upgrade Wizard looks for .bin files is set through Manager's Binary Directory setting. This can be changed using **File | Preferences | Directories** 134. It can also be changed directly from the Upgrade Wizard as follows.

1. Right-click on the list area.
2. Select **Select Directory**.
3. Browse to and highlight the folder containing the .bin files. Click **OK**.
4. The list in the Available column will be updated to show the .bin files in the selected directory that match IP Office units or modules listed.

5.1.7.5 Change Mode

This command is not available whilst a configuration is loaded. When used, it allows the operating mode of a IP Office Basic Edition system to be changed to IP Office Essential Edition.

- For IP Office Release 8.0 and higher, for a system to operate in IP Office Essential Edition it must have an **Essential Mode** license in it configuration. IP Office Essential Edition systems without this license will not provide any telephony functions.
- If this is an existing system, it is recommended that you first use Manager to receives and save a copy of the current configuration locally using **Save Configuration As** 132.
- This process does not default the security settings of the system.
- If this command is used on a system that includes components not supported by the IP Office Essential Edition (currently IP500 ETR6 base cards for ETR phones), the system will restart but those components will be disabled.
- **! Automatic Conversion to IP Office Essential Edition**
  This process can be applied automatically when a configuration for a new or defaulted system running in IP Office Basic Edition is loaded. This is done by selecting the **Default to Standard mode** option in the Manager **Preferences** 133. Only select this option if the only systems you expect to install are IP Office Essential Edition systems.

5.1.7.6 Switch to Standard Mode

IP Office Basic Edition is the default mode assumed by a IP500v2 control unit fitted with an IP Office A-Law or IP Office Mu-Law System SD card.

This option will change the operating mode of the configuration loading in Manager to that of a IP Office Essential Edition system. Manager will automatically switch to its advanced view mode. When the configuration is sent back to the IP Office system, the system will restart in IP Office Essential Edition.

The command provides two options:

- **Default**
  Using this method to switch to IP Office Essential Edition will default the configuration. It is the recommended method for installation of a new installation or for when a IP Office Essential Edition system has been defaulted and needs to be returned to IP Office Essential Edition operation.
- **Best Match**
  Using this method to switch to IP Office Essential Edition mode will attempt to preserve configuration settings; for examples user names, extension numbers, licenses, SIP trunks, etc. However many settings will be flagged as errors by Manager. These should be resolved before sending the configuration to the system.

Note

- For IP Office Release 8.0 and higher, for a system to operate in IP Office Essential Edition it must have an **Essential Mode** license in it configuration. IP Office Essential Edition systems without this license will not provide any telephony functions.
- If this is an existing system, it is recommended that you first use Manager to receives and save a copy of the current configuration locally using **Save Configuration As** 132.
- This process does not default the security settings of the system.
- If this command is used on a system that includes components not supported by the IP Office Essential Edition (currently IP500 ETR6 base cards for ETR phones), the system will restart but those components will be disabled.
- **! Automatic Conversion to IP Office Essential Edition**
  This process can be applied automatically when a configuration for a new or defaulted system running in IP Office Basic Edition mode is loaded. This is done by selecting the **Default to Standard mode** option in the Manager **Preferences** 133. Only select this option if the only systems you expect to install are IP Office Essential Edition systems.

5.1.7.7 Embedded File Management

The contents of the SD memory card used by the system can be viewed through Manager. For further details refer to Embedded File Management 149.

5.1.7.8 Format IP Office SD Card

This command allows suitable SD cards to be formatted by the Manager PC. The IP500v2 supports SD cards with the following format: SDHC minimum 4GB FAT32 format (Single partition, SDHC, class2+, FAT32, SPI & SD bus). Non-Avaya supplied cards of the same format can be used in the IP500v2 system's **Optional SD** slot for additional actions such as backup.

- ⚠ **WARNING: All File Will Be Erased**
  Note that this action will erase any existing files and folders on the card. Once a card has been formatted, the folders and files required for IP Office operation can be loaded onto the card from the Manager PC using the Recreate IP Office SD Card 144 command.

- ⚠ **WARNING:**
  Avaya supplied SD cards should not be formatted using any other method than the format commands within IP Office Manager and IP Office System Status Application. Formatting the cards using any other method will remove the feature key used for IP Office licensing from the card.

1. Insert the SD card into a reader slot on the Manager computer.
2. Using Manager, select **File | Advanced | Format IP Office SD Card**.
3. On Quick systems, select **IP Office A-Law** or **IP Office U-Law.**
   On Partner systems, **select IP Office PARTNER Edition.**
   On Norstar systems, select **IP Office Norstar Edition**.

   This selection just sets the card label shown when viewing the card details. It does not affect the actual formatting. Select the label that matches the files set you will be placing on the card. The other options available are not used for a Basic Edition system.
4. Browse to the card location and click **OK**.
5. The status bar at the bottom of Manager will display the progress of the formatting process.
6. When the formatting is complete, you can use the Recreate IP Office SD Card 144 command to load the IP Office folders and files onto the card from the Manager PC.

5.1.7.9 Recreate IP Office SD Card

This command can be used with a read-writeable SD card on the IP Office Manager PC. It copies the files and folders used by an system when starting. It updates the card with the version of those files installed with the IP Office Manager application. It includes the binary files for the system, external expansion modules and phones. The command also copies all language prompt sets used by Embedded Voicemail.

If the card contains dynamic system files such as SMDR records, they are temporarily backed up by IP Office Manager and then restored after the card is recreated. For the card to be used in a system's **System SD** slot the card must be Avaya SD Feature Key card. The card must be correctly formatted (see Format IP Office SD card 144), however a reformat of an existing working card is not necessary before using recreate to update the card contents.

- The source for the files copied to the SD card are the sub-folders of the ***\Memory Cards*** folder under IP Office Manager's **Working Directory** 134 (normally ***C:\Program Files Avaya\IP Office\Manager***). However, if the Working Directory is changed to a location without an appropriate set of ***\Memory Cards*** sub-folders, the required set of files will not be copied onto the SD card.

1. Note: This process can take up to 20 minutes depending on the PC. Once started the process should not be interrupted.
2. Insert the SD card into a reader slot on the Manager computer.
3. Using Manager, select **File | Advanced | Recreate IP Office SD Card**.
4. On Quick systems, select **IP Office A-Law** or **IP Office U-Law.**
   On Partner systems, **select IP Office PARTNER Edition.**
   On Norstar systems, select **IP Office Norstar Edition**.

   This selection will affect how the IP Office system operates when defaulted with this card present in its **System SD** card slot. The other options available are not used for a Basic Edition system.
5. Browse to the card location and click **OK**.
6. IP Office Manager will prompt whether you want to include Avaya IP Office Web Manager files as part of the recreate process. Those files are necessary if you want to use IP Office Web Manager to manage the IP Office system into which the card will be loaded.
7. Manager will start creating folders on the SD card and copying the required files into those folders.
8. Do not remove the card until the process is completed and Manager displays *"Ready"* in the status bar.

5.1.7.10 Memory Card Command

These commands are used with the memory cards installed in the control unit's **System SD** and **Optional SD** card slots. These command can also be performed from either of the first two extensions in the system.

5.1.7.10.1 Shutdown

This command can be used to shutdown operation of IP500v2 memory cards. This action or a system shutdown 139 must be performed before a memory card is removed from the unit. Removing a memory card while the system is running may cause file corruption.

For IP500v2 systems, shutting down the memory card will disable all services provided by the card including Embedded Voicemail. For IP500v2 systems, features licensed by the memory card will continue to operate for up to 2 hours.

Card services can be restarted by either reinserting the card or using the Start Up 145 command.

5.1.7.10.2 Start Up

This command can be used to restart operation of an IP500v2 memory card that has been shut down 145. The command will start the **Select IP Office** discovery process for selection of the IP Office system.

5.1.7.11 System Status

IP Office System Status is an application that can be used to monitor and report on the status of an IP Office system.

System Status is included on the Avaya System SD memory card and can be start by browsing to the IP address of the system and selecting the System Status link.

5.1.7.12 Add/Display VM Locales

This option is not shown for off-line configuration or configurations loaded from a PC file. Selecting this option displays a list of the Embedded Voicemail prompt languages. Those languages already present on the System SD card or not supported are greyed out. Additional languages can be selected and then uploaded from IP Office Manager to the system.

When editing the system configuration in IP Office Manager, if the locale language selected for the system, a user, a short code or an incoming call route is not already present on the System SD card, IP Office Manager will display an error. **Add/Display VM locales** can then be used to upload the prompts for the required language in order to correct the error.

You can reload languages that are already installed on the System SD card. For example, you may want to reload the languages if new prompts have been added in a maintenance release. To reload existing languages, upgrade the system (File | Advanced | Upgrade 140) with the **Upload System Files** option checked. You can also choose **Upload System Files** from the Embedded File Management utility (File | Advanced | Embedded File Management 143).

The Recreate IP Office SD Card 144 command can be used to locally load all available languages onto an SD card.

5.1.7.13 Initial Configuration

Redisplay the initial configuration menu used during system installation. This allows selection which IP Office operating mode the system should use and sets the basic defaults such as locale and IP address settings.

- **! WARNING**
  Use of this menu erases the existing system configuration and settings.

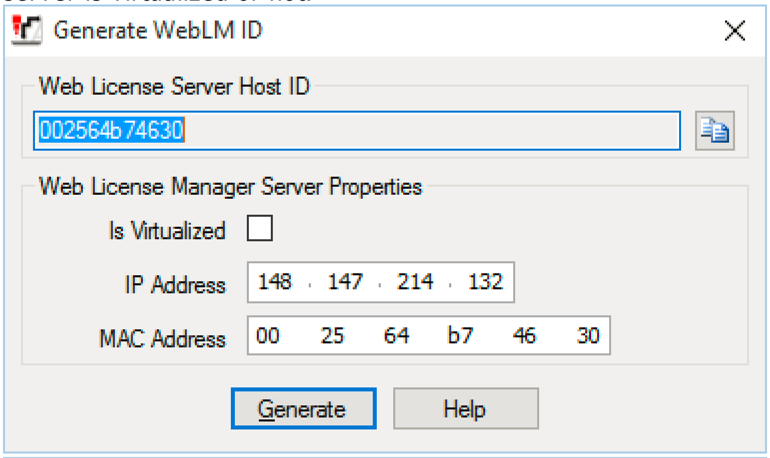
5.1.7.14 Generate WebLM ID

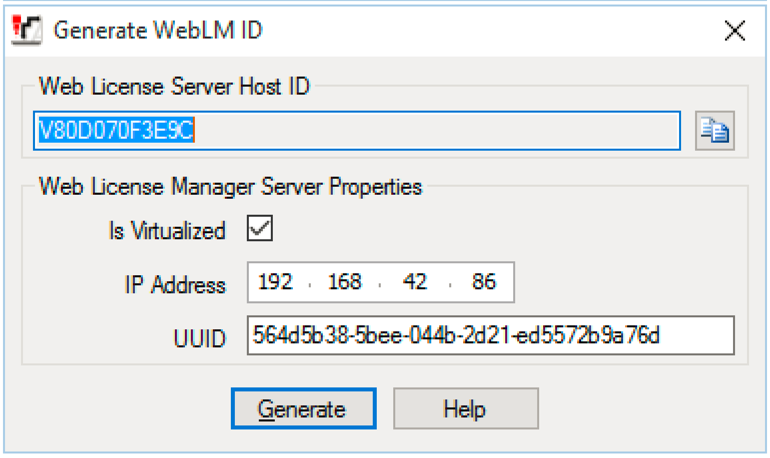
This menu is only used for Server Edition systems.

Any system being upgrading from pre-Release 10 ADI licenses must be migrated to PLDS licenses **before upgrading**. This is done using the files created by the license migration tool 149. However, that tool assumes that the system will also be the licence host. If instead the system is going to use PLDS license hosted by a WebLM server, the system's web license server host ID is required in addition to the files created by the license migration tool. The **Generate WebLM ID** tool provides that additional ID.

To generate the server's Web License Server Host ID:

1. Click **File | Advanced | Generate WebLM ID**. The menu displayed varies depending on whether the server is virtualized or not.





2. Enter the details of the server:

   - **UUID**
     For a virtualized server, the UUID can be obtained as follows:

     a. Using the command line command: *dmidecode -s system-uuid*

     b. From the **uuid.bios** line of the virtual machines vmx file.

     c. From the VSphere client. See http://www-01.ibm.com/support/docview.wss?uid=swg21682150.

3. Click **Generate**.

4. The systems's **Web License Server Host ID** is display in the menu and can be copied.

5.1.7.15 Document History

7th July 2016	3b	Update for IP Office Release 10.0. First source for help.
11th July 2016	3c	Updated list of supported browsers for web management. [110276] Update to include StartTLS field setting. [110569] Update to hunt group modes. [109441]

5.1.8 Exit

The **File | Exit** command exits the Manager application.

5.2 View

5.2.1 Toolbars

This command allows selection of which toolbars should be shown or hidden in configuration mode. A tick mark is displayed next to the name of those toolbars that are currently shown.

5.2.2 Tool Tip

This setting control whether additional tooltips are displayed when Manager is running in Basic Edition.

5.2.3 Advanced View

When there is no configuration loaded in Manager, this command can be used to select the full mode rather than simplified view 31. The full mode is not used by Basic Edition systems, Manager will automatically return to simplified view mode if an Basic Edition system configuration is loaded.

This option is not available when an Basic Edition system configuration is loaded.

5.2.4 Hide Admin Tasks

This settings shows or hides the **Admin Tasks List** 34 available when Manager has a configuration from a system loaded. .

5.2.5 TFTP Log

This command displays the TFTP Log window. This window shows TFTP traffic between Manager and devices that uses TFTP to send and receive files. For example, the TFTP Log below shows an Avaya IP phone requesting and then being sent its software files.

5.3 Tools

5.3.1 Extension Renumber

This tool can be used to change the numbering of user extensions in a system between 2-digit and 3-digit. For 3-digit systems it can also be used to change the numbering of the extensions whereas 2-digits systems use the fixed extension numbers 10 to 57.

It is strongly recommended that these options are only used and changed on a newly installed system. Changing extension numbering affects other services including voicemail and may require extension reconfiguration of hunt groups and trunk call routes.

- **Default Numbering**
  Select whether the systems uses ***2 Digit*** or ***3 Digit*** extension numbering. In 2-digit systems, the user extensions are fixed as 10 to 57. In 3-digit systems the user extension are numbered 100 upwards by default but can be renumbered. In 2-digit mode only 48 extensions are supported, in 3-digit mode a maximum of 100 extensions are supported.

- **Renumber From/Renumber to**
  These options are available for systems set to ***3 Digit*** numbering. They can be used to renumber selected extensions. The extension numbers are restricted to the range 100 to 579.

5.3.2 Import Templates

IP Office Manager can be used to import SIP trunk templates 97 and analog trunk templates. These need to be stored in a specific Manager ***\Templates*** sub-folder.

This command can be used to select a folder containing template files and copy those files into the Manager sub-folder.

The availability of this command is controlled by the **File | Preferences | Visual Preferences | Enable Template Options**.

5.3.3 License Migration

IP Office Release 10.0 and higher only supports licensing via Avaya's Product Licensing and Delivery System (PLDS). This is done via licenses delivered as a non-editable PLDS XML file which is uploaded to the system. The file contains the whole set of licenses for the system. To add or delete a license the whole file must be replaced.

For any system being upgrading from pre-Release 10, you must migrate all the existing licenses to PLDS **before upgrading**. This is done using the license migration tool within IP Office Manager.

The license migration tool extracts all the current license information from the systems configuration and saves it to a file. That file can then be used prepare a software upgrade quote in the Avaya One Source Configurator in order to obtain the required new PLDS XML file.

Notes

- Ensure all licenses are loaded on the system before using the license migration tool to extract the licensing information.
- The license migration tool can only be used with an online configuration.
- The generated file can be read but must not be edited. License migration will fail if the file has been edited.

5.4 Embedded File Management

The contents of the SD memory card used by the system can be viewed through Manager.

- **Embedded Voicemail Files**
  When viewing the memory card, the files related to Embedded Voicemail are visible, however these files are greyed out (ie. cannot be deleted, downloaded or overwritten).
  - Mailbox greetings and messages are shown as ***.clp*** files.
  - The language prompts for Embedded Voicemail functions are stored in separate language sub-folders of **lvmail**.

- **Viewing a Memory Card**
  When **Advanced | Embedded File Management** is selected, the IP Office Manager will go through normal system discovery. When a system is selected, a valid service user name and password for configuration access to that system is requested.
- **Changing the Files View**
  The type of display used in the **Files** pane can be changed by selecting from the **View** menu in the toolbar.
- **Adding Files**
  Files can be added to the card by dragging and dropping or by right-clicking on the Files pane and selecting **Upload** or by using **File | Upload File...**. The IP Office will ask for confirmation if the file already exists on the memory card. The progress of the file upload is then indicated.
- **Deleting Files**
  Existing files can be deleted by right-clicking on them and selecting **Delete**.
- **Downloading Files**
  Files can also be copied from the card by right-clicking on the file and selecting **Download**. Manager will prompt for the download location. Existing files are overwritten if present.
- To exit back to normal configuration operation, select **File | Configuration** from the menu bar. Alternatively, to view the card in another system, select **File | Close File Settings** and then **File | Open File Settings**.

5.4.1 Open File Settings

Select an IP Office system and display the contents of its memory cards if any are present and in use.

5.4.2 Close File Settings

Close the current memory card contents listing without exiting embedded file management mode.

5.4.3 Refresh File Settings

This command can be used to request a file update from the IP Office system.

5.4.4 Upload File

This command can be used to select and upload a file to the memory card in the IP Office system.

5.4.5 Upload System Files

When this command is selected, Manager will upload the software files for IP Office to the System SD card. It includes all IP Office software, phone software and Embedded Voicemail prompts not already present on the System SD card.

- **! WARNING**
  After this command the system will reboot. The reboot will end all calls and services in progress.
  - It copies the binary files used by the IP Office control unit and external expansion modules.
  - It copies the firmware files used by phones supported by the system.
  - For systems configured to running in IP Office Basic Edition, IP Office Basic Edition - PARTNER Mode or IP Office Basic Edition - Norstar Mode mode, the files for IP Office Web Manager are copied.
  - For systems configured to run Embedded Voicemail, the Embedded Voicemail prompts for those supported languages set as the system locale, user locales, incoming call route locales and short code locales are upgraded. In addition the English language prompts are upgraded as follows: **IP Office A-Law/Norstar SD Cards** - UK English, **IP Office U-Law/PARTNER SD Cards** - US English.

5.4.6 Backup System Files

When selected, Manager copies the folders and files from the **System SD** card's ***/primary*** folder to its ***/backup*** folder. Any matching files and folders already present in the ***/primary*** folder are overwritten.

5.4.7 Restore System Files

When selected, Manager copies the folders and files from the **System SD** card's ***/backup*** folder to its ***/primary*** folder. Any matching files and folders already present in the ***/backup*** folder are overwritten.

- **! WARNING**
  After this command the system will reboot. The reboot will end all calls and services in progress.

5.4.8 Upgrade Binaries

This command is available for systems that have an System SD card and Optional SD card installed. When this command is selected, all files except ***config.cfg*** and ***keys.txt*** files in the Optional SD card's ***\primary*** folder are copied to the System SD card.

- **! WARNING**
  After this command the system will reboot. The reboot will end all calls and services in progress.

5.4.9 Upgrade Configuration

This command is available for systems that have an System SD card and Optional SD card installed.

When this command is selected, any ***config.cfg*** and ***keys.txt*** files in the Optional SD card's ***\primary*** folder are copied to the System SD card.

- **! WARNING**
  After this command the system will reboot. The reboot will end all calls and services in progress.

5.4.10 Upload Voicemail Files

Not used with Basic Edition systems.

5.4.11 Copy System Card

This command is available for systems that have a System SD card and Optional SD card installed. When this command is selected, the IP Office will copy the folders and files on its **System SD** card to the **Optional SD** card. Any matching files and folders already present on the **Optional SD** card are overwritten.

This process takes at least 90 minutes and can take longer.

5.4.12 Configuration

This command will exit Embedded File Management and return Manager to configuration editing mode.

Chapter 6.
Appendix: SMDR

6. Appendix: SMDR

The control unit is able to send SMDR (Station Message Detail Reporting) records to the IP address and port specified in the Advanced Parameters 110 settings.

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 157.

Each SMDR record contains call information in a comma-separated format (CSV) format, that is variable-width fields with each field separated by commas. See SMDR Fields 154.

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.

6.1 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.

6.2 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 7. Miscellaneous

7. Miscellaneous

7.1 Management Tools

These tables list the functions and configuration settings accessible from the system administrator tools. They do not cover user administration of their own settings which can also be done using phone based administration or IP Office Web Manager. Note that the names of some features vary depending on which tool is being used for administration.

System Maintenance Functions

	Function	IP Office Manager	IP Office Web Manager	Phone Based Admin	System Status Application
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	–

	Function	IP Office Manager	IP Office Web Manager	Phone Based Admin	System Status Application
Display System Details	**Software Level**	Yes	Yes	Yes	–
	IP Address	Yes	Yes	Yes	–
	Feature Key Number	Yes	Yes	Yes	–
	Installed Hardware	Yes	Yes	–	–
Trunk Templates	**Analog Trunk Templates**	Yes	–	–	–
	SIP Trunk Templates	Yes	–	–	–

System Settings

	Function	IP Office Manager	IP Office 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	Yes	–
	Export	Yes	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	–
	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	–

	Function	IP Office Manager	IP Office Web Manager	Phone Based Admin
	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	IP Office 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	IP Office 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	IP Office 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	IP Office 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	IP Office 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	IP Office 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	IP Office 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	IP Office 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	IP Office Web Manager	Phone Based Admin
DDI	**DID Number**	Yes	Yes	–
	Incoming CLI	Yes	Yes	–
	Destination	Yes	Yes	–

Analog Trunk

	Function	IP Office Manager	IP Office 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	IP Office 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	IP Office 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	IP Office 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	IP Office 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	IP Office 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	–

7.2 What Was New in Release 10.0

The following changes have been made for the IP Office Release 10.0 software.

- **PLDS licensing only**
  IP Office Release 10.0 only supports PLDS licensing. Previous ADI licenses are not supported though existing ADI license entries are still retained in the system configuration until deleted. Existing systems being upgraded to Release 10.0 must go through a license migration process before being upgraded.

  - License migration is done by obtaining a license migration file using IP Office Manager (**Tools | License Migration**) and then submitting that file to Avaya. Note that this process also migrates any virtual licenses entitlements the system has to equivalent PLDS licenses but now associated with the feature key. Only upgrade the system once the replacement PLDS license file for the system has been obtained. Note that following license migration, Avaya will delete all records of any ADI license entitlements it holds for that feature key.

7.3 What Was New in Release 6.1

- **Phantom Users** 17
  Previously user settings were only created and configurable for the physical extensions present in the system (and excluding ports 7 and 8 on ETR6 cards). User settings are now created for all possible user extensions, regardless of whether a matching physical extension port is present or not.
  - Calls to a phantom extension number go directly to that user's mailbox. This applies for normal dialing, DID call routing, line coverage, transfers and routing from an auto-attendant.
  - Within Manager, phantom extensions are indicated by a # shown in front of the extension number (the # is not part of the actual dialed number for the extension.
- **Auto Attendant Enhancements**
  The following changes have been made to auto attendant support for Release 6.1.
  - **9 Auto Attendants**
    Up to 9 auto attendants are now supported.
    - **Transfer to Auto Attendant Action**
      This additional menu action allows calls to be routed from one auto attendant to another. When this occurs, only the menu prompt of the new auto attendant is played.
  - **Language Selection**
    Each auto attendant can be configured with a language selection. The selected language controls the prompts used by the auto attendant actions where applicable.
  - **Time Profile Dependant Menu Actions**
    In addition to controlling with initial greeting is played to callers, each auto attendants time profiles now also control which set of menu actions are available to the callers with the appropriate time dependant menu actions greeting.
  - **Selectable Night Service Mode**
    Previously when the system was put into night service, the auto attendant switched to its out of hours greeting. Each auto attendant now has a **Follow Night Service** setting. If selected, the previously behavior still applies. If not selected, when the system is put into night service, the auto attendant continues to follow its own time profile settings.
  - **Picking Up Auto Attendant Calls**
    Bridging into a call being handled by an auto attendant drops the auto attendant from the call.
  - **Emergency Greeting**
    Each auto attendant can have a recorded emergency greeting and a setting for whether that greeting is active or not. When active, the emergency greeting is played before any of the other auto attendant greetings. When an emergency greeting is active, a warning is displayed on a selected alarm extension. Access to record and either enable or disable an emergency greeting requires entry of the system code.
    - **Transfer to Emergency Greeting Action**
      This additional menu action allows the caller to play and record the emergency greeting. It also allows them to select whether the greeting is active or not. If a system password has been set, it is used to restrict access to this option.
- **Line Enhancements**
  - **Unique Line Ringing**
    Each incoming line, channel or DID can be assigned a ringing pattern. That pattern is then used for incoming calls on that line unless overridden by the user's setting.
  - **Assigning Lines to Auto Attendants**
    Previously assignment of a line to an auto attendant could only be done through the IP Office Manager. This option can now be selected through the administrator menus on the first two extensions in the system.
- **User Setting Enhancements**
  The following changes have been made to user support for Release 6.1.
  - **Immediate Voicemail Coverage**
    The user setting for VMS Cover Ring can now be set to ***0*** for immediate voicemail.
  - **Transfer Return Extension**
    For users who have transfer return enabled, a different extension destination for the return calls can now be specified.
  - **Override Line Ringing**
    For selected users the new Unique Line Ringing can be overridden.
- **Language Control**
  Previously only individual users could be configured with a language selection. For Release 6.1 the default language selection can now be done at the system level. In addition a language setting can be applied to each auto attendant.

- **Embedded Voicemail Enhancements**
  The following changes have been made to the Embedded Voicemail provided to Release 6.1 users.
  - **Caller Post Message Options**
    After leaving a mailbox message, callers can now press **#** rather than hanging up immediately. The caller will hear a prompt informing them whether the message has been saved or whether the messages was too short (less than 3 seconds) and so was not saved.
  - **Skip Your Mailbox Greeting**
    Caller's can skip your mailbox greeting by pressing 1. Instead they immediately hear the tone for the start of recording.
  - **Mark Messages as New**
    You can now change an old or saved message's status back to new while it is being played or just after it has played by dialing ***06***. The message waiting light is relit. However if voicemail email is being used no new message email is sent.
- **Memory Card Commands**
  Before a memory card, whether the System SD card or the Optional SD card is removed from a system, the card should be shutdown. The command to shutdown or startup a memory card can now be performed from a system phone (the first two extensions on the system). Previously the actions could only be done using the Manager application.
- **Display System Information**
  Any extension with a suitable display (ETR18D, ETR34D, 1408 or 1416) can display basic system details. They do this by pressing FEATURE and then dialing the appropriate code for system software level (590), IP address (591) or feature key number (592).
- **Modem Access**
  The V32 analog modem service supported by the first analog port in the system can now be accessed by alternate routes other than incoming calls on that line. Extension 76 is used for modem access and can be specified as the destination in an auto attendant, DID call map or SIP trunk call map.
- **Reset Voicemail Password**
  This system administration feature has been moved from #324 to #325. #324 is now used for override line ringing.
- **One Touch Transfer**
  Users can initiate a transfer using a number of buttons programmed to internal destinations without having to first press the **TRANSFER** button.
- **Analog Trunk Configuration**
  Using IP Office Manager, each analog trunk can be configured between Loop Start or Loop Start ICLID.

7.4 What Was New in Release 7.0

The following additional features are available for an Basic Edition system running IP Office Release 7.0 software.

- **System Mode: Key System or PBX System**
  Systems can now be configured as either key or PBX systems.

- **2/3 Digit Extension Numbering**
  Systems can now be configured to use 2 digit or 3 digit extension numbering. When configured to use 2 digit extension numbering, those numbers are in the range 10 to 57 and cannot be changed. When configured to use 3 digit extension numbering, those numbers are in the range 100 to 579 but can be changed using IP Office Manager (the defaults are 100 to 199).

- **Up to 100 Extensions**
  In 3-digit numbering mode, the systems can now support up to 100 extensions.

- **DND Exceptions**
  Using IP Office Manager, the system administrator can now see and if necessary change the do not disturb status of a user. In addition the system administrator can now edit do not disturb exception numbers for the user.

- **Country and Language Control**
  The system locale settings has been separated into Country and Language values. Selecting a country will also set a default language, however that can be overridden if required using the separate language setting.

- **Embedded Voicemail Enhancements**
  The following changes have been made to Embedded Voicemail operation.

  - **EVM Outcalling**
    Through the mailbox prompts menu, mailbox users can now enable voicemail outcalling. This will call a number that they configured when their mailbox contains new messages.

  - **EVM License**
    The additional ports license that enables 2 additional ports now also enables 5 hours of additional prompt and message storage.

- **Trunk Templates**
  IP Office Manager can be used to import SIP and analog trunk settings from supplied trunk templates.

- **Avaya Nortel Phone Support**
  Systems can now use the TCM8 base card and DS16A or DS30A external expansion modules to support the addition of Avaya Nortel T-Series and M-Series phones.

- **PRI Trunk Support**
  Systems can now be installed with a IP500 PRI-U trunk daughter card. Only one single-port trunk card can be installed, allowing up to 30 E1 PRI trunk channels. Note that BRI and PRI trunks are not supported in the same system.

- **BRI Trunk Support**
  Systems can now be installed with IP500 BRI trunk daughter cards. Multiple cards can be installed up to a maximum capacity of 12 BRI trunk channels. This includes the IP500 Combination card pre-fitted with a BRI trunk daughter card. Note that BRI and PRI trunks are not supported in the same system.

- **Avaya 9500 Series Phone Support**
  Avaya 9500 Series phones are supported. They connect to the system using Digital Station (DS) ports in the same way as existing 1400 Series phones.

- **Button Programming**
  The following changes have been made to button programming features:

  - **Absent Text Inspect**
    The existing **Absent Text** button function used to set and clear a user's absent text can now also be used to see the absent text of another user without having to make a call to that user.

  - **Message Waiting Indicator Inspect**
    This feature can be assigned to a programmable button. In conjunction with **Auto Dial - Intercom** buttons for other user extensions, it can be used to see the current message waiting indicator status of those users.

7.5 What Was New in Release 8.0

The following changes and additional features have been made for an Basic Edition system running IP Office Release 8.0 software.

- **Name Changes**
  The name for these systems has changed from IP Office Essential Edition - Norstar Mode, IP Office Essential Edition - PARTNER Mode and IP Office Essential Edition - Quick Mode to IP Office Basic Edition - Norstar Mode, IP Office Basic Edition - PARTNER Mode and IP Office Basic Edition. This is to clarify the operational and functional distinction from IP Office Essential Edition, IP Office Preferred Edition and IP Office Advanced Edition modes.

- **Changed Default Administrator Password**
  The default password for the Administrator account used for access to the system using IP Office Manager has changed from ***password*** to ***Administrator***. This applies to new systems and to systems that are defaulted.

- **Phone Based Administration During a Call**
  The ability to enter and use phone base administration menus during a call is now supported on all phone types capable of phone based administration. Previously this option was limited to ETR, M-Series and T-Series phones. It includes entering system, centralized and personal telephone administration.

- **IP Office Web Manager**
  Systems can now be configured via web browser. The default access is via the same **Administrator** account as used for IP Office Manager and via a new default **BusinessPartner** account. The **BusinessPartner** account can be used to create additional user accounts for IP Office Web Manager and to set what aspects of the configuration those additional accounts can change.

- **Analog Trunk Unsupervised Disconnect Operation**
  In areas where no disconnect clear signalling or reliable tone disconnect is available, the system operation can be set for unsupervised disconnect on analog trunks. When enabled, unsupervised transfers and trunk to trunk transfers to analog trunks are not allowed.

- **Intuity Mode Embedded Voicemail Support**
  Previous software releases have used IP Office mode key presses to navigate the Embedded Voicemail menus. The system now supports Intuity mode key presses. The mode used by the system is selectable. Intuity mode is the default used for new systems.

- **Prompt Set Upgrade Changes**
  Previously for IP500 V2 systems, when upgrading the system files all files were updated including the sets of prompt files for all languages supported by Embedded Voicemail. For IP Office Release 8.0, the following changes have been made:

  - When upgrading the system files, the current configuration of the system is used to determine which files to include in the upgrade in addition to the system and phone firmware files. This applies when using the **Upload System Files** option during a system upgrade (File | Advanced | Upgrade 140) or when using **Upload System Files** within embedded file management (File | Advanced | Embedded File Management 143),

    - Embedded voicemail prompts are only upgraded if the system is currently configured to use Embedded Voicemail. English language prompts are upgraded as follows: **IP Office A-Law/Norstar SD Cards** - UK English, **IP Office U-Law/PARTNER SD Cards** - US English. Other languages are upgraded if they match the locale languages of the system locale, user locales, incoming call route locales or short code locales in the system's configuration.

    - The files for IP Office Web Manager are only upgraded for systems configured to IP Office Basic Edition, IP Office Basic Edition - PARTNER Mode or IP Office Basic Edition - Norstar Mode mode.

  - When editing the configuration of a system that uses Embedded Voicemail, when the locale of the system, a user, a short code or an incoming call route is change, IP Office Manager will display a warning if the matching set of prompts on the System SD card have not be upgraded.

  - A new option within IP Office Manager, **Add/Display VM locales** 145 can be used to display the upgraded languages on the SD card and to upload an additional language or languages.

  - The **Recreate IP Office SD Card** 144 command still adds all the available Embedded Voicemail language prompts for all languages and all files for IP Office Web Manager.

- **Mobile Twinning No Longer Supported**
  The mobile twinning features available in previous releases are no longer supported. On existing systems upgraded to Release 8.0, the related settings such a **Mobile Twin** buttons are converted to **Remote Call Forward** during the upgrade. Any licenses added for mobile twinning are retained but not used. Customers wanting to use mobile twinning will need to upgrade to IP Office Essential Edition.

- **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.

- **SIP Trunk Enhancements**
  A number of features have been added for SIP trunk operation.

- **Trunk Service Status Checking**
  The system can regularly check the availability of a SIP trunk in order to ensure that outgoing calls are not delayed by attempting to use a trunk which is no longer in service.

- **Call Routing Method Control**
  Which part of the incoming call information should be used as the incoming number can now be selected. The options are to match either the ***Request URI*** or the ***To Header***.

- **Associate Method Control**
  The method by which the system associates a incoming SIP call with a particular SIP trunk can be configured.

- **Fax Transport Support**
  Support for fax calls can be enabled if also supported by the line provider. G711 and/or T38 fax support can be selected.

- **PRACK/100 rel Support**
  This feature is sometimes called 'early media support' and allows features such as in-band call tones and call progress announcements to be played while the call connection process is still in progress.

Chapter 8. Document History

8. Document History

Date	Issue	Change Summary
7th July 2016	**08b**	Update for IP Office Release 10.0. First source for help.
26th September 2016	**08c**	Update to hunt group modes. [109441]
20th February 2018	**09a**	Conversion to new authoring software and output format.
23rd July 2020	**10a**	IP Office 11.1.0.1: Addition of 911-View/Emergency View 119 button feature.

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Template: 30th May 2018