# IP Office 4.1

## Manager: 02. Configuration Settings

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# Contents

# Configuration Settings

## Configuration Settings

This following sections detail the various configuration settings provided for different entry types within the IP Office configuration.

Depending on the type and locale of the IP Office some settings and tabs may be hidden as they are not applicable. Other settings may be grayed out. This indicates that the setting is either for information only or that another setting needs to be enabled first.

The different entry types are:

- **BOOTP**
  Manager settings for the support of IP Office systems requesting software.
- **Operator**
  Manager settings for the editing of configuration settings from pre-3.2 IP Office systems.
- **System**
  Overall settings for the data and telephony operation of the IP Office system.
- **Line**
  Settings for trunks and trunk channels within the IP Office including IP trunks.
- **Control Unit**
  Information summary of the control and expansion units in the IP Office system.
- **Extension**
  Settings for extension ports including IP extensions.
- **User**
  Settings for IP Office users. They may or may not be associated with an extension.
- **Hunt Group**
  Collections of users to which calls can be directed for answer by any one of those users.
- **Short Code**
  These are numbers which when dialed trigger specific IP Office features or are translated for external dialing.
- **Service**
  Configuration settings such as user names and passwords needed for connections to data services such as the Internet.
- **RAS**
  Remote Access Service settings for connecting incoming data calls.
- **Incoming Call Route**
  Entries here are used to match incoming call details to destinations on the IP Office system.
- **WAN Port**
  Configuration settings for the WAN ports provided on some IP Office units.
- **Directory**
  External names and numbers. Used for matching names to incoming calls and for dialing from IP Office user applications.

- **Time Profile**
  Used to control when various IP Office functions are active.
- **Firewall Profile**
  Use to control the types of data traffic that can cross into or out of the IP Office.
- **IP Route**
  These entries are used to determine where data traffic on the IP Office LAN and WAN interfaces should be routed.
- **Least Cost Routing**
  On pre-4.0 IP Office systems LCR is used to rerouted or block outgoing external calls. For IP Office 4.0+ this has been replace by ARS.
- **Account Code**
  Used for call logging and to control the dialing of certain numbers.
- **License**
  License keys are used to enable IP Office features and applications.
- **Tunnel**
  Used to created IPSec and L2TP data tunnels.
- **Logical LAN**
  Used to allow two subnets to be run and routed on the same physical LAN.
- **Wireless**
  Used to provide wireless services on the Small Office Edition.
- **Auto Attendant**
  Used when an Embedded Voicemail card is installed on a Small Office Edition, IP406 V2 or IP500 systems.
- **User Rights**
  Provide templates to control which settings users can change.
- **ARS**
  Automatic Route Selection is used by IP Office 4.0+ to control outgoing external calls. It replaces Least Cost Routes.
- **Authorization Codes**
  Authorization codes are similar to account codes. However, unlike account codes which are useable by any user, each authorization code is only useable by a specific user or users associated with a specific set of user rights.
- **E911 System**
  Available of US systems to support E911 services.

# BOOTP Settings

## BOOTP | BOOTP Entry

BOOTP is protocol used by devices to request software when restarting. For IP Office, it is used when upgrading the control unit within a system or when the core software within the control unit has been erased. When running, Manager can respond to BOOTP requests and, if it finds a matching BOOTP entry for the system, provide the software file indicated by that entry.

BOOTP entries are not part of an IP Office system's configuration settings; instead they are saved on the Manager PC. Normally Manager automatically creates a BOOTP entry for each system with which it has communicated, up to a maximum of 50 entries. However BOOTP entries can be added and edited manually when necessary.

- **File Location**
  The location from which Manager provides files in response to BOOTP is its binaries directory. This can be changed using **File | Change Working Directory** or **File | Preferences | Working Directories**. This directory is also the directory used by Manager when providing files by TFTP.

| Control Unit | Binary File |
|---|---|
| Small Office Edition | ip401ng.bin |
| IP403 | ip403.bin |
| IP406 V1 | ip406.bin |
| IP406 V2 | ip406u.bin |
| IP412 | ip412.bin |
| IP500 | ip500.bin |
| | |

| Expansion Module | Binary File |
|---|---|
| WAN3 10/100 | ipwan3.bin |
| Phone | avpots16.bin |
| Phone V2 | dvpots.bin |
| Analog | naatm16.bin |
| Digital Station | nadcp-16.bin |
| Digital Station V2 | nadcpv2.bin |
| S08 | nas0-16.bin |

- **Disabling BOOTP**
  Manager can be disabled from providing BOOTP support for any systems. Select **File | Preferences | Enable BOOTP and TFTP Server**.

**Settings**

- **Enabled:** *Default = Enabled*
  If unticked, BOOTP support for the matching IP Office system from this Manager PC is disabled.
- **System Name**
  This field is not changeable. It shows the IP Office's system name.
- **MAC Address**
  The MAC address of the IP Office system's control unit. The address can be obtained and or verified in a number of ways:
  - When a system's configuration settings are loaded into Manager, it is shown as the **Serial Number** on the Unit form. On defaulted systems, it is also used as the system name.
  - If the system is requesting software, the MAC address is shown as part of the request in the status bar at the base of the Manager screen.
  - If the system can be pinged, it may be possible to obtain its MAC address using the command **arp -a *<ip address>***.
- **IP Address**
  The IP address of the IP Office system's LAN1.
- **Filename**
  The name of the .bin software file used by that type of control unit. For full details refer to the IP Office Installation Manual. To be transferred to the system this file must exist in the Manager applications Working Directory.
- **Time Offset:** *Default = 0.*
  In addition to performing BOOTP support for IP Office systems the Manager application can also act as a time server (RFC868). This field sets the offset between the time on the PC running Manager and the time sent to the IP Office system in response to its time requests. The field is not used if a specific **Time Server IP Address** is set through the System form in the IP Office's configuration settings.
  - Manager can be disabled from acting as an Internet Time (RFC868) server. Select **File | Preferences | Edit** and untick **Enable time server**.

# Operator Settings

## Operator | Operator

Operator entries are not part of an IP Office system's configuration settings. They are used when a pre-3.2 IP Office configuration is loaded to control what parts of a configuration can be edited.

The table below lists the settings for the default operators provided.

| Operator | View | Edit | New | Delete | Configuration Entry Types |
|---|---|---|---|---|---|
| **Administrator** | ✓ | ✓ | ✓ | ✓ | All configuration entries. |
| **Manager** | ✓ | ✓ | ✓ | ✓ | View all. Other actions **Extension**, **User**, **Hunt Group**, **Short Code**, **Service**, **RAS**, **Incoming Call Route**, **Directory**, **Time Profile**, **Firewall Profile**, **IP Route**, **Least Cost Route**, **Account Code**, **ARS**. |
| **Operator** | ✓ | ✓ | ✗ | ✗ | View all configuration entries. Edit all except **System**, **Line**, **Control Unit** and **Authorization Codes**. |

If when receiving a configuration from a pre-3.2 IP Office system an invalid operator is specified, the settings will be loaded using the **Guest** operator. This additional operator allows a read-only view.

# System Settings

## System Form Overview

 There is only one System entry for each IP Office system.

The following tabs are part of the **System** form:

- **System**
  General settings for the IP Office system.
- **LAN1**
  Network settings for the main RJ45 Ethernet ports on the IP Office control unit. Includes DHCP and RIP settings.
- **LAN2**
  LAN2 is not supported by all IP Office control units.
  - On the Small Office Edition and IP500 control units the LAN2 settings are used for the RJ45 Ethernet port labeled **WAN**.
  - On the IP412 control units LAN2 settings are used for the RJ45 ethernet port labelled **LAN2**.
  - For IP406 V2 control units running IP Office 4.1+, the RJ45 LAN port 8 can be selected to act be LAN2 if required.
- **DNS**
  Specify the Domain Name Server addresses to use for address resolution.
- **Voicemail**
  Details the type and location of the IP Office's voicemail server.
- **Telephony**
  System-wide telephony settings.
- **H.323 Gatekeeper**
  Settings used for VoIP endpoints registering with IP Office and for DiffServ QoS settings applied to VoIP traffic. This tab is only shown on pre-4.0 IP Office systems. For IP Office 4.0+ separate LAN1 and LAN2 H.323 settings are shown on the LAN1 and LAN2 tabs respectively.
- **LDAP**
  Settings to allow the IP Office to include Lightweight Directory Access Protocol database records in its directory.
- **System Events**
  Simple Network Management Protocol (SNMP), email (SMTP), and Syslog settings for the sending of IP Office events.
- **Twinning**
  System wide controls for the use of Mobile Twinning.
- **CDR**
  Call Detail Record settings for the sending of call detail records to a specified IP address.

# System | System

| System \| System | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓Changes to **Locale**, **License Server IP Address** and **Favor RIP Routes over Static Routes** require a reboot.. |

- **Name:** *Default = Control unit MAC address.*
  A name to identify this system. This is typically used to identify the configuration by the location or customer's company name. Some features such as Gatekeeper require the system to have a name. This field is case sensitive and within any network of IP Offices must be unique. Do not use punctuation characters such as #, ?, /, -,. and ,.
- **Contact Information:** *Default = Blank, Software level = 4.1+.*
  This field is only be edited by service user with administrator rights. If Contact Information is entered, it will set the IP Office system under 'special control'.
  - If the contact information is set using Avaya Integrated Management (AIM), warnings that "This configuration is under Integrated Management Control" are given if the configuration is opened using a standalone version of IP Office Manager. See Loading a Configuration.
  - If the contact information is set using a standalone version of Manager, warnings that "This configuration is under special control" are given when the configuration is opened again. This can be used to warn other users of Manager that the IP Office system is being monitored for some specific reason and provide them with contact details of the person doing that monitoring. See Loading a Configuration.
- **Locale**
  This setting sets default telephony and language settings based on the selection. See **Supported Country and Locale Settings**. For individual users the system settings can be overridden through their own locale setting (**User | User | Locale**). If this option is selected, it is the installers responsibility to ensure that the settings selected match those required by the line providers.
  - **Customize Locale Settings:** *Software level = IP Office 4.0 Q2 2007 Maintenance Release +*
    The ***Customize*** locale matches the Saudi Arabia locale but with the following additional controls:
    - **Tone Plan:** *Default = Tone Plan 1*
      The tone plan control dial and ringing tone. The options are:
      - **Tone Plan 1:** United States.
      - **Tone Plan 2:** United Kingdom.
      - **Tone Plan 3:** France.
      - **Tone Plan 4:** Germany.
      - **Tone Plan 5:** Spain.
    - **CLI Type:** *Default = FSK V23*
      This is the method used for CLI signalling on analog lines.
    - **Busy Tone Detection:** *Default = Off.*
      Enables or disables the use of busy tone detection for call clearing. This is a system wide setting.

- **Password:** *Default = password, Software level = 2.1 to 3.1.*
  A password for controlling access to the operation of the Control Unit. This is required to upgrade and reboot and to send or receive configurations from the Control Unit. This is a required option and a prompt is given if left blank. For IP Office 3.2+ systems this setting has become part of the security settings.
- **Monitor Password:** *Default = blank, Software level = 2.1 to 3.1.*
  This password is used by the Monitor and Call Status applications to allow communication with the main unit. If left blank these applications will use the System Password above. For IP Office 3.2+ systems this setting has become part of the security settings.
- **Time Offset:** *Default = 00:00*
  This setting can be used if the IP Office is in a different time zone from its time server. For example, if the IP Office is 5 hours behind the time server, this field should be configured with -5:00 to make the adjustment. The time offset can be adjusted in 15 minute increments. Note: If the time server is a Manager PC, the adjustment can also be done through the Manager BOOTP entry for the system.
- **TFTP Server IP Address:** *Default = 0.0.0.0 (Broadcast)*
  When Manager is running, it can act as the TFTP server for files required by the IP Office control unit and 4600/5600 Series phones. An entry here forces those devices to use the TFTP server at the indicated address. On Small Office Edition, IP406 V2 and IP500 systems, the LAN1 IP Address can be entered to specify the memory card in their **PCMCIA** or **CF TII** slot.
- **Branch Prefix:** *Default = Blank, Range = 0 to 999999999, Software Level = 4.1+.*
  Used to identify the IP Office system within an Avaya SIP for Branch network linked via an SES server. The branch prefixes of each IP Offices within the network must unique and must not overlap. For example 85, 861 and 862 are okay but 86 and 861 overlap. See **SES Line** for more details.
- **Local Number Length:** *Default = Blank (Off), Range = Blank (Off) or 3 to 9, Software level = 4.1+.*
  Set the default length for extension numbers for extensions, users and hunt groups. Entry of an extension number of a different length will cause an error warning within IP Office Manager. This field is intended for IP Office systems being used in an Avaya SIP for Branch network linked via an SES server, where fixed extension lengths must be used. Note that the **Branch Prefix** and the **Local Number Length** must not exceed 15 digits.
- **Time Server IP Address:** *Default = 0.0.0.0 (Default)*
  The IP Office control unit contains a battery backed clock used to maintain system time during normal operation and when mains power is removed. The time is obtained using Internet Time protocol (RFC868) requests. Entering 0.0.0.1 disables time server updates.
  - 0.0.0.0 means default operation. In this mode, following a reboot the IP Office control unit will send 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 broadcast requests.
  - The Voicemail Lite Server, Voicemail Pro Service and the Manager program can all act as time servers for the IP Office, giving the time as set on their host PC.
  - If you are running Manager when the voicemail server starts, 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. Manager can be disabled from acting as a time server by deselecting the **Enable Time Server** option (**File | Preferences | Edit | Preferences**).
- **File Writer IP Address:** *Default = 0.0.0.0 (Disabled)*
  For Small Office Edition, IP406 V2 and IP500 control units only. This field sets the address of the PC allowed to send files to the memory card in their **PCMCIA** or **CF TII** slot.

- **License Server IP Address:** *Default = 255.255.255.255*
  This is the IP address of the server providing license key validation for the IP Office. The serial number of the Feature Key dongle at that address must match the serial number used to generate the licenses in the IP Office system's configuration. See **License** for more details.
  - For parallel and USB Feature Key dongles this address should be the IP address of a PC running the IP Office Feature Key server software. Note that separate IP Offices cannot use the same Feature Key server for license validation, nor can an IP Office validate its licenses against more than one address.
  - For a serial port Feature Key dongle, the address should be set to 0.0.0.0.
  - For IP500 systems this field is hidden as those systems must use a smart card Feature Key dongle inserted into the control unit.
- **Dongle Serial Number**
  This field is for information only. It shows the serial number of the feature key dongle against which the IP Office system last validated its licenses. ***Local*** is shown for a serial port or Smart Card feature key plugged directly into the control unit. ***Remote*** is shown for a parallel or USB feature key connected to a feature Key Server PC.
- **AVPP IP Address:** *Default = 0.0.0.0 (Disabled)*
  Where Avaya 3600 Series SpectraLink wireless handsets are being used with the IP Office, this field is used to specify the IP address of the Avaya Voice Priority Processor (AVPP).
- **Conferencing Center URL:** *Default = Blank (Disabled).*
  This is the root URL of the web server being used to support Conferencing Center, for example ***http://server/***. This address is then by the Phone Manager and SoftConsole applications to launch Conference Center functions. In Phone Manager setting this value enables use of the join conference controls.
- **DSS Status:** *Default = Off*
  This setting affects Avaya display phones with programmable buttons. It controls whether pressing a DSS key set to another user who has a call ringing will display details of the caller. When off, no caller information is displayed.
- **Beep on Listen:** *Default = On (USA)/On (Rest of World)*
  This setting controls whether call parties hear a repeating tone when their call is monitored by another party using the Call Listen feature.
- **Hide auto record:** *Default = On (USA)/Off (Rest of World)*
  During call recording by Voicemail Pro, some Avaya terminals display **REC** or similar to show that the call is being recorded. When on, **Hide auto record** suppresses this recording indication.
- **Favour RIP Routes over Static Routes:** *Default = Off*
  RIP can be enabled on the IP Office **LAN1** and **LAN2** interfaces, and on specific **Services.** When this setting is on, the RIP route to a destination overrides any static route to the same destination in the IP Office's **IP Routes**, regardless of the RIP route's metric. The only exception is RIP routes with a metric of 16 which are always ignored. Note: If a previously learnt RIP route fails, the IP Office applies a metric of 16 five minutes after the failure. When off, any RIP route to a destination for which a static route has been configured is ignored.

# System | LAN1

This tab is used to configure the behavior of the RJ45 Ethernet ports labeled LAN or LAN1 on the IP Office control unit.

For IP Office 4.0+ this form contains 3 sub-tabs: **LAN Settings**, **Gatekeeper** and **Network Topology**. The **Gatekeeper** tab contains settings perviously located on the **System | H323 Gatekeeper** tab.

Depending on the type of IP Office control unit, the relationship between the physical RJ45 Ethernet ports and **LAN1** and **LAN2** within the IP Office configuration varies as follows:

- **IP Office 500**
  This unit has 2 RJ45 Ethernet ports, marked as **LAN** and **WAN**. These form a full-duplex managed layer-3 switch.
  Within the IP Office configuration, the physical LAN port is LAN1, the physical WAN port is LAN2.
- **IP460 V2**
  This unit has 8 RJ45 Ethernet ports marked as **LAN 1** to **8**. These form a full-duplex unmanaged layer-2 LAN switch. Ports are auto-MDI/MDIX. Within the IP Office configuration the physical LAN ports are LAN1.
  - For IP Office 4.1+, port 8 can be configured to act as LAN2 using the **Use Port 8 as LAN2** option on the LAN1 **LAN Settings** tab.
- **IP412**
  This unit has 2 RJ45 Ethernet ports marked as **LAN 1** to **2**. These form a half-duplex managed layer-3 switch. Both ports are fixed MDI crossover ports. Within the IP Office configuration, physical port 1 is LAN1, physical port 2 is LAN2.
- **Small Office Edition**
  This unit has 4 RJ45 Ethernet ports marked LAN 1 to 4. These form a full-duplex unmanaged layer-2 switch. An addition RJ45 Ethernet socket marked as WAN exists. With the LAN ports this acts as a managed layer-3 switch. Within the IP Office configuration, the physical LAN ports are LAN1, the physical WAN port is LAN2.

### LAN Settings

| System \| LAN1 \| LAN Settings | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

- **IP Address:** *Default = 192.168.42.1*
  This is the IP address of the Control Unit on LAN1. If the control unit is also acting as a DHCP server on LAN1 then this address is the starting address of the DHCP address range.
- **IP Mask:** *Default = 255.255.255.0*
  This is the IP subnet mask used with the IP address.
- **Use Port 8 as LAN2:** *Default = Off, Software level = 4.1+.*
  This option is provided for IP406 V2 control units with IP Office 4.1 or higher software only. When selected, LAN port 8 on the control unit acts as **LAN2** for the IP Office control unit. Note that this setting is retained by the IP Office even if the system is defaulted. To change the setting, the value in the configuration should be changed and the configuration sent back to the system for an immediate reboot.
- **Primary Trans. IP Address:** *Default = 0.0.0.0 (Disabled)*
  This setting is only available on control units that support a LAN2. Any incoming IP packets without a service or session are translated to this address if set.

- **RIP Mode:** *Default = None*
  Routing Information Protocol (RIP) is a method by which network routers can exchange information about device locations and routes. Routes learnt using RIP are known as 'dynamic routes'. The IP Office also supports 'static routes' though its **IP Route** entries.
  - **None**
    The LAN does not listen to or send RIP messages.
  - **Listen Only (Passive)**
    Listen to RIP-1 and RIP-2 messages in order to learn RIP routes on the network.
  - **RIP1**
    Listen to RIP-1 and RIP-2 messages and send RIP-1 responses as a sub-network broadcast.
  - **RIP2 Broadcast (RIP1 Compatibility)**
    Listen to RIP-1 and RIP-2 messages and send RIP-2 responses as a sub-network broadcast.
  - **RIP2 Multicast**
    Listen to RIP-1 and RIP-2 messages and send RIP-2 responses to the RIP-2 multicast address.
- **Enable NAT:** *Default = Off*
  This setting is only available on Small Office Edition, IP412 and IP500 systems. It controls whether NAT should be used for IP traffic from LAN1 to LAN2. This setting should not be used on the same LAN interface as a connected WAN3 expansion module.
- **Number of DHCP IP Addresses:** *Default = 200, Range = 1 to 999.*
  This defines the number of sequential IP addresses, including the Control Unit IP address, available for DHCP clients.
- **DHCP Mode**
  This controls the control unit's DHCP mode for LAN1. When doing DHCP; LAN devices are allocated addresses from the bottom of the available address range upwards; Dial In users are allocated addresses from the top of the available range downwards. If the control unit is acting as a DHCP server on LAN1 and LAN2, Dial in users are allocated their address from the LAN1 pool of addresses first.
  - **Server**
    When selected the Control Unit is acting as the DHCP Server on LAN1, allocating address to other devices on the network and to PPP Dial in users.
  - **Disabled**
    When selected the Control Unit will not use DHCP, therefore it will not act as a DHCP server or obtain an IP address from a DHCP server on this LAN.
  - **Dial In**
    This option allows the Control Unit to allocate IP addresses to PPP Dial In users only. It will not allocate IP addresses to local devices on this LAN.
  - **Client**
    The Control Unit obtains its **IP Address** and **IP Mask** from a DHCP server on the LAN.

**Gatekeeper Settings Tab**

These settings are only shown on IP Office 4.0+ systems. On pre-4.0 IP Office systems the H.323 settings were located on a separate **System | H.323 Gatekeeper** tab for the whole system. These settings relate to the support of H323 extension and trunks on the LAN1 interface including SIP trunks.

| **System \| LAN1 \| Gatekeeper Settings** | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✗, **3.0DT** ✗, **3.0** ✗, **3.1** ✗, **3.2** ✗, **4.0** ✓. |
| **Mergeable** | **Pre-3.2** ✗, **3.2+** ✗. |

- **H323 Gatekeeper Enable:** *Default = On*
  This settings enables IP Office gatekeeper operation.
- **SIP Proxy Enabled:** *Default = On*
  This settings enables support of SIP trunks. It also requires entry of a SIP Trunk Channels license.
- **H323 Auto-create Extn:** *Default = On*
  When this option is on, an extension entry is automatically created for H.323 phones registering themselves with the IP Office as their gatekeeper.
- **H323 Auto-create User:** *Default = Off*
  When this option is on and **H323 Auto-create Extn** is also on, when a new H.323 extension is created a matching user record is also created.
- **RTP Port Number Range:** *Software level = 3.0+.*
  For each VoIP call, a receive port for incoming Real Time Protocol (RTP) traffic is selected from a defined range of possible ports, using the even numbers in that range. The Real Time Control Protocol (RTCP) traffic for the same call uses the RTP port number plus 1, that is the odd numbers. For IP Office control units and Avaya H.323 IP phones, the default port range used is 49152 to 53246. On some installations, it may be a requirement to change or restrict the port range used. It is recommended that only port numbers between 49152 and 65535 are used, that being the range defined by the Internet Assigned Numbers Authority (IANA) for dynamic usage.
  - **Port Range (minimum):** *Default = 49152. Range = 1024 to 64510.*
    This sets the lower limit for the RTP port numbers used by the IP Office.
  - **Port Range (maximum):** *Default = 53246. Range = 2048 to 65534.*
    This sets the upper limit for the RTP port numbers used by the IP Office. The gap between the minimum and the maximum must be at least 1024.
- **DiffServe Settings**
  When transporting voice over low speed links it is possible for normal data packets (1500 byte packets) to prevent or delay voice packets (typically 67 or 31 bytes) from getting across the link. This can cause unacceptable speech quality. Therefore it is important that all traffic routers and switches in a network to have some form of Quality of Service mechanism (QoS). QoS routers are essential to ensure low speech latency and to maintain sufficient audible quality.

  IP Office supports the DiffServ (RFC2474) QoS mechanism. This uses a Type of Service (ToS) field in the IP packet header. The IP Office uses this field to prioritize voice and voice signaling packets on its WAN interfaces. Note that the IP Office does not perform QoS for its Ethernet ports including the WAN Ethernet port on the Small Office Edition.

The hex and decimal entry fields for the following values are linked, the hex value being equal to the decimal multiplied by 4.

- **DSCP (Hex):** *Default = B8 (Hex)/46 (decimal), Range = 00 to FC (Hex)/0 to 63 (decimal)*
  The DiffServ Code Point (DSCP) setting applied to VoIP calls. For correct operation, especially over WAN links, the same value should be set at both ends.
- **DSCP Mask (Hex):** *Default = FC (Hex)/63 (decimal), Range = 00 to FC (Hex)/0 to 63 (decimal)*
  Allows a mask to be applied to packets for the DSCP value.
- **SIG DSCP (Hex):** *Default = 88 (Hex)/34 (decimal), Range = 00 to FC (Hex)/0 to 63 (decimal)*
  This setting is used to prioritize VoIP call signaling.

- **Site Specific Option Number (SSON):** *Default = 176, Range 128 to 254.*
  Sets the site specific option number (SSON) used by the IP Office's internal DHCP server. This should match the SSON used by 4600 and 5600 Series IP phones to request installation settings.

**Network Topology Settings**

These settings are used for SIP trunk connections from the LAN. Use of SIP requires entry of SIP Trunk Channels licenses. For further details of IP Office SIP operation refer to the SIP Line section.

STUN (Simple Traversal of UDP through NAT) is a mechanism used with UDP SIP to overcome the effect of NAT firewalls. Test SIP packets are sent to a STUN server. The STUN server replies and includes copies of the packets it received in the reply. By comparing the packet sent to the STUN server and the copies of the packets it received, it is possible to determine the type of NAT firewall and to then modify future SIP packets to overcome negative effects of NAT.

The use of STUN is unnecessary if the SIP ITSP uses a Session Border Controller (SBC).

| System \| LAN1 \| Network Topology Settings | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✗, **3.0DT** ✗, **3.0** ✗, **3.1** ✗, **3.2** ✗, **4.0** ✓. |
| **Mergeable** | **Pre-3.2** ✗, **3.2+** ✗. |

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

- **STUN Server IP Address:** *Default = 69.90.168.13*
  This is the IP address of the SIP ITSP's STUN server. The IP Office will send basic SIP messages to this destination and from data inserted into the replies can try to determine the type NAT changes being applied by any firewall between it and the ITSP.
- **STUN Port:** *Default = 3478, Software level = 4.1+.*
  Defines the port to which STUN requests are sent if STUN is used.
- **Firewall/NAT Type:** *Default = Unknown*
  The settings here reflect different types of network firewalls. Options include **Blocking Firewall**, **Symmetric Firewall**, **Open Internet**, **Symmetric NAT**, **Full Cone NAT**, **Restricted Cone NAT**, **Port Restricted Cone NAT** and **Unknown**.
  - **Open Internet**
    No action required. If this mode is selected, STUN lookups are not performed.
  - **Symmetric Firewall**
    SIP packets are unchanged but ports need to be opened and kept open with keep-alives. If this type of NAT is detected or manually selected, a warning ***'Communication is not be possible unless the STUN server is supported on same IP address as the ITSP'*** will be displayed as part of the manager validation.
  - **Full Cone NAT**
    A full cone NAT is one where all requests from the same internal IP address and port are mapped to the same external IP address and port. Furthermore, any external host can send a packet to the internal host, by sending a packet to the mapped external address. SIP packets need to be mapped to NAT address and Port; any Host in the internet can call in on the open port, that is the local info in the SDP will apply to multiple ITSP Hosts. No warning will be displayed for this type of NAT because the IP Office has sufficient information to make the connection).

    - **Symmetric NAT**
      A symmetric NAT is one where all requests from the same internal IP address and port, to a specific destination IP address and port, are mapped to the same external IP address and port. If the same host sends a packet with the same source address and port, but to a different destination, a different mapping is used. Furthermore, only the external host that receives a packet can send a UDP packet back to the internal host. SIP Packets need to be mapped but STUN will not provide the correct information unless the IP address on the STUN server is the same as the ITSP Host. If this type of NAT/Firewall is detected or manually selected, a warning ***'Communication is not possible unless the STUN server is supported on same IP address as the ITSP'*** will be displayed as part of the manager validation.
    - **Restricted Cone NAT**
      A restricted cone NAT is one where all requests from the same internal IP address and port are mapped to the same external IP address and port. Unlike a full cone NAT, an external host (with IP address X) can send a packet to the internal host only if the internal host had previously sent a packet to IP address X. SIP packets needs to be mapped. Responses from hosts are restricted to those that a packet has been sent to. So if multiple ITSP hosts are to be supported, a keep alive will need to be sent to each host. If this type of NAT/Firewall is detected or manually selected, no warning will be displayed for this type of NAT.
    - **Port Restricted Cone NAT**
      A port restricted cone NAT is like a restricted cone NAT, but the restriction includes port numbers. Specifically, an external host can send a packet, with source IP address X and source port P, to the internal host only if the internal host had previously sent a packet to IP address X and port P. SIP packets needs to be mapped. Keep-alives must be sent to all ports that will be the source of a packet for each ITSP host IP address. If this type of NAT/Firewall is detected or manually selected, no warning will be displayed for this type of NAT. However, some Port Restricted NAT's have been found to be more symmetric in behavior, creating a separate binding for each opened Port, if this is the case the manager will display a warning ***'Communication is not possible unless the STUN server is supported on same IP address as the ITSP'*** as part of the manager validation.
    - **Static Port Block:** *Software level = 4.1+.*
      Use the **RTP Port Number Range** specified on the **Gatekeeper** tab without STUN translation. Those ports must be fixed as open on any NAT firewall involved.

- **Binding Refresh Time (seconds):** *Default = 0 (Never), Range = 0 to 3600 seconds.*
  Having established which TCP/UDP port number to use, through either automatic or manual configuration, the IP Office can send recurring 'SIP Options requests' to the remote proxy terminating the trunk. Those requests will keep the port open through the firewall. Requests are sent every x seconds as configured by this field.
    - Note: If a binding refresh time has not been set you may experience problems receiving inbound SIP calls as they are unable to get through the Firewall. In these circumstances make sure that this value has been configured.
- **Public IP Address:** *Default = 0.0.0.0*
  This value is either entered manually or discovered by the **Run STUN** process. If no address is set, the IP Office LAN1 address is used.
- **Public Port:** *Default = 0*
  This value is either entered manually or discovered by the **Run STUN** process.

- **Run STUN**
  This button tests STUN operation between the IP Office LAN and the STUN Server IP Address set above. If successful the results are used to automatically fill the remaining fields with appropriate values discovered by the IP Office. Before using **Run STUN** the SIP trunk must be configured.
  - When this option is used, a information icon is shown against the fields to indicate that the values were automatically discovered rather than manually entered.
- **Run STUN on startup:** *Default = Off*
  This option is used in conjunction with values automatically discovered using **Run STUN**. When selected, the IP Office will rerun STUN discovery whenever the IP Office is rebooted or connection failure to the SIP server occurs.

# System | LAN2

LAN2 is not supported by all IP Office control units.

- On the Small Office Edition and IP500 control units the LAN2 settings are used for the RJ45 Ethernet port labeled **WAN**.
- On the IP412 control units LAN2 settings are used for the RJ45 ethernet port labelled **LAN2**.
- For IP406 V2 control units running IP Office 4.1+, the RJ45 LAN port 8 can be selected to act be LAN2 if required. This is done using the U**se Port 8 as LAN2** option on the **LAN1 | LAN Settings** tab.

| **System \| LAN2** | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✗, **IP406 V1** ✗, **IP406 V2** ✓*, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

*Optional on IP Office 4.1+.

The fields available for LAN2 are the same as for LAN1 except for the following additional field:

- **Firewall:** *Default = <None> (No firewall)*
  Allows the selection of an IP Office firewall to be applied to traffic routed from LAN2 to LAN1.

# System | DNS

DNS is a mechanism through which the URL's requested by users, such as ***www.avaya.com***, are resolved into IP addresses. These requests are sent to a Domain Name Server (DNS) server, which converts the URL to an IP address. Typically the internet service provider (ISP) will specify the address of the DNS server their customers should use.

WINS (Windows Internet Name Service) is a similar mechanism used within a Windows network to convert PC and server names to IP addresses via a WINS server.

If the IP Office is acting as a DHCP server, in addition to providing clients with their own IP address settings for **LAN1** or **LAN2** it can also provide them with their DNS and WINS settings if requested by the client

| System \| DNS | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

- **DNS Service IP Address:** *Default = 0.0.0.0 (Do not provide DNS/Use DNS forwarding)*
  This is the IP address of an DNS Server. Your Internet service provider or network administrator provides this information. If this field is left blank, the IP office uses its own address as the DNS server for DHCP client and forward DNS requests to the service provider when **Request DNS** is selected in the service being used (**Service | IP**).
    - **Backup DNS Server IP Address:** *Default = 0.0.0.0 (No backup)*
- **DNS Domain:** *Default = Blank (No domain)*
  This is the domain name for your IP address. Your Internet service provider or network administrator provides this. Typically this field is left blank.
- **WINS Server IP Address:** *Default = 0.0.0.0 (Do not provide WINS)*
  This is the IP address of your local WINS server. This is only used by Windows PCs, and normally points to an NT server nominated by your network administrator as your WINS server. Setting a value will result in also sending a mode of "hybrid".
    - **Backup WINS Server IP Address:** Default = 0.0.0.0 (No backup)
- **WINS Scope:** *Default = Blank (no scope)*
  This is provided by your network administrator or left blank.

# System | Voicemail

The following settings are used to set the IP Office's voicemail server type and location. The fields are enabled or grayed out as appropriate to the selected voicemail type.

| System \| Voicemail | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | **Pre-3.2** ✗, **3.2+** ✓*Changes to **Voicemail Type** require a reboot. |

- **Voicemail Type:** *Default = Voicemail Lite/Pro*
  Sets the type of voicemail system being used.
  - **None**
    No voicemail operation.
  - **Centralized Voicemail**
    Select this option when using a Voicemail Pro system installed and licensed on another IP Office in an IP Office Small Community Network. The outgoing line group for connection to the system with the Voicemail Pro should be entered as the **Voicemail Destination**.
  - **Embedded Voicemail**
    Small Office Edition, IP406 V2 and IP500 control units only. Select this option to run embedded voicemail which stores messages and prompts on an Avaya memory card inserted into the control unit.
  - **Group Voicemail**
    This option is used to support third-party voicemail systems attached by extension ports in the group specified as the **Destination**.
  - **Remote Audix Voicemail**
    Select this option if using a remote Avaya Intuity Audix or MultiMessage voicemail system. Requires entry of an ***Audix Voicemail*** license in **Licenses**.
  - **Voicemail Lite/Pro**
    Select this option when using Voicemail Lite or Voicemail Pro. The IP address of the PC being used should be set as the **Voicemail IP Address**. Use of Voicemail Pro requires entry of a ***Voicemail Pro (4 ports)*** license in **Licenses**.
- **Voicemail Destination:** *Default = blank*
  This setting is used when the **Voicemail Type** is set to ***Remote Audix Voicemail*** or ***Centralized Voicemail***. It is used to enter the outgoing line group of the lines configured for connection to the remote voicemail system. It is also used for ***Group Voicemail*** to specify the group connected to the voicemail system.
- **Voicemail IP Address:** *Default = 255.255.255.255*
  This setting is used when the **Voicemail Type** is set to ***Voicemail Lite/Pro***. It is the IP address of the PC running the Voicemail Lite or Voicemail Pro server. If set as 255.255.255.255, the control unit broadcasts on the LAN for a response from a voicemail server. If set to a specific IP address, the system connects only to the voicemail server running at that specific address.
- **Voicemail Password :** *Default = blank, Software level = 2.1 to 3.1.*
  The Voicemail Password is used by the main unit to confirm connection has been made to the correct Voicemail Pro Server. The password entered must correspond to the password set via the Voicemail Pro software. This entry must be left blank when using the standard Voicemail application supplied on the Admin CD. For IP Office 3.2 and higher this value is set through the IP Office security settings.
- **Audix UDP**
  Available if the voicemail type **Remote Audix Voicemail** is selected. Needs to be completed with a four digit number from the Universal Dial Plan of the Avaya Communication Manager system.

- **Maximum Record Time:** *Default = 120 seconds, Range = 30 to 180 seconds, Software level = 3.0+.*
  This field is only available when **Embedded Voicemail** is selected as the **Voicemail Type**. The value sets the maximum record time for messages and prompts.

- **Voicemail Channel Reservation:** *Software level = 4.0+*
  These settings allow the channels between the IP Office and its voicemail sever to be reserved for particular functions. Unreserved channels can be used for any function but reserved channels cannot be used for any function other than that indicated. These settings are not available unless the configuration includes validated licenses for the number of voicemail channels.

  - **Unreserved Channels**
    This setting cannot be changed and by default will show the total number of licensed voicemail channels. This number will decrease as channels are reserved for the following functions.

  - **Mailbox Access:** *Default = 0*
    This setting sets the number of channels reserved for users accessing mailboxes to collect messages.

  - **Auto-Attendant:** *Default = 0*
    This setting sets the number of channels reserved for users directed to Voicemail Pro short code and module start points.

  - **Voice Recording:** *Default = 0*
    This setting sets the number of channels reserved for voice recording other than mandatory voice recording (see below). If no channels are available recording does not occur though recording progress may be indicated.

  - **Mandatory Voice Recording:** *Default = 0*
    This setting sets the number of channels reserved for mandatory voice recording. When no channels are available for a call set to mandatory recording, the call is barred and the caller hears busy tone.

  - **Announcements:** *Default = 0*
    This setting sets the number of channels reserved for announcements. When no channels are available calls continue without announcements.

# System | Telephony

This tab is used to set the default telephony operation of the IP Office. Some settings shown here can be overridden for individual users through their **User | Telephony** tab.

| System \| Telephony | |
|---|---|
| **Control Unit** | **SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓.** |
| **Software Level** | **2.1 ✓, 3.0DT ✓, 3.0 ✓, 3.1 ✓, 3.2 ✓, 4.0 ✓, 4.1 ✓.** |
| **Mergeable** | **Pre-3.2 ✗, 3.2+ ✓*** |

*Changes to **Companding LAW**, **Busy Tone Detection** and **Use External Music on Hold** require a reboot.

- **Default Outside Call Sequence:** *Default = Normal*
  This setting is only used with analog extensions. It sets the ringing pattern used for incoming external calls. For details of the ring types see Ring Tones in the **Telephone Features** section. This setting can be overridden by a user's **User | Telephony | Outside Call Sequence** setting. Note that changing the pattern may cause fax and modem device extensions to not recognize and answer calls.
- **Default Inside Call Sequence:** *Default = Ring Type 1*
  This setting is only used with analog extensions. It sets the ringing pattern used for incoming internal calls. For details of the ring types see Ring Tones in the **Telephone Features** section. This setting can be overridden by a user's **User | Telephony | Inside Call Sequence** setting.
- **Default Ring Back Sequence:** *Default = Ring Type 2*
  This setting is only used with analog extensions. It sets the ringing pattern used for ringback calls such as hold return, park return, voicemail ringback, and Ring Back when Free. For details of the ring types see Ring Tones in the **Telephone Features** section. This setting can be overridden by a user's **User | Telephony | Ringback Call Sequence** setting.
- **Dial Delay Time (secs):** *Default = 4 (USA/Japan) or 1 (ROW), Range = 1 to 99 seconds.*
  This setting sets the time the system waits following a dialed digit before it starts looking for a short code match. In situations where there are potential short codes matches but not exact match, it also sets the delay following the dialing of a digit before dialing complete is assumed. See the **Short Codes** section.
- **Dial Delay Count:** *Default = 0 digits (USA/Japan) or 4 digits (ROW), Range = 0 to 30 digits.*
  This setting sets the number of digits dialed after which the IP Office starts looking for a short code match regardless of the **Dial Delay Time**.
- **Default No Answer Time (secs):** *Default = 15 seconds, Range = 1 to 99999 seconds.*
  This setting controls the amount of time before an alerting call is considered as unanswered. How the call is treated when this time expires depends on the call type.
  - For calls to a user, the call follows the user's Forward on No Answer settings if enabled. If no forward is set, the call will go to voicemail if available or else continues to ring. This timer is also used to control the duration of call forwarding if the forward destination does not answer. It also controls the duration of ringback call alerting. This setting is overridden by the **User | Telephony | No Answer Time** setting for a particular user if different.
  - For calls to hunt groups, this setting controls the time before the call is presented to the next available hunt group member. This setting is overridden by the **Hunt Group | Hunt Group | No Answer Time** setting for a particular hunt group if different.
- **Hold Timeout (secs):** *Default = 120 (US) or 15 (ROW), Range = 0 (Off) to 99999 seconds.*
  This setting controls how long calls remain on hold before recalling to the user who held the call. Note that the recall only occurs if the user has no other connected call. Recalled calls will continue ringing and do not follow forwards or go to voicemail.
- **Park Timeout (secs):** *Default = 300 seconds, Range 0 (Off) to 99999 seconds.*
  This setting controls how long calls remain parked before recalling to the user who parked the

call. Note that the recall only occurs if the user has no other connected call. Recalled calls will continue ringing and do not follow forwards or go to voicemail.

- **Ring Delay:** *Default = 5 seconds, Range = 0 to 98 seconds, Software level = 3.2+.*
  This setting is used when any of the user's programmed appearance buttons is set to Delayed ringing. Calls received on that button will initially only alert visually. Audible alerting will only occur after the ring delay has expired. This setting can be overridden by a ring delay set for an individual user (**User | Telephony | Ring Delay**).
- **Local Dial Tone:** *Default = On*
  For all normal operation this setting should be left enabled as it allows the system to provide dial tone to users (essential for MSN working).
- **Local Busy Tone:** *Default = Off*
  This setting should only be used when the local exchange gives a busy signal via Q.931 but does not provide busy tone.
- **Conferencing Tone:** *Default = Off*
  This settings controls how conference tones are used. When off, a single tone is heard when a new party joins a conference and double-tone is heard when a party leaves a conference. When on, a conference tone is heard every 10 seconds by all conference parties.
- **Inhibit Off-Switch Forward/Transfer:** *Default = Off (Italy = On)*
  When enabled, this setting stops any user from transferring or forwarding calls externally. See Off-Switch Forwarding and Transfer Controls.
- **Allow Outgoing Transfer:** *Default = Off, Software level = 3.0 to 3.2.*
  When not enabled, users are only able to transfer or forward off-switch incoming external calls. When enabled, users can forward both incoming and outgoing external calls. For pre-3.0 IP Office system the default behaviour is to bar outgoing transfers. For IP Office 4.0 and higher the default behaviour is to always allow outgoing transfers.
- **Dial By Name:** *Default = On*
  When on, allows the directory features on various phones to match the dialing of full names. When off, the directory features use the pre-IP Office 1.4 method of first character match only. See **Dial by Name in Appendix A: Configuration Examples**.
- **GSM Silence Suppression:** *Default = Off, Software level = 3.0+.*
  This setting should only be selected if voice quality problems are experienced with calls to voicemail or while recording calls. When on, the IP Office signals silence by generating silence data packets in periods when the voicemail system is not playing prompts. Note that use of this option may cause some timeout routing options in voicemail to no longer work.
- **Show Account Code:** *Default = On*
  This setting controls the display and listing of system account codes:
  - When on
    - When entering account codes through Phone Manager, users can select from a drop-down list of available account codes.
    - When entering account codes through a phone, the account code digits are shown while being dialed.
  - When off
    - Within Phone Manager the drop-down list of available account codes is not useable, instead account codes must be entered using the Phone Managers PIN Code features.
    - When entering account codes through a phone, the account code digits are replaced by **s** characters on the display.

- **Auto Hold**: *Default = On, Software level = 3.0+.*
  Used for users with multiple appearance buttons. When on, if a user presses another appearance button during a call, their current call is placed on hold. When off, if a users presses another appearance button during a call, their current call is disconnected.
- **Use External Music on Hold:** *Default = Off, Software level = 3.1+.*
  The presence of an internal music on hold file overrides the use of any external music on hold source. When this option is selected, following a reboot the IP Office will not attempt to reload an internal music on hold file by TFTP or from the compact flash memory card if available. The only source of music on hold used will be the external port. Note: For 3.1 this option was only supported in the Italy locale.
- **WAN Mode Override:** *Default = Off, Software level = 3.2/4.0 Q2 2007 Maintenance Release.*
  Alters the configuration of the WAN interface from the default to that required for BT X25 link inter-working. Used with IP406 V2 systems only.
- **Companding LAW**
  These settings are used to select the method of audio compression used for external digital lines and internal digital extensions. Note that U-LAW is also called MU-LAW. Most IP Office IP400 control units are available in A-Law and U-Law models. Typically U-Law models are supplied to North American locales, A-Law models are supplied to the rest of the world. IP500 control units are set to default A-Law or U-Law by the licence key inserted into the unit. In addition to the using different companding, A-Law and U-Law models support different default short codes.
    - Some digital phones only support a particular companding method. T3 phones only support A-Law switch mode. 4400 series phones only support U-Law switch mode.
- **Busy Tone Detection:** *Default = System Frequency (Tone defined by system locale)*
  Allows configuration of the IP Office's busy tone detection settings. These are on lines that do not provide reliable disconnect signalling. In that case the IP Office will uses tone disconnect clearing to disconnect such lines after 6 seconds of continuous tone. The default tone  (frequency and on/off cadence) detection used is defined by the system locale. The settings should not be adjusted unless advised by Avaya Technical Support.
- **Default Currency:** *Software level = 4.0+.*
  This setting is used with ISDN Advice of Charge (AOC) services. Note that changing the currency clears all call costs stored by the IP Office except those already logged through Delta Server. The currency is displayed in Phone Manager Pro and included in the IP Office SMDR output.
- **Disconnect Tone:** *Default = Default (Use locale setting); Software level = 4.0+.*
  For digital and IP phones, when the IP Office detects that the far end of a call has disconnected it can either make the near end go idle or play disconnect tone. By default this behaviour depends on the system locale. The **Disconnect Tone** field on the **System | Telephony** tab can be used to override the locale default and force either disconnect tone use or go idle.
    - ***Default:*** Use the system locale specific action for disconnected calls.
    - ***On:*** Play disconnect tone when far end disconnection is detected.
    - ***Off:*** Go idle when far end disconnection is detected.

# System | H.323 Gatekeeper

For IP Office 4.0+ systems these settings have been moved to the **System | LAN1** and if appropriate **System | LAN2** tabs.

H.323 VoIP phones must register with a gatekeeper in order to send and receive H.323 calls. The gatekeeper then controls permission for the phone to make or accept calls.

IP Office can act as the gatekeeper for H.323 phones. For full details relating to Avaya H.323 phones refer to the IP Office IP Phone Installation Manual. For non-Avaya H.323 devices, entry of an ***IP End-Points*** license is required.

IP Office H.323 Gatekeeper (Call Server) operation is supported only on the IP Office LAN1 IP address.

| System \| H.323 Gatekeeper | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✗. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✗, **4.1** ✗. |
| **Mergeable** | ✗. |

- **Gatekeeper Enable:** *Default = On*
  This settings enables IP Office gatekeeper operation.

- **Auto-create Extn:** *Default = On*
  When this option is on, an extension entry is automatically created for H.323 phones registering themselves with the IP Office as their gatekeeper.

- **RTP Port Number Range:** *Software level = 3.0+.*
  For each VoIP call, a receive port for incoming Real Time Protocol (RTP) traffic is selected from a defined range of possible ports, using the even numbers in that range. The Real Time Control Protocol (RTCP) traffic for the same call uses the RTP port number plus 1, that is the odd numbers. For IP Office control units and Avaya H.323 IP phones, the default port range used is 49152 to 53246. On some installations, it may be a requirement to change or restrict the port range used. It is recommended that only port numbers between 49152 and 65535 are used, that being the range defined by the Internet Assigned Numbers Authority (IANA) for dynamic usage.

  - **Port Range (minimum):** *Default = 49152. Range = 1024 to 64510.*
    This sets the lower limit for the RTP port numbers used by the IP Office.

  - **Port Range (maximum):** *Default = 53246. Range = 2048 to 65534.*
    This sets the upper limit for the RTP port numbers used by the IP Office. The gap between the minimum and the maximum must be at least 1024.

- **DiffServe Settings**
  When transporting voice over low speed links it is possible for normal data packets (1500 byte packets) to prevent or delay voice packets (typically 67 or 31 bytes) from getting across the link. This can cause unacceptable speech quality. Therefore it is important that all traffic routers and switches in a network to have some form of Quality of Service mechanism (QoS). QoS routers are essential to ensure low speech latency and to maintain sufficient audible quality.

  IP Office supports the DiffServ (RFC2474) QoS mechanism. This uses a Type of Service (ToS) field in the IP packet header. The IP Office uses this field to prioritize voice and voice signaling packets on its WAN interfaces. Note that the IP Office does not perform QoS for its Ethernet ports including the WAN Ethernet port on the Small Office Edition.

The hex and decimal entry fields for the following values are linked, the hex value being equal to the decimal multiplied by 4.

- **DSCP (Hex):** *Default = B8 (Hex)/46 (decimal), Range = 00 to FC (Hex)/0 to 63 (decimal)*
  The DiffServ Code Point (DSCP) setting applied to VoIP calls. For correct operation, especially over WAN links, the same value should be set at both ends.
- **DSCP Mask (Hex):** *Default = FC (Hex)/63 (decimal), Range = 00 to FC (Hex)/0 to 63 (decimal)*
  Allows a mask to be applied to packets for the DSCP value.
- **SIG DSCP (Hex):** *Default = 88 (Hex)/34 (decimal), Range = 00 to FC (Hex)/0 to 63 (decimal)*
  This setting is used to prioritize VoIP call signaling.

- **Site Specific Option Number (SSON):** *Default = 176, Range 128 to 254.*
  Sets the site specific option number (SSON) used by the IP Office's internal DHCP server. This should match the SSON used by 4600 and 5600 Series IP phones to request installation settings.

# System | LDAP

LDAP (Lightweight Directory Access Protocol) is a software protocol for enabling anyone to locate organizations, individuals, and other resources such as files and devices in a network, whether on the Internet or on a corporate intranet. LDAP is a "lightweight" (smaller amount of code) version of DAP (Directory Access Protocol), which is part of X.500, a standard for directory services in a network. LDAP is lighter because in its initial version, it did not include security features.

In a network, a directory tells you where in the network something is located. On TCP/IP networks, including the Internet, the Domain Name System (DNS) is the directory system used to relate the domain name to a specific network address. However, you may not know the domain name. LDAP allows you to search for an individual without knowing where they're located (although additional information will help with the search).

An LDAP directory is organized in a simple "tree" hierarchy consisting of the following levels:

- The "root" directory (the starting place or the source of the tree), which branches out to
- Countries, each of which branches out to
- Organizations, which branch out to
- Organizational units (divisions, departments, and so forth), which branches out to (includes an entry for)
- Individuals (which includes people, files, and shared resources such as printers)

An LDAP directory can be distributed among many servers. Each server can have a replicated version of the total directory that is synchronized periodically. An LDAP server is called a Directory System Agent (DSA). An LDAP server that receives a request from a user takes responsibility for the request, passing it to other DSA's as necessary, but ensuring a single coordinated response for the user.

LDAP Directory Synchronization allows the telephone number Directory held in the Control Unit to be synchronized with the information on an LDAP server. Although targeted for interoperation with Windows 2000 Server Active Directory, the feature is sufficiently configurable to interoperate with any server that supports LDAP version 2 or higher.

Telephone numbers obtained via the LDAP mechanism are held dynamically in the Directory. Each record retrieved creates a Directory Entry for use with Phone Manager. Please note that the entries are not stored in the configuration and therefore will not be visible via Manager. A maximum of 500 records can be retrieved due to size restraints. Records with exactly the same data in the Name and Number fields will not be duplicated.

Up to 500 LDAP directory entries can be obtained and then displayed in the Phone Manager directory for IP Office users. They do not appear in the Manager directory.

| System \| LDAP | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

- **LDAP Enabled:** *Default = Off*
  This option turns LDAP support on or off.
- **User Name:** *Default = blank*
  Enter the user name to authenticate connection with the LDAP database. To determine the domain-name of a particular Windows 2000 user look on the "Account" tab of the user's properties under "Active Directory Users and Computers". Note that this means that the user name required is not necessarily the same as the name of the Active Directory entry. There should be a built-in account in Active Directory for anonymous Internet access, with prefix "IUSR_" and suffix server_name (whatever was chosen at the Windows 2000 installation). Thus, for example, the user name entered is this field might be: IUSR_CORPSERV@acme.com
- **Password:** *Default = blank*
  Enter the password to be used to authenticate connection with the LDAP database. Enter the password that has been configured under Active Directory for the above user. Alternatively an Active Directory object may be made available for anonymous read access. This is configured on the server as follows:
  - In "Active Directory Users and Computers" enable "Advanced Features" under the "View" menu. Open the properties of the object to be published and select the "Security" tab. Click "Add" and select "ANONYMOUS LOGON", click "Add", click , click "Advanced" and select "ANONYMOUS LOGON", click "View/Edit", change "Apply onto" to "This object and all child objects", click , , .
    Once this has been done on the server, any entry can be made in the User Name field in the System configuration form (however this field cannot be left blank) and the Password field left blank. Other non-Active Directory LDAP servers may allow totally anonymous access, in which case neither User Name nor Password need be configured.
- **Server IP Address:** *Default = blank*
  Enter the IP address of the server storing the database.
- **Server Port:** Default = 389
  This setting is used to indicate the listening port on the LDAP server.
- **Authentication Method:** *Default = Simple*
  Select the authentication method to be used.
  - **Simple:** clear text authentication
  - **Kerberos:** Kerberos 4 LDAP and Kerberos 4 DSA encrypted authentication (for future use).
- **Resync Interval (secs):** *Default = 3600 seconds, Range = 1 to 99999 seconds.*
  The frequency at which the IP Office should resynchronize the directory with the server. This value also affects some aspects of the internal operation.
  - The LDAP search inquiry contains a field specifying a time limit for the search operation and this is set to 1/16th of the resync interval. So by default a server should terminate a search request if it has not completed within 225 seconds (3600/16).
  - The client end will terminate the LDAP operation if the TCP connection has been up for more than 1/8th of the resync interval (default 450 seconds). This time is also the interval at which a change in state of the "LDAP Enabled" configuration item is checked.

- **Search Base / Search Filter:** *Default = blank*
  These 2 fields are used together to refine the extraction of directory entries. Basically the Base specifies the point in the tree to start searching and the Filter specifies which objects under the base are of interest. The search base is a distinguished name in string form (as defined in RFC1779).
  The Filter deals with the attributes of the objects found under the Base and has its format defined in RFC2254 (except that extensible matching is not supported).
  If the Search Filter field is left blank the filter defaults to "(objectClass=*)", this will match all objects under the Search Base.

  The following are some examples applicable to an Active Directory database:

  - To get all the user phone numbers in a domain:
    **Search Base:** cn=users,dc=acme,dc=com
    **Search Filter:** (telephonenumber=*)
  - To restrict the search to a particular Organizational Unit (eg office) and get cell phone numbers also:
    **Search Base:** ou=holmdel,ou=nj,DC=acme,DC=com
    **Search Filter:** (|(telephonenumber=*)(mobile=*))
  - To get the members of distribution list "group1":
    **Search Base:** cn=users,dc=acme,dc=com
    **Search Filter:**
    (&(memberof=cn=group1,cn=users,dc=acme,dc=com)(telephonenumber=*))
- **Number Attributes:** *Default = see below*
  Enter the number attributes the server should return for each entry that matches the Search Base and Search Filter. Other entries could be ipPhone, otherIpPhone, facsimileTelephoneNumber, otherfacsimileTelephone Number, pager or otherPager. The attribute names are not case sensitive. Other LDAP servers may use different attributes.
  - By default the entry is "telephoneNumber,otherTelephone,homePhone=H,otherHomePhone=H,mobile=M,other Mobile=M", as used by Windows 2000 Server Active Directory for Contacts.
  - The optional "=string" sub-fields define how that type of number is tagged in the directory. Thus, for example, a cell phone number would appear in the directory as: John Birbeck M 7325551234

# System | System Events

The IP Office supports a number of methods by which events occurring on the system can be reported. These are in addition to the real-time and historical reports available through the IP Office System Status Application (SSA).

- **SNMP Reporting (IP Office 2.1+)**
  Simple Network Management Protocol (SNMP) allows SNMP clients and servers to exchange information. SNMP clients are built into devices such as network routers, server PC's, etc. SNMP servers are typically PC application which receive and/or request SNMP information. The IP Office SNMP client allows the IP Office system to respond to SNMP polling and to send alarm information to SNMP servers. In order for an SNMP server application to interact with an IP Office, the IP Office MIB files, provided on the IP Office Admin CD, must be compiled into the SNMP server's applications database. For full details refer to the IP Office Installation Manual.
- **SMTP Email Reporting (IP Office 3.2+)**
  IP Office 3.2 and higher can also send alarms to an SMTP email server. This allows users to receive alarms without needing to configure an SNMP server. Using SMTP requires details of a valid SMTP email account user name and password and server address. If SMTP email alarms are configured but for some reason the IP Office cannot connect with the SMTP server, only the last 10 alarms are stored for sending when connection is successful.
- **Syslog Reporting (IP Office 4.1+)**
  IP Office 4.1 and higher can send alarms to a Syslog server (RFC 3164) without needing to configure an SNMP server. In addition Syslog output can include IP Office audit trail events.

Multiple event destinations can be created, each specifying which events and alarms to include, the method of reporting to use (SNMP, Syslog or Email) and where to send the events. Up to 2 alarm destinations can be configured for SNMP, 2 for Syslog and 3 for SMTP email.

**Enabling SNMP System Alarms**

1. Select **System**.
2. Select the **System Events** tab.
3. Tick **SNMP Enabled**.
4. Complete the information in the SNMP Info section by entering the SNMP port and community details to match those expected by your SNMP server. Details of installing the MIB files required for SNMP are included in the IP Office Installation manual.
5. On the **Alarms** sub-tab, setup the required **Trap** alarm (see below).
6. Click **OK**.

**Enabling System Alarms**

1. Select  **System**.
2. Select the **System Events** tab.
    1. If planning to use SNMP, select **SNMP Enabled** and complete the information in the SNMP Info section by entering the SNMP port and community details to match those expected by your SNMP server. Details of installing the MIB files required for SNMP are included in the IP Office Installation manual.
    2. If planning to use SMTP, complete the information in the SMTP Server Configuration section. Enter the details of the SMTP email server and the email account.
    3. If planning to use Syslog, neither the SNMP or SMTP sections require entries.
3. Click **OK**.
4. Alarm destinations can now be configured through the **Alarms** sub-tab.

**Editing Alarm Destinations**

The Alarms section of the System Events tab displays the currently created alarm traps. It shows the event destinations and the types of alarms that will trigger the send of event reports. Up to 2 alarm destinations can be configured for SNMP, 2 for Syslog and 3 for SMTP email.

1. Select the **Alarms** sub-tab.
2. Use the **Add**, **Remove** and **Edit** controls to alter the traps.
3. Click **Add** or select the alarm to alter and then click **Edit**.
4. For a new alarm, set the **Destination** to either ***Trap*** (SNMP) or ***Syslog*** or ***Email*** (SMTP). Note that once a destination has been saved by clicking **OK** it cannot be changed to another sending mode.
5. The remaining details will indicate the required destination information and allow selection of the alarm events to include.
6. When completed, click **OK**.
7. Click **OK** again.

| System \| System Alarms \| SNMP Agent Configuration | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

- **SNMP Enabled:** *Default = Off*
  Enables support for SNMP. This option is not required if using SMTP or Syslog.
- **SNMP Port:** *Default = 161, Range = 0 to 65534.*
  The port on which the IP Office system listens for SNMP polling.
- **Community (Read-only):** *Default = public*
  The SNMP community name to which the IP Office belongs.
- **Device ID**
  This is a text field used to add additional information to alarms.
- **Contact**
  This is a text field used to add additional information to alarms.
- **Location**
  This is a text field used to add additional information to alarms.

| System \| System Alarms \| SMTP Server Configuration | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✗, **3.0DT** ✗, **3.0** ✗, **3.1** ✗, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗ (Syslog can be merged). |

- **IP Address:** *Default = 0.0.0.0*
  This field sets the IP address of the SMTP server being used to forward SNMP alarms sent by email.
- **Port:** *Default = 25, Range = 0 to 65534.*
  This field set the destination port on the SMTP server.
- **Email From Address:** *Default = Blank*
  This field set the sender address to be used with mailed alarms. Depending of the authentication requirements of the SMTP server this may need to be a valid email address hosted by that server. Otherwise the SMTP email server may need to be configured to support SMTP relay.
- **Server Requires Authentication:** *Default = On*
  This field should be selected if the SMTP server being used requires authentication to allow the sending of emails. When selected, the **User Name** and **Password** fields become available.
  - **User Name:** *Default = Blank*
    This field sets the user name to be used for SMTP server authentication.
  - **Password:** *Default = Blank*
    This field sets the password to be used for SMTP server authentication.
  - **Use Challenge Response Authentication (CRAM-MD5):** *Default = Off.*
    This field should be selected if the SMTP uses CRAM-MD5.

**Alarms Sub-Tab Settings**

- **Events**
  This section is used to show and edit the alarm. Up to 2 **Trap** alarms can be configured for the sending of alarms to an SNMP server. Up to 3 **Email** alarms can be configured for the sending of SMTP email messages.
  - **Destination**
    The options are **Trap** (SNMP), **Syslog** or **Email** (SMTP). To use SNMP or Email the appropriate settings must be configured on the Configuration sub-tab. Note that the Destination type will also be grayed out if the maximum number of configurable alarms destinations of that type has been reached. Up to 2 alarm destinations can be configured for SNMP, 2 for Syslog and 3 for SMTP email.
    - **Trap:**
      If selected, the details required in addition to the selected **Events** are:
      - **IP Address:** *Default = 0.0.0.0*
        The IP address of the SNMP server to which trap information is sent.
      - **Port:** *Default = 162, Range = 0 to 65534.*
        The SNMP transmit port
      - **Community:** *Default = Blank*
        The SNMP community for the transmitted traps. Must be matched by the receiving SNMP server.
    - **Email:** *Software level = 3.2+.*
      If selected, the details required in addition to the selected **Events** are:
      - **Email:**
        The destination email address.
    - **Syslog:** *Software level = 4.1+.*
      If selected, the details required in addition to the selected **Events** are:
      - **IP Address:** *Default = 0.0.0.0*
        The IP address of the Syslog server to which trap information is sent.
      - **Port:** *Default = 516, Range = 0 to 65534.*
        The Syslog destination port.
  - **Events:** *Default = None*
    Sets which types of IP Office events should be collected and sent. The table below lists the alarms associated with each type of event. Text in *italics* in the messages is replaced with the appropriate data. The subject line of SMTP email alarms takes the form "*System name*: *IP address* - System Alarm".

| Type | Events | Event State | Message |
|---|---|---|---|
| **Entity** | **Application** | Voicemail operation | The Voicemail server is now operational. |
| | | Voicemail Failure | The Voicemail server is down. |
| | | Voicemail Event - storage OK | The Voicemail server storage is OK. |
| | | Voicemail Event - storage nearly full | The Voicemail server storage is nearly full. |
| | | Voicemail Event - storage full | The Voicemail server storage is full. |
| | | Delta Server Operational | The Delta server is now operational. |
| | | Delta Server Failure | The Delta server is down. |

| Type | Events | Event State | Message |
|---|---|---|---|
| | **Compact Flash Card** | Change | The PC card in *name* has changed. |
| | **Expansion Module** | Operational | Expansion module *name* link is up. |
| | | Failure | Expansion module *name* link is down. |
| | | Error | Expansion module *name* link has a link error. |
| | | Change | Expansion module *name* link has changed. |
| | **Trunk** | Operational | Trunk *n* (*name*) [on expansion module *n*] is now operational. |
| | | Failure | Trunk *n* (*name*) [on expansion module *n*] is down. |
| | **VCM** | Operational | VCM module *name* is now operational. |
| | | Failure | VCM module *name* has failed. |
| **Generic** | **Generic** | Network link failure | Network Interface *name* (*ip address*) has been disconnected. |
| | | Network link operational | Network Interface *name* (*ip address*) has been connected. |
| | | System warm start | System has been restarted (warm start). |
| | | System cold start | System has restarted from power fail (cold start). |
| | | SNMP Invalid community | Invalid community specified in SNMP request. |
| **Licence** | **Licence** | Server operational | The license server is now operational. |
| | | Server failure | The license server is no longer operational. |
| **Loopback** | **Loopback** | Near end line loopback | Trunk *n* (*name*) [on expansion module *n*] is in near end loopback. |
| | | Near end payload loopback | Trunk *n* (*name*) [on expansion module *n*] is in near end loopback with payload. |
| | | Loopback off | Trunk *n* (*name*) [on expansion module *n*] has no loopback. |
| **Phone Change** | **Phone Change** | Phone has been unplugged | The phone with id *n* has been removed from extension *extn* (*unit*, port *n*). |
| | | Phone has been plugged in | The phone with type *type* (id *n*) has been plugged in for extension *extn* (*unit*, port *n*). |
| **Audit Trail** | **Basic Audit** | Events as written to the IP Office Audit Trail. | |

**Notes:**

- **Voicemail Pro Storage Alarms**
  The alarm threshold is adjustable through the Voicemail Pro client.
- **Embedded Voicemail Storage Alarm**s
  A disk full alarm is generated when the embedded voicemail memory card reaches 90% full. In addition a critical space alarm is generated at 99% full (98% for the Small Office) and an OK alarm is generated when the disk space returns to below 90% full.
- **Loopback**
  This type of alarm is only available for systems with a United States locale.

# System | Twinning

These settings are used with Mobile Twinning. See the **User | Twinning** tab for further details. The use of mobile twinning requires entry of a Mobile Twinning license.

| System \| Twinning | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✗, **3.0DT** ✗, **3.0** ✗, **3.1** ✗, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Send Original Party Information for Mobile Twinning:** *Default = On*
  When on, the IP Office will attempt to send the ICLID information provided with the incoming call to the twinning destination. Depending on the services provided by the line provider, this information may not be allowed in which case it may either be removed or the twinned call blocked. If this occurs, the **Calling Party Information for Mobile Twinning** field should be used to send information that is acceptable to the line provider.
- **Calling Party Information for Mobile Twinning:** *Default = Blank (Disabled)*
  This field is useable when **Send Original Part Information for Mobile Twinning** is off.

# System | CDR

Using a specified IP address, the IP Office can send a Call Detail Record (CDR) for each completed external call. A number of different CDR formats can be selected to match the requirements of the call logging/accounting software being used at the destination address.

For further details refer to **Appendix B: CDR Records**.

| System \| CDR | |
|---|---|
| **Control Unit** | **SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓.** |
| **Software Level** | **2.1 ✗, 3.0DT ✗, 3.0 ✗, 3.1 ✓, 3.2 ✓, 4.0 ✓, 4.1 ✓.** |
| **Mergeable** | **Pre-3.2 ✗, 3.2+ ✓.** |

- **Enable CDRs:** *Default = Off.*
  Enables the use of IP Office CDR.
- **Enable intra-switch CDRs:** *Default = Off.*
  When on, includes CDR records for internal calls.
- **Formatting Options**
  These fields are used to select the format and type of CDR records required. They must match the records expected by the call logging application receiving the CDR records.
  - **Record Format:** *Default = Unformatted.*
    Allows selection from a number of common CDR record formats.
  - **Record Options:** *Default = Enhanced.*
    Sets the options to include in the CDR record.
  - **Date Format:** *Default = Day\Month.*
    Sets the date format used in the CDR records.
- **Call Detail Recorder Communications**
  - **IP Address:** *Default = 0.0.0.0.*
    The destination IP address for CDR records.
  - **IP Port:** *Default = 0.*
    The destination IP port for CDR records.
  - **Max CDRs:** *Default = 500. Range = 0 to 1500.*
    The IP Office can cache up to 1500 CDR records if it detects a communications failure with destination address. If the cache is full, the IP Office will begin discarding the oldest records for each new record.
  - **Use UDP:** *Default = Off (Use TCP)*
    When selected, this field switches the sending of CDR record packets to use UDP instead of TCP.
    - If off, TCP is used. In this mode the IP Office will resend missed or corrupted records using the standard TCP protocol. Records are buffered until successfully sent.
    - If on, UDP is used. In this mode the IP Office will not resend missed or corrupt records. Also when using UDP, the IP Office is less likely to detect a communications failure which would triggered record caching.

# System | VCM

This form allows adjustment of the echo control applied by IP400 VCM cards. It does not apply to IP500 VCM cards.

Echoes are typically generated by impedance mismatches when a signal is converted from one type of circuit to another. To resolve this issue, an estimated echo signal can be created from one output and then subtracted from the input to hopefully remove any echo of the output.

| System \| VCM | |
|---|---|
| **Control Unit** | **SOE** ✗, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✗, **3.0DT** ✗, **3.0** ✗, **3.1** ✗, **3.2** ✓*, **4.0** ✓*, **4.1** ✓. |
| **Mergeable** | ✓. |

*IP Office 3.2/4.0 Q2 2007 Maintenance Releases.

- **Echo Return Loss (dB):** *Default = 6dB*
  Allows adjustment of expected echo loss that should be used for the echo cancellation process. The options are ***0dB***, ***3dB***, ***6dB*** and ***9dB***.
- **Nonlinear Processor Mode:** *Default = Adaptive*
  Allows selection of the echo cancellation algorithm between ***Adaptive***, ***Silence*** (attempt to mute background noise caused by echo cancellation) and ***Disabled***.

# Line Settings

## Line Form Overview

The line settings shown in the IP Office configuration will change according to the types of trunk cards installed in the IP Office control unit or added using external expansion modules. Some models of IP Office Small Office Edition control units include up to 4 integral analog trunk ports.

- **WARNING: Changing Trunk Cards**
  Changing the trunk card installed in an IP Office control unit will result in line settings for both the previous trunk card and the currently installed trunk card. In order to change the trunk card type in a particular card slot, the configuration must be defaulted. This does not apply if replacing an existing card with one of a higher capacity or fitting a trunk card into a previously unused slot.

**General Line Operation**
The following are general principles for line operation with an IP Office system. Some particular line types may operate differently.

- **Routing Calls to/from Lines - Line Groups**
  Each trunk channel belongs to an **Incoming Line Group** and an **Outgoing Line Group**. These are used as follows:
  - **Incoming Call Routes - Routing Incoming Calls**
    Within the IP Office configuration, **Incoming Call Routes** are used to determine where calls should be presented. The route used is determined by matching information received with the call such as incoming number, ICLID, call type and the **Incoming Line Group** of the line on which the call arrived.
  - **Short Codes - Routing Outgoing Calls**
    Short codes are used in several areas of the IP Office configuration to match digits dialed. Short codes that result in a number to be dialed includes a Line Group setting. This can specify a matching **Outgoing Line Group** from which a line should be seized if available or for IP Office 4.0 and higher an ARS form.
    - **ARS (Alternate Route Selection)**
      Calls can be routed to ARS forms. These forms for scenarios where different lines are used at different times or where no line in the outgoing line group is available.
- **Clock Quality**
  Calls between systems using digital trunks (for example E1, E1R2, T1 PRI and BRI) require an common clock signal. The IP Office will try to obtain this clock signal from an exchange through one of its digital trunks. This is done by setting the **Clock Quality** setting of that **Line** to ***Network***. If there are multiple trunks to public exchanges, another trunk can be set as **Fallback** should the primary clock signal fail. Other trunks should be set as **Unsuitable**.
- **Removing Unused Trunks**
  In cases where a trunk is not connected, it is important to ensure that the trunk is set as being **Out of Service** within the configuration. Failure to do this may cause the IP Office to attempt to present outgoing calls to that trunk. Similarly, where the number of channels subscribed is less than those supportable by the trunk type, the unsubscribed channels should be disabled. On E1 and BRI trunks this can be done by setting the **Number of Channels** correctly. On all trunk types it can also be done by setting the **Direction** of the unsubscribed channels to ***Incoming***.

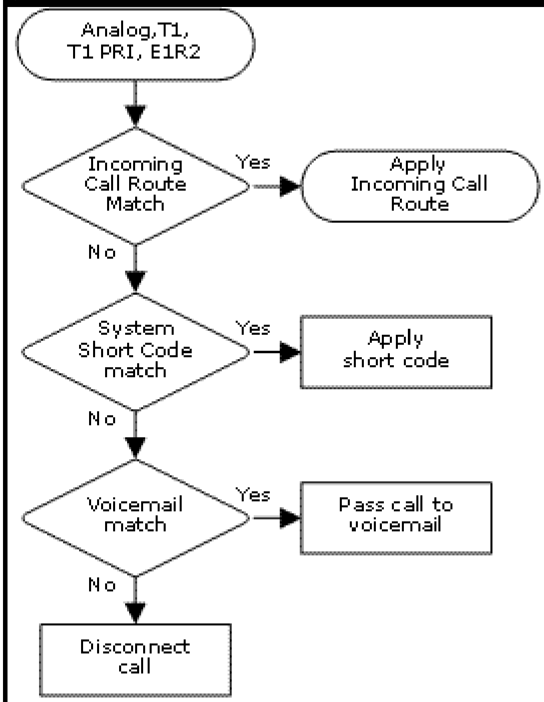
# Line (Analog)

## Analog Line Overview

Analog trunks can be provided within the IP Office systems in the following ways. In all cases the physical ports are labeled as **Analog**. For full details of installation refer to the IP Office Installation manual.

- **Using ICLID**
  The IP Office can route incoming calls using the ICLID received with the call. However ICLID is not sent instantaneously. On analog trunks set to Loop Start ICLID, there will be a short delay while the IP Office waits for any ICLID digits before it can determine where to present the call.
- **Line Status**
  Analog line do not indicate call status other than whether the line is free or in use. Some IP Office features, for example retrieving unanswered forwards and making twinned calls make use of the call status indicated by digital lines. This is not possible with analog lines. Once an analog line has been seized the IP Office has to assume that the call is connected and treats it as having been answered.
- **Dialing Complete**
  The majority of North-American telephony services use en-bloc dialing. Therefore the use of a **;** is recommended at the end of all dialing short codes that use an *N*. This is also recommended for all dialing where secondary dial tone short codes are being used.
- **Ground Start**
  This type of analog trunk is only supported through the Analog Trunk external expansion module.

**Incoming Call Routing: Analog, BRI, T1, T1 ISDN and E1R2 Trunks**




The following options are used to route incoming calls received on this type of trunk.

1. **Incoming Call Route**
   Based on matching the Incoming Line Group plus if set the incoming number and or ICLID.
2. **System Short Code**
   The IP Office checks for a short code match based on the incoming number.
3. **Voicemail Match**
   The IP Office checks for a call flow start point name matching the incoming number.

# Line | Line (Analog)

This tab covers general settings for a line being used with IP Office.

| Line \| Line (Analog) | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

- **Name:** *Default = Blank, Range up to 15 characters. Software level = 4.0 Q2 2007+.*
  This field is only visible on systems where the string ***SHOW_LINEID_NOT_OUTSIDE*** has been entered in the **Source Numbers** tab of the ***NoUser*** user. If a value is entered here, it is used in place of the word ***External*** on calls where no ICLID information is supplied. It does not replace the ***Withheld*** text. The name is also used as the default label for **Line Appearance** buttons set to this line.
- **Line Number**
  This parameter is not configurable, it is allocated by the system.
- **Card/Module:** *Software level = 4.1+.*
  Indicates the card slot being used for the device providing the line.
  - For IP400 control units, **SLOT A** on the control unit is shown as 1, **SLOT B** is shown as 2. Expansion modules are numbered from 4 upward, for example the module on **Expansion Port 1** is shown as 4.
  - For IP Office 500 control units, 1 to 4 match the slots on the front of the control unit from left to right. Expansion modules are numbered from 5 upwards, for example the module on **Expansion Port 1** is shown as 5.
- **Port:** *Software level = 4.1+.*
  Indicates the port on the **Card/Module** above to which the configuration entry relates.
- **Telephone Number**
  Used to remember the external telephone number of this line to assist with loop-back testing. For information only.
- **Incoming Group ID:** *Default = 0, Range 0 to 99999.*
  The **Incoming Group ID** to which a line belongs is used to match it to incoming call routes in the IP Office configuration. The matching incoming call route is then used to route incoming calls. The same ID can be used for multiple lines.
- **Outgoing Group ID:** *Default = 0, Range 0 to 99999.*
  Short codes that specify a number to dial also specify the line group to be used. The IP Office will then seize a line with a matching **Outgoing Group ID**. The same ID can be used for multiple lines.
- **Outgoing Channels:** *Default = 1 (not changeable)*
- **Voice Channels:** *Default = 1 (not changeable)*
- **Prefix:** *Default = Blank.*
  Enter the number to prefix to any ICLID received with incoming calls. If the IP Office has been configured, using short codes, to require users to dial a prefix to make external calls, adding the same prefix to incoming ICLID numbers allows those numbers to be used for return calls.
  - For outgoing calls: For IP Office 3.2 and lower, when a outgoing call is presented to the line with a leading digit to dial that matches the **Prefix**, that digit is stripped from the number. IP Office 4.0 and higher does not strip the prefix, therefore any prefixes not suitable for external line presentation should be stripped using short codes.
- **National Prefix:** *Default = 0 (not changeable)*
- **Line Appearance ID:** *Default = Auto-assigned, Range = 2 to 9 digits, Software level =* 3.0+.
  Allows a number to be assigned to the line to identify it. On phone's that support call appearance

buttons, a Line Appearance button with the same number will show the status of the line and can be used to answer calls on the line. For full details refer to the IP Office Key & Lamp Manual. The line appearance ID must be unique and not match any extension number.

# Line | Analog Options

This tab covers analog line specific settings.

| Line \| Analog Options | |
|---|---|
| **Control Unit** | **SOE ✓**, **IP403 ✓**, **IP406 V1 ✓**, **IP406 V2 ✓**, **IP412 ✓**, **IP500 ✓**. |
| **Software Level** | **2.1 ✓**, **3.0DT ✓**, **3.0 ✓**, **3.1 ✓**, **3.2 ✓**, **4.0 ✓**, **4.1 ✓**. |
| **Mergeable** | ✗. |

- **Channel:**
  Set by the system. Shown for information only.
- **Trunk Type:** *Default = Loop Start*
  Sets the analog line type (***Ground Start***, ***Loop Start***, ***Loop Start ICLID***, ***Out of Service***).
  - **Ground Start**
    Ground Start is only supported on trunks provided by the Analog Trunk 16 expansion module. It requires that the module and the IP Office control unit are grounded. Refer to the IP Office installation manual.
  - **Delay Waiting for Caller ID Information.**
    As the IP Office can use ICLID to route incoming calls, on analog Loop Start ICLID trunks there is a few seconds delay while ICLID is received before the call routing can be determined.
- **Signaling Type:** *Default = DTMF Dialing*
  Sets the signaling method used on the line (***DTMF Dialing*** or ***Pulse Dialing***).
- **Direction:** *Default = Both Directions*
  Sets the allowed direction of operation of the line (***Incoming***, ***Outgoing*** or ***Both Directions***).
- **Bearer:** *Default = Any*
  Sets the type of traffic carried by the line (***Voice***, ***Data*** or ***Any***).
- **Impedance:**
  This field is only available for certain system locales. The range of values available will depend on the system locale.
  - **Brazil:** *Default = 900R*
    Adjustable between 600R and 900R as required by the line provider.
  - **Korea:** *Default = Default, Software level = 3.2 and 4.0 Q2 2007+.*
    In addition to the default impedance settings, and alternate set of impedance values can be selected.
  - **United States:** *Default = Default, Software level = 3.2 and 4.0 Q2 2007+.*
    In addition to the default impedance settings, one of two alternate sets of impedance values can be selected.
- **Allow Analog Trunk to Trunk Connect:** *Default = Not selected (Off).*
  When not enabled, users cannot transfer or forward external calls back off-switch using an analog trunk if the calls was originally made or received on another analog trunk. This prevents transfers to trunks that do not support disconnect clear.
- **BCC:** *Default = Not selected [Brazil locale only]*

- **Secondary Dial Tone:** *Default = Off*
  Configures the use of secondary dial tone on analog lines. This is a different mechanism from secondary dial tone using short codes. This method is used mainly within the Russian locale. When selected, the following additional settings are accessible:
  - **Await time:** *Default = 3000ms, Range = 0 to 25500ms.*
    Used when secondary dial tone (above) is selected. Sets the delay.
  - **After n Digits:** *Default = 1, Range = 0 to 10.*
    Sets where in the dialing string, the delay for secondary dial tone, should occur.
  - **Matching Digit:** *Default =8, Range = 0 to 9.*
    The digit which, when first matched in the dialing string, will cause secondary dial tone delay.
- **Long CLI Line:** *Default = Off*
  The CLI signal on some long analog lines can become degraded and is not then correctly detected. If you are sure that CLI is being provided but not detected, selecting this option may resolve the problem.
- **Modem Enabled:** *Default = Off*
  The first analog trunk on Small Office Edition controls units, IP400 ATM4 trunk cards and IP500 Analog Trunk cards can be set to modem operation (V32 with V42 error correction). This allows the trunk to answer incoming modem calls and be used for system maintenance. When on, the trunk can only be used for analog modem calls. The default system short code ***9000*** can be used to toggle this setting. For the Small Office Edition control unit, when on, the control unit status LED flashes alternate red/green.
- **Ring Persistency:** *Default = Set according to system locale, Range = 0 to 2550ms.*
  The minimum duration of signal required to be recognized.
- **Ring Off Maximum:** *Default = Set according to system locale, Range = 0 to 25500ms.*
  The time required before signaling is regarded as ended.
- **Flash Pulse Width:** *Default = 500ms, Range = 0 to 2550ms.*
- **DTMF Mark:** *Default = 80 (80ms), Range = 0 to 255.*
- **DTMF Space:** *Default = 80 (80ms), Range = 0 to 255.*
- **Intermediate Digit Pause:** *Default = 500ms, Range = 0 to 2550ms.*
- **Voice Recording:** *Default = Low*
  Used to adjust the volume level of calls recorded by voicemail. Options are ***Low***, ***Medium*** and ***High***.
- **Disconnect Clear**
  Disconnect clear (also known as Line Break or Reliable Disconnect) is a method used to signal from the line provider that the call has cleared. The IP Office also uses Tone Disconnect, which clears an analog call after 6 seconds of continuous busy or NU tone, configured through the **Busy Tone Detection** settings on the **System | Telephony** tab.
  - **Enable:** *Default = On*
    Enables the use of disconnect clear.
  - **Units:** *Default = 500ms, Range = 0 to 2550ms.*
    This time must be less than the actual disconnect time period used by the line provider by at least 150ms.
- **Pulse Width On:** *Default = 40ms, Range = 0 to 255ms.*
- **Pulse Width Off:** *Default = 60ms, Range = 0 to 255ms.*
- **Await Dial Tone:** *Default = 3000ms, Range = 0 to 25500ms.*
  Sets how long the system should wait before dialing out.
- **BCC Flash Pulse Width:** *[Brazil locale only] Default = 100 (1000ms), Range = 0 to 255.*

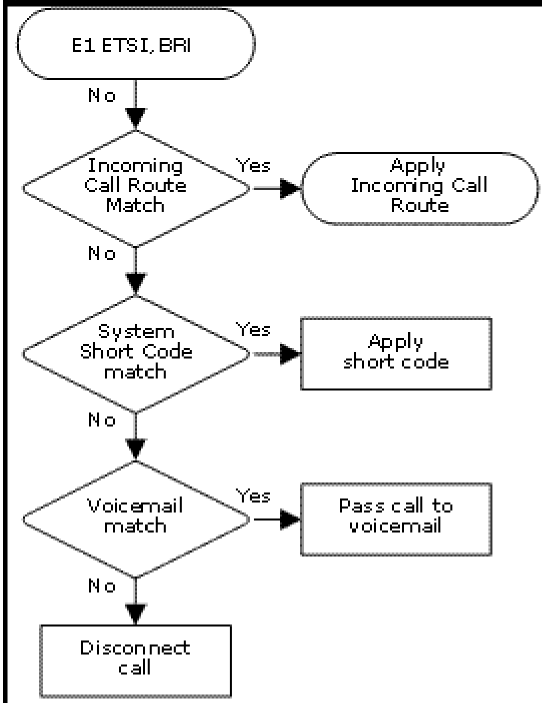
- **Gains:**
  - **Tx (A-D):** *Default = 0dB*
    Sets the analog to digital gain for incoming speech. Range -4.0dB to +3.5dB in 0.5dB steps.
  - **Rx (D-A):** Default = 0dB
    Sets the digital to analog gain for outgoing speech. Range -4.0dB to +3.5dB in 0.5dB steps.
- **Echo Cancellation:** *Default = 16ms.*
  Only useable with the IP400 ATM4 Universal and IP500 Analog trunk card. Allows settings of ***Off***, ***8***, ***16***, ***32***, ***64*** and ***128*** milliseconds. The echo cancellation should only be adjusted as high as required to remove echo problems. Setting it to a higher value than necessary can cause other distortions.

# Line (BRI)

## Line | BRI Overview

BRI trunks are provided by the installation of an Quad BRI trunk card into the IP Office control unit.  This card provides 4 BRI ports. Each port supports 2 channels for calls. For full details of installation refer to the IP Office Installation manual.

**Incoming Call Routing: E1 ETSI, BRI**




The following options are used to route incoming calls received on this type of trunk.

1. **Incoming Call Route**
   The IP Office checks for a match to the Incoming Line Group plus, if set, the incoming number and or ICLID.
2. **System Short Code**
   The IP Office checks for a short code match based on the incoming number.
3. **Voicemail Match**
   The IP Office checks for a call flow start point name matching the incoming number.

**Point-to-Point or Multipoint**
BRI lines can be used in either Point-to-Point or Point-to-Multipoint mode. Point-to-Point lines are used when only one device terminates a line in a customer's office. Point-to-Multipoint lines are used when more than one device may be used on the line at the customer's premises. There are major benefits in using Point-to-Point lines: -

1. The exchange knows when the line/terminal equipment is down/dead, thus it will not offer calls down that line. If the lines are Point-to-Multipoint, calls are always offered down the line and fail if there is no response from the terminal equipment. So if you have two Point-to-Multipoint lines and one is faulty 50% of incoming calls fail.
2. You get a green LED on the Control Unit when the line is connected. With Point-to-Multipoint lines some exchanges will drop layer 1/2 signals when the line is idle for a period.
3. The timing clock is locked to the exchange. If layer 1/2 signals disappear on a line then the Control Unit will switch to another line, however this may result in some audible click when the switchover occurs.

The IP Office's default Terminal Equipment Identifier (TEI) will normally allow it to work on Point-to-Point or Point-to-Multipoint lines. However if you intend to connect multiple devices simultaneously to an BRI line, then the TEI should be set to 127. With a TEI of 127, the IP Office control unit will ask the exchange to allocate a TEI for operation.

Note: When connected to some manufactures equipment, which provides an S0 interface (BRI), a defaulted Control Unit will not bring up the ISDN line. Configuring the Control Unit to a TEI of 127 for that line will usually resolve this.

# Line | BRI Line

**Settings**

| Line \| BRI Line | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

- **Name:** *Default = Blank, Range up to 15 characters. Software level = 4.0 Q2 2007+.*
  This field is only visible on systems where the string ***SHOW_LINEID_NOT_OUTSIDE*** has been entered in the **Source Numbers** tab of the ***NoUser*** user. If a value is entered here, it is used in place of the word ***External*** on calls where no ICLID information is supplied. It does not replace the ***Withheld*** text. The name is also used as the default label for **Line Appearance** buttons set to this line.

- **Card:** *Software level = 4.1+.*
  Indicates the card slot being used for the device providing the line.

  - For IP400 control units, **SLOT A** on the control unit is shown as 1, **SLOT B** is shown as 2. Expansion modules are numbered from 4 upward, for example the module on **Expansion Port 1** is shown as 4.

  - For IP Office 500 control units, 1 to 4 match the slots on the front of the control unit from left to right. Expansion modules are numbered from 5 upwards, for example the module on **Expansion Port 1** is shown as 5.

- **Port:** *Software level = 4.1+.*
  Indicates the port on the **Card** above to which the configuration entry relates.

- **Line Number**
  This parameter is not configurable; it is allocated by the system.

- **Line Sub Type:**
  Select to match the particular line type provided by the line provider. BRI supports ***ETSI***.

- **Telephone Number:**
  Used to remember the external telephone number of this line to assist with loop-back testing. For information only.

- **Prefix:** *Default = Blank.*
  The prefix is used in the following ways:

  - For incoming calls, the ISDN messaging tags indicates the call type (National, International or Unknown). If the call type is unknown, then the number in the **Prefix** field is added to the ICLID.

  - For outgoing calls: For IP Office 3.2 and lower, when a outgoing call is presented to the line with a leading digit to dial that matches the **Prefix**, that digit is stripped from the number. IP Office 4.0 and higher does not strip the prefix, therefore any prefixes not suitable for external line presentation should be stripped using short codes.

- **National Prefix:** *Default = 0*
  This indicates the digits to be prefixed to a incoming national call. When a number is presented from ISDN as a "national number" this prefix is added. For example 1923000000 is converted to 01923000000.

- **International Prefix:** *Default = 00*
  This indicates the digits to be prefixed to an incoming international call. When a number is presented from ISDN as an "international number" this prefix is added. For example 441923000000 is converted to 00441923000000.

- **TEI:** *Default = 0*
  The Terminal Equipment Identifier. Used to identify each device connected to a particular ISDN line. For Point-to-Point lines this is 0. It can also be 0 on a Point to Multipoint line, however if multiple devices are sharing a Point-to-Multipoint line it should be set to 127 which results in the exchange allocating the TEI's to be used.
- **Number of Channels:** *Default = 2. Range = 0 to 2.*
  Defines the number of operational channels that are available on this line.
- **Outgoing Channels:** *Default = 2. Range = 0 to 2.*
  This defines the number of channels available, on this line, for outgoing calls. This should normally be the same as Number of Channels field, but can be reduced to ensure incoming calls cannot be blocked by outgoing calls.
- **Voice Channels:** *Default = 2. Range = 0 to 2.*
  The number of channels available for voice use.
- **Data Channels:** *Default = 2. Range = 0 to 2.*
  The number of channels available for data use. If left blank, the value is 0.
- **Clock Quality:** *Default = Network*
  Sets whether the IP Office takes it clock source for call synchronization and signalling from this line. One line connected to the IP Office should be set to ***Network***. and wherever possible this should be a line connected to the central office exchange. Another line can be set to ***Fallback*** but all other lines should be set to ***Unsuitable***.
- **Supports Partial Rerouting:** *Default = Off, Software level* = 4.0+.
  Partial rerouting (PR) is an ISDN feature. It is supported on external (non-network and QSIG) ISDN exchange calls. When an external call is transferred to another external number, the transfer is performed by the ISDN exchange and the channels to the IP Office are freed. Use of this service may need to be requested from the line provider and may incur a charge.
- **Support Call Tracing:** *Default = Off, Software level* = 4.0+.
  IP Office 4.0+ supports the triggering of malicious caller ID (MCID) tracing at the ISDN exchange. Use of this feature requires liaison with the ISDN service provider and the appropriate legal authorities to whom the call trace will be passed. The user will also need to be enabled for call tracing and be provider with either a short code or programmable button to activate MCID call trace. Refer to Malicious Call Tracing in the **Telephone Features** section for full details.
- **Active CCBS Support:** *Default = Off, Software level* = 4.0+.
  Call completion to a busy subscriber (CCBS). It allows automatic callback to be used on outgoing ISDN calls when the destination is busy. This feature can only be used on point-to-point trunks. Use of this service may need to be requested from the line provider and may incur a charge.
- **Passive CCBS:** *Default = Off, Software level* = 4.0+.
- **Cost Per Charging Unit:** *Software level* = 4.0+.
  Advice of charge (AOC) information can be display on T3/T3IP phones and stored by the IP Office Delta Server application. The information is provided in the form of charge units. This setting is used to enter the call cost per charging unit set by the line provider. 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. Refer to Advice of Charge in the Telephone Features section.

# Line | Channels (BRI)

This tab allows settings for individual channels within the trunk to be adjusted.

| Line \| Channels (BRI) | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

- **Incoming Group ID:** *Default = 0, Range 0 to 99999.*
  The **Incoming Group ID** to which a line belongs is used to match it to incoming call routes in the IP Office configuration. The matching incoming call route is then used to route incoming calls. The same ID can be used for multiple lines.
  - Note: Though the **Incoming Group ID** is shown separately for each channel on the line, they cannot be adjusted separately. If the ID for one channel is changed the new ID is applied to all channels on that line. This also applies to the **Outgoing Group ID**.
- **Outgoing Group ID:** *Default = 0, Range 0 to 99999.*
  Short codes that specify a number to dial also specify the line group to be used. The IP Office will then seize a line with a matching **Outgoing Group ID**. The same ID can be used for multiple lines.
- **Channel Name:** *Default = Blank, Range up to 15 characters. Software level = 4.0 Q2 2007+.*
  This field is only visible on systems where the string ***SHOW_LINEID_NOT_OUTSIDE*** has been entered in the **Source Numbers** tab of the ***NoUser*** user. The value entered here is then used in place of the word  on call where on ICLID information is supplied.
- **Line Appearance ID:** *Default = Auto-assigned, Range = 2 to 9 digits, Software level* = 3.0+.
  Used for configuring Line Appearances with button programming. The line appearance ID must be unique and not match any extension number. Line appearance is not supported for trunks set to QSIG operation and is not recommended for trunks be used for DID.

# Line (E1)

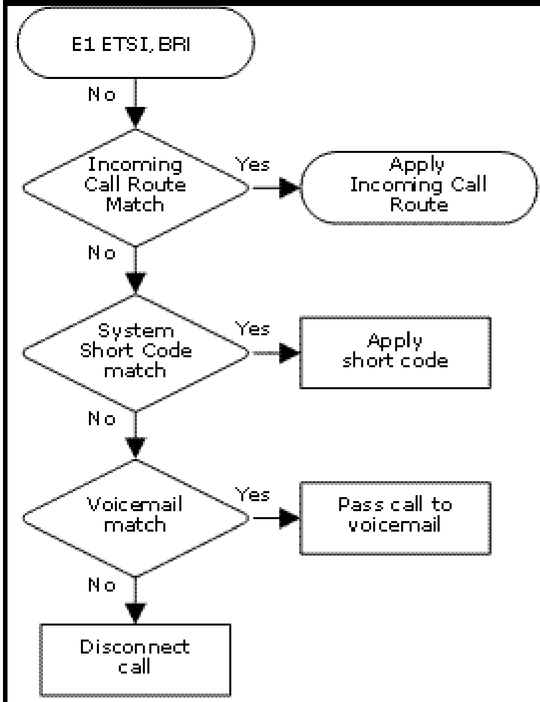
## Line Form (E1 PRI) Overview

E1 PRI trunks are provided by the installation of an E1 PRI trunk card into the IP Office control unit. E1 PRI trunk cards are not supported with the IP Office Small Office Edition control unit. Dual port E1 PRI trunk cards are only supported with the IP412 control unit and in Slot A of the IP406 V2 control unit. For full details of installation refer to the IP Office Installation manual.

Each physical E1 PRI trunk port supports up to 30 channels for calls.

E1 trunks can be set to either ETSI or QSIG operation modes. The selected mode affects how incoming calls are handled.

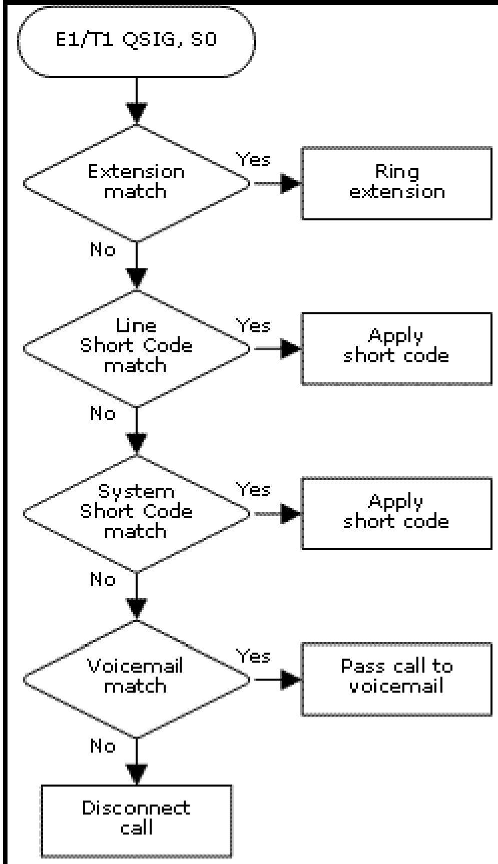
**Incoming Call Routing: E1 ETSI, BRI**




The following options are used to route incoming calls received on this type of trunk.

1. **Incoming Call Route**
   The IP Office checks for a match to the Incoming Line Group plus, if set, the incoming number and or ICLID.
2. **System Short Code**
   The IP Office checks for a short code match based on the incoming number.
3. **Voicemail Match**
   The IP Office checks for a call flow start point name matching the incoming number.

## Incoming Call Routing: E1 QSIG and S0 Lines




The following options are used to route incoming calls received on this type of trunk.

1. **Extension Match**
   The IP Office checks for an extension match based on the incoming number.
2. **Line Short Code Match**
   The IP Office checks for a short code match based on the incoming number.
3. **System Short Code**
   The IP Office checks for a short code match based on the incoming number.
4. **Voicemail Match**
   The IP Office checks for a call flow start point name matching the incoming number.

# Line | PRI Line (E1)

These settings are also used for a US T1 PRI trunk card set to ETSI or QSIG operation.

The IP500 PRI-U card supports E1, T1 and E1-R2 PRI modes. To select the mode required, right-click on the line in the group or navigation pane and select **Change Universal PRI Card Line Type**. The IP Office systems supports 8 B-channels for each IP500 PRI-U port fitted, using in-service channels from port 9 of slot 1 upwards. Additional B-channels up to the capacity of ports installed and PRI mode selected require **IP500 Universal PRI (Additional Channels)** licenses added to the configuration. D-channels are not affected by licensing.

- For ETSI and QSIG trunks, **IP500 Universal PRI (Additional Channels)** licenses are consumed by the number of calls in progress on B-channels.
- For T1 and E1R2 trunks, **IP500 Universal PRI (Additional Channels)** licenses are consumed by the channels in service.

| **Line \| PRI Line (E1)** | |
|---|---|
| **Control Unit** | **SOE** ✗, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

- **Name:** *Default = Blank, Range up to 15 characters. Software level = 4.0 Q2 2007+.*
  This field is only visible on systems where the string ***SHOW_LINEID_NOT_OUTSIDE*** has been entered in the **Source Numbers** tab of the ***NoUser*** user. If a value is entered here, it is used in place of the word ***External*** on calls where no ICLID information is supplied. It does not replace the ***Withheld*** text. The name is also used as the default label for **Line Appearance** buttons set to this line.
- **Card:** *Software level = 4.1+.*
  Indicates the card slot being used for the device providing the line.
  - For IP400 control units, **SLOT A** on the control unit is shown as 1, **SLOT B** is shown as 2. Expansion modules are numbered from 4 upward, for example the module on **Expansion Port 1** is shown as 4.
  - For IP Office 500 control units, 1 to 4 match the slots on the front of the control unit from left to right. Expansion modules are numbered from 5 upwards, for example the module on **Expansion Port 1** is shown as 5.
- **Port:** *Software level = 4.1+.*
  Indicates the port on the **Card** above to which the configuration entry relates.
- **Line Number**
  This parameter is not configurable; it is allocated by the system.
- **Line Sub Type:**
  Select to match the particular line type provided by the line provider. E1 PRI trunks support ***ETSI***, ***ETSI CHI***, ***QSIG A*** or ***QSIG B***.
  - ***ETSI CHI*** is used to send the channel allocation ID (CHI) in the call setup signalling.
  - ***QSIG*** is not supported on IP500 systems without **IP500 Voice Networking** licenses.
- **Telephone Number:**
  Used to remember the external telephone number of this line to assist with loop-back testing. For information only.
- **Incoming Group ID:** *Default = 0, Range 0 to 99999.*
  The **Incoming Group ID** to which a line belongs is used to match it to incoming call routes in the IP Office configuration. The matching incoming call route is then used to route incoming calls. The same ID can be used for multiple lines.

- **Outgoing Group ID:** *Default = 0, Range 0 to 99999.*
  Short codes that specify a number to dial also specify the line group to be used. The IP Office will then seize a line with a matching **Outgoing Group ID**. The same ID can be used for multiple lines.
- **Prefix:** *Default = Blank.*
  The prefix is used in the following ways:
  - For incoming calls, the ISDN messaging tags indicates the call type; National, International or Unknown. If the call type is unknown, then the number in the **Prefix** field is added to the ICLID.
  - For outgoing calls: For IP Office 3.2 and lower, when a outgoing call is presented to the line with a leading digit to dial that matches the **Prefix**, that digit is stripped from the number. IP Office 4.0 and higher does not strip the prefix, therefore any prefixes not suitable for external line presentation should be stripped using short codes.
- **National Prefix:** *Default = 0*
  This indicates the digits to be prefixed to a incoming national call. When a number is presented from ISDN as a "national number" this prefix is added. For example 1923000000 is converted to 01923000000.
- **International Prefix:** *Default = 00*
  This indicates the digits to be prefixed to an incoming international call. When a number is presented from ISDN as an "international number" this prefix is added. For example 441923000000 is converted to 00441923000000.
- **TEI:** *Default = 0*
  The Terminal Equipment Identifier. Used to identify each Control Unit connected to a particular ISDN line. For Point to Point lines this is typically (always) 0. It can also be 0 on a Point to Multi-Point line, however if multiple devices are sharing a Point to Multi-Point line it should be set to 127 which results in the exchange deciding on the TEI's to be used.
- **Number of Channels**
  Defines the number of operational channels that are available on this line. Up to 30 for E1 PRI, 23 for T1 PRI - depending upon the number of channels subscribed.
- **Outgoing Channels**
  This defines the number of channels available, on this line, for outgoing calls. This should normally be the same as Number of Channels field, but can be reduced to ensure incoming calls cannot be blocked by outgoing calls.
- **Voice Channels**
  The number of channels available for voice use.
- **Data Channels**
  The number of channels available for data use.
- **CRC Checking:** *Default = On*
  Switches CRC on or off.
- **Line Signalling:** *Default = CPE*
  This option is not used for lines where the **Line SubType** is set to ***QSIG***. Select either **CPE** (customer premises equipment) or **CO** (central office). The CO feature is intended to be used primarily as a testing aid. It allows PRI lines to be tested in a back-to-back configuration, using crossover cables.
  - The **CO** feature operates on this line type by modifying the way in which incoming calls are disconnected for IP Office configuration in Brazil and Argentina. In these locales, the CO setting uses **Forced-Release** instead of **Clear-Back** to disconnect incoming calls. The Brazilian **Double-Seizure** mechanism, used to police **Collect** calls, is also disabled in CO mode.

- **Clock Quality:** *Default = Network*
  For digital calls the IP Office requires a clock source for call signalling and synchronization. Preferably the clock source from a central office exchange should be used. ***Network*** indicates that this line can be used as the clock source. When multiple ***Network*** sources are available the IP Office will use the first available source. A line set to ***Fallback*** is only used if no line set to ***Network*** is available. Lines set to ***Unsuitable*** are not used.
- **Supports Partial Rerouting:** *Default = Off, Software level* = 4.0+.
  Partial rerouting (PR) is an ISDN feature. It is supported on external (non-network and QSIG) ISDN exchange calls. When an external call is transferred to another external number, the transfer is performed by the ISDN exchange and the channels to the IP Office are freed. Use of this service may need to be requested from the line provider and may incur a charge.
- **Support Call Tracing:** *Default = Off, Software level* = 4.0+.
  IP Office 4.0+ supports the triggering of malicious caller ID (MCID) tracing at the ISDN exchange. Use of this feature requires liaison with the ISDN service provider and the appropriate legal authorities to whom the call trace will be passed. The user will also need to be enabled for call tracing and be provider with either a short code or programmable button to activate MCID call trace. Refer to Malicious Call Tracing in the **Telephone Features** section for full details.
- **Active CCBS Support:** *Default = Off, Software level* = 4.0+.
  Call completion to a busy subscriber (CCBS). It allows automatic callback to be used on outgoing ISDN calls when the destination is busy. This feature can only be used on point-to-point trunks. Use of this service may need to be requested from the line provider and may incur a charge.
- **Passive CCBS:** *Default = Off, Software level* = 4.0+.
- **Cost Per Charging Unit***, Software level* = 4.0+.
  Advice of charge (AOC) information can be display on T3/T3IP phones and stored by the IP Office Delta Server application. The information is provided in the form of charge units. This setting is used to enter the call cost per charging unit set by the line provider. 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. Refer to Advice of Charge in the Telephone Features section.

The following fields are shown for a US T1 trunk card set to ETSI or QSIG operation. These cards have the same settings E1 PRI trunk cards set to ETSI or QSIG but only support 23 channels.

- **CSU Operation**
  Tick this field to enable the T1 line to respond to loop-back requests from the line.
- **Haul Length:** *Default = 0-115 feet*
  Sets the line length to a specific distance.
- **Channel Unit**: *Default = Foreign Exchange*
  This field should be set to match the channel signaling equipment provided by the Central Office. The options are ***Foreign Exchange***, ***Special Access*** or ***Normal***.

# Line | Short Codes

Line short codes can be applied to the digits received with incoming call. The stage at which they are applied also varies depending on the trunk type. See Line Short Codes in the **Short Codes** section.

- **IP Trunks**
  Line short codes are used if Small Community Networking (SCN) is not being used or no SCN user extension match occurs on the digits received.
- **QSIG and SO Trunks**
  S0 trunks, and E1 and T1 trunks set to QSIG mode, use line short code if the digits received do not match an internal extension number.

| **Line \| Short Codes** | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

Short codes can be added and edited using the **Add**, **Remove** and **Edit** buttons. Alternatively you can right-click on the list of existing short code to add and edit short codes.

# Line | Channels (E1 PRI)

This tab allows settings for individual channels within the trunk to be adjusted.

| Line \| Channels (E1 PRI) | |
|---|---|
| Control Unit | SOE ✗, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓. |
| Software Level | 2.1 ✓, 3.0DT ✓, 3.0 ✓, 3.1 ✓, 3.2 ✓, 4.0 ✓, 4.1 ✓. |
| Mergeable | ✗. |

- **Incoming Group ID:** *Default = 0, Range 0 to 99999.*
  The **Incoming Group ID** to which a line belongs is used to match it to incoming call routes in the IP Office configuration. The matching incoming call route is then used to route incoming calls. The same ID can be used for multiple lines.
  - Note: Though the **Incoming Group ID** is shown separately for each channel on the line, they cannot be adjusted separately. If the ID for one channel is changed the new ID is applied to all channels on that line. This also applies to the **Outgoing Group ID**.
- **Outgoing Group ID:** *Default = 0, Range 0 to 99999.*
  Short codes that specify a number to dial also specify the line group to be used. The IP Office will then seize a line with a matching **Outgoing Group ID**. The same ID can be used for multiple lines.
- **Channel Name:** *Default = Blank, Range up to 15 characters. Software level = 4.0 Q2 2007+.*
  This field is only visible on systems where the string ***SHOW_LINEID_NOT_OUTSIDE*** has been entered in the **Source Numbers** tab of the ***NoUser*** user. The value entered here is then used in place of the word  on call where on ICLID information is supplied.
- **Line Appearance ID:** *Default = Auto-assigned, Range = 2 to 9 digits, Software level* = 3.0+.
  Used for configuring Line Appearances with button programming. The line appearance ID must be unique and not match any extension number. Line appearance is not supported for trunks set to QSIG operation and is not recommended for trunks be used for DID.

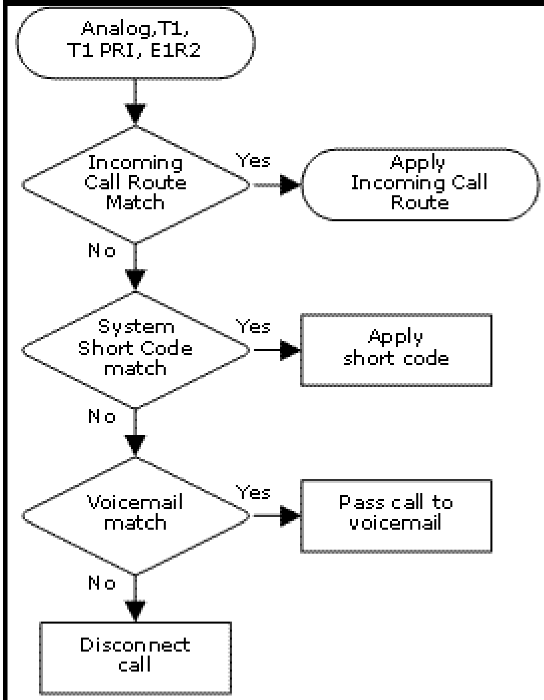
# Line (E1R2)

## Line Form (E1-R2) Overview

E1-R2 PRI trunks are provided by the installation of an E1-R2 PRI trunk card into the IP Office control unit.  E1-R2 PRI trunk cards are not supported with the IP Office Small Office Edition control unit. Dual port E1-R2 PRI trunk cards are only supported with the IP412 control unit and in Slot A of the IP406 V2 control unit. These trunk cards are available in RJ45 connector or coaxial connector versions. For full details of installation refer to the IP Office Installation manual.

Each physical E1 PRI trunk port supports up to 30 channels for calls.

**Incoming Call Routing: Analog, T1, T1 ISDN and E1R2 Trunks**




The following options are used to route incoming calls received on this type of trunk.

1. **Incoming Call Route**
   Based on matching the Incoming Line Group plus if set the incoming number and or ICLID.
2. **System Short Code**
   The IP Office checks for a short code match based on the incoming number.
3. **Voicemail Match**
   The IP Office checks for a call flow start point name matching the incoming number.

# Line | E1-R2 Options (Line)

The IP500 PRI-U card supports E1, T1 and E1-R2 PRI modes. To select the mode required, right-click on the line in the group or navigation pane and select **Change Universal PRI Card Line Type**. The IP Office systems supports 8 B-channels for each IP500 PRI-U port fitted. Additional B-channels up to the capacity of ports installed and PRI mode selected require **IP500 Universal PRI (Additional Channels)** licenses added to the configuration. D-channels are not affected by licensing.

- For ETSI and QSIG trunks, **IP500 Universal PRI (Additional Channels)** licenses are consumed by the number of calls in progress on B-channels.
- For T1 and E1R2 trunks, **IP500 Universal PRI (Additional Channels)** licenses are consumed by the channels in service.

| **Line \| Line (E1-R2)** | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

- **Name:** *Default = Blank, Range up to 15 characters. Software level = 4.0 Q2 2007+.*
  This field is only visible on systems where the string ***SHOW_LINEID_NOT_OUTSIDE*** has been entered in the **Source Numbers** tab of the ***NoUser*** user. If a value is entered here, it is used in place of the word ***External*** on calls where no ICLID information is supplied. It does not replace the ***Withheld*** text. The name is also used as the default label for **Line Appearance** buttons set to this line.
- **Card:** *Software level = 4.1+.*
  Indicates the card slot being used for the device providing the line.
  - For IP400 control units, **SLOT A** on the control unit is shown as 1, **SLOT B** is shown as 2. Expansion modules are numbered from 4 upward, for example the module on **Expansion Port 1** is shown as 4.
  - For IP Office 500 control units, 1 to 4 match the slots on the front of the control unit from left to right. Expansion modules are numbered from 5 upwards, for example the module on **Expansion Port 1** is shown as 5.
- **Port:** *Software level = 4.1+.*
  Indicates the port on the **Card** above to which the configuration entry relates.
- **Line Number:**
  Allocated by the system.
- **Line SubType:** *Default = E1-R2*
  Supported options are **E1-R2**, **ETSI**, **QSIGA** or **QSIGB**.
  - ***QSIG*** is not supported on IP500 systems without **IP500 Voice Networking** licenses.
- **Channel Allocation:** *Default = 30 -> 1*
  The order, 30 -> 1 or 1 -> 30, in which channels are used.
- **Country (Locale):** *Default = Mexico.*
  Select the locale that matches the area of usage. Note that changing the locale will return the MFC Group settings to the defaults for the selected locale. Currently supported locales ***Argentina***, ***Brazil***, ***China***, ***India***, ***Korea***, ***Mexico*** and ***None***.

The table at the base of the form displays the settings for the individual channels provided by the line. For details of the channel settings see Edit Channel (E1-R2).

To edit a channel, either double-click on it or right-click and select **Edit**. This will display the Edit Channel dialog box. To edit multiple channels at the same time select the channels whilst pressing the Shift or Ctrl key. Then right-click and select **Edit**.

# Line | Channels (E1-R2)

This tab allows settings for individual channels within the trunk to be adjusted. To edit a channel, select the required channel or channels and click **Edit**.

| Line \| Channels (E1-R2) | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

The channel settings are split into two sub-tabs, **E1R2 Edit Channel** and **Timers**.

### E1R2 Edit Channel Settings

- **Channel**
  The channel or channels being edited.
- **Incoming Group ID:** *Default = 0, Range 0 to 99999.*
  The Incoming Group ID to which a line belongs is used to match it to incoming call routes in the IP Office configuration. The matching incoming call route is then used to route incoming calls. The same ID can be used for multiple lines.
- **Outgoing Group ID:** *Default = 0, Range 0 to 99999.*
  Short codes that specify a number to dial also specify the line group to be used. The IP Office will then seize a line with a matching **Outgoing Group ID**. The same ID can be used for multiple lines.
- **Line Appearance ID:** *Default = Auto-assigned, Range = 2 to 9 digits, Software level* = 3.0+.
  Though line appearance settings are shown for E1R2 trunks, use of line appearances with E1R2 is not supported.
- **Direction:** *Default = Both Directions*
  The direction of calls on the channel (***Incoming***, ***Outgoing*** or ***Both Directions***).
- **Bearer:** *Default = Any*
  The type of traffic carried by the channel (***Voice***, ***Data*** or ***Any***).
- **Line Signaling Type:** *Default = R2 Loop Start*
  The signaling type used by the channel. Current supported options are: ***R2 Loop Start***, ***R2 DID***, ***R2 DOD***, ***R2 DIOD***, ***Tie Immediate Start***, ***Tie Wink Start***, ***Tie Delay Dial***, ***Tie Automatic***, ***WAN Service*** and ***Out of Service***.
- **Dial Type:** *Default = MFC Dialing*
  The type of dialing supported by the channel; ***MFC Dialing**, **Pulse Dialing** or **DTMF Dialing**.*

### Timers Settings

This sub-tab displays the various timers provided for E1-R2 channels. These should only be adjusted when required to match the line provider's settings.

# Line | MFC Group (E1-R2)

These tabs show the parameter assigned to each signal in an MFC group. The defaults are set according to the **Country (Locale)** on the **Line** tab. All the values can be returned to default by the **Default All** button on the **Advanced** tab.

To change a setting either double-click on it or right-click and select **Edit**.

| Line \| MFC Group (E1-R2) | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

# Line | Advanced (E1-R2)

| Line \| Advanced (E1-R2) | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

- **Zero Suppression:** *Default = HDB3*
  Selects the method of zero suppression used (***HDB3*** or ***AMI***).

- **Clock Quality:** *Default = Network*
  For digital calls the IP Office requires a clock source for call signalling and synchronization. Preferably the clock source from a central office exchange should be used. ***Network*** indicates that this line can be used as the clock source. When multiple ***Network*** sources are available the IP Office will use the first available source. A line set to ***Fallback*** is only used if no line set to ***Network*** is available. Lines set to ***Unsuitable*** are not used.

- **Pulse Metering Bit:** *Default = A Bit*
  Sets which bit should be used to indicate the pulse metering signal; **A Bit**, **B Bit** or **C Bit**.

- **Line Signaling:** *Default = CPE*
  Select either **CPE** or **CO**. The CO feature is intended to be used primarily as a testing aid. It allows T1 and E1 lines to be tested in a back-to-back configuration, using crossover (QSIG) cables.

  - The **CO** feature operates by modifying the way in which incoming calls are disconnected for IP Office configuration in Brazil and Argentina. In these locales, the CO setting uses **Forced-Release** instead of **Clear-Back** to disconnect incoming calls. The Brazilian **Double-Seizure** mechanism used to police **Collect** calls, is also disabled in CO mode.

- **Incoming Routing Digits:** *Default = 4*
  Sets the number of incoming digits used for incoming call routing.

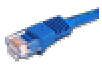
- **CRC Checking:** *Default = Ticked (On)*
  Switches CRC on or off.

- **Default All Group Settings**
  Default the MFC Group tab settings.

- **Line Signaling Timers:**
  To edit one of these timers, either double-click on the timer or right-click on a timer and select the action required.

# Line (T1)

## T1 Line Overview



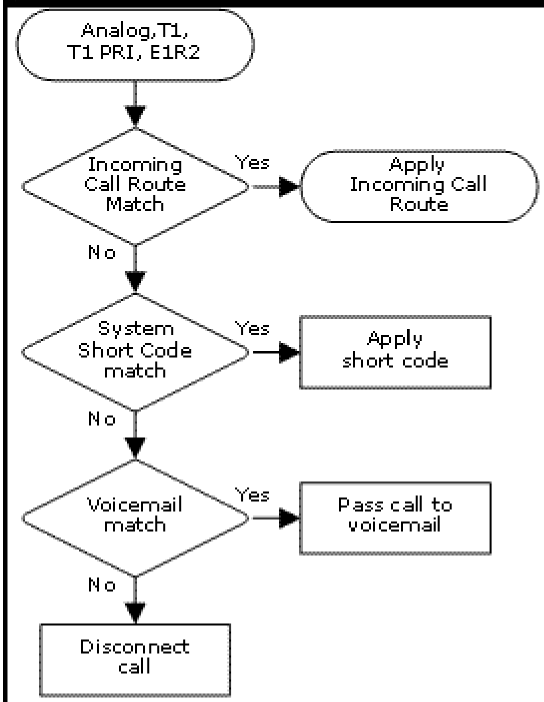
T1 trunks are provided by the installation of an T1 PRI trunk card into the IP Office control unit. The trunks on these cards can be configured for T1, PRI or QSIG operation. For full details of installation refer to the IP Office Installation manual.

Dual port T1 PRI trunk cards are only supported with the IP412 control unit and in Slot A of the IP406 V2 control unit.

Each physical trunk port supports up to 24 channels in T1 mode, 23 channels in PRI and QSIG modes.

- **Dialing Complete**
  The majority of North-American telephony services use en-bloc dialing. Therefore the use of a **;** is recommended at the end of all dialing short codes that use an *N*. This is also recommended for all dialing where secondary dial tone short codes are being used.

**Incoming Call Routing: Analog, T1, T1 ISDN and E1R2 Trunks**




The following options are used to route incoming calls received on this type of trunk.

1. **Incoming Call Route**
   Based on matching the Incoming Line Group plus if set the incoming number and or ICLID.
2. **System Short Code**
   The IP Office checks for a short code match based on the incoming number.
3. **Voicemail Match**
   The IP Office checks for a call flow start point name matching the incoming number.

# Line | Line (T1)

The IP500 PRI-U card supports E1, T1 and E1-R2 PRI modes. To select the mode required, right-click on the line in the group or navigation pane and select **Change Universal PRI Card Line Type**. The IP Office systems supports 8 B-channels for each IP500 PRI-U port fitted. Additional B-channels up to the capacity of ports installed and PRI mode selected require **IP500 Universal PRI (Additional Channels)** licenses added to the configuration. D-channels are not affected by licensing.

- For ETSI and QSIG trunks, **IP500 Universal PRI (Additional Channels)** licenses are consumed by the number of calls in progress on B-channels.
- For T1 and E1R2 trunks, **IP500 Universal PRI (Additional Channels)** licenses are consumed by the channels in service.

| **Line \| Line (T1)** | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

- **Name:** *Default = Blank, Range up to 15 characters. Software level = 4.0 Q2 2007+.*
  This field is only visible on systems where the string ***SHOW_LINEID_NOT_OUTSIDE*** has been entered in the **Source Numbers** tab of the ***NoUser*** user. If a value is entered here, it is used in place of the word ***External*** on calls where no ICLID information is supplied. It does not replace the ***Withheld*** text. The name is also used as the default label for **Line Appearance** buttons set to this line.
  - This name can be overridden by a separate name set for the individual trunk channel.
- **Card:** *Software level = 4.1+.*
  Indicates the card slot being used for the device providing the line.
  - For IP400 control units, **SLOT A** on the control unit is shown as 1, **SLOT B** is shown as 2. Expansion modules are numbered from 4 upward, for example the module on **Expansion Port 1** is shown as 4.
  - For IP Office 500 control units, 1 to 4 match the slots on the front of the control unit from left to right. Expansion modules are numbered from 5 upwards, for example the module on **Expansion Port 1** is shown as 5.
- **Port:** *Software level = 4.1+.*
  Indicates the port on the **Card** above to which the configuration entry relates.
- **Line Number:**
  Allocated by the system.
- **Line Sub Type:** *Default = T1*
  Set to ***T1*** for a T1 line. For ***PRI*** see **Line Form (US PRI)**. If set to **ETSI**, **QSIG A** or **QSIG B** see **Line (E1)**.
- **Channel Allocation:** *Default = 24 -> 1*
  The order, 24 to 1 or 1 to 24, in which channels are used.
- **Prefix:** *Default = Blank*
  Enter the number to prefix to all incoming numbers for callback. This is useful if all users must dial a prefix to access an outside line. The prefix is automatically placed in front of all incoming numbers so that users can dial the number back.
- **Framing:** *Default = ESF*
  Selects the type of signal framing used (***ESF*** or ***D4***).
- **Zero Suppression:** *Default = B8ZS*
  Selects the method of zero suppression used (***B8ZS*** or ***AMI ZCS***).

- **Clock Quality:** *Default = Network*
  For digital calls the IP Office requires a clock source for call signalling and synchronization. Preferably the clock source from a central office exchange should be used. ***Network*** indicates that this line can be used as the clock source. When multiple ***Network*** sources are available the IP Office will use the first available source. A line set to ***Fallback*** is only used if no line set to ***Network*** is available. Lines set to ***Unsuitable*** are not used.
- **Haul Length:** *Default = 0-115 feet*
  Sets the line length to a specific distance.
- **Channel Unit**: *Default = Foreign Exchange*
  This field should be set to match the channel signaling equipment provided by the Central Office. The options are ***Foreign Exchange***, ***Special Access*** or ***Normal***.
- **CRC Checking:** *Default = On*
  Turns CRC on or off.
- **Line Signaling:** *Default = CPE*
  This field affects T1 channels set to **Loop-Start** or **Ground-Start**. The field can be set to either ***CPE*** (Customer Premises Equipment) or ***CO*** (Central Office). This field should normally be left at its default of ***CPE***. The setting ***CO*** is normally only used in lab back-to-back testing.
- **Incoming Routing Digits:** *Default=0 (present call immediately)*
  Sets the number of routing digits expected on incoming calls. This allows the line to present the call to the system once the expected digits have been received rather than waiting for the digits timeout to expire. This field only affects T1 line channels set to ***E&M Tie***, ***E&M DID***, ***E&M Switched 56K*** and ***Direct Inward Dial***.
- **CSU Operation:**
  Tick this field to enable the T1 line to respond to loop-back requests from the line.
- **Enhanced Called Party Number:** *Default = Off*
  This option is not supported for systems set to the **United States** locale. Normally the dialed number length is limited to 15 digits. Selecting this option increases the allowed dialed number length to 30 digits.

# Line | Channels (T1)

The settings for each channel can be edited. Users have the option of editing individual channels by double-clicking on the channel or selecting and editing multiple channels at the same time. Note that the **Line Appearance ID** cannot be updated when editing multiple channels.

When editing a channel or channels, the settings available are displayed on two sub-tabs; **T1 Edit Channel** and **Timers**.

| **Line \| Channels (T1)** | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

**T1 Edit Channel Sub-Tab Settings**

- **Name:** *Default = Blank, Range up to 15 characters. Software level = 4.0 Q2 2007+.*
  This field is only visible on systems where the string ***SHOW_LINEID_NOT_OUTSIDE*** has been entered in the **Source Numbers** tab of the ***NoUser*** user. If a value is entered here, it is used in place of the word ***External*** on calls where no ICLID information is supplied. It does not replace the ***Withheld*** text. The name is also used as the default label for **Line Appearance** buttons set to this line.
  - This name overrides the name set for the line.
- **Channel**
  Allocated by the system.
- **Channel Name:** *Default = Blank, Range up to 15 characters. Software level = 4.0 Q2 2007+.*
  This field is only visible on systems where the string ***SHOW_LINEID_NOT_OUTSIDE*** has been entered in the **Source Numbers** tab of the ***NoUser*** user. The value entered here is then used in place of the word  on call where on ICLID information is supplied.
- **Incoming Group ID:** *Default = 0, Range 0 to 99999.*
  The Incoming Group ID to which a line belongs is used to match it to incoming call routes in the IP Office configuration. The matching incoming call route is then used to route incoming calls. The same ID can be used for multiple lines.
- **Outgoing Group ID:** *Default = 0, Range 0 to 99999.*
  Short codes that specify a number to dial also specify the line group to be used. The IP Office will then seize a line with a matching **Outgoing Group ID**. The same ID can be used for multiple lines.
- **Line Appearance ID:** *Default = Auto-assigned, Range = 2 to 9 digits, Software level* = 3.0+.
  Used for configuring Line Appearances with button programming. The line appearance ID must be unique and not match any extension number. Line appearance is not supported for trunks set to QSIG operation and is not recommended for trunks be used for DID.
- **Direction:** *Default = Bothway*
  The direction of calls on the channel (***Incoming***, ***Outgoing*** or ***Bothway***).
- **Bearer:** *Default = Voice*
  The type of traffic carried by the channel.
- **Type:** *Default = Ground-Start*
  The T1 emulates the following connections (***Ground-Start***, ***Loop-Start***, ***E&M - TIE***, ***E&M - DID***, ***E&M Switched 56K***, ***Direct Inward Dial, Clear Channel 64K*** or ***Out of Service***).  Trunks set to E&M DID will only accept incoming calls.
  - If **E&M-TIE** is selected and the **Outgoing Trunk Type** is set to **Automatic**, no secondary dial tone is provided for outgoing calls on this line/trunk.
- **Dial Type:** *Default = DTMF Dial*
  Select the dialing method required (***DTMF Dial*** or ***Pulse Dial***).

- **Incoming Trunk Type:** *Default = Wink-Start*
  Used for E&M types only. The handshake method for incoming calls (***Automatic***, ***Immediate***, ***Delay Dial*** or ***Wink-Start***).
- **Outgoing Trunk Type:** *Default = Wink-Start*
  Used for E&M types only. The handshake method for outgoing calls (***Automatic***, ***Immediate***, ***Delay Dial*** or ***Wink-Start***).
  - If the line **Type** is set to **E&M-TIE** and the **Outgoing Trunk Type** is set to **Automatic**, no secondary dial tone is provided for outgoing calls on this line/trunk.
- **Tx Gain:** *Default = 0dB*
  The transmit gain in dB.
- **Rx Gain:** *Default = 0dB*
  The receive gain in dB.

**Timers Sub-Tab Settings**
This sub-tab allows various timers relating to operation of an individual channel to be adjusted. These should only be adjusted to match the requirements of the line provider. The following is a list of the default values. To reset a value, click on the current value and then right click and select from the default, minimize and maximize options displayed.

- **Outgoing Seizure:** 10.
- **Wink Start:** 5000.
- **Wink Validated:** 80.
- **Wink End:** 350.
- **Delay End:** 5000.
- **Outgoing Dial Guard:** 590.
- **Outgoing IMM Dial Guard:** 1500.
- **Outgoing Pulse Dial Break:** 60.
- **Outgoing Pulse Dial Make:** 40.
- **Outgoing Pulse Dial Inter Digit:** 720.
- **Outgoing Pulse Dial Pause:** 1500.
- **Flash Hook Generation:** 500.
- **Outgoing End of Dial:** 1000.
- **Answer Supervision:** 300.
- **Incoming Confirm:** 20.
- **Incoming Automatic Delay:** 410.
- **Incoming Wink Delay:** 100.
- **Wink Signal:** 200.
- **Incoming Dial Guard:** 50.
- **First Incoming Digit:** 15000.
- **Incoming Inter Digit:** 5000.
- **Maximum Inter Digit:** 300.
- **Flash Hook Detect:** 240.
- **Incoming Disconnect:** 300.
- **Incoming Disconnect Guard:** 800.
- **Disconnected Signal Error:** 240000.
- **Outgoing Disconnect:** 300.
- **Outgoing Disconnect Guard:** 800.
- **Ring Verify Duration:** 220.
- **Ring Abandon:** 6300.
- **Ping Verify:** 600.
- **Long Ring Duration:** 1100.
- **Silent Interval:** 1100.

# Line | Short Codes

Line short codes can be applied to the digits received with incoming call. The stage at which they are applied also varies depending on the trunk type. See Line Short Codes in the **Short Codes** section.

- **IP Trunks**
  Line short codes are used if Small Community Networking (SCN) is not being used or no SCN user extension match occurs on the digits received.
- **QSIG and SO Trunks**
  S0 trunks, and E1 and T1 trunks set to QSIG mode, use line short code if the digits received do not match an internal extension number.

| Line \| Short Codes | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

Short codes can be added and edited using the **Add**, **Remove** and **Edit** buttons. Alternatively you can right-click on the list of existing short code to add and edit short codes.

# Line (T1 PRI)

## Line Form (T1 PRI) Overview

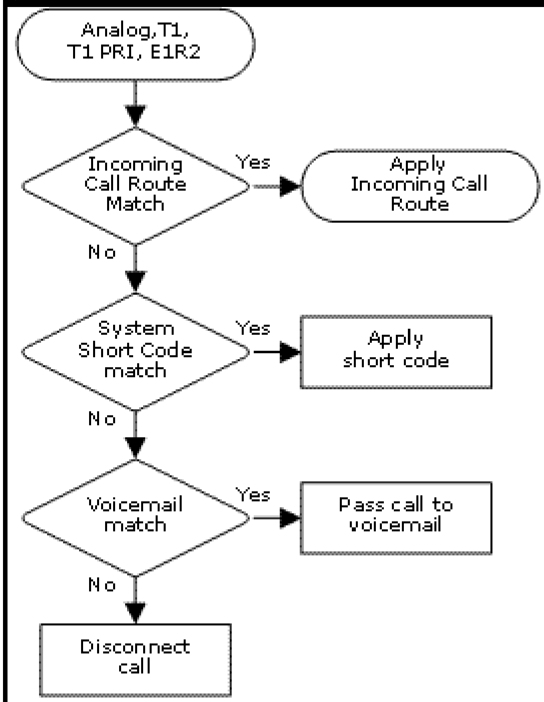
T1 PRI trunks are provided by the installation of an T1 PRI trunk card into the IP Office control unit. The trunks on these cards can be configured for T1 operation or T1 ISDN operation. For full details of installation refer to the IP Office Installation manual.

Dual port T1 PRI trunk cards are only supported with the IP412 control unit and in Slot A of the IP406 V2 control unit.

Each physical trunk port supports up to 24 channels in T1 mode, 23 channels in PRI and QSIG modes.

- **Dialing Complete**
  The majority of North-American telephony services use en-bloc dialing. Therefore the use of a **;** is recommended at the end of all dialing short codes that use an ***N***. This is also recommended for all dialing where secondary dial tone short codes are being used.
- **AT&T Provider Settings**
  For AT&T operation two information elements, TNS (Transit Network Selector) and NSF (Network Specific Facility), are sent in the call setup to the service provider. On IP Office, the values for TNS, NSF and the actual phone number presented to the line are determined by parsing the number dialed through, in sequence, the **TNS**, **Special** and **Call by Call** tabs. These tabs appear when the **Provider** setting on the **Line** tab is set to ***AT&T***. Note also that B-channels within the same line can be brought from different service provides. Additionally some B-channels can be used 'call by call', that is, use a different service provider for each call.

**Incoming Call Routing: Analog, T1, T1 ISDN and E1R2 Trunks**




The following options are used to route incoming calls received on this type of trunk.

1. **Incoming Call Route**
   Based on matching the Incoming Line Group plus if set the incoming number and or ICLID.
2. **System Short Code**
   The IP Office checks for a short code match based on the incoming number.
3. **Voicemail Match**
   The IP Office checks for a call flow start point name matching the incoming number.

# Line | Line (T1 PRI)

The IP500 PRI-U card supports E1, T1 and E1-R2 PRI modes. To select the mode required, right-click on the line in the group or navigation pane and select **Change Universal PRI Card Line Type**.

The IP Office systems supports 8 B-channels for each IP500 PRI-U port fitted. Additional B-channels up to the capacity of ports installed and PRI mode selected require **IP500 Universal PRI (Additional Channels)** licenses added to the configuration. D-channels are not affected by licensing.

- For ETSI and QSIG trunks, **IP500 Universal PRI (Additional Channels)** licenses are consumed by the number of calls in progress on B-channels.
- For T1 and E1R2 trunks, **IP500 Universal PRI (Additional Channels)** licenses are consumed by the channels in service.

| Line \| Line (T1 PRI) | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

- **Name:** *Default = Blank, Range up to 15 characters. Software level = 4.0 Q2 2007+.*
  This field is only visible on systems where the string ***SHOW_LINEID_NOT_OUTSIDE*** has been entered in the **Source Numbers** tab of the ***NoUser*** user. If a value is entered here, it is used in place of the word ***External*** on calls where no ICLID information is supplied. It does not replace the ***Withheld*** text. The name is also used as the default label for **Line Appearance** buttons set to this line.
- **Card:** *Software level = 4.1+.*
  Indicates the card slot being used for the device providing the line.
  - For IP400 control units, **SLOT A** on the control unit is shown as 1, **SLOT B** is shown as 2. Expansion modules are numbered from 4 upward, for example the module on **Expansion Port 1** is shown as 4.
  - For IP Office 500 control units, 1 to 4 match the slots on the front of the control unit from left to right. Expansion modules are numbered from 5 upwards, for example the module on **Expansion Port 1** is shown as 5.
- **Port:** *Software level = 4.1+.*
  Indicates the port on the **Card** above to which the configuration entry relates.
- **Line Number:**
  Allocated by the system.
- **Line SubType:** *Default = PRI*
  Set to ***PRI***. If set to ***T1*** see **Line Form (T1)**. If set to **ETSI**, **QSIG A** or **QSIG B** see **Line (E1)**.
  - ***QSIG*** is not supported on IP500 systems without **IP500 Voice Networking** licenses.
- **Channel Allocation:** *Default = 23 -> 1*
  The order, 23 to 1 or 1 to 23, in which channels are used.
- **Switch Type:** *Default = NI2*
  Options ***4ESS***, ***5ESS***, ***DMS100*** and ***NI2***.
- **Provider:** *Default = Local Telco*
  Select the PSTN service provider (***AT&T***, ***Sprint***, ***WorldCom*** or ***Local Telco***).
- **Prefix:** *Default = Blank*
  Enter the number to prefix to all incoming numbers for callback. This is useful if all users must dial a prefix to access an outside line. The prefix is automatically placed in front of all incoming numbers so that users can dial the number back.
- **Test Number:**
  Used to remember the external telephone number of this line to assist with loop-back testing. For information only.

- **Framing:** *Default = ESF*
  Selects the type of signal framing used (***ESF*** or ***D4***).
- **Zero Suppression:** *Default = B8ZS*
  Selects the method of zero suppression used (***B8ZS*** or ***AMI ZCS***).
- **Clock Quality:** *Default = Network*
  For digital calls the IP Office requires a clock source for call signalling and synchronization. Preferably the clock source from a central office exchange should be used. ***Network*** indicates that this line can be used as the clock source. When multiple ***Network*** sources are available the IP Office will use the first available source. A line set to ***Fallback*** is only used if no line set to ***Network*** is available. Lines set to ***Unsuitable*** are not used.
- **CSU Operation**
  Tick this field to enable the T1 line to respond to loop-back requests from the line.
- **Haul Length:** *Default = 0-115 feet*
  Sets the line length to a specific distance.
- **Channel Unit**: *Default = Foreign Exchange*
  This field should be set to match the channel signaling equipment provided by the Central Office. The options are ***Foreign Exchange***, ***Special Access*** or ***Normal***.
- **CRC Checking:** *Default = On*
  Turns CRC on or off.
- **Line Signaling:**
  The field can be set to either ***CPE*** (Customer Premises Equipment) or ***CO*** (Central Office). This field should normally be left at its default of ***CPE***. The setting ***CO*** is normally only used in lab back-to-back testing.
- **Incoming Routing Digits:** *Default=0 (present call immediately)*
  Sets the number of routing digits expected on incoming calls. This allows the line to present the call to the system once the expected digits have been received rather than waiting for the digits timeout to expire. This field only affects T1 line channels set to ***E&M Tie***, ***E&M DID***, ***E&M Switched 56K*** and ***Direct Inward Dial*.**

# Line | Channels (T1 PRI)

This tab allows settings for individual channels within the trunk to be adjusted. This tab is not available for trunks sets to ETSI or QSIG mode.

| Line \| Channels (T1 PRI) | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

- **Channel**
  Allocated by the system.

- **Channel Name:** *Default = Blank, Range up to 15 characters. Software level = 4.0 Q2 2007+.*
  This field is only visible on systems where the string ***SHOW_LINEID_NOT_OUTSIDE*** has been entered in the **Source Numbers** tab of the ***NoUser*** user. The value entered here is then used in place of the word  on call where on ICLID information is supplied.

- **Incoming Group ID:** *Default = 0, Range 0 to 99999.*
  The Incoming Group ID to which a line belongs is used to match it to incoming call routes in the IP Office configuration. The matching incoming call route is then used to route incoming calls. The same ID can be used for multiple lines.

- **Outgoing Group ID:** *Default = 0, Range 0 to 99999.*
  Short codes that specify a number to dial also specify the line group to be used. The IP Office will then seize a line with a matching **Outgoing Group ID**. The same ID can be used for multiple lines.

- **Line Appearance ID:** *Default = Auto-assigned, Range = 2 to 9 digits, Software level* = 3.0+.
  Used for configuring Line Appearances with button programming. The line appearance ID must be unique and not match any extension number.

- **Direction:** *Default = Both Directions*
  The direction of calls on the channel (***Incoming***, ***Outgoing*** or ***Both Directions***).

- **Bearer:** *Default = Any*
  The type of traffic carried by the channel (***Voice***, ***Data*** or ***Any***).

- **Service:** *Default = No Service or None.*
  If the line provider is set to AT&T, selects the type of service provided by the channel from ***Call by Call***, ***SDN (inc GSDN)***, ***MegaCom800***, ***MegaComWats***, ***Accunet***, ***NLDS***, ***I800***, ***ETN***, ***Private Line***, ***AT&T Multiquest***. For other providers the service options are **None** or **No Service**.

- **Admin:** *Default = In Service*
  Used to indicate the channel status (***In Service***, ***Out of Service*** or ***Maintenance***).

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

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

# Line | TNS (T1 PRI)

This tab is shown when the line **Provider** is set to ***AT&T***. It allows the entry of the Network Selection settings. These are prefixes for alternative long distance carriers. When a number dialed matches an entry in the table, that pattern is stripped from the number before being sent out. This table is used to set field in the TNS (Transit Network Selection) information element for ***4ESS*** and ***5ESS*** exchanges. It is also used to set fields in the NSF information element.

| **Line \| TNS (T1 PRI)** | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

- **TNS Code:**
  The pattern for the alternate long distance carrier. For example: The pattern 10XXX is added to this tab. If 10288 is dialed, 10 is removed and 288 is placed in the TNS and NSF information.

# Line | Special (T1 PRI)

This tab is shown when the line **Provider** is set to ***AT&T***. This table is used to set additional fields in the NSF information element after initial number parsing by the **TNS** tab. These are used to indicate the services required by the call. If the channel is set to Call by Call, then further parsing is done using the entries in the **Call by Call** tab.

| **Line \| Special (T1 PRI)** | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

- **Short code:**
  The number which results from the application of the rules specified in the User or System Short code tables and the Network Selection table and the Call-by-call table to the number dialed by the user.
- **Number:**
  The number to be dialed to line.
- **Special:** *Default = No Operator*
  (***No Operator***, ***Local Operator*** or ***Presubscribed Operator***).
- **Plan:** *Default = National*
  (***National*** or ***International***).

Typical values would be:

| Short code | Number | Service |
|---|---|---|
| 011N | N | No Operator, International |
| 010N | N | Local Operator, International |
| 01N | N | Local Operator, National |
| 00N | N | Presubscribed Operator, National |
| 0N | N | Presubscribed Operator, National |
| 1N | 1N | No operator, National |

# Line | Call By Call (T1 PRI)

This tab is shown when the line **Provider** is set to ***AT&T***. Settings in this tab are only used when calls are routed via a channel which has its Service set to Call by Call.

It allows short codes to be created to route calls to a different services according to the number dialed. Call By Call reduces the costs and maximizes the use of facilities. Call By Call chooses the optimal service for a particular call by including the Bearer capability in the routing decision. This is particularly useful when there are limited resources.

| **Line \| Call By Call (T1 PRI)** | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

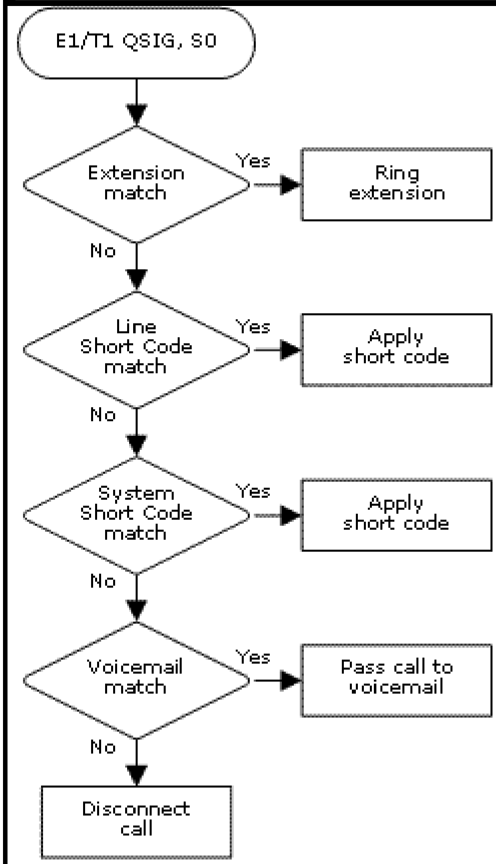
- **Short Code:**
  The number which results from the application of the rules specified in the User or System Short code tables and the Network Selection table to the number dialed by the user.
- **Number:**
  The number to be dialed to line.
- **Bearer:** *Default = Any*
  The type of channel required for the call (***Voice***, ***Data*** or ***Any***).
- **Service:** *Default = AT&T.99*
  The service required by the call (***SDN (inc GSDN)***, ***MegaCom800***, ***MegaCom***, ***Inwats***, ***Wats***, ***Accunet***, ***NLDS***, ***I800***, ***ETN***, ***Private Line***, ***AT&T Multiquest***).

# Line (S0)

## Line Form (S0) Overview

These settings are used for S0 ports provided by an S08 expansion module connected to the IP Office control unit. Though displayed as lines, these BRI ports are used for connection of ISDN2 devices such as video conferencing units or ISDN PC cards. For full details of installation refer to the IP Office Installation manual.

**Incoming Call Routing: E1 QSIG and S0 Lines**




The following options are used to route incoming calls received on this type of trunk.

1. **Extension Match**
   Based on matching an user or hunt group extension number.
2. **Line Short Code Match**
   The IP Office checks for a short code match based on the incoming number.
3. **System Short Code**
   The IP Office checks for a short code match based on the incoming number.
4. **Voicemail Match**
   The IP Office checks for a call flow start point name matching the incoming number.

# Line | Line (S0)

| Line \| Line (S0) | |
|---|---|
| Control Unit | SOE ✗, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓. |
| Software Level | 2.1 ✓, 3.0DT ✓, 3.0 ✓, 3.1 ✓, 3.2 ✓, 4.0 ✓, 4.1 ✓. |
| Mergeable | ✗. |

- **Line Number**
  This parameter is not configurable. It is allocated by the system.
- **Telephone Number:**
  Used to remember the telephone number of this line. For information only.
- **Prefix:** *Default = Blank.*
  On Incoming Calls the ISDN messaging tags the incoming call source location as either National or International (see their respective prefixes above). This determines the addition of the relevant 0 or 00 respectively (0 is default for National and 00 is default for International). If the ISDN message flags the call source as unknown, then the number in the **Prefix** field is added to the ICLID.
  - For outgoing calls: For IP Office 3.2 and lower, when a outgoing call is presented to the line with a leading digit to dial that matches the **Prefix**, that digit is stripped from the number. IP Office 4.0 and higher does not strip the prefix, therefore any prefixes not suitable for external line presentation should be stripped using short codes.
- **National Prefix:** ***Default = 0***
  This indicates the digits to be prefixed to a incoming national call. When a number is presented from ISDN as a "national number" this prefix is added. For example 7325551234 is converted to 17325551234.
- **International Prefix:** ***Default = 00***
  This indicates the digits to be prefixed to an incoming international call. When a number is presented from ISDN as an "international number" this prefix is added. For example 441923000000 is converted to 001441923000000.
- **TEI:** ***Default = 0***
  Not used. The Control Unit will ignore any entry.
- **Number of Channels:** *Default = 2*
  Defines the number of operational channels that are available on this line. 2 for BRI and up to 30 for PRI - depending upon the number of channels subscribed.
- **Outgoing Channels:** *Default = 2*
  This defines the number of channels available, on this line, for outgoing calls. This should normally be the same as Number of Channels field, but can be reduced to ensure incoming calls cannot be blocked by outgoing calls.
- **Voice Channels:** *Default = 2*
  The number of channels available for voice use.
- **Data Channels:** *Default = 2*
  The number of channels available for data use. If left blank the value is 0.
- **Clock Quality:**
  For digital calls the IP Office requires a clock source for call signalling and synchronization. Preferably the clock source from a central office exchange should be used. ***Network*** indicates that this line can be used as the clock source. When multiple ***Network*** sources are available the IP Office will use the first available source. A line set to ***Fallback*** is only used if no line set to ***Network*** is available. Lines set to ***Unsuitable*** are not used.

# Line | Short Codes

Line short codes can be applied to the digits received with incoming call. The stage at which they are applied also varies depending on the trunk type. See Line Short Codes in the **Short Codes** section.

- **IP Trunks**
  Line short codes are used if Small Community Networking (SCN) is not being used or no SCN user extension match occurs on the digits received.
- **QSIG and SO Trunks**
  S0 trunks, and E1 and T1 trunks set to QSIG mode, use line short code if the digits received do not match an internal extension number.

| Line \| Short Codes | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

Short codes can be added and edited using the **Add**, **Remove** and **Edit** buttons. Alternatively you can right-click on the list of existing short code to add and edit short codes.

# Line | Channels (S0)

This tab allows settings for individual channels within the trunk to be adjusted.

| Line \| Channels (S0) | |
|---|---|
| **Control Unit** | **SOE** ✗, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

- **Incoming Group ID:** *Default = 0, Range 0 to 99999.*
  The Incoming Group ID to which a line belongs is used to match it to incoming call routes in the IP Office configuration. The matching incoming call route is then used to route incoming calls. The same ID can be used for multiple lines.
- **Outgoing Group ID:** *Default = 0, Range 0 to 99999.*
  Short codes that specify a number to dial also specify the line group to be used. The IP Office will then seize a line with a matching Outgoing Group ID. The same ID can be used for multiple lines.

# Line (IP)

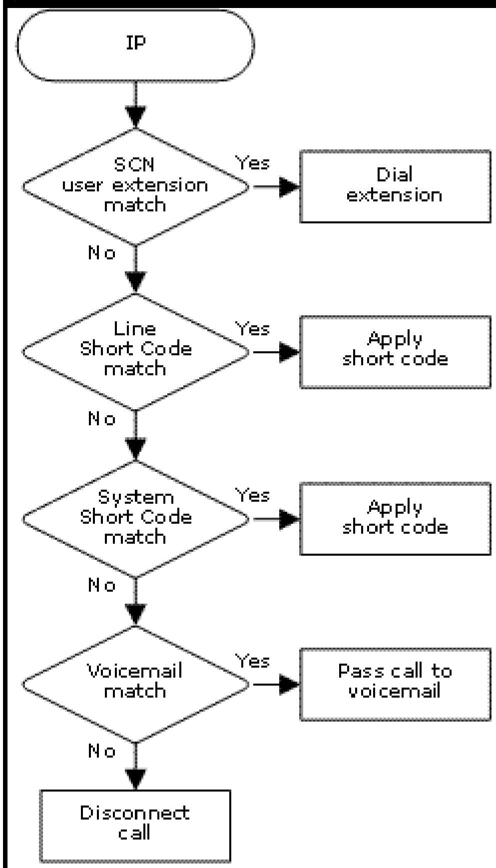
## Line Form (IP) Overview

These lines are added manually. They allow voice calls to be routed over data links within the IP Office system. They are therefore dependent on the IP data routing between the IP Office and the destination having being configured and tested.

**Network Assessments**
Not all data connections are suitable for voice traffic. A network assessment is required for internal network connections. For external network connections a service level agreement is required from the service provider. Avaya cannot control or be held accountable for the suitability of a data connection for carrying voice traffic. Refer to the IP Office Installation Manual for further details of Network Assessments and VoIP requirements.

- IP Trunks are not supported on IP500 systems without **IP500 Voice Networking** licenses.

**Incoming Call Routing: IP Lines**




The following options are used to route incoming calls received on this type of trunk.

1. **SCN User Extension**
   If the line is configured for Small Community Networking, the IP Office checks the incoming digits for a match to a user extension.
2. **Line Short Code**
   The IP Office checks for a short code match based on the incoming number.
3. **System Short Code**
   The IP Office checks for a short code match based on the incoming number.
4. **Voicemail Match**
   The IP Office checks for a call flow start point name matching the incoming number.

# IP Trunk Fallback

IP Office 3.1+ supports IP Trunk Fallback. When setting up a call over an IP trunk, if the remote PBX fails to respond within an adjustable timeout (default 5 seconds), the IP Office can attempt to reroute the call.

IP trunk fallback is performed separately for each call. The use of fallback for a preceding call does not alter the routing attempted for any following call to the same IP trunk.

Within a Small Community Network, calls to remote user are automatically routed to an IP trunk setup for connections to the appropriate remote IP Office. If the remote IP Office doesn't respond to the call attempt with the set timeout, the call is rerouted as follows:

- First the call is rerouted to any other trunk with the same outgoing group. If this is the case, that trunk must be able to route calls to the correct remote IP Office without any further dialing. This would typically require the fallback trunk to be a private leased line.
- If the call is still not connected an attempt is made to reroute the call via either LCR (IP Office 3.1 and 3.2) or ARS (IP Office 4.0+) short code matching of the original dialed number. This later method is the preferred case as the LCR/ARS short code matching allows number manipulation to make the outgoing dialing suitable for rerouting across the PSTN. To make this the preferred case, the IP trunk should be put into its own unique **Outgoing Trunk Group ID**.

In cases where the call is routed to the IP trunk by LCR or ARS in the first place, the timeout used for IP trunk fallback is the **Timeout** defined for the current LCR or ARS form being used. A value of ***0*** disables IP trunk fallback.

**Setting the Default IP Trunk Fallback Timeout**
The default timeout for IP trunk fallback is 5 seconds. This timeout can be changed for specific IP trunks and or all IP trunks. The required timeout is set through the **Source Numbers** tab of the **NoUser** user. The entry or entries take the form ***H323SetupTimerNoLCR line_number timeout*** where the ***line_number*** should be ***ALL*** for all IP trunks or the specific Line Number used for the IP trunk on its Line configuration tab, and the timeout is set in seconds.

# Line | Line (IP)

- IP Trunks are not supported on IP500 systems without **IP500 Voice Networking** licenses.

| Line \| Line (IP) | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

- **Line Number:** *Default = 0, Range = 1 to 249.*
  Enter the line number that you wish. Note that this must be unique.

- **Telephone Number:**
  Used to remember the telephone number of this line. For information only.

- **Incoming Group ID:** *Default = 0, Range 0 to 99999.*
  The Incoming Group ID to which a line belongs is used to match it to incoming call routes in the IP Office configuration. The matching incoming call route is then used to route incoming calls. The same ID can be used for multiple lines.

- **Outgoing Group ID:** *Default = 0, Range 0 to 99999.*
  Short codes that specify a number to dial also specify the line group to be used. The IP Office will then seize a line with a matching **Outgoing Group ID**. The same ID can be used for multiple lines.

- **Prefix:** *Default = Blank.*
  On Incoming Calls the ISDN messaging tags the incoming call source location as either National or International (see their respective prefixes above). This determines the addition of the relevant 0 or 00 respectively (0 is default for National and 00 is default for International). If the ISDN message flags the call source as unknown, then the number in the **Prefix** field is added to the ICLID.

  - For outgoing calls: For IP Office 3.2 and lower, when a outgoing call is presented to the line with a leading digit to dial that matches the **Prefix**, that digit is stripped from the number. IP Office 4.0 and higher does not strip the prefix, therefore any prefixes not suitable for external line presentation should be stripped using short codes.

- **National Prefix:** *Default = 0*
  This indicates the digits to be prefixed to a incoming national call. When a number is presented from ISDN as a "national number" this prefix is added. For example 7325551234" is converted to 17325551234".

- **International Prefix:** *Default = 00*
  This indicates the digits to be prefixed to an incoming international call. When a number is presented from ISDN as an "international number" this prefix is added. For example 441923000000 is converted to 001441923000000.

- **Number of Channels:** *Default = 20, Range 0 to 120.*
  Defines the number of operational channels that are available on this line.

- **Outgoing Channels:** *Default = 20, Range 0 to 120.*
  This defines the number of channels available, on this line, for outgoing calls. This should normally be the same as Number of Channels field, but can be reduced to ensure incoming calls cannot be blocked by outgoing calls.

- **Data Channels:** *Default = 20, Range 0 to 120.*
  The number of channels available for data use. If left blank the value is 0.

- **Voice Channels:** *Default = 20, Range 0 to 120.*
  The number of channels available for voice use.

- **TEI:** *Default = 0, Range = 0 to 127.*
  The Terminal Equipment Identifier. Used to identify each Control Unit connected to a particular ISDN line. For Point to Point lines this is typically (always) 0. It can also be 0 on a Point to Multi-Point line, however if multiple devices are actually sharing a Point to Multi-Point line it should be set to 127 which will result in the exchange deciding on the TEI's to be used by this Control Unit.

# Line | Short Codes

Line short codes can be applied to the digits received with incoming call. The stage at which they are applied also varies depending on the trunk type. See Line Short Codes in the **Short Codes** section.

- **IP Trunks**
  Line short codes are used if Small Community Networking (SCN) is not being used or no SCN user extension match occurs on the digits received.
- **QSIG and SO Trunks**
  S0 trunks, and E1 and T1 trunks set to QSIG mode, use line short code if the digits received do not match an internal extension number.

| **Line \| Short Codes** | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

Short codes can be added and edited using the **Add**, **Remove** and **Edit** buttons. Alternatively you can right-click on the list of existing short code to add and edit short codes.

# Line | VoIP Settings (IP)

| Line \| VoIP Settings (IP) | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

- **Gateway IP Address:** *Default = Blank*
  Enter the IP address of the remote control unit. This address must not be shared by any other IP, IP DECT or SIP line in the IP Office system configuration.
- **Voice Payload Size**
  This is the length of time represented by each VoIP packet in milliseconds. This is automatically defaulted to match the **Compression Mode** selected. Values are G.711/G.729 = 20ms, G.723=30ms.
- **Compression Mode:** *Default = Automatic Selection*
  This defines the compression method to be used for this line.
  - **Automatic Selection** - During call setup the IP Office negotiates the compression mode using the following order of preference: ***G729a***, ***G.723.1***, ***G711 ALAW***, ***G711 ULAW***.
  - If required a specific codec can be selected from: ***G.711 ALAW 64K***, ***G.711 ULAW 64K***, ***G.729(a) 8K CS-ACELP***, ***G.723.1 6K3 MP-MLQ***. If during connect with a specific codec, connect fails, the IP Office will fallback to using automatic selection.
- **H450 Support:** *Default = H450*
  Selects the supplementary service signaling method for use across H.323 connections. Options are ***None***, ***QSIG*** and ***H450***. Note that the selected method must be supported by the remote end. For IP Office to IP Office connections, H450 is preferred.
- **VoIP Silence Suppression:** *Default = Off*
  When selected, this option will detect periods of silence on any call over the line and will not send any data during those silent periods.
  - Note: This feature is not supported on IP lines using G711 between IP400 and IP500 systems.
- **Enable FastStart:** *Default = Off*
  A fast connection procedure. Reduces the number of messages that need to be exchanged before an audio channel is created.
- **Local Tones:** *Default = Off*
  When selected, the tones are generated by the local IP Office system to which the phone is registered. This option should not be used with lines being used for Small Community Networking. For the IP Office Small Office Edition control unit, this field should **not** be enabled.
- **Enable RSVP:** *Default = Disabled (Grayed out)*
- **Out of Band DTMF:** *Default = On*
  When on, DTMF is sent as a separate signal rather than as part of the encoded voice stream ("In Band"). This is recommended for low bit-rate compression modes such as G.729 and G.723 where DTMF in the voice stream can become distorted.

- **Allow Direct Media Path:** *Default = On*
  This settings controls whether H323 calls must be routed via the H323 gatekeeper (the IP Office) or can be routed alternately if possible within the network structure.
  - If enabled, H323 calls can take routes other than through the IP Office. This removes the need for a voice compression channel. Both ends of the calls must support Direct Media. Enabling this option may cause some vendors problems with changing the media path in mid call. Pre-IP Office 4.0, when using direct media path, it is not always possible for the extension to be recorded or monitored.
  - If disabled or not supported one end of the call, all call are routed via the IP Office.
    - On pre-4.0 these calls would then require a voice compression channel even if the IP devices use the same audio codec.
    - On IP Office 4.0 and higher, RTP relay support allows calls between devices using the same audio codec to not require a voice compression channel.
- **Voice Networking:** *Default = Off*
  Also known as "Small Community Networking". This option enables extension number sharing with the remote IP Office system. Extensions on the remote system can then be dialed from the local system.
  - Note: This requires that extension numbers and names on the two systems are unique. Line and group extension numbers are not shared. Remote extension numbers cannot be included in local groups.
  - Full operation requires H450 Support to be enabled over the links used.
- **Fax Transport Support:** *Default = Off*
  When selected, this option will provide support for faxing over a H.323 connection to another IP Office with the same setting.
- **Progress Ends Overlap Send:** *Default = Off, Software level* = 3.2+.
  Some telephony equipment, primarily AT&T switches, over IP trunks send a H323 Progress rather than H323 Proceeding message to signal that they have recognised the digits sent in overlap state. IP Office by default expects an H323 Proceeding message. This option is not available by default. If required, the value **ProgressEndsOverlapSend** must be entered into the **Source Numbers** tab of the ***NoUser*** user.

# Line (IP DECT)

## IP DECT Line Overview

This type of line can be manually added. They are used to route voice calls over an IP data connection to an Avaya IP DECT system. Only one IP DECT line can be added to an IP Office system. Refer to the IP Office IP DECT Installation manual for full details.

For North American locales, an IP DECT line is only supported with the IP Office 4.0 Q2 2007 maintenance release and higher.

# Line | Line (IP DECT)

Currently only one IP DECT line is supported on an IP Office system. For North American locales, an IP DECT line is only supported with the IP Office 4.0 Q2 2007 maintenance release and higher.

| Line \| Line (IP DECT) | |
|---|---|
| **Control Unit** | **SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓.** |
| **Software Level** | **2.1 ✗, 3.0DT ✗, 3.0 ✗, 3.1 ✓, 3.2 ✓, 4.0 ✓, 4.1 ✓.** |
| **Mergeable** | ✗. |

- **Line Number**
  This number is allocated by the system and is not adjustable.
- **Number Of Channels:** *Default = 0 (Grayed out)*
  Defines the number of operational channels that are available on this line. This will match the number of associated IP DECT extensions.
- **Outgoing Channels:** *Default = 0 (Grayed out)*
  This defines the number of channels available, on this line, for outgoing calls. This will match the number of associated IP DECT extensions.
- **Voice Channels:** *Default = 0 (Grayed out)*
  The number of channels available for voice use. This will match the number of associated IP DECT extensions.
- **Incoming Group ID:** *Default = 240 (Grayed out)*
  This number is allocated by the system and is not adjustable. The incoming group ID number should not be used for the routing of calls within IP Office incoming call routes.
- **Outgoing Group ID:** *Default = 240 (Grayed out)*
  This number is allocated by the system and is not adjustable. The outgoing group ID number should not be used to match short codes dialed on the system with trunks to use.
- **Extensions**
  Lists all the DECT extensions associated with the IP DECT line. Adding and deleting IP DECT extensions is done via the Manager extension list.

# Line | Gateway (IP DECT)

| Line \| Gateway (IP DECT) | |
|---|---|
| Control Unit | SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓. |
| Software Level | 2.1 ✗, 3.0DT ✗, 3.0 ✗, 3.1 ✓, 3.2 ✓, 4.0 ✓, 4.1 ✓. |
| Mergeable | ✗. |

- **Gateway IP Address:** *Default = Blank*
  Enter the IP address of the IP DECT base station that will be the Avaya IP DECT Mobility Manager (ADMM). If **Enable DHCP Support** is enabled, this address should be from the range of assignable IP Office DHCP addresses. This address must not be shared by any other IP, IP DECT or SIP line in the IP Office system configuration.
- **Compression Mode:** *Default = Automatic Selection*
  This defines the type of compression which is to be used for calls on this line.
    - **Automatic Selection** - During call setup, the IP Office negotiates the compression mode using the following order of preference: G729a, G.723.1, G711 ALAW, G711 ULAW. This order is an optimum balance of quality and bandwidth for most scenarios.
    - Other available options are: G.711 ALAW 64K, G.711 ULAW 64K, G.729(a) 8K CS-ACELP, G.729 Simple, G.723.1 6K3 MP-MLQ.
- **Gain:** *Default = Default*
  Allow adjustment of the gain applied to calls between the IP DECT system and the IP Office.
- **Enable DHCP Support:** *Default = Off*
  The IP DECT base stations require DHCP and TFTP support. Enable this option if the IP Office is being used to provide that support, using IP addresses from its DHCP range (LAN1 or LAN2) and its TFTP server setting. If not enabled, alternate DHCP and TFTP options must be provided during the IP DECT installation.
    - If it is desired to use the IP Office for DHCP support of the ADMM and IP DECT base stations only, the IP Office address range should be set to match that number of addresses. Those addresses are then taken during the IP Office restart and will not be available for other DHCP responses following the restart.
    - For Small Office Edition and IP406V2 control units, use of the embedded voicemail memory card slot for the TFTP server is recommended for small IP DECT installations. See **System | TFTP IP Server Address**. For other control units, or larger IP DECT installations, the use of a non-embedded TFTP software option other than Manager is recommended.

  When **Enable DHCP Support** is selected, the following fields are also enabled:
    - **Boot File:** *Default = ADMM_RFP_1_0_0.tftp, Range = Up to 31 characters.*
      The name and path of the ADMM software file. The path is relative to the TFTP server root directory.
    - **ADMM MAC Address:** *Default = 00:00:00:00:00:00*
      This field must be used to indicate the MAC address of the IP DECT base station that should load the ADMM software file and then act as the IP DECT system's ADMM. The address is entered in hexadecimal format using comma, dash, colon or period separators.

- **VLAN ID:** *Default = Blank, Range = 0 to 4095.*
  If VLAN is being used by the IP DECT network, this field sets the VLAN address assigned to the base stations by the IP Office if **Enable DHCP Support** is selected.
  - The IP Office itself does not apply or use VLAN marking. It is assumed that the addition of VLAN marking and routing of VLAN traffic is performed by other switches within the customer network.
  - An ID of zero is not recommended for normal VLAN operation.
  - When blank, no VLAN option is sent to the IP DECT base station.
- **Base Station Address List:** Default = Empty
  This box is used to list the MAC addresses of the IP DECT base stations, other than the base station being used as the ADMM and entered in the **ADMM MAC Address field**. Right-click on the list to select **Add** or **Delete**. or use the **Insert** and **Delete** keys. The addresses are entered in hexadecimal format using comma, dash, colon or period separators.

- **Silence Suppression:** *Default = Off*
  When selected H.323 terminals will not send data if they are silent, this is useful when optimizing data traffic.
- **Enable RSVP:** *Default = Disabled (Grayed out)*
  This setting is allocated by the system and is not adjustable. RSVP is not support on the IP DECT system.
- **Out of Band DTMF:** *Default = On (Grayed out)*
  This setting is allocated by the system and is not adjustable. When on, DTMF is sent as a separate signal rather than as part of the encoded voice stream ("In Band"). This is recommended for low bit-rate compression modes such as G.729 and G.723 where DTMF in the voice stream can become distorted.
- **Allow Direct Media Path:** *Default = On*
  When disabled the media (voice) path always passes through the IP Office Control Unit. When enabled the remote end may be told of a new IP address for the media path if for example the call is transferred to a H.323 extension. Enabling this option may cause some vendors problems with changing the media path in mid call.
- **Auto-create Extension:** *Default = Off, Software level* = 4.0+.
  If enabled, subscription of a handset with the IP DECT ADMM causes the auto-creation of a matching numbered extension and user within the IP Office configuration.

# SIP Line

## SIP Overview

IP Office 4.0 and higher supports SIP voice calls through the addition of SIP trunks to the IP Office configuration. This approach allows users with non-SIP phones to make and receive SIP calls.

Use of SIP requires the following:

1. **SIP Service Account**
   An account or accounts with a SIP internet service provider (ITSP). The method of operation and the information provided will vary. The key requirement is a SIP URI, a web address of the form ***mysip@itsp.com***. This is the equivalent of a SIP telephone number for making and receiving calls via SIP.

2. **Voice Compression Channels**
   SIP calls use IP Office voice compression channels in the same way as used for standard IP trunks and extensions. These are provided by the installation of VCM modules within the IP Office control unit. RTP relay is applied to SIP calls where applicable.

3. **Licensing**
   SIP Trunk Channels licenses is required in the IP Office configuration. These set the maximum number of simultaneous SIP calls supported by the IP Office. Multiple licenses can be added to achieve a cumulative maximum number of channels supported.

4. **Firewall Traversal**
   Routing traditional H323 VoIP calls through firewalls often fails due to the effects of NAT (Network Address Translation). For SIP a number of ways of ensuring successful firewall traversal have been created.

   - **STUN (Simple Traverse of UDP NAT)**
     UDP SIP can use a mechanism called STUN to cross firewalls between the switch and the ITSP. This requires the ITSP to provide the IP address of their STUN server and the IP Office to then select from various STUN methods how to connect to that server. The IP Office can attempt to auto-detect the required settings to successfully connect. These settings are part of the **System | LAN1** and **System | LAN2** forms. STUN is not required is the ITSP used Session Border Control (SBC).

   - **TURN (Traversal Using Relay NAT)**
     TCP SIP can use a mechanism called TURN (Traversal Using Relay NAT). IP Office does not currently support this method of firewall traversal.

   - **IP Office Firewall**
     The IP Office firewall between LAN1 and LAN2 is not applied to SIP calls.

5. **SIP Trunks**
   These trunks are manually added to the IP Office configuration. Typically a SIP trunk is required for each SIP ITSP being used. As the configuration provides methods for multiple URI's from that ITSP to use the same trunk. For each trunk at least one SIP URI entry is required. Amongst other things this sets the incoming and outgoing groups for call routing.

6. **Outgoing Call Routing**
   The initial routing uses any standard short code with a dial feature. The short code's **Line Group ID** should be set to match the **Outgoing Group ID** of the **SIP URI** channels to use. However the short code must also change the number dialed into a destination SIP URI suitable for routing by the ITSP. In most cases, if the destination is a public telephone network number, a URI of the form ***123456789@itsp.com*** is suitable. For example:

   - **Code:** 9N#
   - **Feature:** Dial
   - **Telephone Number:** N
   - **Line Group ID:** 100

7. **Incoming Call Routing**
   Incoming SIP calls are routed in the same way as other incoming external calls. The caller and called information in the SIP call header can be used to match **Incoming CLI** and **Incoming Number** settings in normal IP Office **Incoming Call Route** entries.
8. **DiffServe Marking**
   DiffServe marking is applied to calls using the **DiffServer Settings** on the **System | LAN | Gatekeeper** tab of either LAN1 or LAN2 as set by the line's **Use Network Topology Info** setting.

**SIP URIs**
Calls across SIP require URI's (Uniform Resource Identifiers), one for the source and one for the destination. Each SIP URI consists of two parts, the user part (for example mysip) and the domain part (for example itsp.com) to form a full URI (in this case mysip@itsp.com). SIP URI's can take several forms:

- **john.doe@117.53.22.2**
- **Sales@itsp.com**
- **012345678@itsp.com**

Typically each account with a SIP service provider will include a SIP URI or a set of URI's. The domain part is then used for the SIP trunk configured for routing calls to that provider. The user part can be assigned either to an individual user if you have one URI per user for that ITSP, or it can also be configured against the line for use by all users who have calls routed via that line.

**Resource Limitation**
A number of limits can affect the number of SIP call. When one of these limits is reached the following occurs: any further outgoing SIP calls are blocked unless some alternate route is available using ARS; any incoming SIP calls are queued until the required resource becomes available. Limiting factors are:

- the number of licensed SIP channels.
- the number of SIP channels configured for a SIP URI.
- the number of voice compression channels.
  - **SIP Line Call to/from Non-IP Devices**
    Voice compression channel required.
  - **Outgoing SIP Line Call from IP Device**
    No voice compression channel required.
  - **Incoming SIP Line Call to IP Device**
    Voice compression channel reserved until call connected.

**SIP Information Display**
For Delta Server the full from and to SIP URI will be recorded for use by SMDR, CBC and or CCC. For all other applications and for telephone devices, the SIP URI is put through system directory matching the same as for incoming CLI matching. First a match against the full URI is attempted, then a match against the user part of the URI. Directory wildcards can also be used for the URI matching.

**SIP Standards**
The IP Office implementation of SIP conforms to the following SIP RFC's.

| RFC | Description |
|---|---|
| 2833 [7] | RTP Payload for DTMF digits, telephony tones and telephony signals. |
| 3261 [8] | SIP Session Initiation Protocol. |
| 3264 [11] | An Offer/Answer Model with Session Description Protocol (SDP). |
| 3323 [14] | A Privacy Mechanism for SIP |
| 3489 [18] | STUN - Simple Traversal of User Datagram Protocol (UDP) Through Network Address Translators (NAT's). |
| 3824 [24] | Using E.164 Numbers with the Session Initiation Protocol (SIP).<br>E.164 is the ITU-T recommendation for international public telecommunication numbering plans. |

# SIP Incoming Call Routing

Incoming SIP calls are routed using IP Office Incoming Call Routes in the same way as call arriving on other external trunks. The following Incoming Call Route fields are used to determine which route is the best match for a call.

- **Line Group ID**
  This field is matched against the Incoming Group settings of the SIP URI (Line | SIP URI). This must be an exact match.
- **Incoming Number**
  This field can be used to match the called details (TO) in the SIP header of incoming calls. It can contain a number, SIP URI or Tel URI. For SIP URI's the domain part of the URI is removed before matching by incoming call routing occurs. For example, for the SIP URI ***mysip@sipitsp.com*** , only the user part of the URI, ie. ***mysip***, is used for matching.
- **Incoming CLI**
  This field can be used to match the calling details (FROM) in the SDP header of incoming SIP calls. It can contain a number, SIP URI, Tel URI or IP address received with SIP calls. For all types of incoming CLI except IP addresses a partial entry can be used to achieve the match, entries being read from left to right. For IP addresses only full entry matching is supported.

The fields Bearer Capability and Incoming Sub Address are not used for matching of incoming SIP calls. The remain Incoming Call Route fields, including those voice recording, as used as for all call types.

# Line | SIP Line

| Line \| SIP Line | |
|---|---|
| Control Unit | SOE ✓, IP403 ✗, IP406 V1 ✗, IP406 V2 ✓, IP412 ✓, IP500 ✓. |
| Software Level | 2.1 ✗, 3.0DT ✗, 3.0 ✗, 3.1 ✗, 3.2 ✗, 4.0 ✓, 4.1 ✓. |
| Mergeable | ✗. |

- **Line Number:** *Default = Automatically assigned.*
  By default a value is assigned by the IP Office. This value can be changed but it must be unique.
- **ITSP Domain Name:** *Default = Blank.*
  This field is used to enter the domain part of the SIP URI provided by the ITSP. For example, in the SIP URI ***mysip@itsp.com***, the domain part of the URI is ***itsp.com***. For outgoing calls the user part of the SIP URI is determined in a number of ways:
    - For the user making the call, the user part of the FROM SIP URI is determined by the settings of the **SIP URI** channel entry being used to route the call. This will use one of the following:
        - a specific name entered in **Local URI** field of the channel entry.
        - or specify using the primary or secondary authentication name set for the line below
        - or specify using the **SIP Name** set for the user making the call (**User | SIP | SIP Name**).
    - For the destination of the call, the user part of the TO SIP URI is determined by the dialing short codes of the form 9N / N where N is the user part of the SIP URI.
- **ITSP IP Address:** *Default = 0.0.0.0*
  This value is provided by the SIP ITSP. This address must not be shared by any other IP, IP DECT or SIP line in the IP Office system configuration.
- **Primary Authentication Name:** *Default = Blank.*
  This value is provided by the SIP ITSP. Depending on the settings on the Local URI tab associated with the SIP call it may also be used as the user part of the SIP URI.
    - If the **From** field on the **Local URI** being used for the call is set to **Use Authentication Name** and the **Registration** is set to ***Primary***, this value is used as the user part of the SIP URI for calls.
- **Primary Authentication Password:** *Default = Blank.*
  This value is provided by the SIP ITSP.
- **Primary Registration Expiry:** *Default = 3600 minutes.*
  This setting defines how often registration with the SIP ITSP is required following any previous registration.
- **Secondary Authentication Name:** *Default = Blank.*
  This value is provided by the SIP ITSP. Depending on the settings on the Local URI tab associated with the SIP call it may also be used as the user part of the SIP URI.
    - If the **From** field on the **Local URI** being used for the call is set to **Use Authentication Name** and the **Registration** is set to ***Secondary***, this value is used as the user part of the SIP URI for calls.
- **Secondary Authentication Password:** *Default = Blank.*
  This value is provided by the SIP ITSP.
- **Secondary Registration Expiry:** *Default = 3600 minutes.*
  This setting defines how often registration with the SIP ITSP is required following any previous registration.

- **Registration Required:** *Default = Off.*
  If selected, the SIP trunk will register with the ITSP using the value in the **ITSP Domain Name** field.
- **In Service:** *Default = On.*
  When this field is not selected, the SIP trunk is unregistered and not available to incoming and outgoing calls.
- **Use Tel URI:** *Default = Off.*
  Use Tel URI format (for example ***TEL: +1-425-555-4567***) rather than SIP URI format (for example ***mysip@itsp.com***). This affects the **From** field of outgoing calls. The To field for outgoing calls will always use the format specified by the short codes used for outgoing call routing.
- **VoIP Silence Suppression:** *Default = Off.*
  When selected, this option will detect periods of silence on any call over the line and will not send any data during those silent periods.
- **Out of Band DTMF:** *Default = Off.*
  This field is greyed out and cannot be changed. When on, DTMF is sent as a separate signal rather than as part of the encoded voice stream ("In Band"). This is recommended for low bit-rate compression modes such as G.729 and G.723 where DTMF in the voice stream can become distorted.
  - IP Office supports RFC2833 to allow DTMF transport over SIP trunks. IP Office does not support the INFO method used by a few service providers for out-of-band DTMF.
- **Local Tones:** *Default = On.*
  This field is greyed out and cannot be changed. When on, call tones are generated by the local IP Office system to which the phone is registered.
- **Fax T38:** *Default = Off.*
  This field is greyed out and cannot be changed.
- **RE-INVITE Supported:** *Default = Off.*
  When enabled, Re-INVITE can be used during a session to change the characteristics of the session, for example when the target of an incoming call or a transfer does not support the codec originally negotiated on the trunk. Requires the ITSP to also support Re-Invite.
- **Voice Packet Size**
  This is the length of time represented by each VoIP packet in milliseconds. This is automatically defaulted to match the **Compression Mode** selected. Values are G.711/G.729 = 20ms, G.723=30ms.
- **Compression Mode:** *Default = Automatic*
  This defines the compression method to be used for this line.
  - **Automatic Selection** - During call setup the IP Office negotiates the compression mode using the following order of preference: ***G729a***, ***G.723.1***, ***G711 ALAW***, ***G711 ULAW***.
  - If required a specific codec can be selected from: ***G.711 ALAW 64K***, ***G.711 ULAW 64K***, ***G.729(a) 8K CS-ACELP***, ***G.723.1 6K3 MP-MLQ***. If during connect with a specific codec, connect fails, the IP Office will fallback to using automatic selection.

**Network Configuration**

- **Layer 4 Protocol:** *Default = UDP*
  This field sets whether the line uses UDP SIP or TCP SIP.
- **Use Network Topology Info:** *Default = LAN1*
  This field associates the SIP line with the **System | LAN1 | Network Topology** settings of either ***LAN1*** or ***LAN2***. If ***None*** is selected, IP Office STUN lookup is not applied and routing is determined by the IP Office routing tables.
- **Send Port:** *Default = 5060*
  This field sets the port to which IP Office send outgoing SIP calls.
- **Listen Port:** *Default = 5060*
  This field sets the port on which the IP Office listens for incoming SIP calls.

# SIP URI

Having setup the SIP trunk to the SIP ITSP, the SIP URI's registered with that ITSP are entered on this tab. A SIP URI (Uniform Resource Identifier) is similar to an internet email address, for example ***sipname@mysipitsp.com***, or **01555326978@mysipitsp.com** and represents the source or destination for SIP connection. The URI consists of two parts, the user part (eg. ***mysip***) and the host part (eg. ***mysipitsp***).

In the case of IP Office each SIP URI acts as a set of trunk channels. Outgoing calls can then be routed to the required URI by short codes that match that URI's **Outgoing Group** setting. Incoming calls can be routed by incoming call routes that match the URI's **Incoming Group** setting.

| **Line \| SIP URI** | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✗, **IP406 V1** ✗, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✗, **3.0DT** ✗, **3.0** ✗, **3.1** ✗, **3.2** ✗, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

- **Via**
  This field is for information only and cannot be edited. It shows the IP address of the IP Office LAN interface with which the SIP trunk is associated.
- **Local URI:** *Default = Use Authentication Name*
  This field sets the 'From' for outgoing SIP calls using this URI. The value can either be entered manually or the options ***Use Authentication Name*** or ***Use User Data*** selected.
  - ***Use Authentication Name***
    Use the appropriate **Authentication Name** on the SIP Line tab as indicated by the **Registration** setting below.
  - ***Use User Data***
    Use the **SIP Name** value from the **User | SIP** tab of the user making the SIP call.
- **Contact:** *Default = Use Authentication Name*
  This field sets the 'Contact' details for SIP calls using this URI. The value can either be entered manually or the options ***Use Authentication Name*** or ***Use User Data*** selected.
  - ***Use Authentication Name***
    Use the appropriate **Authentication Name** on the SIP Line tab as indicated by the **Registration** setting below.
  - ***Use User Data***
    Use the **Contact** value from the **User | SIP** tab of the user making the SIP call.
- **Display Name:** *Default = Use Authentication Name*
  This field sets the 'Name' value for SIP calls using this URI. The value can either be entered manually or the options ***Use Authentication Name*** or ***Use User Data*** selected.
  - **Use Authentication Name**
    Use the appropriate **Authentication Name** on the SIP Line tab as indicated by the **Registration** setting below.
  - **Use User Data**
    Use the **SIP Display Name (Alias)** value from the **User | SIP** tab of the user making the SIP call.
- **Registration:** *Default = Primary*
  This field sets whether the primary or secondary authentication name values set on the SIP line tab are used.
- **Incoming Group:** *Default = 0, Range 0 to 99999.*
  The **Incoming Group** to which a line belongs is used to match it to incoming call routes in the IP Office configuration. The matching incoming call route is then used to route incoming calls.

- **Outgoing Group:** *Default = 0, Range 0 to 99999.*
  Short codes that specify a number to dial also specify the line group to be used. The IP Office will then seize a line with a matching **Outgoing Group ID**.
- **Max Calls per Channel:** *Default =10*
  This field sets the maximum number of simultaneous calls that can use the URI before the IP Office returns busy to any further calls.

# SES Line

## SES Line

IP Office SES lines are used for connection to an Avaya SIP Enablement Service (SES) server within an Avaya SIP for Branch network. This is a variant of TCP SIP line and requires **SIP Trunk Channels** licenses in the IP Office configuration for the number of simultaneous SIP and or SES calls. The SES server supports up to 1000 branches arranged in a star network configuration.

With an Avaya SIP for Branch network, all extensions are reachable from any branch. Using short codes, the IP Office should be configured to route calls for other branches in the network to the SES server via a SES line. The SES server acts as a SIP proxy. It holds a table of all branch prefixes and uses that table to reroute each call to the appropriate destination branch. The SES server will only route calls to the branch if the number match both the branch prefix and the expected extension number length.

**Related Fields**

There are a number of fields in the IP Office configuration which are used with SES lines but are not on the SES Line form. The following are located on the **System | System** tab:

- **Branch Prefix:** *Default = Blank, Range = 1 to 65535, Software Level = 4.1+.*
  Used to identify the IP Office system within an Avaya SIP for Branch network via an SES server. The branch prefixes of each branch within the network must unique and must not overlap. For example 85, 861 and 862 are okay but 86 and 861 overlap.
  - For ease of routing and maintenance the prefixes should be the same length and begin with the same digit, for example 800, 801, 802 and so on. Routing of calls to the SES line can then be based on the leading digit used for branch prefixes.
- **Local Number Length:** *Default = Blank (Off), Range = Blank (Off) or 3 to 9, Software level = 4.1+.*
  Sets the default length for extension numbers for extensions, users and hunt groups. Entry of an extension number of a different length will cause an error warning within IP Office Manager. This field is intended for IP Office systems being used in an Avaya SIP for Branch network linked via an SES server, where fixed extension lengths must be used. Note that the **Branch Prefix** and the **Local Number Length** must not exceed 15 digits.
  - Using the same local number length for all branches is highly recommended. In addition to simplifying call routing it allows for a common number plan within the network. For example, key services such as reception or security at each branch can be given the same extension number. Those services can then be contacted at any branch simply by knowing the branch prefix and the common number for the service.

**Short Codes**

Calls are routed to the SES line using short codes in the same was as for other line types. The short codes can route calls directly to the SES line's Outgoing Group ID or to an ARS form configured for the SES line. If at all possible, requiring a short code for each branch within the SIP For Branch Network should be avoided.

- **Common Branch Prefix Leading Digit Routing**
  If a common digit has been used at the start of all branch prefixes, that digit can be used as the key for a single SES routing short code. For example, first use a single range for branch prefixes, ie. 80 to 89 in a small network, 800 to 899 in a medium network or 8000 to 8999 in a large network. At each branch a single short code of the form ***8N*** could then be used to start SES routing.
- **Common Branch Prefix and Extension Length Routing**
  If the above method cannot be used, maintaining a common branch prefix and local number length throughout the network is another option to simplify routing. For example, if all branches have a two digit branch prefixes and then 4 digit extension numbers, a short code of the form ***XXXXXX;*** could be used to only match dialed six digit numbers. Longer dialing is still possible so long as the first additional digit is dialed within the **Dial Delay Time** setting of the IP Office system.
- **SES Line Prefix**
  The SES line includes a prefix field that can be added to any calling number information supplied with calls received on a SES line. That prefix can then be used as the mechanism to route return calls back to the SES line. For example, if the line prefix is 3, a short code of the form 3N/N can be used.

**Additional Notes**

1. **Voice Compression Channels**
   SES calls use IP Office voice compression channels in the same way as used for standard IP trunks and extensions. These are provided by the installation of VCM modules within the IP Office control unit. RTP relay is applied to SES calls where applicable.
2. **Licensing**
   SIP Trunk Channels licenses is required in the IP Office configuration. These set the maximum number of simultaneous SIP and SES calls supported by the IP Office. Multiple licenses can be added to achieve a cumulative maximum number of channels supported.
3. **IP Office Firewall**
   The IP Office firewall between LAN1 and LAN2 is not applied to SES calls.
4. **Incoming Call Routing**
   Incoming SES calls are routed as if being internal calls. The dialed branch prefix is removed and the remaining extension number is used as if dialed on switch.
5. **DiffServe Marking**
   DiffServe marking is applied to calls using the **DiffServer Settings** on the **System | LAN | Gatekeeper** tab of either LAN1 or LAN2 as set by the line's **Use Network Topology Info** setting.

**Resource Limitation**

A number of limits can affect the number of SES calls. When one of these limits is reached the following occurs: any further outgoing SES calls are blocked unless some alternate route is available using ARS; any incoming SES calls are queued until the required resource becomes available.

Limiting factors are:

- the number of licensed SIP channels.
- the **Max Calls** setting of the SES line.
- the number of voice compression channels.
  - **SES Line Call to/from Non-IP Devices**
    Voice compression channel required.
  - **Outgoing SES Line Call from IP Device**
    No voice compression channel required.
  - **Incoming SIP Line Call to IP Device**
    Voice compression channel reserved until call connected.
- If the IP Office also used SIP trunks, the same resources are shared between SES and SIP calls.

| Line \| SES Line | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✗, **IP406 V1** ✗, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✗, **3.0DT** ✗, **3.0** ✗, **3.1** ✗, **3.2** ✗, **4.0** ✗, **4.1** ✓. |
| **Mergeable** | ✗. |

- **Line Number:** *Default = Automatically assigned.*
  By default a value is assigned by the IP Office. This value can be changed but it must be unique.
- **SES Domain Name:** *Default = Blank.*
  This field is used to enter the domain part of the SIP URI provided by the ITSP. For example, in the SIP URI ***mysip@itsp.com***, the domain part of the URI is ***itsp.com***. For outgoing calls the user part of the SIP URI is the Branch Prefix and the extension number.
- **SES Address:** *Default = 0.0.0.0*
  This value is the public IP address of the SES server. This address must not be shared by any other IP, IP DECT or SIP line in the IP Office system configuration.
- **Inactivity Timeout (seconds):** *Default = 120, Range = 0 to 99999 seconds.*
  If no SIP messages or signalling have been sent or received during this period the IP Office will close the connection.
- **Outgoing Group:** *Default = 0, Range 0 to 99999.*
  Short codes that specify a number to dial also specify the line group to be used. The IP Office will then seize a line with a matching **Outgoing Group ID**.
- **Max Calls:** *Default =10*
  This field sets the maximum number of simultaneous calls that can use the URI before the IP Office returns busy to any further calls.
- **Prefix:** *Default = Blank.*
  This prefix will be added to any source number received with incoming calls. Normally this should match a dialing short code configured to route matching calls to the SES line's **Outgoing Group ID** number.
- **In Service:** *Default = On.*
  When this field is not selected, the SIP trunk is unregistered and not available to incoming and outgoing calls.
- **VoIP Silence Suppression:** *Default = Off.*
  When selected, this option will detect periods of silence on any call over the line and will not send any data during those silent periods.

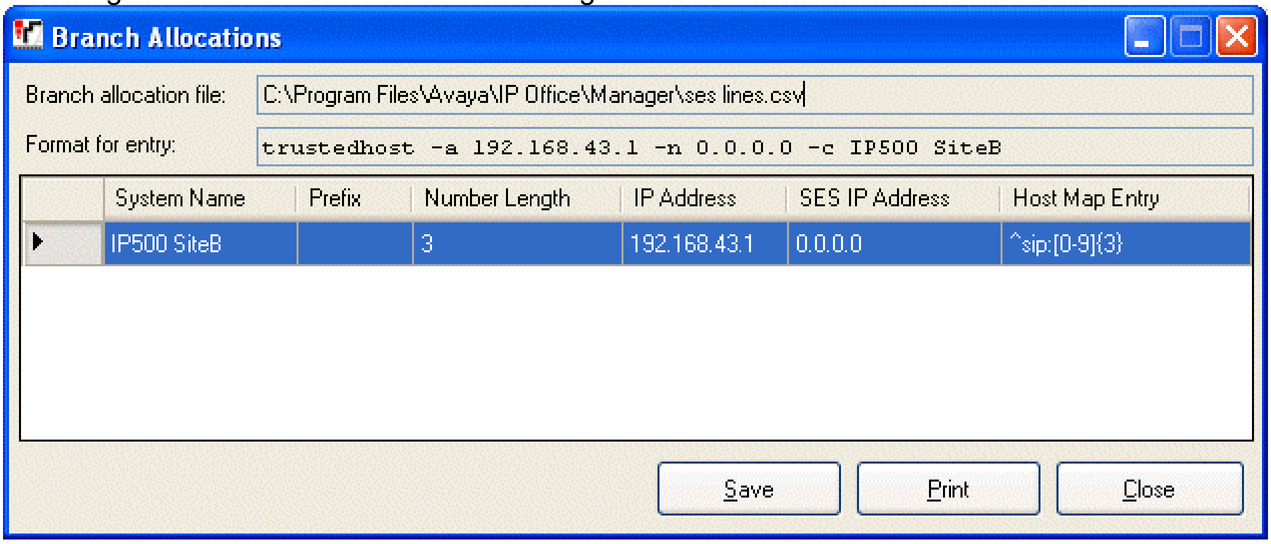
- **Out of Band DTMF:** *Default = Off.*
  This field is greyed out and cannot be changed. When on, DTMF is sent as a separate signal rather than as part of the encoded voice stream ("In Band"). This is recommended for low bit-rate compression modes such as G.729 and G.723 where DTMF in the voice stream can become distorted.
  - IP Office supports RFC2833 to allow DTMF transport over SIP trunks. IP Office does not support the INFO method used by a few service providers for out-of-band DTMF.
- **Local Tones:** *Default = On.*
  This field is greyed out and cannot be changed. When on, call tones are generated by the local IP Office system to which the phone is registered.
- **Fax T38:** *Default = Off.*
  This field is greyed out and cannot be changed.
- **RE-INVITE Supported:** *Default = Off.*
  When enabled, Re-INVITE can be used during a session to change the characteristics of the session, for example when the target of an incoming call or a transfer does not support the codec originally negotiated on the trunk.
- **Voice Packet Size**
  This is the length of time represented by each VoIP packet in milliseconds. This is automatically defaulted to match the **Compression Mode** selected. Values are G.711/G.729 = 20ms, G.723=30ms.
- **Compression Mode:** *Default = Automatic*
  This defines the compression method to be used for this line.
  - **Automatic Selection** - During call setup the IP Office negotiates the compression mode using the following order of preference: ***G729a***, ***G.723.1***, ***G711 ALAW***, ***G711 ULAW***.
  - If required a specific codec can be selected from: ***G.711 ALAW 64K***, ***G.711 ULAW 64K***, ***G.729(a) 8K CS-ACELP***, ***G.723.1 6K3 MP-MLQ***. If during connect with a specific codec, connect fails, the IP Office will fallback to using automatic selection.
- **Network Configuration**
  - **Layer 4 Protocol:** *Default = TCP*
    This field cannot be changed.
  - **Use Network Topology Info:** *Default = LAN1*
    For various settings, this field indicates whether the SES line uses the **System | LAN** settings of either ***LAN1*** or ***LAN2***. For instance the DiffServe Settings applied to outgoing SES calls and the IP address (see description of Via Nat below).
  - **Send Port:** *Default = 5060*
    This field cannot be changed.
  - **Listen Port:** *Default = 5060*
    This field cannot be changed.
  - **Via NAT:** *Default = Off.*
    This option controls which address should be used as the IP address of the IP Office system that is entered into the SES server configuration.
    - If disabled, the address used is the **IP Address** of the IP Office system as set on the **LAN Settings** sub-tab of **LAN1** or **LAN2**. The selection of LAN1 or LAN2 is determined by the **Use Network Topology** field above.
    - If enabled, the address used is the **Public IP Address** as set on the **Network Topology** sub-tab of **LAN1** or **LAN2**. The selection of LAN1 or LAN2 is determined by the **Use Network Topology** field above.

**Prefix Allocation Check**

As SES lines are configured within each IP Office system in the SIP for Branch network, details of the IP Office system can be written to a CSV file. This file can then be used to ensure the following:

1. **Check branch allocation**
   Check that the settings of the IP Office and SES line being added are both complete and do not conflict with those of other IP Office systems in the Distribute Network already added to the CSV file. Select this option and click **Execute** , then select the CSV file containing details of the other IP Offices with SES lines.

2. **List branch allocations**
   Provide the source of the details that need to be entered in the SES server configuration for each IP Office. Select this option and click **Execute** to select and display the CSV file containing details of existing IP Office SES line settings. This file provides the information necessary for matching line entries in the SES server configuration.



# Control Unit Settings

## Control Unit | Control Unit

The Control Unit configuration form gives details for devices connected to the system. This includes some modules within the control unit as well as external expansion modules.

For most units, this information is allocated by the system and is not configurable.

The **New** and **Delete** actions on this form have special functions.

- **New**
  This action is used to added a WAN3 expansion module. If when a WAN3 is added to the system, the WAN3 is not recognised following a system reboot, **New** on this form can be used to scan for the WAN3 module.
- **Delete**
  This action can only be used with external expansion modules. It cannot be applied to the control unit and modules inside the control unit. The action should used with caution as deleting an expansion module will also delete any extensions or lines associated with that expansion module. If the module is physically present, those entries will be recreated following a reboot but with default settings.

| Control Unit \| Control Unit | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | **Pre-3.2** ✗, **3.2+** ✓. |

- **Device Number**
  This is automatically allocated by the system.
- **Unit Type**
  The name of the device.
- **Version**
  The version of software running on each unit.
- **Serial Number**
  This is the number the system uses to tie a physical Control Unit to a device configuration (device number). For the control unit and WAN3 modules this is the MAC address. For a device connected to an Expansion port it is the Expansion port number plus 1.
- **Unit IP Address**
  This field shows the IP address for the control unit (LAN1) and if present, WAN3 expansion module.
- **Interconnect Number:**
  For external expansion modules this is the control unit expansion port used for connection. For other devices this is 0.
- **Module Number**
  For external expansion modules this is the control unit expansion port used for connection. For other devices this is  .

# Extension Settings

## Extension Form Overview

The IP Office supports both physical extensions and IP extensions.

By default, each extension is normally associated with a user and uses that user's directory number and other setting. Users with a login code can move between extensions by logging in and out, so the directory number is not a fixed property of the extension.

- **Physical Extensions**
  Physical extension ports are either integral to the IP Office control unit or added by the installation of an IP Office analog or digital phone expansion module. Extension entries are automatically created for each physical extension port within the system. These ports cannot be added or deleted manually.

  -  **Standard Telephone**
    An analog extension port (**PHONE** or **POT**) or an Avaya digital station port (**DS**) within the IP Office system.
  -  **Quiet Headset**
    Used for analog extension devices that are permanently off-hook and so should not receive dial-tone when not connected.
  -  **IVR Port**
    Used for analog ports connected to devices that require a specific disconnect clear signal at the end of each call.
  -  **Paging Speaker**
    An analog extension port set to be used as a paging speaker connection.

- **IP Extensions**
  These are used for H.323 IP phone devices or applications.

  -  **IP**
    This icon indicates an IP extension. For Avaya IP hardphones the IP extensions is either added manually or by the automatic detection the phone being connected, refer to the IP Office IP Phone Installation Manual. IP extensions can also be added manually to support a Phone Manager Pro PC Softphone or a third-party IP phone device. Note that third-party IP phone devices require entry of an IP End-Points license.
  -  **IP DECT**
    An extension port manually added to match extensions within an Avaya IP DECT system connected to the IP Office via an IP DECT line.

# Extension | Extn

This tab contains settings applicable to most types of extension.

| Extension \| Extn | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. *For IP Office 4.1+, **Base Extension** and **Disable Speakerphone** are mergeable. |

- **Extension ID**
  The physical ID of the extension port. Except for IP extensions, this settings is allocated by the system and is not configurable.
- **Base Extension:** *Range = 2 to 9 digits.*
  This is the directory number of the extension's default associated user. For IP Office 4.1+, changes to the **Base Extension** are mergeable.
  - For IP Office 4.0 and higher, following a restart the system will attempt to log on the last user associated with the extension. For pre-4.0 IP Office systems the extension's default associated user is logged on. In either case this does not occur it that user is set to **Force Login** (**User | Telephone**).
  - If another user logs onto an extension, when they log off, the extension returns to its default associated user unless they have logged on elsewhere or are set to **Force Login**.
  - In the US the **Base Extension** number is used for E911 calls. Any change to an extension's base extension must be matched by changes to the E911 adjunct database, see **E911 Overview**. The extensions default associated user should not be deleted.
  - Extensions associated with IP phones should not be given extension numbers greater than 7 digits.
  - Extension numbers in the range 8897 to 9999 are reserved for use by the IP Office Delta Server.
  - Users for Delta Server, CBC and CCC should only use up to 4 digit extension numbers.
- **Caller Display Type:** *Default = On.*
  Controls the presentation of caller display information. For digital extensions this value is fixed as **On**. Analog extension devices expect caller ID information either in DTMF signalling or FSK (Frequency Shift Keying) signalling, typically before ringing. See Caller Display.
  - **Off:** Disables caller display.
  - **On:** Enables caller display using the caller display type appropriate to the System Locale, see **Supported Country and Locale Settings**. If a different setting is required it can be selected from the list of supported options. For an analog extension connected to a fax server or other device that requires the pass through of DTMF tones, select ***DTMFF***.
  - **UK:** FSK before the first ring conforming to BT SIN 227. Name and number.
  - **UK20:** As per UK but with a maximum length of 20 characters. Name and number.
  - **DTMFA:** Caller ID in the DTMF pattern *A<caller ID>C*. Number only.
  - **DTMFB:** Caller ID in DTMF after call connection. Number only.
  - **DTMFC:** Caller ID in the DTMF pattern *A<caller ID>#*. Number only.
  - **DTMFF:** Sends the called number in DTMF after call connection. Used for fax servers. Number only.
  - **FSKA:** Variant of UK used for BT Relate 1100 phones. Name and number.
  - **FSKB:** ETSI specification with 0.25 second leading ring. Name and number.
  - **FSKC:** ETSI specification with 1.2 second leading ring. Name and number.

  - **FSKD:** Conforms to Belcore specification. Name and number.
  - **DTMFD:** Caller ID in the DTMF pattern *D<caller ID>C*. Number only.
  - All others are currently not used and default to matching UK.

- **Reset Volume after Calls:** *Default = Off.*
  Resets the phone volume after each call. This option is only supported for 4400 Series phones.
- **Device Type**
  This field is for information only. It indicates the type of phone connected to the extension port when the configuration was received from the IP Office.
- **Module**
  This field indicates the external expansion module on which the port is located. **BP** indicates an analogue phone extension port on the base or control unit. **BD** indicates a digital station (DS) port on the control unit. For the IP500 control unit, **BD** and **BP** is also followed by the slot number.
- **Port**
  This field indicates the port number on the **Module** indicated above.
- **Disable Speakerphone:** *Default = Off (Speakerphone enabled), Software level = 4.1+.*
  When selected, disables the fixed **SPEAKER** button if present on the phone using this extension port. An audible beep is sounded when a disabled **SPEAKER** button is pressed. Incoming calls such as pages and intercom calls are still connected but the speech path is not audible until the user goes off-hook using the handset or headset. Similarly calls made or answered using other buttons on the phone are not audible unless the user goes off-hook using the handset or headset. This field is not available for Analog and IP DECT extension ports. Currently connected calls are not affected by changes to this setting.

# Extension | Analog

This tab contains settings that are applicable to analog extensions. These extensions are provided by ports marked as **POT** or **PHONE** on IP Office control units and expansion modules.

| **Extension \| Analog** | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

- **Equipment Classification:** *Default = Standard Telephone*
  Only available for analog extension ports. Note that changing this settings should be followed by a system reboot.
  - **Quiet Headset**
    This option is used with analog extensions where the handset is permanently off-hook and so audio is only desired when a call is connected, for example the handset has been replaced with a headset. This option is typically also used in conjunction with the **User | Telephony Offhook Station** setting in order to make and receive calls using an application such as Phone Manager. Between calls the audio path is silent. Ringing is presented in the audio path. Caller ID is not supported on the phone
  - **Paging Speaker**
    Used for analog ports connected to a paging amplifier. This extension will present busy and cannot be called or be used to make calls. It can only be accessed using dial paging features.
  - **Standard Telephone**
    Use for normal analog phones.
  - **Door Phone 1/Door Phone 2**
    These two options are currently not used and so are grayed out.
  - **IVR Port**
    Used for analog ports connected to devices that require a disconnect clear signal (ie. a break in the loop current) at the end of each call. When selected the **Disconnect Pulse Width** is used. For pre-3.2 IP Office systems, this option was only supported on systems with the locale set to ***United States*** or ***Saudi Arabia***.
- **Message Waiting Lamp Indication Type:** *Default = None*
  Allows the selection of the message waiting indication (MWI) mode for analog and IP DECT extensions.
  - For control unit and Phone V1 module analog extensions, the options **None**, **On**, **51V Stepped**, **81V**, ***Line Reversal A*** and ***Line Reversal B*** are available. **On** defaults the MWI using the system locale,.

| 'On' | 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 Phone V2 module extensions, the additional option **101V** is available.
- **Hook Persistency:** *Default = 100ms, Range = 50 to 255ms.*
  Defines the time frame (in milliseconds) in which the system will wait before determining that the phone is off-hook.

- **Flash Hook Pulse Width**
  The following options are only available for analog extension ports. They define the length of loop break that will be considered a time break recall (TBR) signal.
  - **Use System Defaults:** *Default = Selected (On)*
    Use the default values appropriate to the system's locale. See **Appendix A: Locale Settings**.
  - **Minimum Width:** *Range = 0 to 99 x 10ms.*
    Minimum hook flash length used if **Use System Defaults** is not selected. Shorter breaks are ignored a glitches.
  - **Maximum Width:** *Range = 0 to 255 x10ms***.**
    Maximum hook flash length used if **Use System Defaults** is not selected. Longer breaks are treated as clearing.
- **Disconnect Pulse Width:** *Default = 0ms, Range = 0 to 100ms*
  This setting is used with analog extensions where the **Equipment Classification** has been set to ***IVR Port***. It sets the length of loop current break used to indicate call clearing.

# Extension | VoIP

This tab is only available for IP extensions.

The following are the recommended settings for Avaya IP extensions.

| IP Extension | 3600/4600/5600 Series IP Phones | Phone Manager Pro PC Softphone |
|---|---|---|
| Silence Suppression | Off | Off |
| Enable Faststart | Off | On |
| Local Hold Music | Off | Off |
| Local Tones | Off | Off |

| Extension \| VoIP | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

- **IP Address:** *Default = 0.0.0.0*
  The IP address of the H323 terminal. The default entry accepts connection to any address.
- **MAC Address:** *Default = 000000000000 (Grayed out)*
  This field is grayed out and not used.
- **Voice Payload Size**
  This is the length of time represented by each VoIP packet in milliseconds. This settings is automatically defaulted to match the selected **Compression Mode** selected. Values are G.711/G.729 = 20ms, G.723=30ms. For T3 IP terminals the value 10ms is shown.
- **Compression Mode:** *Default = Automatic Selection*
  This defines the compression method to be used for this extension.
  - **Automatic Selection**
    During call setup the IP Office negotiates the compression mode using the following order of preference: **G729a**, **G.723.1**, **G711 ALAW**, **G711 ULAW**.
  - If required a specific codec can be selected from: ***G.711 ALAW 64K***, ***G.711 ULAW 64K***, ***G.729(a) 8K CS-ACELP***, ***G.723.1 6K3 MP-MLQ***. If during connect with a specific codec, connect fails, the IP Office will fallback to using automatic selection.
- **Gain:** *Default = Default, Software level* = 3.0+.
  Allows adjustment of the received volume. The gain is selectable from -31dB to +31dB in 1 dB increments.
- **H450 Support:** *Default = H450, Software level = 4.0+.*
  Selects the supplementary service signaling method for use with non-Avaya IP devices. Options are ***None***, ***QSIG*** and ***H450***. From h450, hold and transfer are supported. Note that the selected method must be supported by the remote end. None should be used with Avaya IP phones and PC Softphone ***None*** should be used.
- **VoIP Silence Suppression:** *Default = Off*
  When selected H.323 terminals will not send data if they are silent, this is useful when optimizing data traffic.
- **Enable Faststart for non-Avaya IP Phones:** *Default = Off*
  A fast connection procedure. Reduces the number of messages that need to be exchanged before an audio channel is created. Faststart should not be used with Avaya 4600 and 5600 series IP phones.
- **Fax Transport Support:** *Default = Off*
  When selected, this option provides support for faxing over a H.323 connection to another IP Office with the same setting.

- **Out of Band DTMF:** *Default = On*
  When on, DTMF is sent as a separate signal ("Out of Band") rather than as part of the encoded voice stream ("In Band"). The "Out of Band" signaling inserted back into the audio by the remote end. This is recommended for low bit-rate compression modes such as G.729 and G.723 where DTMF in the voice stream can become distorted.
- **Local Tones:** *Default = Off*
  When selected, the H.323 terminals generate their own tones. This option is not supported by Avaya IP phones and Phone Manager Pro PC Softphone.
- **Enable RSVP:** *Default = Disabled (Grayed out)*
- **Allow Direct Media Path:** *Default = On*
  This settings controls whether H323 calls must be routed via the H323 gatekeeper (the IP Office) or can be routed alternately if possible within the network structure.
  - If enabled, H323 calls can take routes other than through the IP Office. This removes the need for a voice compression channel. Both ends of the calls must support Direct Media. Enabling this option may cause some vendors problems with changing the media path in mid call. Pre-IP Office 4.0, when using direct media path, it is not always possible for the extension to be recorded or monitored.
  - If disabled or not supported one end of the call, all call are routed via the IP Office.
    - On pre-4.0 these calls would then require a voice compression channel even if the IP devices use the same audio codec.
    - On IP Office 4.0 and higher, RTP relay support allows calls between devices using the same audio codec to not require a voice compression channel. T3 IP phones must be configured to 20ms packet size to use RTP relay.
    - With the IP Office 4.0 Q2 2007 maintenance release, previous restrictions on T3 IP phones using direct media were removed. The phone must have firmware T246 or higher as provided on the IP Office Administrator Applications CD.
- **VPN Phone Allowed:** *Default = Off, Software level = 4.1+.*
  Indicates that the extension can be used with Avaya IP Office VPNremote Phone firmware. The phones must be licensed using a **VPN IP Extensions** licenses added to the IP Office configuration for the number of VPN IP phones required. Note that the field is greyed out if there are no available licenses.

# Extension | IP DECT

This tab is displayed for IP DECT extensions. These are created manually after a IP DECT line has been added to the configuration. They should match the extensions configured on the IP DECT system. Refer to the IP Office IP DECT Installation manual.

| Extension \| IP DECT | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✗, **3.0DT** ✗, **3.0** ✗, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

- **DECT Line ID**
  Use the drop-down list to select the IP DECT line from the IP Office to the Avaya IP DECT system.
- **Message Waiting Lamp Indication Type:** *Default = On*
  Allows selection of the message waiting indication to use with the IP DECT extension. Options are: ***None***, ***On***.

# User Settings

## User Form Overview

Users are the people who use the IP Office system. They do not necessary have to be an extension user, for example users are used for RAS dial in data access. In addition, more users can be created than there are extensions, with users logging on to an extension when they want to receive calls.

By default, a user is automatically created to match each extension. They are numbered from 201 upwards and the first 16 are placed in the hunt group Main (200), which is the default destination for incoming calls.

- **Standard User:** A standard user.
- **No User:** Used to apply settings for extensions which currently have no associated user.
- **Remote Manager:** Used as the default settings for dial in user connections.
- **Hot Desking User:** Users with a **Login Code** can move between extensions by logging on and off.

**When a user is deleted**
When a user is deleted, any calls in progress continue until completed. The ownership of the call is shown as the NoUser user. Merging the deletion of a user causes all references to that deleted user to be removed from the system.

**Changing a user's extension**
Changing a user's extension automatically logs the user out of their current extension and into their new extension provided that this new extension exists and the user doesn't have Forced Login enabled. If Forced Login is enabled, then the user remains on the current extension being used until they logs out and login at the new extension.

Note that changing a user's extension number affects the user's ability to collect Voicemail messages from their own extension. Each user's extension is set up as a "trusted location" under the Source Numbers tab of the User configuration form. This "trusted location" allows the user to dial *17 to collect Voicemail from his own extension. Therefore if the extension number is changed so must the "trusted location".

The following related configuration items are automatically updated when a user extension is changed:

- User, Coverage and Bridged Appearance buttons associated with the user.
- Hunt group membership (disabled membership state is maintained).
- Forwards and Follow Me's set to the user as the destination.
- Incoming call routes to this destination.
- Dial in source numbers for access to the user's own voicemail.
- Direct call pickup buttons are updated.
- The extension number of an associated extension is updated.

**Creating a User Rights Based on an Existing User**

1. Select **User Rights**.
2. In the group pane, right-click and select **New User Rights from a User**.
3. Select the user and click **OK**.

**Associating User Rights to a User**

1. Select **User Rights** or **User**.
2. In the group pane, right-click and select **Apply User Rights to Users**.
3. Select the user rights to be applied.
4. On the **Members of this User Rights** sub tab select the users to which the user rights should be applied as their **Working Hours User Rights**.
5. On the **Members when out of hours** sub tab select which users should use the selected user rights as their out of hours user rights.
6. Click **OK**.

or

1. Select the required user to display their settings in the details pane.
2. Select the **User** tab.
3. Use **Working Hours User Rights** drop-down to select the user rights required.
4. If required a **Working Hours Time Profile** and **Out of Hours User Rights** can be selected.
5. Click **OK**.

**Copy User Rights Settings over a User's Settings**
This process replaces a user's current settings with those that are part of the selected user rights. It does not associate the user with the user rights.

1. Select **User Rights**.
2. In the group pane, right-click and select **Copy user rights values to users**.
3. Select the user rights to be applied.
4. Click **OK**.

# User | User

Users are the people who use the system or are Dial In users for data access. A system User may or may not have an Extension Number that physical exists - this is useful if users do not require a physical extension but wish to use system features, for example voicemail, forwarding etc.

- **No User** is used to apply settings to extensions which have no associated user.
- **Remote Manager** is used as the default settings for dial in connections.

A symbol indicates that in IP Office 3.2 the settings can be set and locked by the user's associated user rights.

| User \| User | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Name:** *Range = Up to 15 characters*
  This is the user's account name used for RAS Dial In, Caller Display and voicemail mailbox. As the display on Caller Display telephones is normally only 16 digits long it is useful to keep the name short. Only alphanumeric characters and space are supported in this field. Do not use punctuation characters such as #, ?, /, -, _ and ,. Do not start names with a numeric character. This field is case sensitive and must be unique.
  - Voicemail uses the name to match a user to their mailbox. Changing a user's name will route their voicemail calls to a new mailbox. Note however that Voicemail Pro will treat names such as "Steve Smith", "steve smith" and "STEVE SMITH" as being the same.
- **Password:** *Default = Blank*
  This password is used for user's with Dial In access. It is also used for Phone manager, SoftConsole and TAPI. This is not their voicemail mailbox password, that is set through the **User | Voicemail** tab.
- **Full Name:** *Default = Blank*
  Use this field to enter the entire user's name. When set, the **Full Name** is used in place of the **Name** for display by phones and IP Office user applications. Only alphanumeric characters and spaces are supported in this field. Do not use punctuation characters such as #, ?, /, -, _ , ^, > and ,. The entry in this field also cannot start with either a space or a numeral
- **Extension:** *Range = 2 to 9 digits.*
  Any number up to 9 digits. In general all extensions should have the same number of digits. This setting can be left blank for users used just for dial in data connections.
  - Users for Delta Server, CBC and CCC should only use up to 4 digit extension numbers.
  - Users associated with IP phones or who may log in as such devices should not be given extension numbers greater than 7 digits.
  - Extension numbers in the range 8897 to 9999 are reserved for use by the IP Office Delta Server.
- **Locale:** *Default = Blank (Use system locale)*
  Configures the language used for voicemail prompts played to the user, assuming the language is available on the voicemail server. See **Supported Country and Locale Settings**. On a digital extension it also controls the display language used for messages from the IP Office. Note however that some phones have their own menu options for the selected language for the phone menus.

- **Priority:** *Default = 5, Range = 1 (Lowest) to 5 (Highest)*
This setting is used by Least Cost Routing to determine what happens when the number being called by the user matches an LCR short code set to the Busy feature. If the user's priority is higher than the priority of the LCR tab on which the match occurred, LCR will look for an alternate match on the next LCR tab. If an alternate match is found it is used, otherwise the user receives busy. User's whose priority is not higher than the LCR tabs always receive busy.
- **Restrictions:** *Default = None, Software level = Up to 3.1.*
Sets which set of User Restrictions applies to the user. See User Restrictions.
- **Phone Manager Type:** *Default = Lite, Software level = Up to 3.1*.
Determines the mode in which the user's copy of the Phone Manager application will operate. Modes are **Lite**, **Pro** and **VoIP** (Phone Manager Pro PC Softphone). Note that the number of users able to simultaneously use Pro and VoIP modes is controlled by licenses entered into the IP Office configuration. *In IP Office 3.2 configurations this option has moved to the **Phone Manager Options** tab.
- **Book a Conferencing Center in Phone Manager:** *Default = Off, Software level = Up to 3.1.*
When enabled, displays links in the user's Phone Manager application for access to the IP Office Conferencing Center application if installed. Note that to book a conference requires the user to have a Conferencing Center user ID and password. This feature also requires the **Conference Center IP Address** and **Conference Center URL** to be set (**System | System**). *In IP Office 3.2 configurations this option has moved to the **Phone Manager Options** tab.
- **Ex Directory:** *Default = Off*
When on, the user does not appear in the directory list shown by the Phone Manager application and on phones with a directory function. The user is still visible in other applications such as the SoftConsole directory.
- **User Rights View:** *Software level = 3.2+.*
This field affects Manager only. It allows you to switch between displaying the user settings as affected by their associated **Working Hours User Rights** or **Out of Hours User Rights**.
- **Working Hours Time Profile:** *Default = <None> (Continuous), Software level = 3.2+.*
If set, the selected time profile defines when the user's **Working Hours User Rights** are applied. Outside the time profile, the user's **Out of Hours User Rights** are applied.
- **Working Hours User Rights:** *Default = Blank (No rights restrictions), Software level = 3.2+.*
This field allows selection of user rights which may set and lock some user settings. If a **Working Hours Time Profile** has been selected, the **Working Hours User Rights** are only applied during the times defined by that time profile, otherwise they are applied at all times.
- **Out of Hours User Rights:** *Default = Blank (No rights restrictions), Software level = 3.2+.*
This field allows selection of alternate user rights that are used outside the times defined by the user's **Working Hours Time Profile**.

# User | Voicemail

If a voicemail server application is being used on your system, each user has use of a voicemail mailbox. You can use this form to enable this facility and various user voicemail settings.

A 🔒 symbol indicates that in IP Office 3.2 the settings can be set and locked by the user's associated user rights.

| User \| Voicemail | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Voicemail Code:** *Default = Blank, Range = 0 (no code) to 15 digits.*
  A code (1-15 digits) used by the voicemail server to validate access to this mailbox. If remote access is attempted to a mailbox that has no voicemail code set, the prompt *"Remote access is not configured on this mailbox"* is played. Whether the caller will be prompted to enter this code varies as follows:
  - **Embedded Voicemail**
    The voicemail code is used if set.
  - **Trusted Source Access**
    The voicemail code is required when accessing the mailbox from a location that is not set as a trusted number in the user's **Source Numbers** list. Also Voicemail Pro call flows containing an action where the action's PIN code set to $ will prompt the user for their voicemail code.
  - Users can set their own code through the mailbox telephone user interface. In this case the user is forced to enter at least 4 digits.
  - Codes set through the Voicemail Pro telephone user interface are restricted to valid sequences. For example, attempting to enter a code that matches the mailbox extension, repeat the same number (1111) or a sequence of numbers (1234) are not allowed. If these types of code are required they can be entered through Manager.
- **Voicemail Email:** *Default = Blank*
  When a new voicemail message is received by the user, the WAV file created can be sent to an email account. This field is used to store the user's email address. Whether to send the email is set through the Voicemail Email options below. This address is also used for Voicemail Email Reading if enabled and for Phone Manager instant messaging. This setting is not used for IMS operation.
- **Voicemail On:** *Default = On* 🔒
  This feature enables the user's voicemail mailbox to answer calls which ring unanswered or arrive when the user is busy. It does not disable the voicemail mailbox being used as the target for other functions such as call recording or messages forwarded from other mailboxes.
- **Voicemail Help:** *Default = Off*
  For voicemail systems running IP Office mailbox mode, this option controls whether users retrieving messages are automatically given an additional prompt *"For help at any time press 8."* If switched off, users can still press 8 for help. For voicemail systems running in Intuity emulation mode, this option has no effect. On those systems the default access greeting always includes the prompt *"For help at any time, press *4"* (*H in the US locale).
- **Voicemail Ringback:** *Default = Off* 🔒
  When enabled and a new message has been received, the voicemail server calls the user's extension to attempt to deliver the message each time the telephone is put down. Voicemail will not ring the extension more than once every 30 seconds.

- **Voicemail Email Reading:** *Default = Off*
  When you log into you voicemail box, it will detect your email messages and read them to you. This email text to speech feature is set-up through Voicemail Pro.
- **Voicemail Email Mode:** *Default = Off*
  Controls the method of operation of Voicemail Email above. These settings are not used by IMS.
  - **Off**
    Do not automatically send a new message to the email account
  - **Copy**
    Copy all messages to the email account
  - **Forward**
    Forward all messages to the email account and delete from the Voicemail Server.
  - **Alert**
    Send an email message without attaching the Voicemail file. This may be used with Email gateways to Pagers or Mobile telephone Short Message Services. Includes the caller's **Caller ID** if available.
- **Reception / Breakout (DTMF 0):** *Default = Blank*
  When connected to a user's voicemail, the caller can press 0 (*0 on Embedded Voicemail) to be transferred to either an internal number or external number. Enter here the telephone number to be used. The user should announce this facility in their greeting message, for example *"John Smith is not available today, you may leave a message or press 0 for Reception"*.
- **Breakout (DTMF 2/3):** *Default = Blank, , Software level = 3.2+.*
  These two fields are supported by Voicemail Pro (Intuity mode) on IP Office 3.2+ and Embedded Voicemail on IP Office 4.0+. They allow users to have two further breakout numbers in addition to the **Reception** number above. These two additional breakouts are triggered by the caller pressing 2 or 3 (*2 or *3 on Embedded Voicemail). These additional numbers allow a simple personal auto attendant to be created. The options setup should be included in the user's mailbox greeting.

# User | DND

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

The user can switch do not disturb on/off using short codes, pre-programmed keys on their phone or IP Office Phone Manager application. See Do Not Disturb in the **Telephone Features** section.

A 🔒 symbol indicates that in IP Office 3.2 the settings can be set and locked by the user's associated user rights.

| User \| DND | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Do Not Disturb:** *Default = Off* 🔒
  When checked the user's extension is considered busy, except for calls coming from sources listed in their **Do Not Disturb Exception List**. When a user has do not disturb in use, their normal extension will give alternate dialtone when off hook.
- **Do Not Disturb Exception List:** *Default = Blank*
  This is the list of telephone numbers that are still allowed through when Do Not Disturb is set. For example this could be an assistant or an expected phone call. Internal extension numbers or external telephone numbers can be entered. If you wish to add a range of numbers, you can either enter each number separately or make use of the wildcards  and "X" in the number. For example, to allow all numbers from 7325551000 to 7325551099, the DND Exception number can be entered as either 73255510XX or 73255510N.

# User | Short Codes

Short codes entered in this list can only be dialed by the user. They will override any matching user rights or system short code. See Short Codes for details.

- **WARNING**
  User dialing of emergency numbers must not be blocked by the addition of short codes. If short codes are added, the users ability to dial emergency numbers must be tested and maintained.

A symbol indicates that in IP Office 3.2 the settings can be set and locked by the user's associated user rights.

| User \| Short Codes | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

Short codes can be added and edited using the **Add**, **Remove** and **Edit** buttons. Alternatively you can right-click on the list of existing short code to add and edit short codes.

- ***FWD**
  Short codes of this form are inserted by the IP Office. They are used in conjunction with the User | Forwarding settings to remember previously used forwarding numbers. They can be accessed on that tab by using the drop-down selector on the forwarding fields.

# User | Source Numbers

This form is used to enter values that have special usages. These are entered using the **Add**, **Edit** or **Remove** buttons.

| **User \| Source Numbers** | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

The following types of entry can be added to a user's source numbers.

- **V<*Caller's ICLID*>**
  For systems using Voicemail Lite or Voicemail Pro, strings prefixed with a **V** indicate numbers from which access to the users mailbox is allowed without requiring entry of the mailbox's voicemail code. This is referred to as "trusted source".
    - For Voicemail Pro running in Intuity mode, trusted source is used for calls from programmable buttons set to Voicemail Collect and Visual Voice. Other controls are prompted for the mailbox number and then password.
- **R<*Caller's ICLID*>**
  To allow data access only from a specified number prefix the number with a "R", for example ***R7325551234***.
- **H<*Group Name*>**
  Allows the user to receive message waiting indication of new group messages. Enter **H** followed by the group name, for example **HMain**.
    - On suitable display extensions, the hunt group name and number of new messages is displayed. Refer to the appropriate telephone user guide.
    - If the user is using Phone Manager, the Messages tab shows the hunt group name and number of new messages.
    - If the user is not a member of the group, a voicemail code must be set for the group's mailbox. See **Voicemail Code** on the **Hunt Group | Voicemail** tab.
- **P<*Telephone Number*>**
  This entry sets the destination for callback (outbound alert) calls from voicemail. Enter **P** followed by the telephone number including any necessary external dialing prefix, for example ***P917325559876***. This facility is only available when using Voicemail Pro through which a default Callback or a user specific Callback start point has been configured. Refer to the Voicemail Pro documentation. This feature is separate from voicemail ringback.
- **RESERVE_LAST_CA=** *Software level* = Up to 3.2.
  Used for users with multiple call appearance buttons. When present, this string stops the users last call appearance button from being used to receive incoming calls. This ensures that the user always has a call appearance button available to make outgoing calls and to initiate transfers and conferences. For IP Office 4.0 and higher this option has been replaced by a **Reserve Last CA** option on the **User | Telephony** tab.
- **AT<string>**
  Strings beginning with **AT** can be used with a user called ***DTEDefault*** to configure the default settings of the IP Office control unit's DTE port.

**NoUser User Source Numbers**

The following source numbers can also be used on the Source Numbers tab of the ***NoUser*** user. These affect all users on the system. Note that these source numbers typically require a reboot of the IP Office system to become effective.

- **H323SetupTimerNoLCR** Software level = 3.2 only.
  Used to set the fallback time from VoIP trunks to non-VoIP trunks within LCR. See **IP Trunk Fallback**.
- **LONGER_NAMES**
  Used to increase the length of names sent for display on DS phones. See Caller Display.
- **HIDE_CALL_STATE**
  Used to hide the call status information, for example ***Dial***, ***Conn***, etc, on DS phones. Used in conjunction with the **LONGER_NAMES** option.
- **ACD_QUEUE_DELAY=*nn*** *Software level = Up to 3.2.*
  Used to change the timeout for still queued messages. The parameter ***nn*** can be replace with a time in seconds between 20 and 180. For IP Office 4.0+ this has been replaced by **Hunt Group | Announcements**.
- **ProgressEndsOverlapSend**
  See **Line | VoIP**.
- **VM_TRUNCATE_TIME=*X*** *(Range X = 0 to 7 seconds, Software level = 4.0+)*
  On analog trunks, call disconnection can occur though busy tone detection. When such calls go to voicemail to be recorded or leave a message, when the call ends the IP Office indicates to the voicemail system how much to remove from the end of the recording in order to remove the busy tone segment. This amount varies by system locale and the defaults are listed below. For some systems it may be necessary to override the default if analog call recordings are being clipped or include busy tone.  That can be done by adding a **VM_TRUNCATE_TIME=** setting with a value from 0 to 7 seconds.
  - **New Zealand, Australia, China and Saudi Arabia:** 5 seconds.
  - **Korea:** 3 seconds.
  - **All other locales:** 7 seconds.
- **SHOW_LINEID_NOT_OUTSIDE** *(Software level = 4.0 Q2 2007 Maintenance release+)*
  By default, for calls where no incoming caller ID (ICLID) information is available, the IP Office inserts the word   wherever ICLID information is normally displayed. The NoUser source number value **SHOW_LINEID_NOT_OUTSIDE** can be used to make available within the configuration for each trunk and channel **Line Name** and **Channel Name** fields. The text entered into those fields is then used with external calls without ICLID information.
  - This feature is not used with SIP, IP DECT, E1R2 and S0 lines.
  - On T1 lines, a **Name** field is also made available for individual channels and if set overrides the line name field.
  - This feature does not override the display of ***Withheld*** if the caller has withheld their ICLID information.
  - Where a **Name** is entered, that value is used to identify calls with no ICLID information.
  - For line appearance buttons, if set the **Name** replaces the **Line Appearance ID** as the default button label.
- **DISTINCT_HOLD_RINGBACK** *(Software level = 4.1+)*
  Used to display a specific message about the call type for calls returning after timing out from being parked or held. If set, such call display **Return Call - Held** or **Return Call – Parked** rather than connected party name or line name.

- **ExtendDirectLimit <*optional limit*>** *(Software level = 4.1+)*
  This command allows the number of directory entries that can be added to the configuration to be controlled. By default Manager imposes a limit of 1000 directory entries. If this command is used without specifying an optional limit, the limit set is determined by the type of IP Office control unit; refer to the table below. Note that specifying or allowing a large directory limit may affect the performance of applications that use and display the directory, for example IP Office Phone Manager.
- **ExtendLDAPDirectLimit <*optional limit*>** *(Software level = 4.1+)*
  This command allows the number of LDAP directory entries that the IP Office will read to be controlled. By default the IP Office will only support up to 500 LDAP directory entries. If this command is used without specifying an optional limit, the limit set is determined by the type of IP Office control unit; refer to the table below. Note that specifying or allowing a large directory limit may affect the performance of applications that use and display the directory, for example IP Office Phone Manager.

| Control Unit | IP Office Directory Entries | LDAP Entries |
|---|---|---|
| **Small Office Edition** | 100 | 1000 |
| **IP406 V2** | 2500 | 10000 |
| **IP412** | 10000 | 10000 |
| **IP500** | 10000 | 10000 |

# User | Telephony

This form allows you to set telephony related features for the user. These override any matching setting in the **System | Telephony** tab. For details of the ringing tones, see Ring Tones. ***DefaultRing*** uses the system default setting set through the **System | Telephony** tab.

A 🔒 symbol indicates that in IP Office 3.2 the settings can be set and locked by the user's associated user rights.

| **User \| Telephony** | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Outside Call Sequence:** *Default = Default Ring (Use system setting)*
  Applies only to analog phones. Sets the ring pattern used for external calls to the user. The distinctive ring patterns used for other phones are fixed. Note that changing the pattern for users associated with fax and modem device extensions may cause those devices to not recognize and answer calls.
- **Inside Call Sequence:** *Default = Default Ring (Use system setting)*
  Applies only to analog phones. Sets the ring pattern used for internal calls to the user. The distinctive ring patterns used for other phones are fixed.
- **Ring Back Sequence:** *Default = Default Ring (Use system setting)*
  Applies only to analog phones. Sets the ring pattern used for ringback calls to the user. The distinctive ring patterns used for other phones are fixed.
- **No Answer Time:** *Default = Blank (Use system setting), Range = 1 to 99999 seconds.* 🔒
  Sets how long a call rings the user before following forwarded on no answer if set or going to voicemail. Leave blank to use the system default setting.
- **Wrap-up Time (secs):** *Default = 2 seconds, Range 0 to 99999 seconds.* 🔒
  Specifies the amount of time after ending one call before another call can ring. During this interval the user is treated as still being on a call. You may wish to increase this in a "call center" environment where users may need time to log call details before taking the next call. It is recommended that this option is not set to less than the default of 2 seconds. 0 is used to allow immediate ringing.
- **Transfer return Time (secs):** *Default = Blank (Off), Range 1 to 99999 seconds.* 🔒
  Sets the delay after which any call transferred by the user, which remains unanswered, should return to the user. Note that pre-3.2 IP Office the recall only occurs if the user has no other connected call. For IP Office 3.2+ transfer returns will occur if the user has available call appearance buttons. Recalled calls will continue ringing and do not follow forwards or go to voicemail.
- **Individual Coverage Time (secs):** *Default = 10 seconds, Range 1 to 99999 seconds, Software level 3.0+.* 🔒
  This function sets how long the phone will ring at your extension before also alerting at any call coverage users. This time setting should not be equal to or greater than the **No Answer Time**.
- **Login Code:** *Default = Blank, Range = Up to 31 digits.*
  The code that has to be entered, as part of the login sequence, to allow a user to make use of any telephone as if it was their phone. This login code can be used for hot desking as well as logging back onto your phone after it has been used by a hot desker. This entry must be at least 4 digits for DS port users. Login codes of up to 15 digits are supported with **Extn Login** buttons. Login codes of up to 31 digits are supported with **Extn Login** short codes.
  - IP Office 4.0+ supports hot desking between IP Office systems. To log on at a remote IP Office requires that IP Office to have a **Advanced Small Community Networking** license.

- **Login Idle Period (secs):** *Default = Blank (Off), Range = 0 (Off) to 99999*
  If the telephone is not used for this period; the user currently logged in is automatically logged off. This option should be used only in conjunction with **Force Login** (see below).
- **Monitor Group:** *Default = <None>*
  Sets the Hunt Group whose members the user can monitor if silent monitoring is setup. See Call Listen.
- **Call Cost Mark-Up:** *Default = 100, Software level = 4.0+.*
  This setting is used for ISDN advice of charge (AOC). The markup is applied to the cost calculations based on the number of units and the line base cost per charging unit. The field is in units of 1/100th, for example an entry of 100 is a markup factor of 1. This value is included in the IP Office SMDR output.
- **Status on No Answer:** *Default = Logged On, Software level = 4.0+.*
  Hunt groups can change the status of call center agents (users with a login code and set to forced login) who do not answer a hunt group call presented to them before it is automatically presented to the next agent. Use of this is controlled by the **Agent's Status on No Answer Applies To** setting of the hunt group.
  - **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.
- **Multi-Line Options**
  These options are used when the user is associated with a phone that has multiple call appearance buttons.
  - **Ring Delay:** *Default = Blank (Use system setting), Range = 0 (use system setting) to 98 seconds, Software level = 3.2+.*
    This setting is used when any of the user's programmed appearance buttons is set to Delayed ringing. Calls received on that button will initially only alert visually. Audible alerting will only occur after the ring delay has expired.
  - **Ringing Line Preference:** *Default = On, Software level = 3.0+.*
    For users with multiple appearance buttons. When the user is free and has several calls alerting, ringing line preference assigns currently selected button status to the appearance button of longest waiting call. Ringing line preference overrides idle line preference.
  - **Idle Line Preference:** *Default = On, Software level = 3.0+.*
    For users with multiple appearance buttons. When the user is free and has no alerting calls, idle line preference assigns the currently selected button status to the first available appearance button.

  - **Delayed Ring Preference:** *Default = Off, Software level = 4.0+.*
    This setting is used in conjunction with appearance buttons set to delayed or no ring. It sets whether ringing line preference should use or ignore the delayed ring settings applied to the user's appearance buttons.
    - When on, ringing line preference is only applied to alerting buttons on which the ring delay has expired.
    - When off, ringing line preference can be applied to an alerting button even if it has delayed ring applied. This is the same as pre-4.0 ringing line preference operation.
  - **Answer Pre-Select:** *Default = Off, Software level = 4.0+.*
    Normally when a user has multiple alerting calls, only the details of the call on current selected button are shown. Pressing any of the alerting buttons will answer the call on that button, going off-hook will answer the current selected button. Enabling **Answer Pre-Select** allows the user to press any alerting button to make it the current selected button and displaying its call details without answering that call until the user either presses that button again or goes off-hook. Note that when both **Answer Pre-Select** and R**inging Line Preference** are enabled, once current selected status is assigned to a button through ringing line preference it is not automatically moved to any other button.
- **Reset Longest Idle Time:** *Default = All Calls, Software level = 4.0+.*
  This setting is used in conjunction with hunt groups set to Longest Waiting (also known as Idle and Longest Waiting). It defines what type of calls reset the idle time of users who are members of these hunt groups. Options are **All Calls** and **External Incoming**.
- **Call Waiting On:** *Default = Off*
  For users on phones without appearance buttons, if the user is on a call and a second call arrives for them, an audio tone can be given in the speech path to indicate a waiting call (the call waiting tone varies according to locale). The waiting caller hears ringing rather than receiving busy. There can only be one waiting call, any further calls receive normal busy treatment. If the call waiting is not answered within the no answer time, it follows forward on no answer or goes to voicemail as appropriate. User call waiting is not used for users on phones with multiple call appearance buttons. Call waiting can also be applied to hunt group calls, see ***Hunt Group | Hunt Group | Call Waiting***.
- **Answer Call Waiting on Hold (Analog):** *Default = On*
  Applies to analog and IP DECT extension users only. If the user has a call waiting and places their current call on hold, the call waiting is automatically connected.
- **Busy on Held:** *Default = On*
  If on, when the user has a call on hold, new calls receive busy treatment. They will follow the users forward on busy setting or are diverted to voicemail. Otherwise busy tone (ringing for incoming analog calls) is played. This overrides call waiting when the user has a call on hold. The use of Busy on Held for users with multiple call appearance buttons is deprecated and Manager will prompt whether it should switch off busy on held for such a user.
- **Outgoing Call Bar:** *Default = Off*
  When enabled, this setting stops a user from making any external calls. On most Avaya display phones, this causes a **B** to be displayed.
- **Offhook Station:** *Default = Off*
  If on, the user's extension can be taken on/off-hook by applications such as Phone Manager. Only use this setting if the user's main telephone is capable of handsfree speech or is an analog extension setup as a **Quite Headset** (**Extension | Analog**). On pre-4.0 systems, calls to a busy destination are cleared immediately without hearing busy tone. On 4.0+ calls to a busy destination present busy tone before being cleared.

- **Can Intrude:** *Default = Off*
  Check this option if the User can interrupt other user's calls. This setting and the setting below are used to control the use of the following short code and button features: **Call Intrude**, **Call Listen**, **Call Steal** and **Dial Inclusion**.
- **Cannot be Intruded:** *Default = On (Pre-4.0 - Off in Italy)*
  If checked, this user's calls cannot be interrupted or acquired. In addition to the features listed above, this setting also affects whether other users can use their appearance buttons to bridge into a call to which this user has been the longest present user.
- **Force Login:** *Default = Off*
  If checked, the user must login using their **Login Code** to use any extension including an extension to which they are the default associated user (Base Extension). For example, if **Force Login** is ticked for user A and user B has logged onto A's phone, when B logs off user A is not automatically associated with their normal phone and instead must log back on. If Force Login was not ticked, A would be automatically logged back on.
  - Note that users with a **Login Code** and set to **Forced Login** are treated as call center agents. These users consume CCC agents licenses and their status is reported within CBC and CCC applications.
- **Force Account Code:** *Default = Off*
  If checked, the user must enter a valid account code to make an external call.
- **Force Authorization Code:** *Default = Off, Software level = 3.2+.*
  If checked, the user must enter a valid authorization code to make an external call. That authorization code must be one associated with the user or the user rights to which the user belongs. See **Authorization Codes**.
- **Can Accept Collect Calls:** *Default = Off [Brazil Only]*
  Determines whether the user is able to receive and accept collect calls.
- **System Phone:** *Default = Off*
  Allows the user, on DS phones with a **Menu** key, to alter the date and time displayed on the phones. On those phones it is accessed by pressing **Menu | Menu | Func | Setup**. Also allows SoftConsole users to use the SoftConsole's **Send Message** function.
  - IP Office 4.0 Q2 2007 maintenance release allows users with this setting enabled to use a short code to set the IP Office system date and time. Refer to Date and Time in the **Telephony Features** section.
  - IP Office 4.1+ allows users with this setting enabled to use a programmable button to set the time, date and to view the IP Office system type and version.
- **Inhibit Off-Switch Transfers:** *Default = Off, Software level* = 3.2+*.*
  When enabled, this setting stops the user from transferring or forwarding calls externally. Note that all user can be barred from forwarding or transferring calls externally by the **System | Telephony | Inhibit Off-Switch Transfers** setting.
- **Reserve Last CA:** *Default = Off, Software level = 4.0+.*
  Used for users with multiple call appearance buttons. When present, this string stops the users last call appearance button from being used to receive incoming calls. This ensures that the user always has a call appearance button available to make outgoing calls and to initiate transfers and conferences. For pre-4.0 IP Office this option is set by adding the **RESERVE_LAST_CA=** option on the **User | Source Numbers** tab.
- **Can Trace Calls:** *Default = Off, Software level = 4.0+.*
  This settings controls whether the user is able to make used of ISDN MCID controls.
- **Remote Homeworker/Agent:** *Default = Off, Software level = Up to 3.2 only.*
  Select if the user has been configured as a remote extension on an Avaya INDeX telephone system. Refer to the INDeX Level 10 documentation for full details. Only available in Locales where the Avaya INDeX switch is supported.

- **Abbreviated Ring:** *Default = On, Software level = 4.1+.*
  This option controls the type of alerting ring given for additional appearance calls when a call is already connected. When on, additional calls alert with a single short ring. When off, additional calls alert will normal ringing. The audible alert is still subject to ring delay settings if enabled.

# User | Forwarding

This form can be used to check and adjust a user's call forwarding and follow me settings.

Follow Me is intended for use when the user is present to answer calls but for some reason is working at another extension. For example; temporarily sitting at a colleague's desk or in another office or meeting room. As a user, you would use Follow Me instead of Hot-Desking if you don't have a login code or you don't want to interrupt you colleague also receiving their own calls.

Forwarding is intended for use when, for some reason, the user is unable to answer a call. They may be busy on other calls, unavailable or simply don't answer. Calls may be forwarded to internal or, subject to the user's call barring controls, external numbers.

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.

Note that analog lines doe not provide call progress signalling. Therefore calls forwarded off-switch via an analog line are treated as answered and are not recalled.

A 🔒 symbol indicates that in IP Office 3.2 the settings can be set and locked by the user's associated user rights.

| User \| Forwarding | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Follow Me Number:** *Default = Blank. Range = Internal extension number.*
  Redirects the user's calls to the internal extension number entered. If the redirected call receives busy or is not answered, it follows the user's forwarding and or voicemail settings as if it had been presented to their normal extension. When a user has follow me in use, their normal extension will give alternate dialtone when off hook.
  - Calls targeting longest waiting type hunt groups ignore Follow Me.
  - Calls triggered by actions at the user's original extension, for example voicemail ringback, ignore Follow Me.
  - Park, hold and transfer return calls will go to the extension at which the user initiated the park, hold or transfer action.

- **Forward Unconditional:** *Default = Off*
  This option, when checked and a **Forward Number** is also set, forwards all external calls immediately. Additional options allow this forwarding to also be applied to internal calls and to hunt group calls if required. Using **Follow Me** overrides **Forward Unconditional**. When a user has forward unconditional in use, their normal extension will give alternate dialtone when off hook.
  - **Forward Number:** *Default = Blank. Range = Internal or External number. Up to 32 characters.*
    This option sets the destination number to which calls are forwarded when **Forward Unconditional** is checked. The number can be an internal or external number. This option is also used for **Forward on Busy** and **Forward on No Answer** if no separate **Forward Number** is set for those features.
  - **Forward Internal Calls:** *Default = On, Software level = 3.2+.*
    This option, when checked, sets that internal calls should be also be forwarded immediately when forward unconditional is active.
  - **Forward Hunt Group Calls:** *Default = Off*
    Hunt group calls are not normally presented to a user who has forward unconditional active. Instead they are presented to the next available member of the hunt group. This option, when checked, sets that hunt group calls are also forwarded when forward unconditional is active. The group's **Ring Type** must be ***Sequential*** or ***Rotary***, not ***Collective*** or ***Longest Waiting***. The call is forwarded for the period defined by the hunt group's **No Answer Time** after which it returns to the hunt group if unanswered. Note also that hunt group calls cannot be forwarded to another hunt group.
- **Forward On Busy:** *Default = Off*
  When checked and a forward number is set, external calls are forwarded when the user's extension is busy. The number used is either the **Forward Number** set for **Forward Unconditional** or if set, the separate **Forward Number**. Having **Forward Unconditional** active overrides **Forward on Busy**.
  - If the user has **Busy on Held** selected, if forward on busy is active it is applied when the user is free to receive calls but already has a call on hold.
  - If the user's phone has multiple call appearance buttons, the system will not treat them as busy until all the call appearance buttons are in use unless the last appearance button has been reserved for outgoing calls only.
- **Forward On No Answer:** *Default = Off*
  When checked and a forward number is set, calls are forwarded when the user does not answer within their set **No Answer** time (**User | Telephony**). Having **Forward Unconditional** active overrides **Forward on No Answer**.
  - **Forward Number:** *Default = Blank. Range = Internal or External number. Up to 32 characters.*
    If set, this number is used as the destination for **Forward On Busy** and **Forward On No Answer** when on. If not set, the **Forward Number** set for **Forward Unconditional** is used.
  - **Forward Internal Calls:** *Default = On, Software level = 3.2+.*
    When checked, this option sets that internal calls should be also be forwarded when forward on no answer or forward on busy is active.

# User | Dial In

Use this dialogue box to enable dial in access for a remote user. An Incoming Call Route and RAS service must also be configured.

| User \| Dial In | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Dial In On:** *Default = Off*
  When enabled, dial in access into the system is available via this User account.
- **Dial In Time Profile:** *Default = <None>*
  Select the Time Profile applicable to this User account. A Time Profile (configured via the Time Profile configuration form) can be used to set time restrictions on dial in access via this User account. Dial In is allowed during the times set in the Time Profile form. If left blank, then there are no restrictions.
- **Dial In Firewall Profile:** *Default = <None>*
  Select the Firewall Profile to restrict access to the system via this User account. If blank, there are no Dial In restrictions. Firewall profiles are created in the Firewall Profile configuration form.

# User | Voice Recording

This tab is used to activate the automatic recording of user's external calls. This requires Voicemail Pro to be installed and running. Call recording also requires available conference resources similar to a 3-way conference.

This tab also allows the destination for manual triggered and automatic recordings to be changed from the normal default of the user's own mailbox.

IP Office 4.0+ introduces the following changes to user recording:

- Calls to and from IP devices including those using Direct media can be recorded.
- User recording stops if the call is transferred to another user.
- Calls parked or held by the user pause recording until the user unparks the call or takes it off hold.

| User \| Voice Recording | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Record Outbound:** *Default = None*
  Select whether outgoing external calls are recorded. Options for recording are:
  - **On:** Record the call if possible.
  - **Mandatory:** If not possible to record, return busy tone.
  - **Percentages of calls:** Various percentages of calls made by the user will be recorded.
- **Record Inbound:** *Default = None*
  The same as Record Outbound but applied to inbound external calls to the user.
- **Record Time Profile:** *Default = <None> (Any time)*
  Used to select a time profile during which automatic call recording setting above apply.
- **Auto Recording Mailbox:** *Default = <user's own mailbox>*
  Sets the mailbox into which automatically triggered recordings are placed.
- **Voice Recording Library (Auto):** *Default = Not Enabled, Software level = 3.0+.*
  When checked, this setting overrides the **Auto Recording Mailbox** setting and marks the recording for collection by the ContactStore for IP Office application. Note that Voicemail Pro still performs the actual recording.
- **Manual Recording Mailbox:** *Default = <user's own mailbox>*
  Sets the mailbox into which recordings triggered by the user are placed.
- **Voice Recording Library (Manual):** *Default = Blank (Not Enabled), Software level = 3.0+.*
  When checked, this setting overrides the **Manual Recording Mailbox** setting and marks the recording for collection by the ContactStore for IP Office application. Note that Voicemail Pro still performs the actual recording.

# User | Coverage

Call coverage allows calls ringing at one extension (the 'Sender') to also be presented and answered at other defined extensions (the 'Covering Extensions').

| User \| Coverage | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✗. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✗, **3.1** ✗, **3.2** ✗, **4.0** ✗, **4.1** ✗. |
| **Mergeable** | ✓. |

- **Covering Extension**
  The number of the extension that will be receiving the calls from the selected extension.
- **Covering User**
  This is the user's account name associated with the covering extension.

**To add a covering extension**

1. Right-click within the **Coverage** window and select **Add**.
2. Choose from the list of extension/users.
3. Click **OK**.

**Senders**

Senders are extensions that share their alerting calls with other extensions, referred to as their covering extensions.

The only calls that are not shared are:

- Hunt Group calls that alert at the sender.
- Automatic Intercom calls.
- Calls that have been forwarded/diverted to the sender.
- Paging calls.
- Calls that are being covered for another station.
- Calls from one of their covering extensions.

**Covering Extensions**

When the sender's extension rings, the covering extensions also ring and show the call on a free call appearance button. The display indicates that the call is from the sender by showing the incomings call's name or number and the sender's name.

Covering Extensions can receive their own calls as well as calls for the Sender. A Covering Extension can receive a call when:

- Send All Calls/Do Not Disturb is not active.
- Forwarding/Divert is not active.
- They have an available Call Appearance button to accept the call.

**Notes**
To help covering extensions handle coverage calls efficiently it is suggested that the following buttons are programmed.

- **Program additional Call Appearance buttons**
  Covering extensions must have enough call appearance buttons for their own calls and for the extensions they are covering. By default each extension has three call appearance buttons. A suggested minimum extra is one less than the number of call appearance buttons on the sender's extension.
- **Program a Voicemail Collect button for the Sender**
  This will allow the covering extension to transfer a call directly to the sender's voicemail.
- **Program an Automatic Intercom button for the Sender**
  This allows the covering extension to place a voice announcement. If you do not wish to make voice announcement calls, use Dial Intercom instead.
- **Program a Send All Calls button**
- **Program a Drop Button**
  This helps in transferring calls.

**Call Alerting Scenarios**
Listed below are examples of how calls to the sender's extension are handled in specific scenarios.

- **Sender and Covering Extensions available**
  An incoming call alerts both the sender's and covering extension's on call appearance buttons. It alert the sender's extension for their set No Answer Time and then alerts the covering extension only until the call is answered or the caller hangs up.
- **Sender available/Covering Extension not available**
  An incoming call alerts the sender only. The call remains alerting until it is answered or the caller hangs up.
- **Sender not available/Covering Extension available**
  The call will alert the covering extension but not the sender. The call remains alerting until the call is answered or the call hangs up.
- If voicemail is available and enabled for the sender, then in all the above scenarios, following the sender's No Answer Time the call is redirected to the Sender's voicemail.
- **Sender and Covering Extension not available**
  The caller hears busy tone or is redirected to the sender's voicemail.

# User | Button Programming

This tab is used to assign functions to the programmable keys provided on many Avaya DS and IP telephones. For full details on Button Programming refer to the IP Office Button Programming Manual.

- **T3 Phones:** T3 phone buttons have default functions. These are not shown in the configuration file but can be overridden by settings added to the configuration file. Buttons left blank or set to call appearance will use the phone's default function for that button.

A symbol indicates that the button has been set and locked by the user's associated user rights.

**Editing a Button**

1. Select the button row and then click **Edit**.
2. Enter a custom label if required.
3. Click on the **...** button.
4. Use the window that appears to select the required action.
5. Enter or select the required setting for the action in the **Action Data** field.
6. Click **OK**.
7. Click **OK** again.

or

1. Select the button row.
2. Right-click on the **Action** field and select the required action.
3. Right-click on the **Action Data** field and enter or select the required value.
4. If required, right-click on the **Label** field and enter the required label.
5. Repeat for any other buttons.
6. Click **OK**.

| **User \| Button Programming** | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Button No.**
  The number of the DSS key against which the function is being set. To set a function against a button double-click it or select it and then click **Edit**.

- **Label**
  This is a text label for display on the phone. It will be used on phones where the button status is indicated by the adjacent display space rather than LED's. Depending on the phone type, only up to eight characters may be displayed.

- **Action**
  Defines the action taken by the button.

- **Action Data**
  This is a parameter used by the selected action. The options here will vary according to the selected button action.

- **Display All**
  The number of button displayed is based on the phone associated with the user when the configuration was loaded. This can be overridden by selecting **Display All Buttons**. This may be necessary for users who switch between different phones using hot desking or have an expansion unit attached to their phone.

# User | Menu Programming

Some Avaya DS phones have a **Menu** key, sometimes marked with an icon.

When **Menu** is pressed, a number of default functions are displayed. The **<** and **>** keys can be used to scroll through the functions while the keys below the display can be used to select the required function.

The default functions can be overwritten by selections made within this tab.

<table>
<tr><th colspan="2">User | Menu Programming</th></tr>
<tr><td>Control Unit</td><td>SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓.</td></tr>
<tr><td>Software Level</td><td>2.1 ✓, 3.0DT ✓, 3.0 ✓, 3.1 ✓, 3.2 ✓, 4.0 ✓, 4.1 ✓.</td></tr>
<tr><td>Mergeable</td><td>✓.</td></tr>
</table>

# User | Twinning

Twinning allows a user's calls to be presented to both their current extension and to another number. The IP Office supports two modes of twinning:

| | Internal | Mobile |
|---|---|---|
| **Twinning Destination** | IP Office extensions only | External numbers only. |
| **Supported in** | All locales.* | All locales. |
| **License Required** | No | Yes |

*IP Office 4.0 Q2 2007 maintenance release and higher only. Prior to IP Office 4.0 Q2 2007 maintenance release, Internal twinning was not supported in North American locales.

**Internal Twinning**

Internal twinning is available on IP Office 3.1 and higher systems. It can be used to link two IP Office extensions to act as a single extension. Typically this would be used to link a users desk phone with some form of wireless extension such as a DECT or WiFi handset.

Internal twinning is an exclusive arrangement, only one phone may be twinned with another. When twinned one acts as the primary phone and the other as the secondary phone. With internal twinning in operation, calls to the user's primary phone are also presented to their twinned secondary phone. Other users cannot dial the secondary phone directly.

- If the primary or secondary phones have call appearance buttons, they are used for call alerting. If otherwise, call waiting tone is used, regardless of the users call waiting settings. In either case, the **Maximum Number of Twinned Calls** setting applies.
- Calls to and from the secondary phone are presented with the name and number settings of the primary.
- The twinning user can transfer calls between the primary and secondary phones.
- Calls will ring at the secondary if the primary is logged off or set to do not disturb.
- Logging off or setting do not disturb at the secondary only affects the secondary.
- User buttons set to monitor the status of the primary also reflect the status of the secondary.
- Depending on the secondary phone type, calls alerting at the secondary but then answered at the primary may still be logged in the secondary's call log. This occurs if the call log is a function of the phone rather than the IP Office system.
- Call alerting at the secondary phone ignoring any **Ring Delay** settings applied to the appearance button being used at the primary phone. The only exception is buttons set to No Ring, in which case calls are not twinned.

**Mobile Twinning**
This licensed method of twinning can be used with external numbers. Calls routed to the secondary remain under control of the IP Office and can be pulled back to the primary if required. If either leg of an alerting twinned call is answered, the other leg is ended.

A number of controls are available in addition to those on this tab.

- **Button Programming Actions**
  The **Emulation | Twinning** action can be used to control mobile twinning operation. Set on the primary extension, when that extension is idle the button can be used to set the twinning destination and to switch twinning on/off. When a twinned call has been answered at the twinned destination, the button can be used to retrieve the call at the primary extension.
- **Short Code Features**
  The following short code actions are available for use with mobile twinning.
  - **Set Mobile Twinning Number.**
  - **Set Mobile Twinning On.**
  - **Set Mobile Twinning Off.**
  - **Mobile Twinned Call Pickup.**
- **Caller ID**
  The options on the **System | Twinning** tab can be used to control which caller ID is sent with calls sent to the twinned destination. The use of those options may be restricted by the trunk type carrying the twinned call and the services provided by the line provider.

Mobile twinning is only applied to normal calls. It is not applied to:

- Intercom, dial direct and page calls since these are answered automatically.
- Calls alerting on line appearance, bridged appearance and call coverage buttons.
- Returning held, returning parked, returning transferred and automatic callback calls.
- Follow me calls.
- Additional calls when the primary extension is active on a call or the twinning destination has a connected twinned call.

User BLF indicators and application speed dials set to the primary user will indicate busy when they are connected to a twinned call including twinned calls answered at the mobile twinning destination.

**Analog Lines**
These types of lines do not provide call progress signalling. Once a twinned call has been sent to an analog line, the IP Office assumes that it has been answered and stops ringing the primary extension.

**Mobile Twinning in a Small Community Network**
In order for mobile twinning to be used with SCN extensions as the destination, a short code or short codes must be added in order to route the calls to the correct SCN link. For example, a primary user at site A wants to twin with an extension at Site B, the two sites being linked by an IP trunk in Outgoing Group Id of 1. At Site A, add the system short code **8N/Dial/N/1**. For the primary to twin with extension 300 at Site B, the mobile twinning number should be entered 8300.

**Settings**

A symbol indicates that in IP Office 3.2 the settings can be set and locked by the user's associated user rights.

- **Twinning Type:** *Default = None, Software Level = 3.2+.*
  This control is used to select the type of twinning. For US systems the options are ***None*** and ***Mobile***. For other locales the options are ***None***, ***Internal*** and ***Mobile***. The ***Mobile*** option is only shown if the system number of licensed mobile twinning users has not been exceeded.

**Internal Twinning Settings**
The following settings are available when the **Twinning Type** is set to ***Internal***.

| User \| Twinning \| Internal | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✗, **3.0DT** ✗, **3.0** ✗, **3.1** ✓, **3.2** ✓, **4.0*** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

* For North American locales only supported for IP Office 4.0 Q2 2007 maintenance release.

- **Twinned Handset:** *Default = Blank.*
  For internal twinning, the drop-down list can be used to select an available user as the twinned calls destination. Users not displayed in the list are already twinned with another user. If the list is grayed out, the user is a twinning destination and the primary to which they are twinned is displayed. The secondary phone must be on the same IP Office.
- **Maximum Number of Twinned Calls:** *Default = 1.*
  If set to one, when either the primary or secondary phone are in use, any additional incoming call receives busy treatment. If set to two, when either phone is in use, it receives call waiting indication for any second call. Any further calls above two receive busy treatment.
- **Twin Bridge Appearances:** *Default = Off, Software level = 4.1+.*
  By default only calls alerting on the primary phone's call appearance buttons also alert at the secondary. When this option is enabled, calls alerting on a bridged appearance button at the primary can also alert at the secondary.
- **Twin Coverage Appearances:** *Default = Off, Software level = 4.1+.*
  By default only calls alerting on the primary phone's call appearance buttons also alert at the secondary. When this option is enabled, calls alerting on a coverage appearance button at the primary can also alert at the secondary.
- **Twin Line Appearances:** *Default = Off, Software level = 4.1+.*
  By default only calls alerting on the primary phone's call appearance buttons also alert at the secondary. When this option is enabled, calls alerting on a line appearance button at the primary can also alert at the secondary.

**Mobile Twinning Settings**

The following settings are available when the **Twinning Type** is set to ***Mobile***. The use of mobile twinning requires entry of a Mobile Twinning license into the configuration.

| **User \| Twinning \| Mobile** | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✗, **3.0DT** ✗, **3.0** ✗, **3.1** ✗, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Twinned Mobile Number:** *Default = Blank.*
  This field sets the destination number for mobile twinned calls. This can be an external or internal number. It is subject to normal short code processing and should include any external dialing prefix if necessary. For an SCN number, see the required short code detailed in the note above.

- **Twinning Time Profile:** *Default = <None> (Any time)*
  This field allows selection of a time profile during which mobile twinning will be used.

- **Mobile Dial Delay:** *Default = 2 seconds*
  This setting controls how long calls should ring at the user's primary extension before being routed to ring at the twinning destination number. This setting may be used at the user's choice, however it may also be a necessary control. For example, if the twinning number is a mobile device that has been switched off, the mobile service provider may immediately answer the call with their own voicemail service. This would create a scenario where the user's primary extension does not ring or ring only briefly.

- **Hunt group calls eligible for mobile twinning:** *Default = Off*
  This setting controls whether hunt group calls ringing the user's primary extension should also be presented to the mobile twinning number.

- **Forwarded calls eligible for mobile twinning:** *Default = Off*
  This setting controls whether calls forwarded to the user's primary extension should also be presented to the mobile twinning number.

# User | T3 Options

This tab is only applicable to users using Avaya T3 phones. It is divided into several sub-tabs.

| User \| T3 Options | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✗, **3.0DT** ✗, **3.0** ✗, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

**T3 Telephony**

- **Third Party Forwarding**
  Avaya T3 phone users can be given menu options to change the forwarding settings of other users. In addition to the following controls, this functionality is protected by the forwarding user's login code.
  - **Allow Third Party Forwarding:** *Default = Off*
    Sets whether this user can change the forwarding settings of other users.
  - **Protect from Third Party Forwarding:** *Default = On*
    Sets whether this user's forwarding settings can be changed by other users.
- **Advice of Charge**
  - **Display Charges:** *Default = On, Software level = 4.0+.*
    This setting is used to control whether the user sees ISDN AOC information when using a T3 phone.

**Hunt Group Settings**

Avaya T3 phones are able to display the status (in-service, out-of-service or night-service) of up to 5 hunt groups. The phone user can change the service status of those groups.

T3 phones can also display the phone user's membership status (enabled or disabled) for up to 5 hunt groups and allow the user to change that status.

- **Hunt Group Service Status**
  This list shows all the hunt groups on the IP Office system. Up to 5 of these groups can be checked. The user is then able to view and change the service status of the checked groups through their T3 phones menus.
- **Hunt Group Membership Status**
  This list shows the hunt groups of which the user is a member. Up to 5 of these groups can be checked. The user is then able to view and change their membership status for the checked groups through their T3 phones menus.

**Personal Directory**

T3 phones are able to display a personal directory of numbers to speed dial. Each user can have up to 100 personal directory numbers. Unlike system directory numbers, these entries are not matched against incoming ICLID numbers.

- **Name**
  Enter the text, without spaces, to be used to identify the number.
- **Number**
  Enter the number, without spaces, to be dialed.

# User | Phone Manager Options

This tab is used to configure the user's Phone Manager options.

A 🔒 symbol indicates that in IP Office 3.2 the settings can be set by the user's associated user rights.

| User \| Phone Manager Options | |
|---|---|
| **Control Unit** | **SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓.** |
| **Software Level** | **2.1 ✗, 3.0DT ✗, 3.0 ✗, 3.1 ✗, 3.2 ✓, 4.0 ✓, 4.1 ✓.** |
| **Mergeable** | ✓. |

- **Allow user to modify Phone Manager settings:** *Default = On* 🔒
  This setting is used with the **Phone Manager Status Options**, **Screen Pop Options** and **Hide Options** below. It controls whether those options are applied every time the user starts Phone Manager or only the first time the user starts Phone Manager.
  - If this setting is enabled, then the IP Office configuration setting of those options are only applied the first time a user starts Phone Manager on a PC. Those settings become part of the user's Phone Manager profile on that PC. They can be changed by the user through Phone Manager. On subsequent Phone Manager starts the Manager settings are ignored.
  - If this setting is not enabled, the IP Office configuration settings are applied every time the user starts Phone Manager and cannot be overridden by the user.
- **Agent Mode:** *Default = Off* 🔒
  This option controls the setting of the **Agent Mode** option on the **Configure Preferences | Agent Mode** tab within Phone Manager Pro. When enabled, the user has additional toolbar controls for **Busy Wrap Up**, **Busy Not Available** and **Select Group**. Note that the options on the Phone Manager Pro **Agent Mode** tab can be greyed out from user changes by the **Agent Mode** setting in **Configuration Options** below.
- **Phone Manager Type:** *Default = Lite* 🔒
  Determines the mode in which the user's copy of the Phone Manager application operates. Note that the number of users able to simultaneously use modes other than **Lite** is controlled by licenses entered into the IP Office configuration. This setting cannot be changed by the user. * For pre-3.2 IP Office systems this setting is located on the **User | User** tab.
  - **Lite**
    Basic Phone Manager mode. This mode does not require any licenses.
  - **Pro**
    Advanced Phone Manager mode that enables a range of additional functions. This mode requires an available Phone Manager Pro license, otherwise the application will run in Phone Manager Lite mode.
  - **Phone Manager PC Softphone**
    This is the VoIP IP phone mode of Phone Manager Pro. This mode requires both an available Phone Manager Pro license and a Phone Manager Pro IP Audio Enabled license. The user must be associated with an VoIP extension within the IP Office configuration.
  - **Phone Manager Pro Telecommuter:** *Software level = 4.1+.*
    This version of phone Manager Pro is supported with Phone Manager 4.1+. It allows the user to make and receive calls via an external phone specified at Phone Manager login. This mode requires an available Phone Manager Pro license, otherwise the application will run in Phone Manager Lite mode.

- **VoIP Mode:** *Default = On*
  This option only appears if the selected **Phone Manager Type** is ***Phone Manager PC Softphone***. It sets the **Enable VoIP** control within the user's Phone Manager PC Softphone.
- **Book a Conference in Phone Manager:** *Default = Off*
  When enabled, displays links in the user's Phone Manager application for access to the IP Office Conferencing Center application if installed. Note that to book a conference requires the user to have a Conferencing Center user ID and password. This feature also requires the **Conference Center URL** to be set (**System | System**). This setting cannot be changed by the user. * For pre-3.2 IP Office systems this setting is located on the **User | User** tab.
- **Configuration Options**
  These options allow the user to changes the settings on the indicated configure preferences tabs within Phone Manager.
  - The controllable tabs for Phone Manager Lite are **Telephone** and **Do Not Disturb**.
  - The additional controllable tabs for Phone Manager Pro and Phone Manager PC Softphone are **Screen Pop**, **Compact Mode**, **Agent Mode**, **Voicemail** and in IP Office 4.0 and higher **Mobile Twinning**.
- **Screen Pop Options**
  These options allow selection of the Phone Manager Pro/Phone Manager PC Softphone screen pop options **Ringing**, **Answering**, **Internal**, **External** and **Outlook**.
  - The **Allow user to modify Phone Manager settings** option controls whether these settings are applied only when Phone Manager is first started or every time Phone Manager is started.
- **Phone Manager Status Options**
  These options allow selection of the tabs to show within the call history area of the user's Phone Manager.
  - The tabs selectable for Phone Manager are **All**, **Missed**, **Status** and **Messages**.
  - The additional tabs selectable for Phone Manager Pro and PC Softphone are **Incoming**, **Outgoing** and **Account Codes**.
  - The **Allow user to modify Phone Manager settings** option controls whether these settings are applied only when Phone Manager is first started or every time Phone Manager is started.
- **Hide Options**
  These options allow selection of the Phone Manager Pro/Phone Manager PC Softphone options **Hide on close** and **Hide on no calls**.
  - The **Allow user to modify Phone Manager settings** option controls whether these settings are applied only when Phone Manager is first started or every time Phone Manager is started.

# User | Hunt Group Memberships

This tab displays the hunt group of which the user has been made a member. The tick boxes indicate whether the user's membership of each of those groups is currently enabled or disabled.

# User | Announcements

IP Office 4.0 supports user announcements with Voicemail Pro 4.0 and higher. Announcements are played to callers waiting to be answered.

- User announcements are supported with Voicemail Pro 4.0+ and Embedded Voicemail.
- If no voicemail channel is available for an announcement when required, the announcement is not played.
- In conjunction with Voicemail Pro, IP Office 4.0+ allows a number of voicemail channels to be reserved for announcements. See **System | Voicemail**.
- With Voicemail Pro, the announcement can be replace by the action specified in a Queued (1st announcement) or Still Queued (2nd announcement) start point call flow for the user. Refer to the Voicemail Pro Installation and Maintenance documentation for details.
- Calls can be answered while hearing announcements. If it is a mandatory requirement that announcements should be heard before a call is answered then a Voicemail Pro call flow should be used before the call is presented to the user.
- If a call is rerouted, for example forwarded, the announcement plan of the original users is still applied until the call is answered. The exception is calls rerouted to a hunt group at which point the hunt group announcement settings are applied.
- For announcements to be used effectively, either the user's no answer time must be extended beyond the default 15 seconds or **Voicemail On** should be deselected.

Voicemail Pro provides default announcements "*I'm afraid all the operators are busy at the moment, but please hold and you will be transferred when somebody becomes available"* and *"I'm afraid all the operators are still busy but please hold and you will be transferred when somebody becomes available".* These defaults are used for announcement 1 and announcement 2 respectively if no specific hunt group announcement has been recorded. Embedded Voicemail does not provide any default announcement.

- **Voicemail Pro**
  There is no mechanism within the telephony user interfaces (TUI) to record user announcements. To provide custom announcements, user queued and still queued start points must be configured with Voicemail Pro with the required prompts played by a generic action.
- **Embedded Voicemail**
  Embedded Voicemail does not include any default announcement or method for recording an announcement. The Record Message short code feature is provided to allow the recording of hunt group announcements. The telephone number field of short codes using this feature requires the extension number followed by either ".1" for announcement 1 or ".2" for announcement 2. For example, for a hunt group with extension number 300, the short codes ***1N# | Record Message | N".1"*** and ***2N# | Record Message | N".2"*** could be used to allow recording of the announcements by dialing *1300# and *2300#.

| User \| Announcements | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✗, **3.0DT** ✗, **3.0** ✗, **3.1** ✗, **3.2** ✗, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Announcements On:** *Default = Off, Software level = 4.0+.*
  Enables use of user announcements.
- **Wait before 1st announcement:** *Default = 10 seconds. Range = 0 to 9999 seconds.*
  This setting sets the time delay from the calls presentation to the user, either ringing or queued, after which the first announcement should be played to the caller.
- **Flag call as answered:** *Default = Off.*
  Normally for reporting purposes with the IP Office CCC application, a call is not regarded as answered until it has been answered by a person or by a Voicemail Pro action set to **Flag call as answered**. This setting allows the call to be marked as answered once the caller has heard the first announcement.
- **Post announcement tone:** *Default = Silence.*
  Following the first announcement, you can select whether the caller should hear ***Music on Hold***, ***Ringing*** or ***Silence*** until answered or played another announcement.
- **2nd Announcement:** *Default = On.*
  If selected, a second announcement can be played to the caller if they have still not been answered.
- **Wait before 2nd announcement:** *Default = 20 seconds. Range = 0 to 9999 seconds.*
  This setting sets the wait between the 1st and the 2nd announcement.
- **Repeat last announcement:** *Default = On.*
  If selected, the last announcement played to the caller is repeated until they are answered or hang-up.
- **Wait before repeat:** *Default = 20 seconds.  Range = 0 to 9999 seconds.*
  If **Repeat last announcement** is selected, this setting sets wait applied between each repeat of the last announcement.

# User | SIP

This tab is only available when a SIP trunk with a SIP URI entry has been added to the IP Office configuration.

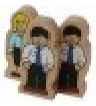
Various fields within the URI settings used by SIP trunks can be set to **Use User Data**. When that is the case, the values from this tab are used inserted into the URI when the user makes or receives a SIP call.

| User \| SIP | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✗, **3.0DT** ✗, **3.0** ✗, **3.1** ✗, **3.2** ✗, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **SIP Name:** *Default = User name.*
  The value from this field is used when the **From** field of the SIP URI being used for a SIP call is set to ***Use User Data***.
- **SIP Display Name (Alias):** *Default = User name.*
  The value from this field is used when the **Display Name** field of the SIP URI being used for a SIP call is set to ***Use User Data***.
- **Contact:** *Default = User name.*
  The value from this field is used when the **Contact** field of the SIP URI being used for a SIP call is set to ***Use User Data***.
- **Anonymous:** *Default = Off.*
  If the **From** field in the SIP URI is set to ***Use User Data***, selecting this option inserts ***Anonymous*** into that field rather than the **SIP Name** set above.

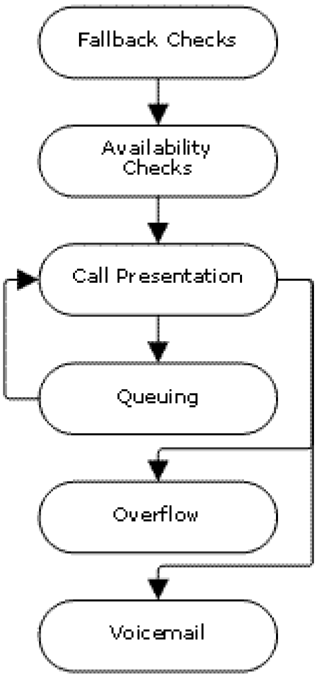
# Hunt Group Settings

## Hunt Group Overview



A hunt group is a collection of users accessible through a single directory number. Calls to that hunt group can be answered by any available member of the group. The order in which calls are presented can be adjusted by selecting different group types and adjusting the order in which group members are listed. For a full overview of hunt groups, see Hunt Groups in the **Telephone Features** section.

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




- **Call Presentation**
  The order in which the available members of the hunt group are used for call presentation is selectable.
- **Availability**
  There are a range of factors which control whether hunt group calls are presented to a user in addition to that user being a member of the hunt group.
- **Queuing**
  This optional feature allows calls to be queued when the number of calls to be presented exceeds the number of available hunt group members to which call can be presented.
- **Announcements**
  On systems with a voicemail server (Voicemail Pro, Voicemail Lite or Embedded Voicemail), announcements can be played to callers waiting to be answered. That includes calls that are ringing and calls that are queued.
- **Overflow**
  This optional feature can be used to redirect calls to an overflow group or groups when not answered within a set time.
- **Fallback**
  A hunt group can be taken out of operation manually or using a time profile. During fallback, calls can be redirected to a fallback group or sent to voicemail or just receive busy tone. Two types of fallback are supported; night service and out of service.
- **Voicemail**
  Calls can be redirected to voicemail. The IP Office allows selection of whether hunt group calls remain in the hunt group mailbox or are copied to the individual mailboxes of the hunt group members. When messages are stored in the hunt group's own mailbox, selection of who receives message waiting indication is possible.

Changing the name of a hunt group has the following effects:

- A new empty mailbox is created on voicemail with the new hunt group name.
- Entries in other groups' **Overflow** lists will be updated.
- **Out-of-Service** and **Night-Service** fallback references are updated.

Modifying the extension number of a hunt group updates the following:

- Group buttons.
- Overflow, Out of Service Fallback and Night Service Fallback group entries.
- Incoming call route entries.

When a hunt group is deleted, all references to the deleted group will be removed including:

- Entry in Incoming call routing table.
- Transfer target in internal auto-attendant.
- Overflow, Night-Service or Fallback-Service on other groups.
- DSS keys monitoring group status.

# Hunt Groups in a Small Community Network (SCN)

In a Small Community network, the extension numbers of users are automatically shared between IP Office systems and become diallable from other systems without any further programming.

The above does not apply to hunt group extension numbers. In order for users on one system to dial a hunt group on another system, short codes are required to route the call to the correct system. Through the addition of **Advanced Small Community Networking** licenses, IP Office 4.0 introduces a number of options that make hunt groups automatically useable across an SCN without the addition of short codes.

- **Advanced Hunt Groups**
  Each hunt group can be set as being 'advertised'. The hunt group can then be dialed from other systems within the SCN. The hunt groups extension number and name must be unique within the network. Non-advertised hunt group numbers remain local only to system hosting the hunt group.

- **Distributed Hunt Groups**
  Hunt groups on a system can include users located on other IP Office systems within the SCN network. Distributed hunt groups are automatically advertised to other systems within the SCN. Note that distributed hunt groups can only be edited on the system on which they were created.

# Hunt Group | Hunt Group

| Hunt Group \| Hunt Group | |
|---|---|
| **Control Unit** | **SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓.** |
| **Software Level** | **2.1 ✓, 3.0DT ✓, 3.0 ✓, 3.1 ✓, 3.2 ✓, 4.0 ✓, 4.1 ✓.** |
| **Mergeable** | ✓. |

- **Name:** *Range = Up to 15 characters*
  The name to identify this Hunt Group. Only alphanumeric characters with no spaces should be used. Do not start names with a numeric character. This field is case sensitive and must be unique.
    - Voicemail uses the name to match a group and its mailbox. Changing a group's name will route its voicemail calls to a new mailbox. Note however that Voicemail Pro will treat names such as "Sales", "sales" and "SALES" as being the same.
- **Extension:** *Range = 2 to 9 digits.*
  This sets the directory number for calls to the hunt group.
    - Users for CBC and CCC should only use up to 4 digit extension numbers.
    - Extension numbers in the range 8897 to 9999 are reserved for use by the IP Office Delta Server.
- **Overflow Time:** *Default = Blank, Range = 1 to 99999 seconds.*
  Calls that ring the hunt group members without being answered can be redirected to an overflow group or groups. This requires an overflow group or groups to be added to the **Overflow Group List** and for those groups to be ***In Service***. The **Overflow Time** setting is then used to determine when the overflow groups should be used.
    - If **Queuing** is off and all members of the hunt group are busy, a call presented to the group will overflow immediately, irrespective of the **Overflow Time**.
    - If **Queuing** is on and all members of the hunt group are busy, a call presented to the group may queue for up to the **Overflow Time** before overflowing.
    - If the call is currently ringing a hunt group member when the **Overflow Time** expires, it will complete ringing using the set **No Answer Time** before overflowing.
    - If the **Overflow Time** is ***0*** or blank, a call will overflow when it has rung each available hunt group member without being answered.
    - One or more calls in the group reaching their **Overflow Time** will permit other calls within the group to overflow if they are queued, or have rung each available hunt group member without being answered.
    - When a call overflows, it is still controlled by a number of settings of the hunt group from which it has overflowed.
        - Calls that overflow use the announcement settings of the original group from which they overflowed.
        - Calls that overflow use the **Voicemail Time** of the original group from which they overflowed.
        - Calls that overflow are included in the **Queue Length** and **Calls In Queue Threshold** of the hunt group from which they overflowed. They are not included in those values for the hunt group to which they overflow.
- **No Answer Time (secs):** *Default = Blank (Use System setting). Range = 1 to 99999 seconds.*
  The number of seconds an extension rings before the call is passed to another extension in the list. This applies to all telephones in this group and the Overflow Groups (if used). If left blank the system Default No Answer Time (15 seconds by default) is used. For collective hunt groups the idea of moving to the next member when the **No Answer Time** expires does not apply. Calls will ring until the call is answered or another timeout is applied.

- **Voicemail Answer Time:** *Default = 45 seconds, Range = Off, 1 to 99999 seconds, Software level = 4.0+.*
  This setting sets how long a call should be presented to a hunt group, and its overflow groups if set, before going to voicemail. When exceeded the call goes to voicemail (if available) regardless of any announcements, overflow, queuing or any other actions.
- **Ring Mode:** *Default = Sequential*
  Sets the order in which each extension in a Hunt Group is rung. The **Extension List** is used to provide the basic order, with those users who are available being used for hunt group calls.
  - **Collective** *(also known as Group)*
    All telephones in the Extension List ring simultaneously.
  - **Sequential** *(also known as Linear and Hunt)*
    Each extension is rung in order, one after the other, starting from the first extension in the list each time.
  - **Rotary** *(also known as Circular)*
    Each extension is rung in order, one after the other. However, the last extension used is remembered. The next call received rings the next extension in the list.
  - **Longest Waiting** *(also known as Idle and Most Waiting)*
    The extension that has been unused for the longest period rings first, then the extension that has been idle second longest rings, etc. For extensions with equal idle time, 'linear' mode is used.
- **Agent's Status on No-Answer Applies To:** *Default = None (No status change), Software level = 4.0+.*
  For call center agents, that is hunt group members with a login code and set to forced login, the IP Office can change the agent's status if they do not answer a hunt group call presented to them before being automatically presented to the next available agent.
  - This setting defines what type of hunt group calls should trigger use of the agents **Status on No Answer** setting. The options are ***None***, ***Any Call*** and ***External Inbound Calls Only***.
  - The new status is set by the agent’s **Status on No Answer** setting, set through their **User | Telephony** settings.
  - This action is only applied if the call is unanswered at the agent for the **No Answer Time** or longer. It does not apply if the call is presented and then, before the **No Answer Time** expires is answered elsewhere. It is does not apply if the caller disconnects.
- **Call Waiting On:** *Default = Off, Software level = 3.0+.*
  Hunt group call waiting is only supported for group's set to the **Ring Type** of ***Collective***. When on, user's in the group already on a call, receive call waiting indication when a call to the hunt group call is waiting to be answered. 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.
- **Central System:** *Software level = 4.0+.*
  This field is only visible if the IP Office has an **Advanced Small Community Networking** license. The field is for information only. It displays the IP Office system where the hunt group was created and can be configured.

- **Advertise Group:** *Default = Off, Software level = 4.0+.*
  This field is only visible if the IP Office has an **Advanced Small Community Networking** license. If ***On***, details of the hunt group are advertised to the other systems within a Small Community Network and the hunt group can be dialled from those other systems without the need for routing short codes.
  - Advertised groups must have an extension number that is unique within the SCN. If an advertised hunt group's extension number conflicts with a local groups extension number, the advertised group is ignored.
  - Group's set as advertised will appear in the configuration of other IP Office systems that also have an **Advanced Small Community Networking** license. However an advertised group can only be edited on the IP Office system on which it was created.
  - Hunt groups that contain members from other IP Office systems are automatically advertised.
  - Advertised groups are not saved as part of the configuration file when **File | Save Configuration As** is used.
- **Extension List**
  This is an ordered list of the users who are members of the hunt group. For Sequential and Rotary groups it sets the order in which members are used.
  - Repeated numbers can be used, for example 201, 202, 201, 203, etc. Each extension will ring for the number of seconds defined by the **No Answer Time** before moving to the next extension in the list, dependent on the **Hunt Type** chosen.
  - The check box next to each member indicates the status of their membership. Checked boxes appear for members whose membership is enabled. The order of the users can be changed by dragging the existing entries to the required position.
  - To add entries select **Add**. The available users are shown in the left-hand column. Those already in the group are shown in the right-hand column. Select a user or users and use the **>>** and **<<** buttons to move them between the columns.
  - If the system has a **Advanced Small Community Networking** license, then users on remote systems in a Small Community Network with the same license can be included. Groups containing remote members are automatically advertised within the SCN.
- **Overflow Group List**
  If a call cannot be answered by the extensions shown in the **Extension List**, the call can be passed to the available members of another hunt group or groups listed here. The hunt type and no answer time of the overflow group are applied while it rings members of that group.

# Hunt Group | Voicemail

IP Office supports voicemail for hunt groups in addition to individual user voicemail mailboxes.

- **When is voicemail used?**
  If voicemail is available and enabled for a hunt group, it is used in the following scenarios.
  - **Voicemail Answer Time**
    This timeout is supported by IP Office 4.0 and higher. The default is 30 seconds. When exceeded the call goes to voicemail (if available) regardless of any announcements, overflow, queuing or any other actions.
  - **Unanswered Calls**
    When a call has rung unanswered at all the available hunt group members. If overflow is being used that will include being unanswered by the available overflow group members.
  - **Queue Limit Reached**
    If queuing is being used, it overridesuse of voicemail prior to expiry of the **Voicemail Answer Time**, unless the number of queued callers exceeds the set **Queue Limit**. By default there is no set limit.
  - **Night Service**
    When the hunt group is in night service with no Night Service Fallback Group set.
  - **Out of Service**
    When the hunt group is out of service with no Out of Service Fallback Group set.
  - **Automatic Call Recording**
    Incoming calls to a hunt group can be automatically recorded using the settings on the **Hunt Group | Voice Recording** tab.
- **Which Mailbox is Used**
  For IP Office 3.2 and earlier, when voicemail was invoked, the mailbox of whichever hunt group was currently handling the call was used, for example the mailbox of the overflow or night service hunt group might be used if the call had gone from the original hunt group to an overflow or night server hunt group. For IP Office 4.0 and higher, the mailbox of the originally targeted hunt group is used even if the call has overflowed or gone to a night server hunt group.
- **Who Receives Message Waiting Indication?**
  By default no user is configured to receive message waiting indication when a hunt group voicemail mailbox contains new messages. Message waiting indication is configured by adding a **H*groupname*** entry to a user's **SourceNumbers** tab (**User | Source Numbers**).
- **Accessing Hunt Group Messages**
  By default no mechanism is provided for access to specific hunt group mailboxes.
  - **Phone Manager**
    User's with hunt group message waiting indication can access the hunt group mailbox through Phone Manager.
  - **Intuity Emulation Mailbox Mode**
    For IP Office systems using Intuity emulation mode mailboxes, the hunt group extension number and voicemail code can be used during normal mailbox access
  - **IP Office Mailbox Mode**
    For IP Office mode mailbox access, short codes are required to access the mailbox directly.
- **Broadcast**
  The voicemail system can be instructed to automatically forward messages to the individual mailboxes of the hunt group members. The messages are not stored in the hunt group mailbox.

| Hunt Group \| Voicemail | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Voicemail Code:** *Default = Blank, Range = 0 to 15 digits*
  A code (1-15 digits) used by the voicemail server to validate access to this mailbox. If remote access is attempted to a mailbox that has no voicemail code set, the prompt *"Remote access is not configured on this mailbox"* is played. Whether the caller will be prompted to enter this code varies as follows:
  - **Embedded Voicemail**
    The voicemail code is used if set.
  - **Voicemail Lite/Voicemail Pro in IP Office mode**
    The voicemail code is required when accessing the mailbox from a location that is not set as a trusted number in the user's **Source Numbers** list. Also Voicemail Pro call flows containing an action where the action's PIN code set to $ will prompt the user for their voicemail code.
  - **Voicemail Pro in Intuity Emulation mode**
    By default the voicemail code is required for all mailbox access. The first time the mailbox is accessed the user will be prompted to change the password. Also if the voicemail code setting is left blank, the caller will be prompted to set a code when they next access the mailbox. The requirement to enter the voicemail code can be removed by adding a customized user or default collect call flow, refer to the Voicemail Pro manuals for full details. Also Voicemail Pro call flows containing an action where the action's PIN code set to $ will prompt the user for their voicemail code.
  - Codes set through the Voicemail Pro telephone user interface are restricted to valid sequences. For example, attempting to enter a code that matches the mailbox extension, repeat the same number (1111) or a sequence of numbers (1234) are not allowed. If these types of code are required they can be entered through Manager.
- **Voicemail Email:** *Default = Blank (Disabled)*
  Messages for this hunt group can be sent to an email account. Enter the email address, for example jbloggs@bloggs.com. Select the required Voicemail Email mode below. The Voicemail message is received by the email application as a .wav file and played through the speakers of the PC. Refer to the Voicemail Installation & Administration Manual for full details. This entry is not used for IMS.
- **Voicemail On:** *Default = On*
  This field enables the use of voicemail to take messages for the hunt group. It does not disable the voicemail mailbox being used as the target for other functions such as call recording or messages forwarded from other mailboxes.
- **Voicemail Help:** *Default = Off*
  For voicemail systems running IP Office mailbox mode, this option controls whether users retrieving messages are automatically given an additional prompt *"For help at any time press 8."* If switched off, users can still press 8 for help. For voicemail systems running in Intuity emulation mode, this option has no effect. On those systems the default access greeting always includes the prompt *"For help at any time, press *4."*
- **Broadcast:** *Default = Off, Software level = 3.0+.*
  If a voicemail message is left for the hunt group and **Broadcast** is enabled, copies of the message are forwarded to the mailboxes of the individual group members. The original message in the hunt group mailbox is deleted unless it occurred as the result of call recording.
- **Voicemail Email mode:** *Default = Off*
  If a Voicemail Email address has been entered above, select one of the following modes:

- **Off**
  Voicemail messages or notifications are not sent to the email account automatically.
- **Copy**
  A copy of the Voicemail message is sent to the email account.
- **Forward**
  Voicemail messages are sent to the email account and deleted from the Voicemail server.
- **Alert**
  Notification that a new Voicemail message has been received is sent to the email account.

# Hunt Group | Fallback

Fallback settings can be used to make a hunt group unavailable and to set where the hunt group's calls should be redirected at such times. Hunt groups can be manually placed ***In Service***, ***Out of Service*** or in ***Night Service***. Additionally using a time profile, a group can be automatically placed in ***Night Service*** when outside the Time Profile settings.

**Summary: Fallback redirects a hunt group's calls when the hunt group is not available, for example outside normal working hours.  It can be triggered manually or using an associated time profile.**

- **Hunt Group Service States**
  A hunt group can be in one of three states; **In Service**, **Out of Service** and **Night Service**. When In service, calls are presented as normal. In any other state calls are redirected.
- **Call Redirection During Fallback**
  The following options are possible when a hunt group is either Out of Service or in Night Service.
  - **Fallback Group**
    If an **Out of Service Fallback Group** or **Night Service Fallback Group** has been set, calls are redirected to that group.
  - **Voicemail**
    If no fallback group has been set but voicemail is available, calls are redirected to voicemail.
  - **Busy Tone**
    If no fallback group has been set and voicemail is not available, busy tone is returned to calls.
- **Manually Controlling the Service State**
  Manager and or short codes can be used to change the service state of a hunt group. The short code actions can also be assigned to programmable buttons on phones.
  - The  icon is used for a hunt group manually set to Night Service mode.
  - The  icon is used for a hunt group manually set to Out of Service mode.
- **Time Profile**
  A time profile can be associated with the hunt group. When outside the time profile, the hunt group is automatically place into night service. When inside the time profile, the hunt group uses manually selected mode.
  - When outside the time profile and therefore in night service, manual night service controls cannot be used to override the night service. However the hunt group can be put into out of service.
  - When a hunt group is in ***Night Service*** due to a time profile, this is not indicated within Manager.
  - For IP Office 4.0+, time profile operation does not affect hunt groups set to ***Out of Service***.

| Hunt Group \| Fallback | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Time Profile:** *Default = <None> (No automatic night service)*
  This field allows selection of a previously created **Time Profile**. That profile then specifies the times at which it should use the manually selected Service Mode settings. Outside the period defined in the time profile, the hunt group behaves as if set to Night Service mode.

  Please note that when a hunt group is in Night Service due to it associated time profile, this is not reflected by the Service Mode on this tab. Note also that the manual controls for changing a hunt group's service mode cannot be used to take a hunt group out of time profile night service.

- **Out of Service Fallback Group:** *Default = <None> (Voicemail or Busy Tone)*
  This field sets the alternate hunt group destination for calls when this hunt group is in ***Out of Service*** mode. If left blank, calls are redirected to voicemail if available or otherwise receive busy tone.

- **Night Service Fallback Group:** *Default = <None> (Voicemail or Busy Tone)*
  This field sets the alternate hunt group destination for calls when this hunt group is in ***Night Service*** mode. If left blank, calls are redirected to voicemail if available or otherwise receive busy tone.

- **Service Mode:** *Default = In Service*
  This field is used to manually select the current service mode for the hunt group.

  - **Out of Service**
    When selected, calls are redirected using the **Out of Service Fallback Group** setting. This setting can also be manually controlled using the short code and button programming features **Set Hunt Group Out of Service** and **Clear Hunt Group Out of Service**.

  - or **: In Service**
    When selected the hunt group is enabled. This is the default mode.

  - **Night Service**
    When selected, calls are redirected using the **Night Service Fallback Group** setting. This setting can also be manually controlled using the short code and button programming features **Set Hunt Group Night Service** and **Clear Hunt Group Night Service**.

### Hunt Group Fallback Controls

<table>
<tr><th colspan="2">Hunt Group Fallback</th></tr>
<tr><td>Manager</td><td>Hunt group fallback selection is done through the Hunt Group | Fallback tab.<br>A time profile if required is set through the Time Profile | Time Profile tab.</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>Set Hunt Group Night Service</td><td>✓</td><td>*20*N#</td><td>✓- Toggles.</td></tr>
<tr><td>Clear Hunt Group Night Service</td><td>✓</td><td>*21*N#</td><td>✓</td></tr>
<tr><td>Set Hunt Group Out of Service</td><td>✗</td><td>✗</td><td>✓- Toggles.</td></tr>
<tr><td>Clear Hunt Group Out of Service</td><td>✗</td><td>✗</td><td>✓</td></tr>
</table>
Note that for a hunt group using a time profile, these controls only are only applied when the hunt group is within the specified time profile period. When outside its time profile, the hunt group is in night service mode and cannot be overridden.</td></tr>
<tr><td>Phone Manager</td><td>There are no specific controls for the operation of hunt group fallback.</td></tr>
<tr><td>SoftConsole</td><td>There are no specific controls for the operation of hunt group fallback.</td></tr>
<tr><td>Voicemail</td><td>There are no specific controls for the operation of hunt group fallback.</td></tr>
</table>

# Hunt Group | Queuing

- **When is a Call Queued**
  The definition of when a call is in a queue can vary:
  - For pre-4.0 IP Office systems, calls to a hunt group were only queued when the number of calls waiting exceeded the number of available hunt group members that could be ringing. Using that definition, calls that were actually ringing were not regarded as queued.
  - For IP Office 4.0+, any calls waiting to be answered at a hunt group are regarded as being queued. The **Normalise Queue Length** control allows selection of whether features that are triggered by the queue length should include or exclude ringing calls.
- **Additional Calls**
  Once one call is queued, any further calls are also queued. When an available hunt group member becomes idle, the first call in the queue is presented.
- **How Many Calls Can be Queued?**
  Calls are added to the queue until the hunt group's **Queue Limit**, if set, is reached.
  - When the queue limit is reached, any further calls are redirected to the hunt group's voicemail if available.
  - If voicemail is not available excess calls receive busy tone. An exception to this are analog trunk and T1 CAS trunk calls which will remain queued regardless of the queue limit if no alternate destination is available.
  - If an existing queued call is displaced by a higher priority call, the displaced call will remain queued even if it now exceeds the queue limit.
- **Queue Announcements**
  The method of hunt group announcements depends on the IP Office software level:
  - For pre-4.0 IP Office systems with Voicemail Pro or Voicemail Lite, announcements are applied to queued calls.
  - For IP Office 4.0+, hunt group announcements are separate from queuing. Announcements can be used even if queuing is turned off and are applied to ringing and queued calls. See **Hunt Group | Announcements**.
- **Queue Monitoring**
  There are several methods of displaying a hunt group queue.
  - **Group Button**
    On phones, with programmable buttons, the Group function can be assigned to monitor a specified group. The button indicates when there are calls ringing within the group and also when there are calls queued. The button can be used to answer the longest waiting call.
  - **Phone Manager and SoftConsole**
    Both these applications can display queue monitors for selected hunt groups, 2 using Phone Manager, 7 using SoftConsole. This requires the hunt group to have queuing enabled. These queues can be used to answer calls.
- **What Happens When A Hunt Group Members Becomes Available**
  When a hunt group member becomes available, the first call in the queue is presented to that member. If several members become available, the first call in the queue is simultaneously presented to the all the free members.
- **Overflow Calls**
  Calls that overflow at counted in the queue of the original hunt group from which they overflow and not that of the hunt group to which they overflow. This affects the **Queue Limit** and **Calls in Queue Threshold**.

| Hunt Group \| Queuing | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Queuing On:** *Default = On*

  This settings allows calls to this hunt group to be queued. The normal icon is replaced .

- **Queue Limit:** *Default = No Limit. Range = No Limit, 1 to 999 calls.*
  This setting can be used to limit the number of calls that can be queued. Calls exceeding this limit are passed to voicemail if available or otherwise receive busy tone. This value is affected by **Normalize Queue Length** setting.

  - If voicemail is not available excess calls receive busy tone. An exception to this is analog trunk and T1 CAS trunk calls which will remain queued regardless of the queue limit if no alternate destination is available. This is due to the limited call status signalling supported by those trunks which would otherwise create scenarios where the caller has received ringing from the local line provider and then suddenly gets busy from the IP Office, creating the impression that the call was answered and then hung up.
  - If priority is being used with incoming call routes, high priority calls are place ahead of lower priority calls. If this would exceed the queue limit the limit is temporarily increased by 1.
  - If an existing queued call is displaced by a higher priority call, the displaced call will remain queued even if it now exceeds the queue limit.

- **Normalize Queue Length:** *Default = Off, Software level = 4.0+.*
  Prior to IP Office 4.0 the call queue only included calls waiting to ring and did not include calls actually ringing. For IP Office 4.0+ calls both waiting to ring and ringing are regarded as being queued. This therefore affects the use of the **Queue Limit** and **Calls in Queue Alarm** thresholds. If **Normalize Queue Length** is enabled, the number of hunt group members logged in and not on DND is added to those thresholds.

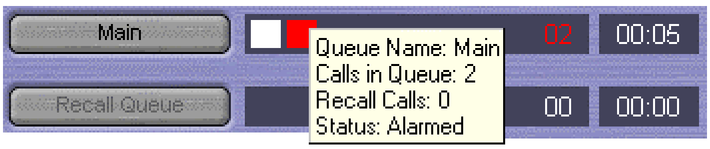
  - **Example:**
    A customer has two products that it is selling through a call center with 10 available agents; one product with a $10 margin and one with a $100 margin. Separate hunt groups with the same 10 members are created for each product.

    - The $100 product has a **Queue Limit** of 5 and **Normalize Queue Length** is on. The maximum number of $100 calls that can be waiting to be answered will be 15 (10 ringing/connected + 5 waiting to ring).
    - The $10 product has a **Queue Limit** of 5 and **Normalize Queue Length** is off. The maximum number of $10 calls that can be waiting to be answered is 5 (5 ringing/connected).

- **Queue Ring Time (secs):** *Default = 10 seconds. Range = 0 to 99999 seconds, Software level = Up to 3.2 only.*
  On systems with Voicemail Lite or Voicemail Pro, the voicemail system can provide announcements to queued callers. This setting controls the time before the first queued announcement is played to a queued caller. For IP Office 4.0+ this has been replaced by the **Hunt Group | Announcement** tab controls.

- **Calls In Queue Alarm:** *Software Level = 4.1+.*
  The IP Office can be set to send an alert to a specified extension when the number of calls queued for the hunt group reaches the specified threshold.
  - **Calls In Queue Threshold:** *Default = 0 (Off), Range = 0 to 99, Software level = 4.1+.*
    Alerting is triggered when the number of queued calls reaches this threshold. Alerting will stop only when the number of queued calls drops back below this threshold. This value is affected by **Normalize Queue Length** setting.
  - **Analog Extension to Notify:** *Default = <None>, Software Level = 4.1+.*
    This should be set to the extension number of a user associated with an analogue extension. The design intention is that the analogue extension port should be connected to a loud ringer or other alerting device and is not used for making or receiving calls. The user should not be a member of any hunt groups or the queuing alarm target for any other hunt group queue. Attempting to answer the alerting device will just give ring tone. The alert does not follow user settings such as forwarding, follow me, DND, call coverage, etc or receive ICLID information.

**Hunt Group Queue Settings**

<table>
<tr><th colspan="2">Hunt Group Queue Settings</th></tr>
<tr><td>Software Level</td><td>Hunt group queuing is enabled using the Queuing On option on the Hunt Group | Queuing tab. When enabled, the icon is used for the hunt group.</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>Group</td><td>✗</td><td>✗</td><td>✓</td></tr>
</table></td></tr>
<tr><td>Phone Software Level</td><td>Phone manager Pro can be used to monitor up to two hunt group queues. This is configured by clicking and then on the Queue ID tab selecting the two hunt groups. During normal operation the Phone Manager user then has access to a Queue tab which is automatically given focus when calls become queued.</td></tr>
<tr><td>SoftConsole</td><td>SoftConsole can display up to 7 hunt group queues (an eight queue is reserved for recall calls). They are configured by clicking and selecting the Queue Mode tab. For each queue alarm threshold can be set based on number of queued calls and longest queued call time. Actions can then be selected for when a queue exceeds its alarm threshold; Automatically Restore SoftConsole, Ask me whether to restore SoftConsole or Ignore the Alarm.


Within the displayed queues, the number of queued calls is indicated and the time of the longest queued call is shown. Exceeding an alarm threshold is indicated by the queue icons changing from white to red. The longest waiting call in a queue can be answered by clicking on the adjacent button.</td></tr>
</table>

# Hunt Group | Voice Recording

When the IP Office system has a Voicemail Pro server installed, that server can be used for automatic recording of external calls. By default call recordings are placed into the hunt group's mailbox. Call recording also requires available conference resources similar to a 3-way conference.

- IP Office 4.0+ introduces the following changes to hunt group recording:
  - Calls to and from IP devices including those using Direct media can be recorded.
  - Hunt group recording stops if the call is transferred to another user who is not a member of the hunt group.
  - Calls parked or held by the user, pause recording until a member of the hunt group unparks the call or takes it off hold.
- For IP Office 4.1+, a destination mailbox other than the hunt group's own mailbox can be specified as the destination for recordings.

| Hunt Group \| Voice Recording | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Record Inbound:** *Default = None*
  Select whether inbound external calls to group members should be recorded. Options are **On**, **Mandatory** and then various percentages of the calls received by the group members.
  - **On**
    Record the call if possible.
  - **Mandatory**
    If not possible to record, return busy tone to the caller.
- **Record Time Profile:** *Default = <None> (Any time)*
  Used to select a time profile during which automatic call recording setting above apply.
- **Auto Recording Mailbox:** *Default = <hunt group's own mailbox>, Software level = 4.1+.*
  Sets the mailbox into which automatically triggered recordings are placed.
- **Voice Recording Library:** *Default = Off, Software level = 3.0+.*
  When checked, this setting overrides the **Auto Recording Mailbox** setting and marks the recording for collection by the ContactStore for IP Office application. Note that Voicemail Pro still performs the actual recording.

# Hunt Group | Announcements

This tab is new for IP Office 4.0 and higher. Unlike pre-4.0 IP Office systems, this method of using announcements is independent of hunt group queuing.

Announcements are played to callers waiting to be answered. This includes callers being presented to hunt group members, ie. ringing, and callers queued for presentation.

- IP Office 4.0+ supports hunt group announcement using Voicemail Pro, Voicemail Lite and Embedded Voicemail.
- If no voicemail channel is available for an announcement when required, the announcement is not played.
- In conjunction with Voicemail Pro, IP Office 4.0+ allows a number of voicemail channels to be reserved for announcements. See **System | Voicemail**.
- With Voicemail Pro, the announcement can be replace by the action specified in a Queued (1st announcement) or Still Queued (2nd announcement) start point call flow for the hunt group. Refer to the Voicemail Pro Installation and Maintenance documentation for details.
- Calls can be answered while hearing announcements. If it is a mandatory requirement that announcements should be heard before a call is answered then a Voicemail Pro call flow should be used before the call is presented to the hunt group.
- If a call is rerouted to a hunt group's Night Service Group or Out of Service Fallback Group, the announcements of the new group are applied.
- If a call overflows, the announcements of the original group are still applied, not those of the overflow group.
- For announcements to be used effectively, the hunt group's **Voicemail Answer Time** must be extended or **Voicemail On** must be deselected.

**Recording the Hunt Group Announcement**
Voicemail Lite and Voicemail Pro provide the default announcement *"I'm afraid all the operators are still busy but please hold and you will be transferred when somebody becomes available".* This default is used for announcement 1 and announcement 2 if no specific hunt group announcement has been recorded. Embedded Voicemail does not provide any default announcement.

The maximum length for announcements is 10 minutes. New announcements can be recorded using the following methods:

- **Voicemail Lite**
  Access the hunt group mailbox and press **3**. Then press either **3** to record the 1st announcement for the hunt group or **4** to record the 2nd announcement for the hunt group.
- **Voicemail Pro - IP Office Mode**
  This is the same as Voicemail Lite above.
- **Voicemail Pro - Intuity Emulation Mode**
  There is no mechanism within the Intuity telephony user interface (TUI) to record hunt group announcements. To provide custom announcements, hunt group queued and still queued start points must be configured with Voicemail Pro with the required prompts played by a generic action.
- **Embedded Voicemail**
  Embedded Voicemail does not include any default announcement or method for recording an announcement. The Record Message short code feature is provided to allow the recording of hunt group announcements. The telephone number field of short codes using this feature requires the hunt group number followed by either ".1" for announcement 1 or ".2" for announcement 2. For example, for a hunt group with extension number 300, the short codes ***1N# | Record Message | N".1"** and ***2N# | Record Message | N".2"** could be used to allow recording of the announcements by dialing *1300# and *2300#.

| Hunt Group \| Announcements | |
|---|---|
| Control Unit | SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓. |
| Software Level | 2.1 ✗, 3.0DT ✗, 3.0 ✗, 3.1 ✗, 3.2 ✗, 4.0 ✓, 4.1 ✓. |
| Mergeable | ✓. |

- **Announcements On:** *Default = Off.*
  This setting enables or disables announcements for the hunt group.
- **Wait before 1st announcement:** *Default = 10 seconds. Range = 0 to 9999 seconds.*
  This setting sets the time delay from the calls presentation to the hunt group, either ringing or queued, after which the first announcement should be played to the caller. If **Synchronize Calls** is selected (see below), the actual wait may vary between immediate (0) and the wait interval plus the announcement length.
- **Flag call as answered:** *Default = Off.*
  Normally for reporting purposes with the IP Office CCC application, a call is not regarded as answered until it has been answered by a person or by a Voicemail Pro action set to **Flag call as answered**. This setting allows the call to be marked as answered once the caller has heard the first announcement.
- **Post announcement tone:** *Default = Music on hold.*
  Following the first announcement, you can select whether the caller should hear ***Music on Hold***, ***Ringing*** or ***Silence*** until answered or played another announcement.
- **2nd Announcement:** *Default = On.*
  If selected, a second announcement can be played to the caller if they have still not been answered.
- **Wait before 2nd announcement:** *Default = 20 seconds. Range = 0 to 9999 seconds.*
  This setting sets the wait between the 1st and the 2nd announcement. If **Synchronize Calls** is selected (see below), the actual wait may vary between immediate (0) and the wait interval plus the announcement length.
- **Repeat last announcement:** *Default = On.*
  If selected, the last announcement played to the caller is repeated until they are answered or hang-up.
- **Wait before repeat:** *Default = 20 seconds. Range = 0 to 9999 seconds.*
  If **Repeat last announcement** is selected, this setting sets wait applied between each repeat of the last announcement. If **Synchronize Calls** is selected, this value is greyed out and set to match the **Wait before 2nd announcement** setting.

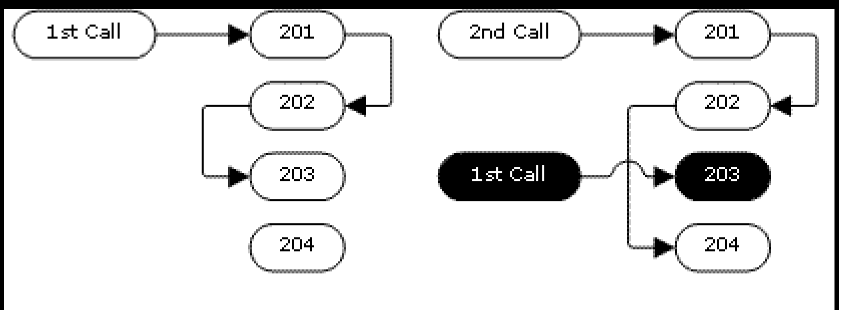
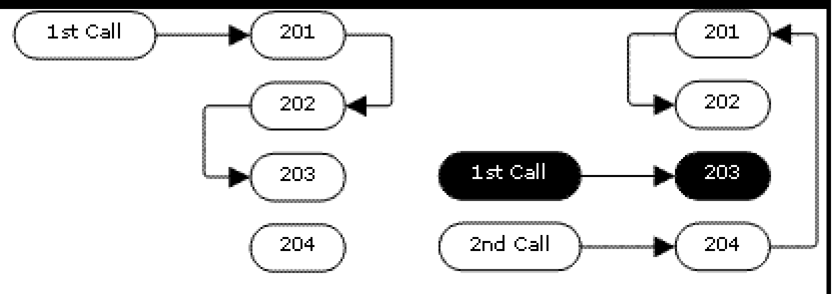
- **Synchronize calls:** *Default = Off*
  This option can be used to restrict how many voicemail channels are required to provide the groups announcements.
  - When **Synchronize calls** is off (the default), the announcement pattern is followed individually for each call. This requires a separate voicemail channel each time an announcement is played to each caller. While this ensure accurate following of the timing of the announcement pattern, it does not make efficient use of voicemail channels.
  - When **Synchronize calls** is on, if a required announcement is already being played to another caller, pending callers for the same announcement wait until it has been completed and can be restarted. In addition when a caller has waited for the set wait period and the announcement is started, any other callers waiting for the same announcement hear it even if they have not waited for the full wait period. Using this the maximum number of voicemail channels ever needed by the hunt group for announcements is 1 or 2 depending on the number of selected announcements. When on, the wait for each announcement may vary.
    - **Note: Interaction with Voicemail Pro Queued and Still Queued Start Points**
      If either custom Queued or Still Queued start point call flows are being used for the announcements, when **Synchronize Calls** is enabled those call flows will support the playing of prompts only. Voicemail Pro actions such as **Speak ETA**, **Speak Position**, **Menu**, **Leave Mail**, **Transfer** and **Assisted Transfer**, etc. are not supported.

# Hunt Group Operation

## Hunt Group Types

At its most basic, a hunt groups settings consist of a hunt group name, an extension number, a list of hunt group members and a hunt type selection. It is the last two settings which determine the order in which incoming calls are presented to hunt group members.

The available hunt types are; **Collective**, **Sequential**, **Rotary** and **Longest Waiting**. These work are follows:

<table>
<tr><th colspan="2">Collective Group (formerly Group Group)</th></tr>
<tr><td>An incoming call is presented simultaneously to all the available hunt group members.</td><td>
</td></tr>
<tr><th colspan="2">Sequential Group (formerly Hunt or Linear Group)</th></tr>
<tr><td>An incoming call is presented to the first available member in the list. If unanswered, it is presented to the next available member in the list.<br><br>The next incoming call uses the same order. It is presented to the available members starting again from the top of the list.</td><td>
</td></tr>
<tr><th colspan="2">Rotary Hunt Type (formerly Circular Group)</th></tr>
<tr><td>This hunt type operates similarly to Sequential. However the starting point for call presentation is the first available member after the last member to answer a call.</td><td>
</td></tr>
<tr><th colspan="2">Longest Waiting Hunt Type (formerly Idle or Most Idle)</th></tr>
<tr><td colspan="2">This hunt type does not present calls to hunt group members in the order that they are listed. It presents calls using the order of how long the available hunt group members have been idle.<br><br>An incoming call is first presented to the available member who has been idle the longest. If unanswered it is presented to the next longest idle member.</td></tr>
</table>

# Call Presentation

**Summary: Calls are presented to each available hunt group member in turn. If having been presented to all the available members, none answers, the call is redirected to voicemail if available, otherwise it continues to be presented to the next available member.**

In addition to the summary, options exist to have calls queued or to have calls redirected to overflow groups.

- **First and Next Available Members**
  The first available member to which a call is presented and the order of the next available members to which a call is presented are determined by the hunt group's **Hunt Type** setting.
- **Additional Calls**
  When additional calls are waiting to be presented, additional available hunt group members are alerted using the hunt group type. The way additional calls are presented if there are available members depends on the IP Office software level.
  - **Pre-IP Office 4.0**
    Additional calls ring around the group separately. This means that additional calls may be answered ahead of the first call.
  - **IP Office 4.0 and Higher**
    When any member answers a call it will be the first waiting call that is answered.
- **No Available Members**
  If the number of incoming calls exceeds the number of available members to which calls can be presented, the following actions are usable in order of precedence.
  - **Queuing**
    If queuing has been enabled for the hunt, it is applied to the excess calls up to the limits specified for the number of queued calls or length of time queued.
  - **Voicemail**
    If voicemail has been enabled for the hunt group, excess calls are directed to voicemail.
  - **Busy Tone**
    Busy tone is returned to the excess calls (except analogue and T1 CAS calls which remain queued).
- **No Answer Time**
  This value is used to determine how long a call should ring at a hunt group member before being presented to the next available hunt group member. The **System | Telephony | No Answer Time** setting is used unless a specific **Hunt | Hunt Group | No Answer Time** is set.
- **Voicemail**
  If voicemail is being used, if having been presented to all the available group members the call is still not answered then it goes to voicemail. For IP Office 4.0 and higher, the call will also go to voicemail when the hunt groups **Voicemail Answer Time** is exceeded.
  - For IP Office 3.2 and earlier, when voicemail was invoked, the mailbox of whichever hunt group was currently handling the call was used, for example the mailbox of the overflow or night service hunt group might be used if the call had gone from the original hunt group to an overflow or night server hunt group. For IP Office 4.0 and higher, the mailbox of the originally targetted hunt group is used even if the call has overflowed or gone to a night server hunt group.
- **Calls Not Being Answered Quick Enough - Overflow**
  In addition to ringing at each available member for the **No Answer Time**, a separate **Overflow Time** can be set. When a calls total ring time against the group exceeds this, the call can be redirected to an overflow group or groups.

- **No Available Member Answers**
  If a call has been presented unanswered to all the available members, either of two actions can be applied. If voicemail is available, the call is redirected to voicemail. If otherwise, the call will continue being presented to hunt group members until answered or, if set, overflow is used.
- **Call Waiting**
  For hunt groups using the ***Group*** hunt type, call waiting can be used.

# Hunt Group Member Availability

**Summary: Details when a hunt group member is seen as being available to be presented a hunt group call.**

The Hunt Group settings within Manager list those users who are members of the hunt group and therefore may receive calls directed to that hunt group. However there are a range of factors that can affect whether a particular hunt group member is available to take hunt group calls at any time.

- **Existing Connected Call**
  Users with an existing connected call are not available to further hunt group calls. This is regardless of the type of connected call, whether the user has available call appearance buttons or is using call waiting.
  - **Hunt Group Call Waiting**
    For Group type hunt groups, call waiting can be enabled.
- **Logged On/Logged Off**
  IP Office allows user's to log on and off extensions, a process known as 'hot desking'. Whilst a user is logged off they are not available to receive hunt group calls.
- **Membership Enabled/Disabled**
  The IP Office provides controls to temporarily disable a users' membership of a hunt group. Whilst disabled, the user is not available to receive calls directed to that hunt group.
- **Do Not Disturb**
  This function is used by users to indicate that they do not want to receive any calls. This includes hunt group calls. In call center environments this state is also known as 'Busy Not Available'. See Do Not Disturb.
- **Busy on Held**
  When a user has a held call, they can receive other calls including hunt group calls. The Busy on Held settings can be used to indicate that the user is not available to further calls when they have a held call.
- **Forward Unconditional**
  Users set to Forward Unconditional are by default not available to hunt group calls. The IP Office allows the forwarding of hunt group calls to be selected as an option.
- **Idle /Off Hook**
  The hunt group member must be on-hook and idle in order to receive hunt group call ringing.
- **No Available Members**
  If queuing has been enabled, calls will be queued. If queuing has not been enabled, calls will go to the overflow group if set, even if the overflow time is not set or is set to 0. If queuing is not enabled and no overflow is set, calls will go to voicemail. If voicemail is not available, external calls go to the incoming call routes fallback destination while internal calls receive number unobtainable indication.

## Hunt Group Member Availability Settings

<table>
<tr><th colspan="2">Hunt Group Member Availability Settings</th></tr>
<tr><td>Software Level</td><td>Forwarding and do not disturb controls for a user are found on the User | Forwarding and User | DND tabs.<br><br>Enabling and disabling a users hunt group membership is done by ticking or unticking the user entry in the hunt group's extensions list on the Hunt Group | Hunt Group tab.</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>Hunt Group Enable</td><td>✓</td><td>✗</td><td>✓HGEna - Toggles.</td></tr>
<tr><td>Hunt Group Disable</td><td>✓</td><td>✗</td><td>✓HGDis</td></tr>
<tr><td>Forward Hunt Group On</td><td>✓</td><td>✓-*50</td><td>✓FwDH+ - Toggles</td></tr>
<tr><td>Forward Hunt Group Off</td><td>✓</td><td>✓-*51</td><td>✓FwDH-</td></tr>
<tr><td>Busy on Held</td><td>✓</td><td>✗</td><td>✓BusyH</td></tr>
<tr><td>Do Not Disturb On</td><td>✓</td><td>✓-*08</td><td>✓DNDOn - Toggles</td></tr>
<tr><td>Do Not Disturb Off</td><td>✓</td><td>✓-*09</td><td>✓DNDOf</td></tr>
<tr><td>Extn Login</td><td>✓</td><td>✓-*35*N#</td><td>✓Login</td></tr>
<tr><td>Extn Logout</td><td>✓</td><td>✓-*36</td><td>✓Logof</td></tr>
</table></td></tr>
<tr><td>Phone Software Level</td><td>DND, Forwarding and Busy on Held can all be controlled through Phone Manager. They are accessed by clicking and then selecting the Do Not Disturb, Forwarding or Telephone tabs respectively.<br><br>Phone Manager Pro users can select agent mode by clicking , selecting the Agent Mode tab and selecting Agent Mode. In this mode, Phone Manager provides icons for Busy Wrap Up (Hunt group disable) and Busy Not Available (DND). It also allows individual selection of which group memberships are enabled.<br><br>Phone Manager can also be used to log on and log off when the application is started or stopped.</td></tr>
<tr><td>SoftConsole</td><td>A SoftConsole user can view and edit a user's settings. Through the directory, select the required user. Their current status including DND, Logged In and hunt group membership states are shown and can be changed. Forwarding settings can be accessed by then selecting Forwarding.</td></tr>
</table>

# Example Hunt Group

The follow are simple examples of how a department might use the facilities of an IP Office hunt group.

| 1. Basic Hunt Group | |
|---|---|
| **Scenario** | The Sales department want all sales related calls to be presented first to Jane, then Peter and finally Anne. |
| **Actions** | 1. Create a hunt group named ***Sales*** and assign it an extension number.<br>2. Set the **Hunt Type** to ***Sequential***.<br>3. Add Jane, Peter and Ann to the **Extension List** in that order.<br>4. Turn off queuing on the **Queuing** tab and voicemail on the **Voicemail** tab.<br>5. Route relevant calls to the Sales group by selecting it as the destination in the appropriate Incoming Call Routes. |
| **Results** | Any call received by the Sales hunt group is first presented to Jane if she is available. If Jane is not available or does not answer within 15 seconds the call is presented to Peter. If Peter is not available or does not answer within 15 seconds the call goes Anne. Since voicemail is not on, the call will continue to be presented around the group members in that order until it is answered or the callers hangs up. |

| 2. Adding Voicemail Support | |
|---|---|
| **Scenario** | An IP Office voicemail server (Voicemail Lite or Voicemail Pro) has now been added to the IP Office system. The Sales department wants to use it to take messages from unanswered callers. When messages are left, they want Jane to receive message waiting indication. |
| **Actions** | 1. Open the ***Sales*** hunt group settings and select **Voicemail On** on the **Voicemail** tab.<br>2. Select the **User** settings for Jane. On the **Source Numbers** tab, add the entry ***HSales***. |
| **Results** | Once a call to the Sales group has been presented to all the available members, if it is still unanswered then the call will be redirected to the group's voicemail mailbox to leave a message. When a message has been left, the message waiting indication lamp on Jane's phone is lit. |

| 3. Using the Queuing Facility | |
|---|---|
| **Scenario** | The Sales department now wants calls queued when no one is available to answer. However if the number of queued calls exceeds 3 they then want any further callers directed to voicemail. |
| **Actions** | 1. Open the ***Sales*** hunt group settings and select **Queuing On** on the **Queuing** tab.<br>2. Set the **Queue Limit** to ***3***. |
| **Results** | When the Sales group are all on calls or ringing, any further calls to the group are queued and receive queuing announcements from the voicemail server. When the number of queued calls exceeds 3, any further calls are routed to the group's voicemail mailbox. |

| 4. Using Out of Service Fallback | |
|---|---|
| **Scenario** | During team meetings, the Sales department want their calls redirected to another group, for this example Support. |
| **Actions** | 1. Open the ***Sales*** hunt group settings and select the **Fallback** tab. In the **Out of Service Fallback Group** field select the ***Support*** group.<br>2. Create a system short code ***98 / 300 / Set Hunt Group Out of Service**.<br>3. Create a system short code ***99 / 300 / Clear Hunt Group Out of Service**. |
| **Results** | Prior to team meetings, dialing *98 puts the Sales group into out of service mode. Its calls are then redirected to the Support group. Following the meeting, dialing *99 puts the Sales group back In Service. |

| 5. Using a Night Service Time Profile | |
|---|---|
| **Scenario** | Outside their normal business hours the Sales department want their group calls automatically sent to voicemail. This can be done using a time profile and leaving the Night Service Fallback Group setting blank. |
| **Actions** | 1. Create a **Time Profile** called ***Sales Hours*** and in it enter the time during which the Sales department are normally available.<br>2. Open the ***Sales*** hunt group settings and select the **Fallback** tab.<br>3. In the **Time Profile** field select ***Sales Hours***. |
| **Results** | Outside the normal business hours set in the time profile, the Sales hunt group is automatically put into Night Service mode. Since no Night Service Fallback Group has been set, calls are redirected to voicemail. |

# CBC/CCC Agents and Hunt Groups

The use of and reporting on hunt groups is a key feature of call center operation. For IP Office, reporting is provided through the Compact Business Center (CBC) or Compact Contact Center (CCC) applications.

In order for these applications to provide hunt group and hunt group user (agent) reports, the following rules apply:

- The hunt group names must be restricted to a maximum of 12 characters.
- The hunt group and user extension numbers should be a maximum of 4 digits.
- Hunt group members should be given a **Login Code** and set to **Force Login**.
- The agent state Busy Not Available is equivalent to Do Not Disturb. The agent state Busy Wrap Up is equivalent to hunt group disable.

# Short Code Settings

## Short Code | Short Code

This form is used to create System Short Codes. System short codes can be dialed by all IP Office users. However the system short code is ignored if the user dialing matches a user short code or user restriction short code. For full details on short code usage and parameter see the section Short Codes.

| Short Code \| Short Code | |
|---|---|
| **Control Unit** | **SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓.** |
| **Software Level** | **2.1 ✓, 3.0DT ✓, 3.0 ✓, 3.1 ✓, 3.2 ✓, 4.0 ✓, 4.1 ✓.** |
| **Mergeable** | ✓. |

- **Short Code**
  The dialing digits used to trigger the short code. Maximum length 31 characters.
- **Telephone Number**
  The number dialed by the short code or parameters for the short code feature. This field can contain numbers and characters. For example, it can contain Voicemail Pro start point names, user names, hunt group names and telephone numbers (including those with special characters). Maximum length 31 characters.
  - **Dialing Complete**
    The majority of North-American telephony services use en-bloc dialing. Therefore the use of a **;** is recommended at the end of all dialing short codes that use an *N*. This is also recommended for all dialing where secondary dial tone short codes are being used.
- **Line Group ID:** *Default = 0*
  For short codes that result in the dialing of a number, that is short codes with a Dial feature, this field is used to enter the initially routing destination of the call. The drop down can be used to select the following from the displayed list:
  - **Outgoing Group ID**
    The **Outgoing Group ID**'s current setup within the IP Office configuration are listed. If an **Outgoing Group ID** is selected, the call will be routed to the first available line or channel within that group.
  - **ARS**
    For IP Office 4.0+ systems, the ARS entries currently configured in the system are listed. If an ARS entry is selected, the call will be routed by the setting within that ARS entry. Refer to ARS Overview.
- **Feature**
  Select the action to be performed by the short code. See Short Code Features for a listing.
- **Locale:** *Default = Blank*
  For short codes that route calls to voicemail, this field can be used to set the prompts locale that should be used if available on the voicemail server. See
- **Force Account Code:** *Default = Off.*
  For short codes that result in the dialing of a number, this field trigger the user being prompted to enter a valid account code before the call is allowed to continue.
- **Force Authorization Code:** *Default = Off*
  This option is only shown on systems where authorization codes have been enabled, see **Authorization Codes**. If selected, then for short codes that result in the dialing of a number, the user is required to enter a valid authorization code in order to continue the call.

# Service Settings

## Service Form Overview

Services are used to configure the settings required when a user or device on the IP Office LAN needs to connect to a off-switch data service such as the Internet or another network. Services can be used when making data connections via trunk or WAN interfaces.

Once a service is created, it can be used as the destination for an IP Route entry. One service can also be set as the **Default Service**. That service will then be used for any data traffic received by the IP Office for which no IP Route is specified.

The IP Office supports three types of service:

- **Normal Service**
  This type of service should be selected when for example, connecting to an ISP.
- **WAN Service**
  This type of service is used when creating a WAN link. A User and RAS Service will also be created with the same name. These three entries are automatically linked and each open the same form. Note however, that this type of Service cannot be used if the Encrypted Password option is checked. In this case the RAS Service name must match the Account Name. Therefore either create each entry manually or create an Intranet Service.
- **Intranet Service**
  This type of service can be selected to automatically create a User with the same name at the same time. These two entries are linked and will each open the same form. The User's password is entered in the Incoming Password field at the bottom on the Service tab. An Intranet Services shares the same configuration tabs as those available to the WAN Service.

# Service | Service

| Service \| Service | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Name**
  The name of the service. It is recommended that only alphanumeric characters be used.
- **Account Name**
  The User Name that is used to authenticate the connection. This is provided by the ISP or remote system.
- **Password:** *Default = Blank*
  Enter the password that is used to authenticate the connection. This is provided by the ISP or remote system.
- **Telephone Number:** *Default = Blank*
  If the connection is to be made via ISDN enter the telephone number to be dialed. This is provided by the ISP or remote system.
- **Firewall Profile:** *Default = Internet01 if present, otherwise <None>*
  From the list box select the Firewall Profile that is used to allow/disallow protocols through this Service.
- **Encrypted Password:** *Default = Off*
  When enabled the password is authenticated via CHAP (this must also be supported at the remote end). If disabled, PAP is used as the authentication method.
- **Default Route:** *Default = Off*
  When enabled this Service is the default route for data packets unless a blank IP Route has been defined in the IP Office **IP Routes**. A green arrow appears to the left of the Service in the Configuration Tree. Only one Service can be the default route. If disabled, a route must be created under IP Route.
- **Incoming Password:** *Default = Blank*
  Shown on WAN and Intranet services. Enter the password that will be used to authenticate the connection from the remote Control Unit. (If this field has appeared because you have created a Service and User of the same name, this is the password you entered in the User's Password field).

# Service | Bandwidth

These options give the ability to make ISDN calls between sites only when there is data to be sent or sufficient data to warrant an additional call. The calls are made automatically without the users being aware of when calls begin or end. Using ISDN it is possible to establish a data call and be passing data in less that a second. Note: the system will check Minimum Call Time first, then Idle Period, then the Active Idle Period.

| Service \| Bandwidth | |
|---|---|
| **Control Unit** | **SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓.** |
| **Software Level** | **2.1 ✓, 3.0DT ✓, 3.0 ✓, 3.1 ✓, 3.2 ✓, 4.0 ✓, 4.1 ✓.** |
| **Mergeable** | ✓. |

- **Minimum No of Channels:** *Default = 1. Range = 1 to 30.*
  Defines the number of channels used to connect for an outgoing connection. The initial channel must be established and stable, before further calls are made.
- **Maximum No of Channels:** *Default = 1. Range = 1 to 30.*
  Defines the maximum number of channels to can be used. This field should contain a value equal to or greater than the Minimum Channels field.
- **Extra BW Threshold:** *Default = 50%. Range = 0 to 100%.*
  Defines the utilization threshold at which extra channels are connected. The value entered is a %. The % utilization is calculated over the total number of channels in use at any time, which may be one, two etc.
  - For example, if Minimum Channels set to 1, Maximum Channels set to 2 and Extra Bandwidth set to 50 - once 50% of first channel has been used the second channel are connected.
- **Reduce BW Threshold:** *Default = 10%. Range = 0 to 100%.*
  Defines the utilization threshold at which additional channels are disconnected. The value entered is a %. Additional calls are only dropped when the % utilization, calculated over the total number of channels in use, falls below the % value set for a time period defined by the Service-Idle Time. The last call (calls - if Minimum Calls is greater than 1) to the Service is only dropped if the % utilization falls to 0, for a time period defined by the Service-Idle Time. Only used when 2 or more channels are set above.
  - For example, if Minimum Channels set to 1, Maximum Channels set to 2 and Reduce Bandwidth is set to 10 - once the usage of the 2 channels drops to 10% the number of channels used is 1.
- **Callback Telephone Number:** *Default = Blank*
  The number that is given to the remote service, via BAP, which the remote Control Unit then dials to allow the bandwidth to be increased. Incoming Call routing and RAS Services must be appropriately configured.
- **Idle Period (secs):** *Default = 10 seconds. Range = 0 to 999999 seconds.*
  The time period, in seconds, required to expire after the line has gone idle. At this point the call is considered inactive and is completely closed.
  - For example, the 'Idle Period' is set to X seconds. X seconds before the 'Active Idle Period' timeouts the Control Unit checks the packets being transmitted/received, if there is nothing then at the end of the 'Active Idle Period' the session is closed & the line is dropped. If there are some packets being transmitted or received then the line stays up. After the 'Active Idle Period' has timed out the system performs the same check every X seconds, until there are no packets being transferred and the session is closed and the line dropped.

- **Active Idle Period (secs):** *Default = 180 seconds. Range = 0 to 999999 seconds.*
  Sets the time period during which time the line has gone idle but there are still active sessions in progress (for example an FTP is in process, but not actually passing data at the moment). Only after this timeout will call be dropped.
  - For example, you are downloading a file from your PC and for some reason the other end has stopped responding, (the remote site may have a problem etc.) the line is idle, not down, no data is being transmitted/ received but the file download session is still active. After the set time period of being in this state the line will drop and the sessions close. You may receive a remote server timeout error on your PC in the Browser/FTP client you were using.
- **Minimum Call Time (secs):** *Default = 60 seconds. Range = 0 to 999999 seconds.*
  Sets the minimum time that a call is held up after initial connection. This is useful if you pay a minimum call charge every time a call is made, no matter the actual length of the call. The minimum call time should be set to match that provided by the line provider.
- **Extra BW Mode:** *Default = Incoming Outgoing*
  Defines the mode of operation used to increases bandwidth to the initial call to the remote Service.
  - **Outgoing Only**
    Bandwidth is added by making outgoing calls.
  - **Incoming Only**
    Bandwidth is added by the remote service calling back on the BACP number (assuming that BACP is successfully negotiated).
  - **Outgoing Incoming**
    Uses both methods but bandwidth is first added using outgoing calls.
  - **Incoming Outgoing**
    Uses both methods but bandwidth is first added using incoming BACP calls.

# Service | IP

The fields in this tab are used to configure network addressing for the services you are running. Depending on how your network is configured, the use of Network Address Translation (NAT) may be required.

| Service \| IP | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **IP Address:** *Default = 0.0.0.0 (address assigned by ISP)*
  An address should only be entered here if a specific IP address and mask have been provided by the Service Provider. Note that if the address is in a different domain from the IP Office then NAT is automatically enabled.
- **IP Mask:** *Default = 0.0.0.0 (use NAT)*
  Enter the IP Mask associated with the IP Address if an address is entered.
- **Primary Transfer IP Address:** *Default = 0.0.0.0 (No transfer)*
  This address acts as a primary address for incoming IP traffic. All incoming IP packets without a session are translated to this address. This would normally be set to the local mail or web server address.
    - For IP Office control units supporting a LAN1 and LAN2, the primary transfer address for each LAN can be set through the **System | LAN1** and **System | LAN2** tabs.
- **Request DNS:** *Default = Off*
  When selected, DNS information is obtained from the service provider. To use this, the DNS Server addresses set in the IP Office configuration (**System | DNS**) should be blank. The PC making the DNS request should have the IP Office set as its DNS Server. For DHCP clients the IP Office will provide its own address as the DNS server.
- **Forward Multicast Messages:** *Default = On*
  By default this option is on. Multicasting allows WAN bandwidth to be maximized through the reduction of traffic that needs to be passed between sites.
- **RIP Mode:** *Default = None*
  Routing Information Protocol (RIP) is a method by which network routers can exchange information about device locations and routes. RIP can be used within small networks to allow dynamic route configuration as opposed to static configuration using.
    - **None**
      The LAN does not listen to or send RIP messages.
    - **Listen Only (Passive)**
      Listen to RIP-1 and RIP-2 messages in order to learn RIP routes on the network.
    - **RIP1**
      Listen to RIP-1 and RIP-2 messages and send RIP-1 responses as a sub-network broadcast.
    - **RIP2 Broadcast (RIP1 Compatibility)**
      Listen to RIP-1 and RIP-2 messages and send RIP-2 responses as a sub-network broadcast.
    - **RIP2 Multicast**
      Listen to RIP-1 and RIP-2 messages and send RIP-2 responses to the RIP-2 multicast address.

# Service | Autoconnect

Fields in this tab enable you to set up automatic connections to the specified Service.

| Service \| Autoconnect | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Auto Connect Interval (mins):** *Default = 0 (disabled). Range = 0 to 99999 minutes.*
  This field defines how often this Service will automatically be called ("polled"). For example setting 60 means the system will call this Service every hour in the absence of any normally generated call (this timer is reset for every call; therefore if the service is already connected, then no additional calls are made). This is ideal for SMTP Mail polling from Internet Service Providers.

- **Auto Connect Time Profile:** *Default = <None>*
  Allows the selection of any configured Time Profiles. The selected profile controls the time period during which automatic connections to the service are made. It does NOT mean that connection to that service is barred outside of these hours. For example, if a time profile called "Working Hours" is selected, where the profile is defined to be 9:00AM to 6:00PM Monday to Friday, then automatic connection to the service will not be made unless its within the defined profile. If there is an existing connection to the service at 9:00AM, then the connection will continue. If there is no connection, then an automatic connection will be made at 9:00AM.

# Service | Quota

Quotas are associated with outgoing calls, they place a time limit on calls to a particular IP Service. This avoids excessive call charges when perhaps something changes on your network and call frequency increases unintentionally.

| Service \| Quota | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Quota Time (mins):** *Default = 240 minutes. Range = 0 to 99999 minutes.*
  Defines the number of minutes used in the quota. When the quota time is used up no further data can be passed to this service. This feature is useful to stop things like an internet game keeping a call to your ISP open for a long period.
  - **Warning:** Setting a value here without selecting a Quota period below will stop all further calls after the **Quota Time** has expired.
- **Quota:** *Default = Daily, Range = None, Daily, Weekly or Monthly*
  Sets the period during which the quota is applied. For example, if the **Quota Time** is 60 minutes and the **Quota** is set to ***Daily***, then the maximum total connect time during any day is 60 minutes. Any time beyond this will cause the system to close the service and prevent any further calls to this service. To disable quotas select ***None*** and set a **Quota Time** of zero.
  - Note: The **ClearQuota** feature can be used to create Short Codes to refresh the quota time.

# Service | PPP

Fields in this tab enable you to configure Point to Point Protocol (PPP) in relation to this particular service. PPP is a protocol for communication between two computers using a Serial interface.

| **Service \| PPP** | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Chap Challenge Interval (secs):** *Default = 0 (disabled). Range = 0 to 99999 seconds.*
  The period between CHAP challenges. Blank or 0 disables repeated challenges. Some software such as Windows 95 DUN does not support repeated CHAP challenges.
- **Bi-Directional Chap:** *Default =Off.*
- **Header Compression:** *Default = None selected*
  Enables the negotiation and use of IP Header Compression. Supported modes are ***IPHC*** and ***VJ***. ***IPHC*** should be used on WAN links.
- **PPP Compression Mode:** *Default = MPPC*
  Enables the negotiate and use of compression. Do not use on VoIP WAN links.
  - **Disable**
    Do not use or attempt to use compression.
  - **StacLZS**
    Attempt to use STAC compression (Mode 3, sequence check mode).
  - **MPPC**
    Attempt to use MPPC compression. Useful for NT Servers.
- **PPP Callback Mode:** *Default = Disable*
  - **Disable**
    Callback is not enabled
  - **LCP:** (Link Control Protocol)
    After authentication the incoming call is dropped and an outgoing call to the number configured in the Service is made to re-establish the link.
  - **Callback CP:** (Microsoft's Callback Control Protocol)
    After acceptance from both ends the incoming call is dropped and an outgoing call to the number configured in the Service is made to re-establish the link.
  - **Extended CBCP:** (Extended Callback Control Protocol)
    Similar to Callback CP except the Microsoft application at the remote end prompts for a telephone number. An outgoing call is then made to that number to re-establish the link.

- **PPP Access Mode:** *Default = Digital64*
  Sets the protocol, line speed and connection request type used when making outgoing calls. Incoming calls are automatically handled (see RAS services).
  - **Digital64**
    Protocol set to Sync PPP, rate 64000 bps, call presented to local exchange as a "Data Call".
  - **Digital56**
    As above but rate 56000 bps.
  - **Voice56**
    As above but call is presented to local exchange as a "Voice Call".
  - **V120**
    Protocol set to Async PPP, rate V.120, call presented to local exchange as a "Data Call". This mode runs at up to 64K per channel but has a higher Protocol overhead than pure 64K operation. Used for some bulletin board systems as it allows the destination end to run at a different asynchronous speed to the calling end.
  - **V110**
    Protocol is set to Async PPP, rate V.110. This runs at 9600 bps, call is presented to local exchange as a "Data Call". It is ideal for some bulletin boards.
  - **Modem**
    Allows Asynchronous PPP to run over an auto-adapting Modem to a service provider (requires a Modem2 card in the main unit)
- **Data Pkt. Size:** *Default = 0, Range = 0 to 2048.*
  Sets the size limit for the Maximum Transmissible Unit.
- **BACP:** *Default = Off*
  Enables the negotiation and use of BACP/BCP protocols. These are used to control the addition of B channels to increase bandwidth.
- **Incoming traffic does not keep link up:** *Default = On*
  When enabled, the link is not kept up for incoming traffic only.
- **Multilink/QoS:** *Default = Off*
  Enables the negotiation and use of Multilink protocol (MPPC) on links into this Service. Multilink must be enabled if there is more than one channel that is allowed to be Bundled/Multilinked to this RAS Service.

# Service | Fallback

These options allow you to set up a fallback for the Service. For example, you may wish to connect to your ISP during working hours and at other times take advantage of varying call charges from an alternative carrier. You could therefore set up one Service to connect during peak times and another to act as fallback during the cheaper period.

You need to create an additional Service to be used during the cheaper period and select this service from the **Fallback Service** list box (open the Service form and select the **Fallback** tab).

If the original Service is to be used during specific hours and the Fallback Service to be used outside of these hours, a Time Profile can be created. Select this Time Profile from the Time Profile list box. At the set time the original Service goes into Fallback and the Fallback Service is used.

A Service can also be put into Fallback manually using short codes, for example:

- **Put the service  into fallback:**
    - **Short Code:** *85
    - **Telephone Number:**
    - **Line Group ID:** 0
    - **Feature:** SetHuntGroupNightService

- **Take the service  out of fallback:**
    - **Short Code:** *86
    - **Telephone Number:**
    - **Line Group ID:** 0
    - **Feature:** ClearHuntGroupNightService

| **Service \| Fallback** | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **In Fallback:** *Default = Off*
  This option indicates whether the Service is in Fallback or not. A service can be set into fallback using this setting. Alternatively a service can be set into fallback using a time profile or short codes.

- **Time profile:** *Default = <None> (No automatic fallback)*
  Select the time profile you wish to use for the service. The time profile should be set up for the hours that you wish this service to be operational, out of these hours the Fallback Service is used.

- **Fallback Service:** *<None>*
  Select the service that is used when this service is in fallback.

# Service | Dial In

Only available for WAN and Intranet Services. This tab is used to define a WAN connection.

| Service \| Dial In | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

**To define a WAN connection**

1. Right-click within the **DialIn** window and select **Add**.
2. Enter **WAN** if the service is being routed via a WAN port on a WAN3 expansion module.

# RAS Settings

## RAS | RAS

A Remote Access Server (RAS) is a piece of computer hardware which sits on a corporate LAN and into which employees dial on the public switched telephone network to get access to their email and to software and data on the corporate LAN.

This form is used to create a RAS service that the system offers Dial In users. A RAS service is needed when configuring modem dial in access, digital (ISDN) dial in access and a WAN link. Some systems may only require one RAS service since the incoming call type can be automatically sensed.

| **RAS \| RAS** | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Name**
  A textual name for this service. If Encrypted Password (see below) is used this name must match the Account Name entered in the Service Form.

- **Extension**
  Enter an extension number if this service is to be accessed internally.

- **COM Port**
  For future use.

- **TA Enable:** *Default = Off*
  Select to enable or disable - if enabled RAS will pass the call onto a TA port for external handling.

- **Encrypted Password:** *Default = Off*
  This option is used to define whether Dial In users are asked to use PAP or CHAP during their initial logon to the RAS Service. If the Encrypted Password box is checked then Dial In users are sent a CHAP challenge, if the box is unchecked PAP is used as the Dial In Authorization method.

# RAS | PPP

PPP (Point-to-Point Protocol) is a Protocol for communication between two computers using a Serial interface, typically a personal computer connected by phone line to a server.

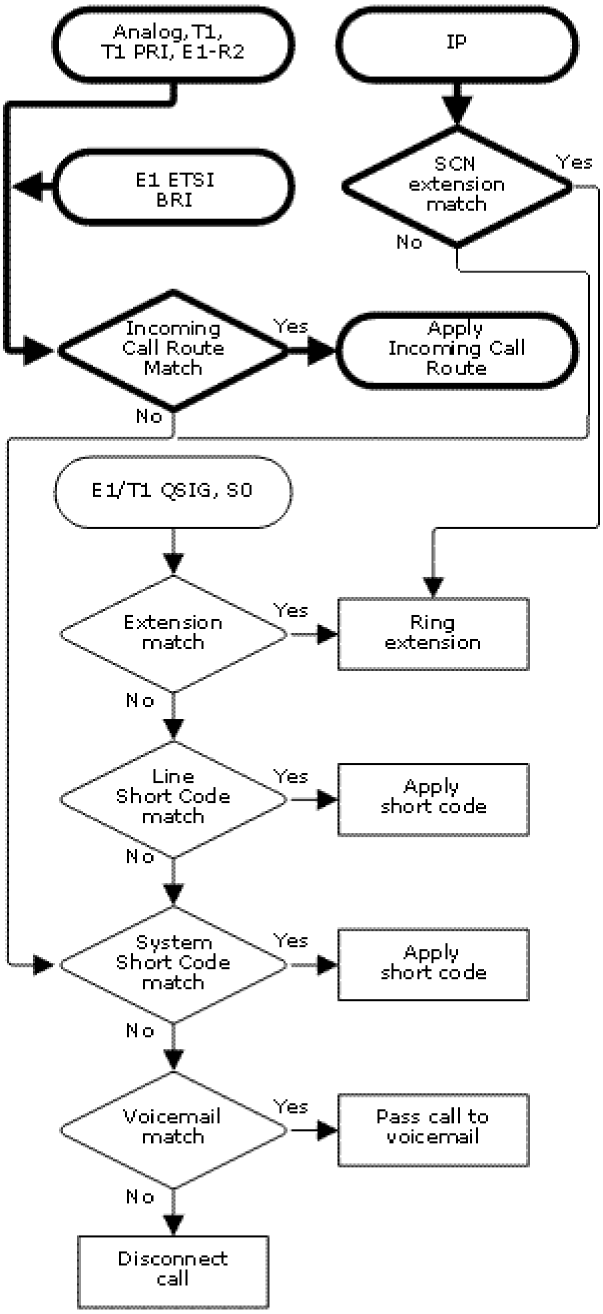
| RAS \| PPP | |
|---|---|
| Control Unit | SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓. |
| Software Level | 2.1 ✓, 3.0DT ✓, 3.0 ✓, 3.1 ✓, 3.2 ✓, 4.0 ✓, 4.1 ✓. |
| Mergeable | ✓. |

- **CHAP Challenge Interval (secs):** *Default = 0 (disabled). Range = 0 to 99999 seconds.*
  The period between successive CHAP challenges. Blank or 0 disables repeated challenges. Some software, for example Windows 95 DUN, does not support repeated CHAP challenges.
- **Header Compression:** *Default = Off*
  Enables the negotiation and use of IP Header Compression as per RFC2507, RFC2508 and RFC2509.
- **PPP Compression Mode:** *Default = MPPC*
  This option is used to negotiate compression (or not) using CCP. If set to MPPC or StacLZS the system will try to negotiate this mode with the remote Control Unit. If set to Disable CCP is not negotiated.
  - **Disable**
    Do not use or attempt to use compression.
  - **StacLZS**
    Attempt to use and negotiate STAC compression (the standard, Mode 3)
  - **MPPC**
    Attempt to use and negotiate MPPC (Microsoft) compression. Useful for dialing into NT Servers.
- **PPP Callback Mode:** *Default = Disable*
  - **Disable:**
    Callback is not enabled
  - **LCP:** (Link Control Protocol)
    After authentication the incoming call is dropped and an outgoing call to the number configured in the Service will be made to reestablish the link.
  - **Callback CP:** (Microsoft's Callback Control Protocol)
    After acceptance from both ends the incoming call is dropped and an outgoing call to the number configured in the Service is made to reestablish the link.
  - **Extended CBCP:** (Extended Callback Control Protocol)
    Similar to Callback CP however the Microsoft application at the remote end will prompt for a telephone number. An outgoing call will then be made to that number to reestablish the link.
- **Data Pkt. Size:** *Default = 0, Range = 0 to 2048.*
  This is the number of data bytes contained in a Data Packet.
- **BACP:** *Default = Off*
  Enable/Disable - Allows negotiation of the BACP/BCP protocols. These are used to control the addition of additional B channels to simultaneously improve data throughput.
- **Multilink:** *Default = Off*
  Enable/Disable – When enabled the system attempts to negotiate the use of the Multilink protocol (MPPC) on the link(s) into this Service. Multilink must be enabled if the more than one channel is allowed to be Bundled/Multilinked to this RAS Service.

# Incoming Call Route Settings

## Incoming Call Route Overview

Incoming call routes are used to determine the destination of voice and data calls received by the IP Office. Determining which incoming call route is used is based on the call matching number of criteria. In order of priority, these are:




The bearer capability indicated, if any, with the call. For example voice, data, video.

- The incoming line group ID of the trunk or trunk channel on which the call was received.
- The incoming number received with the call.
- The incoming sub address received with the call.
- The incoming caller ID of the call.

Each incoming route can include a fallback destination for when the primary destination is busy. It can also include a time profile which control when the primary destination is used. Outside the time profile calls are redirected to a night service destination.

- For IP Office 4.1+, multiple time profiles can be associated with an incoming call route. Each time profile used has its own destination and fallback destination specified.

On IP Office systems where a large number incoming call routes need to be setup for DID numbers, the MSN/DID Configuration tool can be used. Select **Tools | MSN Configuration**.

# Incoming Call Routing Examples

**Example: Incoming Numbers**
For this example, the customer has subscribes to receive two 2-digit DID numbers. They want calls on one routed to a Sales hunt group and calls on the other to a Services hunt group. Other calls should use the normal default route to hunt group Main.

The following incoming call routes were added to the configuration to achieve this:

| Line Group | Incoming Number | Destination |
|---|---|---|
| 0 | 77 | Sales |
| 0 | 88 | Services |
| 0 | *blank* | Main |

Note that the incoming numbers could have been entered as the full dialed number, for example 7325551177 and 7325551188 respectively. The result would still remain the same as incoming number matching is by default performed on the right hand digits.

Right-hand matching gets complicated when the number of incoming digits is greater than the number of digits specified in the Incoming Number field. Consider the example below for when the incoming number digits 77 are received. The entries 677 and 77 have the same number of matching digit places and no non-matching places. However the 77 entry is the shorter match and so is used by the IP Office.

| Line Group | Incoming Number | Destination |
|---|---|---|
| 0 | 677 | Support |
| **0** | **77** | **Sales** |
| 0 | 7 | Services |
| 0 | *blank* | Main |

In the following example the 677 entry is used as the match for 77 as it has more matching digits than the 7 entry and no non-matching digits.

| Line Group | Incoming Number | Destination |
|---|---|---|
| **0** | **677** | **Support** |
| 0 | 7 | Services |
| 0 | *blank* | Main |

If this case the digits 777 are received. The 677 entry had a non-matching digit, so it isn't a match. The entry 7 is used as it has one matching digit and no non-matching digits.

| Line Group | Incoming Number | Destination |
|---|---|---|
| 0 | 677 | Support |
| **0** | **7** | **Services** |
| 0 | *blank* | Main |

# Incoming Call Route | Standard

Incoming call routes are used to match call received with destinations. Routes can be based on the incoming line group, the type of call, incoming digits or the caller's ICLID. If a range of MSN/DID numbers has been issued, this form can be populated using the MSN Configuration tool (see **MSN Configuration**).

- **Default Blank Call Routes**
  By default the configuration contains two incoming calls routes; one set for Any Voice calls (including analog modem) and one for Any Data calls. Whilst the destination of these default routes can be changed it is strongly recommended that they are not deleted.
    - Deleting the default call routes, may cause busy tone to be returned to any incoming external call that does not match another incoming call route.
    - Setting any route to a blank destination field, may cause the incoming number to be matched against system short codes for a match. This may lead to the call being rerouted off-switch.
- Calls received on S0 trunks and PRI E1 trunks set to QSIG operation do not use incoming call routes. Routing for these is based on incoming number received as if dialed on-switch. Line short codes on those trunks can be used to modify the incoming digits.
- If there is no matching incoming call route for a call, matching is attempted against system short codes and finally against voicemail nodes before the call is dropped.
- **SIP Calls**
  For SIP calls, the following fields are used for call matching:
    - **Line Group ID**
      This field is matched against the Incoming Group settings of the SIP URI (Line | SIP URI). This must be an exact match.
    - **Incoming Number**
      This field can be used to match the called details (TO) in the SIP header of incoming calls. It can contain a number, SIP URI or Tel URI. For SIP URI's the domain part of the URI is removed before matching by incoming call routing occurs. For example, for the SIP URI ***mysip@sipitsp.com*** , only the user part of the URI, ie. ***mysip***, is used for matching.
    - **Incoming CLI**
      This field can be used to match the calling details (FROM) in the SDP header of incoming SIP calls. It can contain a number, SIP URI, Tel URI or IP address received with SIP calls. For all types of incoming CLI except IP addresses a partial entry can be used to achieve the match, entries being read from left to right. For IP addresses only full entry matching is supported.

| Incoming Call Route \| Standard | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Bearer Capability:** *Default = Any Voice*
  The type of call selected from the list of standard bearer capabilities: **Any**, **Any Voice**, **Any Data**, **Speech**, **Audio 3K1**, **Data 56K**, **Data 64K**, **Data V110**, **Video**.
- **Line Group ID:** *Default = 0, Range = 0 to 99999.*
  Matches against the Incoming Line Group to which the trunk receiving the call belongs.
- **Incoming Number:** *Default = Blank (Match any unspecified)*
  Matches to the digits presented by the line provider. A blank entry matches all calls that do not match other entries. By default this is a right-to-left match.
  - **– :** Using a **-** in front of the number causes a left-to-right match.
  - **X or x :** Use **X**'s to enter a single digit wild card character. For example - *91XXXXXXXXXXX* will only match when 13 digits, starting with 91, are received. **N**, **n** and **?** cannot be used.
  - **i :** The **i** character does not affect the incoming number matching. It is used for Outgoing Caller ID Matching, see notes below.
- **Incoming Sub Address:** *Default = Blank (Match all)*
  Matches any sub address component sent with the incoming call. If this field is left blank, it matches all calls.
- **Incoming CLI:** *Default = Blank (Match all)*
  Enter a number to match the caller's ICLID provided with the call. This field is matched left-to-right. Number options are:
  - Full telephone number.
  - Partial telephone number, for example just the area code.
  - **! :** Matches calls where the ICLID was withheld.
  - **? :** for number unavailable.
  - Blank for all.
- **Locale:** *Default = Blank (Use system setting)*
  This option specifies the language prompts, if available, that voicemail should use for the call if it is directed to voicemail.
- **Priority:** *Default = 1, Range = 1 (lowest) to 3 (highest).*
  This setting allows calls to be assigned a priority.  Internal calls are always assigned priority 1. Note that using priority is not recommended for destinations where queue ETA and position messages are being provided by Voicemail Pro.
  - In situations where calls are queued, high priority calls are placed before calls of a lower priority. If this causes the queue length to exceed the hunt group's Queue Length limit, the limit is temporarily raised by 1, ie. calls already accepted as queued are not rerouted by the arrival of a higher priority call.

- **Destination:** *Default = Blank, Software Level = Up to 4.0 only.*
  For IP Office 4.1+ this option has moved to the **Incoming Call Route | Destinations** tab. Select the destination for the call from the drop-down list box which contains all available extensions, users, groups, RAS services and voicemail. System short codes and dialing numbers can be entered manually. Once the incoming call is matched the call is passed to that destination.
  - **Drop-Down List Options**
    The following options appear in the drop-down in the following order:
    - **Voicemail** allows remote mailbox access with Embedded Voicemail, Voicemail Lite and Voicemail Pro. Callers are asked to enter the extension ID of the mailbox required and then the mailbox access code.
    - **User Names**
    - **Hunt Groups Names**
    - **AA:*Name*** directs calls to an Embedded Voicemail auto-attendant services.
  - **Manually Entered Options**
    The following options can be entered manually into the field.
    - **VM:*Name*** Directs calls to the matching start point in Voicemail Pro.
    - A **.** matches the **Incoming Number** field. This can be used even when **X** wildcards are being used in the **Incoming Number** field.
    - A **#** matches all **X** wildcards in the Incoming Number field. For example, if the Incoming Number was *-91XXXXXXXXXXX*, the Destination would be *XXXXXXXXXXX*.
    - Text and number strings entered here are passed through to system short codes, for example to direct calls into a conference. Note that not all short code features are supported.
- **Tag:** *Default = Blank (No tag), Software level = 4.1+.*
  Allows a text tag to be associated with calls routed by this incoming call route. This tag is displayed with the call within applications such as Phone Manager and SoftConsole (note that those application can also be used to change the tag text when transferring calls).
- **Fallback Extension:** *Default = Blank (No fallback), Software level = Up to 4.0 only.*
  For IP Office 4.1+ this option has moved to the **Incoming Call Route | Destinations** tab. Defines an alternate destination which should be used when the current destination, set in the **Destination** or **Night Service Destination** field, cannot be obtained. For example if the primary destination is a hunt group returning busy and without queuing or voicemail.
- **Night Service Profile:** *Default = <None> (No night service), Software level = Up to 4.0 only.*
  A time profile during which the **Night Service Destination** should be used rather than the **Destination**.
- **Night Service Destination:** *Default = Blank, Software level = Up to 4.0 only.*
  Set the destination to be used during periods defined by the **Night Service Profile**. The same range of values can be used as for the **Destination** field.

# Outgoing Caller ID Matching

In cases where a particular **Incoming Number** is routed to a specific individual user, the IP Office will attempt to use that Incoming Number as the user's caller ID when they make outgoing calls. This requires that the Incoming Number is a full number suitable for user as outgoing caller ID and acceptable to the line provider.

When this is the case, the character **i** can also be added to the **Incoming Number** field. This character does not affect the incoming call routing. However when the same **Incoming Number** is used for an outgoing caller ID, the calling party number plan is set to ISDN and the type is set to National. This option may be required by some network providers.

# Incoming Call Route | Voice Recording

This tab is used to activate the automatic recording of incoming calls that match the incoming call route. This requires Voicemail Pro to be installed and running. Call recording also requires available conference resources similar to a 3-way conference.

- Calls to and from IP devices including those using Direct media can be recorded.
- Incoming call route recording continues for the duration of the call on the system.
- Calls parked or held by the user pause recording until the call is unparked or taken off hold.

| Incoming Call Route \| Voice Recording | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✗, **3.0DT** ✗, **3.0** ✗, **3.1** ✗, **3.2** ✗, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Record Inbound:** *Default = None*
  The same as Record Outbound but applied to inbound external calls to the user.
  - **On:** Record the call if possible.
  - **Mandatory:** If not possible to record, return busy tone.
  - **Percentages of calls:** Various percentages of calls made by the user will be recorded.
- **Record Time Profile:** *Default = <None> (Any time)*
  Used to select a time profile during which automatic call recording setting above apply.
- **Recording Mailbox:** *Default = <None>*
  Sets the mailbox into which automatically triggered recordings are placed.
- **Voice Recording Library:** *Default = Not Enabled*
  When checked, this setting overrides the **Recording Mailbox** setting and marks the recording for collection by the ContactStore for IP Office application. Note that Voicemail Pro still performs the actual recording.

# Incoming Call Route | Destinations

IP Office 4.1+ allows multiple time profiles to be associated with an incoming call route. For each time profile, a separate **Destination** and **Fallback Extension** can be specified.

When multiple entries are added, they are resolved from the bottom up. The entry used will be the first one, working from the bottom of the list upwards, that is currently 'true', ie. the current day and time or date and time match those specified by the Time Profile. If no match occurs the **Default Value** options are used.

Once a match is found, the IP Office does not use any other destination set even if the intended **Destination** and **Fallback Extension** destinations are busy or not available.

| Incoming Call Route \| Standard | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✗, **IP406 V1** ✗, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✗, **3.0DT** ✗, **3.0** ✗, **3.1** ✗, **3.2** ✗, **4.0** ✗, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Time Profile**
  This column is used to specify the time profiles used by the incoming call routes. The entry displays a drop-down list of existing time profiles from which a selection can be made. To remove an existing entry, select it by clicking on the button on the left of the row, then right-click on the row and select **Delete**.
  - **Default Value**
    This entry is fixed and is used if no match to a time profile below occurs.
- **Destination:** *Default = Blank*
  Select the destination for the call from the drop-down list box which contains all available extensions, users, groups, RAS services and voicemail. System short codes and dialing numbers can be entered manually. Once the incoming call is matched the call is passed to that destination.
  - **Drop-Down List Options**
    The following options appear in the drop-down in the following order:
    - **Voicemail** allows remote mailbox access with Embedded Voicemail, Voicemail Lite and Voicemail Pro. Callers are asked to enter the extension ID of the mailbox required and then the mailbox access code.
    - **User Names**
    - **Hunt Groups Names**
    - **AA:*Name*** directs calls to an Embedded Voicemail auto-attendant services.
  - **Manually Entered Options**
    The following options can be entered manually into the field.
    - **VM:*Name*** Directs calls to the matching start point in Voicemail Pro.
    - A **.** matches the **Incoming Number** field. This can be used even when **X** wildcards are being used in the **Incoming Number** field.
    - A **#** matches all **X** wildcards in the Incoming Number field. For example, if the Incoming Number was *-91XXXXXXXXXX*, the Destination would be *XXXXXXXXXXX*.
    - Text and number strings entered here are passed through to system short codes, for example to direct calls into a conference. Note that not all short code features are supported.
- **Fallback Extension:** *Default = Blank (No fallback)*
  Defines an alternate destination which should be used when the current destination, set in the

**Destination** field cannot be obtained. For example if the primary destination is a hunt group returning busy and without queuing or voicemail.

# WAN Port Settings

## WAN Port Overview

These entries are used to configure the operation of IP Office WAN ports. These are the 37-way D-type WAN ports found on the rear of all IP Office control units except the Small Office Edition.

Additional WAN ports can be added by the installation of up to two WAN3 expansion modules, each module providing 3 additional WAN ports. On the Small Office Edition control unit, a single WAN port can be added by the installation of a WAN trunk card at the rear of the unit. For full details of installing additional WAN ports, refer to the IP Office Installation Manual.

Physical WAN are not supported on the IP500.

**Creating a Virtual WAN Port**
WAN services can be run over a T1 PRI trunk connection. This requires creation of a virtual WAN port. For full details refer to Using a Dedicated T1/PRI ISP Link in **Appendix A**.

1. Select **WAN Port**.
2. Click and select PPP.
3. In the **Name** field, enter either ***LINEx.y*** where:
    - ***LINE*** must be in uppercase.
    - ***x*** is the line number. For a PRI/T1 module in Slot A, this will be ***1***. For a PRI/T1 module 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.
4. 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.
    - Note: The maximum number of channels that can be used will be limited by the number of data channels supported by the IP Office Control Unit and not already in use.
5. In the **RAS Name** field, select the RAS name created when the new Service of that name was created.
6. Click **OK**.

# WAN Port | WAN Port

Use this form to configure the leased line connected to the WAN port on the Control Unit. Normally this connection is automatically detected by the IP Office control unit. If a WAN Port is not displayed, connect the WAN cable, reboot the Control Unit and receive the configuration. The WAN Port configuration form should now be added.

| WAN Port \| WAN Port | |
|---|---|
| **Control Unit** | **SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓.** |
| **Software Level** | **2.1 ✓, 3.0DT ✓, 3.0 ✓, 3.1 ✓, 3.2 ✓, 4.0 ✓, 4.1 ✓.** |
| **Mergeable** | ✗. |

- **Name**
  The physical ID of the Extension port,. This parameter is not configurable; it is allocated by the system.
- **Speed**
  The operational speed of this port. For example for a 128K connection, enter 128000. This should be set to the actual speed of the leased line as this value is used in the calculation of bandwidth utilization. If set incorrectly, additional calls may be made to increase Bandwidth erroneously.
- **Mode:** *Default = SyncPPP*
  Select the protocol required:
  - **SyncPPP**
    For a data link.
  - **SyncFrameRelay**
    For a link supporting Frame Relay.
- **RAS Name**
  If the **Mode** is ***SyncPPP***, selects the RAS service to associate with the port. If the **Mode** is ***SyncFrameRelay***, the RAS Name is set through the **DCLIs** tab.

# WAN Port | Frame Relay

This tab is only available for Frame Relay entries. These show **SyncFrameRelay** as the **Mode** on the WAN Port tab.

| WAN Port \| Frame Relay | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

- **Frame Management Type**
  This must match the management type expected by the network provider. Selecting **AutoLearn** allows the IP Office to automatically determine the management type based on the first few management frames received. If a fixed option is required the following are supported: **Q933 AnnexA 0393**, **Ansi AnnexD** , **FRFLMI** and **None**.
- **Frame Learn Mode**
  This parameter allows the DLCIs that exist on the given WAN port to be provisioned in a number of different ways.
  - **None**
    No automatic learning of DLCIs. DLCIs must be entered and configured manually.
  - **Mgmt**
    Use LMI to learn what DLCIs are available on this WAN.
  - **Network**
    Listen for DLCIs arriving at the network. This presumes that a network provider will only send DLCIs that are configured for this particular WAN port.
  - **NetworkMgmt**
    Do both management and network listening to perform DLCI learning and creation.
- **Max Frame Length**
  Maximum frame size that is allowed to traverse the frame relay network.
- **Fragmentation Method**
  Options are **RFC1490** or **RFC1490+FRF12**.

# WAN Port | DLCIs

This tab is only available for Frame Relay entries. These show **SyncFrameRelay** as the **Mode** on the WAN Port tab.

The tab lists the DLCIs created for the connection. These can be edited using the **Add**, **Edit** and **Remove** buttons.

| WAN Port \| DLCIs | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

- **Frame Link Type:** *Default = PPP*
  Data transfer encapsulation method. Set to the same value at both ends of the PVC (Permanent Virtual Channel).
  - **None**
  - **PPP**
    Using PPP offers features such as out of sequence traffic reception, compression and link level connection management.
  - **RFC 1490**
    RFC 1490 encapsulation offers performance and ease of configuration and more inter-working with third party CPE.
  - **RFC1490 + FRF12**
    Alternate encapsulation to PPP for VoIP over Frame Relay. When selected all parameters on the **Service | PPP** tab being used are overridden.
- **DLCI:** *Default = 100*
  This is the Data Link Connection Identifier, a unique number assigned to a PVC end point that has local significance only. Identifies a particular PVC endpoint within a user's physical access channel in a frame relay.
- **RAS Name**
  Select the RAS Service you wish to use.
- **Tc:** *Default = 10*
  This is the Time Constant in milliseconds. This is used for measurement of data traffic rates. The Tc used by the IP Office can be shorter than that used by the network provider.
- **CIR:** (Committed Information Rate) *Default = 64000 bps*
  This is the Committed Information Rate setting. It is the maximum data rate that the WAN network provider has agreed to transfer. The committed burst size (***Bc***) can be calculated from the set Tc and CIR as *Bc = CIR x Tc*. For links carrying VoIP traffic, the ***Bc*** should be sufficient to carry a full VoIP packet including all its required headers. See the example below.
- **EIR:** (Excess Information Rate) *Default = 0 bps*
  This is the maximum amount of data in excess of the CIR that a frame relay network may attempt to transfer during the given time interval. This traffic is normally marked as ***De*** (discard eligible). Delivery of De packets depends on the network provider and is not guaranteed and therefore they are not suitable for UDP and VoIP traffic. The excess burst size (***Be***) can be calculated as *Be = EIR x Tc*.

**Example: Adjusting the Tc Setting**
G.729 VoIP creates a 20 byte packet every 20ms. Adding typical WAN PPP headers results in a 33 byte packet every 20ms.

For a Committed Information Rate (CIR) of 14Kbps, with the Time Constant (Tc) set to 10ms; we can calculate the Committed Burst size:

Bc = CIR x Tc = 14,000 x 0.01 = 140 bits = 17.5 bytes.

Using 10ms as the **Tc**, a full G.729 VoIP packet (33 bytes) cannot be sent without exceeding the Bc. The most likely result is lost packets and jitter.

If the Tc is increased to 20ms:

Bc = CIR x Tc = 14,000 x 0.02 = 280 bits = 35 bytes.

The Bc is now sufficient to carry a full G.729 VoIP packet.

**Notes**

1. Backup over Frame Relay is not supported when the Frame Link Type is set to RFC1490.
2. When multiple DLCIs are configured, the WAN link LED is switched off if any of those DLCIs is made inactive, regardless of the state of the other DLCIs. Note also that the WAN link LED is switched on following a reboot even if one of the DLCIs is inactive. Therefore when multiple DLCIs are used, the WAN link LED cannot be used to determine the current state of all DLCIs.
3. When the Frame Link Type is set to RFC1490, the WAN link LED is switched on when the WAN cable is attached regardless other whether being connected to a frame relay network.

# WAN Port | Advanced

The settings on this tab are used for Frame Relay connections.

| WAN Port \| Advanced | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

- **Address Length:**
  The address length used by the frame relay network. The network provider will indicate if lengths other than two bytes are to be used.
- **N391:** *Full Status Polling Counter*
  Polling cycles count used by the CPE and the network provider equipment when bidirectional procedures are in operation. This is a count of the number of link integrity verification polls (T391) that are performed (that is Status Inquiry messages) prior to a Full Status Inquiry message being issued.
- **N392:** *Error Threshold Counter*
  Error counter used by both the CPE and network provider equipment. This value is incremented for every LMI error that occurs on the given WAN interface. The DLCIs attached to the given WAN interface are disabled if the number of LMI errors exceeds this value when N393 events have occurred. If the given WAN interface is in an error condition then that error condition is cleared when N392 consecutive clear events occur.
- **N393:** *Monitored Events Counter*
  Events counter measure used by both the CPE and network provider equipment. This counter is used to count the total number of management events that have occurred in order to measure error thresholds and clearing thresholds.
- **T391:** *Link Integrity Verification Polling Timer*
  The link integrity verification polling timer normally applies to the user equipment and to the network equipment when bidirectional procedures are in operation. It is the time between transmissions of Status Inquiry messages.
- **T392:** *Polling Verification Timer*
  The polling verification timer only applies to the user equipment when bidirectional procedures are in operation. It is the timeout value within which to receive a Status Inquiry message from the network in response to transmitting a Status message. If the timeout lapses an error is recorded (N392 incremented).

# Directory Settings

## Directory | Directory Entry

Directory entries are used to store external telephone numbers and to associate names with those numbers. They have two main functions:

- **Making External Calls**
  Directory entries can displayed and then used to make calls from IP Office applications such as Phone Manager, SoftConsole and Conference Center. They can also be used to make calls from Avaya digital phones that support the **Dir** function on a programmable key.
- **Name Matching**
  Directory entries can be used to match the ICLID received with an incoming call to a name. That name is then display by IP Office applications and phones receiving the call. Note that the Phone Manager and SoftConsole applications have their own directories which are also used for name matching and, for that user, can override the system directory name match.
  - Name matching is not performed on trunks where a name is supplied with the incoming call, for example QSIG trunks.

A maximum of 1000 entries are supported in the IP Office system directory. The IP Office also supports LDAP (Lightweight Directory Access Protocol). Directory entries obtained by LDAP are only shown in the directory of the Phone Manager and SoftConsole applications. They are not shown or used in the IP Office configuration.

| Directory \| Directory Entry | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Name**
  Enter the text, without spaces, to be used to identify the number.
- **Number**
  Enter the number, without spaces, to be matched with the above name. Any brackets or - characters used in the number string are ignored. The directory number match is done on reading from the left-hand side of the number string. Note that if the system has been configured to use an external dialing prefix, that prefix should be added to directory numbers.
  - ? Wild-card characters can be used at the right-hand end of the number string. For example:
    - **Name:** ***Holmdel***
    - **Number:** ***732555????***
      The number must be padded with question marks to the correct length; in this example local numbers are 10 digits long so 4 question marks are required. This displays ***Holmdel:1234*** for a call from ***7325551234***.

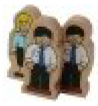
# Time Profile Settings

## Time Profile Overview

Time Profiles are used by different IP Office services to change their operation when required. In most areas where time profiles can be used, not setting a time profile is taken as meaning 24-hour operation.

- Pre-IP Office 4.1 time profiles consist of recurring weekly patterns of days and times when the time profile is in effect.
- IP Office 4.1 time profiles can include time periods on specified calendar days when the time profile is in effect. Calendar entries can be entered for the current and following calendar year.

Time profiles are used by the following entry types:



**Hunt Group** can use time profiles in the following ways:

- A time profile can be used to determine when a hunt group is put into night service mode. Calls then go to an alternate Night Service Fallback group if set, otherwise to voicemail if available or busy tone if not. See **Hunt Group | Fallback**.
- For automatic voice recording, a time profile can be used to set when voice recording is used. See **Hunt Group | Voice Recording**.

**Service** can use time profiles in the following ways:

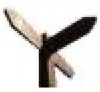
- A time profile can be used to set when a data service is available. Outside its time profile, the service is either not available or uses an alternate fallback service if set.
- For services using auto connect, a time profile can be used to set when that function is used. See **Service | Autoconnect**.

**User** can use time profiles in the following ways:

- Users being used for Dial In data services such as RAS can have an associated time profile that defines when they can be used for that service. See **User | Dial In**.
- Users can be associated with a working hours and an out of hours user rights. A time profile can then be used to determine which user rights is used at any moment. See **User | User**.
- For automatic voice recording, a time profile can be used to set when that voice recording is used. See **User | Voice Recording**.
- For mobile twinning, a time profile can be used to define when twinning should be used. See **User | Twinning**.

**Incoming Call Routes** can use an alternate night service destination. A time profile is then used to set when that destination is used. See **Incoming Call Route | Incoming Call Route**.

- For IP Office 4.0 and higher incoming call routes can also use time profiles to specify when calls should be recorded.
- For IP Office 4.1 and higher, multiple time profiles can be associate with an incoming call route, each profile specifying a destination and fall back destination.



**Least Cost Route** use time profiles to determine when the routes should be used. See **Least Cost Route | LCR**.

**ARS** forms use time profile to determine when the ARS form should be used or calls rerouted to an out of hours route.

**Account Codes** can use automatic voice recording triggered by calls with particular account codes. A time profile can be used to set when this function is used. See **Account Code | Voice Recording**.

**Auto Attendant** supported by embedded voicemail on IP406 V2 and Small Office systems, use time profiles to control the different greetings played to callers. See **Auto Attendant | Actions**.

For a time profile with multiple entries, for example a week pattern and some calendar entries, the profile is valid when any entry is valid.

| **Time Profile \| Time Profile** | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | **Pre-3.2** ✗, **3.2+** ✓. |

- **Name:** *Range = Up to 31 characters*
  This name is used to select the time profile from within other tabs.
- **Time Entry List**
  This list shows the current periods during which the time profile is active. Clicking on an existing entry will display the existing settings and allows them to be edited if required. To remove an entry, selecting it and then click on **Remove** or right-click and select **Delete**.
- **Recurrence Pattern (Weekly Time Pattern)**
  When a new time entry is required, click **Add Recurring** and then enter the settings for the entry using the fields displayed. Alternately right-click and select **Add Recurring Time Entry**. This type of entry specific a time period and the days on which it occurs, for example 9:00 - 12:00, Monday to Friday. A time entry cannot span over two days. For example you cannot have a time profile starting at 18:00 and ending 8:00. If this time period is required two Time Entries should be created - one starting at 18:00 and ending 11:59, the other starting at 00:00 and ending 8:00.
  - **Start Time**
    The time at which the time period starts.
  - **End Time**
    The time at which the time period ends.
  - **Days of Week**
    The days of the week to which the time period applies.
- **Recurrence Pattern (Calendar Date)** *Software Level = 4.1+.*
  When a new calendar date entry is required, click **Add Date** and then enter the settings required. Alternately right-click and select **Add Calendar Time Entry**. Calendar entries can be set for up to the end of the next calendar year.
  - **Start Time**
    The time at which the time period starts.
  - **End Time**
    The time at which the time period ends.
  - **Year**
    Select either the current year or the next calendar year.
  - **Date**
    To select or de-select a particular day, double-click on the date. Selected days are shown with a dark gray background. Click and drag the cursor to select or de-select a range of days.

# Firewall Profile Settings

## Firewall | Standard

The IP Office can act as a firewall, allowing only specific types of data traffic to start a session across the firewall and controlling in which direction such sessions can be started.

IP Office firewall profiles can be applied in the following areas of IP Office operation.

- **System**
  A firewall profile can be selected to be applied to traffic between LAN1 and LAN2.
- **User**
  Users can be used as the destination of incoming RAS calls. For those users a firewall profile can be selected on the user's **Dial In** tab.
- **Service**
  Services are used as the destination for IP routes connection to off-switch data services such as the internet. A Firewall Profile can be selected for use with a service.
- **Logical LAN**
  Where a logical LAN is created for use as an IP Route destination, a Firewall Profile can be selected for use with the logical LAN.

If Network Address Translation (NAT) is used with the firewall (which it typically is), then you must also configure a Primary Incoming Translation Address (see IP tab of the Service configuration form) if you wish sessions to be started into your site (typically for SMTP) from the Internet.

By default, any protocol not listed in the standard firewall list is dropped unless a custom firewall entry is configured for that protocol.

| **Firewall \| Standard** | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Name:** *Range = Up to 31 characters*
  Enter the name to identify this profile.

- **Protocol Control**
  For each of the listed protocols, the options ***Drop***, ***In*** *(Incoming traffic can start a session)*, ***Out*** *(Outgoing traffic can start a session)* and ***Both Directions*** can be selected. Once a session is started, return traffic for that session is also able to cross the firewall.

| Protocol | Default | Description |
|---|---|---|
| **TELNET** | Out | Remote terminal login. |
| **FTP** | Out | File Transfer Protocol. |
| **SMTP** | Out | Simple Mail Transfer Protocol. |
| **TIME** | Out | Time update protocol. |
| **DNS** | Out | Domain Name System. |
| **GOPHER** | Drop | Internet menu system. |
| **FINGER** | Drop | Remote user information protocol. |
| **RSVP** | Drop | Resource Reservation Protocol. |
| **HTTP** | Out | Hypertext Transfer Protocol. |
| **POP3** | Out | Post Office Protocol. |
| **NNTP** | Out | Network News Transfer Protocol. |
| **SNMP** | Drop | Simple Network Management Protocol. |
| **IRC** | Out | Internet Relay Chat. |
| **PPTP** | Drop | Point to Point Tunneling Protocol. |
| **IGMP** | Drop | Internet Group Membership Protocol. |
| **H323** | Drop | This option is not supported and so is grayed out. |

- **IP Office Service Control:** *, Software level = 4.0+.*
  For each of the listed services, the options **Drop**, **In**, **Out** and **Both Directions** can be selected. Once a session is started, return traffic for that session is also able to cross the firewall.

| Protocol | Default | Description |
|---|---|---|
| **SSI** | In | System Status Application access. |
| **SEC** | Drop | TCP security settings access. |
| **CFG** | Drop | TCP configuration settings access. |

# Firewall | Custom

The tab lists custom firewall settings added to the firewall profile. The Add, Edit and Remove controls can be used to amend the settings in the list.

| Firewall \| Custom | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Notes**
  For information only. Enter text to remind you of the purpose of the custom firewall entry.
- **Remote IP Address**
  The IP address of the system at the far end of the link. Blank allows all IP addresses.
- **Remote IP Mask**
  The mask to use when checking the Remote IP Address. When left blank no mask is set, equivalent to 255.255.255.255 - allow all.
- **Local IP Address**
  The address of devices local to this network (pre-translated). Blank allows all IP addresses.
- **Local IP Mask**
  The mask to use when checking the Local IP Address. When left blank no mask is set, equivalent to 255.255.255.255 - allow all.
- **IP Protocol**
  The value entered here corresponds to the IP Protocol which is to be processed by this Firewall profile: 1 for ICMP, 6 for TCP, 17 for UDP or 47 for GRE. This information can be obtained from the "pcol" parameter in a Monitor trace.
- **Match Offset**
  The offset into the packet (0 = first byte of IP packet) where checking commences for either a specific port number, a range of port numbers, or data.
- **Match Length**
  The number of bytes to check in the packet, from the Match Offset point, that are checked against the Match Data and Match Mask settings.
- **Match Data**
  The values the data must equal once masked with the Match Mask. This information can be obtained from "TCP Dst" parameter in a Monitor trace (the firewall uses hex so a port number of 80 is 50 in hex)
- **Match Mask**
  This is the byte pattern, which is logically ANDed with the data in the packet from the offset point. The result of this process is then compared against the contents of the "Match Data" field.
- **Direction**
  The direction that data may take if matching this filter.

| | |
|---|---|
| **Drop** | All matching traffic is dropped. |
| **In** | Incoming traffic can start a session. |
| **Out** | Outgoing traffic can start a session. |
| **Both Directions** | Both incoming and outgoing traffic can start sessions. |

# Example Custom Firewall Entries

**Example: Dropping NetBIOS searches on an ISPs DNS**

We suggest that the following filter is always added to the firewall facing the Internet to avoid costly but otherwise typically pointless requests from Windows machines making DNS searches on the DNS server at your ISP.

- **Direction:** Drop
- **IP Protocol:** 6 (TCP)
- **Match Offset:** 20
- **Match Length:** 4
- **Match Data:** 00890035
- **Match Mask:** FFFFFFFF

**Example: Browsing Non-Standard Port Numbers**

The radio button for HTTP permits ports 80 and 443 through the firewall. Some hosts use non-standard ports for HTTP traffic, for example 8080, 8000, 8001, 8002, etc. You can add individual filters for these ports as you find them.

You wish to access a web page but you cannot because it uses TCP port 8000 instead of the more usual port 80, use the entry below.

- **Direction:** Out
- **IP Protocol:** 6 (TCP)
- **Match Offset:** 22
- **Match Length:** 2
- **Match Data:** 1F40
- **Match Mask:** FFFF

A more general additional entry given below allows all TCP ports out.

- **Direction:** Out
- **IP Protocol:** 6 (TCP)
- **Match Offset:** 0
- **Match Length:** 0
- **Match Data:** 00000000000000000000000000000000
- **Match Mask:** 00000000000000000000000000000000

**Example: Routing All Internet Traffic through a WinProxy**
If you wish to put WinProxy in front of all Internet traffic via the Control Unit. The following firewall allows only the WinProxy server to contact the Internet : -

1. Create a new Firewall profile and select **Drop** for all protocols
2. Under **Custom** create a new Firewall Entry
3. In Notes enter the name of the server allowed. Then use the default settings except in Local IP Address enter the IP address of the WinProxy Server, in Local IP Mask enter 255.255.255.255 and in Direction select Both Directions.
4. **Stopping PINGs**
   You wish to stop pings - this is ICMP Filtering. Using the data below can create a firewall filter that performs the following; Trap Pings; Trap Ping Replies; Trap Both.
   - **Trap Pings:** Protocol = 1, offset = 20, data = 08, mask = FF
   - **Trap Ping Replies:** Protocol = 1, offset = 20, data = 00, mask = FF
   - **Trap Both:** Protocol = 1, offset = 20, data = 00, mask = F7, Traps Both.

# IP Route Settings

## IP Route Overview

The IP Office acts as the default gateway for its DHCP clients. It can also be specified as the default gateway for devices with static IP addresses on the same subnet as the IP Office. When devices on LAN1 and LAN2 want to send data to IP addresses on a different subnet, they will send that data to their default gateway for onward routing.

The IP Route table is used by the IP Office to determine where data traffic should be forwarded. This is done by matching details of the destination IP address to IP Route entries and then using the Destination specified by the matching IP route. These are referred to as 'static routes'.

- **Automatic Routing (RIP)**
  The IP Office can support RIP (Routing Information Protocol) on LAN1 and or LAN2. This is a method through which the IP Office can automatically learn routes for data traffic from other routers that also support matching RIP options, see **RIP**. These are referred to as 'dynamic routes'.
- **Dynamic versus Static Routes**
  By default, static routes entered into the IP Office override any dynamic routes it learns by the use of RIP. This behavior is controlled by the **Favor RIP Routes over static routes** option on the **System | System** tab.
- **Static IP Route Destinations**
  The IP Office allows the following to be used as the destinations for IP routes:
  - **LAN1**
    Direct the traffic to the IP Office's LAN1.
  - **LAN2**
    On IP Office Small Office Edition, IP412 and IP500 systems, traffic can be directed to LAN2. For IP Office 4.1+, LAN port 8 on IP406 V2 control units can be enabled as LAN2.
  - **Service**
    Traffic can be directed to a service. The service defines the details necessary to connect to a remote data service.
  - **Logical LAN**
    Traffic can be directed to a logical LAN already added to the configuration.
  - **Tunnel**
    Traffic can be directed to an IPSec or L2TP tunnel.
- **Default Route**
  IP Office provides two methods of defining a default route for IP traffic that does not match any other specified routes. Use either of the following methods:
  - **Default Service**
    Within the settings for services, one service can be set as the **Default Route** (**Service | Service**).
  - **Default IP Route**
    Create an IP Route entry with a blank IP Address and blank IP Mask set to the required destination for default traffic.

# Viewing the Routing Table

An IP Office's routing table can be viewed using the IP Office Monitor application. This application can be installed from the IP Office Admin CD. Full details of using Monitor are not covered here.

The routing tab includes both static and dynamic routes.

1. Start **Monitor** and select the IP Office system whose routing table you want to view.
2. Select **Filters | Trace Options**.
3. Select the **Routing** tab.
4. Tick **Routing Table**.
5. If required you can also select to view **Routing Table Changes** plus **RIP In** and **RIP Out** messages.
6. The routing table is sent to the monitor trace once every minute.

```
Destination     Netmask         Gateway      Interface    Metric Type
0.0.0.0         0.0.0.0         0.0.0.0      LAN1          0      S
255.255.255.255 255.255.255.255 0.0.0.0      LAN1          0      I
192.168.44.0    255.255.255.0   0.0.0.0      LAN1          0      I
192.168.99.0    255.255.255.0   0.0.0.0      RemoteManager 0      S
192.168.42.0    255.255.255.0   192.168.44.1 LAN1          0      S
```

The **Type** indicates:

- **I** = Internal routes.
- **S** = Static route set in the IP Route table.
- **R** = RIP route resolved from RIP messages.
- **T** = Temporary route to a specific IP address accessed via a service.

# IP Route | IP Route

This tab is used to setup static IP routes from the IP Office. These are in addition to RIP if RIP is enabled on LAN1 and or LAN2.

| IP Route \| IP Route | |
|---|---|
| **Control Unit** | **SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓.** |
| **Software Level** | **2.1 ✓, 3.0DT ✓, 3.0 ✓, 3.1 ✓, 3.2 ✓, 4.0 ✓, 4.1 ✓.** |
| **Mergeable** | ✓. |

- **IP Address**
  The IP address to match for ongoing routing. Any packets meeting the IP Address and IP Mask settings are routed to the entry configured in the Destination field. When left blank then an IP Address of 255.255.255.255 (all) is used.
- **IP Mask**
  The Subnet Mask used to mask the IP Address for ongoing route matching. If blank the mask used is 255.255.255.255 (all).
  - A 0.0.0.0 entry in the IP Address and IP Mask fields routes all packets for which there is no other specific IP Route available. The **Default Route** option with Services can be used to do this if a blank IP route is not added.
- **Gateway IP Address:** *Default = Blank*
  The address of the gateway where packets for the above address are to be sent. If this field is set to 0.0.0.0 or is left blank then all packets are just sent down to the Destination specified, not to a specific IP Address. This is normally only used to forward packets onto another Router on the local LAN.
- **Destination**
  Allows selection of ***LAN1***, ***LAN2*** (if supported) and any configured Service, Logical LAN or Tunnel.
- **Metric:** *Default = 1*
  The number of "hops" this route counts as.
- **Proxy ARP:** *Default = Off*
  This allows the IP Office to respond on behalf of this IP address when receiving an ARP request.

# RIP Dynamic Routing

Routing Information Protocol (RIP) is a protocol which allows routers within a network to exchange routes of which they are aware approximately every 30 seconds. Through this process, each router becomes adds routes in the network to its routing table.

Each router to router link is called a 'hop' and routes of up to 15 hops are created in the routing tables. When more than one route to a destination exists, the route with the lowest metric (number of hops) is added to the routing table.

When an existing route becomes unavailable, after 5 minutes it is marked as requiring 'infinite' (16 hops). It is then advertised as such to other routers for the next few updates before being removed from the routing table. The IP Office also uses 'split horizon' and 'poison reverse'.

RIP is a simple method for automatic route sharing and updating within small homogeneous networks. It allows alternate routes to be advertised when an existing route fails. Within a large network the exchange of routing information every 30 seconds can create excessive traffic. In addition the routing table held by each IP Office is limited to 100 routes (including static and internal routes).

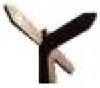
RIP is supported with IP Office system's from Level 2.0 upwards. The normal default is for RIP to be disabled. It can be enabled on LAN1, LAN2 and individual services.

- **Listen Only (Passive):**
  The IP Office listens to RIP1 and RIP2 messages and uses these to update its routing table. However the IP Office does not respond.
- **RIP1:**
  The IP Office listens to RIP1 and RIP2 messages. It advertises its own routes in a RIP1 sub-network broadcast.
- **RIP2 Broadcast (RIP1 Compatibility):**
  The IP Office listens to RIP1 and RIP2 messages. It advertises its own routes in a RIP2 sub-network broadcast. This method is compatible with RIP1 routers.
- **RIP2 Multicast:**
  The IP Office listens to RIP1 and RIP2 messages. It advertises its own routes to the RIP2 multicast address (249.0.0.0). This method is not compatible with RIP1 routers.

Broadcast and multicast routes (those with addresses such as 255.255.255.255 and 224.0.0.0) are not included in RIP broadcasts. Static routes (those in the **IP Route** table) take precedence over a RIP route when the two routes have the same metric.

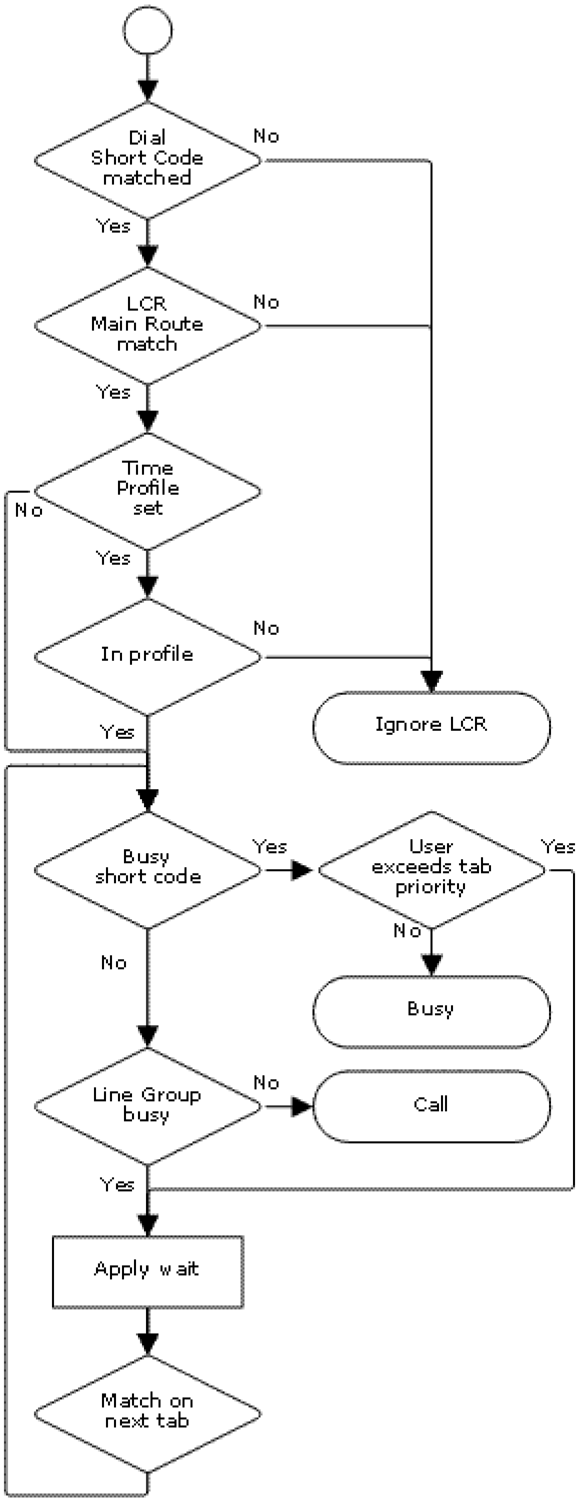
# Least Cost Routing Settings

## Least Cost Routing Overview



**Summary: Least cost routes allow short code matching on the number being dialed from the system rather than the number originally dialed by the user or application.**

For IP Office 4.0, Least Cost Routes have been replaced by ARS. When an IP Office system is upgraded to IP Office 4.0, the LCR entries are automatically replaced by ARS entries and appropriate short codes.




When a line, user, user rights or system short code results in a number to be dialed off-switch, the resulting telephone number to be dialed can be further processed by matching to Least Cost Route (LCR) short codes.

LCR short codes are grouped in sets. Within each set, the short codes are grouped into tabs called **Main Route**, **Alternate Route 1** and **Alternate Route 2**. Each tab also has a priority and a timeout setting.

- **Using a Time Profile**
  Each LCR set can have an associated time profile. LCR sets without a time profile are active all the time. LCR sets with a time profile are only active within the times defined by that profile.
- **Which Number is Used For Matching**
  The telephone number output by the original matched short code is checked against the Main Route tab short codes of the active LCR sets.
  - If a match is found, that set is used for processing.
  - If no match is found, the calls is dialed without LCR.
- **Returning Busy**
  If the LCR short code match is set to the **Busy** feature:
  - If the user's priority is higher than the LCR tabs, the IP Office will immediately look for a matching short code on the next tab and use that short code if found.
  - Otherwise the user receives busy tone.
- **Switching Outgoing Line Groups**
  If the LCR short code match is a dial feature, the IP Office will attempt to seize a line from the outgoing line group specified by the LCR short code.
  - If a line cannot be seized within the time specified on the LCR tab, the IP Office will look in the next tab for an alternate LCR short code match. If an alternate match is found it is used.

# Least Cost Routing Example

Site A has two outgoing line groups. Outgoing line group 0 contains external lines to the public telephone network. Outgoing line group 1 contains private lines to Site B.

**Requirements**

- **Scenario 1**
  The external public number for Site B is 123456. The internal speed dial number is 600. When a user dials 600, the administrator want the call to be routed by the private lines if possible.
- **Scenario 2**
  The sales hot line at Site B has the public number 654321. The administrator only want high priority users at Site A to be able to dial that number to test its performance.

**Settings**

- **System Short Code 1:** 600/123456/Dial/0.
- **System Short Code 2:** 654321/N/Dial/0.
- **User 1:** Priority 2. **User 2:** Priority 4.

| Least Cost Route "SiteB" | Main Route | Alternate Route 1 |
|---|---|---|
| **Timeout** | 10 | 30 |
| **Priority** | 3 | 5 |
| **Short Codes** | 123456/N/Dial/1 | 123456/N/Dial/0 |
| | 654321/N/Busy | 654321/N/Dial/0 |

**Effects**

**Scenario 1**

When a user dials 123456, it matches system short code 1. That short code specifies dialing Site B via the public lines (Outgoing line group 0).

The number to be dialed is checked against the least cost routing **Main Route** tabs for any match. In this example a match occurs in the ***SiteB*** least cost route. The short code there specifies dialing the number using the private lines (Outgoing line group 1).

If the IP Office cannot seize a line for the call from that group within 10 seconds, it looks for an alternate short code match in Alternate Route 1 tab of the Site B least cost route. In this example that match changes the call to using the public lines (Outgoing line group 0).

**Scenario 2**

When a user dials 654321, it matches system short code 2. That short code specifies dialing the Site B sale hot line number via the public lines (Outgoing line group 0).

Since this short code is set to a **Dial** feature, the number to be dialed is checked against the least cost routing **Main Route** tabs for any match. In this example a match occurs in the ***SiteB*** least cost route. The short code there specifies Busy and so returns busy to callers.

User 1 has a priority of 2. They will receive busy tone when they dial 654321.

User 2 has a priority of 4 which is higher than the Main Route tab in the Site B least cost route. Therefore the IP Office will immediately check for a further match in the Alternate Route 1 tab. In this example the short code match for 654321 in the Alternate 1 tab allows the number to be dialed to the public lines.

# Least Cost Routing | LCR

| Least Cost Routing \| LCR | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✗. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✗, **4.1** ✗. |
| **Mergeable** | ✓. |

- **Name**
  The name to identify the LCR set.
- **Time Profile:** *Default = Blank*
  Selects a time profile that is used to define when this least cost route can be used. If no profile is selected the route settings apply at all times.

# Least Cost Routing | Main Route

This tab is used for the initial short code matching. The match is performed on the telephone number outputs by a Line, User, User Rights or System short code that resulted in a number to dial. If a match is found it is used, otherwise the call is dialed using the original short code.

| Least Cost Routing \| Main Route | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✗. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✗, **4.1** ✗. |
| **Mergeable** | ✓. |

- **Timeout (secs):** *Default = 30 seconds. Range = 1 to 99999 seconds or 0 (No IP trunk fallback).* If an LCR short code match is found, the IP Office will attempt to seize a line in the outgoing line group specified by that short code. If, after this timeout period has expired, the IP Office still cannot seize a line, it will look for a alternate short code match in the **Alternate Route 1** tab.
- **Priority:** *Default = 5, Range 1 (lowest) to 5 (highest).* Normally, if the LCR short code match is a short code set to Busy, the user will receive busy tone. For users whose own **Priority** setting (**User | User | Priority**) is higher than the Main Route tab's, the IP Office will look for an alternate short code match in the **Alternate Route 1** tab.
- **Allow Bump:** *Default = Off* When a line indicated by the LCR short code match cannot be seized because lines in that outgoing line group are being used for a multilink PPP data call, this options allows a line to be seized from the data call.
- **Short Code List** These are the short codes used for matching against the telephone number output Line, User, User Right or System dial short code.
    - The only short code features that should be used in a Least Cost Route short code are: **Dial**, **Dial3K1**, **Dial56K**, **Dial64K**, **DialEmergency**, **DialSpeech**, **DialV110**, **DialV120**, **DialVideo** and **Busy**.
    - The **;** character and **[ ]** characters cannot be used.
    - Short codes can be added and edited using the **Add**, **Remove** and **Edit** buttons. Alternatively you can right-click on the list of existing short code to add and edit short codes.

# Least Cost Routing | Alternate Route 1

This tab of a Least Cost Route is used in the following cases:

- The short code match on the **Main Route** tab is set to **Busy** but the user has a higher priority than that tab.
- The **Main Route** tab timeout has expired while trying to seize a line from the outgoing line group specified by the short code match on that tab.

In either case the IP Office will look for an alternate short code match on this tab. The match is performed on the telephone number outputs by a Line, User, User Rights or System short code that resulted in a number to dial. If a match is found it is used, otherwise the call receives busy.

| Least Cost Routing \| Alternate Route 1 | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✗. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✗, **4.1** ✗. |
| **Mergeable** | ✓. |

- **Timeout (secs):** *Default = 30 seconds. Range = 1 to 99999 seconds or 0 (No IP trunk fallback).*
  If an LCR short code match is found, the IP Office will attempt to seize a line in the outgoing line group specified by that short code. If, after this timeout period has expired, the IP Office still cannot seize a line, it will look for a alternate short code match in the **Alternate Route 2** tab.
- **Priority:** *Default = 5, Range 1 (lowest) to 5 (highest).*
  Normally, if the LCR short code match is a short code set to Busy, the user will receive busy tone. For users whose own **Priority** setting (**User | User | Priority**) is higher than the Main Route tab's, the IP Office will look for an alternate short code match in the **Alternate Route 2** tab.
- **Allow Bump:** *Default = Off*
  When a line indicated by the LCR short code match cannot be seized because lines in that outgoing line group are being used for a multilink PPP data call, this options allows a line to be seized from the data call.
- **Short Code List**
  These are the short codes used for matching against the telephone number output Line, User, User Right or System dial short code.
  - The only short code features that should be used in a Least Cost Route short code are: **Dial**, **Dial3K1**, **Dial56K**, **Dial64K**, **DialEmergency**, **DialSpeech**, **DialV110**, **DialV120**, **DialVideo** and **Busy**.
  - The **;** character and **[ ]** characters cannot be used.
  - Short codes can be added and edited using the **Add**, **Remove** and **Edit** buttons. Alternatively you can right-click on the list of existing short code to add and edit short codes.

# Least Cost Routing | Alternate Route 2

This tab of a Least Cost Route is used in the following cases:

- The short code match on the **Alternate Route 1** tab is set to **Busy** but the user has a higher priority than that tab.
- The **Alternate Route 1** tab timeout has expired while trying to seize a line from the outgoing line group specified by the short code match on that tab.

In either case the IP Office will look for an alternate short code match on this tab. The match is performed on the telephone number outputs by a Line, User, User Rights or System short code that resulted in a number to dial. If a match is found it is used, otherwise the call receives busy.

| **Least Cost Routing \| Alternate Route 2** | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✗. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✗, **4.1** ✗. |
| **Mergeable** | ✓. |

- **Timeout (secs)**
  Not used. This is the last tab within a Least Cost Route.
- **Priority**
  Not used. This is the last tab within a Least Cost Route.
- **Allow Bump:** *Default = Off*
  When a line indicated by the LCR short code match cannot be seized because lines in that outgoing line group are being used for a multilink PPP data call, this options allows a line to be seized from the data call.
- **Short Code List**
  These are the short codes used for matching against the telephone number output Line, User, User Right or System dial short code.
  - The only short code features that should be used in a Least Cost Route short code are: **Dial**, **Dial3K1**, **Dial56K**, **Dial64K**, **DialEmergency**, **DialSpeech**, **DialV110**, **DialV120**, **DialVideo** and **Busy**.
  - The **;** character and **[ ]** characters cannot be used.
  - Short codes can be added and edited using the **Add**, **Remove** and **Edit** buttons. Alternatively you can right-click on the list of existing short code to add and edit short codes.

# Account Code Settings

## Account Code Overview

Account codes are commonly used to control cost allocation and out-going call restriction. The account code used on a call is included in the call information output by the system's call log. Incoming calls can also trigger account codes automatically by matching the Caller ID stored with the account code.

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

The maximum recommended number of accounts codes is 1000.

**Setting a User to Forced Account Code**

1. Receive the system configuration if one is not opened.
2. In the left-hand panel, click  **User**. The list of existing user is shown in the right-hand panel.
3. Double-click the required user.
4. Select the **Telephony** tab.
5. Tick the **Force Account Code** option.
6. Click **OK**.
7. Merge the configuration.

**Forcing Account Code Entry for Specific Numbers**
Account code can be set a being required for any dialing that matches a particular short code. This is done by ticking the Force Account Code option found in the short code settings. Note that the account code request happens when the short code match occurs. Potentially this can be in the middle of dialing the external number, therefore the use of X wildcards in the short code to ensure full number dialing is recommended.

**Entering Account Codes**
The method for entering account codes depends on the type of phone being used. Refer to the relevant telephone User's Guide for details.

- **Account Code Button**
  The **Account Code Entry** action (**User | Button Programming | Emulation | Account Code Entry**) and **Set Account Code** action (**User | Button Programming | Advanced | Set | Set Account Code**) can be assigned to a programmable button on some phones. They both operate the same. The button can be preset with a specific account code or left blank to request account code entry when pressed. The button can then be used to specify an account code before a call or during a call.
- **Phone Manager**
  The IP Office Phone Manager application can be used to enter account codes before or during calls. For full details refer to the Phone Manager documentation.
  - To enter an account code before making a call or during a call select Actions | Account Code. A valid account code can then be selected from the Account Code drop down.
  - The **Account Codes** tab can be used to create icons to speed dial specific numbers and account codes that are regularly used.
- **Setting an Account Code using Short Codes**
  The **Set Account Code** feature allows short codes to be created that specify an account code before making a call.

- **Show Account Code Setting**
  This System | Telephony setting controls the display and listing of system account codes:
  - When on
    - When entering account codes through Phone Manager, users can select from a drop-down list of available account codes.
    - When entering account codes through a phone, the account code digits are shown while being dialed.
  - When off
    - Within Phone Manager the drop-down list of available account codes is not useable, instead account codes must be entered using the Phone Managers PIN Code features.
    - When entering account codes through a phone, the account code digits are replaced by **s** characters on the display.

# Account Code | Account Code

This tab is used to define an individual account code.

| Account Code \| Account Code | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Account Code**
  Enter the account code required. The code can include alphabetic characters for users dialing via Phone Manager. It can also include wildcards; **?** matches a single digit and ***** matches any digits.
- **Caller ID**
  A caller ID can be entered and used to automatically assign an account code to calls made to or received from caller ID.

# Account Code | Voice Recording

This tab is used to activate the automatic recording of external calls when the account code is entered at the start of the call or automatically assigned by call ID matching when the call is received. This requires Voicemail Pro to be installed and running. Call recording also requires available conference resources similar to a 3-way conference.

The recordings are placed in the mailbox of the user making or receiving the call. This option cannot be triggered by entry of the account code during a call.

- IP Office 4.0+ introduces the following changes to user account code recording:
    - Calls to and from IP devices including those using Direct media can be recorded.
    - User account code recording stops if the call is transferred to another user.
    - Calls parked or held by the user pause recording until the user unparks the call or takes it off hold.
- For IP Office 4.1+, the destination mailbox for the recording can be specified.

| Account Code \| Voice Recording | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Record Outbound:** *Default = None*
  Select whether outbound external calls are recorded. Options are **On**, **Mandatory** and then various percentages of calls made by the user.
    - **On:** Record the call if possible.
    - **Mandatory:** If not possible to record, return busy tone to the caller.
- **Record Inbound:** *Default = None, Software level = Up to 3.2 only.*
  Select whether inbound external calls are recorded. Requires the account code to be assigned to the call by matching the caller ID associated with the account code on the **Account Code | Account Code** tab. For IP Office 4.0 and higher, recording calls from a particular caller ID can be setup using incoming call route voice recording.
- **Record Time Profile:** *Default = Blank (Any time)*
  Used to select a time profile during which the settings above are used for call recording. Outside this period calls are not recorded using these settings.
- **Auto Recording Mailbox:** *Default = <None> (Calling user's mailbox), Software level = 4.1+.*
  Sets the mailbox into which automatically triggered recordings are placed.
    - **Voice Recording Library (Auto):** *Default = Off, Software level = 3.0+.*
      When checked, this setting marks the recording for collection by the ContactStore for IP Office application rather than being placed in the user's mailbox. Note that Voicemail Pro still performs the actual recording.

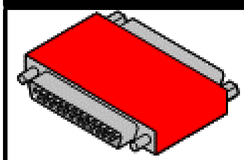
# License Settings

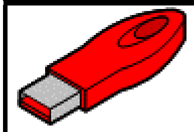
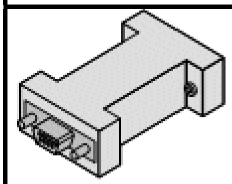
## License | License

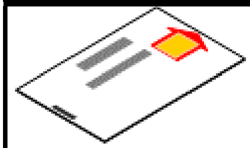
This form is used to display the function, value and status of license keys entered into the IP Office configuration. License keys are 32 character strings uniquely based on the feature they active and the serial number of a Feature Key dongle being used by the IP Office control unit.

IP Office Feature Key dongles are available in several types. Each IP Office system only supports license validation against a single dongle and vice versa. The licenses in the IP Office systems configuration must match the serial number of the Feature Key dongle. Depending on the dongle type, it is installed either directly on the IP Office control unit or onto a PC running the IP Office Feature Key Server application.

- Each IP Office will only support license validation against one feature key dongle.
- If being used, a Feature Key Server PC will only validate licenses for the first IP Office to which it connects after starting.
- For parallel and USB feature keys, the address of the PC hosting the dongle and running the IP Office Feature Key Server software is set by the **License Server IP Address** setting on the **System | System** tab. For serial key dongles, the address is set to 0.0.0.0.
- Note that for the IP500 control unit, a smart card Feature Key dongle must be present even if no licensed features are being used. For the IP500 control unit, the dongle fitted controls default operation.

| IP Office Feature Key Dongle | | Server PC Required | License Server IP Address | SOE | IP403 | IP406 v1 | IP406 V2 | IP412 | IP500 |
|---|---|---|---|---|---|---|---|---|---|
|  | **Parallel** This type of feature key dongle is plugged into the parallel port of a PC running the IP Office Feature Key Server software. | ✓ | 255.255.255.255 or server PC address. | ✓ | ✓ | ✓ | ✓ | ✓ | ✗ |

| IP Office Feature Key Dongle | | Server PC Required | License Server IP Address | SOE | IP403 | IP406 v1 | IP406 V2 | IP412 | IP500 |
|---|---|---|---|---|---|---|---|---|---|
|  | **USB** This type of feature key dongle is plugged into the USB port of a PC running the IP Office Feature Key Server software. | ✓ | 255.255.255.255 or server PC address. | ✓ | ✓ | ✓ | ✓ | ✓ | ✗ |
|  | **Serial** This type of feature key dongle is plugged into the 9-pin serial port on the back of IP Office Small Office Edition and IP406 V2 control units. No separate PC running IP Office Feature Key software is required. | ✗ | 0.0.0.0 | ✓ | ✗ | ✗ | ✓ | ✓ | ✓ |

| IP Office Feature Key Dongle | | Server PC Required | License Server IP Address | SOE | IP403 | IP406 v1 | IP406 V2 | IP412 | IP500 |
|---|---|---|---|---|---|---|---|---|---|
|  | **Smart Card** This type of feature key dongle, a credit-card sized memory card, is used with the IP500 control unit. No separate PC running IP Office Feature Key software is required. | ✗ | Not used. | ✗ | ✗ | ✗ | ✗ | ✗ | ✓ |

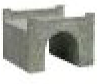
**Importing License Keys**
It is recommended that licenses are cut and pasted electronically. This removes the chances of errors due to mistyping and misinterpretation of characters fonts. Where multiple licences need to be added, the CSV import option can be used (**File | Import/Export | Import**). Licenses imported this way may be listed as invalid until the configuration is saved and then reloaded.

| License \| License | |
|---|---|
| **Control Unit** | **SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓.** |
| **Software Level** | **2.1 ✓, 3.0DT ✓, 3.0 ✓, 3.1 ✓, 3.2 ✓, 4.0 ✓, 4.1 ✓.** |
| **Mergeable** | ✓. |

- **License Key**
  This field is used to enter the 32-character license key.
- **License Type**
  For information only. If the key is recognized, its function will be listed here. **Invalid** indicates that the License Key has not been recognized as enabling any feature, regardless of the Feature Key dongle serial number. For a list of licenses and their purpose refer to the IP Office Installation manual.
- **License Status**
  For information only. This field indicates the current validation status of the license key against the serial number of the Feature Key dongle being used by the IP Office system.
  - **Unknown** is shown for newly entered licenses until the configuration is sent to the IP Office and then retrieved again.
  - **Valid** is shown if the license key matches the Feature Key dongle serial number.
  - **Invalid** is shown if the license key does not match the Feature Key dongle serial number.
- **Instances**
  For information only. Some licenses enable a number of port, channels or users. When that is the case, the number of such is indicated here. Multiple licences for the same feature are usually cumulative.
- **Expiry Date**
  For information only. License can be set to expire within a set period from their issue by Avaya. The expiry date is shown here.

# Tunnel Settings

## Tunnel



Tunneling allows additional security to be applied to IP data traffic. This is useful when sites across an unsecure network such as the public internet. The IP Office supports two methods of tunneling, L2TP and IPSec. Once a tunnel is created, it can be used as the destination for selected IP traffic in the **IP Route** table.

Two types of tunnelling are supported:

-  **L2TP - Layer 2 Tunneling Protocol**
  PPP (Point to Point Protocol) authentication normally takes place between directly connected routing devices. For example when connecting to the internet, authentication is between the customer router and the internet service provider's equipment. L2TP allows additional authentication to be performed between the routers at each end of the connection regardless of any intermediate network routers. The use of L2TP does not require an IP Office license.

-  **IPSec**
  IPSec allows data between two locations to be secured using various methods of sender authentication and or data encryption. The use of IPSec requires entry of an IPSec Tunneling license (IP400 IPSec VPN RFA) into the IP Office at each end.

# L2TP Tunnel

## Tunnel | Tunnel (L2TP)

| Tunnel \| Tunnel (L2TP) | |
|---|---|
| Control Unit | SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓. |
| Software Level | 2.1 ✓, 3.0DT ✓, 3.0 ✓, 3.1 ✓, 3.2 ✓, 4.0 ✓, 4.1 ✓. |
| Mergeable | ✗. |

- **Name:** *Default = Blank.*
  A unique name for the tunnel. Once the tunnel is created, the name can be selected as a destination in the IP Route table.

- **Local Configuration**
  The account name and password is used to set the PPP authentication parameters.

  - **Local Account Name**
    The local user name used in outgoing authentication.

  - **Local Account Password/Confirm Password**
    The local user password. Used during authentication.

  - **Local IP Address**
    The source IP address to use when originating an L2TP tunnel. By default (un-configured), IP Office uses the IP address of the interface on which the tunnel is to be established as the source address of tunnel.

- **Remote Configuration**
  The account name and password is used to set the PPP authentication parameters.

  - **Remote Account Name**
    The remote user name that is expected for the authentication of the peer.

  - **Remote Account Password/Confirm Password**
    The password for the remote user. Used during authentication.

  - **Remote IP Address**
    The IP address of the remote L2TP peer or the local VPN line IP address or the WAN IP address.

- **Minimum Call Time (Mins):** *Default = 60 minutes, Range = 1 to 999.*
  The minimum time that the tunnel will remain active.

- **Forward Multicast Messages:** *Default = On*
  Allow the tunnel to carry multicast messages when enabled.

- **Encrypted Password:** *Default = Off*
  When enabled, the CHAP protocol is used to authenticate the incoming peer.

# Tunnel | L2TP (L2TP)

| Tunnel \| L2TP | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

- **Shared Secret/Confirm Password**
  User setting used for authentication. Must be matched at both ends of the tunnel. This password is separate from the PPP authentication parameters defined on the **L2TP|Tunnel** tab.

- **Total Control Retransmission Interval:** *Default = 0, Range = 0 to 65535.*
  Time delay before retransmission.

- **Receive Window Size:** *Default = 4, Range = 0 to 65535.*
  The number of unacknowledged packets allowed.

- **Sequence numbers on Data Channel:** *Default = On*
  When on, adds sequence numbers to L2TP packets.

- **Add checksum on UDP packets:** *Default = On.*
  When on, uses checksums to verify L2TP packets.

- **Use Hiding:** *Default = Off*
  When on, encrypts the tunnel's control channel.

# Tunnel | PPP (L2TP)

| Tunnel \| PPP (L2TP) | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

- **CHAP Challenge Interval (secs):** *Default = 0 (Disabled), Range = 0 to 99999 seconds.*
  Sets the period between CHAP challenges. Blank or 0 disables repeated challenges. Some software (such as Windows 95 DUN) does not support repeated challenges.
- **Header Compression:** *Default = None*
  Select header compression. Options are: ***IPHC*** and/or ***VJ***.
- **PPP Compression Mode:** *Default = MPPC*
  Select the compression mode for the tunnel connection. Options are: ***Disable***, ***StacLZS*** or ***MPPC***.
- **Multilink / QoS:** *Default = Off*
  Enable the use of Multilink protocol (MPPC) on the link.
- **Incoming traffic does not keep link up:** *Default = On*
  When enabled, the link is not kept up when the only traffic is incoming traffic.
- **LCP Echo Timeout (secs):** *Default = 6, Range = 0 to 99999 seconds.*
  When a PPP link is established, it is normal for each end to send echo packets to verify that the link is still connected. This field defines the time between LCP echo packets. Four missed responses in a row will cause the link to terminate.

# IP Security Tunnel

## Tunnel | Main (IPSec)

| Tunnnel \| Main (IPSec) | |
|---|---|
| Control Unit | SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓. |
| Software Level | 2.1 ✓, 3.0DT ✓, 3.0 ✓, 3.1 ✓, 3.2 ✓, 4.0 ✓, 4.1 ✓. |
| Mergeable | ✗. |

- **Name:** *Default = Blank*.
  A unique name for the tunnel. Once the tunnel is created, the name can be selected as a destination for traffic in the IP Route table.
- **Local Configuration**
  The IP Address and IP Mask are used in conjunction with each other to configure and set the conditions for this Security Association (SA) with regard to inbound and outbound IP packets.
    - **IP Address**
      The IP address or sub-net for the start of the tunnel.
    - **IP Mask**
      The IP mask for the above address.
    - **Tunnel Endpoint IP Address**
      The local IP address to be used to establish the SA to the remote peer. If left un-configured, IP Office will use the IP address of the local interface on which the tunnel is to be configured.
- **Remote Configuration**
  The IP Address and IP Mask are used in conjunction with each other to configure and set the conditions for this Security Association (SA) with regard to inbound and outbound IP packets.
    - **IP Address**
      The IP address or sub-net for the end of the tunnel.
    - **IP Mask**
      The IP mask for the above address.
    - **Tunnel Endpoint IP Address**
      The IP address of the peer to which a SA must be established before the specified local and remote addresses can be forwarded.

# Tunnel | IKE Policies (IPSec)

| Tunnel \| LKE Policies (IPSec) | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

- **Shared Secret/Confirm Password**
  The password used for authentication. This must be matched at both ends of the tunnel.
- **Exchange Type:** *Default = ID Prot*
  **Aggressive** provides faster security setup but does not hide the ID's of the communicating devices. **ID Prot** is slower but hides the ID's of the communicating devices.
- **Encryption:** *Default = DES CBC*
  Select the encryption method used by the tunnel. The options are: ***DES CBC***, ***3DES*** or ***Any***.
- **Authentication:** *Default = MD5*
  The method of password authentication. Options are: ***MD5***, ***SHA*** or ***Any***.
- **DH Group:** *Default = Group 1*
- **Life Type:** *Default = KBytes*
  Sets whether **Life** (below) is measured in seconds or kilobytes.
- **Life:** *Range = 0 to 99999999.*
  Determines the period of time or the number of bytes after which the SA key is refreshed or re-calculated.

# Tunnel | IPSec Policies (IPSec)

| Tunnel \| IPSec Policies | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

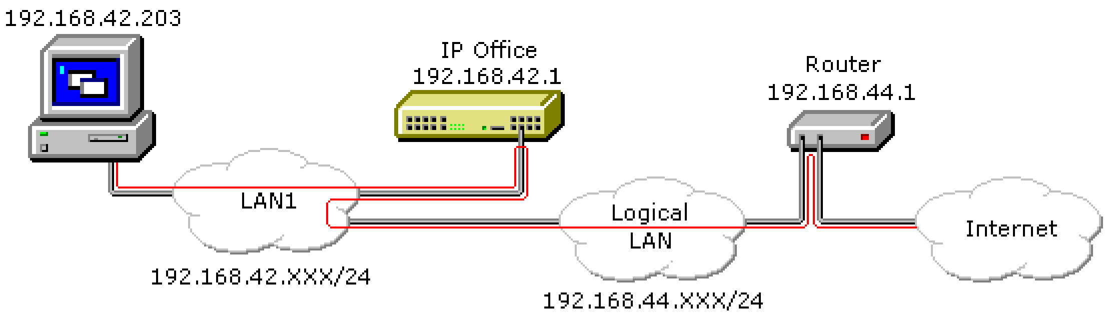
- **Protocol:** *Default = ESP*
  ESP (Encapsulated Security Payload) or AH (Authentication Header, no encryption).
- **Encryption:** *Default = DES*
  Select the encryption method used by the tunnel. The options are: ***DES CBC***, ***3DES*** or ***Any***.
- **Authentication:** *Default = HMAC MD5*
  The method of password authentication. Options are: ***HMAC MD5***, ***HMAC SHA*** or ***Any***.
- **Life Type:** *Default = KBytes*
  Sets whether **Life** (below) is measured in seconds or kilobytes.
- **Life**
  Determines the period of time or the number of bytes after which the SA key is refreshed or re-calculated.

# Logical LAN Settings

## Logical LAN

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.




| Logical LAN \| Logical LAN | |
|---|---|
| Control Unit | SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓. |
| Software Level | 2.1 ✓, 3.0DT ✓, 3.0 ✓, 3.1 ✓, 3.2 ✓, 4.0 ✓, 4.1 ✓. |
| Mergeable | ✗. |

- **Name:** *Default = Blank. Range = Up to 31 characters.*
  A unique name for the logical LAN. This name becomes selectable as a destination in the IP Route table.
- **IP Address:** *Default = 0.0.0.0*
  The IP address provided by the internet service provider for the logical LAN.
- **IP Mask:** *Default = 0.0.0.0*
  The IP address mask provided by the internet service provider for the logical LAN.
- **Gateway IP Address:** *Default = 0.0.0.0*
  The IP address of the router on the logical LAN.
- **Gateway Mac Address:** *Default = 00:00:00:00:00:00*
  The MAC address of the router. If the MAC address isn't known, from a PC that can ping the router's IP address, use the command ***arp -a <ip address***>.
- **Firewall Profile:** *Default = Blank*
  This field allows selection of an existing IP Office firewall profile that should be applied to traffic to and from the logical LAN.
- **Enable NAT:** *Default = On (Grayed out).*
  NAT is applied to all traffic from the IP Office LAN to the logical LAN. The use of NAT is not compatible with H.323 VoIP operation, therefore a VPN tunnel should also be applied to traffic being routed

# Wireless Settings

## Wireless Overview

The Small Office Edition control unit can act as an 802.11b wireless access point. To do this requires the insertion of an Avaya supplied IP Office wireless card into one of the control unit's PCMCIA slots and entry of a Small Office Edition WiFi license into the configuration. The IP Office Wireless settings can then be configured.

In order to connect to the IP Office LAN, wireless devices must be configured to match the IP Office Wireless settings. Additionally the wireless device must match the control unit's LAN1 or LAN2 network settings unless using IP Office DHCP.

# Wireless | SSID

This tab is used to set the general identity of the wireless connection to the IP Office LAN.

| Wireless \| SSID | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✗, **IP406 V1** ✗, **IP406 V2** ✗, **IP412** ✗, **IP500** ✗. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

- **Network Name:** *Default = IP Office Wireless.Net*
  A unique name used to identify and distinguish the IP Office wireless LAN from other wireless LAN's. This is the wireless LAN's Service Set Identifier (SSID).
- **Wireless Mac Address**
  Displays a list of the MAC addresses of the devices currently connected to the wireless LAN.
- **Frequency/Channel:** *Default = 6*
  The 802.11b wireless frequency band is sub-divided into a number of channels. In locations where there are multiple wireless LAN's or multiple access points to the same wireless LAN, each access point should use a separate channel. Devices connecting to a wireless LAN will automatically connect to the channel providing the strongest signal.
  - The number of channels available is country specific. In the US channels 1 to 11 are available. In most of Europe, channels 1 to 13 are available. In Japan only channel 14 is available.
  - The channel frequencies overlap. For instance, channel 2 shares part of the same frequency band as channels 1 and 3. In areas with multiple access points or LAN, use widely spaced channels. For example uses channels 1, 6 and 11 on different access points.
- **Accept Any:** *Default = Off*
  If on, allows any wireless device to connect to the wireless LAN without having to have a matching wireless network name (SSID) set. When off, only devices configured with a matching wireless network name can connect to the wireless LAN.

# Wireless | Security

This tab allows for additional security through the use of WEP wireless encryption keys. If enabled, in addition to encrypting the wireless traffic, only devices using a matching encryption key can connect to the wireless LAN.

| Wireless \| Security | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✗, **IP406 V1** ✗, **IP406 V2** ✗, **IP412** ✗, **IP500** ✗. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✗. |

- **Encryption:** *Default = Disabled*
  Allows selection of 50/64 bit or 128 bit security. Note: 50/64 bit encryption is also know as 40/64 encryption in some locales.
- **Alpha/Hex:** *Default = Hex*
  Switch key entry between hexadecimal and alphabetic entry modes.
- **Key 1/4**
  Allows entry of the security key and selection of which key is the current key to use.

# User Restrictions Settings

## User Restrictions Overview

**User Restrictions** are only available in pre-IP Office 3.2 configurations. For IP Office 3.2 and higher they have been replaced by **User Rights**.

Within Manager, users can be grouped by the types of numbers they are allowed to dial or not allowed to dial. For example, those who are allowed to dial 1900 or international numbers.

The **User Restriction** form allows named groups of dialing short codes/restrictions to be created. These short codes can then be applied to a user by associating them with the **User Restriction** name rather than having to recreate the short codes for each user.

**To set up a restriction within the User Restriction form**

1. Click **User Restriction** form within the Configuration Tree.
2. Enter a name for the restriction.
3. Click the **Short Code** List tab and create a short code.
4. Merge the configuration.

**To apply a User Restriction to a specific user**

1. Click the **User** form within the Configuration Tree.
2. Double-click the user for whom you want this restriction applied.
3. Within the **User** tab, click the **Restriction** drop down box and select the **User Restriction** you want applied to this user.
4. Merge the configuration.

# Restrictions

**User Restrictions** are only available in pre-IP Office 3.2 configurations. For IP Office 3.2 and higher they have been replaced by **User Rights**.

| User Restrictions \| Restrictions | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✗. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✗, **4.0** ✗, **4.1** ✗. |
| **Mergeable** | ✓. |

- **Name:** *Default = Blank*
  A name used to identify the set of user restrictions and allow its selection through the **Restrictions** field in each individual user's **User** settings.
- **Priority:** *Default = 5 (highest), Range 0 to 5*
  The priority that should be applied to user calls if routed via a Least Cost Route. This overrides the priority of the individual user.
- **Outgoing Call Bar:** *Default = Off.*
  When on, bars users making external calls.

# User Restrictions | Short Codes

**User Restrictions** are only available in pre-IP Office 3.2 configurations. For IP Office 3.2 and higher they have been replaced by **User Rights**.

- **WARNING**
  User dialing of emergency numbers must not be blocked by the addition of short codes. If short codes are added, the users ability to dial emergency numbers must be tested and maintained.

Allows entry of short codes for dialing by associated users. These short codes override any match system short codes but not individual user short codes.

| User Restrictions \| Short Codes | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✗. |
| **Software Level** | **2.1** ✓, **3.0DT** ✓, **3.0** ✓, **3.1** ✓, **3.2** ✗, **4.0** ✗, **4.1** ✗. |
| **Mergeable** | ✓. |

# User Rights Settings

## User Rights Overview

User Rights act as templates for users, locking selected user settings to the template value. For most of the settings within the user rights tabs, the following options can be selected from an adjacent drop down list. Note that some settings are grouped and are set and locked as a group.

- **Apply User Rights Value**
  Apply the value set in the user rights to all associated users.
    - The matching user setting is grayed out and displays a lock symbol.
    - Users attempting to change the settings using short codes receive inaccessible tone.
    - Within the user's Phone Manager the associated fields are grayed out or hidden.
- **Not Part of User Rights**
  Ignore the setting.

**Adding User Rights**

1. Select **User Rights**.
2. Click and select **User Rights**.
3. Enter a name.
4. Configure the user rights as required.
5. Click **OK**.

**Creating User Rights Based on an Existing User**

1. Select **User Rights**.
2. In the group pane, right-click and select **New User Rights from User**.
3. Select the user and click **OK**.

**Associating User Rights to a User**

1. Select **User Rights** or **User**.
2. In the group pane, right-click and select **Apply User Rights to Users**.
3. Select the user rights to be applied.
4. On the **Members of this User Rights** sub tab select the users to which the user rights should be applied as their **Working Hours User Rights**.
5. On the **Members when out of hours** sub tab select which users should use the selected user rights as their out of hours user rights.
6. Click **OK**.

or

1. Select the required user to display their settings in the details pane.
2. Select the **User** tab.
3. Use **Working Hours User Rights** drop-down to select the user rights required.
4. If required a **Working Hours Time Profile** and **Out of Hours User Rights** can be selected.
5. Click **OK**.

**Copy User Rights Settings over a User's Settings**
This process replaces a user's current settings with those that are part of the selected user rights. It does not associate the user with the user rights.

1. Select **User Rights** or **User**.
2. In the group pane, right-click and select **Copy user rights values to users**.
3. Select the user rights to be applied.
4. Click **OK**.

**Default User Rights**

For defaulted IP Office systems, the following user rights are created as a part of the default configuration. Fields not listed are not part of the user rights.

✓ = Set to **On**. ✗ = Set to **Off**. – = Not part of the user rights.

| User Rights | Call Center Agent | Boss | Application | Default | IP Hard Phone | Mailbox | Paging | T3 |
|---|---|---|---|---|---|---|---|---|
| **Priority** | ✓ 5 | ✓ 5 | ✓ 5 | ✓ 5 | ✓ 5 | ✓ 5 | ✓ 5 | ✓ 5 |
| **Voicemail** | ✓ | – | – | – | – | ✓ | – | – |
| **Voicemail Ringback** | ✗ | ✗ | ✗ | ✗ | ✗ | ✗ | – | ✗ |
| **Outgoing Call Bar** | ✗ | ✗ | ✗ | ✗ | ✗ | ✗ | ✗ | ✗ |
| **No Answer Time** | ✓ 0 | ✓ 0 | ✓ 0 | ✓ 0 | ✓ 0 | ✓ 0 | ✓ 0 | ✓ 0 |
| **Transfer Return Time** | ✓ 0 | ✓ 0 | ✓ 0 | ✓ 0 | ✓ 0 | ✓ 0 | ✓ 0 | ✓ 0 |
| **Individual Coverage Time** | ✓ 10 | ✓ 10 | ✓ 10 | ✓ 10 | ✓ 10 | ✓ 10 | ✓ 10 | ✓ 10 |
| **Busy on Held** | ✓ | ✗ | ✓ | ✗ | ✗ | – | – | ✗ |
| **Call Waiting** | ✗ | ✗ | ✓ | ✗ | ✗ | ✗ | ✗ | ✓ |
| **Can be Intruded** | ✗ | ✗ | ✗ | ✗ | ✗ | ✗ | ✗ | ✗ |
| **Cannot be Intruded** | ✗ | ✗ | ✓ | ✓ | ✓ | ✗ | ✗ | ✗ |
| **Force Login** | ✓ | – | – | – | – | – | – | – |
| **Force Account Code** | ✗ | ✗ | ✗ | ✗ | ✗ | ✗ | ✗ | ✗ |
| **Button Programming** | 1: a=<br>2: b=<br>4: HGEna<br>5: DNDOn<br>6: Busy | 1: a=<br>2: b=<br>3: c=<br>6: DNDOn<br>7: Dial *17 | – | 1: a=<br>2: b=<br>3: c= | 1: a=<br>2: b=<br>3: c=<br>6: Dial *17 | – | – | – |
| **Phone Manager Type** | ✓ Pro | ✓ Lite | ✓ Pro | ✓ Lite | ✓ Lite | ✓ Lite | ✓ Lite | ✓ Lite |

# User Rights | User

This tab is used to set and lock various user settings.

| User Rights \| User | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✗, **3.0DT** ✗, **3.0** ✗, **3.1** ✗, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Name**
  The name for the user rights . This must be set in order to allow the user rights to be selected within the **User Rights** drop down list on the **User | User** tab of individual users.
- **Locale:** *Default = Blank*
  Sets and locks the language used for voicemail prompts to the user, assuming the language is available on the voicemail server. On a digital extension it also controls the display language used for messages from the IP Office to the phone. See **Supported Country and Locale Settings**.
- **Priority:** *Default = 5, Range 1 (Lowest) to 5 (Highest)*
  Sets and locks the user's priority setting for least cost routing.
- **Voicemail On:** *Default = On*
  Sets and locks the user's voicemail on setting.
- **Voicemail Ringback:** *Default = Off*
  Sets and locks the user's voicemail ringback setting.
- **Do Not Disturb:** *Default = Off*
  Sets and locks the user's DND status setting.
- **Outgoing Call Bar:** *Default = Off*
  When set, bars the user from making external calls.

# User Rights | Short Codes

This tab is used to set and lock the user's short code set. The tab operates in the same way as the **User | Short Codes** tab.

- **WARNING**
  User dialing of emergency numbers must not be blocked by the addition of short codes. If short codes are added, the users ability to dial emergency numbers must be tested and maintained.

Where the same short code exists in both the **User | Short Codes** tab and the associated **User Rights | Short Codes** tab, the IP Office will use the user short code.

| User Rights \| Short Codes | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✗, **3.0DT** ✗, **3.0** ✗, **3.1** ✗, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

Short codes can be added and edited using the **Add**, **Remove** and **Edit** buttons. Alternatively you can right-click on the list of existing short code to add and edit short codes.

# User Rights | Telephony

This tab allows various user telephony settings to be set and locked. These match settings found on the **User | Telephony** tab.

| User Rights \| Telephony | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✗, **3.0DT** ✗, **3.0** ✗, **3.1** ✗, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **No Answer Time:** *Default = Blank (Use system setting), Range = 1 to 99999 seconds.*
  Sets how long a call rings the user before following forwarded on no answer if set or going to voicemail. Leave blank to use the system default setting.
- **Transfer return Time (secs):** *Default = Blank (Off), Range 1 to 99999 seconds.*
  Sets the delay after which any call transferred by the user, which remains unanswered, should return to the user if possible.
- **Wrap up Time (secs):** *Default = 2 seconds, Range 0 to 99999 seconds.*
  Specifies the amount of time after ending one call before another call can ring. You may wish to increase this in a "call center" environment where users may need time to log call details before taking the next call. It is recommended that this option is not set to less than the default of 2 seconds. 0 is used for immediate ringing.
- **Individual Coverage Time (secs):** *Default = 10 seconds, Range 1 to 99999 seconds.*
  This function sets how long the phone will ring at your extension before also alerting at any call coverage users. This time setting should not be equal to or greater than the **No Answer Time**.
- **Call Waiting On:** *Default = Off*
  For users on phones without appearance buttons, if the user is on a call and a second call arrives for them, an audio tone can be given in the speech path to indicate a waiting call (the call waiting tone varies according to locale). The waiting caller hears ringing rather than receiving busy. There can only be one waiting call, any further calls receive normal busy treatment. If the call waiting is not answered within the no answer time, it follows forward on no answer or goes to voicemail as appropriate. User call waiting is not used for users on phones with multiple call appearance buttons. Call waiting can also be applied to hunt group calls, see ***Hunt Group | Hunt Group | Call Waiting***.
- **Busy on Held:** *Default = On*
  If on, when the user has a call on hold, new calls receive busy tone (ringing for incoming analog call) or are diverted to voicemail if enabled, rather than ringing the user. Note this overrides call waiting when the user has a call on hold.
- **Can Intrude:** *Default = Off*
  Check this option if the User can interrupt other user's calls. This setting and the setting below are used to control the use of the following short code and button features: **Call Intrude**, **Call Listen**, **Call Steal** and **Dial Inclusion**.
- **Cannot be Intruded:** *Default = On*
  If checked, this user's calls cannot be interrupted or acquired. In addition to the features listed above, this setting also affects whether other users can use their appearance buttons to bridge into a call to which this user has been the longest present user.
- **Force Login:** *Default = Off*
  If checked, the user must login using their **Login Code** to use an extension. For example, if **Force Login** is ticked for User A and user B has logged onto A's phone, after B logs off A must log back. If Force Login was not ticked, A would be automatically logged back on.
- **Force Account Code:** *Default = Off*
  If checked, the user must enter a valid account code to make an external call.
- **Inhibit Off-Switch Transfers:** *Default = Off*
  When enabled, this setting stops the user from transferring or forwarding calls externally. Note

that all user can be barred from forwarding or transferring calls externally by the **System | Telephony | Inhibit Off-Switch Transfers** setting.

# User Rights | Button Programming

This tab is used to set and lock the user's programmable button set. When locked, the user cannot use **Admin** or **Admin1** buttons on their phone to override any button set by their user rights.

Buttons not set through the user rights can be set through the user's own settings.

When **Apply user rights value** is selected, the tab operates in the same manner as the **User | Button Programming** tab.

| User Rights \| Button Programming | |
|---|---|
| Control Unit | SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓. |
| Software Level | 2.1 ✗, 3.0DT ✗, 3.0 ✗, 3.1 ✗, 3.2 ✓, 4.0 ✓, 4.1 ✓. |
| Mergeable | ✓. |

### Adding Blank Buttons

There are scenarios where users are able to program their own buttons but you may want to force certain button to be blank. This can be done through the user's associated User Rights as follows:

1. Assign the action **Emulation | Inspect** to the button. Enter some spaces as the button label.
2. When pressed by the user, this button will not perform any action. However it cannot be overridden by the user.

# User Rights | Menu Programming

This tab is used to set and lock the user's programmable button set.

When **Apply User Rights value** is selected, the tab operates in the same manner as the **User | Menu Programming** tab.

| User Rights \| Menu Programming | |
|---|---|
| Control Unit | SOE ✓, IP403 ✓, IP406 V1 ✓, IP406 V2 ✓, IP412 ✓, IP500 ✓. |
| Software Level | 2.1 ✗, 3.0DT ✗, 3.0 ✗, 3.1 ✗, 3.2 ✓, 4.0 ✓, 4.1 ✓. |
| Mergeable | ✓. |

# User Rights | Phone Manager

This tab is used to set and lock which parts of Phone Manager the associated users can use or adjust.

| User Rights \| Phone Manager | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✗, **3.0DT** ✗, **3.0** ✗, **3.1** ✗, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Allow user to modify Phone Manager settings:** *Default = On*
  This setting is used with the **Phone Manager Status Options**, **Screen Pop Options** and **Hide Options**. It controls whether those options are applied every time the user starts Phone Manager or only the first time the user starts Phone Manager.
  - If this setting is enabled, then the IP Office configuration setting of those options are only applied the first time a user starts Phone Manager on a PC. Those settings become part of the user's Phone Manager profile on that PC. They can be changed by the user through Phone Manager. On subsequent Phone Manager starts the Manager settings are ignored.
  - If this setting is not enabled, the IP Office configuration settings are applied every time the user starts Phone Manager and cannot be overridden by the user.
- **Agent Mode:** *Default = Off*
  This option controls the setting of the **Agent Mode** option on the **Configure Preferences | Agent Mode** tab within Phone Manager Pro. When enabled, the user has additional toolbar controls for **Busy Wrap Up**, **Busy Not Available** and **Select Group**. Note that the options on the Phone Manager Pro **Agent Mode** tab can be greyed out from user changes by the **Agent Mode** setting in **Configuration Options** below.
- **Phone Manager Type:** *Default = Lite*
  Determines the mode in which the user's copy of the Phone Manager application operates. This setting cannot be changed by the user. * For pre-3.2 IP Office systems this setting is located on the **User | User** tab.
  - **Lite**
    Basic Phone Manager mode. This mode does not require any licenses.
  - **Pro**
    Advanced Phone Manager mode that enables a range of additional functions. This mode requires an available Phone Manager Pro license, otherwise the application will run in Phone Manager Lite mode.
  - **Phone Manager PC Softphone**
    This is the VoIP IP phone mode of Phone Manager Pro. This mode requires both an available Phone Manager Pro license and a Phone Manager Pro IP Audio Enabled license. The user must be associated with an VoIP extension within the IP Office configuration.
  - **Phone Manager Pro Telecommuter:** *Software level = 4.1+.*
    This version of phone Manager Pro is supported with Phone Manager 4.1+. It allows the user to make and receive calls via an external phone specified at Phone Manager login. This mode requires an available Phone Manager Pro license, otherwise the application will run in Phone Manager Lite mode.
- **Enable VoIP:** *Default = Off*
  This option only appears if the selected **Phone Manager Type** is ***Phone Manager PC Softphone***. It enables or disables the matching setting on the user's Phone Manager PC Softphone.

- **Configuration Options**
  These options allow the user access to the indicated configure preferences tabs within Phone Manager. The controllable tabs for Phone Manager Lite are **Telephone** and **Do Not Disturb**. The additional controllable tabs for Phone Manager Pro and Phone Manager PC Softphone are **Screen Pop**, **Compact Mode**, **Agent Mode**, **Voicemail** (**Voicemail** and **Voicemail Ringback** controls only) and in IP Office 4.0 and higher **Mobile Twinning**. .
- **Screen Pop Options**
  These options allow selection of the Phone Manager Pro/Phone Manager PC Softphone screen pop options **Ringing**, **Answering**, **Internal**, **External** and **Outlook**. The **Allow user to modify Phone Manager settings** option controls whether these settings are applied only when Phone Manager is first started or every time Phone Manager is started.
- **Phone Manager Status Options**
  These options allow selection of the tabs to show within the call history area of the user's Phone Manager. The tabs selectable for Phone Manager are **All**, **Missed**, **Status** and **Messages**. The additional tabs selectable for Phone Manager Pro and PC Softphone are **Incoming**, **Outgoing** and **Account Code**. The **Allow user to modify Phone Manager settings** option controls whether these settings are applied only when Phone Manager is first started or every time Phone Manager is started.
- **Hide Options**
  These options allow selection of the Phone Manager Pro/Phone Manager PC Softphone options **Hide on close** and **Hide on no calls**. The **Allow user to modify Phone Manager settings** option controls whether these settings are applied only when Phone Manager is first started or every time Phone Manager is started.

# User Rights | Twinning

This tab is used to set and lock the following settings relating to the use of mobile twinning. Use of mobile twinning requires entry of a mobile twinning license.

| User Rights \| Twinning | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✗, **3.0DT** ✗, **3.0** ✗, **3.1** ✗, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Mobile Dial Delay**
  Sets and locks the dial delay applied to calls eligible for mobile twinning.
- **Hunt group calls eligible for mobile twinning**
  Sets whether mobile twinning is applied to hunt group calls.
- **Forwarded calls eligible for mobile twinning**
  Sets whether mobile twinning is applied to forwarded calls.

# User Rights | User Rights Membership

The tabs display the users associated with the user rights. and allows these to be changed.

| User Rights \| User Rights Membership | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✗, **3.0DT** ✗, **3.0** ✗, **3.1** ✗, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Members of this User Rights**
  This tab indicates those users associated with the user rights. If the user has an associated **Working hours time profile**, their association to the user rights applies only during the periods defined by the time profile. If the user does not have an associated **Working hours time profile**, they are associated with the user rights at all times.

- **Members when out of service**
  This tab indicates those users associated with the user rights outside the time periods defined by their **Working hours time profile**. The **Members when out of service** tab is not populated unless there are time profiles available within the configuration.

# Auto Attendant Settings

## Auto Attendant Overview

The IP Office Small Office Edition, IP406 V2 and IP500 control units support embedded voicemail. This is setup by adding an Avaya embedded voicemail memory card to the control unit and then selecting ***Embedded Voicemail*** as the **Voicemail Type** on the **System | Voicemail** tab. For full details refer to the Embedded Voicemail Installation Manual.

The IP406 V2 and IP500 support up to 4 simultaneous calls to embedded voicemail services. The Small Office Edition supports up to 10 simultaneous calls to embedded voicemail depending on available voice compression channels. A call from an IP device to voicemail uses two voice compression channels on the Small Office Edition.

- For pre-IP Office 4.1 systems up to 4 auto-attendant services are supported.
- For IP Office 4.1 and higher systems up to 40 auto-attendant services are supported. Due to this increase in the number of possible auto attendant, the method of automatic short code creation for recording prompts has been amended.

In addition to basic mailbox functionality, embedded voicemail can also provide auto-attendant operation. Each auto attendant can use existing time profiles to select the greeting given to callers and then provide follow on actions relating to the key presses 0 to 9, * and #.

- **Time Profiles**
  Each auto attendant can use up to three existing time profiles, on each for Morning, Afternoon and Evening. These are used to decide which greeting is played to callers. They do not change the actions selectable by callers within the auto attendant. If the time profiles overlap or create gaps, then the order of precedence used is morning, afternoon, evening.
- **Greetings**
  Four different greetings are used for each auto attendant. One for each time profile period. This is then always followed by the greeting for the auto-attendant actions. By default a number of system short codes are automatically created to allow the recording of these greetings from an IP Office extension. See below.
- **Actions**
  Separate actions can be defined for the DTMF keys 0 to 9, * and #. Actions include transfer to a specified destination, transfer to another auto-attendant transfer to a user extension specified by the caller (dial by number) and replaying the greetings.
  - IP Office 4.0+ supports a **Fax** action. This can be used to reroute fax calls when fax tone is detected by the auto-attendant.
- **Short Codes**
  Adding an auto attendant automatically adds a number of system short codes. These use the **Auto Attend** short code feature. These short codes are used to provide dialing access to record the auto attendant greetings.
  - For pre-IP Office 4.1 system, 4 short codes are added for each auto attendant. These use the form AA:Name.1 where Name is the Auto Attendant name.
  - For IP Office 4.1+ these take the form short code ***81XX*** with the number ***"AA:.1"*** where ***N*** is the replaced with the auto attendant number when dialing. Four such short codes (***81XX***, ***82XX***, ***83XX*** and ***84XX***) are added, one each for the morning, afternoon, evening and menu options greetings.
- **Routing Calls to the Auto Attendant**
  The telephone number format **AA:*Name*** can be used to route callers to an auto attendant. It can be used in the destination field of incoming call routes and telephone number field of short codes set to the **Auto Attend** feature.

# Auto Attendant | Auto Attendant

This tab is used to define the name of the auto attendant service and the time profiles that should control which auto attendant greetings are played.

| Auto Attendant \| Auto Attendant | |
|---|---|
| **Control Unit** | **SOE ✓, IP403 ✗, IP406 V1 ✗, IP406 V2 ✓, IP412 ✗, IP500 ✓.** |
| **Software Level** | **2.1 ✓, 3.0DT ✓, 3.0 ✓, 3.1 ✓, 3.2 ✓, 4.0 ✓, 4.1 ✓.** |
| **Mergeable** | **Pre-3.2 ✗, 3.2+ ✓.** |

- **Name:** *Range = Up to 12 characters*
  This field sets the name for the auto-attendant service. External calls can be routed to the auto attendant by entering **AA:*Name*** in the destination field of an Incoming Call Route.
- **Maximum Inactivity:** *Default = 8 seconds; Range = 1 to 20 seconds, Software level = 3.0+.*
  This field sets how long after playing the prompts the Auto Attendant should wait for a valid key press. If exceeded, the caller is either transferred to the Fallback Extension set within the Incoming Call Route used for their call or else the caller is disconnected.
- **Enable Local Recording:** *Default = On, Software level = 4.0+ (Q2 2007 Maintenance release)*
  When off, use of short codes to record auto-attendant prompts is blocked. The short codes can still be used to playback the greetings.
- **AA Number:** *Software level = 4.1+.*
  This number is assigned by the IP Office and cannot be changed. It is used in conjunction with short codes to access the auto attendant service or to record auto attendant greetings.
- **Morning/Afternoon/Evening/Menu Options:**
  Each auto-attendant can consist of three distinct time periods, defined by associated time profiles. A greeting can be recorded for each period. The appropriate greeting is played to callers and followed by the Menu Options greeting which should list the available actions.
  - **Time Profile**
    The time profile that defines each period of auto-attendant operation. When there are overlaps or gaps between time profiles, precedence is given in the order morning, afternoon and then evening.
  - **Short code**
    These fields indicate the system short codes automatically created to allow recording of the time profile greetings and the menu options prompt.
  - **Recording Name:** *Default = Blank. Range = Up to 31 characters, Software level = 4.0+ (Q2 2007 Maintenance release) .*
    This field appears next to the short code used for recording auto-attendant prompts.
    - For IP Office 4.0 this field is only useable for IP Offices being managed through the Avaya Integrated Management (AIM) application. As part of the Auto Attendant template this field defines the name of the file that should sent to the IP Office system when the template is merged with that IP Office system's configuration settings. The .wav file is automatically converted to the required format for the control unit before being transferred.
    - For IP Office 4.1+ this field can be used with all systems supporting embedded voicemail. The utility for converting .wav files to the correct format is provided with Manager and can be launched via File | Advanced | LVM Greeting Utility. Files then need to be manually transferred to the embedded voicemail memory card. For full details refer to the IP Office Embedded Voicemail Installation manual.

# Auto Attendant | Actions

This tab defines the actions available to callers dependant on which DTMF key they press. To change an action, select the appropriate row and click **Edit**. When the key is configured as required click **OK**.

| Auto Attendant \| Actions | |
|---|---|
| Control Unit | SOE ✓, IP403 ✗, IP406 V1 ✗, IP406 V2 ✓, IP412 ✗, IP500 ✓. |
| Software Level | 2.1 ✓, 3.0DT ✓, 3.0 ✓, 3.1 ✓, 3.2 ✓, 4.0 ✓, 4.1 ✓. |
| Mergeable | Pre-3.2 ✗, 3.2+ ✓. |

- **Key**
  The standard telephone dial pad keys, 0 to 9 plus * and #.
  - For IP Office 4.0+ the option Fax is available. This can be set to a Blind Transfer to the required fax destination and will then be triggered by fax tone detection. If left as **Not Defined**, fax calls will follow the incoming call routes fallback settings once the auto-attendant **Maximum Inactivity Time** set on the **Auto Attendant | Auto Attendant** tab is reached.
- **Action**
  The following actions can be assigned to each key.
  - **Not Defined**
    The corresponding key takes no action.
  - **Blind Transfer:** *Software level = 4.0+.*
    Transfer the call to the selected destination. This is an unsupervised transfer, if the caller is not answered they will be handled as per a direct call to that number.
  - **Dial By Number:** *Software level = 4.0+.*
    This option allows callers with DTMF phones to dial the extension number of the user they require. No destination is set for this option. The prompt for using this option should be included in the auto attendant Menu Options greeting. A uniform length of extension number is required for all users and hunt group numbers.
  - **Transfer to Attendant:** *Software level = 4.0+.*
    This action can be used to transfer calls to another existing auto attendant.
  - **Transfer to Operator:** *Software level = Up to 3.2 only*
    Transfer the caller to the selected destination. Operates the same as the Normal Transfer option below.
  - **Normal Transfer:**
    Transfer the caller to the selected destination. This is an unsupervised transfer, if the caller is not answered they will be handled as per a direct call to that number. If no destination is set, the caller can dial the user extension number that they require.
  - **Replay Greeting**
    Replay the auto-attendant greetings again.
- **Destination**
  Sets the destination extension or group number. Note that if the destination field is left blank, callers can dial the user extension number that they require. Note however that no prompt is provided for this option so it should be included in the auto attendant Menu Options greeting.

# Authorization Codes Settings

## Overview of Authorization Codes

Authorization codes are not shown by default. Manager must be modified in order to support authorization codes. Similarly in order to record authorzation codes used with calls in the IP Office SMDR, the UP Office Delta Server software must be modified.

Each authorization code is assocaited with a particular user or user rights set. The user or users associated with the user rights, can then dial numbers which are set to trigger forces authorization code entry. Once a code is entered, the short code settings of the user or user rights with which the code is assocaited are used to complete the call.

This can be used to allow authorized user to make otherwise restricted calls from any extension without first havinf to log on to that extension and then log off ater the call. Authorization code usage can be reorded with the call details by the IP Office Delta Server in ots SMDR output, including valid/invalid code entry and the code used.

- **Example**
  The system short code for dialinginternational calls is set to require a forced authorization code. This bars most users from making international calls. An authorization code is created and assocaited with User A. User A also has a short code that allows international dialing with no restriction. They can now go to any extension, start dialing an international number and when authorization code entry is triggered, enter their code to complet the call using their international dialing short code.

- ⚠ **Warning: Changing PC Registry Settings**
  Avaya accepts no liability for any issues arising from the editing of a PC's registry settings. If you are in any doubt about how to perform this process you should not proceed. It is your responsibility to ensure that the registry is correctly backed up before any changes are made.

**Enabling Authorization Codes in Manager**.
To enable support for authorization codes within Manager requires a change to the Manager PC registrysettings. Once this change is made, various authorization codes related to features are visible when Manager is restarted and a configuration from a 3.2 or higher system is loaded.

1. Close Manager.
2. Locate the registry key ***HKEY_CURRENT_USERS\Software\Avaya\IP400\Manager\EnableAuthorisationCodes*** and change its value from **0** to **1**.
3. Restart Manager and load a configuration from an IP Office 3.2 or higher system.

### Enabling Authorization Codes in Delta Server

The use of authorization codes can be included in the SMDR output logged by the IP Office Delta Server application. Again this requires changes to the registry of the PC running the Delta Server application.

1. Open the registry and locate the ***HKEY_LOCAL_MACHINES\Software\Avaya\CCCServer\Setup*** registry keys.
2. Add two new ***DWORD*** registry keys and set their values to **1**. They are:
   - ***AllowAuthorization***.
   - ***ShowAllowAuthorization***.
3. Open the browser to the Delta Server configuration screens.
4. Select SMDR. An **Add Authorization Fields to SMDR** option should now be available. Select this to enable logging of authorization codes to the SMDR log file.

Authorization codes are only logged to the SMDR log file. The two new fields are added to the end of each call log record in the SMDR log file. The first new field is the authorization code used or ***n/a*** if no authorization code was used. The second field is ***1*** for valid authorization or ***0*** for invalid authorization.

### Forcing Authorization Codes

There are two methods to force a user to enter an authorization code in order to complete dialing an external call.

- **To Force Authorization Codes on All External Calls**
  A user can be required to enter an authorization code for all external call. This is done by selecting **Force Authorization Code** on the **User | Telephony** tab.
- **To Force Authorization Codes on Specific Calls**
  To require entry of an authorization code on a particular call or call type, the **Force Authorization Code** option should be selected in the short code settings. This can be used in user, user rights or system short codes in order to apply its effect to a user, group of users or all users respectively. You need to ensure that the user cannot dial the same number by any other methos that would by pass the shortcode, for example with a different prefix.

### Entering an Authorization Code

Where possible, when an authorization code is required, the user can enter it through their phones display, However this is not possible for all types of phone, for example it is not possible with analog phones and Avaya XX01 or XX02 phones. The users of these devices must either enter the suthorization code using Phone Manager or by using a shortcode set to the **Set Authorization Code** feature immediately before making the call.

When entry of an authorization code is triggered, the user can enter any authorization code with which they are either directly assocaited or associated through their current user rights.

Note

1. If account code entry is setup for a particular number, calls forwarded or transferred to that number will also trigger account code entry.
2. On systems using line appearances to BRI trunk channels to make outgoing calls, account code entry may not be triggered. This can be resolved by adding a short code such as **[9]XN;/Dial/XN0** (adjust the prefix and line group number as necessary).

| Authorization Code \| Authorization Code | |
|---|---|
| **Control Unit** | **SOE** ✓, **IP403** ✓, **IP406 V1** ✓, **IP406 V2** ✓, **IP412** ✓, **IP500** ✓. |
| **Software Level** | **2.1** ✗, **3.0DT** ✗, **3.0** ✗, **3.1** ✗, **3.2** ✓, **4.0** ✓, **4.1** ✓. |
| **Mergeable** | ✓. |

- **Authorization Code**: *Range = Up to 12 digits.*
  The digits used for the authorization code. Each code must be unique. Wildcards are not useable with authorization codes.
  - **User Rights**
    This field is used to select the user right with which the authorization code is assocaited. The authorization code can then be used to authoriza calls made by users currently assocaited with that set of user rights.
  - **User**
    This field is used to select a user with which the authorization code is associated. The authorization code can then be used to authoriza calls made by that user.

Error! Not a valid filename.

**Error! Not a valid filename.**

Error! Not a valid filename.

# Appendix: CDR Records

## CDR Record Formats

There are a number of formats available for CDR output. Each format consist of two types of records; date records and call detail records.

- **Date Records**
  A date record is sent each time a CDR connection is started and then once a day (at midnight). The date can be in month/day or day/month format, as selected on the **System | CDR** tab.
- **Call Detail Records**
  Call detail records are sent at the termination of a call. For some formats, additional fields can be selected using the ***Normal***, ***Enhanced***, or ***ISDN*** options on the **System | CDR** tab.

| CDR Record Formats | | | |
|---|---|---|---|
| **Record Format** | **Record Options** | | |
| | **Normal** | **Enhanced** | **ISDN** |
| **Printer** | ✓ | ✓ | ✓ |
| **59-Character** | ✓ | ✗ | ✗ |
| **Expanded** | ✓ | ✓ | ✗ |
| **LSU** | ✓ | ✓ | ✓ |
| **LSU Expanded** | ✓ | ✗ | ✗ |
| **INT Direct** | ✓ | ✗ | ✗ |
| **INT ISDN** | ✓ | ✗ | ✗ |
| **INT Process** | ✓ | ✗ | ✗ |
| **Teleseer** | ✓ | ✓ | ✓ |
| **Unformatted** | ✓ | ✓ | ✗ |

# CDR Record Fields

The following list describes the fields which, depending on the selected report format and options, may be included in the CDR records.

Those fields shown in *italics* are not supported by IP Office CDR. Where the report format includes such a field, the data is replaced by a space or spaces. Similarly fields not appropriate to the call type are replaced by a space or spaces as appropriate.

- **Access Code Dialed**
  The access code the user dialed to place an outgoing call. On IP Office this will be the digit used to trigger secondary dial tone if used.
- **Access Code Used**
  The number of the line used for an outgoing call.
- **Account Code**
  This field may contain a number to associate call information with projects or account numbers. For some formats, a long account code overwrites spaces on the record that are assigned to other fields.
- ***Attendant Console*** *- Not supported by IP Office CDR.*
- ***Authorization Code -*** *Not supported by IP Office CDR.*
- ***Bandwidth -*** *Not supported by IP Office CDR.*
- **BCC (Bearer Capability Class)**
  This field identifies the type of ISDN call. Any one of the following may appear in this field.
  - 0 = Voice Grade Data and Voice.
  - 1 = Mode 1 (56 Kbps synchronous data).
  - 2 = Mode 2 (less than 19.2 Kbps synchronous or asynchronous data).
  - 3 = Mode 3 (64 Kbps data for LDAP protocol).
  - 4 = Mode 0 (64 Kbps data clear).
- **Calling Number**
  For outgoing or intra-switch calls, this field contains the extension number of the originating telephone user. For incoming and tandem calls, this field contains the trunk access code in standard formats. The fifth digit is the first digit of a 5-digit dialing plan. In formats where the field is less than 7 digits, this also shows the trunk access code of the incoming call.

  This field shows the calling party number in Unformatted or Expanded records. If the calling party number is not available, this field is blank for both formats.
- **Calling Number/Incoming Trunk Access Code**
  For incoming calls this field contains the incoming trunk access code. For outgoing calls, this field contains the calling extension.
- **Carriage Return**
  The ASCII carriage return character followed by a line feed indicates the end of a call record.

- **Condition Code**
  The condition code indicates what type of call the record describes. For example, condition code C indicates a conference call, 7 indicates an ARS call, etc. The table below shows condition codes for most record formats. The 59-character format uses different condition codes from those used for other record types.

| Code | 59 | Description |
|---|---|---|
| 0 | - | Identifies an outgoing intra-switch call (a call that originates and terminates on the switch). |
| 9 | I | Identifies an incoming external call. |
| A | - | Identifies an outgoing external call. |
| C | L | Identifies a conference call. |
| E | N | An incomplete external call, due to all trunks being busy or out of service. Incoming trunk calls to a busy terminal do *not* generate a CDR record. |
| G | - | Indicates a call terminating to a ringing station. |
| H | - | Indicates that a ringing call that was then  abandoned. |
| I | - | Indicates a call attempt to a busy station. |

CDR can also record the ring time to answer or abandon for incoming calls originated by the trunk group. In addition, CDR indicates if the incoming destination is busy. This record is separate from the normal call duration record printed for an answered call. This information is indicated by the condition code.

When an incoming call is terminated to an internal destination, the call is tracked from the time ringing feedback is given to the originator. If the call is answered, a CDR record is printed with the condition code **G** and the duration reflects the time between the start of ringing and the answer of the call. If the call is abandoned before being answered, the system prints a record with the condition code **H** and the duration reflects the time between the start of ringing and the time the call was abandoned. If the destination is busy, a CDR record is printed with the condition code **I** and a duration of 0.

- **Dialed Number**
  This field contains the number dialed. If it is an outgoing call, the field contains the number dialed by a system user. If it is an incoming call, the field contains the extension that was dialed. If more than 18 digits are dialed, the least significant digits (starting from the right) are truncated.
- **Duration**
  This is the duration of the call or call segment. It is recorded in hours, minutes and tenths of minutes. Calls are rounded down in 6-second increments. Therefore, a call of 5-second duration will be indicated as 0 duration. If 9999 appears in this field, this call was in progress when a time change was made in the switch.
- **Feature Flag**
  1 for a data call, 0 for voice calls.
- ***Format Code -*** *Not supported by IP Office CDR.*
- ***FRL -*** *Not supported by IP Office CDR.*
- **Incoming Circuit Id.**
  This field identifies the trunk used for an incoming call. For outgoing calls this field is blank.
- **Incoming Trunk Access Code**
  This field contains the access code of the incoming trunk group.
- ***ISDN Network Service*** *- Not supported by IP Office CDR.*
- ***ISDN CC*** *- Not supported by IP Office CDR.*
- ***IXC (Interexchange Carrier Code) -*** *Not supported by IP Office CDR.*

- **Line Feed**
  The ASCII line feed character follows a carriage return to terminate CDR records.
- ***MA-UUI (Message Associated User-to-User Signaling) -*** *Not supported by IP Office CDR.*
- ***Node Number*** *- Not supported by IP Office CDR.*
- **Null**
  Used to terminate and divide CDR Records (usually in triplets) when needed.
- **Outgoing Circuit Id.**
  For outgoing calls, this field identifies the trunk used.
- ***Packet Count*** *- Not supported by IP Office CDR.*
- ***PPM (Periodic Pulse Metering) -*** *Not supported by IP Office CDR.*
- ***Resource Flag*** *- Not supported by IP Office CDR.*
- **Space**
  The ASCII space character separates other CDR fields or fills unused record locations.
- ***TSC Flag*** *- Not supported by IP Office CDR.*
- **Time**
  This fields contains the time that the call ended, or the time that a user dropped from a multi-party call.

# Call Splitting

Call splitting keeps track of calls where more than two parties are involved. These can be calls that are transferred or conferenced. When any of these situations arise, CDR produces a separate record for each new party involved in the call.

### Conference

Caller **A** makes an incoming trunk call to switch party **B** (201). They talk for 2 minutes, then **B** conferences in **C** (202), and **D** (203). The entire group talks for another 8 minutes, at which point B drops off the call. This produces a record for segment **A–B**.

**A**, **C** and **D** continue to talk for another 5 minutes. All remaining parties drop, producing two more records; **A–C** and **A–D**. Note that each record shows the incoming trunk ID as the calling number.

| Segment | Duration | Condition Code | Calling Number | Dialed Number |
|---|---|---|---|---|
| **A-B** | 0:10:0 | C | 123 | 201 |
| **A-C** | 0:13:0 | C | 123 | 202 |
| **A-D** | 0:13:0 | C | 123 | 203 |

### Transfer

**A** calls **B** (201). They talk for 1 minute, then **B** transfers the call to **C** (202). CDR generates a record for segment **A–B**. **A** and **C** talk for 5 minutes. CDR then generates a record for segment **A–C**.

| Segment | Duration | Condition Code | Calling Number | Dialed Number |
|---|---|---|---|---|
| A-B | 0:01:0 | 9 | 123 | 201 |
| A-C | 0:05:0 | 9 | 123 | 202 |

### Trunk to Trunk Transfer

A calls switch party **B** (201), they talk for one minute. **B** transfers the call to public-network party **E** (5665555), they talk for 4 minutes. Note that the duration of the original incoming trunk call includes the time after the call was transferred to an outgoing trunk, until all trunk parties drop.

| Segment | Duration | Condition Code | Access Code Used | Calling Number | Dialed Number |
|---|---|---|---|---|---|
| A-B | 0:05:0 | 9 | | 123 | 201 |
| A-E | 0:04:0 | 9 | 345 | 123 | 5665555 |

# Record Formats

## 59 Character (Normal) CDR Record Formats

Those fields shown in *italics* are not supported by IP Office CDR and are replaced by a space or spaces.

| Date Record | |
|---|---|
| **Position** | **Field Description** |
| 1-2 | Month * |
| 3-4 | Day * |
| 5 | Carriage return |
| 6 | Line feed |
| 7-9 | Null |

*Leading 0 added if needed.

| Call Details Record | |
|---|---|
| **Position** | **Field Description** |
| 1-2 | Time of day-hours |
| 3-4 | Time of day-minutes |
| 5 | Duration-hours |
| 6-7 | Duration-minutes |
| 8 | Duration-tenths of minutes |
| 9 | Condition code |
| 10-12 | Access code dialed |
| 13-15 | Access code used |
| 16-30 | Dialed number |
| 31-35 | Calling number |
| 36-50 | Account code |
| 51 | *FRL* |
| 52 | *IXC* |
| 53-55 | Incoming circuit ID |
| 56-58 | Outgoing circuit ID |
| 59 | Carriage return |
| 60 | Line feed |
| 61-63 | Null |

# Expanded (Normal) CDR Record Formats

Those fields shown in *italics* are not supported by IP Office CDR and are replaced by a space or spaces. Those positions not listed are spaces.

| Date Record | |
|---|---|
| **Position** | **Field Description** |
| 1-2 | Month * |
| 4-5 | Day * |
| 6 | Carriage return |
| 7 | Line feed |
| 8-10 | Null |

*Leading 0 added if needed.

| Call Details Record | |
|---|---|
| **Position** | **Field Description** |
| 1-2 | Time of day-hours |
| 3-4 | Time of day-minutes |
| 6 | Duration-hours |
| 7-8 | Duration-minutes |
| 9 | Duration-tenths of minute |
| 11 | Condition code |
| 13-16 | Access code dialed |
| 18-21 | Access code used |
| 23-37 | Dialed number |
| 39-48 | Calling number |
| 50-64 | Account code |
| 66-72 | *Authorization code* |
| 77 | *FRL* |
| 79-81 | Incoming circuit ID |
| 83-85 | Outgoing circuit ID |
| 87 | Feature flag |
| 89-90 | *Attendant console* |
| 92-95 | Incoming trunk access code |
| 97-98 | *Node number* |
| 100-102 | *ISDN NSV* |
| 104-106 | *IXC* |
| 108 | Bearer Capability Class |
| 110 | *MA-UUI* |
| 112 | *Resource flag* |
| 114-117 | *Packet count* |
| 119 | *TSC flag* |
| 121-129 | Reserved |
| 131 | Carriage return |
| 132 | Line feed |
| 133-135 | Null |

# Expanded (Enhanced) CDR Record Formats

Those fields shown in *italics* are not supported by IP Office CDR and are replaced by a space or spaces. Those positions not listed are spaces.

| Date Record | |
|---|---|
| **Position** | **Field Description** |
| 1-2 | Month * |
| 4-5 | Day * |
| 6 | Carriage return |
| 7 | Line feed |
| 8-10 | Null |

*Leading 0 added if needed.

| Call Details Record | |
|---|---|
| **Position** | **Field Description** |
| 1-2 | Time of day-hours |
| 3-4 | Time of day-minutes |
| 6 | Duration-hours |
| 7-8 | Duration-minutes |
| 9 | Duration-tenths of minutes |
| 11 | Condition code |
| 13-16 | Access code dialed |
| 18-21 | Access code used |
| 23-37 | Dialed number |
| 39-48 | Calling number |
| 50-64 | Account code |
| 66-72 | *Authorization code* |
| 74-75 | *Time in queue* |
| 77 | *FRL* |
| 79-81 | Incoming circuit ID |
| 83-85 | Outgoing circuit ID |
| 87 | Feature flag |
| 89-90 | *Attendant console* |
| 92-95 | Incoming TAC |
| 97-98 | *Node number* |
| 100-102 | *ISDN NSV* |
| 104-107 | *IXC* |
| 109 | Bearer Capability Class |
| 111 | *MA-UUI* |
| 113 | *Resource flag* |
| 115-118 | *Packet count* |
| 120 | *TSC flag* |
| 122-123 | *Bandwidth* |
| 125-130 | *ISDN CC (digits 1–6)* |
| 131-135 | *ISDN CC (digits 7–11)* /PPM count (1–5) |
| 136-146 | Reserved for future use |
| 147 | Carriage return |
| 148 | Line feed |
| 149-151 | Null |

# INT-Direct (Normal) CDR Record Formats

Those fields shown in *italics* are not supported by IP Office CDR and are replaced by a space or spaces. Those positions not listed are spaces.

**Date Record**

| Position | Field Description |
|---|---|
| 1-2 | Month * |
| 3-4 | Day * |
| 5 | Carriage return |
| 6 | Line feed |
| 7-9 | Null |

*Leading 0 added if needed.

**Call Details Record**

| Position | Field Description |
|---|---|
| 1-2 | Day of month |
| 3-4 | Month |
| 5-6 | Year |
| 8-9 | Time of day-hours |
| 10-11 | Time of day-minutes |
| 13 | Duration-hours |
| 14-15 | Duration-minutes |
| 16 | Duration-tenths of minutes |
| 18 | Condition code |
| 20-22 | Access code dialed |
| 23-25 | Access code used |
| 27-44 | Dialed number used |
| 46-50 | Calling number |
| 52-66 | Account code |
| 68-72 | *PPM count* |
| 74-75 | Incoming circuit ID |
| 77-78 | Outgoing circuit ID |
| 79 | Carriage return |
| 80 | Line feed |

# INT-ISDN (Normal) CDR Record Formats

Those fields shown in *italics* are not supported by IP Office CDR and are replaced by a space or spaces. Those positions not listed are spaces.

| Date Record | |
|---|---|
| **Position** | **Field Description** |
| 1-2 | Month * |
| 3-4 | Day * |
| 5 | Carriage return |
| 6 | Line feed |
| 7-9 | Null |

*Leading 0 added if needed.

| Call Details Record | |
|---|---|
| **Position** | **Field Description** |
| 1-2 | Time of day-hours |
| 3-4 | Time of day-minutes |
| 5 | Space |
| 6 | Duration-hours |
| 7-8 | Duration-minutes |
| 9 | Duration-tenths of minutes |
| 11 | Condition code |
| 13-16 | Access code dialed |
| 18-21 | Access code used |
| 23-37 | Dialed number |
| 39-48 | Calling number |
| 50-64 | Account code |
| 66-72 | *Authorization code* |
| 74 | Line feed |
| 76 | *FRL* |
| 78 | Incoming circuit ID (hundreds) |
| 79 | Incoming circuit ID (tens) |
| 80 | Incoming circuit ID (units) |
| 82-84 | Outgoing circuit ID |
| 86 | Feature flag |
| 88-89 | *Attendant console (1st digit)* |
| 91-94 | Incoming trunk access code |
| 96-97 | Node number |
| 99-101 | *ISDN NSV* |
| 103-106 | *IXC* |
| 108 | Bearer Capability Class |
| 110 | *MA-UUI* |
| 112 | *Resource flag* |
| 114-119 | Reserved |
| 120-124 | *PPM* |
| 132 | Carriage return |
| 133 | Line feed |
| 134-136 | Null |

# INT-Process (Normal) CDR Record Formats

Those fields shown in *italics* are not supported by IP Office CDR and are replaced by a space or spaces. Those positions not listed are spaces.

| Date Record | |
|---|---|
| **Position** | **Field Description** |
| 1-2 | Month * |
| 3-4 | Day * |
| 5 | Carriage return |
| 6 | Line feed |
| 7-9 | Null |

*Leading 0 added if needed.

| Call Details Record | |
|---|---|
| **Position** | **Field Description** |
| 1-2 | Format code |
| 3-4 | Time of day-hours |
| 5-6 | Time of day-minutes |
| 7 | Duration-hours |
| 8-9 | Duration-minutes |
| 10 | Duration-tenths of minutes |
| 12 | Condition code |
| 14-16 | Access code dialed |
| 17-19 | Access code used |
| 21-38 | Dialed number (digits 1–18) |
| 39-43 | Calling number (digits 1–5) |
| 45-59 | Account code (digits 1–15) |
| 61 | *IXC* |
| 62 | *FRL* |
| 66-67 | Incoming circuit ID (digits 1–2) |
| 71-72 | Outgoing circuit ID (digits 1–2) |
| 74-78 | *PPM (digits 1–5)* |
| 79 | Carriage return |
| 80 | Line feed |
| 81-83 | Null |

# LSU (Normal) CDR Record Formats

Those fields shown in *italics* are not supported by IP Office CDR and are replaced by a space or spaces. Those positions not listed are spaces.

| Date Record | |
|---|---|
| **Position** | **Field Description** |
| 1-2 | Hour * |
| 3 | Colon (:) |
| 4-5 | Minute * |
| 6 | Blank |
| 7-8 | Month * |
| 9 | Slash (/) |
| 10-11 | Day * |
| 12 | Carriage return |
| 13 | Line feed |
| 14-16 | Null |

*Leading 0 added if needed.

| Call Details Record | |
|---|---|
| **Position** | **Field Description** |
| 1 | Duration-hours |
| 2-3 | Duration-minutes |
| 4 | Duration-tenths of minutes |
| 5 | Condition code |
| 6-8 | Access code dialed |
| 9-11 | Access code used |
| 12-26 | Dialed number |
| 27-30 | Calling number (digits 2–5) |
| 31-35 | Account code (first 5 digits) |
| 36-42 | *Authorization code*<br>or digits 6–12 of account code |
| 43-44 | Space or digits 13–14 of account code |
| 45 | *FRL* or digit 15 of account code |
| 46 | Calling number (1st digit) |
| 47-48 | Incoming circuit ID (tens, units) |
| 49 | Feature flag |
| 50-52 | Outgoing circuit ID (tens, units, hundreds) |
| 53 | Incoming circuit ID (hundreds) |
| 54 | *IXC* |
| 55 | Carriage return |
| 56 | Line feed |
| 57-59 | Null |

# LSU (Enhanced) CDR Record Formats

Those fields shown in *italics* are not supported by IP Office CDR and are replaced by a space or spaces. Those positions not listed are spaces.

**Date Record**

| Position | Field Description |
|---|---|
| 1-2 | Hour * |
| 3 | Colon (:) |
| 4-5 | Minute * |
| 6 | Blank |
| 7-8 | Month * |
| 9 | Slash (/) |
| 10-11 | Day * |
| 12 | Carriage return |
| 13 | Line feed |
| 14-16 | Null |

*Leading 0 added if needed.

**Call Details Record**

| Position | Field Description |
|---|---|
| 1 | Duration-hours |
| 2-3 | Duration-minutes |
| 4 | Duration-tenths of minutes |
| 5 | Condition code |
| 6-9 | *IXC* |
| 10-12 | Access code used |
| 13-27 | Dialed number |
| 28-31 | Calling number |
| 32-35 | Account code (digits 1–4) |
| 36-42 | *Authorization code*<br>or digits 6–12 of account code |
| 43-45 | *ISDN NSV* |
| 46 | 1st digit of a 5-digit calling number |
| 47-48 | Incoming circuit ID (tens, units) |
| 49 | Feature flag |
| 50-52 | Outgoing circuit ID (tens, units, hundreds) |
| 53 | Incoming circuit ID (hundreds) |
| 54 | *FRL* |
| 55 | Carriage return |
| 56 | Line feed |
| 57-59 | Null |

# LSU (ISDN) CDR Record Formats

Those fields shown in *italics* are not supported by IP Office CDR and are replaced by a space or spaces. Those positions not listed are spaces.

| Date Record | |
|---|---|
| **Position** | **Field Description** |
| 1-2 | Hour * |
| 3 | Colon (:) |
| 4-5 | Minute * |
| 6 | Blank |
| 7-8 | Month * |
| 9 | Slash (/) |
| 10-11 | Day * |
| 12 | Carriage return |
| 13 | Line feed |
| 14-16 | Null |

*Leading 0 added if needed.

| Call Details Record | |
|---|---|
| **Position** | **Field Description** |
| 1 | Duration-hours |
| 2-3 | Duration-minutes |
| 4 | Duration-tenths of minutes |
| 5 | Condition code |
| 6-8 | *IXC* |
| 9-11 | Access code used |
| 12-26 | Dialed number |
| 27-30 | Calling number (digits 2–5) |
| 31-35 | Account code (digits 1–5) |
| 36-42 | *Authorization code* or digits 6–12 of account code |
| 43-44 | *ISDN NSV* or digits 13–14 of account code |
| 45 | *ISDN NSV (3rd digit) or FRL*, or digit 15 of account code |
| 46 | Calling number (1st digit) |
| 47-48 | Incoming circuit ID (tens, units) |
| 49 | Feature flag |
| 50-52 | Outgoing circuit ID (tens, units, hundreds) |
| 53 | Incoming circuit ID (hundreds) |
| 54 | *FRL* |
| 55 | Carriage return |
| 56 | Line feed |
| 57-59 | Null |

# LSU-Expanded CDR Record Formats

Those fields shown in *italics* are not supported by IP Office CDR and are replaced by a space or spaces. Those positions not listed are spaces.

**Date Record**

| Pos. | Field Description |
|---|---|
| 1-2 | Hour * |
| 3 | Colon (:) |
| 4-5 | Minute * |
| 6 | Blank |
| 7-8 | Month * |
| 9 | Slash (/) |
| 10-11 | Day * |
| 12 | Carriage return |
| 13 | Line feed |
| 14-16 | Null |

*Leading 0 added if needed.

**Call Details Record**

| Pos. | Field Description |
|---|---|
| 1-2 | Time of day-hours |
| 3-4 | Time of day-minutes |
| 6 | Duration-hours |
| 7-8 | Duration-minutes |
| 9 | Duration-tenths of minutes |
| 11 | Condition code |
| 13-15 | Access code dialed |
| 16-18 | Access code used |
| 20-34 | Dialed number |
| 36-39 | Calling number |
| 41-45 | Account code |
| 47-53 | *Authorization code* |
| 58 | *FRL* |
| 60 | Calling number (1st digit) |
| 62-63 | Incoming circuit ID (tens, units) |
| 65 | Feature flag |
| 67-68 | Outgoing circuit ID (tens, units) |
| 70 | Outgoing circuit ID (hundreds) |
| 72 | Incoming circuit ID (hundreds) |
| 73 | *IXC* |
| 74 | Carriage return |
| 75 | Line feed |
| 76-78 | Null |

# Printer (Normal) CDR Record Formats

Those fields shown in *italics* are not supported by IP Office CDR and are replaced by a space or spaces. Those positions not listed are spaces.

| Date Record | |
|---|---|
| **Position** | **Field Description** |
| 1-2 | Month * |
| 4-5 | Day * |
| 6 | Carriage return |
| 7 | Line feed |
| 8-10 | Null |

*Leading 0 added if needed.

| Call Details Record | |
|---|---|
| **Position** | **Field Description** |
| 1-2 | Time of day-hours |
| 3-4 | Time of day-minutes |
| 6 | Duration-hours |
| 7-8 | Duration-minutes |
| 9 | Duration-tenths of minutes |
| 11 | Condition code |
| 13-15 | Access code dialed |
| 17-19 | Access code used |
| 21-35 | Dialed number |
| 37-41 | Calling number |
| 43-57 | Account code |
| 59-65 | *Authorization code* |
| 70 | *FRL* |
| 72 | *IXC* |
| 74-76 | Incoming circuit ID |
| 78-80 | Outgoing circuit ID |
| 82 | Feature flag |
| 83 | Carriage return |
| 84 | Line feed |

# Printer (Enhanced) CDR Record Formats

Those fields shown in *italics* are not supported by IP Office CDR and are replaced by a space or spaces. Those positions not listed are spaces.

**Date Record**

| Position | Field Description |
|---|---|
| 1-2 | Month * |
| 4-5 | Day * |
| 6 | Carriage return |
| 7 | Line feed |
| 8-10 | Null |

*Leading 0 added if needed.

**Call Details Record**

| Position | Field Description |
|---|---|
| 1-2 | Time of day-hours |
| 3-4 | Time of day-minutes |
| 6 | Duration-hours |
| 7-8 | Duration-minutes |
| 9 | Duration-tenths of minutes |
| 11 | Condition code |
| 13-16 | *IXC* |
| 18-21 | Access code used |
| 23-37 | Dialed number |
| 39-43 | Calling number |
| 45-59 | Account code |
| 61-67 | *Authorization code* |
| 69-71 | *ISDN NSV* |
| 73 | *FRL* |
| 75-77 | Incoming circuit ID |
| 79-81 | Outgoing circuit ID |
| 83 | Feature flag |
| 84 | Carriage return |
| 85 | Line feed |

## Printer (ISDN) CDR Record Formats

Those fields shown in *italics* are not supported by IP Office CDR and are replaced by a space or spaces. Those positions not listed are spaces.

| Date Record | |
|---|---|
| **Position** | **Field Description** |
| 1-2 | Month * |
| 4-5 | Day * |
| 6 | Carriage return |
| 7 | Line feed |
| 8-10 | Null |

*Leading 0 added if needed.

| Call Details Record | |
|---|---|
| **Position** | **Field Description** |
| 1-2 | Time of day-hours |
| 3-4 | Time of day-minutes |
| 6 | Duration-hours |
| 7-8 | Duration-minutes |
| 9 | Duration-tenths of minutes |
| 11 | Condition code |
| 13-15 | *IXC* |
| 17-19 | Access code used |
| 21-35 | Dialed number |
| 37-41 | Calling number |
| 43-57 | Account code |
| 59-65 | *Authorization code* |
| 67-68 | *ISDN NSV (hundreds, tens)* |
| 70 | *ISDN NSV (units)* |
| 72 | *FRL* |
| 74-76 | Incoming circuit ID |
| 78-80 | Outgoing circuit ID |
| 82 | Feature flag |
| 83 | Carriage return |
| 84 | Line feed |

# Teleseer (Normal) CDR Record Formats

Those fields shown in *italics* are not supported by IP Office CDR and are replaced by a space or spaces. Those positions not listed are spaces.

| Date Record | |
|---|---|
| **Position** | **Field Description** |
| 1-2 | Month * |
| 3-4 | Day * |
| 5 | Carriage return |
| 6 | Line feed |
| 7-9 | Null |

*Leading 0 added if needed.

| Call Details Record | |
|---|---|
| **Position** | **Field Description** |
| 1-3 | Space |
| 4-5 | Time of day-hours |
| 6-7 | Time of day-minutes |
| 8 | Duration-hours |
| 9-10 | Duration-minutes |
| 11 | Duration-tenths of minutes |
| 12 | Condition code |
| 13-15 | Access code dialed |
| 16-18 | Access code used |
| 19-33 | Dialed number |
| 34-38 | Calling number |
| 39-53 | Account code |
| 54 | *FRL* |
| 55 | *IXC* |
| 56-58 | Incoming circuit ID |
| 59-61 | Outgoing circuit ID |
| 62 | Feature flag |
| 63-69 | *Authorization code* |
| 70-76 | Space |
| 77 | Carriage return |
| 78 | Line feed |
| 79-81 | Null |

# Teleseer (Enhanced) CDR Record Formats

Those fields shown in *italics* are not supported by IP Office CDR and are replaced by a space or spaces. Those positions not listed are spaces.

| Date Record | |
|---|---|
| **Position** | **Field Description** |
| 1-2 | Month * |
| 3-4 | Day * |
| 5 | Carriage return |
| 6 | Line feed |
| 7-9 | Null |

*Leading 0 added if needed.

| Call Details Record | |
|---|---|
| **Position** | **Field Description** |
| 1-3 | Space |
| 4-5 | Time of day-hours |
| 6-7 | Time of day-minutes |
| 8 | Duration-hours |
| 9-10 | Duration-minutes |
| 11 | Duration-tenths of minutes |
| 12 | Condition code |
| 13-16 | *IXC* |
| 17-19 | Access code used |
| 20-34 | Dialed number |
| 35-39 | Calling number |
| 40-54 | Account code |
| 55 | *ISDN NSV (units)* |
| 56 | *FRL* |
| 57-59 | Incoming circuit ID |
| 60-62 | Outgoing circuit ID |
| 63 | Feature flag |
| 64-70 | *Authorization code* |
| 71-72 | *ISDN NSV (hundreds, tens)* |
| 77 | Carriage return |
| 78 | Line feed |
| 79-81 | Null |

# Teleseer (ISDN) CDR Record Formats

Those fields shown in *italics* are not supported by IP Office CDR and are replaced by a space or spaces. Those positions not listed are spaces.

| Date Record | |
|---|---|
| **Position** | **Field Description** |
| 1-2 | Month * |
| 3-4 | Day * |
| 5 | Carriage return |
| 6 | Line feed |
| 7-9 | Null |

*Leading 0 added if needed.

| Call Details Record | |
|---|---|
| **Position** | **Field Description** |
| 1-3 | Space |
| 4-5 | Time of day-hours |
| 6-7 | Time of day-minutes |
| 8 | Duration-hours |
| 9-10 | Duration-minutes |
| 11 | Duration-tenths of minutes |
| 12 | Condition code |
| 13-15 | *IXC* |
| 16-18 | Access code used |
| 19-33 | Dialed number |
| 34-38 | Calling number |
| 39-53 | Account code |
| 54 | *ISDN NSV (units)* |
| 55 | *FRL* |
| 56-58 | Incoming circuit ID |
| 59-61 | Outgoing circuit ID |
| 62 | Feature flag |
| 63-69 | *Authorization code* |
| 70-71 | *ISDN NSV (hundreds, tens)* |
| 77 | Line feed |
| 78-80 | Null |

# Unformatted (Normal) CDR Record Formats

Those fields shown in *italics* are not supported by IP Office CDR and are replaced by a space or spaces. Those positions not listed are spaces.

| Date Record | |
|---|---|
| **Pos.** | **Field Description** |
| 1-2 | Hour * |
| 3 | Colon (:) |
| 4-5 | Minute * |
| 6 | Blank |
| 7-8 | Month * |
| 9 | Slash (/) |
| 10-11 | Day * |
| 12 | Carriage return |
| 13 | Line feed |
| 14-16 | Null |

*Leading 0 added if needed.

| Call Details Record | |
|---|---|
| **Pos.** | **Field Description** |
| 1-2 | Time of day-hours |
| 3-4 | Time of day-minutes |
| 5 | Duration-hours |
| 6-7 | Duration-minutes |
| 8 | Duration-tenths of minutes |
| 9 | Condition code |
| 10-13 | Access code dialed |
| 14-17 | Access code used |
| 18-32 | Dialed number |
| 33-42 | Calling number |
| 43-57 | Account code |
| 58-64 | *Authorization code* |
| 67 | *FRL* |
| 68-70 | Incoming circuit ID |
| 71-73 | Outgoing circuit ID |
| 74 | Feature flag |
| 75-76 | *Attendant console* |
| 77-80 | Incoming TAC |
| 81-82 | Node number |
| 83-85 | *ISDN NSV* |
| 86-88 | *IXC* |
| 89 | Bearer Capability Class |
| 90 | *MA-UUI* |
| 91 | *Resource flag* |
| 92-95 | *Packet count* |
| 96 | *TSC flag* |
| 97-100 | Reserved |
| 101 | Carriage return |
| 102 | Line feed |
| 103-105 | Null |

# Unformatted (Enhanced) CDR Record Formats

Those fields shown in *italics* are not supported by IP Office CDR and are replaced by a space or spaces. Those positions not listed are spaces.

| Date Record | |
|---|---|
| **Position** | **Field Description** |
| 1-2 | Hour * |
| 3 | Colon (:) |
| 4-5 | Minute * |
| 6 | Blank |
| 7-8 | Month * |
| 9 | Slash (/) |
| 10-11 | Day * |
| 12 | Carriage return |
| 13 | Line feed |
| 14-16 | Null |

*Leading 0 added if needed.

| Call Details Record | |
|---|---|
| **Position** | **Field Description** |
| 1-2 | Time of day-hours |
| 3-4 | Time of day-minutes |
| 5 | Duration-hours |
| 6-7 | Duration-minutes |
| 8 | Duration-tenths of minutes |
| 9 | Condition code |
| 10-13 | Access code dialed |
| 14-17 | Access code used |
| 18-32 | Dialed number |
| 33-42 | Calling number |
| 43-57 | Account code |
| 58-64 | *Authorization code* |
| 65-66 | *Time in queue* |
| 67 | *FRL* |
| 68-70 | Incoming circuit ID |
| 71-73 | Outgoing circuit ID |
| 74 | Feature flag |
| 75-76 | *Attendant console number* |
| 77-80 | Incoming TAC |
| 81-82 | *Node number* |
| 83-87 | *ISDN NSV* |
| 88-89 | *IXC* |
| 90 | Bearer Capability Class |
| 91 | *MA-UUI* |
| 92 | *Resource flag* |
| 93-96 | *Packet count* |
| 97 | *TSC flag* |
| 98-99 | *Bandwidth* |
| 100-105 | *ISDN CC (digits 1–6)* |
| 106-110 | *ISDN CC (digits 7–11)/PPM* |
| 111-114 | Reserved for future use |
| 115 | Carriage return |
| 116 | Line feed |
| 117-119 | Null |

# Index

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



Avaya
Unit 1, Sterling Court
15 - 21 Mundells
Welwyn Garden City
Hertfordshire
AL7 1LZ
England

Tel: +44 (0) 1707 392200
Fax: +44 (0) 1707 376933

Web: http://www.avaya.com/ipoffice/knowledgebase