IP Office 4.1

Manager: 04. Telephony Features

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Telephone Features

Advice of Charge (ISDN Only)

IP Office 4.0 supports advice of charge (AOC) on outgoing calls to ISDN exchanges that provide AOC information. It supports AOC during a call (AOC-D) and at the end of a call (AOC-E). This information is included in the IP Office Delta Server 5.2.5 and higher output.

AOC is only supported on outgoing ISDN exchange calls. It is not supported on incoming calls, reverse charge calls, QSIG and non-ISDN calls. Provision of AOC signalling will need to be requested from the ISDN service provider and a charge may be made for this service.

For users, display of AOC information is only supported on T3 phones, T3 IP phones and Phone Manager.

- The user who makes an outgoing call is assigned its charges whilst they are connected to the call, have the call on hold or have the call parked.
- If AOC-D is not available, then all indicated by AOC-E are assigned to the user who dialed the call.
- If AOC-D is available:
  - If the call is transferred (using transfer, unpark or any other method) to another user, any call charges from the time of transfer are assigned to the new user.
  - If the call is manually transferred off-switch, the call charges remain assigned to the user who transferred the call.
  - If the call is automatically forwarded off switch, subsequent call charges are assigned to the forwarding user.
  - AOC-D information will only be shown whilst the call is connected. It will not be shown when a call is parked or held.
  - Call charges are updated every 5 seconds.
- For conference calls all call charges for any outgoing calls that are included in the conference are assigned to the user who setup the conference, even if that user has subsequently left the conference.

Enabling AOC Operation

1. **Set the System Currency**
   The **Default Currency** setting on the **System | Telephony** tab is by default set to match the system locale. Note that changing the currency clears all call costs stored by the IP Office except those already logged through Delta Server.
2. **Set the Call Cost per Charge Unit for the Line**
   AOC is indicated by the ISDN exchange in charge units rather than actual cost. The cost per unit is determined by the IP Office using the **Call Cost per Charge Unit** setting which needs to be set for each line. 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.
3. **Enable User AOC Display**
   By default users do not see call charges. The setting **Display Charges** on the **User | T3 Options** tab is used to switch this option on or off. Note that the display of AOC information is only supported on T3 phones.
4. **Applying a Call Cost Markup**
   It may be a requirement that the call cost per unit applied to a user calls has a mark-up (multiplier) applied to it. This can be done using the Call Cost Markup setting on the **User | Telephony** tab. The field is in units of 1/100th, for example an entry of 100 is a markup factor of 1.

AOC Short Codes
A number of short code features exist that can be used with AOC. These features can only be used with T3 terminals.

- **AOC Previous Call**
  Displays the call costs of the user's previous call if AOC information was provided with that call.
- **AOC Total**
  Display the cumulative total cost of the user's calls for which AOC information is available.
- **AOC Reset Total**
  Set the cumulative total (units and cost) for the user's calls back to zero.

Malicious Call Tracing (MCID)

MCID (Malicious Caller ID) is an ISDN feature. It is supported by IP Office 4.0+ on BRI and PRI trunks to ISDN service provider who provide MCID.

When used it instructs the ISDN exchange to perform a call trace on the users current or last call and to keep a record of the call trace for the legal authorities. Trace information is not displayed by or provided to the IP Office and IP Office phones.

The use of MCID is subject to local and national legal requirements that will vary. The feature may also not be enabled until specifically requested from the service provider. You should consult with your ISDN service provider and with appropriate legal authorities before attempting to use MCID.

Successful use of this feature is indicated by a tone on the phone and a "User Registered" message on T3 phones.

Activating MCID

1. **Liaise with the ISDN Service Provider**
   MCID should not be used with first confirming its usage with the ISDN service provider.
2. **Enabling MCID Call Tracing on a Line**
   BRI and PRI lines include a **Support Call Tracing Option**. which by default is off.
3. **Enabling MCID Call Tracing for a User**
   First the user must be allowed to use call tracing. Each user has a **Can Trace Calls** option on their **User | Telephony** tab. This option is off by default.
4. **Providing an Active MCID Control**
   The user needs to be provides with a mechanism to trigger the MCID call trace at the exchange. This can be done using either a short code or a programmable button.
   - **MCID Activate Button**
     The action **MCID Activate** (**Advanced | Miscellaneous | MCID Activate**) can be assigned to a programmable buttons. It allows a malicious call trace to be triggered during a call.
   - **MCID Activate Short Codes**
     The feature **MCID Activate** can be used to create a short code to triggering a malicious call trace.

Call Barring

Call barring can be applied in a range of ways.

- **Barring a User From Making Any External Calls**
  For each user, **Outgoing Call Bar** can be selected (**User | Telephony**) to stop that user from making any outgoing calls.
- **Barring Particular Numbers/Number Types**
  IP Office short codes are used to match user dialing and then perform a specified action. Typically the action would be to dial the number to an external line. However, short codes that match the dialing of particular numbers or types of numbers can be added and set to another function such as **Busy**. Those short codes can be added to a particular user, to a User Rights associated with several users or to the system short codes used by all users.
  - The IP Office allows short codes to be set at user, user rights, system and least cost route. These have a hierarchy of operation which can be used to achieve various results. For example a system short code for a particular number can be set to busy to bar dialing of that number. For a specific user, a user short code match to the same number but set to Dial will allow that user to override the system short code barring.
- **Using Account Codes**
  The IP Office configuration can include a list of account codes. These can be used to restrict external dialing only to users who have entered a valid account code.
  - **Forcing Account Code Entry for a User**
    A user can be required to enter an account code before the system will return dialing tone. The account code that they enter must match a valid account code stored in the IP Office configuration. The setting for this is **Forced Account Code** (**User | Telephony | Forced Account Code**).
  - **Forcing Account Code Entry for Particular Numbers**
    Each IP Office short code has a **Force Account Code** option. Again the account code entered must match a valid account code stored in the IP Office configuration. for the call to continue.
- **Barring External Transfers and Forwards**
  A user cannot forward or transfer calls to a number which they cannot normally dial. In addition there are controls which restrict the forwarding or transferring of external calls back off-switch. See Off-Switch Transfer Restrictions.

Caller Display

Caller display displays details about the caller and the number that they called. On internal calls the IP Office provides this information. On external calls it uses the Incoming Caller Line Identification (ICLID) received with the call. The number is also passed to IP Office applications and can be used for features such as call logging, missed calls and to make return calls.

Analog extension can be configured for caller display via the IP Office configuration (**Extension | Extn | Caller Display Type**).

- **Adding the Dialing Prefix**
  Some IP Office systems are configured to require a dialing prefix in front of external numbers when making outgoing calls. When this is the case, the same prefix must be added to the ICLID received to ensure that it can be used for return calls. The prefix to add is specified through the **Prefix** field of each line.
- **Directory Name Matching**
  The IP Office configuration contains a directory of names and numbers. If the ICLID of an incoming call matches a number in the directory, the directory name is associated with that call and displayed on suitable receiving phones.
  - The IP Office SoftConsole and Phone Manager applications also have directories that can be used for name matching. If a match occurs, it overrides the system directory name match for the name shown by that application.
- **Note: Caller ID can not be forwarded**
  If an extension is forwarded or transferred to another extension, the Caller ID of the forwarding extension is received, not the Caller ID of the original call.

Extended Length Name Display

In some locales, it may be desirable to change the way names are displayed on phones in order to maximize the space available for the called or calling name. There are two hidden controls which can be used to alter the way the IP Office displays calling and called information.

These controls are activated by entering special strings on the **Source Numbers** tab of the **NoUser** user. These strings are:

- **LONGER_NAMES**
  This setting has the following effects:
  - On DS phones, the call status display is moved to allow the called/calling name to occupy the complete upper line and if necessary wrap-over to the second line.
  - For all phone types:
    - On incoming calls, only the calling name is displayed. This applies even to calls forwarded from another user.
    - On outgoing calls, only the called name is displayed.
- **HIDE_CALL_STATE**
  This settings hides the display of the call state, for example **CONN** when a call is connected. This option is typically used in conjunction with **LONGER_NAMES** above to provide additional space for name display.

Call Intrusion

Call intrusion allows a user to join another users existing conversation. Once the intrusion has occurred, all parties can hear and talk to each other. Note that intruding uses IP Office conference resources.

The ability to intrude is controlled by two IP Office configuration settings, the **Can Intrude** setting of the person intruding (**User | Telephony | Can Intrude**) and the **Cannot Be Intruded** setting of any other users in the call (**User | Telephony | Cannot Be Intruded**). By default no users can intrude and all users cannot be intruded.

- **Bridging**
  Users with call appearance buttons may be able to bridge into other calls. This is similar to intrusion but subject to different operation. Refer to the IP Office Key & Lamp Operation Manual.
- **Privacy**
  IP Office 4.0+ provides privacy features that allow users to indicate that a call cannot be intruded on. See **Private Calls**.

Below is an example of a short code, which can be used to attempt call intrusion. Using it the intruder would dial *90*N#, replacing the N with the extension number of the user into whose call they need to intrude.

- **Short Code:** *90**N*#
- **Telephone No:** *N*
- **Feature:** CallIntrude

The **Dial Inclusion** short code feature can be used instead of **Call Intrude**. It allows the intruder and the intrusion target to talk without the third party hearing them. During this type of intrusion, all parties hear a repeated intrusion tone. When the intruder hangs-up the original call parties are reconnected.

Call Tagging

Call tagging associates a text string with a call. That string remains with the call during transfers and forwards including across a Small Community Network (SCN).

On Avaya display phones, the text is shown whilst a call is alerting and is then replace by the calling name and number when the call is connected. On analogue phones with a caller ID display, the tag text replace the normal caller information.

IP Office applications such as Phone Manager and SoftConsole display any call tag associated with a call. If the call is parked, the tag is shown on the call park slot button used.

A call tag can be added when making a call using Phone Manager or SoftConsole. A tag can also be added to a call by an Voicemail Pro Assisted Transfer action.

Private Calls

IP Office 4.0 and higher provides features that allow users to make calls are being private. Private calls cannot be recorded, bridged into, intruded on or silently monitored. Use of private calls can be changed during a call. Enabling privacy during a call will stop any current recording, intrusion or monitoring.

Privacy only applies to the speech part of the call. Call details are still recorded in the IP Office Delta Server SMDR output and other IP Office call status displays.

Note that use of private calls is separate from the user's intrusion settings. If a user is set to **Cannot be Intruded**, switching private calls off does not affect that status. To allow private calls to be used to fully control the user status, **Cannot be Intruded** should be disabled for that user.

- **Button Programming**
  The button programming action **Advanced | Call | Private Call** can be used to switch privacy on/off. Unlike the short code features it can be used during a call to apply or remove privacy from current calls rather than just subsequent calls. On suitable phones the button indicates the current status of the setting.
- **Short Codes**
  A number of short code features are available for privacy.
  - **Private Call**
    Short codes using this feature toggle private status on/off for the user's subsequent calls.
  - **Private Call On**
    Short codes using this feature enable privacy for all the user's subsequent calls until privacy is turn off.
  - **Private Call Off**
    Short codes using this feature switch off the user's privacy if on.

Call Pickup

Call pickup allows a user to answer a call ringing at another phone.

The following default short codes can be used:

- ***30 - Call Pickup Any**
  Answers the longest ringing call on the IP Office system. On large IP Office systems it is recommended that this short code is removed as it becomes difficult for users to predict which call they are answering.
- ***31 - Call Pickup Group**
  Pickup the longest ringing call to the hunt groups of which the user is a member.
- ***32*N# - Call Pickup Extn**
  Pick up the call ringing at a specific extension. When dial, ***N*** is replaced by the extension number.
- ***53*N# - Call Pickup Members**
  Pick up any call ringing on another extension that is a member of the Hunt group specified. The call picked up does not have to be a hunt group call. When dial, ***N*** is replaced by the hunt group extension number.

Call Waiting

Call waiting allows a user who is already on a call to be made aware of a second call waiting to be answered.

User Call Waiting

Call waiting is primarily a feature for analog extension users. The user hears a call waiting tone and depending on the phone type, information about the new caller may be displayed. The call waiting tone varies according to locale.

For Avaya digit phone with multiple call appearance buttons, call waiting settings are ignored as additional calls are indicated on any free call appearance button.

To answer a call waiting, either end the current call or put the current call on hold, and then answer the new call. Hold can then be used to move between the calls.

Call waiting for a user can be enabled through the IP Office configuration (**User | Telephony | Call Waiting On**), through the Phone Manager application and through programmable phone buttons.

Call waiting can also be controlled using short codes. The following default short codes are available when using Call Waiting.

- ***15 - Call Waiting On**
  Enables call waiting for the user.
- ***16 - Call Waiting Off**
  Disables call waiting for the user.
- ***26 - Clear Call and Answer Call Waiting**
  Clear the current call and pick up the waiting call.

Hunt Group Call Waiting

Call waiting can also be provided for hunt group calls (**Hunt Group | Hunt Group | Call Waiting On**). The hunt group type must be ***Collective***.

On pre-IP Office 4.0 systems, all the group members must have their own call waiting setting switched on, this no longer applies for IP Office 4.0+.

On phones with call appearance buttons, the call waiting indication takes the form of an alert on the next available call appearance button. On other phones, call waiting indication is given by a tone in the speech path (the tone is locale specific).

- For pre-IP Office 4.0, all the users in the group must also have their own **Call Waiting** setting set to ***On***.
- For IP Office 4.0+, the user's own **Call Waiting** setting is overridden when they are using a phone with call appearances. Otherwise the user's own **Call Waiting** setting is used in conjunction with the hunt group setting.

Parking Calls

Parking a call is an alternative to holding a call. A call parked on the IP Office system can be retrieved by any other user if they know the system park slot number used to park the call. When the call is retrieved, the action is known as **Unpark Call** or **Ride Call**. While parked, the caller hears music on hold if available.

Each parked call requires a park slot number. Attempting to park a call into a park slot that is already occupied causes an intercept tone to be played. Most park functions can be used either with or without a specified park slot number. When parking a call without specifying the park slot number, the system automatically assigns a number based on the extension number of the person parking the call plus an extra digit 0 to 9. For example if 220 parks a call, it is assigned the park slot number 2200, if they park another call while the first is still parked, the next parked call is given the park slot number 2201 and so on.

The **Park Timeout** setting in the IP Office configuration (**System | Telephony | Park Timeout**) controls how long a call can be left parked before it recalls to the user that parked it. The default time out is 5 minutes. Note that the recall only occurs if the user is idle has no other connected call.

There are several different methods by which calls can be parked and unparked. These are:

Using Short Codes

The short code features, **ParkCall** and **UnparkCall**, can be used to create short codes to park and unpark calls respectively. The default short codes that use these features are:

- ***37*_N#_** - Parks a call in park slot number ***N***.
- ***38*_N#_** - Unparks the call in park slot number ***N***.

Using the Phone Manager and SoftConsole Applications

The Phone Manager and SoftConsole applications all support park buttons. Phone Manager provides 4 park slot buttons, numbered 1 to 4. SoftConsole provides 16 park slot buttons numbered 1 to 16 by default. In both cases the park slot number for each button can be changed if required. Clicking on the buttons allows the user to park or unpark calls in the park slot associated with each button.

In addition, when a call is parked in one of those slots by another user, the application user can see details of the call and can unpark it at their extension.

Using Programmable Buttons

The **Call Park** feature can be used to park and unpark calls. If configured with a specified park slot number, the button can be used to park a call in that slot, unpark a call from that slot and will indicate when another user has parked a call in that slot. If configured without a number, it can be used to park up to 10 calls and to unpark any of those calls.

Phone Defaults

Some telephones support facilities to park and unpark calls through their display menu options (refer to the appropriate telephone user guide). In this case parked calls are automatically put into park slots matching the extension number.

Ring Back When Free

Also called 'callback' and 'automatic callback'. If the user called is busy and you want to be informed when they become free, you can dial any digit while listening to the busy tone and then hang up. This sets a **Ring Back When Free**. When the busy extension becomes free, the system will ring your telephone when also idle and, when answered, call the original target extension.

Note

- If the extension called has multiple call appearance buttons, you will not receive busy until all its call appearance buttons are in use.
- You must be idle (have no call connected or held) to receive the ringback.

This feature can also be set via the followings methods. These options also allow a ringback to be set when the target just rings, often known as a ringback when next used.

- **Using a button programmed to the Ring Back When Free function**
  If the button has status indication, it will show that a ring back has been set. The button can also be pressed again to cancel the ringback.
- **Use a short code set to the Ring Back When Free function**
  A short code set to this function can be used to set a ringback on any extension without having to actually make a call.
- **IP Office Phone Manager**
  This application provides the user with a button to set a ringback.
- Analog Phones
  Analog phone users can set a ringback when free by pressing any DTMF key before hanging up when they hear busy tone.

Message Waiting Indication

Message waiting indication (MWI) or a message lamp is supported for a wide variety of phones. It is used to provide the user with indication of when their voicemail mailbox contains new messages. It can also be configured to provide them with indication when selected hunt group mailboxes contain new messages.

Avaya digital and IP phones all have in-built message waiting lamps. Also for all phone users, the IP Office Phone Manager application provides message waiting indication.

Analog Phone Message Waiting Indication

For analog phones, IP Office 3.1 provides support for a variety of analog message waiting indication (MWI) methods. Those methods are **51V Stepped**, **81V**, **101V** and **Line Reversal**. The **101V** method is only supported when using a Phone V2 expansion module.

81V is typically used in European countries. **51V Stepped** is used in most other countries. However the actual method used for a particular model of analog phone should be confirmed with the phone manufacturer's documentation.

The method used for an individual analog extension is set for the **Extn | Extn | Message Waiting Lamp Indication Type** field. This field also provides options for None (no MWI operation) and On. On selects a default message waiting indication method based on the system locale.

'On' Method	Locale
81V	Belgium, Denmark, Finland, France, Germany, Greece, Hungary, Iceland, Italy, Netherlands, Norway, Poland, Portugal, Russia, Saudi Arabia, Sweden, Switzerland, United Kingdom.
51V Stepped	Argentina, Australia, Brazil, Canada, Chile, China, Columbia, Japan, Korea, Mexico, New Zealand, Peru, South Africa, Spain, United States.

- For the United Kingdom system locale (***eng***), the default **Caller Display Type** (***UK***) allows updates of an analog phone's ICLID display whilst the phone is idle. The IP Office uses this facilities to display the number of new messages and total number of messages in the users own mailbox. This feature is not supported with other Caller Display Types.

Hunt Group Message Waiting Indication

By default no message waiting indication is provided for hunt group voicemail mailboxes. Message waiting indication can be configured by adding an **H** entry followed by the hunt groups name to the Source Numbers tab of the user requiring message waiting indication for that hunt group. For example, for the hunt group Sales, add ***HSales***. Hunt group message waiting indication does not require the user to be a member of the hunt group

Ring Tones

Ring tones can be defined in the following terms:

- **Distinctive Ringing - Inside, Outside and Ringback**
  A distinctive ring tone can be given for each of the different call types: an internal call, an external call and a ringback calls (voicemail calls, ringback when free calls, calls returning from park, hold or transfer).
  - The distinctive ringing patterns used for all non-analog phones are fixed as follows:
    - **Internal Call:** Repeated single-ring.
    - **External Call:** Repeated double-ring.
    - **Ringback Call:** Repeated single-ring followed by two short rings.
  - For analog extensions, the ringing pattern used for each call type can be set through the IP Office configuration in Manager. This is done using the settings on **System | Telephony** and or **User | Telephony** tabs.
  - For non-analog extension the ringing pattern used for each call type by the IP Office is not configurable.
- **Personalized Ringing**
  This term refers to control of the ringing sound through the individual phones. For non-analog phones, while the distinctive ringing patterns cannot be changed, the ringer sound and tone may be personalized depending on the phone's own options. Refer to the appropriate IP Office telephone user guide.

Analog Phone Ringing Patterns

For analog phone users, the distinctive ringing pattern used for each call type can be adjusted. From the **System | Telephony** tab, the default ring tone for each call type can be configured. The setting for an individual user associated with the analog extension can be altered from the system default through the **User | Telephony** tab.

Note that changing the pattern for users associated with fax and modem device extensions may cause those devices to not recognize and answer calls.

The selectable ringing patterns are:

- **RingNormal**
  This pattern varies to match the **Locale** set in the **System | System** tab. This is the default for external calls.
- **RingType1:** 1s ring, 2s off, etc. This is the default for internal calls.
- **RingType2:** 0.25s ring, 0.25s off, 0.25s ring, 0.25s off, 0.25s ring, 1.75s off, etc. This is the default for ringback calls.
- **RingType3:** 0.4s ring, 0.8s off, …
- **RingType4:** 2s ring, 4s off, …
- **RingType5:** 2s ring, 2s off, …
- **RingType6:** 0.945s ring, 4.5s off, …
- **RingType7:** 0.25s ring, 0.24 off, 0.25 ring, 2.25 off, …
- **RingType8:** 1s ring, 3s off, …
- **RingType9:** 1s ring, 4s off, …
- **RingType0:** Same as **RingNormal** for the United Kingdom locale.
- **Default Ring:** Shown on the **User | Telephony** tab. Indicates follow the settings on the **System | Telephony** tab.

Music on Hold (MOH)

The IP Office can provide music on hold (MOH) in from either an internally stored file or from an externally connected audio input.

- **Legal Requirements**
  You must ensure that any MOH source you use complies with copyright, performing rights and other local and national legal requirements.
- **Internal Music on Hold File**
  The IP Office can use an internal music on hold file that it stores in its non permanent memory. If the IP Office loses power or is restarted, the file is loaded as follows:
  - Following a reboot, the IP Office will try using TFTP to download a file called ***holdmusic.wav***. The file properties should be: PCM, 8kHz 16-bit, mono, maximum length 30 seconds.
  - The initial source for download is the system's configured TFTP server (**System | System | TFTP Server IP Address**). The default for this is a broadcast to the local subnet for any PC running a TFTP server.
  - Manager acts as a TFTP server while it is running. If Manager is used as the TFTP server then the ***holdmusic.wav*** file should be placed in the Manager applications working directory.
  - If no successful TFTP download occurs, the IP Office will automatically look for a **holdmusic.wav** file on the control unit's compact flash memory card if present and will download that file. (Small Office Edition, IP406 V2 and IP500 control units with IP Office 3.1 or higher)
  - If IP Office has not loaded a hold music file it will retry loading a hold music file approximately every five minutes.
  - If an internal music on hold file is downloaded, the IP Office will automatically write a copy of that file to its compact flash memory card if present. This will overwrite any existing music on hold file stored on that card. (Small Office Edition, IP406 V2 and IP500 control units with IP Office 3.1 or higher)
  - If an internal music on hold file is downloaded, that file is used and overrides any external music on hold source if also connected.
  - All the above operation can be cancelled by selecting **Use External Music on Hold** (**System | Telephony**) and restarting the IP Office.
- **External MOH:**
  An external music source can be connected to the IP Office control unit. Connect a line out audio source to the 3.5mm port marked AUDIO on the back of the control unit.
  - If the control unit downloads an internal **holdmusic.wav** file, the external audio port is ignored.
  - The IP Office can be forced to use the external port and not download an internal music on hold file by selecting **Use External Music on Hold** (**System | Telephony**) and restarting the IP Office.
- **Default Music on Hold Tones**
  On these systems; if no external source is connected, no internal music on hold file is available and **Use External Music on Hold** is not selected; then the system will use a default tone for music on hold. The tone used is double beep tone (425Hz repeated (0.2/0.2/0.2/3.4) seconds on/off cadence). On IP Office 3.0(50) and higher this option is only supported for the Italian locale. On IP Office 4.0 and higher this option is supported in all locales.

Checking Music on Hold

The IP Office has a default system short code that allows you to listen to a system's current music on hold.

1. At an idle extension, dial ***34**.
2. You will hear the system's music on hold.

System Date and Time

The IP Office control unit contains a battery backed clock which is used to maintain system time during normal operation and when mains power is removed. The time is normally obtained using Time protocol (RFC868) requests. Note that this is different from Network Time Protocol (NTP).

Following a reboot the IP Office control unit sends out a time request on its LAN interfaces. It first makes the request to the Voicemail Server IP address in its configuration and, if it receives no reply, it then makes a broadcast request.

The Voicemail Lite Server, Voicemail Pro Server/Service and the Manager program can all act as Time servers, giving the time as set on their host PC. If you are running Manager when the Voicemail Server starts, then Voicemail does not start as a time server. It is therefore recommended that you have no copy of Manager running when you start or restart the Voicemail Server.

A specific address for the time server that should be used can be set in the IP Office configuration (**System | System | Time Server IP Address**). Setting this address to 0.0.0.1 also disables the IP Office's time update requests.

When using a time server located in a different time zone from the IP Office, there are two mechanisms for applying an offset to the time. If Manager is acting as the time server, the time offset for each can be specified through the BOOTP entry for the system. Alternatively, the offset can specified in the IP Office configuration (**System | System | Time Offset**).

Manually Setting the System Time

For IP Office systems without access to a time server, a number of methods to manually set the system time exist. These are available in the IP Office 4.0 Q2 2007 maintenance release and higher.

The following methods are only supported on these phones: 2410, 2420, 4412, 4424, 4612, 4624, 4610, 4620, 4621, 5410, 5420, 5610, 5620, 5621, 6412, 6424. They are not supported on any analog, T3, T3 IP or wireless (IP DECT or 3600 Series) phones.

- Note: If the IP Office system that is still configured for automatic time updates (on by default), any manually set time and date will be overridden on the next automatic update. The use of automatic updates can be disabled by setting the **System | System | Time Server IP Address** to ***0.0.0.1***.

Using a Programmable Button

The following method uses a programmable button. It is support on IP Office 4.1+.

Enabling a User Button for Time and Date

1. Select the user who can adjust the system time and date.
    - If necessary create a new user who can be used to log on and set the time and date before logging off.
2. On the **User | Telephony** tab, ensure that the option **System Phone** is selected.
3. On the **User | Button Programming** tab select and double-click on an available button.
4. Select **Emulation > Self-Administer** as the action.
5. In the **Action Data** field enter either ***2*** or ***2A***. Use ***2A*** for a button that will request the users **Login Code** if they have one set.
6. Enter an appropriate label for the button.
7. Adding another button set to the **Emulation > Time of Day** action is recommended as this will allow the user to view and confirm the systems current time.
8. Send the updated configuration to the IP Office.

Using the Programmable Button

1. When the user presses the programmed button, the options **Date** and **Time** are displayed.
2. To set the date press **Date**.
    1. The current system date is displayed.
    2. Enter the new date, using two digits for day and month. Use the ***** or **#** keys to insert the **/** separators. The format for date enter matches the locale of the IP Office system.
    3. During entry the key labeled **<<<** can be used to backspace.
    4. When the full date has been entered as required, press **Next**.
    5. Press **Done**.
    6. The phone will return to idle.
3. To set the time press **Time**.
    1. The current system time is displayed.
    2. Enter the new time using 24-hour clock format. Use the ***** or **#** keys to insert the **:** separator.
    3. When the full time has been entered as required, press **Done**.
    4. The phone will return to idle.

Using the *9000* Maintenance Short Code

This method uses the default short code provided for toggling the modem enabled state of the first analog trunk in a system. Note that this method will switch off the maintenance mode if currently enabled. This method is supported on IP Office 4.0 Q2 2007 Maintenance release and higher.

Enabling use of the Maintenance Short Code

1. Select the user who can adjust the system time and date.
    - If necessary create a new user who can be used to log on and set the time and date before logging off.
2. On the **User | Telephony** tab, ensure that the option **System Phone** is selected.
3. Send the updated configuration to the IP Office.

Using the Maintenance Short Code

1. Dial the short code *9000* and wait for the confirmation beeps.
2. Press **Hold**.
    - If the user has a **Login Code** it will be requested. Enter the code.
3. The options **Date** and **Time** are displayed.
4. To set the date press **Date**.
    1. The current system date is displayed.
    2. Enter the new date, using two digits for day and month. Use the ***** or **#** keys to insert the **/** separators. The format for date enter matches the locale of the IP Office system.
    3. During entry the key labeled **<<<** can be used to backspace.
    4. When the full date has been entered as required, press **Next**.
    5. Press **Done**.
    6. The phone will return to idle.
2. To set the time press **Time**.
    1. The current system time is displayed.
    2. Enter the new time using 24-hour clock format. Use the ***** or **#** keys to insert the **:** separator.
    3. When the full time has been entered as required, press **Done**.
    4. The phone will return to idle.

The 'No User' User

It is possible to have an extension which has no default associated user. This can occur for a number of reasons:

- The extension has no Base Extension setting associating it with a user who has the same setting as their Extension to indicate that they are the extension's default associated user.
- The extension's default associated user has logged on at another extension. Typically they will be automatically logged back on at their normal extension when they log off the other phone.
- The extension's default associated user cannot be automatically logged on as they are set to Forced Login.

Phones with no current user logged in are associated with the setting of the **NoUser** user in the IP Office configuration. This user cannot be deleted and their Name and Extension setting cannot be edited. However their other settings can be edited to configure what functions are available at extensions with no currently associated user.

By default the NoUser user has **Outgoing Call Bar** enabled so that the extension cannot be used for external calls. For IP Office 4.0+ the users first programmable button is set to the **Login** action.

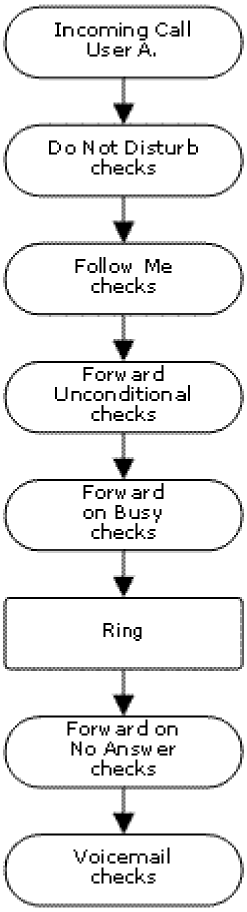
NoUser Source Numbers

The **SourceNumbers** tab of the ***NoUser*** user is used to configure a number of special options. These are then applied to all users on the IP Office system. For details refer to the **User | Source Numbers** section.

Forward and Transferring Calls

DND, Follow Me and Forwarding

This section contains topics looking at how users can have their calls automatically redirected. As illustrated, there is a order of priority in which the redirect methods are used.



1. **Do Not Disturb (DND)**
   Redirect all calls to voicemail if available, otherwise return busy tone. DND overrides all the redirect method below unless the calling number is in the user's DND Exception Numbers List.
2. **Follow Me**
   Redirect all calls to another extension that the users is temporarily sharing. Follow Me overrides Forward Unconditional. The Follow Me destination is busy or does not answer, the user's Forward on Busy or Forward on No Answer options can be used if set.
3. **Forward Unconditional**
   Redirect the user's external calls to another number. That number can be any number the user can normally dial including external numbers. Forwarding of hunt group and internal calls is optional. Forward Unconditional overrides Forward on Busy and Forward on No Answer.
4. **Forward on Busy**
   Redirects the user's external calls when the system sees the user as being busy. Uses the same number as Forward Unconditional unless a separate Forward on Busy Number is set. Forwarding internal calls is optional. Forward on Busy overrides Forward on No Answer.
5. **Forward on No Answer**
   Redirects the user's external calls when they ring for longer than the user's No Answer Time. Uses the same number as Forward Unconditional unless a separate Forward on Busy Number is set. Forwarding internal calls is optional.

Notes

- **Retrieving Externally Forwarded Calls**
  Where a call is forwarded to an external destination and receives busy or is not answered within the forwarding user's No Answer Time, the system will attempt to retrieve the call. If forwarded on a trunk that does not indicate its state, for example an analog loop start trunks, the call is assumed to have been answered.
- **Off-Switch Forwarding Restrictions**
  User forwarding is subject to the same restrictions as transferring calls. To bar a user from forwarding calls to an external number, the **Inhibit Off-Switch Transfers** option on the **User | Telephony** tab should be selected. To bar all users from forwarding calls to external numbers the **Inhibit Off-Switch Transfers** option on the **System | Telephony** tab should be selected. See Off-switch Transfer Restrictions.
- For IP Office 4.0 and higher, when transferring a call to another extension that has forwarding enabled, the type of call being transferred is used. For example, if transferring an external call, if the transfer target has forwarding of external calls enabled then the forward is used.

Do Not Disturb (DND)

Summary: Redirect all calls to busy tone or to voicemail if available except those in your DND exceptions list.

Do Not Disturb (DND) is intended for use when the user is present but for some reason does not want to be interrupted. Instead calls are sent to voicemail if available, otherwise they receive busy tone.

- **Exceptions**
  Specific numbers can be added to the user's **Do Not Disturb Exception List**. Calls from those numbers override DND. N and X wildcards can be used at the end of exception numbers to match a range of numbers. For external numbers, this uses the incoming caller line ID (ICLID) received with the call.
- **Priority**
  Enabling DND overrides any Follow Me or forwarding set for the user, except for calls in the user's **Do Not Disturb Exception List**.
- **Phone**
  When enabled, the phone can still be used to make calls. An **N** is displayed on DS phones. When a user has do not disturb in use, their normal extension will give alternate dialtone when off hook.
- **Applied to**

Call Types Blocked		
Internal	✓	Busy or voicemail.
External	✓	Busy or voicemail.
Hunt Group	✓	Not presented.
Page	✓	Not presented.
Follow Me	✗	Rings.
Forwarded	✓	Busy.
VM Ringback	✗	Rings
Automatic Callback	✗	Rings
Transfer Return	✗	Rings.
Hold Return	✗	Rings.
Park Return	✗	Rings.

Do Not Disturb Controls

<table>
<tr><th colspan="2">Do Not Disturb</th></tr>
<tr><td>Manager</td><td>A user's DND settings can be viewed and changed through the User | DND tab within the IP Office's configuration settings.</td></tr>
<tr><td>Controls</td><td>The following short code features/button programming actions can be used:
<table>
<tr><th>Feature/Action</th><th>Short Code</th><th>Default</th><th>Button</th></tr>
<tr><td>Do Not Disturb On</td><td>✓</td><td>*08</td><td>✓- Toggles.</td></tr>
<tr><td>Do Not Disturb Off</td><td>✓</td><td>*09</td><td>✓</td></tr>
<tr><td>Do Not Disturb Exception Add</td><td>✓</td><td>*10*N#</td><td>✓</td></tr>
<tr><td>Do Not Disturb Exception Delete</td><td>✓</td><td>*11*N#</td><td>✓</td></tr>
</table></td></tr>
<tr><td>Phone Manager</td><td>Users can enable DND and set exception numbers by clicking and select the Do Not Disturb tab.

When a user has DND enabled, it is indicated by DND in the title bar. It is indicated by a blue cross symbol on Speed Dial icons set to the user.</td></tr>
<tr><td>SoftConsole</td><td>A SoftConsole user can view and edit a user's DND settings except exception numbers. Through the directory, select the required user. Their current status including DND is shown. Double-click on the details to adjust DND on or off.</td></tr>
<tr><td>Voicemail</td><td>If voicemail is available, it is used instead of busy tone for callers not in the users exceptions list.

For Voicemail Pro, the Play Configuration Menu action can be used to let callers switch DND on or off.</td></tr>
</table>

Follow Me

Summary: Have your calls redirected to another user's extension, but use your coverage, forwarding and voicemail settings if the call receives busy tone or is not answered.

Follow Me is intended for use when a user is present to answer calls but for some reason is working at another extension such as temporarily sitting at a colleague's desk or in another office or meeting room. Typically you would use Follow Me if you don't have a Hot Desking login code or if you don't want to interrupt your colleague from also receiving their own calls.

- **Priority**
  Follow Me is overridden by DND except for callers in the user's DND Exception Numbers List. Follow Me overrides Forward Unconditional but can be followed by the user's Forward on Busy or Forward on No Answer based on the status of the Follow Me destination.
- **Destination**
  The destination must be an internal user extension number. It cannot be a hunt group extension number or an external number.
- **Duration**
  The Follow Me user's no answer timeout is used. If this expires, the call either follows their Forward on No Answer setting if applicable, or goes to voicemail is available. Otherwise the call continues to ring at the destination.
- **Phone**
  When enabled, the phone can still be used to make calls. When a user has follow me in use, their normal extension will give alternate dialtone when off hook.
- **Exceptions**
  - The Follow Me destination can make return calls which will not be redirected.
  - The call coverage settings of the user are applied to their Follow Me calls. The call coverage settings of the destination are not applied to Follow Me calls it receives.
- **Calls Forwarded**

Call Types Redirected		
Internal	✓	Redirected.
External	✓	Redirected.
Hunt Group	✓	Redirected*.
Page	✓	Redirected.
Follow Me	✗	Not redirected.
Forwarded	✓	Redirected.
VM Ringback	✗	Not redirected.
Automatic Callback	✗	Not redirected.
Transfer Return	✗	Not redirected.
Hold Return	✗	Not redirected.
Park Return	✗	Not redirected.

*Except calls for "Longest Waiting" type hunt groups.

Follow Me Controls

<table>
<tr><th colspan="2">Follow Me</th></tr>
<tr><td>Manager</td><td>A user's Follow Me settings can be viewed and changed through the User | Forwarding tab within the IP Office's configuration settings. Note that on this tab, entering a Follow Me Number also enables Follow Me.</td></tr>
<tr><td>Controls</td><td>The following short code features/button programming actions can be used:
<table>
<tr><th>Feature/Action</th><th>Short Code</th><th>Default</th><th>Button</th></tr>
<tr><td>Follow Me Here</td><td>✓</td><td>*12*N#</td><td>✓</td></tr>
<tr><td>Follow Me Here Cancel</td><td>✓</td><td>*13*N#</td><td>✓</td></tr>
<tr><td>Follow Me To</td><td>✓</td><td>*14*N#</td><td>✓</td></tr>
</table></td></tr>
<tr><td>Phone Manager</td><td>Users can set a Follow Me To Number and enable Follow Me through the Forwarding tab. Click and select the Forwarding tab.

When a user has Follow Me enabled, it is indicated by FollowTo and the destination in the title bar. It is not indicated on Speed Dial icons set to the user. Other status icons remain linked to the user's normal telephone and not to the status of the destination.</td></tr>
<tr><td>SoftConsole</td><td>A SoftConsole user can view and edit a user's Follow Me settings. Through the directory, select the required user. Their current status including Follow Me is shown. Double-click on the details and select Forwarding to alter their forwarding settings including Follow Me.</td></tr>
<tr><td>Voicemail</td><td>For calls initially targeted to the user but then redirected, when voicemail is invoked the mailbox of the user is used and not the mailbox of the destination.

For Voicemail Pro, the Play Configuration Menu action can be used to let callers alter or set their current Follow Me destination.</td></tr>
</table>

Forward Unconditional

Summary: Have your calls redirected immediately to another number including any external number that you can dial.

- **Priority**
  This function is overridden by DND and or Follow Me if applied. Forward Unconditional overrides Forward on Busy and Forward on No Answer.
- **Destination**
  The destination can be any number that the user can dial. If external and **Inhibit Off-Switch Transfers** is applied, the caller is directed to voicemail if available, otherwise they receive busy tone.
- **Duration**
  The destination is rung using the forwarding user's **No Answer Time**. If this expires, the call goes to voicemail if available. Otherwise the call continues to ring at the destination. Calls to an external destination sent on trunks that do not signal their state, for example analog loop start trunks, are assumed to have been answered.
- **Phone**
  When enabled, the phone can still be used to make calls. An **D** is displayed on DS phones. When a user has forward unconditional in use, their normal extension will give alternate dialtone when off hook.
- **Calls Forwarded**

Call Types Forwarded		
Internal	✓	Optional.
External	✓	Forwarded.
Hunt Group	✓	Optional.*
Page	✗	Not presented.
Follow Me	✗	Rings.
Forwarded	✓	Forwarded.
VM Ringback	✗	Rings.
Automatic Callback	✗	Rings.
Transfer Return	✗	Rings.
Hold Return	✗	Ring/hold cycle.
Park Return	✗	Rings.

*Optional only for calls targeting sequential and rotary type groups.

Forward Unconditional Controls

<table>
<tr><th colspan="2">Forward Unconditional</th></tr>
<tr><td>Manager</td><td>A user's forwarding settings can be viewed and changed through the User | Forwarding tab within the IP Office's configuration settings.</td></tr>
<tr><td>Controls</td><td>The following short code features/button programming actions can be used:
<table>
<tr><th>Feature/Action</th><th>Short Code</th><th>Default</th><th>Button</th></tr>
<tr><td>Forward Number</td><td>✓</td><td>*07*N#</td><td>✓</td></tr>
<tr><td>Forward Unconditional On</td><td>✓</td><td>*01</td><td>✓- Toggles.</td></tr>
<tr><td>Forward Unconditional Off</td><td>✓</td><td>*02</td><td>✓</td></tr>
<tr><td>Forward Hunt Group Calls On</td><td>✓</td><td>✗</td><td>✓- Toggles.</td></tr>
<tr><td>Forward Hunt Group Calls Off</td><td>✓</td><td>✗</td><td>✓</td></tr>
<tr><td>Disable Internal Forwards</td><td>✓</td><td>✗</td><td>✗</td></tr>
<tr><td>Enable Internal Forwards</td><td>✓</td><td>✗</td><td>✗</td></tr>
<tr><td>Disable Internal Forward Unconditional</td><td>✓</td><td>✗</td><td>✗</td></tr>
<tr><td>Enable Internal Forward Unconditional</td><td>✓</td><td>✗</td><td>✗</td></tr>
<tr><td>Set No Answer Time</td><td>✓</td><td>✗</td><td>✓</td></tr>
</table></td></tr>
<tr><td>Phone Manager</td><td>Users can set a forward destination number and enable Forward Unconditional through the Forwarding tab. Click and select the Forwarding tab.

When a user has Forward Unconditional enabled, it is indicated by Fwd unconditional and the destination in the title bar. It is indicated by a green arrow symbol on Speed Dial icons set to that user.</td></tr>
<tr><td>SoftConsole</td><td>A SoftConsole user can view and edit a user's forwarding settings. Through the directory, select the required user. Their current forwarding status is shown. Double-click on the details and select Forwarding to alter their forwarding settings.</td></tr>
<tr><td>Voicemail</td><td>For calls initially targeted to the user but then redirected, when voicemail is invoked the mailbox of the user is used and not the mailbox of the destination.

For Voicemail Pro, the Play Configuration Menu action can be used to let callers set their current forwarding destination and switch Forwarding Unconditional on/off.</td></tr>
</table>

Forward on Busy

Summary: Have your calls redirected when you are busy to another number including any external number that you can dial.

The method by which the system determines if a user is 'busy' to calls depends on factors such as whether they have multiple calls appearance buttons or **Call Waiting** and or **Busy on Held** set. See **Busy**.

- **Priority**
  This function is overridden by DND and or Forward Unconditional if applied. It can be applied after a Follow Me attempt. It overrides Forward on No Answer.
- **Destination**
  The destination can be any number that the user can dial. The Forward Unconditional destination number is used unless a separate number Forward on Busy Number is set. If **Inhibit Off-Switch Transfers** is applied, the caller is directed to voicemail if available, otherwise they receive busy tone.
- **Duration**
  The destination is rung using the forwarding user's No Answer Time. If this expires, the call goes to voicemail is available. Calls to an external destination sent on trunks that do not signal their state, for example analog loop start trunks, are assumed to have been answered.
- **Phone**
  Forward on Busy is not indicated and normal dial tone is used.
- Calls Forwarded

Call Types Forwarded		
Internal	✓	Optional.
External	✓	Forwarded.
Hunt Group	✗	Not presented
Page	✗	Not presented.
Follow Me	✗	Rings.
Forwarded	✓	Forwarded.
VM Ringback	✗	Rings.
Automatic Callback	✗	Rings.
Transfer Return	✗	Rings.
Hold Return	✗	Ring/hold cycle.
Park Return	✗	Rings.

Forward on Busy Controls

<table>
<tr><th colspan="2">Forward on Busy</th></tr>
<tr><td>Software Level</td><td>A user's forwarding settings can be viewed and changed through the User | Forwarding tab within the IP Office's configuration settings.</td></tr>
<tr><td>Controls</td><td>The following short code features/button programming actions can be used:
<table>
<tr><th>Feature/Action</th><th>Short Code</th><th>Default</th><th>Button</th></tr>
<tr><td>Forward Number</td><td>✓</td><td>*07*N#</td><td>✓</td></tr>
<tr><td>Forward on Busy Number</td><td>✓</td><td>*57*N#</td><td>✓</td></tr>
<tr><td>Forward on Busy On</td><td>✓</td><td>*03</td><td>✓- Toggles.</td></tr>
<tr><td>Forward on Busy Off</td><td>✓</td><td>*04</td><td>✓</td></tr>
<tr><td>Disable Internal Forwards</td><td>✓</td><td>✗</td><td>✗</td></tr>
<tr><td>Enable Internal Forwards</td><td>✓</td><td>✗</td><td>✗</td></tr>
<tr><td>Disable Internal Forward Busy or No Answer</td><td>✓</td><td>✗</td><td>✗</td></tr>
<tr><td>Enable Internal Forward Busy or No Answer</td><td>✓</td><td>✗</td><td>✗</td></tr>
<tr><td>Set No Answer Time</td><td>✓</td><td>✗</td><td>✓</td></tr>
</table></td></tr>
<tr><td>Phone Software Level</td><td>Users can set a forward destination number and enable Forward on Busy through the Forwarding tab. Click and select the Forwarding tab.

When a user has Forward on Busy enabled, it is indicated by Fwd on Busy and the destination in the title bar. It is not indicated by Speed Dial icons set to that user.</td></tr>
<tr><td>SoftConsole</td><td>A SoftConsole user can view and edit a user's forwarding settings. Through the directory, select the required user. Their current forwarding status is shown. Double-click on the details and select Forwarding to alter their forwarding settings.</td></tr>
<tr><td>Voicemail</td><td>For calls initially targeted to the user but then redirected, when voicemail is invoked the mailbox of the user is used and not the mailbox of the destination.

For Voicemail Pro, the Play Configuration Menu action can be used to let callers set the forward destination. It cannot however be used to enable Forward on Busy or set a separate Forward on Busy number.</td></tr>
</table>

Forward on No Answer

Summary: Have your calls redirected another number if it rings without being answered.

- **Priority**
  This function is overridden by DND and Forward on Busy if applied. It can be applied after a Follow Me attempt. Forward Unconditional overrides Forward on Busy and Forward on No Answer.
- **Destination**
  The destination can be any number that the user can dial. The Forward Unconditional destination number is used unless a separate number Forward on Busy Number is set. If **Inhibit Off-Switch Transfers** is applied, the caller is directed to voicemail if available, otherwise they receive busy tone.
- **Duration**
  The destination is rung using the forwarding user's No Answer Time. If this expires, the call goes to voicemail is available. Otherwise the call continues to ring at the destination. Calls to an external destination sent on trunks that do not signal their state, for example analog loop start trunks, are assumed to have been answered.
- **Phone**
  Forward on No Answer is not indicated and normal dial tone is used.
- **Calls Forwarded**

Call Types Forwarded		
Internal	✓	Optional.
External	✓	Forwarded.
Hunt Group	✗	Not presented.
Page	✗	Not presented.
Follow Me	✗	Rings..
Forwarded	✓	Forwarded.
VM Ringback	✗	Rings.
Automatic Callback	✗	Rings.
Transfer Return	✗	Rings.
Hold Return	✗	Ring/hold cycle.
Park Return	✗	Rings.

Forward on No Answer Controls

<table>
<tr><th colspan="2">Forward on No Answer</th></tr>
<tr><td>Manager</td><td>A user's forwarding settings can be viewed and changed through the User | Forwarding tab within the IP Office's configuration settings.</td></tr>
<tr><td>Controls</td><td>The following short code features/button programming actions can be used:
<table>
<tr><th>Feature/Action</th><th>Short Code</th><th>Default</th><th>Button</th></tr>
<tr><td>Forward Number</td><td>✓-</td><td>*07*N#</td><td>✓</td></tr>
<tr><td>Forward on Busy Number</td><td>✓-</td><td>*57*N#</td><td>✓</td></tr>
<tr><td>Forward on No Answer On</td><td>✓-</td><td>*05</td><td>✓- Toggles.</td></tr>
<tr><td>Forward on No Answer Off</td><td>✓-</td><td>*06</td><td>✓</td></tr>
<tr><td>Enable Internal Forwards</td><td>✓</td><td>✗</td><td>✗</td></tr>
<tr><td>Disable Internal Forwards</td><td>✓</td><td>✗</td><td>✗</td></tr>
<tr><td>Enable Internal Forward Busy or No Answer</td><td>✓</td><td>✗</td><td>✗</td></tr>
<tr><td>Disable Internal Forward Busy or No Answer</td><td>✓</td><td>✗</td><td>✗</td></tr>
<tr><td>Set No Answer Time</td><td>✓</td><td>✗</td><td>✓</td></tr>
</table></td></tr>
<tr><td>Phone Manager</td><td>Users can set a forward destination number and enable Forward on No Answer through the Forwarding tab. Click and select the Forwarding tab.

When a user has Forward on No Answer enabled, it is indicated by Fwd on No Answer and the destination in the title bar. It is not indicated by Speed Dial icons set to that user.</td></tr>
<tr><td>SoftConsole</td><td>A SoftConsole user can view and edit a user's forwarding settings. Through the directory, select the required user. Their current forwarding status is shown. Double-click on the details and select Forwarding to alter their forwarding settings.</td></tr>
<tr><td>Voicemail</td><td>For calls initially targeted to the user but then redirected, when voicemail is invoked the mailbox of the user is used and not the mailbox of the destination.

For Voicemail Pro, the Play Configuration Menu action can be used to let callers set the forward destination. It cannot however be used to enable Forward on Busy or set a separate Forward on Busy number.</td></tr>
</table>

Determining a User's Busy Status

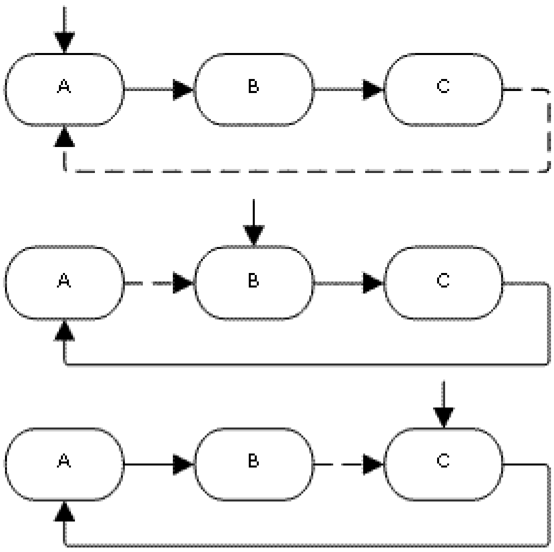
Various IP Office features allow users to handle more than one call at a time. Therefore the term "busy" has different meanings. To other users it means whether the user is indicated as being busy. To the system it means whether the user is not able to receive a call. The latter is used to trigger 'busy treatment', either using a user's Forward on Busy settings or redirecting calls to voicemail or just returning busy tone.

- **Busy Indication**
  The user busy indication provided to programmable buttons, and to the Phone Manager and SoftConsole applications, is based on the monitored user's hook switch status. Whenever the user is off-hook, they will be indicated as being busy regardless of call waiting or call appearance settings.
- **Busy to Further Calls**
  Whether a user is busy to receive further call is based on a number of factors as illustrated and described below.
  - **Logged In and Present**
    Is the user logged into an extension and is that extension physically connected to the IP Office system.
  - **Busy on Held**
    If a user enables their Busy on Held setting, whenever they have a call on hold, they are no longer available to any further incoming calls.
  - **Appearance Buttons**
    A user's call appearance button are used to receive incoming calls. Normally, whilst the user has any free call appearance buttons, they are available to receive further calls. Exceptions are:
    - **Reserve Last Appearance**
      Users with appearance buttons require a free call appearance button to initiate transfers or conferences. Therefore it is possible through the user's configuration settings to reserve their last call appearance button for outgoing calls only.
    - **Other Appearance Buttons**
      Calls may also be indicated on line, call coverage and bridged appearance buttons.
- **Call Waiting**
  Users of phones without appearance buttons can use call waiting. This adds an audio tone, based on the system locale, when an additional call is waiting to be answered. Only one waiting call is supported, any further calls receive busy treatment.
- **Hunt Group Calls**
  A user's availability to receive hunt group calls is subject to a range of other factors. See Member Availability.

Chaining and Loops

Chaining is the process where a call forward to an internal user destination is further forwarded by that user's own forwarding settings.

- **Follow Me Calls**
  Follow Me calls are not chained. They ignore the forwarding, Follow Me and Do Not Disturb settings of the Follow Me destination.
- **Voicemail**
  If the call goes to voicemail, the mailbox of the initial call destination before forwarding is used.
- **Looping**
  When a loop would be created by a forwarding chain, the last forward is not applied. For example the following are scenarios where A forwards to B, B forwards to C and C forwards to A. In each case the final forward is not used as the destination is already in the forwarding chain.



- **Maximum Number of Forwards**
  A maximum of 10 forwarding hops are supported for any call.

Transferring Calls

The following are some of the methods usable to transfer calls.

- **Supervised Transfer**
  This is a transfer where the user waits for the transfer destination to answer and potentially talks to that party before completing the transfer. However the initial consultation stage is presented as an internal call with internal call details and ringing.
- **Unsupervised Transfer**
  This is a transfer completed whilst the destination is still ringing.
- **Automatic Transfer - Forwarding**
  The IP Office allows users to automatically transfer calls using forwarding options such as Forward on Busy, Forward on No Answer and Forward Unconditional. For full details see **DND, Follow Me and Forwarding**.
- **Transfers to a Forwarded Extension**
  For IP Office 4.0 and higher, when transferring a call to another extension that has forwarding enabled, the type of call being transferred is used. For example, if transferring an external call, if the transfer target has forwarding of external calls enabled then the forward is used.

Manual Transfer Methods			
Tool	**Unsupervised Transfer**	**Supervised Transfer**	**Reclaim**
Analog Phone	1. Press **R**. Note that broken dial tone is heard while a call is on hold. 2. Dial the transfer destination number. 3. Hang-up.	1. Press **R**. 2. Dial the transfer destination number. 3. If the destination answers and accepts the call, hang-up. 4. If the called party does not answer or does not want to accept the call, press **R** again. 5. To return to the original caller press **R**.	***46**
Avaya DS Phone	1. Press **Transfer**. 2. Dial the transfer destination number. 3. Press **Transfer** again to complete the transfer.	1. Press **Transfer**. 2. Dial the transfer destination number. 3. If the destination answers and accepts the call, press **Transfer** again to complete the transfer. 4. If the called party does not answer or does not want to accept the call, press **Drop**. 5. To return to the original caller press it's call appearance button.	***46**
Phone Software Level	1. Click . 2. Enter the transfer destination in the **Number** box. 3. Select **Blind Transfer** button or click .	1. Click . 2. Enter the transfer destination in the **Number** box. 3. Select **Transfer**. The original call will be put on Hold. 4. Once the call has been answered you can talk with the transfer target. 5. To transfer the call, click . 6. To cancel the transfer and reconnect the held call press End.	**Function | Reclaim**

Off-Switch Transfer Restrictions

Users cannot transfer calls to a destination that they cannot normally dial. This applies to manual transfers and also to automatic transfers (forwarding). In addition to call barring applied through short codes, the following IP Office settings may restrict a users ability to transfer calls.

- **Outgoing Call Bar:** *Default = Off (User | Telephony)*
  When enabled, this setting stops a user from making any external calls. It therefore stops them making any external transfers or forwards.
- **Inhibit Off-Switch Forward/Transfer:** *Default = Off (System | Telephony)*
  When enabled, this setting stops any user from transferring or forwarding calls externally.
- **Inhibit Off-Switch Forward/Transfer:** *Default = Off (User | Telephony)*
  When enabled, this setting stops a specific user from transferring or forwarding calls externally.
  - When either system or user **Inhibit Off-Switch Forward/Transfer** is enabled, it affects the operation of the user's Phone Manager application and phone. In Phone Manager, the destination fields on the **Forwarding** tab are changed to drop-down lists containing only internal destinations. User attempts to set an external forward destination via a short code will receive error tone. User attempt to set an external forward destination via a programmable button on their phone will not have a Next option allowing the number to be saved.
- **Analog Trunk to Trunk Connection:** *Default = Off (Line | Analog)*
  When not enabled, users cannot transfer or forward calls on one analog trunk back off-switch using another analog trunk.
- **Allow Outgoing Transfer:** *Default = Off (System | Telephony)*
  When not enabled, users are only able to transfer or forward back off-switch incoming external calls. For IP Office 4.0 and higher this option is not present, instead outgoing transfers are allowed by default.

Conferencing

Calls can be transferred using a conference from which the conference originator then drops. The user will normally be allowed to setup conference calls that can include two external parties. However, if the user then drops from the conference, if any transfer restriction applies then the whole conference will be dropped.

Centrex Transfer

Centrex Transfer is a feature provided by some line providers on external analog lines. It allows the recipient of a calls on such a line to transferred that call to another external number. The transfer is then performed by the line provider and the line is freed. Without Centrex Transfer, transferring an external call to another external number would occupy both a incoming and outgoing line for the duration of the call.

IP Office 4.0 and higher supports Centrex Transfer using the existing Flash Hook features.

The following are the supported controls and usages for Centrex Transfer on IP Office:

- **Centrex Transfer Button Operation**
  The action **Flash Hook** (**Advanced | Miscellaneous | Flash Hook**) can be assigned to programmable buttons on DS and IP phones. This button can be configured with or without a telephone number for an automatic or manual transfer respectively.
  - **Manual Transfer**
    If the programmable button is setup as a Flash Hook button without a target telephone number, pressing the button returns dial tone to the user. They can then dial the required transfer number and when they hear ringing or an answer, hang up to complete the Centrex Transfer.
  - **Automatic Transfer**
    If the programmable button is setup as a Flash Hook button with a target telephone number, pressing the button performs the Centrex Transfer to the numbers as a single action.
- **Centrex Transfer Short Code Operation**
  The short code feature **Flash Hook** can be used with IP Office short codes. It can be setup with or without a telephone number in the same way as a Flash Hook programmable button detailed above. The line group must be the group of analog lines from the Centrex service line provider.
  - **Centrex Transfer Operation for Analog Extensions**
    Most analog phones have a button that performs the action of sending a hook flash signal. The marking of the button will vary and for example may be any of **R**, **H**, **Recall** or **Hold**. For IP Office analog extensions, pressing this button sends a hook flash to the IP Office to hold any current call and return dial tone.
    - To perform a Centrex Transfer, pressing the analog extension's hook flash button should be followed by the dialing of a **Flash Hook** short code.
    - For analog extension users with **Call Waiting** enabled, pressing the hook flash button during a call will hold the current call and connect any call waiting. Therefore it is recommend that analog extension users wanting to use Centrex Transfer should not also have **Call Waiting** enabled.
  - **Transfer from Voicemail/Auto-Attendant**
    This operation is only supported through Voicemail Pro using an Assisted Transfer action with the destination set to a Flash Hook short code.

Additional Notes

- **Addition Prefix Dialing**
  In some cases the Centrex service provider may require a prefix for the transfer number. If that is the case, that prefix must be inserted in the button programming or the short code used for the Centrex Transfer.
- **Phone Manager and SoftConsole**
  Centrex Transfer is not supported for calls being held and transferred through IP Office applications such as Phone Manager and SoftConsole.
- **Conference Calls**
  Centrex Transfer is not supported with conference calls.

Conferencing

Conferencing Overview

IP Office systems support the following conference capabilities:

<table>
<tr><th>Control Unit</th><th>DSP</th><th>Conference Capability</th></tr>
<tr><td>Small Office Edition</td><td>✓</td><td>Supports up to 24 conference parties with a maximum of 6 parties in any particular conference.</td></tr>
<tr><td>IP403</td><td>✗</td><td rowspan="4">Support multiple conferences totaling up to 63 parties. For example:
• 21 x 3-way conferences.
• 1 x 10-way conference (10 parties) plus 11 x 3-way conferences (33 parties) and free capacity for 20 more conference parties to join new or existing conferences.</td></tr>
<tr><td>IP406 V1</td><td>✗</td></tr>
<tr><td>IP406 V2</td><td>✓</td></tr>
<tr><td>IP500</td><td>✓</td></tr>
<tr><td>IP412</td><td>✗</td><td>The IP412 supports two 63 party conference banks. When a new conference is started, the bank with the most free capacity is used for that conference. However once a conference is started on one conference bank, that conference cannot use any free capacity from the other conference bank.</td></tr>
</table>

Notes:

- **DSP Conferencing**
  Systems that use DSP for conferencing automatically apply silence suppression to quite parties in conferences involving 10 or more parties.
- **Other Use of Conference Resources**
  System features such as call intrusion, call recording and silent monitoring all use conference resources for their operation.
- **Ending Conferences**
  The behaviour for the IP Office ending a conference varies as follows:
  - For pre-4.0 IP Office systems, if a conference has two parties, and one party leaves, the conference call is ended. This may affect conferences that are just beginning but currently only contain the first two parties to join.
  - For IP Office 4.0, a conference remains active until the last extension or trunk with reliable disconnect leaves. Connections to voicemail or a trunk without reliable disconnect (for example an analog loop-start trunk) will not hold a conference open.
- **Analog Trunk Restriction**
  In conferences that include external calls, only a maximum of two analog trunk calls are supported. This limit is not enforced by the IP Office software.
- **Recording Conferences**
  If call recording is supported, conference calls can be recorded just like normal calls. Note however that recording is automatically stopped when a new party joins the conference and must be restarted manually. This is to stop parties being added to a conference after any "advice of recording" message has been played.
- **IP Office Conferencing Center**
  If IP Office Conferencing Center is installed, 5 resources are reserved for use by the system. The maximum number of callers in any one conference and the total number of people on conference calls is reduced by 5. The maximum number of conferences on the system for IP406 V2, IP412 and IP Office 500 is reduced by 2.
- **IP Trunks and Extensions**
  Conferencing is performed by services on the IP Office's non-IP interface. Therefore a voice compression channel is required for each IP trunk or extension involved in the conference.
- **Call Routing**
  A short code routing calls into a conference can be used as an Incoming Call Route destination.

Default Conference Handling

The methods below use the IP Office's default system short codes.

To start/add to a conference:

1. Place your first call or the existing conference on hold. Existing conference parties will still be able to talk to each other.
2. Call the new party.
    - If not answered, or diverted to voicemail, or answered but the party does not want to join the conference; put them on hold and dial ***52** to clear the call.
3. If answered and the other party wants to join the conference, put them on hold and dial ***47**.
4. All held calls are now in conference.
    - Digital display extensions will see **CONF** followed by the conference number.

To exit a conference:

1. Parties wanting to leave a conference can simply hang-up.

Using Conference Meet Me

Each conference on the IP Office is assigned a conference number. This number is displayed on suitable display phones.

Conference Meet Me allows users to join or start a specific numbered conference. This method of operation allows you to advertise a conference number and then let the individual parties join the conference themselves.

Through the **Button Programming** tab (also called **Digital Telephony**) within IP Office Manager, the **Conference Meet Me** function can be assigned to a DSS key (select **Advanced | Call | Conference Meet Me**). This allows simple one key access by internal users to specific conferences.

- For IP Office 4.1+ the following enhancements apply to use buttons programmed to Conference Meet Me:
    - When the conference is active, any buttons associated with the conference ID indicate that active state.
    - Calls can be transferred to the conference by pressing **Transfer** and then pressing the **Conference Meet Me** button.

Note:

- Conference Meet Me can create conferences that include only one or two parties. These are still conferences using slots from the IP Office's conference capacity.
- Conference Meet Me is not supported on IP Office IP500 systems running in IP Office Standard Edition mode.

Example 1: Meet Me to a user specified conference

The following example system short code allows any extension to dial *67* and then the number of the conference which they want to join followed by #. For example dialing *67*600# will put the user into conference 600.

- **Short Code:** *67*N#
- **Telephone Number:** N
- **Feature:** Conference Meet Me

Example 2: Meet Me to a system specified conference number

The following example system short code allows the dialing extension to join a specific conference, in this case 500.

- **Short Code:** *500
- **Telephone Number:** 500
- **Feature:** Conference Meet Me

If you are asked to add a party to a conference, having a conference meet me short code is very useful. With the conference in progress, call the new party. When they answer, hold the call, dial the conference meet me short code and then hang-up.

Hot Desk Operation

Hot Desking

Hot desking allows users to log in at another phone. Their incoming calls are rerouted to that phone and their user settings are applied to that phone. There are a number of setting and features which affect logging in and out of IP Office phones.

- In order to hot desk, a user must be assigned a **Login Code** (**User | Telephony | Login Code**) in the IP Office configuration.
- By default, each IP Office extension has an **Base Extension** setting containing a directory number. This associates the extension with the user who has the matching Extension directory number settings as being that extension's default associated user.
    - By leaving the **Base Extension** setting for an extension blank, it is possible to have an extension with no default associated user. All extensions in this state use the settings of a special user named ***NoUser*** . On suitable phones the display will show **NOT LOGGED ON**.
    - You can create users whose **Extension** directory number is not associated with any physical extension. These users must have a login code in order to log in at a phone when they need to make or receive calls. In this way the IP Office system can support more users than it has physical extensions.
- When another user logs in at an extension, they take control of that phone. Any existing user, including the default associated user, is logged off that phone.
    - Any user settings not applicable to the type of phone on which the user has logged on become inaccessible. For example some programmable button features will become inaccessible if the phone at which a user logs on does not have a sufficient number of programmable buttons.
    - Note that settings that are stored by the phone rather than by the IP Office remain with the phone and do not move when a user hot desks.
- When a user logs off or is logged off by someone else logging on, they are automatically logged back in at the extension for which they are the default associated user if no one else is logged on at that extension. However this does not happen for users set to **Forced Login**.
- For each user, you can configure how long the extension at which they are logged in can remain idle before they are automatically logged out. This is done using the **Login Idle Period** option. This option should only be used in conjunction with **Force Login**.
- Logged in users who are members of a hunt group can be automatically logged off if they do not answer hunt group calls presented to them. This is done by selecting ***Logged Off*** as the user's **Status on No Answer** setting.
- Calls to a logged off user are treated as if the user is busy until the user logs on.
- Logging in and out at a phone can be done either using IP Office short codes or programmable buttons.
    - The default system short code for logging in, is ***35*_N#_** where the user replaces ***N*** with their extension number and then login code separated by a *. This uses the short code feature **ExtnLogin**. If the user dials just a login code as ***N***, it is checked against the user with the same extension number as the extension's base extension number.
    - The default system short code for logging out is ***36**. This uses the short code feature **ExtLogout**. For IP Office 4.0 and higher this feature cannot be used by a user who does not have a login code or by the default associated user of an extension unless they are set to forced login.
    - The **ExtnLogin** and **ExtnLogout** features can be assigned to programmable buttons on suitable Avaya phones. The **ExtnLogin** button will then prompt the user to enter their details.

Remote Hot Desking

IP Office 4.0+ supports hot desking between IP Office systems within a Small Community Network.

The system on which the user is configured is termed their 'home' IP Office, all other systems are 'remote' IP Offices. To log on at a remote IP Office requires that IP Office to have a **Advanced Small Community Networking** license. A license is not necessary on the user's home IP Office.

- **User Settings**
  When a user logs on to a remote IP Office system;
  - The users incoming calls are rerouted across the SCN.
  - The users outgoing calls uses the settings of the remote IP Office.
  - The users own settings are transferred. However some settings may become unusable or may operate differently.
    - Users settings are transferred but user rights are not.
      - If user rights with the same name exist on the remote system then they will be used. The same applies for user rights applied by time profiles, if a time profile with the same name also exists on the remote system .
    - Appearance buttons configured for users on the home system will no longer operate.
    - Various other settings may either no longer work or may work differently depending on the configuration of the remote system at which the user has logged on. For example T3 phones the personal directory is not transferred with the user.
- **Break Out Dialing**
  In some scenarios a hot desking user logged on at a remote system will want to dial a number using the system short codes of another system. This can be done using either short codes with the Break Out feature or a programmable button set to Break Out. This feature can be used by any user within the Advanced Small Community Network but is of significant use to remote hot deskers.

Call Center Agents

On IP Office systems with a call center application such as Compact Contact Center (CCC) or Compact Business Center (CBC), logging in and logging off is a key part of tracking and reporting on call center agents. It also controls call distribution as, until the agent logs in, their hunt group membership is seen as disabled.

For IP Office, CCC, CBC and Delta Server, an agent is defined as being a user with a **Login Code** and set to **Forced Login**. Those users consume a CCC agent license.

Hot Desking Examples

The following are example of different ways that the hot desking settings can be used.

Scenario 1: Occasional Hot Desking

In this scenario, a particular user, for this example extension 204, needs to occasionally work at other locations within the building.

1. A **Login Code** is added to the user's configuration settings, for this example ***1234***.
2. The user can now log in when needed at any other phone by dialing ****35*204*1234#***. The phone's default associated user is logged off by this and their calls get busy treatment. User 204 is also logged off their normal phone and their calls now rerouted to the phone at which they have logged on.
3. When finished, the user can dial ****36*** to log off.
4. This logs the phone's normal default user back on. Its also logs the hot desking user back on at their normal extension.

Scenario 2: Regular Hot Desking

This scenario is very similar to the one above. However the user doesn't want to be automatically logged back in on their normal phone until they return to its location.

1. A **Login Code** is added to the user's configuration settings, for this example 1234.
2. The **Forced Login** option is selected.
3. When the user logs out of the phone that they are currently using, they are no longer automatically logged in on their normal extension. When they return to it they must dial ****35*204*1234#*** to log in.
4. Whilst not logged in anywhere, calls to the user receive busy treatment.

Scenario 3: Full Hot Desking

Similar to the scenarios above but this time the user doesn't have a regular phone extension that they use. In order to make and receive calls they must find a phone at which they can log in.

1. The user is given an **Extension** directory number that is not matched by the extension directory number setting of any existing extension.
2. They are also given a **Login Code** and a **Login Idle Period** is set, for this example 3600 seconds (an hour). Forced Login isn't required as the user has no default extension at which they might be automatically logged in by the IP Office system.
3. The user can now log in at any available phone when needed.
4. If at the end of the business day they forget to log off, the **Login Idle Period** will eventually log them off automatically.

Scenario 4: Call Center Hot Desking

In this scenario, the phone extensions have no default extension number. Several phones set like this might be used in a call center where the agents use whichever desk is available at the start of their shift. Alternatively a set of desks with such phones might be provided for staff that are normally on the road but occasionally return to the office and need a temporary desk area to complete paper work.

1. For the extensions, the Extension setting is left blank. This means that those phones will be associated with the **NoUser** user's settings and display **NOT LOGGED ON**.
2. The call center agents or road-warrior users are configured with Extension directory numbers that also don't match any existing physical extensions. They are all given **Login Code** numbers.
3. The users can log in at any of the extensions when required. When they log off or log in elsewhere, the extensions return to the **NoUser** setting.

Automatic Log Off

Normally a user can either log themselves off or be logged off by another user logging on. The follow methods can be used by the IP Office system to automatically log off a user. These methods only apply to users with a Login Code and set to Forced Login.

Idle Timeout

The **User | Telephony** setting **Login Idle Period** can be used to automatically log out the user after a set period of phone inactivity. The period can be set between 1 to 99999 seconds and is based on call inactivity other than ringing calls.

Unanswered Calls

Users who are members of hunt groups are presented with hunt group calls when they are logged in and not already on a call. If the user is logged in but not actually present they will continue to be presented with hunt group calls. In this scenario it can be useful to log the user off.

1. **For the hunt group**
   On the **Hunt Group | Hunt Group** tab, use the **Agent's Status on No Answer Applies** to setting to select which types of unanswered hunt group calls should change the user's status. The options are:
   - **None.**
   - **Any Calls.**
   - **External Inbound Calls Only.**
2. **For the user**
   On the **User | Telephony** tab, the Status on No Answer setting is used. This sets what the user's status should be changed to if they do not answer a hunt group call. The options are:
   - **Logged On**
     If this option is selected, the user's status is not changed.
   - **Busy Wrap-Up**
     If this option is selected the user's membership status of the hunt group triggering the action is changed to disabled. The user can still make and receive calls and will still continue to receive calls from other hunt groups to which they belong.
   - **Busy Not Available**
     If this option is selected the user's status is changed to do not disturb. This is the equivalent of DND and will affect all calls to the user.
   - **Logged Off**
     If this option is selected the users status is changed to logged off. In that state the cannot make calls and cannot receive calls. Hunt group calls go to the next available agent and personal calls treat the user as being busy.

Data Routing

Overview of Data Routing

The IP Office system is a network router. In this role it can connect users on its LAN to remote services by using WAN links and telephone trunk connections. It can also allow users to dial-in and then act as if they were using a PC on the LAN.

As well as being a network router, the IP Office is a telephone system. These dual roles allow it to support a range of functions that involve traffic between the network and telephony interfaces. These functions use internal data channels. The number of internal data channels that can be connected from the system's LAN interface to its telephony interface at any time is restricted.

- An internal data channel is a connection between the system's telephony and LAN interfaces. For example a Voicemail connection, an internet connection or a RAS user.
- Calls using a VCM channel (for example VoIP calls and Avaya 4600 and 5600 Series phones) do not use a data channel.
- The number of data channels in use does not necessarily match the number of users:
    - Several LAN network users, browsing the internet using the same service to an ISP would be a single data channel.
    - Several dial-in network users would each have a separate data channel.
- The maximum number of data channels that can be simultaneously in use for voicemail is restricted. These channels also require entry of an appropriate license.

The restriction depends on the type of Control Unit being used.

IP Office Control Unit	Internal Data Channels	Maximum Data Channels for Voicemail
Small Office Edition	18	10
IP403	18	10
IP406 V1	24	20
IP406 V2	40	20
IP412	100	30

Network Address Translation (NAT)

NAT allows the addresses used within your LAN to be replaced by a different address when connecting to an external service.

Typically a service provider will allocate you a single IP address to be used when connecting to their service. NAT allows all your user's traffic to appears to be coming from that single address without having to change any of your user's real addresses. This is useful as internally most networks use addresses that have been reserved for public use within networks but are not valid for routing across the internet (since the same addresses may be being used on other networks). Also as stated it allows multiple users to use the same service simultaneously.

The use of NAT is automatically enabled if the IP Office Service being used includes an IP address that is not in the same domain as the its LAN1 IP address.

An exception to the above applies for the Small Office Edition and IP412 control units. These units have two LAN's, LAN1 and LAN2. For these units, on each LAN, **Enable NAT** can be selected and then applied to traffic between the two LAN's.

Dynamic Host Configuration Protocol (DHCP)

The IP Office can act as a simple DHCP server. When switched on with a defaulted configuration, the Control Unit request IP address information from a DHCP server. If it gets no response it assumes the role of DHCP server for the LAN.

In DHCP Server mode, by default the Control Unit issues itself the address 192.168.42.1. It allocates 200 addresses for DHCP clients, 192.168.42.1 to 19.168.42.200. This leaves 192.168.42.201 to 192.168.42.254 available for any computers that need to be allocated a fixed or static IP address. 192.168.42.255 is not used as this is a broadcast address for the LAN.

Examples

Simple ISDN Internet Connection

In this example, we want all non-local data traffic to be routed to the Internet. The Internet Service Provider (ISP) has provided the account details required. Using the IP Office's Network Address Translation (NAT), a single account can be used for all users.

1. Select **Service** and add a normal service. Change the following settings and click **OK**.
    - **Name:** Internet
    - **Account Name:** As provided by the ISP.
    - **Password:** As provided by the ISP.
    - **Telephone Number:** As provided by the ISP.
    - Check **Request DNS**.
3. Select **IP Route** and add a new route. Change the following settings and click **OK**.
    - Leave the **IP Address** and **IP Mask** blank. This will then match any data traffic that isn't matched by any other IP Route entry.
    - Select the service created above as the **Destination**.

Alternate

In the example above, a default IP Route was created which then routed all traffic to the required Service. An alternate method to do this with IP Office is to select Default Route within the Service settings.

ISDN Link Between IP Offices

To create a data link between two sites via ISDN configure the Control Unit as per the following example:

At Site A on IP address 192.168.43.1

1. **Create a Normal Service:**
   The Service name can be any text and is used to identify this particular Service. The Account Name and password are presented to the remote end, therefore must match the User name and password configured at Site B. The Telephone Number is the number of the remote end.
2. **Create an IP Route:**
   In the IP Address field enter the network address of the remote end, not the IP address of the Control Unit. Under Destination select the Service created above.
3. **Create a User:**
   Under the **Dial In** tab tick **Dial In On**. This User account is used to authenticate the connection from the Site B. Note that as the Service and User have the same names, these two configuration forms are automatically linked and become an Intranet Service. The User password is displayed at the bottom of the Service tab as the Incoming Password.
4. **Setup RAS:**
   Check the default RAS settings "DialIn" are available, otherwise create a new one. If the RAS settings are given the same name as the Service and User they are automatically linked and become a WAN Service. Ensure that the Encrypted Password option is not checked when using a WAN Service.
5. **Setup an Incoming Call Route:**
   Check the default Incoming Call Route is available, otherwise create a new one. If the Incoming Number is left blank, the Incoming Call Route accepts data calls on any number. Under **Destination** select the RAS service created above. The Bearer Capability should be AnyData.

At Site B on IP address 192.168.45.1

1. Repeat the above process but altering the details to create an route from Site B to Site A.

Using a Dedicated T1/PRI ISP Link

This section shows an example of a dedicated WAN PPP link to an Internet Service Provider (ISP) over a set of T1 or T1 PRI line channels. The ISP must support this mode of connection and will need to provide details of the required settings. If multiple channels are to be used, then the ISP must support Multilink PPP.

1. **Create a New WAN Service**
   A service is used to define connection settings such as name, password, bandwidth, etc.
   1. Select **Service** to display the existing services.
   2. Click on and select **WAN Service**.
   3. Select the **Service** tab.
      1. In the **Name** field enter an appropriate name, such as "**Internet**". Note that the IP Office will also automatically create User entry and a RAS entry with the same name.
      2. Enter the **Account Name**, **Password** and **Telephone Number** details provided by the ISP.
      3. For the **Firewall Profile** select the firewall created previously.
   4. Click the **Bandwidth** tab.
      1. Set the **Maximum No. of Channels** to the maximum number of channels that the service should use. In this example, 12 channels were used.
      2. Leave all the other entries at their default values.
   5. If the ISP has allocated IP address details these are entered through the **IP** tab. If the **IP Address** and **IP Mask** define a different domain from the IP Office LAN, then NAT is automatically applied.
      1. Click the **IP** tab.
      2. In the **IP Address** field enter the IP address specified by the ISP.
      3. In the **IP Mask** field enter the IP Mask specified by the ISP.
   6. The settings shown are typical. The actual settings must match those required by the ISP. For example, if Cisco routers are being used then IPHC needs to be ticked.
      1. Click the **PPP** tab.
      2. Ensure that the following options are selected. Leave all other options at their default settings.
         - **Multilink.**
         - **Compression Mode: Disable.**
         - **Callback Mode: Disable.**
         - **Access Mode: Digital64**
      3. Click **OK**.

2. **Create the Virtual WAN Port**
   In this stage, a WAN port is defined that actually uses T1 or T1 ISDN trunk channels.

    1. Select **WAN Port** to display existing ports.
    2. Click on and select **WAN Port**.
    3. In the **Name** field, enter either ***LINEx.y*** where:
        - ***LINE*** must be in uppercase.
        - ***x*** is the line number. For a trunk card in Slot A, this will be ***1***. For a trunk card in Slot B, this will be ***5***.
        - ***y*** is the lowest numbered channel number to be used by the WAN link minus 1. For example, if the lowest channel to be used is channel 1 then y = 1 - 1 = 0.
    2. In the **Speed** field, enter the total combined speed of the maximum number of channels sets in the Service. In this example, 12 channels x 64000 bits = 76800.
    3. Set the **Mode** to **SyncPPP**.
    4. In the **RAS Name** field, select the name used for the Service.
    5. Click **OK**.

3. **Create an IP Route**
   By creating an IP route with blank IP address details, it becomes the default route for outgoing IP traffic.

    1. Select **IP Route** to display existing routes.
    2. Click on and select **IP Route**.
    3. Leave the **IP Address** and **IP Mask** fields blank.
    4. In the **Destination** field, select the WAN service.
    5. Leave the **Metric** at default value of **1**.
    6. Click **OK**.

4. **Configure the Line Channels**
   This stage of the process differs according to the type of trunk being used.

    1. **T1 Trunk**
       Use the following for a T1 trunk.

        1. Click **Line** to display the existing lines.
        2. Double-click on the line previously entered in the WAN Port settings.
        3. Check that the **Channel Allocation** order matches that required by the ISP. Cisco routers typically use ***1->24***.
        4. Select the channels to be used in the WAN PPP link and change their **Channel Type** to "***Clear Channel 64k***".
        5. Click **OK**.
        6. Click **OK** again.
        7. Send the configuration to the IP Office and reboot.

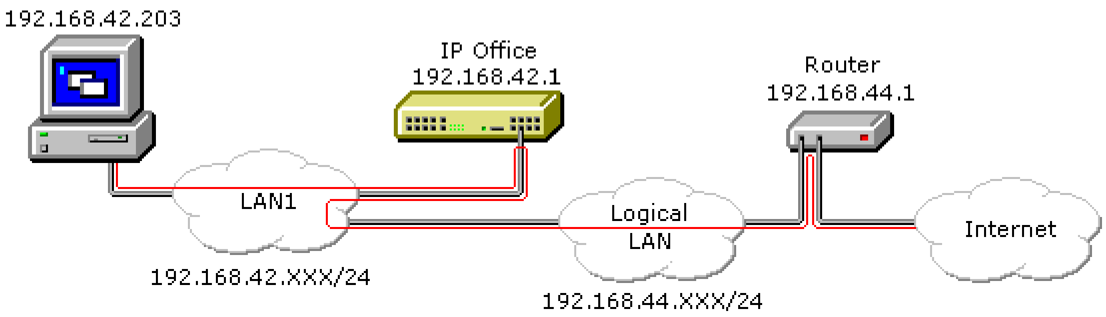
2. **T1 PRI Trunk**
   Use the following for a T1 PRI trunk.

   1. In the left-hand panel, click on **Line** to display the list of existing lines.
   2. Double-click on the line previously entered in the WAN Port settings.
   3. Check that the **Channel Allocation** order matches that required by the ISP. Cisco routers typically use ***1->23***.
   4. Select the channels to be used in the WAN PPP link and change their **Admin** to "***Out of Service***".
   5. Click **OK**.
   6. Click **OK** again.
   7. Send the configuration to the IP Office and reboot.

Logical LAN Connection

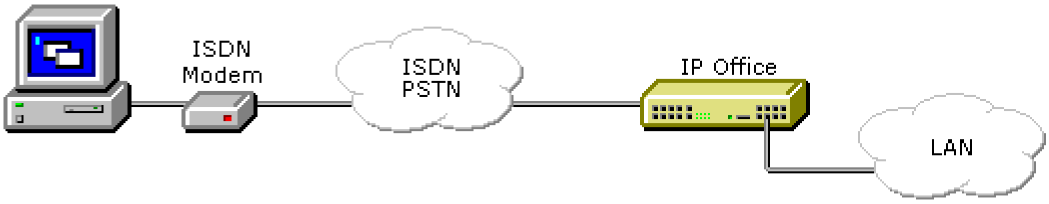
IP Office Small Office Edition and IP412 control units support two separate LAN interfaces (LAN1 and LAN2). These are separately addressed and the IP Office's IP route table and firewalls can be used to control traffic between device attached to the two LAN's.

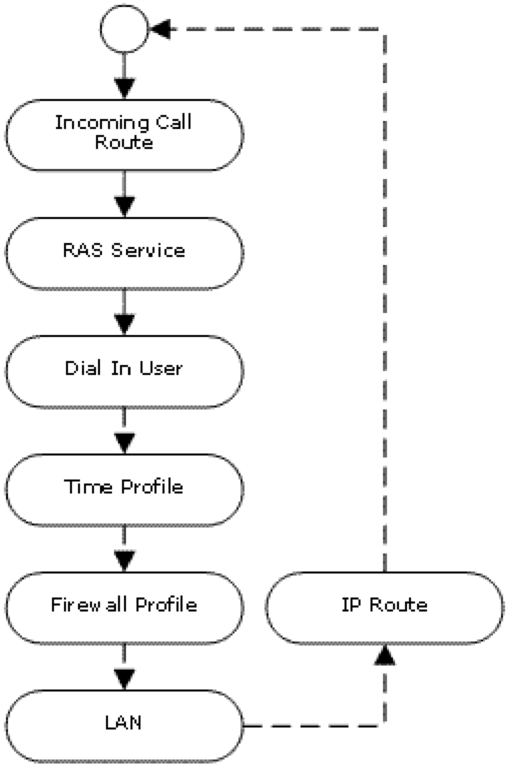
On other IP Office control units only a single LAN (LAN1) is available. A logical LAN allows these systems to support a second separately addressed LAN on the same interface. Traffic between the IP Office LAN1 and the logical LAN can then be controlled by the IP Office's IP route table and firewalls.



Direct Remote Access

The IP Office support remote access for incoming data calls on trunks.

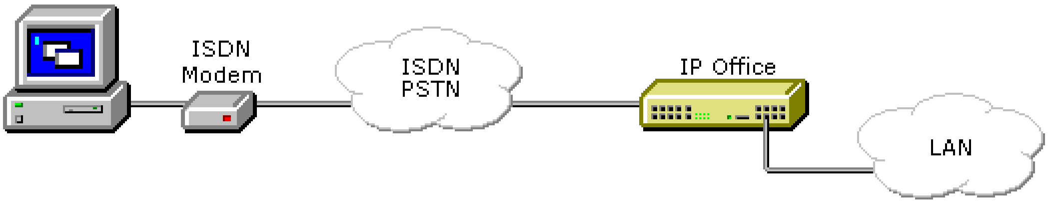




To do remote access, an incoming call is passed through the following elements of the IP Office configuration.

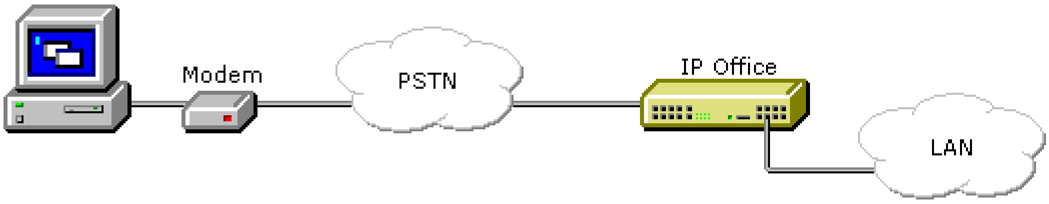
- **Incoming Call Route**
  A Incoming Call Route is used to match incoming remote access calls and pass them to a RAS service as the destination.
- **RAS Service**
  The RAS service defines settings relating to the data traffic methods useable with the call.
- **User**
  The user defines the name and password required for the RAS service. The user must have **Dial In On** enabled.
  - An **R** setting on the user's **Source Number** tab can be used to define the ICLID from which RAS calls are accepted.
- **Time Profile**
  The user settings can specify a time profile. The time profile then controls when remote access is allowed.
- **Firewall Profile**
  The user settings can specify a firewall profile. The firewall profile then control what traffic is allowed through the remote access connection.
- **System | LAN**
  The IP Office can provide DHCP support for remote access connections when it is set to ***Server*** or ***Dial in*** modes. Alternatively the remote access client can use a static IP address on the IP Office's subnet.
- **IP Route**
  If the remote access client uses a IP address that is from a different subnet from the IP Office, then a IP route entry is required for returning data. The RAS service is set as the destination.

ISDN Remote Access Example



1. **Create a User**
   The required details are:
   - In the **User** tab:
     Enter a **Name** and **Password**. IP Office is case sensitive. Remember to take care with passwords as this is a remote access link into your network.
   - In the **Dial In** tab:
     Ensure that **Dial In On** is ticked. The Firewall Profile and Time Profile are optional.
2. **Create a RAS Entry**
   - In the **RAS** tab:
     Enter the same name as the user that you created earlier. Again, remember this is case sensitive.
3. **Create an Incoming Call Route**
   - Set the **Bearer Capability** to ***Any Data***.
   - In the **Destination** drop-down list, select the RAS entry created above.
   - The values that you enter for any of the other fields will depend on whether the remote user will be calling in on a particular line, number or from a set ICLID.
4. **Is a Return IP Route Needed ?**
   Go to Step 6.
5. **Create a IP Route** *(Optional)*
   If the remote user has an IP address that is not in the same domain as the IP Office, then an IP Route is needed for return data. This is not necessary if the remote user's dial-up connection method is set to 'Obtain an IP Address Automatically' and the IP Office's DHCP mode is set to Server or DialIn.
   - Enter the IP Address and IP Mask of the remote system.
   - In the **Destination** drop-down list select the RAS entry created above.

Analog Remote Access Example



Configuration for a connection from an analog modem call is very similar to the ISDN example. However the IP Office must be able to answer modem calls. This can be done in the following ways;

- **Modem Cards**
  For all IP Office control units except the Small Office Edition, a modem card can be installed. This module allows the IP Office system to answer V.90 analog modem calls. The Internal Modem Card allows the IP Office system to support 12 simultaneous modem calls (4 only on the IP403). The Modem 2 card allows the IP Office system to support 2 simultaneous modem calls.
- **Analog Trunk Modem Mode**
  On systems with an ATM4 trunk card and on the Small Office edition, the first analog trunk can be set to answer V.32 modem calls. This is done by checking the **Modem Enabled** option on the analog line settings or using the default short code ***9000*** to toggle this service on or off.
- When using an analog modem, the **Bearer Capability** of the incoming call route used should be ***Any Voice***.

Example: Creating a VoIP Link via the WAN Port Using PPP

A VoIP link across a leased line requires the Control Unit at both ends to have a Voice Compression Module installed. These provide for a fixed number of channels to use VoIP at any time. They are used to compress voice down to either 6k3 (G723) or 8k (G729) and provide echo cancellation.

Both ends must using the same version of software and configured to use the same speed and compression.

At Site A on IP address 192.168.42.1.

1. **Create a Normal Service:**
   The Account Name and password is presented to the remote end, therefore must match the User name and password configured at Site B. The Encrypted Password option can only be used if the remote end also supports CHAP.
2. **Create a User:**
   Under the Dial In tab tick Dial In On. This User account is used to authenticate the connection from the Site B. As the Service and User have the same name these two configuration forms are automatically linked and become an Intranet Service. The User password is displayed at the bottom of the Service tab as the Incoming Password.
   - **Name:** SiteB
   - **Dial In | Dial In On:** Enabled.
3. **Create a RAS service:**
   If CHAP is to be used on this link, then the **Encrypted Password** option must be checked in the Service and in the RAS service. The name of the RAS service must match the name of the Service at Site B. If the RAS service is given the same name as the Service and User, they are automatically linked and become a WAN Service. Ensure that the Encrypted Password option is **not** checked when using a WAN Service.
4. **Edit the WANPort:**
   Note - do not create a new WANPort, this is automatically detected. If a WANPort is not displayed, connect the WAN cable, reboot the Control Unit and receive the configuration. The WANPort configuration form should now be added.
   - **RAS Name:** SiteA
5. **Create an IP Route:**
   The IP Address is the network address of the remote end. Under Destination select the Service created above.
6. **Create a new Line:**
   The Line Number and Line Group ID must be unique, in other words, not used by any other line. The Gateway IP Address is the IP Address of the Control Unit at the remote end. The Compression Mode used is dependent on the Voice Compression Card the Control Unit is running and the speed of the link.
7. **Create a Short Code:**
   To route all calls where the number dialed starts with 8 via Line Group ID 1, therefore via the VPN Line created above.
   - **Short Code:** 8N
   - **Telephone Number:** N
   - **Line Group ID:** 1
   - **Feature:** Dial

At Site B on IP address 192.168.45.1

1. Repeat the above steps for VoIP traffic from Site B to Site A.

- Note: For the IP Office Small Office Edition Control Unit, enabling Local Tones under the Line and Extension VoIP tabs is recommended.

Example: Creating a VoIP Link via the WAN Port Using Frame Relay

To create a VoIP link via the WAN port using frame relay, the first step is to attach a WAN cable and reboot the Control Unit. After this, receive a copy of the configuration.

Both ends must using the same version of software and configured to use the same speed and compression.

__At Site A__

1. Create a WAN Service:
    - On the Service Tab:
      The Name is "FR_link". The Account Name should be "FR_Link" and all password fields (both Password and Incoming Password) should be left blank.
    - On the **PPP** Tab:
      Check the **MultiLink/QoS** box.
      Set the **Header Compression Mode** to ***IPHC***.
    - On the Dial In Tab:
      If you are using a WAN 3 module, you must add "WAN" as the Dial In Service number.
2. On the **Wan Port** Form:
    - In the WanPort Tab
      Set the speed to match the link. Set the **RAS Name** to ***DialIn***. Set the **Mode** as ***SyncFrameRelay***.
    - In the FrameRelay Tab
      Set the appropriate Frame Relay Management Type. The other default settings are appropriate for a basic Frame Relay Connection.
    - In the DCLI tab
      Set the RAS Name to "FR_link".
      Frame Link Type = PPP
      DLCI set to the network setting
3. **Create a RAS service:**
   Encrypted Password option is **not** checked when using a WAN Service. Have the Name = "FR_Link"
4. **Create an IP Route:**
   The IP Address is the network address of the remote end. Under Destination select the "FR_link" that was created above.
5. **Create a new Line:**
   The Line Number and Line Group ID must be unique, in other words, not used by any other line. The Gateway IP Address is the IP Address of the Control Unit at the remote end.
6. **Create a Short Code:**
   To route all calls where the number dialed starts with 8 via Line Group ID 1, therefore via the VPN Line created above.
    - **Short Code:** 8N
    - **Telephone Number:** N
    - **Line Group ID:** 1
    - **Feature:** Dial

__At Site B__

1. Repeat the above steps for VoIP traffic from Site B to Site A.
    - Note: For the IP Office Small Office Edition Control Unit, enabling Local Tones under the Line and Extension VoIP tabs is recommended.

Voice over IP

Overview of VoIP

Depending on the model of control unit the IP Office control unit supports up to 128 voice compression channels. Either 3 or 16 of these channels are pre-installed in Small Office Edition controls units. On other units they are added by installing Voice Compression Modules (VCM's). The type and number of VCM modules supported by each control unit type varies.

The voice compression channel improves call quality and can be used to compress voice down to either 6k3 (G723) or 8k (G729/Netcoder) and provides echo cancellation (required for high latency circuits).

The bandwidth required for a VoIP call is made up of two parts, one of which is due to the actual digitization of the analog voice the other is required by the protocol which is used to wrap the digitized voice up and transport it to the remote site. VoIP calls require an overhead of 40 bytes per packet (RTP/UDP/IP Header overhead). This overhead is increased on a LAN by a further 12 bytes Ethernet or by 7 bytes over a PPP WAN link.

When transporting voice over low speed links (WANs) it is possible that normal data packets (typically 1500 byte IP packets) can prevent or delay the voice data from getting across the link. This can cause a very unacceptable speech quality. Thus it is vital that the routers in the network that carry voice have some form of Quality of service mechanism (QoS).

The Control Unit supports the DiffServ (RFC 2474) Quality of Service mechanisms (QoS) which is based upon a Type of Service (ToS) field in the IP header. The software will prioritize voice, fragment large packets and provide VoIP header compression to minimize the WAN overhead.

Typically the VoIP WAN overhead is 47 bytes on 20 byte payload this is 235% overhead. On the WAN protocol this is reduced to 11 bytes (8 bytes data, 2 bytes CRC and 1 byte HDLC flag) on the same 20-byte packet this is only 55%, and 180% saving. This overhead must be included when calculating the actual link speeds required to support voice traffic. For example an 8Kbps compression voice path actually required 12.4Kbps of WAN bandwidth when using QoS or 26.8Kbps if using standard non QoS routers.

QoS routers are also required to ensure low speech latency and to maintain sufficient audible quality. At present our header compression is based upon the latest standards (RFC 2507/2508/2509). For efficiency we operate below PPP (non-standard) - reducing the overheads further and allow data fragmentation to be performed more effectively (keeping latency low). It is therefore required to place our equipment at both ends to operate at full efficiency.

VoIP Protocols

The H.323 Stack within the core software supports the following protocols:-

- H.323 (V2)(1998), Packet-based multimedia communications systems
- Q.931, ISDN user-network interface layer 3 specification for basic call control
- H.225.0 (1998), Call signaling protocols and media stream packetization for packet-based multimedia communication systems
- RTP/RTCP
- H.245 (1998), Control protocol for multimedia communication
- Audio CODECs:
  - G.711 A-law/U-law
  - G.723.1 MP-MLQ
  - G.729 Annex A - CS-ACELP (Not supported by NetMeeting)
- Silence Suppression
- Fax Relay
- Local End Echo Cancellation 25ms (except transparent - no cancellation)
- Out of band DTMF
- Internet Standards/Specification (in addition to TCP/UDP/IP)
  - RFC 1889 - RTP/RTCP
  - RFC 2507,2508,2509 - Header Compression
  - RFC 2474 - DiffServ

Performance

The following table is the maximum ratings tested in the lab. For deployment we recommend that more bandwidth be made available for normal data.

	56K	64K	128K	256K	2M	LAN
G729.1 (8k)	4	5*	6	18	20!	20
NetCoder (8k)	4	5*	9*	18!	20	20
G.723 (6.4k)	5	5*	9	18	20	20
ADPCM (32k)	1	1	6	5	20!	20
G.711 (64k)	X	X	1	3	16!	20
Transparent (64k)	X	X	1	3	14!	20

- * - data transfer is affected at higher channel connectivity
- ! - channel connectivity at higher levels is affected by data transfer

Implementation

A Control Unit plays the part of a Gateway between H.323 terminals and phones connected to the Control Unit (and also external lines). H.323 is configured on a Control Unit as a VPN line, specifying the IP address of a remote gateway and the audio compression to be used.

IP phones can be configured as extensions. An example is NetMeeting, which is configured to use the Control Unit as a Gatekeeper, with an account name that should match the name of a user configured on the Control Unit.

IP extensions are automatically created when an IP phone registers with the Gatekeeper (depending on a configuration option). If the user is not found a new user and extension are created, allowing the phone to be used immediately.

Basic call setup (without a Gatekeeper)	• Call setup using H.225.0 encapsulated in Q.931 messages • Capability exchange using H.245 • Establishment of audio communication using H.245 OpenLogicalChannel • Audio using RTP/RTCP
Fast connect procedure	• Call setup using H.225.0 in Q.931 messages, with H.245 OpenLogicalChannel messages embedded in the H.225.0 messages • Audio using RTP/RTCP
Overlap sending	• Support for overlap sending, where a SetupAck is sent in response to the Setup message
Gatekeeper	• Gatekeeper support allows IP extensions to be automatically configured when they register with the gatekeeper.
Jitter buffer	• 5 frames of jitter buffer
Quality of Service	• Layer3 - DiffServ TOS Field set to DSCP 6 on generated packets. WAN links optimize for this traffic when set to "PPPSyncVoice". At present normal LAN and normal ISDN traffic is not prioritized. • Layer4 - UDP Port Marking - all RTP/UDP traffic is sent within a configurable UDP port range. The default range is C000-CFFF (hex) (49152-53247)

Voice Packet Payload Sizing/Latency (Default)

Codec	Payload	Latency
Transparent 64K G711	80bytes	10ms
ADPCM 32K	40bytes	10ms
ADPCM 16K	20bytes	10ms
G.711 ALAW	160bytes	20ms
G.729A	20bytes	20ms
G.723 (6K3)	24bytes	30ms
Netcoder 8K	20bytes	20ms
G.726-32K	80bytes	20ms
G.726-16K	40bytes	20ms

Small Community Networking

Small Community Networking

IP Office systems linked by IP trunks can enable voice networking to form a Small Community Networking (SCN). Within an SCN the separate IP Office systems 'learn' each other's extension numbers and user names. This allows extension calls between systems and support for a range of internal call features, see Supported SCN Features.

IP Office Small Community Networking is supported for a maximum of 500 extensions across up to 16 IP Office systems.

To set up a small community network, the following are required:

- A working IP trunk between the IP Office systems that has been tested for correct voice and data traffic routing.
    - On IP Office 500 systems, IP trunks require the entry of **IP500 Voice Networking** licenses. For IP Office 4.0, IP Trunks are not supported on systems running in Standard Edition mode. For IP Office 4.1+ IP trunks are supported in Standard Edition mode
- VCM modules are required in all systems.
- The extension and group numbering on each system must be unique.
- The extension and group names on each system must be unique.
- We also recommend that all names and numbers (groups, line, services, etc) on the separate IP Office systems are kept unique. This will reduce potential maintenance confusion.
- All systems should use the same set of **Telephony** timers, especially the **Default No Answer Time**.

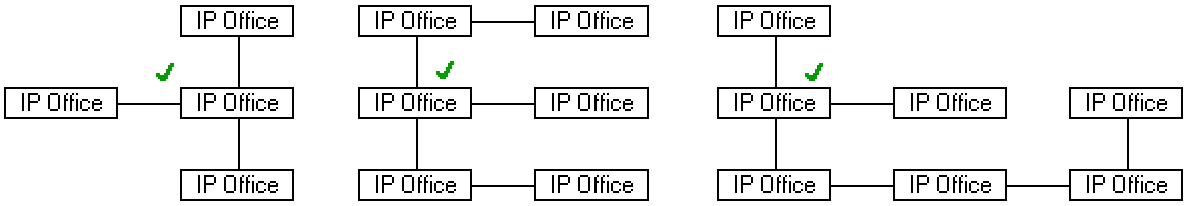
Software Level Interoperation
SCN is supported between IP Office systems with the same major software level or one level of difference in major software level. For example between 3.2 and 3.1 (same major level) and between 4.0 and 3.2 (one level of difference).

This option is intended mainly to allow the phased upgrading of sites within a Small Community Network. It is still recommended that all systems within a network are upgraded to the same level where possible. Within a SCN using differing levels of software, network features will be based on the lowest level of software within the network.

Supported SCN Network Layouts

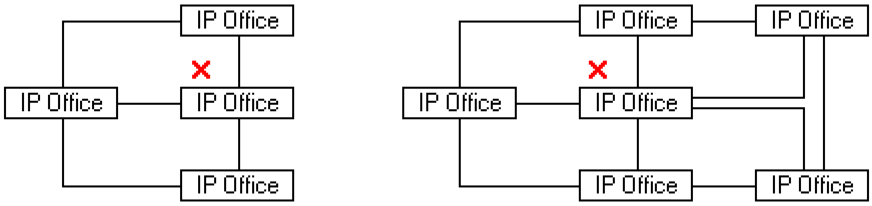
The IP Office systems within a Small Community network should only be connected in a star and or serial layout. For each IP Office there should only be one possible route to any other IP Office even if that route is via intermediate IP Offices.

The following are examples of supported star and serial layouts. Note that the maximum number of hops for user information transfers is 5.



The use of 'mesh' layouts connections is not supported. A mesh layout is one where there is more than one possible route between any two IP Offices.

The following are examples of mesh layouts.



SCN Signalling

IP Office SCN uses a signalling similar to RIP is order to update each other of there presence. This traffic can be seen in the IP Office Monitor application as AVRIP packets. This traffic is broadcast and sent to port 50795 on which each IP Office system listens. Each IP Office in the SCN broadcasts an regular update every 30 seconds. Additionally BLF updates are broadcast when applicable up to a maximum of every 5 seconds. Typically the volume is less than 1Kbps per IP Office system.

Supported SCN Features

Standard SCN Features

The following features are supported across all Small Community Networks:

- **User to user dialing**
- **Absence Text/Call Tagging**
- **Anti-tromboning**
- **Hold**
- **Transfer**
- **Forwarding**
- **Paging**
- **Directed Call Pick-Up**
- **Callback When Free**
- **Conference**
- **Ring Back**
- **Directory**
  Inclusion of SCN users and groups in the **DIR** directory of each system. Directory entries are not shared.
- **User DSS/BLF**
- **Voicemail Support - Centralized Voicemail using Voicemail Pro**
  Within a Small Community Network, a single Voicemail Pro can be used to provide voicemail services for all the IP Office systems. For full details of installation and setup refer to the Voicemail Pro documentation. The Voicemail Pro is licensed and hosted by a chosen central IP Office system and provides full operation for that system. The voicemail features supported for the other remote IP Offices are:
  - **User mailboxes.**
  - **Call recording.**
    Recording of incoming call routes is only supported for destinations on the same IP Office system, not for remote SCN destinations.
  - **Dial by Name.**
  - **Auto Attendants.**
  - **Breakout**
    Requires that the numbers used are routable by the system hosting the voicemail server.
  - **Announcements**
    Using IP Office 4.0 announcements. Pre-4.0 announcements are only supported for queues on the system hosting the voicemail server.

Advanced SCN Networking Features

IP Office 4.0+ allows a number of additional feature to be enabled by addition of a **Advanced Small Community Networking** license. The presence of this license allows the IP Office system to host hunt groups containing members from other systems ("Distributed Hunt Groups") and to host users from other IP Office 4.0+ systems in the SCN logging on ("Remote Hot Desking").

- **Advertised Hunt Groups**
  A hunt group can be set to be 'advertised'. This requires an ***Advanced Small Community Networking*** license. Hunt groups that are advertised can be dialed by users on other systems within the Small Community Network (SCN) without the need for short codes.
- **Distributed Hunt Groups**
  Hunt groups on a system can include users located on remote IP Office systems within the SCN network. A license is required on each system. A distributed hunt group can only be edited on the system on which they were created.
- **Remote Hot Desking**
  Users can hot desk between IP Office systems within the network. The system on which the user configured is termed their 'home' IP Office, all other systems are 'remote' IP Offices. To log on at a remote IP Office requires the remote IP Office to have a license. A license is not necessary on the user's home IP Office.
    - When a user logs on to a remote IP Office system;
        - The users incoming calls are rerouted across the SCN.
        - The users outgoing calls uses the settings of the remote IP Office.
        - The users own settings are transferred. However some settings may become unusable or may operate differently.
            - Users settings are transferred but user rights are not.
                - If user rights with the same name exist on the remote system then they will be used. The same applies for user rights applied by time profiles, if a time profile with the same name also exists on the remote system .
            - Appearance buttons configured for users on the home system will no longer operate.
            - Various other settings may either no longer work or may work differently depending on the configuration of the remote system at which the user has logged on. For example T3 phones the personal directory is not transferred with the user.
- **Breakout Dialing**
  This feature allows the user to select an IP Office system in the network from a displayed list and then dial a subsequent number as if dialing locally on the select system. This feature is triggered either by a programmable button or short code.

Note that both Distributed Hunt Groups and Remote Hot Desking are not supported for use with CBC and CCC.

Enabling Small Community Networking

Setup the VoIP Line from System A to System B

1. On System A, receive the system configuration.
2. Click the **Line** configuration form to display a list of existing lines.
3. Right-click on the displayed list and select **New**.
4. In the **Line** tab for the VoIP line set the following:
   - Set a unique **Line Number**.
   - In the **Telephone Number** field, enter a description of the link.
   - Set the **Outgoing Group ID** to a unique value.
5. In the **VoIP** tab for the VoIP line, set the following:
   - Ensure that **Voice Networking** is ticked. This enables the exchange of directory and user information between the IP Office systems and is the key enabler of Small Community Networking.
   - For the **Gateway IP Address** enter the IP address of System B.
   - Select the preferred **Compression Mode**. The same mode must be used by all VoIP lines and extensions within the network.
   - Check that the **H450 Support** option is set to ***H450***. This enables various Supplementary Signaling Services across the VoIP connection. QSIG can be used if H450 is not supported across the VoIP connection. However QSIG supports fewer supplementary signaling features.
   - Do not change any other options from their default settings without testing. For Small Community Networking **Enable Fast Start** and **Local Tones** should not be used.
6. Load the configuration and reboot System A. Note: Configuration changes and additions to VoIP line settings cannot be merged.

Setup the VoIP Line from System B to System A

7. On the remote system, repeat the previous steps to create a VoIP VPN link to System A.
   - Ensure that the **Compression Mode** selected in the **VoIP** tab of the VoIP line is the same at both the central and remote system.
   - Load the configuration and reboot the remote IP Office.

Test Small Community Networking

8. Test by making calls between extensions on the different systems.

Short Code Programming for Small Community Networks

With Small Community Networking enabled, the IP Offices 'learn' each others extension numbers and route extension calls appropriately.

However the same does not apply to dialing group and other numbers meant for the remote IP Office. To allow these to be routed correctly across the VoIP VPN links, short codes can be used.

Scenario
We want a short code on System A which will correctly route any 3000 range number to System B. This will allow System B group numbers to be dialed from System A.

To achieve the above scenario, we will add a new system short code. By using a system short code it becomes available to all users.

1. Receive the configuration from System A.
2. Click the **Short code** configuration form to display a list of existing system short codes.
3. Right-click on the displayed list and select New.
4. Enter the short code settings as follows:
    - **Short Code: *3XXX***
      This will match any four-digit number beginning with 3.
    - **Telephone Number: .**
      The . indicates that the short code should output the digits as dialed.
    - **Line Group ID: *3000***
      This should match the **Outgoing Group ID** given to the VoIP VPN line connected to System B.
    - **Feature: *Dial***
5. Click **OK**.
6. If the only change made to the configuration was this short code, load the new configuration using merge, otherwise load the configuration and reboot.
7. A similar system short code can be added to System B's configuration to route 2XXX dialing to System A.

Appendix: Configuration Examples

Transactional Pad

Connecting a Transactional Pad

A transaction pad (T-PAD, credit card "swipe" terminal) can use the ISDN (B channel) trunks, via the 25-pin D-type connector on the rear of the system control unit. This allows for faster transactions then provided by conventional modem connectivity.

The control unit supports a single DTE port. This DTE port has an AT command interface. Certain AT commands may be sent to the serial port so that it runs an X.25 T-PAD interface.

The ISDN link between the Control Unit and the transaction pad is digital. The transaction pad does not require a modem.

Configuration Parameters

In order to connect to a remote server, the DTE port needs:

- the Phone number of the remote server
- local_nua
- nui
- lower_channel (defaults to 1024)
- upper_channel (defaults to 1279)

AT commands need to be issued to the DTE port to enable the interface. The following AT commands are relevant to transaction pad operation:

- **ATB6** - set connection mode as TPAD
- **AT%A** - set the local nua.
- **AT%I** - set the nui.
- **AT%L** - set the lower channel limit (defaults to 1024).
- **AT%U** - set the upper channel limit (defaults to 1279).
- **AT&A** - set an autodial number.
- **AT&D=1** - dial autodial number whenever DTR is raised (by pad).

Example:

```
ATB6
AT%A=1234
AT%I=test_host
AT%L=1048
AT%U=1052
AT&A 01923111111
AT&D=1
```

Additional information of IP Office DTE port AT commands can be found in the "IP Office AT Commands Manual"..

Configuration Auto-Load

Though AT commands can be issue to the DTE port through a serial communications program such as Hyperterminal, DTE port settings and parameters are not saved in the IP Office Control Unit's flash memory. Thus they are lost during any reboot.

In order to 'permanently' set the parameters they need to be added to the configuration through the IP Office Manager application. This is done through a configured user called **DTEDefault**.

Create a user called **DTEDefault** and add the required initial AT commands into the **SourceNumbers** table. These commands are then automatically reloaded following any reboot.

Tracing

There are a number of locations where the transaction can be traced using the IP Office Monitor application.

- **Options/DTE**
  - **DTE Command Tx/Rx**
    This trace information is output when the DTE port is in AT mode
  - **DTE Filter Tx/Rx**
    This is trace information of the Serial communication between the DTE port and the pad. Other trace information will appear from time to time.

Paging

Paging from IP Office

Paging to and from IP Office phone's is covered by the appropriate telephone user guides. This section covers paging to 3rd-party paging equipment (centrally amplified paging systems or self-amplified speakers).

Typically, 3rd-party paging equipment uses analog connections. The IP Office can provide analog connections via either analog trunks or analog extensions. In terms of flexibility of operation once installed, the use of an analog extension port for paging is the preferred solution.

WARNINGS:

- The Paging Equipment must provide isolation to the IP Office analog port or an additional interface device must be fitted.
- The Paging Equipment (and separate interface device if used) must conform to the local and national telecommunications device regulation:
    - USA: FCC approval.
    - European Union: CE marked indicating compliance with the EMC (EN41003) and Low Voltage (EN60950) directives.
    - All other countries: use equipment that complies with locals and national telecommunication device regulation.
- Failure to observe the notes above could result in damage to the IP Office or the 3rd-party equipment.

Universal Paging Access Module

For the US, the Universal Paging Access Module (UPAM) is recommended as the interface device between the IP Office and the Paging Equipment.

The UPAM:

- Supports analog extension or trunk (loop or ground start) connection.
- Requires a 24V or 48V power supply if used with trunk connections.
- Provides a pre-announce tone heard at the paging extension and over the paging speakers. On/Off selectable.
- Provides a confirmation tone heard by the pager only (not supported for ground start trunks). On/Off selectable.
- Has Paging Time control which sets the maximum page time (6 to 35 seconds) if its other disconnect controls are disabled.
- Supports background music input via an RCA-type jack.

Universal Paging Access Module

For the US, the Universal Paging Access Module (UPAM) is recommended as the interface device between the IP Office and the Paging Equipment.

The UPAM:

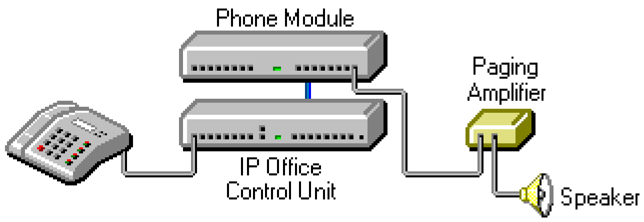
- Supports analog extension or trunk (loop or ground start) connection.
- Requires a 24V or 48V power supply if used with trunk connections. This may be optional for station connections.
- Provides a pre-announce tone heard at the paging extension and over the paging speakers. On/Off selectable.
- Provides a confirmation tone heard by the pager only (not supported for ground start trunks). On/Off selectable.
- Has Paging Time control which sets the maximum page time (6 to 35 seconds) if its other disconnect controls are disabled.
- Supports background music input via an RCA-type jack.

Paging via an Analog Extension Port (POT Port)

IP Office analog extension ports are marked as POT. These can be used for the connection of third-party paging equipment.

The IP Office 401 (not available in North America) and IP Office 403 Control Units have integral POT ports.

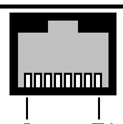
POT ports can also be installed by the addition of an IP Office Phone Module to the system.



- The Paging Amplifier must provide isolation or an additional isolation device should be fitted.
- The Paging Amplifier (and separate isolation device if used) must conform to the local and national telecommunications device regulation.
- If not done automatically, it may be necessary to set the **Paging Amplifier** to give priority to the VOX input.

Do the following to set up a page via an analog extension port:

1. **IP Office POT Port Wiring Connection**
    - Connections to POT ports should use a twin-pair cable wired as follows:

PHONE RJ45 Socket	Pin Number	Description
 Pin 8 Pin 1	1 to 3	Do not use.
	4	A: Ring
	5	B: Tip
	6 to 8	Do not use.

The POT ports are rated as follows:

- Off-Hook Current = 25mA.
- Ring Voltage = 40V rms.
- REN = 2

2. **Configure the Analog Extension**
    1. Start IP Office Manager and receive the configuration from the IP Office.
    2. Click the **Extension** icon to display the list of extensions.
    3. Double-click on the extension that will be used for the paging equipment connection.
    4. In the **Extn** tab, set the following:
        - Set the **Equipment Classification** to ***Paging Speaker***. In this mode the extension connects the speech path immediately without any ringing.
        - Set **Caller Display Type** to ***Off***.
    5. Click on **OK**.

3. **Configure the Analog Extension User**
    1. Click the **User** icon to display the list of users.
    2. Double-click the user currently associated with the extension above.
    3. In the **User** tab, set the following set the **Name** to ***Paging*** or similar to indicate the function.
    4. In the **Voicemail** tab untick **Voicemail On**.
    5. Click **OK**.

4. **Create a Short Code for Paging the Extension**

    This stage is optional. Since the connection is via a extension with an associated user, page calls can be made using the appropriate user name or number (see "Making Page Calls"). If you skip short code creation, send the new configuration to the IP Office and reboot.

    Do the following to create a short code:

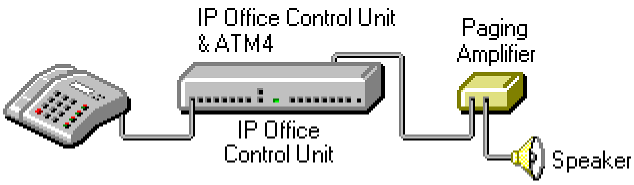
    1. Click on the **ShortCode** icon to display the list of short codes.
    2. Right-click on the list and select **New**.
    3. Enter the settings for the short code that users should dial when to make a paging call:
        - **Short Code:** ****78***
          The numbers users should dial to do a page. *78 is just an example.
        - **Telephone Number:** ***201***
          The analog extension connected to the paging equipment.
        - **Feature:** ***DialPaging***
          Note that ***DialPaging*** is used for an analog extension connection. ***Dial*** is used for an analog trunk connection. .

4. Click on **OK**.
5. Send the new configuration to the IP Office and reboot.

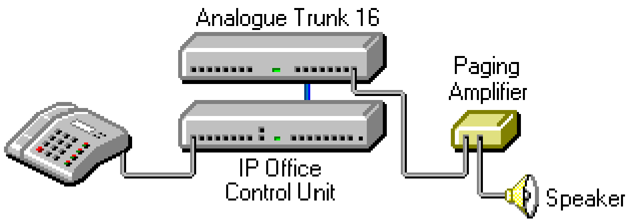
Paging via an Analog Trunk Port

You can use the analog trunk ports provided by the ATM4 or ATM16 modules.

The ATM4 is an internal module installed into the IP Office Control Unit. Note that the ATM4 only provides Loop Start analog trunk ports.



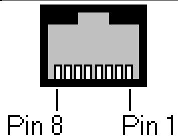
The ATM16 is an external expansion module. It supports both Loop Start and Ground Start analog trunks.



- The Paging Amplifier must provide isolation or an additional isolation device should be fitted.
- The Paging Amplifier (and separate isolation device if used) must conform to the local and national telecommunications device regulation.
- The paging connection must provide power in order to be seen as a real trunk by the IP Office.

Do the following to set up a page via an analog trunk port:

1. **Connection**
   The analog trunk ports on IP Office modules are RJ45 sockets. Connections to these should use a single-pair cable wired cable as follows:

RJ45 Socket	Pin Number	Description
	1 to 3	Do not use.
	4	A: Ring
	5	B: Tip
	6 to 8	Do not use.

2. **Configure the line via the Line configuration form in Manager:**
   1. Receive the configuration from the IP Office.
   2. Click on the **Line** icon to display the list of installed lines.
   3. In the **Line** tab for the analog line, set the following:
      - In the **Telephone Number** field enter a note indicating that this is the line to the paging equipment.
      - Set the **Outgoing Group ID** to a unique value, that is one not used by any other line. This number will be used in a short code that routes page calls to this line.
   4. In the **Analog** tab, set the following:
      - Set the **Trunk Type** to ***Loop Start***. Note: This is the only option with ATM4 trunks. With ATM16 trunks ***Ground Start*** can be used if required by the paging equipment.
      - Leave the remaining values at their defaults unless the instructions of the paging equipment manufacturer indicate that other values are required.
   5. Click **OK**.
3. **Create a Short Code for the Paging Trunk**
   1. Click the ShortCode icon to display the list of system short codes.
   2. Right-click on the list and select New.
   3. Enter the settings for the short code that users should dial when to make a paging call:
      - **Short Code: *88**
        This is the numbers users should dial to do a page. *88 is just an example.
      - **Telephone Number:**
      - **Line Group ID:** ***20***
        This must match the Outgoing Group ID set for the analog trunk.
      - **Feature:** ***Dial***
        Note that ***Dial*** is used for an analog trunk connection. ***DialPaging*** is used for an analog extension connection.
   4. Click **OK**.
   5. Send the new configuration to the IP Office and reboot.

Making Page Calls

Making Page Calls

Having setup and tested the paging equipment, users can begin to use it.

If the paging device has been connected via an analog extension port, then the page call features provided for different phones can also be used to page the extension number. Refer to the appropriate phone user guide. Otherwise users can dial the short code setup for paging.

The following methods can be used to make page calls.

Paging via a DSS Key

For extensions with DSS keys, paging can be assigned to one of those keys. The following method programs the key via the Manager application.

1. Start Manager and load the IP Office configuration.
2. Click **Users** to display the list of Users. In the list, double-click the user whose DSS keys you want to edit.
3. Select the **Button Programming** tab (**Digital Telephony** tab on UK English systems).
4. For the required DSS button, select ***Dial*** as the **Action**. For the Telephone number enter the paging short code or the extension number or the extension name in quotes.
5. Click **OK**.
6. Save the new configuration.

Paging from Phone Manager

You can add a speed dial to Phone Manager in order to make paging calls.

1. Within the users Phone Manager, select the Speed Dials tab.
2. Right-click on the tab area.
    - If paging via a analog extension port, select **Add User** and select the appropriate user.
    - If paging via an analog trunk port, select **New**. Enter a name and enter the paging short code as the number.

Group Paging

If the paging connection is via an extension port, that extension can be included in a group with other pageable extensions. This allows page calls to be heard via the speaker and over pageable telephones.

To set up group paging:

1. Create a Hunt Group with all the users required as members.
2. Create a short code to call the Hunt Group using the DialPaging feature:
    - **Short Code:** *81
    - **Telephone Number:** 305
    - **Line Group ID:** 0
    - **Feature:** DialPaging

- Note TransTalk 9040 MDW sets do not receive page calls, but may make them.

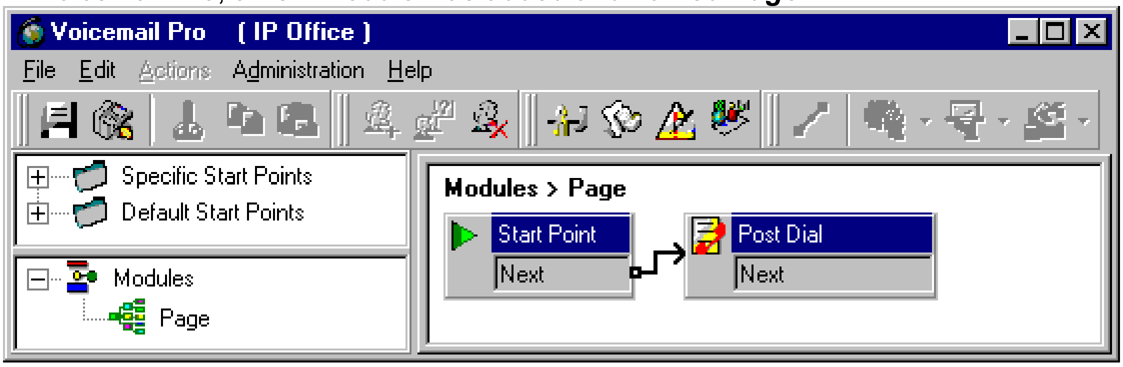
Paging Via Voicemail Pro

Voicemail Pro can be used to deliver pre-recorded announcements. This can be useful when the same announcement is repeated frequently. This method requires the paging port to be an analog extension.

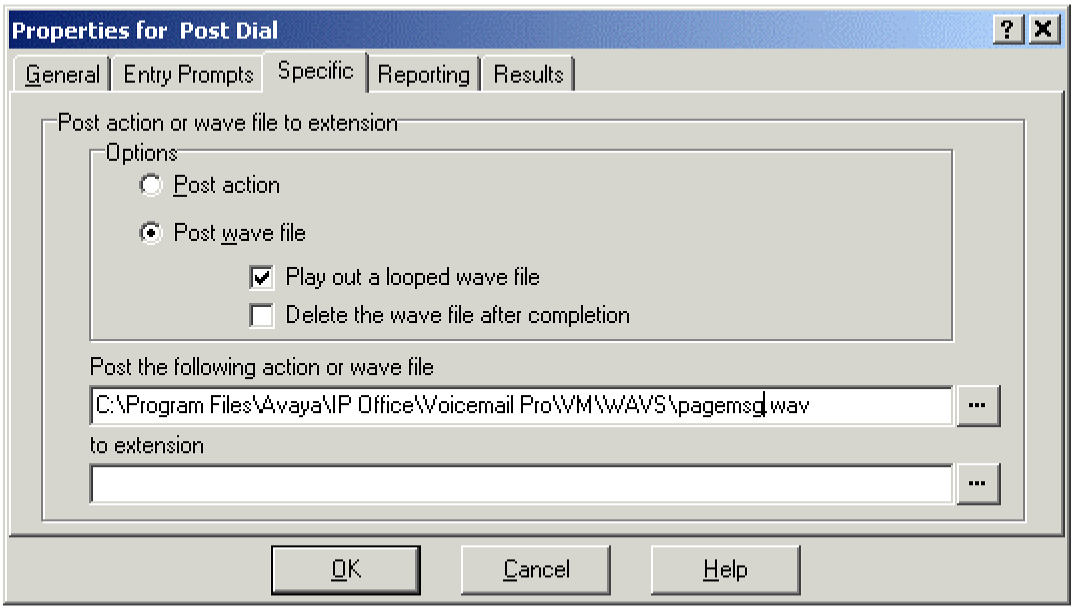
This method also removes the feedback loop that can occur on some sites as the page is first recorded and then played.

Example 1

1. In Voicemail Pro, a new Module was added and named ***Page***.



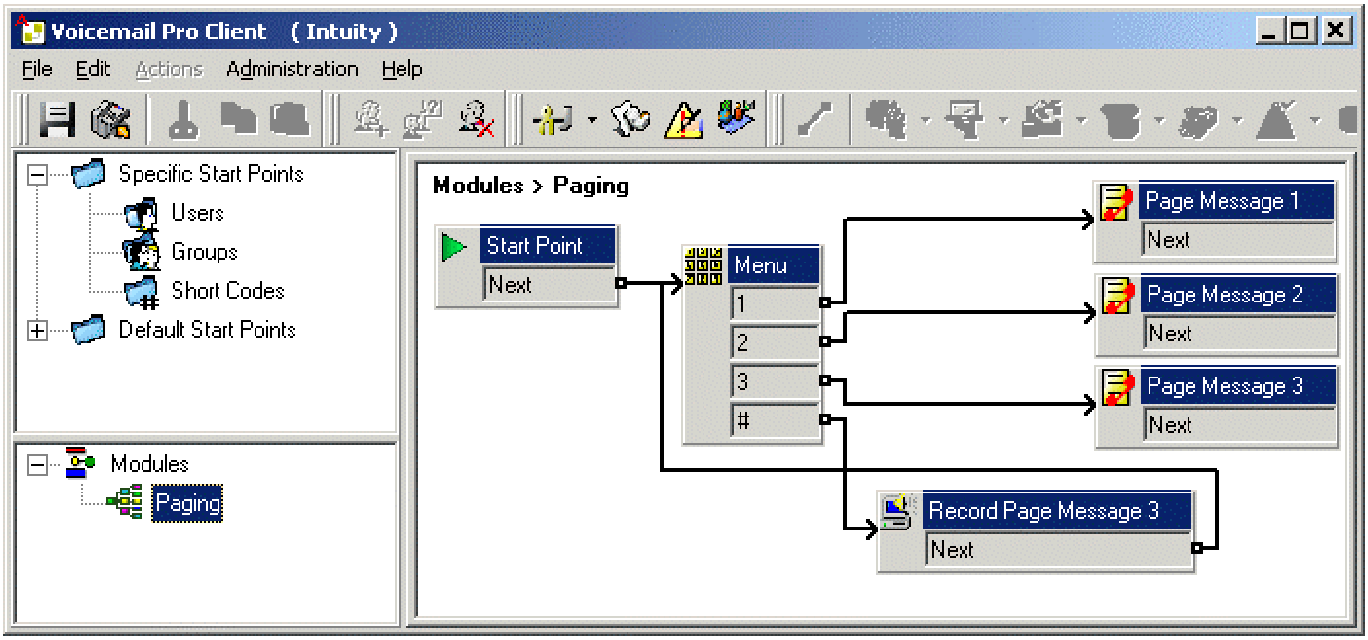
2. A **Post Dial** action was added to the module. The properties of the Specific tab were set as shown:



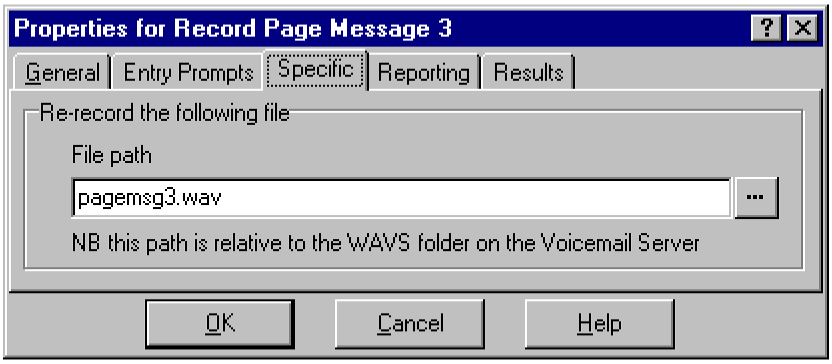
3. We then saved and made live the new Voicemail Pro call flow.
4. In Manager we received the IP Office configuration and created a new short code.
    - **Short Code:** *80
    - **Telephone Number :** "Page"
    - **Feature:** VoicemailCollect.

5. The new IP Office configuration was then merged.

Example 2
This example builds on example 1 by allowing the user to select which message is played from a menu. In this example the user can press 1, 2 or 3 for different messages. They can also re-record the message associated with option 3 by pressing #.



A **Play List** action was added and in this example set to record ***pagemsg3.wav***. Note that just the file name was specified as this action saves files relative to the Voicemail Server's WAVS folder.



In the **Post Dial** action that plays back ***pagemsg3.wav*** note that the full file path needs to be used.

In IP Office Manager, we then added a short code that triggers the module **"Paging"** using the **VoicemailCollect** feature.

Dial By Name

Dial By Name

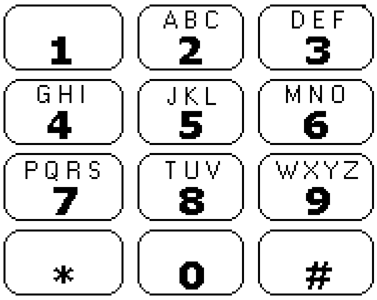
IP Office includes a Dial Name feature for making internal and external calls. It allows users to make calls by dialing the name on their telephone keypad and making a selection from the displayed matches or dialing further characters to improve the match.

When used to make internal calls, the name matches are based on the User Names and Full Names programmed into the system. If a user has a Full Name programmed then that takes precedence over their User Name.

When used to make external calls, the name matches are based on entries in the IP Office Directory.

Within a Small Community Network, remote User Names, Full Names and Group Names are shared and thus are available for the Dial By Name feature. Directories are not shared within a the Small Community Network, so only the Directory of the user's local IP Office is available.

The Dial Name feature uses the ITU key character layout:



<u>Selecting Dial Name Mode</u>:

1. Double-click **System**.
2. Select the **Telephony** tab.
3. The **Dial By Name** checkbox operates as follows.
    - When checked, matching is based on the series of characters dialed by the user.
    - When unchecked, matching is based just on the first character selected.

<u>Setting User Full Names</u>:
The process below details doing this through Manager.

1. In Manager, receive the IP Office's configuration.
2. Click **User** to display the list of users.
3. Double-click the required user to display their **User** form.
4. In the **Full Name** field enter the name required. Do not use characters other than Aa to Zz and 0 to 9.
5. Click **OK**.
6. Repeat for all users required.

Adding Directory Entries:
Note that Directory entries are also used for other functions such as name matching against received ICLID on incoming calls.

1. Click  **Directory** to display a list of current entries.
2. Either double-click on an entry to change and right-click on the list and select **New**.
3. Enter the **Name** and **Number** and click **OK**.
4. Repeat for all entries required.

Using Dial Name

The Directory function can be assigned to a programmable key on most DS phones. It is also available through the phone menu on phones with a **Menu** key.

1. Press the programmable button that has been set to the **Dir** function.
   - On phones with a **Menu** key, press **Menu** and select **Dir**. Alternatively, press **Menu** twice, then press ▶ and then select **Dir**.
2. Select from **INDeX** (internal extensions), **Group** (Hunt Groups) or **Extrn** (numbers in the IP Office Directory).
3. The next steps depend on which mode of working your system is using:
   - **Dial Name Mode** (System | Telephony | Dial by Name checked)
     1. Using the letter keys, start dialing the name that you want. For example, for names starting with ***John*** dial ***5646***. Ignore any spaces in the name.
     2. The display will show the first match to the letters entered so far. Either enter further letters or use the ◀ and ▶ keys to scroll through the other matches found.
     3. If **NO MATCH** is displayed press ◀ to go back to the previous step.
     4. When the name you want is shown, select **Call**.
     5. If you cannot find the name you want press **Exit**.
   - **Classic Mode** (System | Telephony | Dial By Name not checked)
     1. Press the dial pad button that matches the first letter of the name you want. For example, to select ***L*** press the **5** key three times.
     2. Use the ◀ and ▶ keys to scroll through the matching entries. You can press another key on the dialing pad to select a different first letter.
     3. When the name you want is shown, select **Call**.
     4. If you cannot find the name you want press **Exit**.

Index

Any comments or suggestions regarding this document should be sent to "wgctechpubs@avaya.com".

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