IP Office Technical Bulletin

Bulletin No: 127

Date: November 2010

Title: General Availability (GA) of IP Office Release 6.1

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Date: 29th November 2010

Region: GLOBAL

General Availability (GA) of IP Office Release 6.1 Software

Avaya is delighted to announce the launch and availability of IP Office Release 6.1. This version introduces additional telephones and telephone features, SCN Management enhancements, SIP and other core features, as well as many IP Office applications improvements.

1. Product Overview

Avaya IP Office Release 6.1 is the latest advancement in converged voice and data technology from Avaya. IP Office combines high-end voice and data applications, allowing the smallest of businesses to deliver cutting edge customer service.

IP Office Release 6.1 will be supported on the following control units:

- IP406v2 (64Mb PCS 8 and later), IP412, IP500, IP500v2

Licensing In order to upgrade to Release 6.1 from a previous release, an Upgrade License must be purchased. There are two variants:

- Upgrade to 6.1 for sites with over 32 users or when expansion modules are used
- Upgrade to 6.1 for small sites up to 32 users where no external expansion modules are used (applies *only* to IP500v1 and IP500v2; not available for IP406v2 or IP412)

For new installations, no upgrade license is required on the IP500v2 platform to run IP Office 6.1. Also no upgrade license is required on Essential Edition - PARTNER® Version systems or Essential Edition – Norstar™ Version systems.

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IP Office Release 6.1 will NOT be supported on the following control units:

- IP401, IP403, IP406 (v1), IP406v2 (16Mb PCS7 or earlier), Small Office Edition

Note: *The WAN3 and WAN3 10/100 expansion modules were supported on the IP406v2 and IP412 platforms in Release 5. In Release 6.0 & 6.1 this support has been removed.*

********** **Application Support** **********

Please note that the following applications are not supported with IP Office Release 6.1 and will not connect to the system.

Compact Contact Center (CCC)
Delta Server
Call Status
Conferencing Center
Feature Key Server*

Note: *The IP Office Feature Key Server application is not supported for systems running IP Office Release 6.0 & 6.1, meaning that the use of parallel port and USB port Feature Key dongles attached to a server PC is not supported. Therefore IP406v2 and IP412 systems being upgraded to IP Office Release 6.1 must also be upgraded to use a serial port feature key. Please speak with your Distributor about swapping licenses to a new feature key dongle.*

IP Office Release 6.1 is the entry-level software required to support the following new hardware:

- AVAYA 1000 Series Video/SIP Devices 1010 & 1040 *Note (1)*
- Grandstream GXV3140 IP Multimedia Phone
- Polycom VVX1500
- 1100/1200 Series Nortel SIP Phones *Note (2)*
- Essential Edition Norstar SD Card

1) *IP Office IP500v1/IP500v2 will now support SIP HD Video over SIP trunks – the Avaya 1000 series will be supported up to the basic feature set of making a call and receiving a call including Video*

2) *The following phones from the Avaya Nortel range of SIP phones are supported: 1120E, 1140E, 1220 and 1230 on IP500v1/IP500v2*

IP Office Release 6.1 also supports the following new features:

IP Office Core Software: Version 6.1

- IP Office Essential Edition – PARTNER Version new features
- IP Office Essential Edition – Norstar Version for MENA Regions
- Embedded Voicemail – Skip Mailbox Greeting
- Music on Hold – Multiple External Sources
- Automatic Call Log Expiry
- Mobile Twinning Handover
- Show last Call Duration
- Restrict Analogue Extension Ringer Voltage
- Time Support
- Restart IP Phones on System Upgrade
- Multiple SIP Proxies and Resilience
- SIP Interoperability with Nortel BCM & CS1000 Systems
- SIP Busy Lamp Field, Call Forward and Conf 3rd Party Add Support
- Remote Loopback PRI/TI Trunk Modules
- Simplified language supported for 1408 & 1416 telephones in CHS locale
- Call presentation enhancements in 1400 & 1600 telephones
- Call Timer on 1408 & 1416, 1608 & 1616 telephones
- Set based Admin of Feature Labels on 9600 telephones
- System Administration and Admin 2 enhancements

IP Office Manager: Version 8.1

- Small Community Network Management
- Simplified Manager for Partner Version
- Simplified Manager for Norstar Version

IP Office Preferred Edition (formerly Voicemail Pro): Version 6.1

- Additional Generic Action String Manipulation Options
- Post Call Completion Call Flows
- Automatic Call Recording for Internal Calls

one-X Portal for IP Office: Version 6.1

- Adjustable Layout
- Multi Directory search
- Call Assistant

Customer Call Reporter: Version 6.1.1.84

- Microsoft SQL 2008 Support
- Visual Redesign
- New Supervisor Pages
- Force Agent State
- New Statistics
- Report Templates
- Message Color

IP Office Application Server

- Linux CentOS 5.5 Operating System
- One DVD installation
- Incorporating one-X Portal for IP Office
- Incorporating IP Office Preferred Edition (formerly Voicemail Pro)

IP Office Softphone

- BLF (BLFs use dialog event defined in RFC 4235) and Call Pickup Support

2. IP Office Hardware

The following new hardware is supported with IP Office Release 6.1 software on IP Office only:

2.1. SIP Conference/Video Telephones

IP Office will now support SIP Video telephony over SIP Trunks. There are four current models tested and approved to a basic level of making and receiving of Video Telephony calls.

- Polycom VVX 1500
- Avaya Aura Video Telephone Models 1010 & 1040
- Grandstream GVX3140 IP Multimedia Phone

2.2. Support for 1100 & 1200 Series IP Telephones

“Nortel” SIP Telephones 1120, 1140, 1220, and 1230 are supported on IP500 & IP500v2 IP Office only
IP Office IP500v1 & IP500v2 will support our existing basic feature set on the Nortel 1220/1230 and the 1120/1140 SIP feature phones in 6.1. This is aimed at former Nortel Business Partners who are making the transition to IP Office as an alternative solution. The feature set will later be expanded in future releases.

1120E Telephone - The IP Phone 1120E is a multi-line (four lines), intermediate-level desktop SIP phone. 1120E delivers advanced communications capabilities including support for a high-resolution, pixel-based, fully-backlit, eight-level greyscale graphical display, an IEEE integrated phone switch with Gigabit Ethernet LAN and PC ports and support for a local Tools Menu delivering simpler administration.

1140 Telephone - The IP Phone 1140E is a multi-line professional-level desktop SIP phone. 1140E delivers advanced communications capabilities including support for a high-resolution, fully-backlit, pixel-based graphical monochrome eight-level greyscale display, an integrated phone switch with Gigabit Ethernet LAN and PC ports and support for a local Tools Menu for simpler administration.


1120 SIP Telephone


1140 SIP Telephone

1120 / 1140 Expansion Module - Eighteen self-programmable line/feature keys per module are provided and up to three modules can be supported with an IP Phone 1120E or IP Phone 1140E.
The current system limit of 1024 applies.


1140 with Expansion Module

1220 Telephone - The SIP Phone 1220 is an intermediate-level desktop multi-line IP Phone that supports up to four line/programmable feature keys, eight fixed telephony keys, four context-sensitive soft keys and six shortcut/feature keys. It's positioned for office workers with moderate call activity.
1230 Telephone - The SIP Phone 1230 is a premium-intermediate multi-line IP Phone that supports up to 10 line/programmable feature keys, eight fixed telephony keys, four context-sensitive soft keys and six shortcut/feature keys. It's best suited for power users with advanced communication needs such as managers, knowledge workers and administrative assistants.


1220 Telephone


1230 Telephone

1220/1230 Expansion Modules - 12-Key LCD (Liquid Crystal Display) KEM that is self-labelling which are both supported on the IP Phone 1220 and 1230. 12 key LCD KEM cascades up to 7 modules. The current system limit of 1024 applies.

1220 With Expansion Module

2.3. Summary of the Terminal's capabilities

IPOffice 6.1 will support these sets with SIP 4.0 firmware version. The sets will register as Avaya SIP terminals and will use Avaya SIP Endpoint license. The terminal firmware supports all basic SIP VoIP capabilities.

There are some differences in the terminal User Interface and supported capabilities when the terminal registers with an IPOffice as compared with CS1000.

Number of options supported via Soft Key in CS1000 deployment is not available. These are: DND, Conf, Fwd, and Redirect. All these functions will be available via IPOffice short codes only.

Speed dial programming is done via local feature programming menu on the terminal.

The following terminals SIP firmware capabilities are not supported:

- Send IM
- Presence state
- Auto-programming

Directory download from the IPOffice is not supported. Local Directory is available.

BLF buttons can be programmed via Manager only. This is the only button programming information which will be downloaded to the terminals. Call pickup function is not supported in Release 6.1. The buttons will show BUSY or IDLE state only. RINGING state will not be shown.

None of the other button programming done via the Manager will be downloaded to the sets.

2.4. 11xx and 12xx Connection to IP Office

Configuration files (firmware, languages, etc.) should only be downloaded from the IPO. Use of IPO Manager or another HTTP server to download the files is not supported. The 11xx & 12xx telephones are similar to IP Office IP Telephones by the fact that they will load a configuration file, a settings file, a firmware file, and up to 5 language files.

NOTES:
- 1120e,1140e, 1220, 1240 IP sets must be at Nortel BCM6.0 UNISTIM GA F/W level (062AC7M) or loaded with the basic Boot Loader typically when new out of the box
- 1165e, 1210, 1110 IP sets are NOT Supported
- Supported on IP500 and IP500v2 6.1 core software

The configuration & settings files are auto generated by the IP Office (similar to Avaya IP 1600 series telephones).

Example configuration files would be:

- 1120eSIP.cfg
- 11404SIP.cfg
- 1220SIP.cfg
- 1230SIP.cfg

The configuration files would tell the telephone over HTTP what firmware, language files and what settings file to use e.g. 11xxsettings.txt, 12xxsettings.txt

The firmware files vary dependant on telephone type and firmware release version, but look like: SIP12x0e04.00.0.03.bin.

The language files end with .lng and therefore look like: Spanish.lng

To configure settings on the phone – **press the “Globe” (looks like a planet on icon type IP sets) or “Services” (on English Text type IP sets) key twice**
It will ask for an Admin password: 26567*738 (Color*set)

2.4.1. 11xx & 12xx Telephone Reset Procedure

If the terminal was previously deployed as a SIP terminal with CS1K or AS500, it is advisable to rest the terminal to the factory defaults. As specified by the procedure below:

This reset procedure does not change or replace the existing firmware that the IP set used to boot with.

Here are the steps for resetting an 11xx or 12xx IP set to factory defaults.

1) Look at the back of your IP set. You will notice a white label with three lines. Look at the bottom line on which you see six consecutive pairs of hexadecimal values that are sometimes presented with ":" between each pair. This is the MAC address of the

IP set. For example a MAC address can look like "A1 B2 C3 D4 E5 F6" or "A1:B2:C3:D4:E5:F6". Write down the MAC address value.

2) You need to translate the MAC address value into a series of equivalent keypad numbers. Existing digits remain digits and letters are mapped to the keypad number on which the actual letter appears. Empty spaces are ignored in the translation. For example "A1" translates to "11". Using the MAC address in the previous step we have "A1 B2 C3 D4 E5 F6" translated to value "212223343536".

3) On the IP phone at any point in time, you can press the following sequence on your keypad:

**73639<Translated MAC>## (or **renew<Translated MAC>##)

4) You'll be presented with a question on the IP phone screen to "Reset to Factory Settings". Press the "Yes" soft key to complete the reset to factory procedure. Press the "No" soft key to quit.

2.4.2. Enable 11xx Telephone on IP Office

On the IP Office Manager check the following as with any SIP telephone connection:

- In System, uncheck “Avaya HTTP Clients only” – this can be enabled once the 11xx or 12xx phones have loaded their firmware, they will then be recognised as an Avaya phone.
- Use Auto Create User preferably or you can create one in Manager.
- Set DHCP in Server Mode – in System / LAN1/2 / DHCP Pools, uncheck “Apply to Avaya IP Phones Only”.
- Set Call Waiting on the User entry to enable Call Transfer
- System / LAN1/2 / VoIP, check “SIP Registrar enabled”
- Avaya IP Endpoints license key enabled with appropriate number of endpoints licensed.
- Optional : Connect SysMon showing the HTTP (Services Tab) events so you can monitor progress

Connect 11xx Telephone using DHCP method

- Connect the Telephone to a PoE port and it will come up with Unistim (no BootC for 11xx) regardless if it’s new or redeployed from the BCM network.
- Once the phone has booted:
  - Select **Service** key (English labelled set) or **globe** icon (icon labelled set)
  - Select item **3 Network configuration**
  - Soft keys display **Apply Auto -- Cancel**
  - Press **Auto**
  - Navigate down to item ***12 Provision Server***
  - Deselect **check box**
  - Press **Config**
  - Populate IPO’s IP address at **Provision:<0.0.0.0>**

- Press **Config**
- Press **Apply**

- IP Phone will boot and update the IP Phone with SIP FW configuration derived from the IP Office. It will read and load:
  - The Configuration Files
  - The Firmware File
  - The Language Files (up to 5)
- It then displays **Avaya** on the LCD
- The phone then resets
- It will finally ask you to enter the User Login e.g. 250
  - Press “**Login**”
  - It will ask for a password (default is 0000)
  - Press the “**Next**” key twice
- The phone should now be logged in as User 250

Connect 11xx Telephone using Non DHCP method

- Connect the Telephone to a PoE port and it will come up with Unistim (no BootC for 11xx)
- Once the phone has booted: Set manual mode on the terminal
  1. Select **Service** key (English labelled set) or globe icon (icon labelled set)
  2. Select item **3 Network configuration**
  3. Soft keys display **Apply Auto -- Cancel**
  4. Press **Auto**
  5. Right navigate: **DHCP Enable**
  6. Deselect **check box** by pressing the “**Man**” soft key
  7. Navigate down to item ***12 Provision Server***
  8. Deselect **check box** by pressing the “**Man**” soft key
  9. Press “**Cfg” soft key**
  10. DCHP y/n displayed
  11. Select DHCP = 0 (for No)

Provisioning - required settings on the terminal

- Populate: **Set IP address, NetMask** and **Gateway** values as appropriate to your network
  - (e.g. Set IP = 192.168.43.114, NetMask = 255.255.255.0, Gateway=192.168.1.1)
- Navigate down to **Provision Server**
- Populate IPO’s IP address (e.g.192.168.43.1)
- Press “**OK**”
- Displays DNS
- Press “**OK**”
- Press **Apply**

- The IP phone will boot and update the IP Phone with SIP firmware configuration derived from the IP Office. It will read and load:
  - The Configuration Files
  - The Firmware File
  - The Language Files (up to 5)
- It then displays **Avaya** on the LCD

- The phone then resets
- It will finally ask you to enter the User Login e.g. 250
- Press "**Login**"
- It will ask for a password (default is 0000)
- Press the "**Next**" key twice
- The phone should now be logged in as User 250

2.4.3. Enable 12xx Telephone on IP Office

NOTE:

- New 1220/1230 IP sets should have latest BCM6.0 level Unistim and initially loads up BootC upon Power up.
- 1220/1230 IP sets previously used in BCM will require Unistim FW upgrade to BCM6.0 level and will need to load BootC (by pressing 2 and UP key upon Power up.

Connect 12xx Telephone Using DHCP

This is similar to the 11xx, but you first need to manually configure the IPOffice/HTTP Server address.

- Connect the 12xx telephone into a PoE port.
- Set DHCP option on the terminal
- The phone will boot and ask if you want **Manual Configuration**?
- Press the 4 buttons under the LCD left to right one after the other :
    - **OK Auto BkSpc Clear**
- It will display **DHCP 0=N 1=Y**
- Select **1** for DHCP
- Use the Scroll down key to get to "**Provision Server**"
- It will display "**0.0.0.0**" – press the backspace key to clear then populate using the **IP Address of the IP Office** (e.g. 192.168.43.1)
- Press "**OK**"
- Displays DNS – Press "**OK**"
- Press "**Apply**"
- The Phone will reboot and update the IP Phone with SIP FW configuration derived from the IP Office.
- It will now go through the process to load files the same as the 11xx as described above.

Connect 12xx Telephone Using Non- DHCP

- Connect the 12xx telephone into a PoE port.
- Set Manual option on the terminal
- The phone will boot and ask if you want **Manual Configuration**?
- Press the 4 buttons under the LCD left to right one after the other :
    - **OK Auto BkSpc Clear**
- It will display **DHCP 0=N 1=Y**
- Press the "**Auto**" key – to show the list of features that enable automatically.
- Scroll down to **DHCP Enable**

- Deselect the **check box** by pressing the “**Man**” soft key
- Scroll down to get to “**12. Provision Server**”
- Deselect the **check box** by pressing the “**Man**” soft key
- Press the “**Cfg**” button
- DCHP y/n displayed
- Select DHCP = 0 (for No)

Provisioning required settings on the terminal

- Populate : **Set IP address, NetMask** and **gateway** values as appropriate to your network
- (e.g. Set IP = 192.168.43.114, NetMask = 255.255.255.0, Gateway=192.168.1.1)
- Navigate down to **Provision Server**
- Populate IPO’s IP address (e.g.192.168.43.1)
- Press “**OK**”
- Displays DNS – Press “**OK**”
- Press “**Apply”**
- The Phone will reboot and update the IP Phone with SIP FW configuration derived from the IP Office.
- It will now go through the process to load files the same as the 11xx as described above.

2.5. System SD Cards

There are now 4 types of System SD cards on 6.1:

- IPOffice µ-Law
- IPOffice A-Law
- IPOffice Partner Version (µ-Law)
- IPOffice Norstar Version (A-Law) *

** Norstar Version is supported in Middle-East and Africa regions only*

On occasion, IP Office SD cards are built at the factory with a release that may not be the current GA software. Due to delays inherent in shipping and managing inventory, one cannot be sure that when ordered, the SD card will have the GA release software on it. In addition to overlaps that occur naturally at the point of a new release, maintenance builds are performed regularly and can also result in an SD card with a non GA version of the software. Those installing or maintaining IP Office should always be sure to check online at http://support.avaya.com – “Downloads” section for the latest software build and if more recent than the one they have, download it. Downloads should always be done in advance of going to the customer location to ensure availability of a high speed Internet connection. If the installer finds that the SD card that comes with the system is out of date, it can be easily updated using the manager program, either via an SD card reader in the laptop or directly while installed in the IP500 v2 chassis.

2.6. Avaya IP Office Telephone Support

IP Office Release 6.1 will support the following telephones (pending regional availability, not all telephones available for new sales). Not supported on Essential Edition PARTNER® or Norstar systems unless otherwise stated.

- 2400/5400 series digital telephones
- 1400 series digital telephones (supported on IP Office, PARTNER® and Norstar systems)
- Selected 4600/5600 series IP telephones
- Selected 1600 series IP telephones
- Selected 1600-I series IP telephones
- 9620L/9620C/9630G/9640/9640G/9650/9650C/9608/9621/9641 IP telephones
- 4406, 4412, 4424D+ telephones (DS expansion modules only, no longer sold new)
- Selected 6400 series telephones (no longer sold new)
- T3 IP and T3 digital telephones
- 3701/3711 (IP DECT)
- 3720/3725 (DECT R4)
- 3616/3620/3626/3641/3645 (Wi-Fi, 3616/3620/3626 no longer sold new)
- 3810 (900MHz)
- 1120e, 1140e, 1220, 1230 Series SIP Telephones
- ETR6/ETR6D/ETR18/ETR18D/ETR34D/3910/3920 telephones (Essential Edition – PARTNER® Version only)
- Analog phones
- Associated DSS units (XM24, EU24, BM32, DBM32, SBM24, DSS 4450, T3 DSS, 12xx 12-Key LCD KEM, 11xx 18-Key Exp Module)

2.7. Material Codes for IP Office Release 6.1 Hardware

The following table lists the new material codes for the hardware that is being supported with IP Office Release 6.1:

Material Code	Description
700500948	IPO SYSTEM SD CARD NORSTAR

3. Essential Edition – PARTNER® new features

Supported on IP500v2 configured as an Essential Edition Partner system. This is available in North America only.

Availability

The Essential Edition - PARTNER® Version is offered and supported in all the current PARTNER® ACS R8 markets and distribution channels.

Current PARTNER® ACS R8 countries include:
- Canada
- United States
- Mexico – regions that support North American telephony (US) standards
- Countries that support North American telephony (US) standards – specifically the Caribbean, Puerto Rico and U.S. Virgin Islands.

Languages supported are US English (default), Latin American Spanish and Canadian French – the same languages as currently supported on the PARTNER® ACS.

Partner – New Features

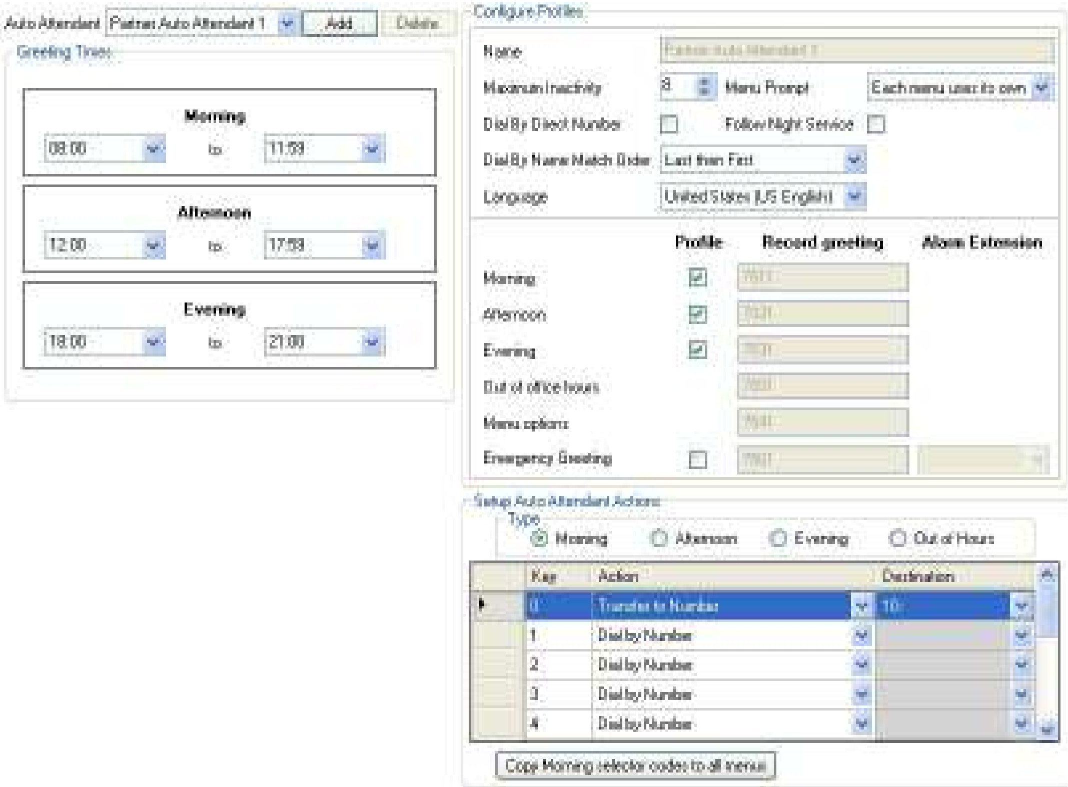
IP Office Essential Edition – R6.1 is an incremental software release built on top of IP Office Essential Edition – R6

PARTNER Version enhancements are focused in the following areas:
- Auto Attendant
- Voice Mail
- New Features
- Service and Support

3.1. Voicemail and Automated Attendant Enhancements

Multiple Auto Attendants – Presently only one automated attendant is configurable. 6.1 will provide the ability to support up to nine automated attendants which is the same as the Partner ACS. There will also be support to enable Auto Attendant Sub-Menus – the ability to link Auto Attendants and their greetings. Using Simplified manager, lines and DID's can be assigned to any one of the 9 Auto Attendants.

- The system supports up to 9 Automated Attendants (AA).
  - This is an increase from R6 which supported 1 AA.
  - AA's are identified by a number (1-9).
- Each AA has its own greeting (morning/afternoon/evening), its own set of selector codes, separate VMS Hunt and Delay Schedules, and its own unique time profiles.
- The AA can be accessed by dialing the AA dial codes (to record greetings, menu prompts, etc.)

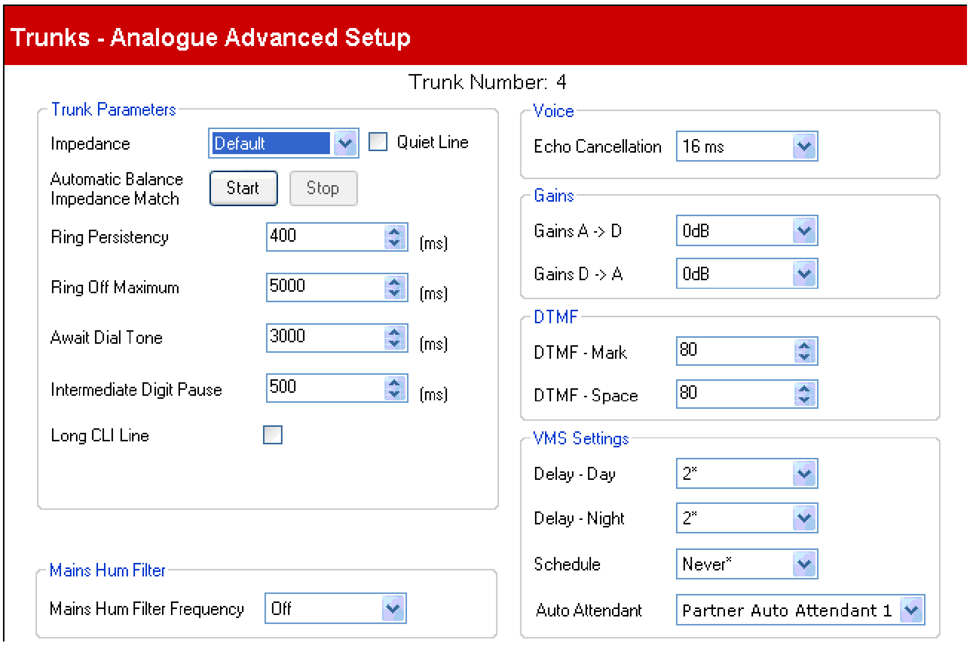


<u>**Auto Attendant Dial Selector Codes**</u>

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

3.2. Multiple Auto Attendants and Sub Menus

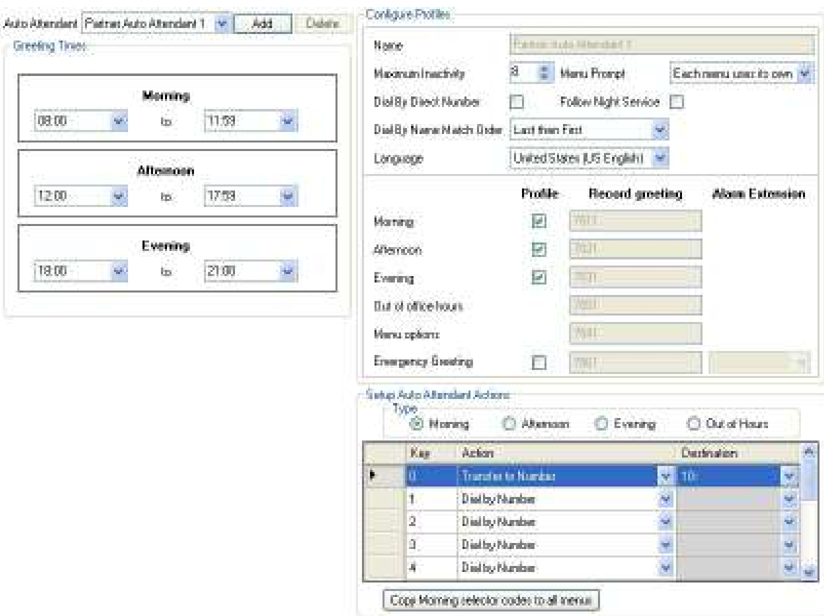
With Multiple Auto Attendants, the administrator must specify which lines are serviced by which AA. Lines can be configured to ring at any one of the 9 AA, using the System Programming Procedure, *Assign Lines to AA.* The default value for each line is AA 1.



The system also supports up to 8 sub-menus. Sub-menus can be assigned to a specific AA, or can be shared among all AA. The total number of AA and sub-menus cannot exceed 9.

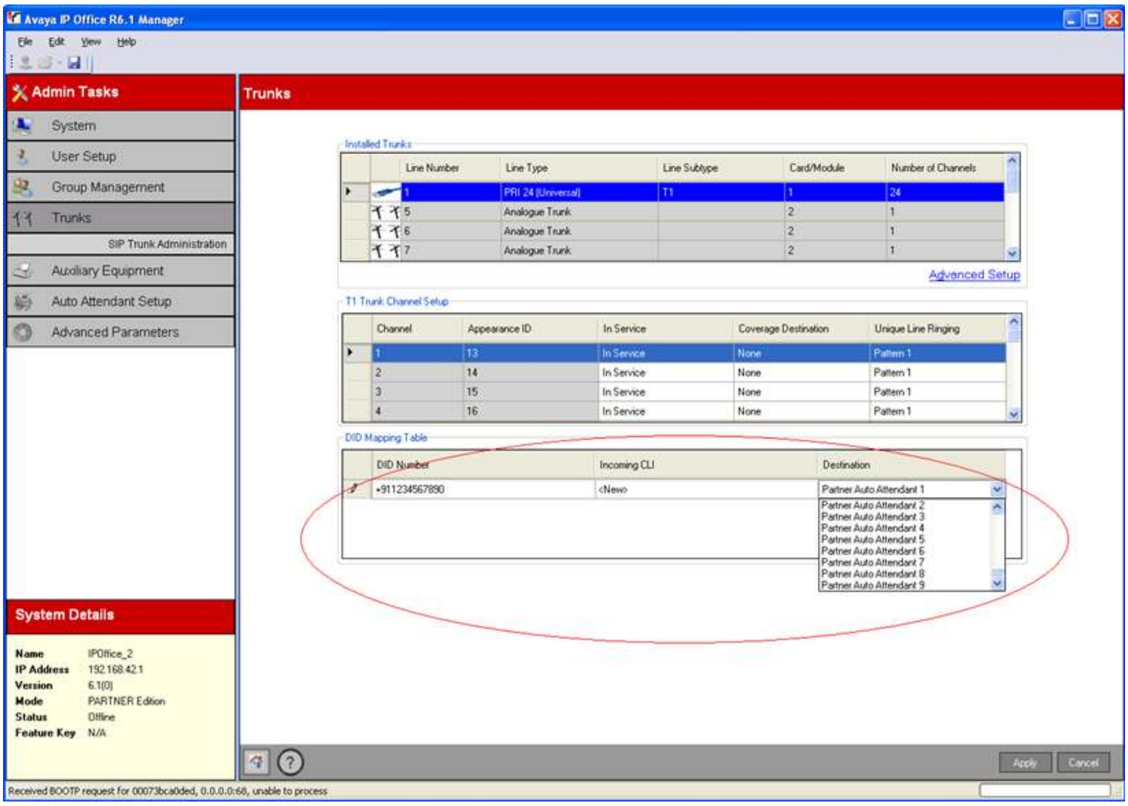
Some examples of sub-menu applications:

1 x AA with up to 8 sub-menus
2 x AA with up to 7 sub-menus that can be assigned to any/all AA
3 x AA with up to 6 sub-menus that can be assigned to any/all AA



3.3. DID Auto Attendant Support

DID number(s) can be mapped to any of the 9 AA's. When DID calls are routed to the AA, the call terminates at the AA without delay. If someone accesses a DID call while it is being serviced by the AA, the AA is dropped from the call.



3.4. Time Profiles

- Selector codes are available for each of the 9 x AA for morning, afternoon, evening, and out-of-hours time profiles.
- These codes are used to select the appropriate Day/Night service mode or AA time profile.
- Unique menu prompts can be recorded for each of the time profiles.
- When a line is covered to an AA that is following Night Service, and the system is in Day mode, and there are no active time profiles (or gaps exist between active time profiles and the current time), then the out-of-hours menu prompts and selector codes tables apply.
- One of the four possible menu prompts can be used as a common menu prompt for all time profiled menu prompts (based upon the GUI configuration).
- If no menu prompt is recorded, after 8 seconds of inactivity, the system shall transfer the call to extension 10.

3.5. Emergency Auto Attendant

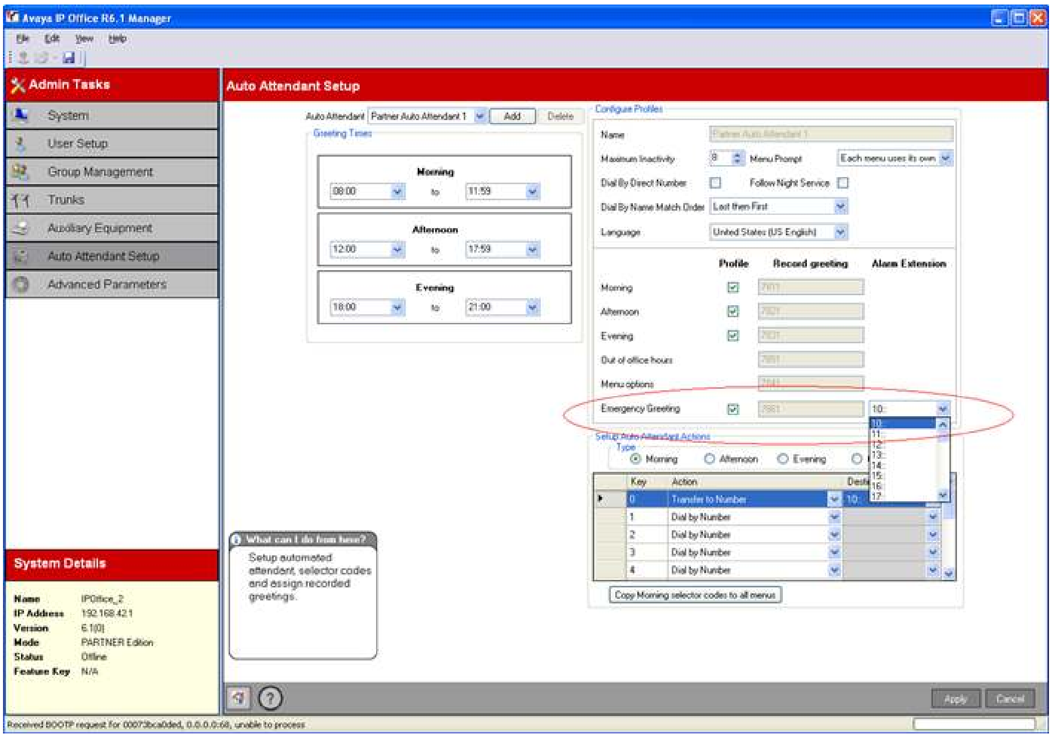
This feature provides for recording and activating an emergency greeting either locally using Simplified Manager or remotely using any touch tone telephone. Administrators need to be aware that they must setup a new automated attendant selector code action for each of the time profiles so that they can remotely access this feature using a touch-tone phone. A designated Alarm extension can be assigned for each Automated Attendant to receive a message on its display when the Emergency Greeting is active. This is a reminder message to the user to shut off the Emergency Greeting in a timely manner after the emergency ends. Administrators will be required to enter a security code in order that they can enable/disable the emergency greeting.

- Each AA supports an Emergency Greeting (e.g. business closing to due inclement weather) that can be remotely activated/deactivated and recorded. The greeting can be activated/deactivated using the Simplified Manager, or by using any touch-tone telephone.

- The administrator must establish a new AA selector code action for each of the time profiles, so that they can be access remotely.

- A dial code is assigned to record the Emergency Greeting in the same way that a normal Greeting is recorded. The dial code is *786x*, where x is the AA number.
- The Emergency Greeting defaults to inactive, with no default greeting.

- When the Emergency Greeting is active, it is played prior to the AA greeting.



The greeting can be recorded and activated by dialing into the AA and mapping a key to the *Transfer to Emergency Greeting* action.

The greeting is administered as follows:

- Call into the AA.
- Press the key that has been mapped to *Transfer to Emergency Greeting* action. A dial code is assigned to record the Emergency Greeting in the same

way that a normal Greeting is recorded. The dial code is *786x*, where x is the AA number.
- Enter the access code when prompted by the system.
- If the access code is valid, one of the following prompts is played:
  - *The Emergency Greeting is active*, or
  - *The Emergency Greeting is inactive*.
- Then, he following prompts are played for the user to choose from:
  - *To hear the recording, press 1.*
  To change the recording, press 2.
  - *To save the recording, press 3.*
  - *To activate the Emergency Greeting, press 4.*
  - *To deactivate the Emergency Greeting, press 5.*

Security
- Each AA supports the capability to require that a user enter the system password prior to activating/deactivating/recording the Emergency Greeting.
- The system-wide password is shared among all 9 AA.
- If the system password is not set, the system compares the entered password with a null password. After 3 attempts, the call is dropped.
- There is no mechanism to enter an empty password, so it will not be possible to access the Emergency Greeting.
- Simplified Manager activation/deactivation of the Emergency Greeting uses the existing Remote Access password.

Alarm Extension
- A designated alarm extension can be assigned to each AA to receive a message on its display when the Emergency Greeting is active.
  - This also serves as a reminder message to the user to shut off the Emergency Greeting in a timely manner after the emergency ends.
  - By default, extension 10 is the designated alarm extension for AA 1.
- An alarm extension can be the recipient of multiple AA Emergency Greeting alarms.
  - The extension's display will show a string of up to 9 digits, representing the particular AA's where the Emergency Greeting is active.
  - *NNNNNNNNN,* where the worst case scenario is that all 9 AA have an active alarm.
  - Note: If an alarm extension has only a 16 character display, it should not be an alarm extension for more than 2 AA. Otherwise the display information will be truncated and information will be lost.

3.6. Different Selector Codes for Time-Profile Operation

- Selector codes are available for each of the 9 AA for morning, afternoon, evening, and out-of-hours time profiles.
  - These codes are used to select the appropriate Day/Night service mode or AA time profile.
  - Unique menu prompts can be recorded for each of the time profiles.

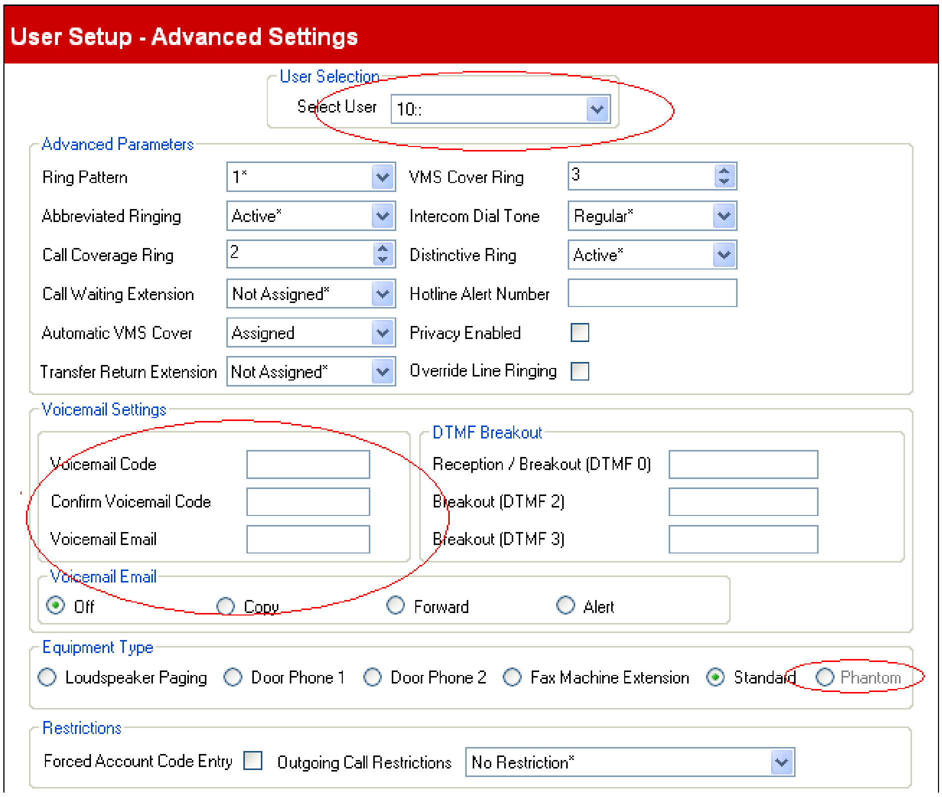
- When a line is covered to an AA that is following Night Service, and the system is in Day mode, and there are no active time profiles (or gaps exist between active time profiles and the current time), then the out-of-hours menu prompts and selector codes tables apply.
- One of the four possible menu prompts can be used as a common menu prompt for all time profiled menu prompts (based upon the GUI configuration).
    - If no menu prompt is recorded, after 8 seconds of inactivity, the system shall transfer the call to extension 10.

3.7. Phantom Mailbox Support without Hardware

Phantom Mailbox Support is implemented to provide mailbox support for all extensions 10 to 57 that are not physical represented on the system – known as phantom extensions. This was a feature of Partner ACS that allows flexible auto attendant functionality.

Provides voice mailbox support for all extensions (10-57), without requiring the physical hardware to be present. Extensions without hardware are called *phantom extensions*.

- Note: Ports 7 and 8 of an ETR-6 module can be used for phantom extensions even though you cannot plug a phone into them.
- Calls to a phantom extension go directly to voice mail.
- DTMF breakout service can be used to transfer the call.
- Phantom voice mailboxes:
    - Default like a normal mailbox.
    - Can be accessed remotely if the remote access feature of the mailbox is enabled.
    - Voicemail to Email can be activated for a phantom mailbox via the Simplified Manager.



3.8. End of Recording Options

A caller who has finished recording a message into a user's voice mailbox can press the ``#'' key signalling the end of message recording. At that time the system will play one of two messages:

If the message was less than three seconds, the system will play the following prompt: ``Message too short. Deleted. Goodbye.''

If the message is longer than three seconds, the system will play the following prompt: ``Your message has been sent. Goodbye.''

3.9. User can mark Old or Saved Message as new

Press *06 to move an Old or saved Message to the end of the list of New Messages. This option will also be reflected in 14xx Visual Voice, A new option is added to the *Visual Voice -> Listen -> Old/Saved* sub-menus to mark a message as *New*. . When a message is marked as *New*, the system does not initiate a new Voicemail to Email message, but the extension's message waiting light is illuminated.

3.10. Bypass Greeting

Callers that are directed into a User's mailbox can press "1" to bypass the recorded greeting and immediately record a message.

Available in Partner/Norstar Versions and IP Office Mode

3.11. Immediate Coverage to Mailbox

Configuring a "0" Rings option in a user's 1400 TUI or via Simplified Manager Cover Rings configuration will send calls immediately into the user's voice mailbox without alerting the user's extension, or the caller having to wait one ring cycle.

3.12. Automated Attendant

Each automated attendant can have it's own language setting which overrides the primary System Language. This is useful for implementing an automated attendant sub-menu that is in a different language than the automated attendant's main menu. This is configured via Simplified Manager. The language options are English/French/Spanish.

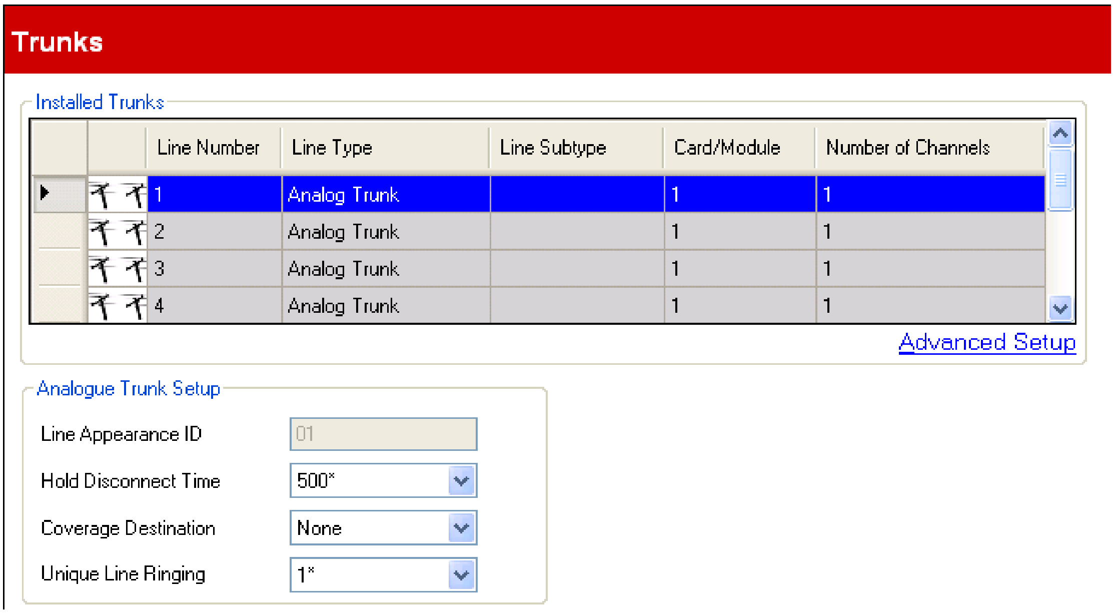
3.13. Selection of Primary System Language

Simplified Manager – The default language is English, but Simplified Manager "System Setup" will provide the option of changing the "Language" via a drop down list – options are English (default), French & Spanish.

3.14. Unique Line Ringing

The ability to assign different ring patterns in Simplified manager to incoming lines. This feature allows users to differentiate by sound which line is ringing by assigning a ring pattern to it.

Once a ring pattern is assigned to a line, incoming calls on that line ring with the assigned ring pattern.
8 ring patterns are available.
Unique Line Ringing does not apply to DID calls because the channel used by the PTT is not guaranteed.
If the Override Line Ringing feature is active, then incoming calls on a line that ring at an extension, ring with the configured extension ring pattern.
A new System Programming Code, #209, is available.
The default of Unique Line Ringing is pattern 1.



3.15. One-Touch Call transfer

A user can initiate a transfer of a connected internal or external call by touching a programmed inside Autodial (AD-I Auto Dial Intercom) button. An inside Autodial button is defined as one that has been programmed for one of the following purposes: Intercom Ring of an extension, Intercom Voice Signal of an extension, Group Calling Ring, Group Calling Page, Group Hunting Ring, Group Hunting Voice Signal, and Simultaneous Page.

This feature allows a user to automatically transfer a connected call with one-touch -- by pressing a pre-programmed AD-I (Auto Dial Intercom) for the destination extension. (The *Transfer* button is not used.)

The transfer is completed when the transferring station goes on-hook, or touches the *Transfer* button or *Complete* soft-key.

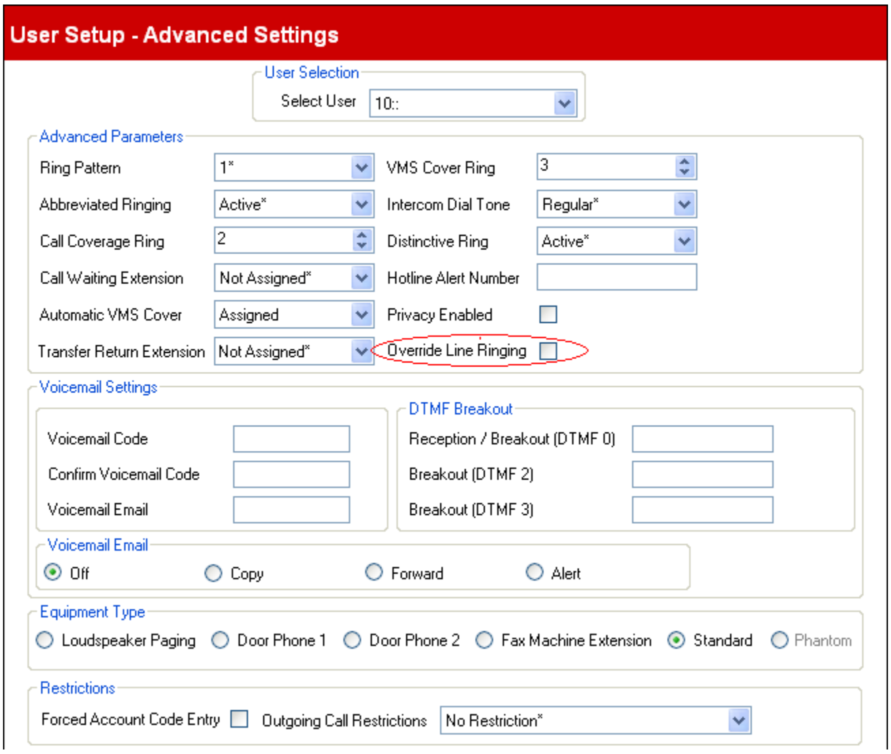
The user initiating the transfer is able to consult and complete the transfer or blind transfer after successfully initiating a one-touch transfer.

The LED's at the telephone initiating the transfer, as well as other appearances of the call in the system behave the same as if the user touched the *Transfer* button to initiate the transfer.

3.16. Override Line Ringing

There is a new system programming selection in the 1408 and 1416 user interface and Simplified Manager. When toggled on, this configuration will override the line ringing pattern with the station ringing pattern for calls to that station where the line ringing pattern would have applied.

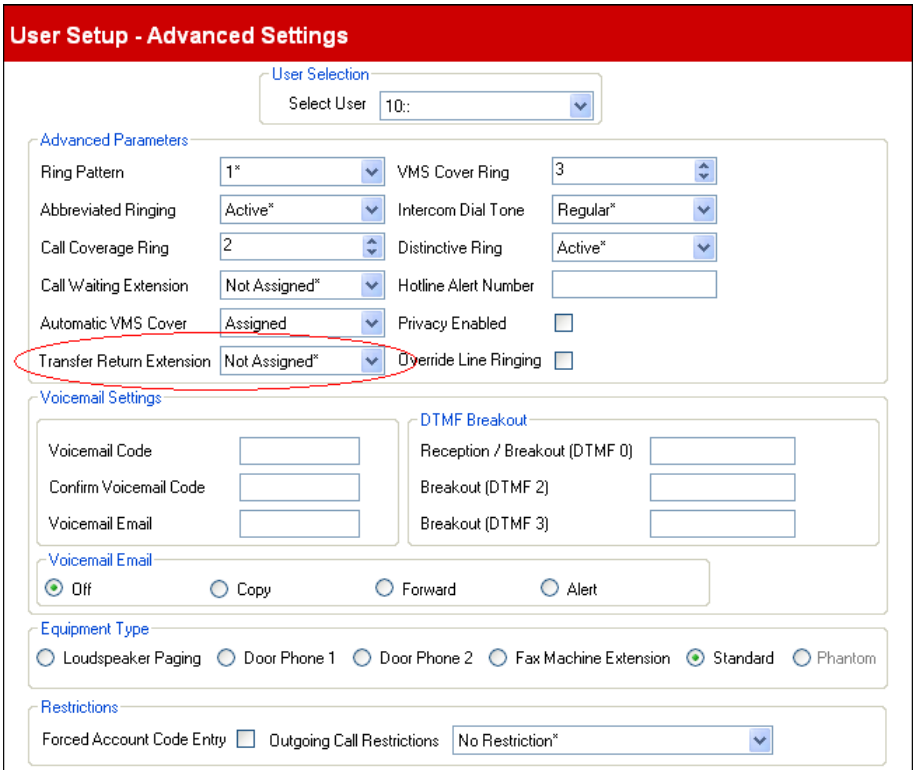
This feature is used to specify whether extensions receiving calls on lines follow the Unique Line Ringing Pattern or the Extension's Ring Pattern.
When this feature is active, the unique line ringing pattern will be overridden by the extension's ringing pattern.
A new System Programming Code, #324, is available.
The default is “Not Active” (i.e. do not override).
Note: #324 was defined in R6 to be *Reset Voice Mail Password*. *Reset Voice Mail Password* is now #325.



3.17. Transfer return to programmable Extension

This provides a programmable option to have incomplete transfer attempts ring back at a pre-programmed extension. The feature is programmable on a per extension basis.

This feature provides an option to re-route unanswered transferred calls to an alternate extension which is programmable on a per extension basis.
The existing System Programming option, *Transfer Return Rings*, will be used to indicate when transferred calls are returned to the Transfer Return extension.
If a call is routed to a transfer return extension and there are no available intercom appearances to terminate the call, the call will continue to alert at the transfer destination until the transfer return extension becomes available.
A new System Programming code, #306, allows a transfer return extension to be configured for each extension.
The default value for each transfer return extension is its own extension.

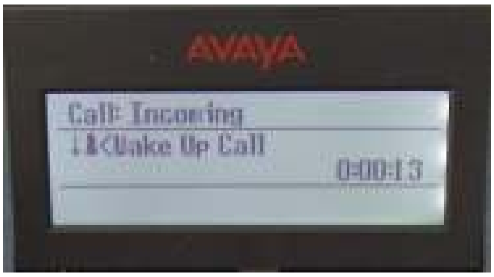


3.18. Wake Up Service

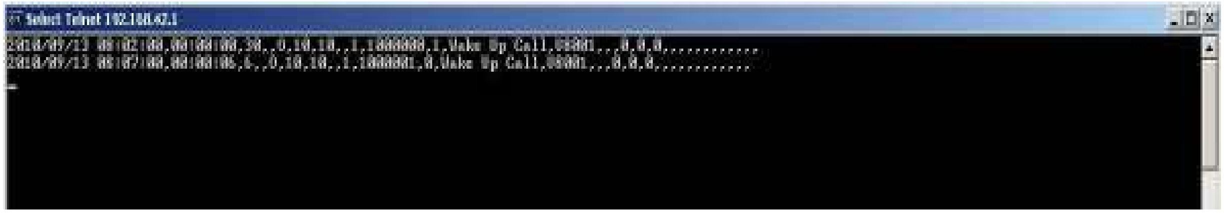
This feature allows a system telephone (Extension 10) to setup a Wake up Call on behalf of another user.

This feature allows a system telephone (x10) to set a Wake Up Call on behalf of another user. When a Wake up Call is scheduled, the system will place an intercom call to the target extension at the scheduled time.
Wake Up calls ring for approximately 30 seconds (and override DND).
If the target extension is busy on a call or has an alerting call, the Wake Up call is delivered to the telephone as an intercom call.
If the Wake Up call is answered, the user will hear Music on Hold if it is active; otherwise, they will hear silence.
If the Wake Up call is not answered, the system will make a 2nd attempt 5 minutes later.
If the call is not answered after the 2nd attempt, then the Wake Up call is abandoned.
A scheduled Wake Up call occurs once in a 24 hour period.

Telephone displays will indicate a Wake Up call

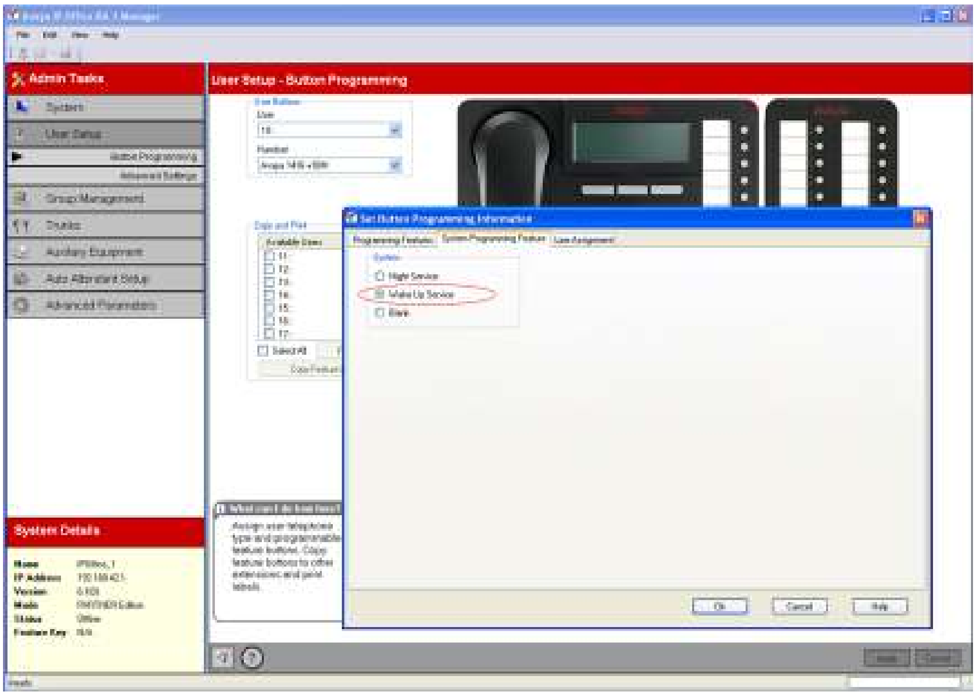


SMDR records will identify the unanswered and answered Wake Up calls.



If a Wake Up call has been scheduled for an extension, and the call hasn't been delivered yet, a new Wake Up time can be scheduled for that extension. The Wake Up call is rescheduled for the new time.

- To schedule a Wake Up call for an extension:
    - Press the Wake Up Service Button at x10 (x10 is a system telephone).
    - Dial the 2 digit target extension number (or use Intercom Auto Dial Button)
    - Enter the Wake Up Time in 24 hour notation using the dial pad.
    - When the time is successfully entered, the Wake Up call is scheduled.
- Similar procedures exist to review a scheduled Wake Up time, or to cancel a scheduled Wake Up call.
- A new *Wake Up Service Button* is programmable on x10 only using a new System Programming code, #115.
    - The default value is Not Assigned.



3.19. Service and Support Enhancements

1. SD Card shutdown/start-up via TUI - The system provides an independent TUI command option to shut down and restart the System SD card slot and the Optional SD card slot, to allow for removal and reinstallation of the System SD and Optional SD Cards. On the 1400 telephone, the User must dial #733 while in System Administration, the user is then taken into the

Memory card sub menu. Once any of the memory card commands have been started, exiting the screen does not abort the operation.

2. Display System release via TUI - A new Feature code 590 shall be added to the system. Any user in the system can enter this feature code by touching the soft key labelled with Feature on the 1400 telephones and dialling the digits 590 on the dial-pad.

NOTE: *In R6,* ***Feature 59*** *was used to display the system release information. This has been changed to* ***Feature 590*** *in R6.1 to accommodate additional system information using the 59 prefix digits.*

3. Display System IP Address via TUI - A new Feature code 591 is added to the system. This feature code will display the IP address associated with the system's primary Ethernet interface. This provides access to read the system's IP address without requiring a PC running Simplified Manager.

4. Display SD card Feature Key Number via TUI - A new Feature code 592 is added to the system. This code shall display the system SD card's feature key. This provides access to the feature key without requiring a PC running Simplified Manager or shutting down the system and removing the SD card to read the feature key from the label.

Remote Access to built-in Modem (x76) via selector code or DID - This feature provides for unattended access to the built-in modem by calling in to the system via a DID number or by transferring to the modem using an automated attendant.

The Simplified Manager Auto Attendant Setup allows configuring extension 76 as the destination for a *Transfer to* Number action. The menu shall show: ''76::Modem''.

The DID mapping table for T1/PRI trunks and the call by call table for SIP trunks will allow mapping a DID number or numbers to the modem destination. The menu drop-down displays: ''76::Modem''. ``Modem'' is a fixed label assigned by the Simplified Manager. These configuration changes are merged without a system reboot.

TUI Menu Optimization - When navigating *Admin->Telephone Programming* on a 1408 or a 1416 terminal, the first menu option presented is *Button Programming*. All of the other telephone programming options have been shifted down and maintain their current order.

3.20. IP Office Essential Edition – PARTNER® Manuals and Guides

IP Office Essential Edition – PARTNER® Version Manager Manual
IP Office Essential Edition – PARTNER® Version Quick Installation Guide
IP Office Essential Edition – PARTNER® Version Installation Manual
IP Office Essential Edition – PARTNER® Version Phone Based Administration Manual
IP Office Essential Edition – PARTNER® Version ETR Phone User Guide
IP Office Essential Edition – PARTNER® Version 1400 Phone User Guide

4. Essential Edition Norstar

MENA Market only – “Essential Edition Norstar”. Supported on IP500v2 with 1400 Series Telephones only. The expansion of the IP Office Essential Edition – PARTNER® Version into the Middle East and Africa (MENA) region and is referred to as the “Essential Edition Norstar” Version. This system is aimed at Norstar Business Partners as an alternative solution to the Norstar system in the MENA region only. A new set of locales as well as some existing locales supported in IP Office mode are supported. The Essential Edition Norstar is provisioned using a pre-programmed SD card that will launch in the “Norstar Version mode. Customers will be able to select country specific defaults as a configuration item “system locale” on the Norstar Version.

4.1. System Overview

- Capacity:
  - 48 stations
  - 56 trunks
- Trunks Supported:
  - Analog (up to 4 ATMS, 16 trunk capacity)
  - SIP
- Terminals Supported:
  - 1400 digital telephones only and button modules (BM32)
  - Analog devices (POT's, etc.)
- Fixed 2-digit dial plan (same as Partner Version)
- Embedded voice mail is provided for all users (same as Partner Version)
- Essential Edition Norstar shares the same feature set as Essential Edition PARTNER excluding the support of the PARTNER ETR telephones.

The IP500v2 in Norstar Version only supports the following cards/expansion modules and components:

- IP 500 V2 (700476005)
- IP 500 V2 SD Card “Norstar” Version (700500948)
- IPO 500 EXTN CARD DGTL STA 8 (700417330)
- IPO 500 EXTN CARD PHONE 2 (700431778)
- IPO 500 EXTN CARD PHONE 8 (700417231)
- IPO IP500V2 COMBINATION CARD ATM (700476013)
- IPO 500 TRNK ANLG 4 UNI (700417405)
- IPO 500 EXP MOD PHONE 16 (700449507)
- IPO 500 EXP MOD DGTL STA 16 (700449499)
- 1403 TELSET FOR IPO (700469927)
- 1408 TELSET FOR CM/IPO/IE UpN (700469851)
- 1416 TELSET FOR CM/IPO/IE UpN (700469869)
- BUTTON MOD FOR 1400 SERIES (700469968)
- 1 Expansion module (DS16 or POTS16) only is supported in slot 1.

Any cards and telephones that are plugged in and not supported will remain disabled and unusable

4.2. Supported Locales

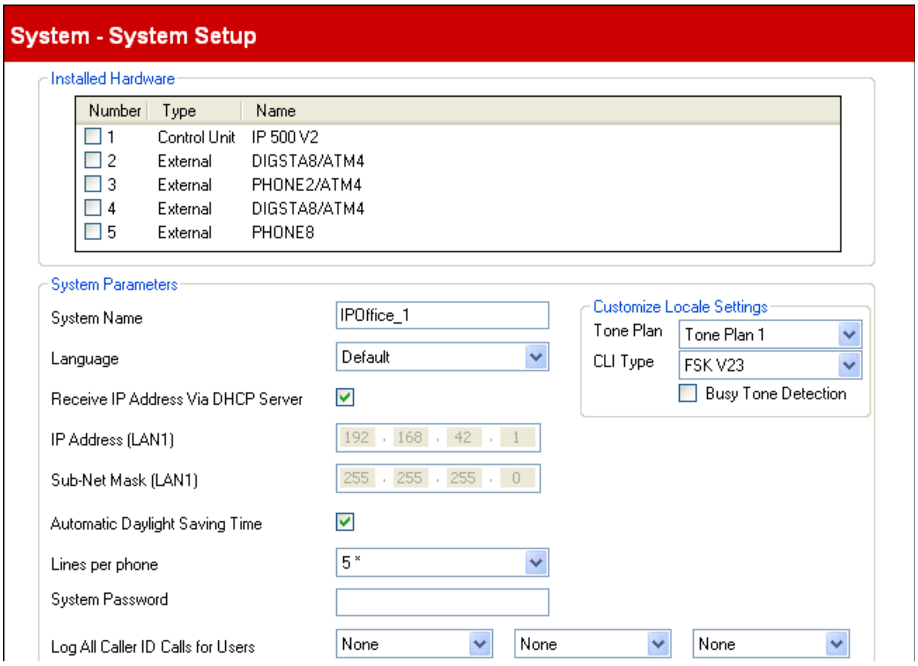
The following locales are supported in Essential Edition Norstar:

- Bahrain
- Egypt
- Kuwait
- Morocco
- Oman
- Pakistan
- Qatar
- Saudi Arabia
- South Africa
- Turkey
- United Arab Emirates
- Default

Norstar Version defaults to the *MEA* locale. *MEA* is equivalent to the *CUS* locale available in IP Office mode. Tones, Cadences and Analog Trunk parameters are defaulted based upon the system locale. The new locales incorporate the Automatic Impedance Match for Analog Trunks.

4.3. Supported Languages

- UK English (Default)
- European French
    - Available for telephone displays, Embedded Voice Mail Prompts, and Simplified Manager screens.
    - The telephone displays are not available in Arabic and default to English.
    - Embedded Voice Mail prompts played to the user are Arabic, French or English. The default is English for all locales except Morocco where the default is French.



4.4. Embedded Voice Mail

- The default language used by Embedded Voice Mail is based upon the system locale.
- The user locale defaults to UK English so that the voicemail prompts will be played in English.
  - The user locale can be changed to provide prompts in a different language. For example, if the user locale is changed to Egypt, voice prompts are played in Arabic.
  - External calls that terminate into voice mail in locales where the default is Arabic, will always hear prompts in English because the incoming call routes are defaulted to UK English (this setting cannot be changed).

4.5. Analogue Trunks

For the Default (MEA) locale all analogue trunks are treated as unsupervised trunks i.e. the call must be cleared at both ends for the trunk to be idled. If the call goes to voicemail/auto attendant then the trunk will use busy tone detection to clear the call. This also means that you cannot do an unsupervised transfer.

The system will only process busy tone detect clear when the call is connected to voicemail. In all other cases where you have two telephones at either end then the call must be dropped at both ends.

This only applies to the MEA locale. For all other locales it behaves in the same was as IP Office busy tone disconnect-clear being the default operation.

- Current IP Office operation assumes that analog trunks support either disconnect clear or busy tone detection to detect disconnect signals on trunk calls.

- In the MEA region, trunk disconnection is done by busy tone detection, but with a much wider variation in the frequency range.
- For Norstar Version, in the MEA locale only, analog trunks are managed as unsupervised trunks.
- A new filter has been added to detect busy tone detection in a much broader frequency range.
- Trunk-to-Trunk transfers are barred.
- Analog trunks will not allow unsupervised transfers to internal extensions.

4.6. Set Based Administration

Set-Based Administration menus follow the current PARTNER Version structure. Norstar Set-Based Administration is NOT supported on Norstar Version. 1416 and 1408 digital telephones can be used to perform Set-Based Administration.

4.7. Licensing

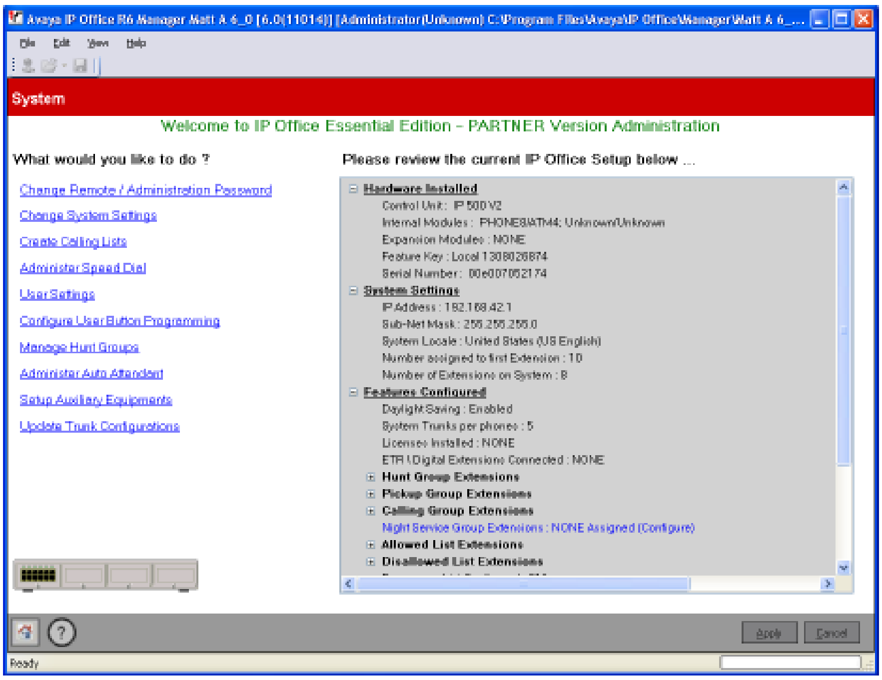
Norstar Version, out-of-the-box, maintains the same license defaults as PARTNER Version:

- 2 channels for Embedded Voice Mail
- 3 SIP Trunk licenses
- 3 Mobility licenses.

4.8. Services and Support

SysMon and SSA will refer to the product as Norstar Version. The system mode and core software version is displayed when *Feature 590* is dialed from any 1408 or 1416 telephone in the system and will display the system software version.

For example, NV 6.1 (N), where N is the build number.



By using the 10 System Hyperlinks, you are able to quickly and easily administer the PARTNER® Version system in a short time.

4.9. Essential Edition Norstar Manuals and Guides

Further details can be found in the IP Office Essential Edition - NORSTAR Version Manager documentation:

IP Office Essential Edition – Norstar Version Manager Manual
IP Office Essential Edition – Quick Installation Guide
IP Office Essential Edition – Norstar Version Installation Manual
IP Office Essential Edition – Norstar Version Phone Based Administration Manual
IP Office Essential Edition – Norstar Version 1400 Phone User Guide

5. Core Software Features

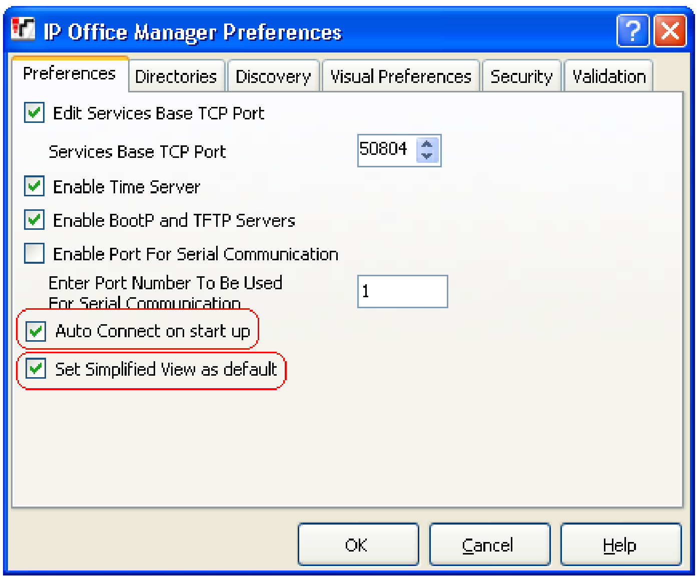
The following features are IP Office version only (not Partner or Norstar) unless otherwise stated.

5.1. Simplified Manager

A new mode of operation has been provided within the IP Office Manager, known as Simplified Manager, which is used to administer the system when it is running in Essential Edition - PARTNER® Version and Essential Edition – Norstar Version.

Although system administration can be completed via either the TUI or GUI method, some advanced features such as SIP and PRI/T1 administration require the Manager application.

The default mode of operation for the Manager application is set via Preferences. If you have selected “Set Simplified View as default” and “Auto Connect on start up” then the application searches for any IP500v2 units running Essential Edition - PARTNER® Version or Essential Edition – Norstar Version on your network and will automatically connect using the default login and password.



Once the system is contacted and the configuration open, you are then presented with a “Simplified Manager” Screen:

5.2. Embedded Voicemail – Skip Mailbox Feature

Supported in IPOffice, Partner and Norstar versions

Caller's can skip your mailbox greeting by pressing 1. Instead they immediately hear the tone for the start of recording.

5.3. Music on Hold – Multiple External Sources

Multiple Music-on-Hold sources are supported on IP Office only IP500, IP500V2, IP406V2, IP412 with Phone Modules.

Based on field feedback we have enhanced the existing IP Office MoH functionality with up to 4 external MoH inputs, and set the MoH source for outbound calls. IP Office will add the ability to take MoH from an analogue extension port (maximum of four). ETR ports are not supported.

An analog extension port is specified as XTN: followed by the base extension number. For example XTN:216 for base extension 216. The analog extension must be one that has had its Equipment Classification set to “MOH Source”. A suitable interface device is required to provide the audio input to the extension port. It must look to the IP Office like an off-hook analogue telephone. For example a transformer with a 600 Ohm winding (such as a Bogen WMT1A) or a dedicated MoH device with a 600Ohm output designed for connection to a PBX extension port which is providing loop current can be used.

Unlike playing WAV files using an Analogue Extension port will not use a Data Channel. There will be a new Equipment Classification option for Analogue Extension ports; "MoH Source". Extensions configured as MoH source will count as extensions for licensing - specifically in the limit of 32 extensions for Small System Upgrade License. An Extension configured as MoH Source will have a specific icon in Manager.

The new source can be used for incoming calls and for outgoing calls. The “h” character can be used in the Telephone Number field of short codes to specify the hold music to associate with calls routed by that short code. The format h(X) is used where X is the source number. Please review the IP Office Manager Help for more details.

- Ability to take MOH from an analog extension port (Max of 4)
- ETR ports not supported
- New Equipment Classification option for analog extension ports called "MOH Source"
- Extensions configured as MOH source count as extensions against licensing - specifically in the limit of 32 extensions for Small System Upgrade License
- An Extension configured as MOH Source will have a specific icon in Manager
- A new Short Code telephone number character 'h' will be added to select the MOH source that will be used when a call is placed on hold by the system originating the call

- An analog extension port is specified as XTN: followed by the base extension number.
- The IP Office will re-load any WAV files used for MoH if they have been changed on SD card.

5.4. Simplified Chinese Support

On IP Office, 1408 and 1416 telephones now support Simplified Chinese language strings for CHS System or User Locale.

5.5. Call Presentation

1400 & 1600 Telephones

On IP Office, when 1400 and 1600 telephones are presented a call with a call Tag String i.e. from an Operator, one-X Portal client or via a Voicemail Pro auto attendant, it will display the call state and Tag text on the top line when connected. In previous versions of software, the Tag would be lost once the call was connected.

This is not supported 1403 and 1603 Telephones.

When a call is presented to the telephone with no Tag String (receiving a transferred call with no associated Tag string) the call information will be arranged:

Top line
- Call: (Responding Party)

Application line(s):
- Calling Party > Redirecting Party

5.6. Per Call – Call Timer

Permits the ‘Timer’ feature to toggle the selected appearance’s Call Timer display on/off during a call. Supported Telephones:

- *1408 & 1416*
- *1608 & 1616*

Quickly prioritise the display of a particular call’s Timer or call information on telephones with limited display space. New calls always take the User’s Call Timer configuration for initially displaying the Call Timer.

5.7. Set Based Administration of Feature Labels

Users may change the displayed label for their features using Text Entry via a new option in the Self Administer Menu feature.

Supported Telephones: *9600 and 96x1 (96x1 are not available at 6.1 GA)*

5.8. System Administration and Admin 2 Menu Enhancements

Telephones configured as System Phones in Manager / User and have a Profile set as Level 2 can display in a programmed Admin 2 button:

- LAN interface IP addresses
- Feature Key Serial Number

5.9. Show last Call Duration

1400, 1600 and 9600 telephones will briefly display the duration of a call after it is ended. This setting is a user accessible option through the telephone menu **Features | Call Settings | Show Last Call Duration** (except on 1403 and 1603 telephones where it is on by default).

5.10. Mobile Twinning Handover

Mobile Twinning handover is the ability to swap calls between the desk telephone and the twinned mobile telephone.

- A user configured with the Twinning button feature may send a connected call from their desk telephone to their mobile by pressing the mobile twinning button. This feature utilises a simple “blind” transfer to the twinned mobile. The caller will hear ringtone once the mobile is ringing until connection. This feature will be enhanced further in a future release.
- During and after the transfer the call may be retrieved from the mobile as before.

5.11. Set Based Admin of Feature Labels

Users may change the displayed label for their features using Text Entry via a new option in the Self Administer Menu feature.

Supported Telephones:

- *9600*
- *96x1(Not available until 2011)*

5.12. Automatic Call Log Expiry

For call log entries written into a user's centralized call log, an expiry time can be set after which the call log entry is automatically deleted from the users call log. This is configured in Manager / User / Telephony / Call Log – Delete Entries after (hours: minutes) – for more detail, see section 7.1

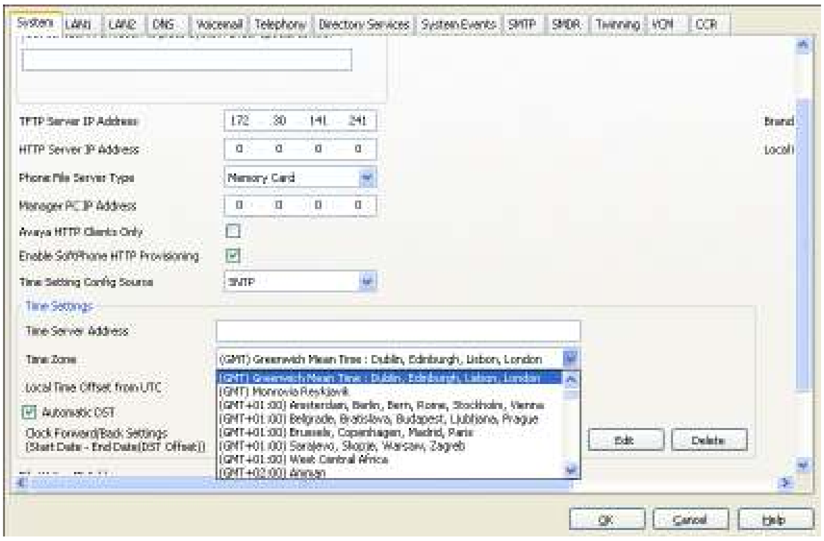
5.13. NTP and Automatic DST Time Services

IP Office systems now have the ability to obtain time information from an NTP server. This is configured in Manager / System.

- Up to three NTP servers are configurable (IP address or Name)
    - Names will be resolved using DNS

- IP Office will accept a list of time server addresses but only use one at any given time

- New SNTP (RFC4330) client alongside previous RFC868 client
  - Only one time source (NTP/RFC868) per system
  - Pre-6.1 scenario still possible (time from VMPro/Manager PC)



- As NTP only provides UTC information, added support for Automatic Daylight Saving Time (ADST) is now available in Manager / System

  - System time is set to DST and back according to ADST rules for the time zone the system is located in
  - Time zone information and ADST transition dates provided by Manager
  - Retained simplified ADST functionality for Partner Edition
  - The default Time zone settings are set by the system locale

Supported on IP500v2, IP500, IP406v2, IP412

5.14. Forced IP Telephone Restart at IP Office Upgrade

For IP500v2 systems where the SD card acts as the telephone file server, there is now the ability to upgrade IP Office and IP telephones in a single pass. When upgrading IP Office and uploading system files, Avaya IP telephones can now be forced to reboot even when the system is not under ENM control.

After reboot and during Upload of system files, IP telephones are not served by HTTP until all telephone binaries have been transferred to the SD card. When telephone binary upload is complete, IP telephones get served again and perform their individual upgrade procedure using up-to-date files. This is only for IP telephones that are local DHCP clients.

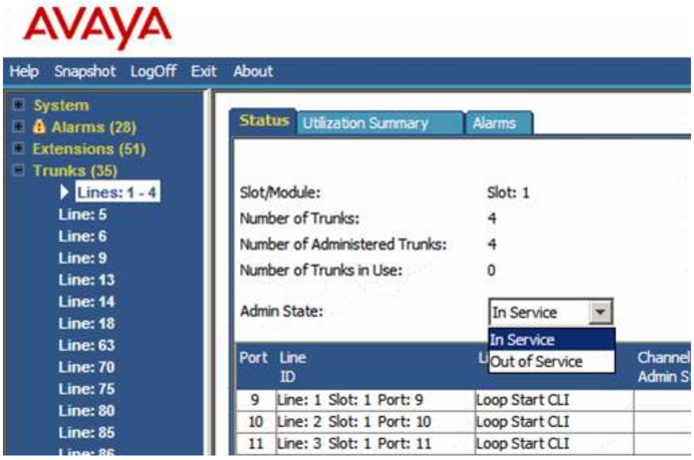
5.15. Remote PRI/T1 Loopback

SSA now has the ability to generate/analyze BERT test patterns on IP500 PRI Trunk cards (single and dual).

SSA menus to select Line or Channel loopback on IP500 PRI Trunk cards

- Lines - Payload or Line Loopback
- Channels - Payload Loopback

To enable loopback or to run a bit error rate test the effect channels or line must be first set Out of Service, and cannot have a call in progress. SSA can be used to disconnect active calls, then once idle, SSA will command the IP Office to generate/analyze test patterns. SSA / IP Office will only support the testing of a single channel/circuit at a time



Initiated tests report results

- Elapsed Time
- Errored Seconds
- Errors Detected
- Errors Injected
- Bit Error Rate

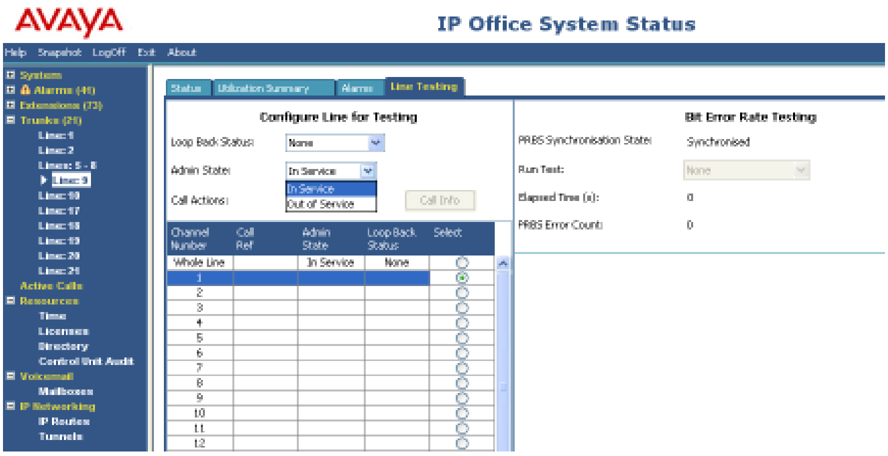
SSA buttons

- Reset Counters
- Inject Errors
- Stop Tests

5.16. Trunks/Channels “Out of Service” Diagnostics

SSA provides the system engineer with the ability to place TDM trunks Out of Service without requiring a reboot. There is a new tab on the Trunks form in SSA for “Line Testing”

In and Out of Service options are delivered via Manager (PRI/T1 Trunk) and SSA (Trunks / Status). The existing options in Manager can be “merged”.



5.17. Licensing Changes

Software Upgrade: All systems upgraded to Release 6.1 from a previous version outside of the 90 day entitlement period will require the Software Runtime License 3. This will display in Manager / License section as: Software Upgrade 3.

New IP500v2 systems do not require a Software License to run 6.1

This license *Software Upgrade 3* can be entered into the configuration using Manager prior to upgrade to 6.1, or after upgrade. IP Office services will not function until a valid runtime license is entered. The Manager Upgrade Wizard provides a reminder window at the time of upgrade to ensure a license is available either before or post upgrade.

5.18. SIP Telephone Feature Enhancements – BLF and Conference 3rd Party Add

Supported on IP500, IP500V2, 406V2, IP412

Additional features have been added for SIP endpoints that are used as key systems:

Busy lamp Field – If a User number is programmed into a SIP Endpoint BLF that matches the configuration, Idle / Busy indication will be supported. This is primarily intended for support of the:

- Nortel 11xx and 12xx series telephones DDS Keys (IP500 & IP500v2 only)
- Avaya Softphone Speed Dial Buttons.

Although we have received confirmation that this feature works on various SIP endpoints of different manufacture, we cannot guarantee support for other SIP endpoints and pre-testing is recommended.

- Previously unsupported features of RFC 5359 – Call Forward, Conf 3rd party add are now supported on SIP endpoints.

5.19. SIP Trunk Interoperability on IP Office with BCM and CS1000 Systems

Supported on IP500, IP500V2, 406V2, IP412

Interoperability between BCM Release 6.0, CS1000 and IP Office Release 6.1 will support the following features:

Basic Call
- Basic Call Completion
- Handling of Busy Called party
- DTMF and Ring Back Tone
- Hold and Retrieve
- Call Waiting Presentation
- Called Number Display
- Calling Name Display
- Abandoned Call

Call Redirection (at node level – no network optimization)
- Call Forward
- Call transfer
- Call Redirection

Conferencing
- 3-Party and Multiparty Conferencing

6. SIP Trunk Resiliency

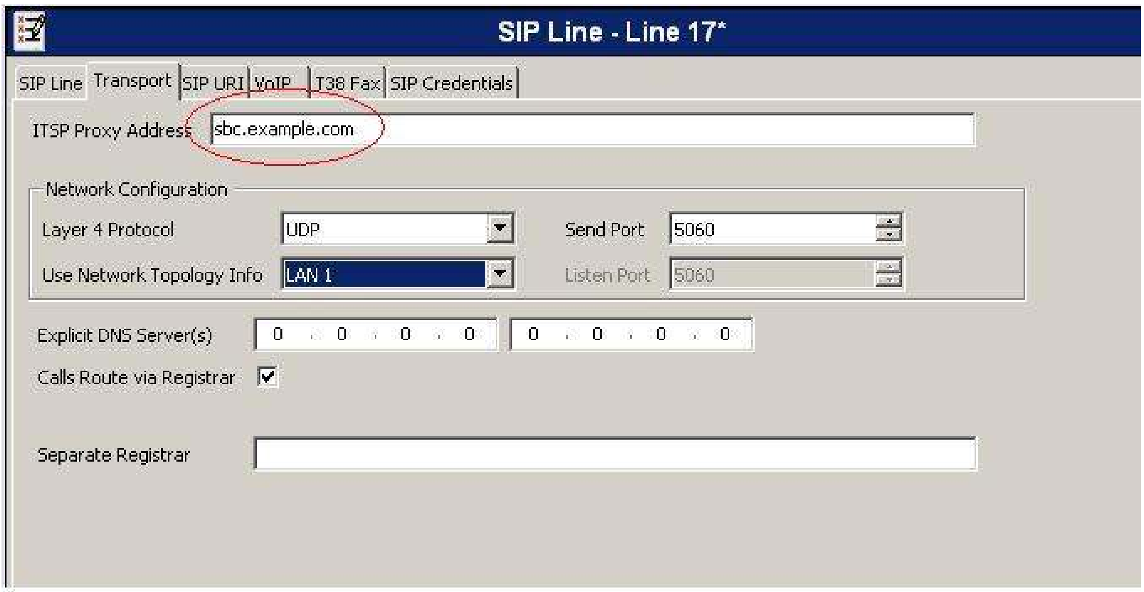
Supported on IP500, IP500v2, 406v2, IP412

The following SIP trunk enhancements have been delivered in IP Office Release 6.1 in response to requests from SIP Providers. The main focus for the enhancements to SIP configuration and functionality:

- Implement Outbound Proxy via DNS Business Driver
- Implement Support for enhanced DNS - SRV lookup, multiple records (RFC3263)

A new tab is added to the SIP configuration form labelled “Transport”. The new tab contains 4 new fields:

- ITSP Proxy Address (free form text field, max 79 chars.)
- A DNS address (e.g. “sbc.example.com”)
- A DNS address with load balancing post-fix (e.g. "sbc.example.com 1...4")
- A “Calls route via Registrar” checkbox



6.1. ITSP Proxy Address

This is a free-form text field max 79 characters. The following formats are supported:

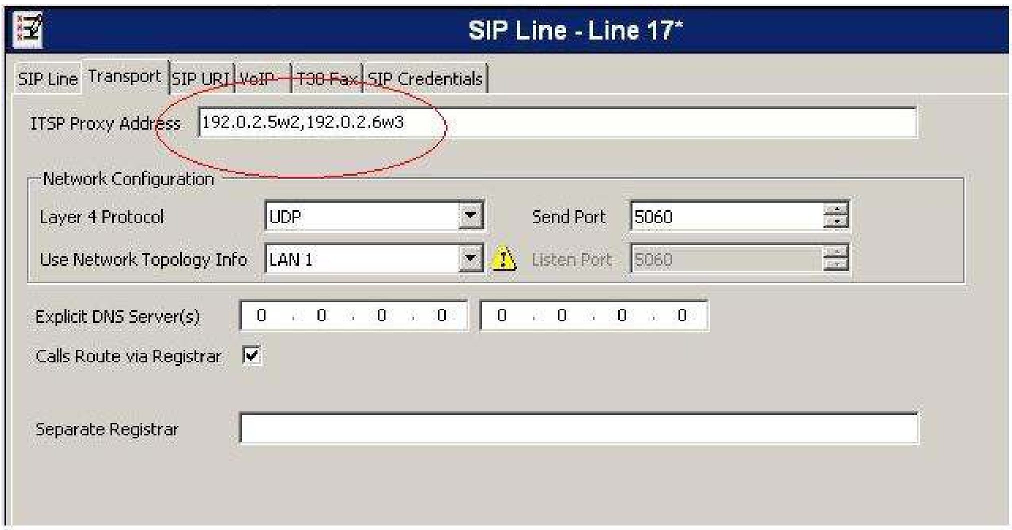
- Empty - the address is resolved from the Domain name
- An IP Address (e.g. "192.0.2.5")
- A list of IP Addresses, separated by comma or space. Maximum 4 addresses allowed.
- Addresses can include indication of relative call weight (each address’ call weight is compared to the others). If an address if followed by a 'w', this indicates a relative weight. The default weight is 1. A higher weight will route proportionally more calls. Received calls do not count in the weighting.
    - This is done by adding “***wN***” suffix to the address where ***N*** is the weighting value (i.e. address1***w2***,address2,address3***w3***)
    - Entries with higher w= value take higher number of calls
    - Entries with no w= value default to 1
    - Entries with w=0 can be used to disable an address

Example Entry: 192.0.2.5w3 192.0.2.6w2" - the first Proxy will be given 1.5 times the weight of calls

- A DNS address (e.g. "sbc.example.com")

- A DNS address with load balancing post-fix (e.g. "sbc.bloxham.co.uk w2"(1...4)").

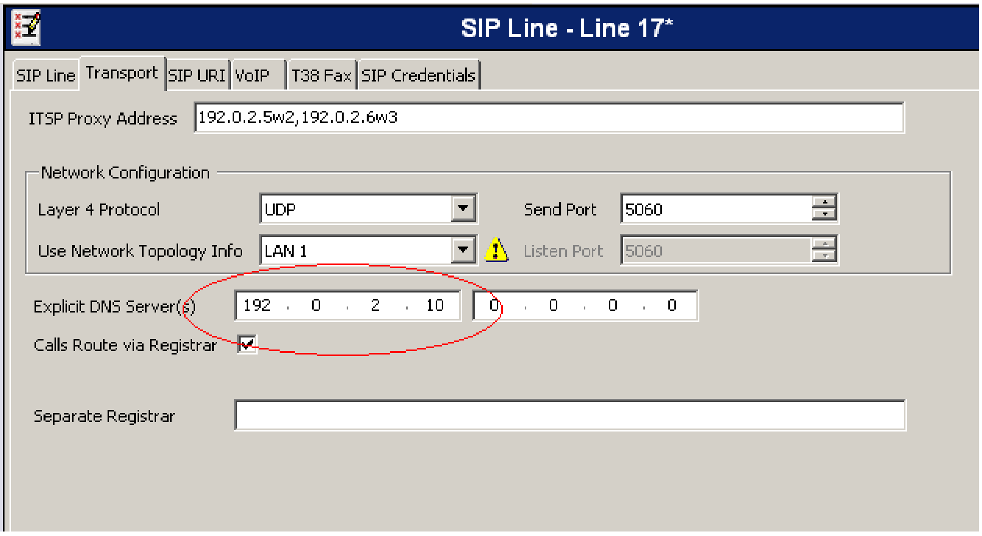
- The weight where not specified will default to 1. Where the weight is 0, this address should not be used for any new transactions.



6.2. Explicit DNS Server

Two fields containing the IP Addresses of DNS Servers. If specific DNS servers should be used for SIP trunk operation rather than the general DNS server specified or obtained for the IP Office system, the server addresses can be specified here (Default = 0.0.0.0 Off). If Explicit DNS Server(s) are configured, a DNS request will be sent out to see whether the proxy server has disappeared from those being offered. An Active-In Maintenance proxy server should not be used for a new transaction (INVITE or REGISTER) until:

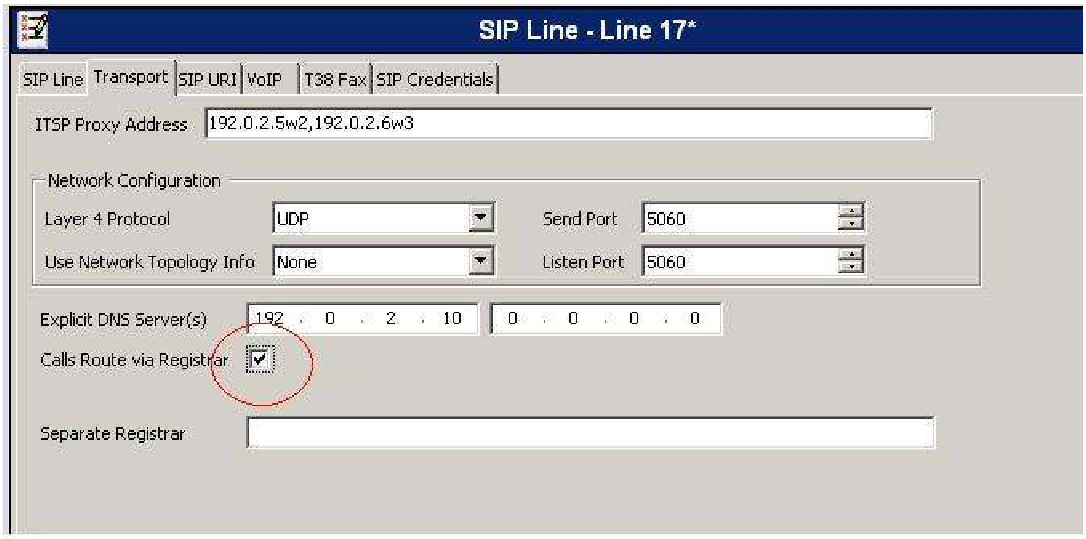
- There is a change in DNS responses indicating the proxy has become active
- The configuration does not leave any better option available. In this case, there is a throttle so that no more than 5 failures (without successes) in 1 minute should be allowed
- A configuration merge has occurred where the proxy string is changed
- 10 minutes has expired



6.3. Calls Route via Registrar Checkbox – Tied Balancing

A "tied" system means that a call must be sent to the same proxy server as the registration with which it is associated. In this case, it is the registrations which are load-balanced rather than the call distribution. This is specified by the check box: "Calls route via registrar".

- If selected, SIP ***registrations*** are balanced according to the weight of each resolved proxy. Once registered, all calls that are made go through the proxy where the associated **registration** has occurred
- All calls are routed via the same proxy as used for registration so the ***balancing is applied to registrations, versus calls***
- If multiple ITSP proxy addresses have been specified, the weighting for those addresses is applied to the registrations.

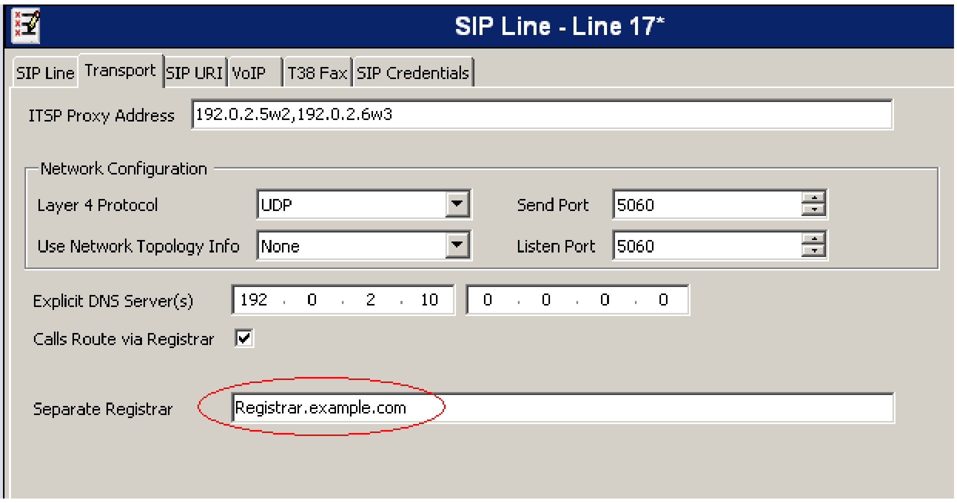


6.4. Separate Registrar

This field allows the SIP registrar address to be specified if it is different from that of the SIP proxy. The address can be specified as an IP address or DNS name

- A proxy server is considered *Active* once the IP Office has received a response to an INVITE, REGISTER or OPTIONS
- A proxy server is considered *Active - In Maintenance* if the server responds to INVITE with *503 - Service Unavailable*.

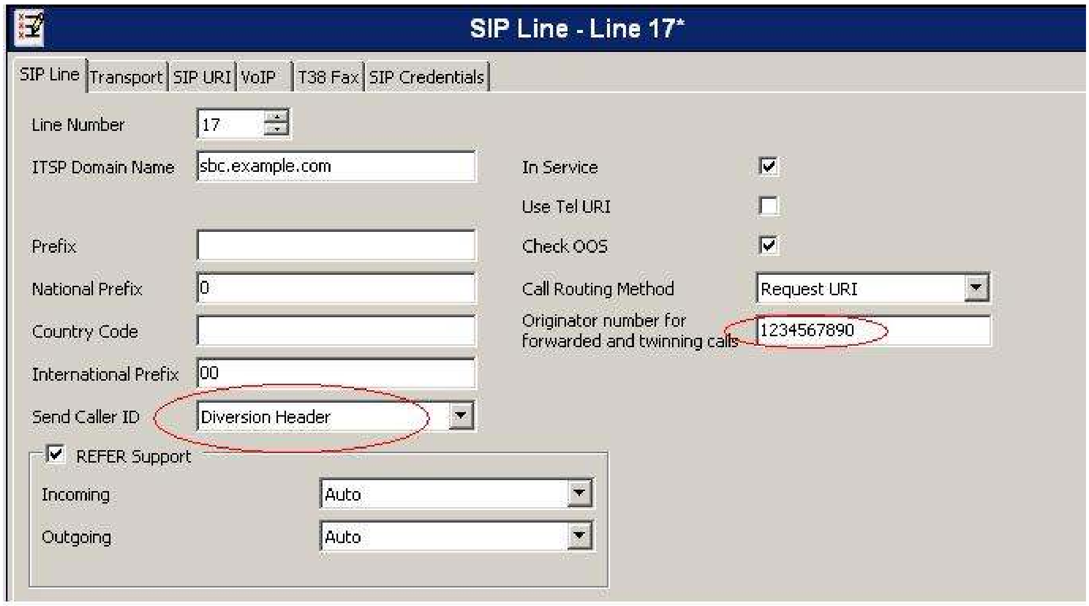
Please review the 6.1 Manager Help files for further information.



6.5. Diversion Header for Call Forward

This feature provides CLIP and CLIR feature to forwarded calls that were previously available only to twinning. This feature is available to any type of Send Caller ID (Remote Party ID, PAI – discouraged). Customers using a mobile twinning CLIR feature will use this feature automatically after upgrade (no special configuration required).

- The caller number can be pre-defined on the trunk
- If trunk uses Internal Data and caller is internal, the number will be mapped to SIP trunk Internal Data
- Reverse route lookup will be attempted for all group IDs belonging to that trunk
- For external calls, prefix manipulation will be used

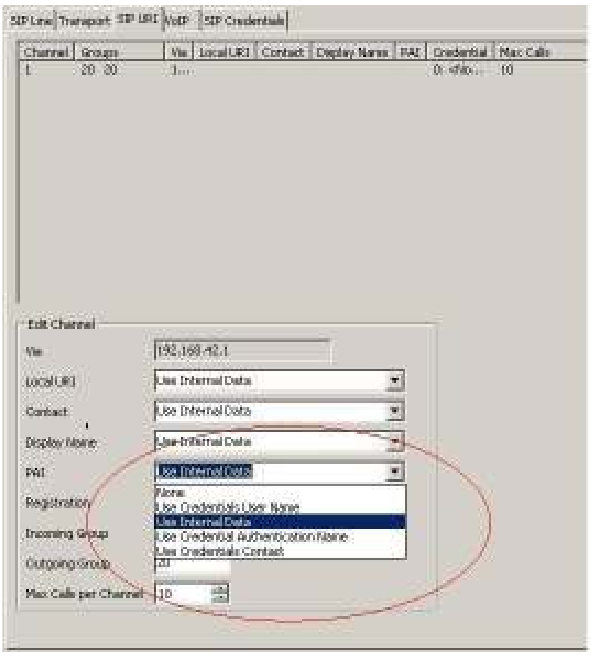


6.6. Configuring P-Asserted IS Header (PAI) per SIP Trunk URI

P-Asserted ID header is now configurable per SIP call (SIP URI configuration item). Possible values:

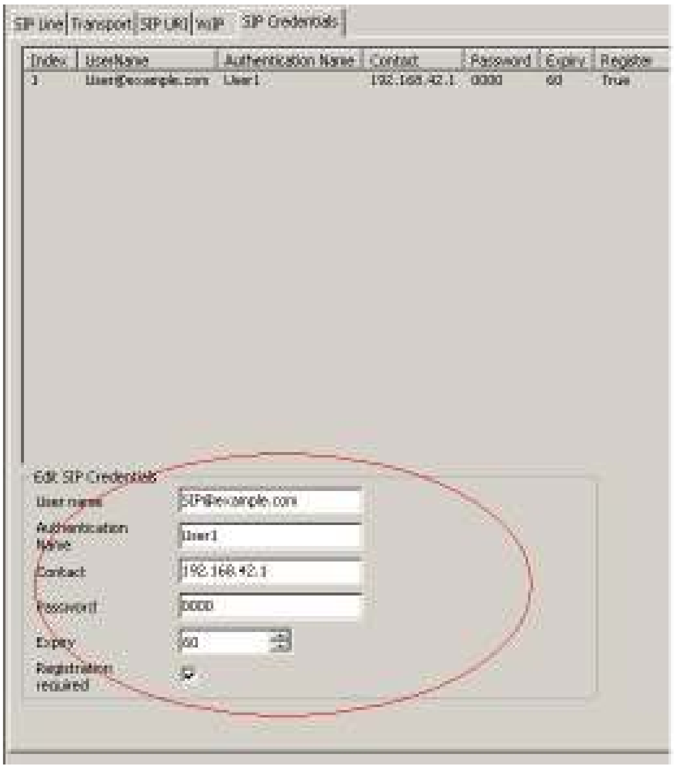
- None
- Use Credentials Authentication Name
- Use Credentials User Name
- Use Internal Data
- Use Credentials Contact

This feature is required for interoperability for many SIP providers. In order for this feature to work, Send Caller ID ***MUST NOT be set to P Asserted ID***



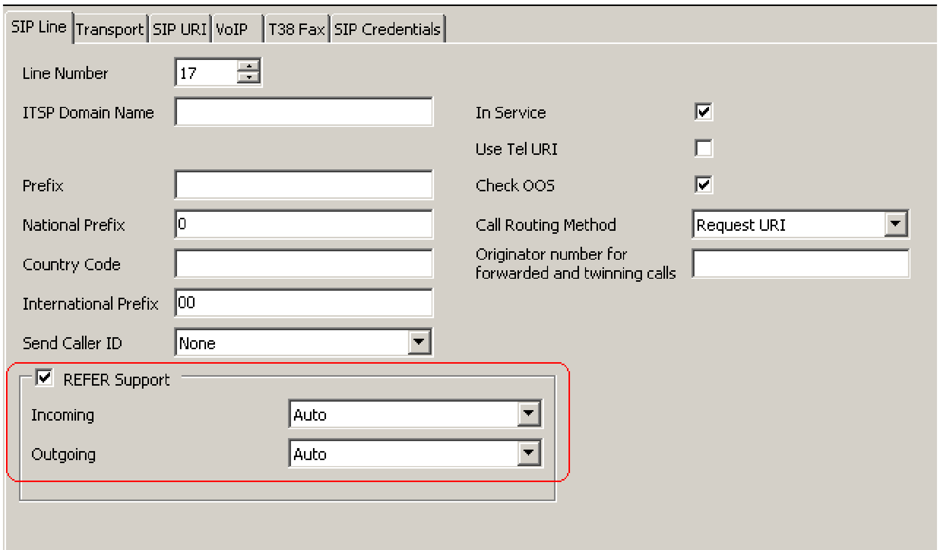
6.7. SIP Credentials Enhancement

- New configurable Contact field added. If not specified, the User Name value will be used.
- User Name and Authentication name can contain a domain name (can be Fully Qualified Domain Names).
- For the Contact field, Domain Name will always be the IP Office IP address or domain.
- SIP Credentials are now merged into the system configuration now so adding/modifying/deleting a credential will not require a reboot.
- You can enter up to 30 sets of ITSP account names and passwords on this tab.



6.8. REFER Support

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



- **Incoming –** Default = Auto - Select whether REFER can or should be used when an attempt to transfer an incoming call on the trunk results in an outgoing call on another channel on the same trunk. The options are:
  - **Always -** Always use REFER for call transfers that use this trunk for both legs of the transfer. If REFER is not supported, the call transfer attempt is stopped.
  - **Auto -** Request to use REFER if possible for call transfers that use this trunk for both legs of the transfer. If REFER is not supported, transfer the call via the system as for the ***Never*** setting below.
  - **Never -** Do not use REFER for call transfers that use this trunk for both legs of the transfer. The transfer can be completed but will use 2 channels on the trunk.

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

6.9. SIP trunk Video support

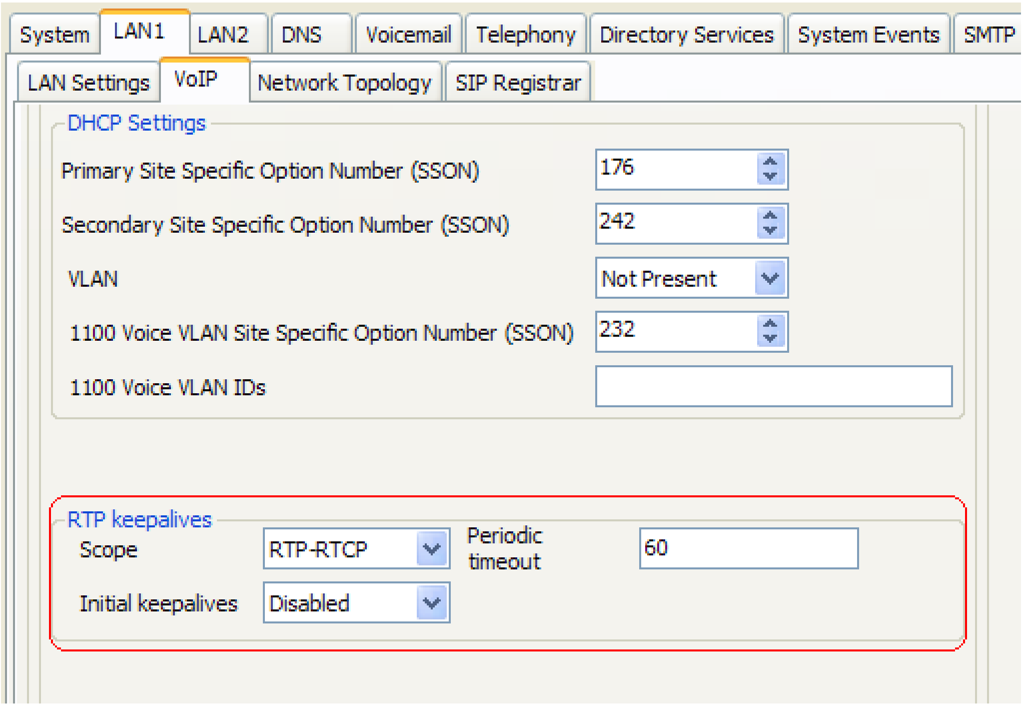
Video is now supported on SIP trunks. If STUN is enabled on the trunk, it will be applied to video data as well, if the addresses are not multicast.

6.10. Hosted NAT Traversal

Hosted NAT Traversal is used in order to avoid no speech path in forwarded or twinned calls through SIP trunks.

- If the call is coming from SIP and the destination is on SIP, it is possible that SIP Relay will be used (through the IP Office backplane). Depending on the Firewall configuration, this might result in no speech path (RTP packets will never be sent because the firewall is blocking every received packet).

- The Hosted NAT Traversal solution to this problem is implemented on Release 6.1. IP Office will generate UDP packets with no data using same source and destination address and port that the RTP stream would use. This method can be deployed in the initial packets and can be performed periodically. Both RTP and RTCP firewall holes can be created this way.
- See in the diagram below System/LAN1/VoIP/RTP keep-alives section. Some providers may have issues with these types of packets; this feature should only be applied after testing.



6.11. SIP STUN Changes

The Release 6.1 Manager will now have a blank entry in: *System / LAN1/2 / Network Topology / STUN Server IP Address*. It is recommended that if a STUN Server is required, it is advisable to consult with the SIP Trunk provider for a recommended STUN Server.

7. SeCom Retirement Home Integration Enhancements (Europe)

Secom is an Avaya solution for Retirement Home solutions in Europe. It is not available outside of the European Market. For more details on this application, please visit www.avaya.com or consult your sales channel. To support the SeCom solution, two new features are enabled:

Auto Delete of Call Log entries
In certain circumstances entries in call logs present a security, or breach of privacy, risk. In meeting or consulting rooms for example. IP Office allows deletion of call log entries from a User's centralised call log after a timeout, configured in Manager. Only the User's call log entries will be deleted, not any hunt group entries – therefore the User should not be configured to have any groups in their call log. The feature will only apply to the centralised call log and therefore only the telephones that it

supports; 1600 Series, 1400 series, 9600 series on IP500v2. Centralised call log is also used by one-X Portal for IP Office.

Shortcode to set/clear User's MWI from another port
IP500v2 only The ability for the SeCom server to set and clear Message Waiting Indication and Do Not Disturb on a User's telephone

7.1. Set a User's DND from a Shortcode on another telephone

IP500v2 As above, there will be a Shortcode feature to enable/disable DND from another telephone or interface.

7.2. Restrict Ring Voltage on Analogue Extension Ports

IP500v2 - A system wide configuration entry to restrict the ring voltage on analogue extension ports on the system to a maximum of 40V Peak-Peak as required by safety regulations. When set, Manager will restrict the selection of MWI types to None and the Line Reversal options, any ports set for High Voltage (81V or 101V), or 51V stepped MWI will be forced to Line Reversal A. This is required to be supported in all the European locals, specifically Germany, Switzerland, and Norway. The configuration item is a check-box "Restrict Analogue Extension Ringer Voltage" default un-checked in the System -> Telephony -> Analogue Extensions.

8. IP Office Voicemail Pro (Preferred Edition)

IP Office Only

8.1. Voicemail Pro on Linux

The IP Office Voicemail Pro Server (Preferred Edition) is now available on Linux as part of the new IP Office Applications Server DVD. However, please note that the Linux version does not support the following capabilities in R6.1:

- TTS
- VB-Scripting
- Use of the MAPI Interface/Exchange Integration:
- Web Voicemail: Expect access via one-X Portal
- VPNM
- 3rd Party Dbase

The VmPro Client is however only available to run on a Windows O/S, though can be used to Administer the Voicemail Pro Server whether it is running on Windows or Linux.

Note: The Voicemail Pro Client login credentials to a Linux based Voicemail Pro is:

User name: *Administrator*   Password: *Administrator*

The Linux based Voicemail Pro Server will provision an Apache based Web Services interface, whilst the Windows based server will continue to utilize IIS Web Services.

8.2. Improved Data Throughput

Further to the release of Backup and Distributed Voicemail Pro configurations in Release 6, distributed to primary Voicemail Pro data throughput has been improved to reduce latency from a Message being left at distributed to user availability via the (Primary (Active) Voicemail Pro and Msg waiting indication on the telephone.

Just as in IP Office R6.0, a multi-site network (SCN) of IP Office PBXs can be configured to use a single centralized Voicemail Pro server. This centralized Voicemail Pro server provides the media store for all of the voicemails left for any of the users across the multi-site network, as well as for announcements, auto-attendants, campaigns, and recorded calls. To avoid a single-point of failure scenario, a backup Voicemail Pro server can be put into place.

Distributed Voice Messaging
To take advantage of this feature, Voicemail Pro servers can be put into place at multiple sites across the multi-site network. Not all sites need to have a Voicemail Pro server. For sites that do have a local Voicemail Pro server, the server's resources will be used to increase performance, reduce WAN traffic, and to provide an extra level of resiliency beyond having a centralized backup Voicemail Pro server.

The Voice Ports and Text-to-Speech Ports of the local Voicemail Pro server at a site are used to service the needs of users local to that site. When the multi-site network is healthy and access to the central Voicemail Pro server is active, all voicemails are sent to the central site, so users can access them from anywhere. During a temporary break in communications with the central Voicemail Pro site, the local Voicemail Pro server remains available to provide announcements (for queues, still queued, and conferences), auto attendants, call recording, campaigns, and the recording of voicemails. However, the retrieval of voicemails will require access to the centralized Voicemail Pro server.

Note: All Voicemail Pro servers on the multi-site network (whether centralized, distributed or back up) will require a valid Preferred Edition (Voicemail Pro) license.

Distributed and Backup Voicemail pro configuration utilises SMTP to perform the inter-server synchronisation of configuration and voice recordings between servers.

8.3. Voicemail Pro Operating System Mix

- The Operating System type of Voicemail Pro used as the Backup server, must be the same as that used as on the Primary Voicemail Pro. That is if the Primary Voicemail Pro is Windows based, the Backup VmPro must be a Windows based also.
- If the Primary VmPro is Linux based, the Backup Voicemail Pro is to be Linux based also.
- Backup data from a Windows VmPro can not be retrieved by a Linux VmPro.
- Backup data from a Linux VmPro can not be retrieved by a Windows VmPro.
- Distributed Voicemail Pro and Primary Voicemail Pro servers can be any mix of Windows or Linux.

8.4. Email Settings in the Voicemail Pro Client

Email Settings are now configured in the 6.1 Voicemail Pro Client rather than the previous Windows versions in Control Panel. The Client will display these settings under the “Administration/Preferences/Email section – and will show three tabs:

- **MAPI** - There is no change to these settings from previous versions.

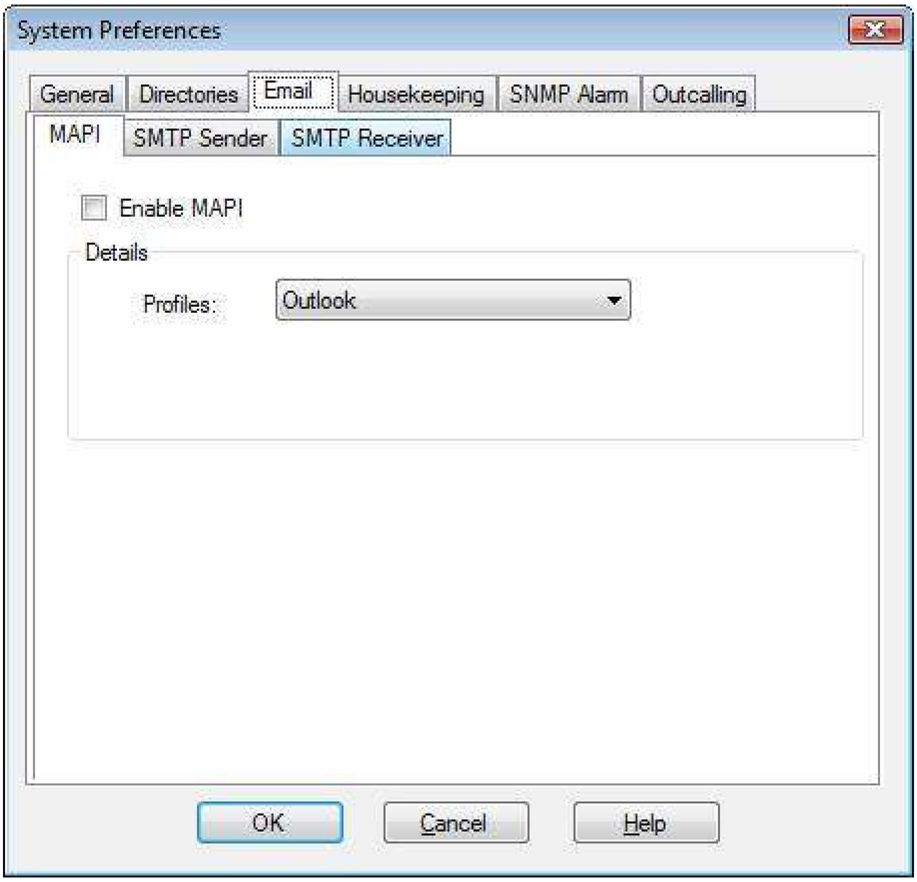
- **SMTP Sender** - Enter the required settings in here – on an upgrade, they should be ported from the Voicemail settings in Control Panel

- **SMTP Receiver** - Important – You have two SMTP Receiver Settings in a drop down list – “Internal” or External”.
  - On a Windows Voicemail Pro – ensure that the SMTP Entry is set to “External” - This will use the C;\Inetpub\mailroot\Drop folder. It may default to the Internal setting on an upgrade which would inhibit email functions and Distributed/Backup Voicemail would not synchronise between the VMPro Servers.
  - On the Linux (CentOS) server, use the “Internal” setting.

There are three pages for configuration:

MAPI Support.
This is used to specify the profile that is to be used by the Voicemail Pro server for connecting to Mail using Extended MAPI. Generally this will be an exchange server. The profiles are obtained from the Voicemail Pro server and will also be used to determine the name of the default mailbox to use and the Exchange Server that is being used. This page is also used to indicate that mail will be sent using Extended MAPI by default.

If “Enable MAPI” is checked then Voicemail Pro will send email using Extended MAPI. Of course VPNM/VPIM and synchronisation messages are still sent using SMTP.
This dialogue will list all the MAPI profiles configured on the Voicemail Pro server for use by the Voicemail Pro server allowing the administrator to select the profile that the Voicemail Pro server should use for exchange integration support.



SMTP Sender.

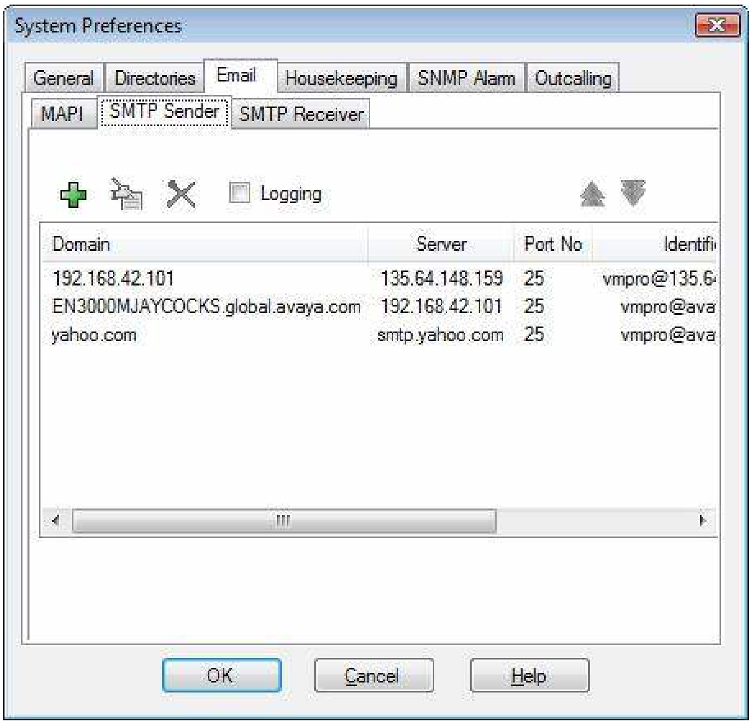
This dialogue allows the configuration for the transmission of SMTP messages. If the logging option is also enabled then transmission logs are created and stored within the Voicemail Pro logs directory.

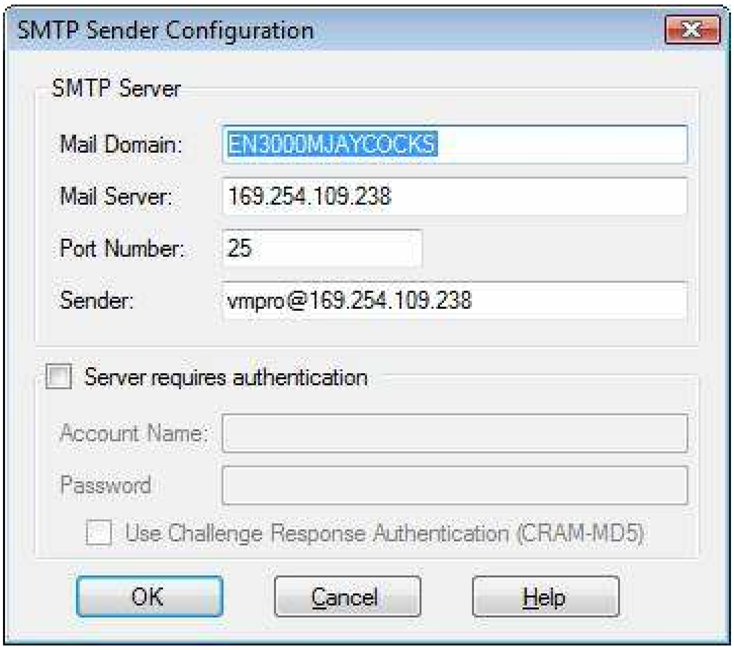
The list defines the parameters required to transmit SMTP mail. The first entry defines the system wide default parameters, whilst subsequent entries allow configuration for specific mail domains. For example, it is possible for default messages to be targeted at an Exchange Server, but then specify settings for sending messages to domains such as yahoo.com, hotmail.com (etc).

Each entry has the following parameters:

- Domain - this is usually the server name itself, you may need to enter the Fully Qualified Domain Name which would be the Computer Name & primary DNS suffix i.e. EN3000IPO3.global.domain.com - or just its own IP Address will be sufficient – For the first entry this specifies the mail domain that Voicemail Pro is using. It will also specify what mail domain Voicemail Pro will accept messages for. For subsequent entries this specifies that for recipients of that mail domain, Voicemail Pro will use those settings for transmitting the message to that user. This entry will need to be the fully qualified name; if DNS is not working then the IP address should be used.
- Mail Server - This specifies the address or fully qualified DNS name of the SMTP server to connect to for sending the message. This could be an Exchange Server or an internet based SMTP service.
- Port No - This specifies the port on that server to connect to. This is nominally 25.

- Identifier - This is important for mail servers that will only accept email from valid accounts. If there is no such restriction it can be blank.
- Auth, User, Password - If the mail server requires authentication then this should be completed accordingly.



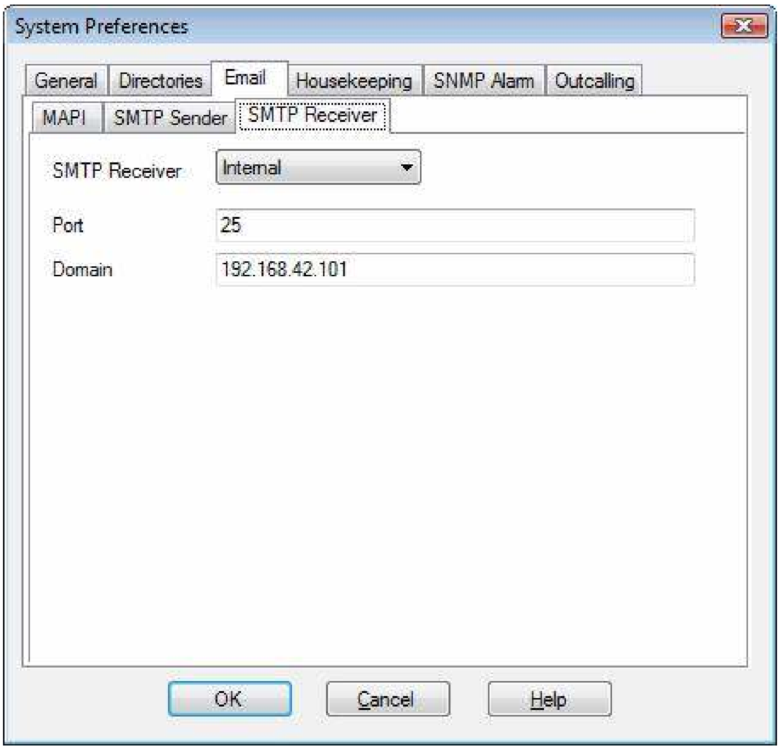


SMTP Receiver

The SMTP Receiver specifies how Voicemail Pro will receive email. It will use either an internal SMTP server or allow use of an SMTP server external to Voicemail Pro (e.g. IIS (SMTP)) where the location for the messages can be retrieved.

For an internal SMTP server the following is required:

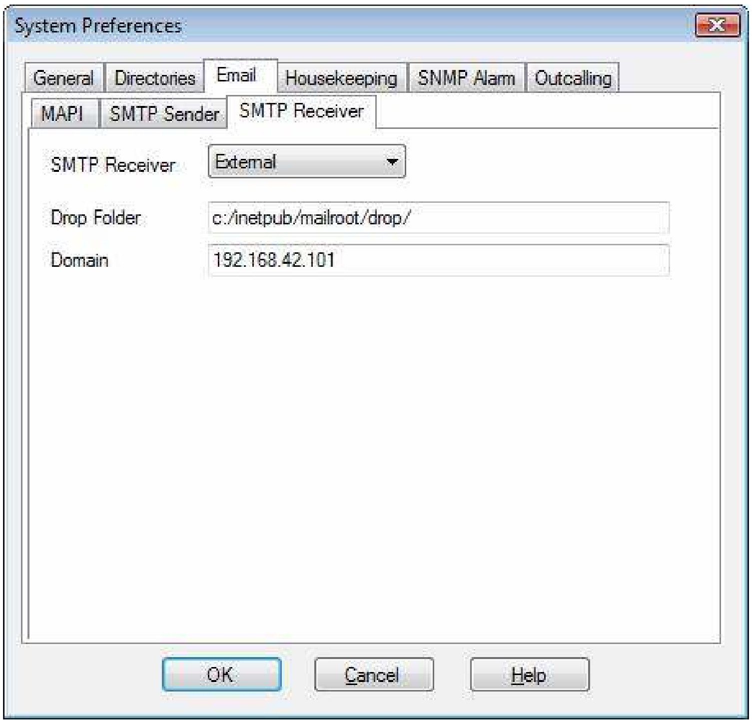
- The port that Voicemail Pro will listen on for SMTP mail. By default this is 25.
- The mail domain (Name or IP Address) that Voicemail will accept messages for. All other messages are ignored.



This is usually used on the Linux variant of Voicemail Pro where there are no onboard IIS Services (like a Windows Server).

For an external SMTP server (Typically on a Windows Server) the following is required:

- The location that Voicemail Pro will scan for incoming messages.
- The mail domain that Voicemail will accept messages for. All other messages are ignored.



This is usually used if the VMPro is a Windows Server with its own IIS Services – inetpub/mailroot/drop

8.5. Known Issues in SMTP and MAPI Configuration

- **SMTP messages via Exchange and workaround:**

For SMTP messages, the VMPRO is now using the binary mime format as it allows the Voicemail Pro server to send larger messages which enhances the speed between a Primary and a Distributed or Primary and Backup server. Exchange servers support binary mime; but some of the major internet suppliers of email services do not. These services will usually utilise mine64 format for SMTP messages. Exchange does not provide a conversion mechanism from binary mime to mime64 encoding therefore it cannot forward the message on in mime64 if it receives a rejection from the internet email service. However, if VMPRO connects directly to an SMTP server that does not support binary mime it will fall back to it mime64 which is supported by default on these SMTP servers and successfully transmit the message.

You can configure a workaround for this issue configuring another external SMTP server in the 'SMTP Sender' form – this fixes the problem. This issue is being investigated and will be resolved in a future release.

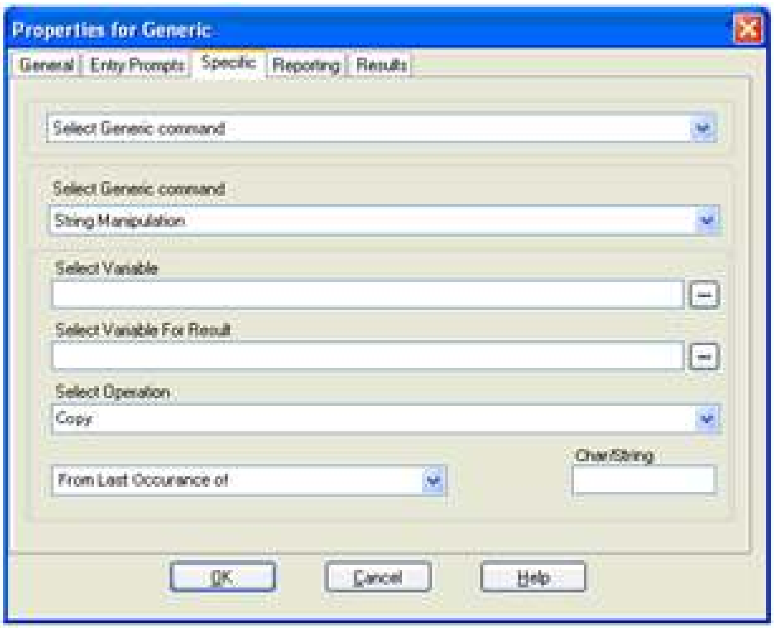
- **SMTP Sender - entries not saved and workaround**

If the MAPI configuration is not valid or is utilising a non MAPI email client (Outlook Express) – you may find that the email settings in the Voicemail Pro Client (Administration/System preferences/Email) are slow to open and may not save new settings entered in the SMTP Sender section. This is because the Voicemail Pro cannot resolve MAPI. Please ensure the MAPI settings are correct and this will not occur. This will be improved in a future revision of Voicemail Pro

8.6. Generic Action Enhancements

The ‘Generic Action’ has been enhanced to accommodate basic string manipulation functions that will allow:

- The length of a string
- The length of a string to a char/char sequence
- The length of a string from a char/char sequence
- Copying of a string to a char/char sequence
- Copying of a string from a char/char sequence



These enhancements provide string manipulation functions that can be used via standard call-flow elements. The Voicemail Pro can be supplied multiple elements in a data string, from which the individual items can be extracted using the ‘Generic’ action, without the need to create bespoke software solutions or use complex scripting languages such as VB.

8.7. Complete Sequence Function

Complete Sequence is the ability for the Voicemail Pro to complete call-flow tasks once a call has been terminated has been introduced in this release. The capability will allow call independent tasks, such as the sending of multiple emails, and data manipulation tasks and database read/write tasks to continue to completion, or till a configured timeout expires, even after the incoming call terminates.

The Complete Sequence checkbox is found on the Specific Tab of the Start Point Action. There is a Timeout Setting in seconds to allow for tasks such as database interactions to finish before completion of the callflow.

8.8. Automatic Call Recording for Internal Calls

Voicemail Pro now has the ability to record internal calls as well as external. The user and hunt group options for call recording can now be set to work for internal calls. Previously they only applied to external calls.

8.9. Command Line Options

The Command Line Options will allow another application such as the IP Office Manager to invoke a Voicemail Pro Client that can connect to a VmPro Server selected by the invoking application using the User and Password criteria provided.

Note: *See the new enhancements in IP Office Manager for more information on this feature*

8.10. Voicemail Pro Manuals and Guides

Please review the following manuals and guides for more information on Voicemail Pro - Preferred Edition:

IP Office Release 6.1- Voicemail pro Installation and Maintenance Manual
IP Office Release 6.1 – IP Office Application Server Installation and Maintenance Manual

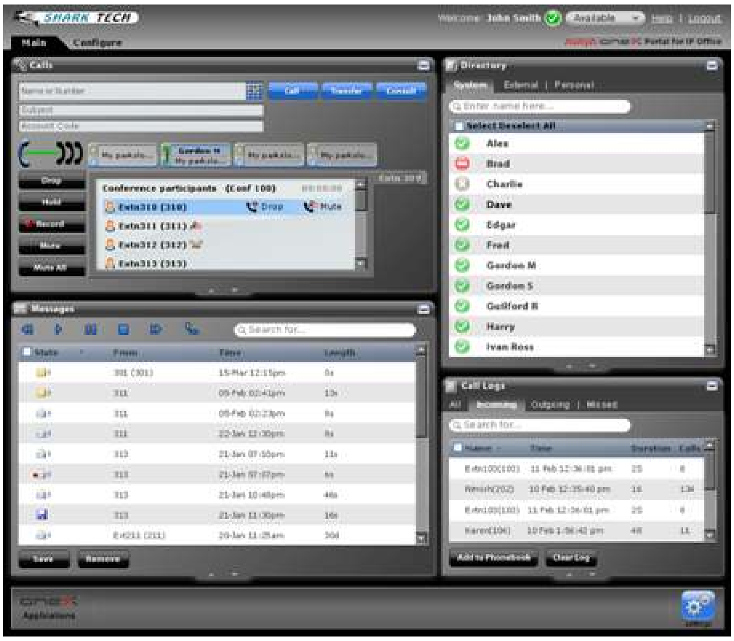
9. IP Office one-X Portal for IP Office

9.1. IP Office one-X Portal - New Look and Feel

There is a new look and feel for the one-X Portal for IP Office client. Rather than the 50/50 layout for gadgets, which was used in Release 6.0, the gadgets are now displayed in the Web browser with a 60/40 ratio, and they can be resized, moved (drag and move), minimized, and restored.

one-X Portal Server on Windows - Minimum PC Specification

RAM	2 GB
Hard Disk	10 GB
Operating System	Windows 2003 or Windows 2008 (32 bit and 64 bit)



9.2. Show/Hide Gadgets

You can minimise any of the Gadgets sown on the Main tab by selecting the minimise icon in the top right of the gadget. The gadget is reduced in size to just its title bar

9.3. New Skins

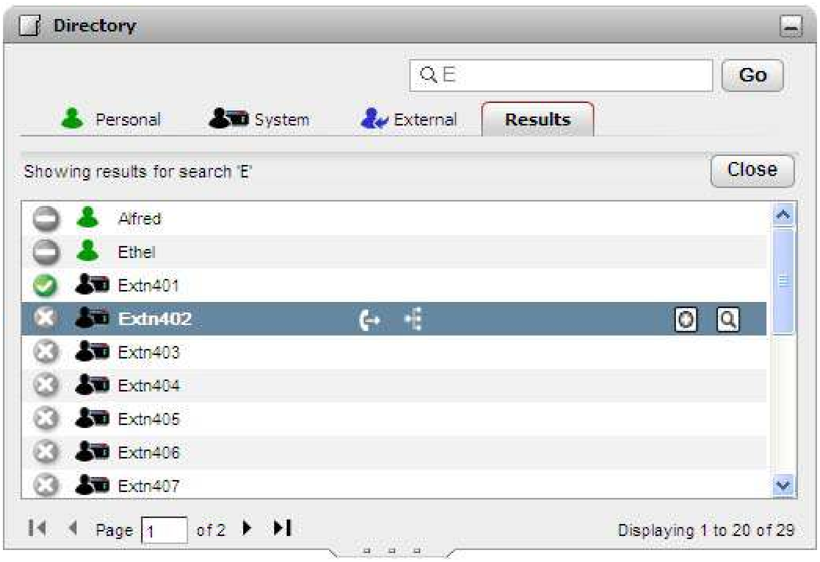
In the Settings area of the new one-X Portal client, the user can click on one of the skin icons to select it and apply it to the one-X Portal for IP Office. The user can click on the Reset to Factory icon and Click on the Layouts icons to select the ratio of columns widths it represents.

9.4. Instant Message Chat

one-X Portal for IP Office allows the user to have instant message chat sessions with other one-X Portal for IP Office users. Select another user in the directory gadget with the chat icon. You can only do this if they have one-X Portal for IP Office client running on their computer. You can use an instant message session even when on a call to the same user that you are messaging.

9.5. Multi-Directory Search

One-X Portal now allows a multi-directory search to allow the user to search all the directories. The User enters the search name into the search box at the top of the directory gadget and clicks to “GO” button. The results are shown on the Results tab.



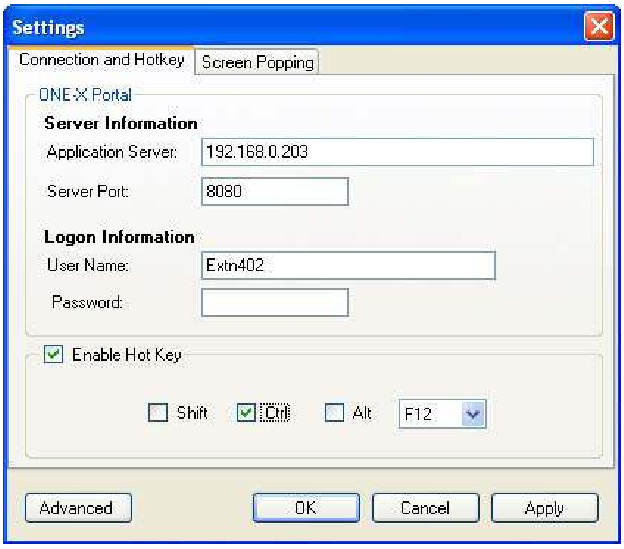
9.6. one-X Portal® for IP Office - Call Assistant

one-X Portal® for IP Office supports a Call Assistant Plug-In which is a desktop application that installs on the user's PC. Call Assistant can be used to answer calls, drop calls, launch the one-X Portal® for IP Office browser, and configure screen options

Installation

One-X Call Assistant can be installed directly from the one-X Portal Client – check on the client "Configuration" Tab and scroll down to the Avaya Call Assistant section

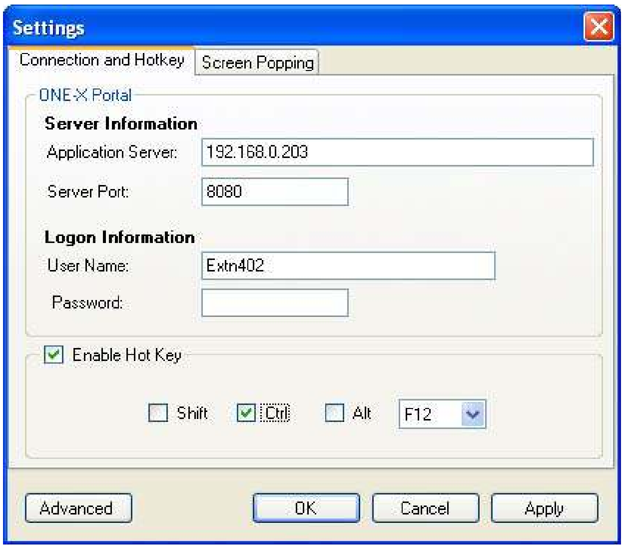
When installed, the one-X Call Assistant displays its settings page:



The **Server Information** fields will have been completed using the information entered during the software installation. Check that this is correct.

- In the **Logon Information** section enter the user name and password that you use to login to one-X Portal for IP Office.
- Click **OK**.

The one-X Call Assistant icon should appear in the Windows task bar and will show a message when it connects to one-X Portal for IP Office. If the icon remains with a red dot instead of green dot then it has failed to connect to one-X Portal for IP Office. You will need to check the connection settings you entered by right clicking on the icon and selecting **Settings**.



Server Information
This information should be supplied by your system administrator. In most cases it will match the web address you use in your web browser to access one-X Portal for IP Office.

Application Server
This will match the part of your one-X Portal for IP Office web address between the **//** and **:** characters.

Server Port
This will match the number shown in your one-X Portal for IP Office web address after the **:** character. The normal default is ***8080***.

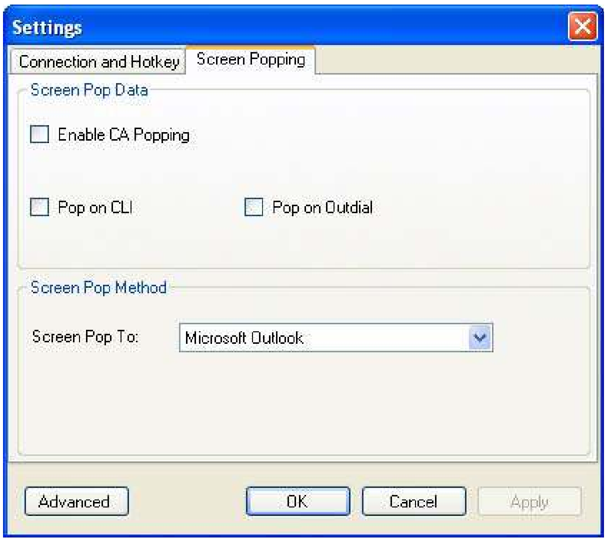
Logon Information
This information will match the user name and password that you use to login to one-X Portal for IP Office.

User Name
Enter the user name that you use to login to one-X Portal for IP Office.

Password
Enter the password that you use to login to one-X Portal for IP Office.

Enable Hot Key
You can use these options to use one-X Call Assistant to make calls.

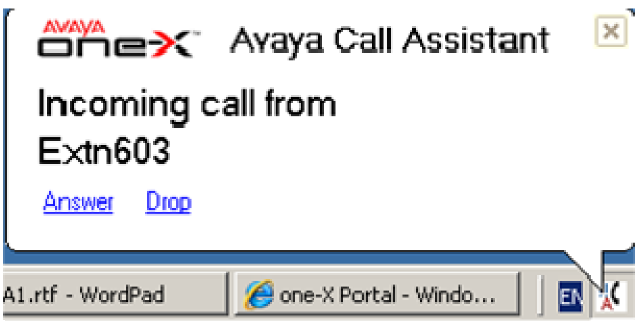
Screen Popping
These options are used to set when the one-X Call Assistant should user screen popping and which application it should pop.



Advanced
The advanced button is used to access settings that are not normally needed by most users.

- **Local IP address:** If you're PC supports more than one network, it may be necessary to specify which network address the one-X Call Assistant should use.

The one-X Call Assistant can be used to display information about calls. This information appears as a popup message which includes links for actions that you can perform (answer the call, drop the call, etc).



The one-X Call Assistant can also pass attempt to match call information to contacts in your Microsoft Outlook 2003 or 2007 and display the matching contact. This Outlook screen popping can be done on the number received with incoming calls and or on the number you dial when making a call. You need to have Outlook running for this feature to work.

Hot Key Dialling

- The one-X Call Assistant can be used to make a call by dialling a number selected in a Windows application. To do this, simply select or highlight the number in the Windows application and then press your one-X Call Assistant's hot key combination.

- The number selected must be suitable for dialling. For example you telephone system uses an external dialling prefix that must be present in the number. It is not possible to edit the number before the one-X Call Assistant attempts to dial it.

- It is not possible to predicate which hot key combination will always work. There will be scenarios where the hot key combination you use for one-X Call Assistant will match one used by the application containing the number that you wish to dial. In that case the application function may take precedence over one-X Call Assistant dialling.

9.7. one-X Portal for IP Office – Linux

As part of the IP Office Application Server DVD, the one-X Portal Server can now also be installed on the CentOS 5.5 Linux operating system as well as Windows servers. Functionality is the same on either operating system with no loss of features.

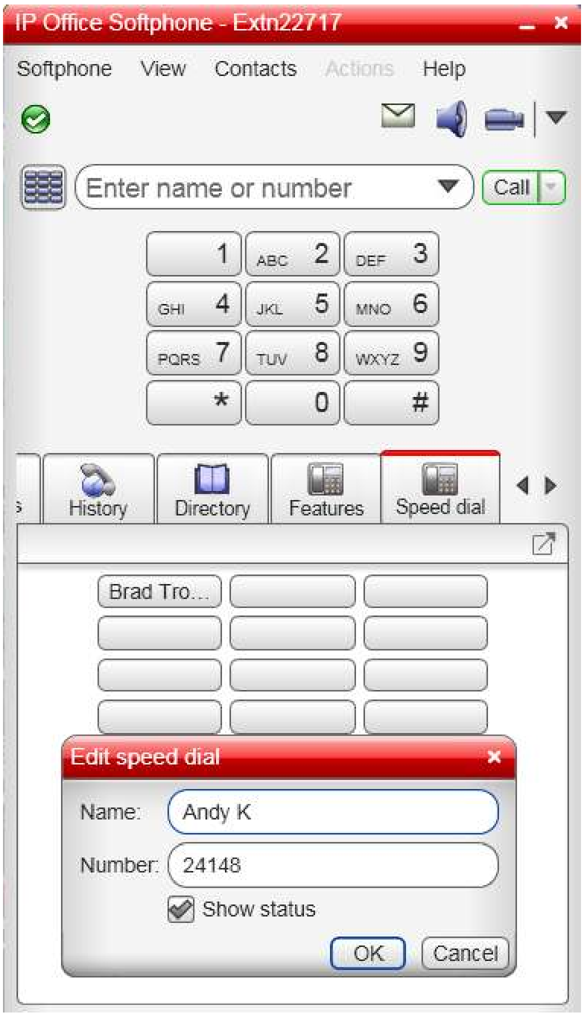
Please review the following manuals for complete installation and configuration information:

IP Office 6.1 – one-X Portal for IP Office Installation Manual
IP Office 6.1 – one-X Portal for IP Office User Guide
IP Office 6.1 – IP Office Application Server Installation and Maintenance Manual

10. IP Office Softphone

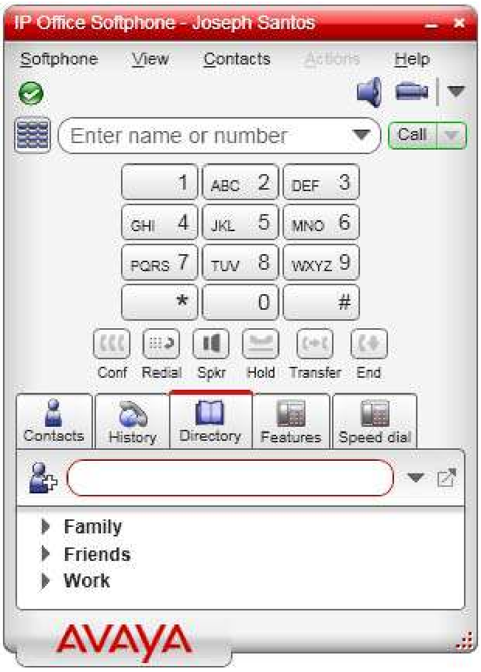
IP Office Video Softphone has been enhanced to provide:

- BLF with Pickup feature – The Speed Dial feature allows the user to configure extension numbers onto the speed dial buttons. If these entries match Users in the IP Office configuration the button will display Busy/Idle indication on that button. The button when selected can be used to call the programmed destination.



- The same BLF button can be used to pickup a ringing call from that programmed User.

- Do Not Disturb – This is enabled/disabled by use of an icon displayed in the top left of Softphone – green tick icon top right of the above screenshot.
- Updated HD Video Codec – provides better video quality and compatibility with other SIP video conferencing telephones.

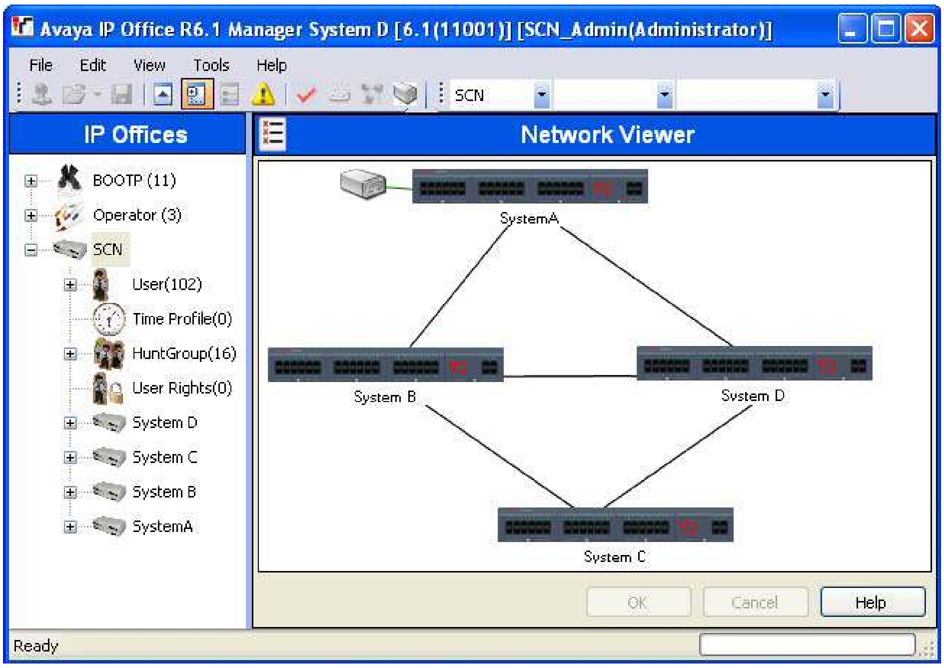


IP Office Softphone can be installed from the IP Office Administration Suite CD or download.

Please review the *IP Office Release 6.1 – IP Office Softphone User Guide* for installation and user guide information.

11. IP Office Manager – Small Community Network (SCN) Management

IP Office Manager 8.1 and higher supports the ability to load and manage the configurations of the IP Office systems in a SCN at the same time. This requires IP Office Manager to be enabled for SCN discovery and at least one IP Office in the SCN to have Release 6 or higher software.



When the configurations of the systems in an SCN are loaded, Manager switched to SCN management mode. This differs from normal IP Office configuration mode in a number of ways:

- A network viewer is available. In addition to giving a graphical view of the SCN, the view can be used to add and remove links between the systems in the SCN.
- In the configuration tree, the entries for users and hunt groups on all systems are grouped together.
- Time Profiles and User Right common to all systems are grouped together.
- The configuration settings for each system in the SCN can be accessed and edited.

11.1. Enabling SCN Discovery

In order for the **Select IP Office** menu to groups systems in an SCN and allow loading of all the SCN configurations, Manager must be enabled for SCN discovery.

1. Select **File | Preferences**
2. Select the **Discovery** tab
3. Select the **SCN Discovery** option
4. Check that the other discovery setting are sufficient to discover all the IP Office systems in the SCN
5. Click **OK**.

11.2. Creating a Common Admin Account

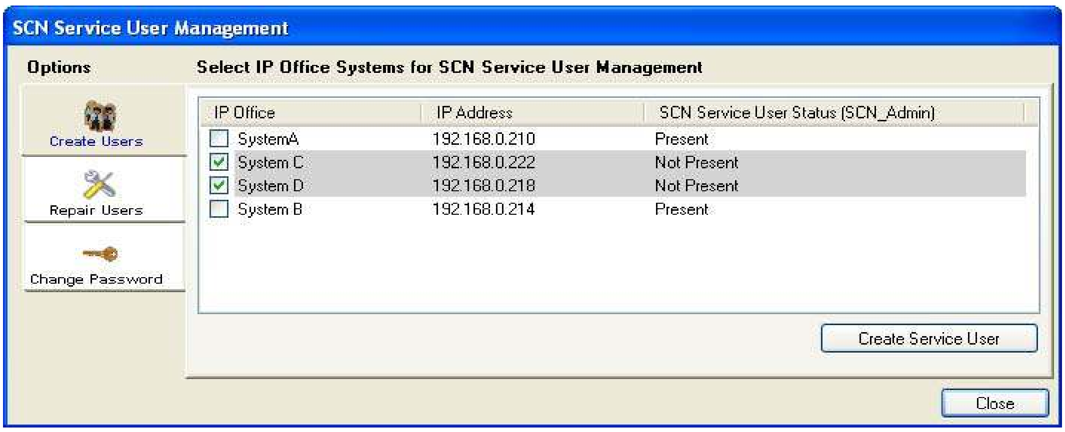
When managing multiple systems, it may be useful to create a common user name and password on all the systems for configuration access. This tool can be used to create a new service user account, **SCN_Admin**, for configuration access. This process requires you to have a user name and password for security configuration access to each of the IP Office systems.

1. Select **Tools | SCN Service User Management**. The option is not shown if an IP Office Essential Edition - PARTNER® Version or IP Office Essential Edition - Norstar Version system configuration is loaded. If no configuration is loaded, and the option is not shown, select **View | Advanced View**.

2. The **Select IP Office** menu will display the list of discoverable IP Office systems.

3. Select the systems for which you want to create a common configuration account. Click **OK**. A user name and password for security configuration access to each system is request. Enter the values and click **OK**. If the same values can be used for all systems enter those values, select Use above credentials for all remaining, selected IPOs. If each system requires a different security user names and password, deselect Use above credentials for all remaining, selected IPOs

4. The systems will be listed and whether they already have an SCN_Admin account is shown.



5. To create the **SCN_Admin** account on each system and set the password for those account click on **Create Service User**.

6. Enter the common password and click **OK**.

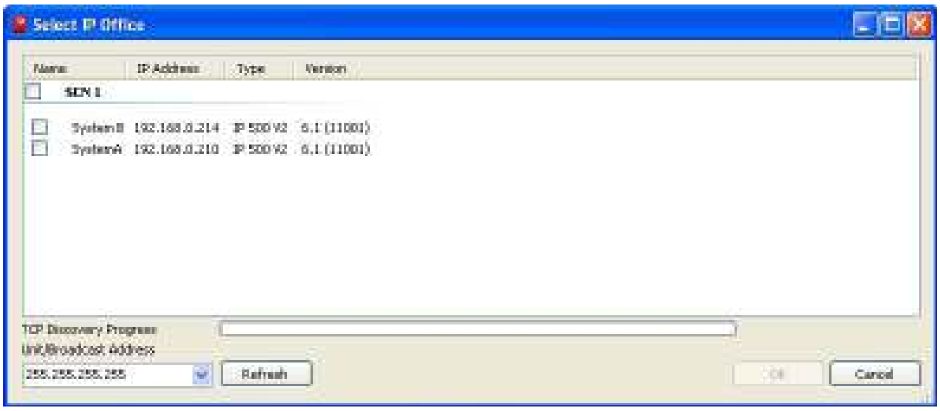
7. The password can be changed in future using the **Change Password** option.
8. Click **Close**

11.3. Loading an SCN Configuration

If Manager is configured with **SCN Discovery** enabled, the **Select IP Office** menu will display the SCN networks it discovers.

With no configuration loaded, click on or select **File | Open Configuration**

The **Select IP Office** menu is displayed. Any systems in an SCN will be grouped together.



To load the configuration of all the systems in the SCN, click the check box next to the SCN name and then click **OK**

.

If a warning icon is displayed next to the **SCN** check box, it indicates that not all the IP Office systems known to be in the SCN were discovered. Hovering the cursor over the icon will display details of the missing systems. Loading the SCN configuration at this time would not include the configuration of the missing system or systems. The missing systems:

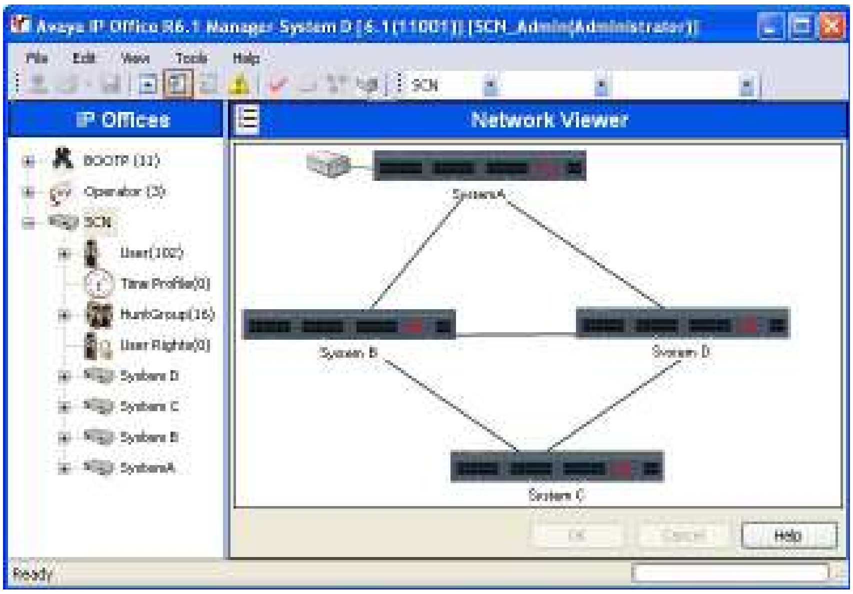
- May be disconnected
- The discovery settings for the Manager PC may be incorrect.
- The data routing between the Manager PC and the missing IP Offices may be incorrect or blocked

Enter the name and password for configuration access to each system. If the systems all have a common user name and password (see Common Administrator Access below), select **Use above credentials for all remaining selected IPOs**. Click **OK**.

Manager will load and display the combined configurations in SCN Management mode.

11.4. Editing an SCN Configuration

When the configuration of an SCN is loaded, Manager displays the configuration in a different way from when the configuration of a single IP Office system is loaded. The main differences are in how configuration entries are grouped in the configuration tree.



Clicking on the SCN icon displays the Network Viewer which shows the lines between the IP Offices in the SCN.

SCN Configuration Entries

Certain entries from each of the systems in the SCN are grouped together in the configuration tree differently from when just a single system configuration is loaded. There are two types, unique SCN entries and shared SCN entries:

.

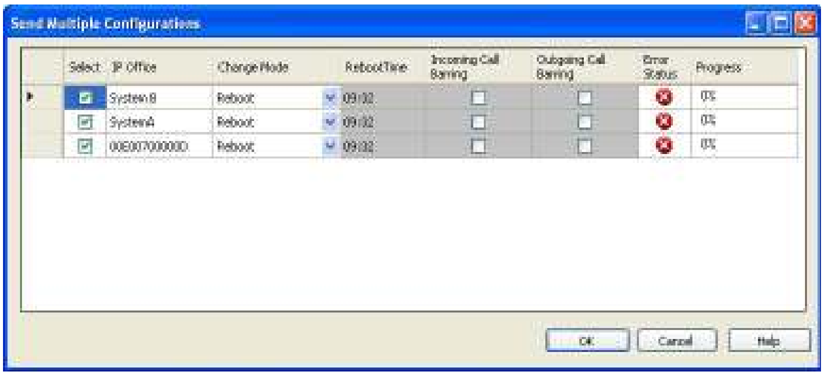
- **Unique Entries**
  They can be edited here and the system to which they belong is indicated in the group pane and in the title bar of the details pane. However, to add or delete these types of entry must be done within the configuration entries of the particular system that will host the entry's configuration details:
  All users in the SCN are shown under the **User** icon.
  All hunt groups in the SCN are shown under the **Hunt Group** icon.

- **Shared Entries**
  Shared entries are configuration items that exist on all systems in the SCN, having the same name and settings on each system. Editing the shared entry

updates the matching copy in the configuration of each system. Similarly, adding or deleting a shared entry adds or deletes from the individual system configurations. If the copy of the shared entry within an individual configuration is edited, it is no longer a shared entry for the SCN though the individual entries on other system will remain. Changing the individual entries back to matching will turn the entries back into a shared entry:
Shared time profiles are shown under the **Time Profile** icon.
Shared user rights are shown under the **User Rights** icon.

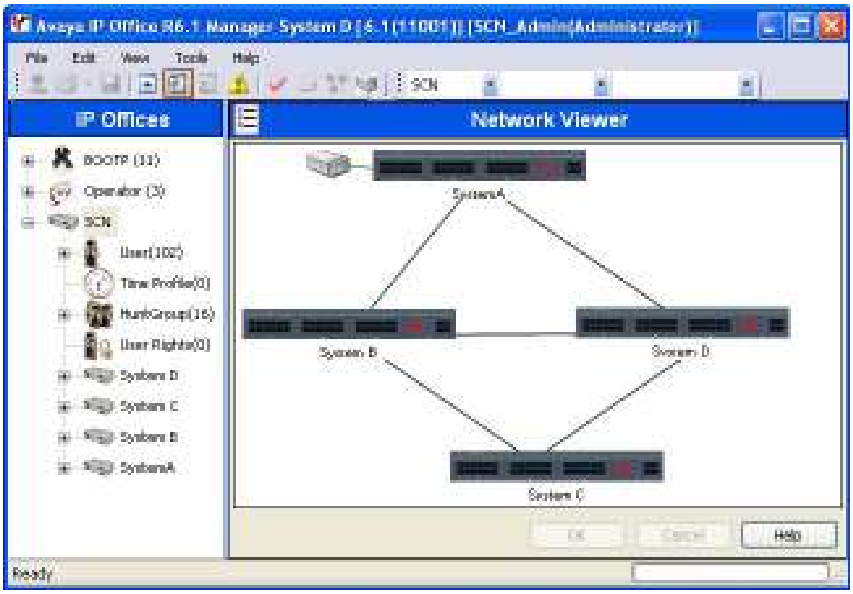
Individual System Configurations
The full configuration for each system in the SCN can be accessed and edited as required. It is possible to copy and paste configuration entries between systems using the configuration tree.

Saving Changes
When the save icon or **File | Save Configuration** is selected, the menu for multiple configuration saves is displayed. It provides similar options are for a normal single configuration save. Note that when working in SCN Management mode, after saving configuration changes the Manager will always close the displayed configuration.



11.5. Using the Network Viewer

Clicking on SCN in the configuration tree displays the Network Viewer. This shows each of the IP Office systems in the SCN and the links between each of the systems. Systems with attached Voicemail Pro servers are also indicated.



Green
IP Office with Voicemail Pro system.
Black
SCN line between two IP Office systems.
Red
Incorrect SCN line between systems (probably one-way connection). Right-click on the line and select **Repair**.

You can use the Network Viewer to perform a range of functions:

- Arrange the View
- Launch System Status
- Launch Voicemail Pro
- Add an SCN Line
- Add an IP Office
- Remove an SCN Line
- Remove an IP Office from the SCN
- Repairing an SCN Line
- Add a Background Image

Adding a line within the SCN
By right clicking on the starting IP office for the link, then select **Connect To** your can use the network viewer to add an SCN link between two IP Offices in the SCN that are currently linked. This process will add new H323 SCN line entries to the configurations of each of the IP Offices.

Start System Status
If the System Status Application is also installed on the Manager PC, you can start it for a particular system.

1. Right click on the system and select System Status.
2. The application is started and the login form pre filled with the IP address of the system.

Start Voicemail Pro Client
If the Voicemail Pro client is also installed on the Manager PC, you can start it for the any system with an associated Voicemail Pro server. Right click on the voicemail server icon and select Launch VMPro Client.

Add a Background Image
You can select an image file to be displayed in the background of the Network Viewer display. This file is not saved as part of the configuration in any way, ie. If the image file is moved or deleted it is not longer used by IP Office Manager.

1. Right click on the general background area of the network viewer and select Background Image.
2. Select Set Background Image to browse to the location of the file to be used.
3. The Visible option can be used to switch the display of the background image on or off.

11.6. System Inventory

When working in SCN Management mode, clicking on the System icon for a particular system displays a system inventory page for that system.

For more information on IP Office SCN Management please refer to the:
IP Office Release 6.1 – Manager 8.1 Manual

12. Customer Call Reporter 6.1 (IP Office Advanced Edition)

CCR Windows Server PC Requirements:

	Minimum Specification
Processor	Intel Pentium D945 Dual Core or AMD Athlon 64 4000+.
RAM Memory	2 GB (4 GB recommended)
Hard Disk Space	30 GB
Operating System	Windows 2003 SP2 Server, 2003 R2 Server, Small Business Server 2003 R2, 2008 Server & 2008 Server R2 (32-bit and 64-bit)

12.1. Microsoft SQL 2008 Support

IP Office Customer Call Reporter is now supported using Microsoft SQL 2008.

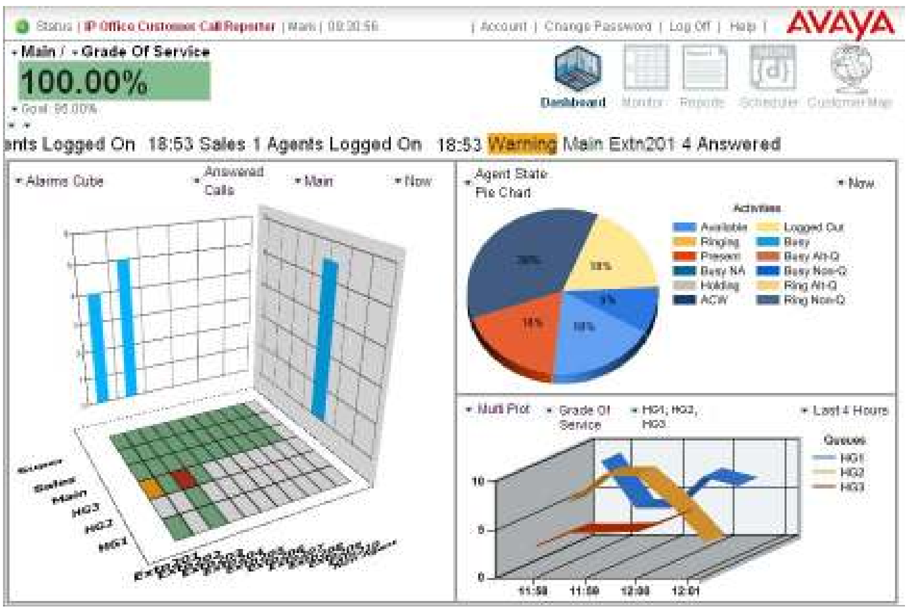
12.2. Visual Redesign

Some elements of the browser display have been redesigned.

- Many of the tabs used by Supervisors to move between the available pages have been replaced by icons.
- The colour used for warnings has been changed from yellow to orange.

12.3. Dashboard Display

For a supervisor account, this is the default page shown after logging in. It consists of a number of adjust information elements; a dashboard goal, a statistics information ticker and a set of graph display panels. The main part of the dashboard is divided into a large display panel and two small display panels. Each panel has a header row of icons which can be clicked to select the type of item displayed in the panel and to then adjust its settings. You can move your mouse over each highlighted area to get more information.

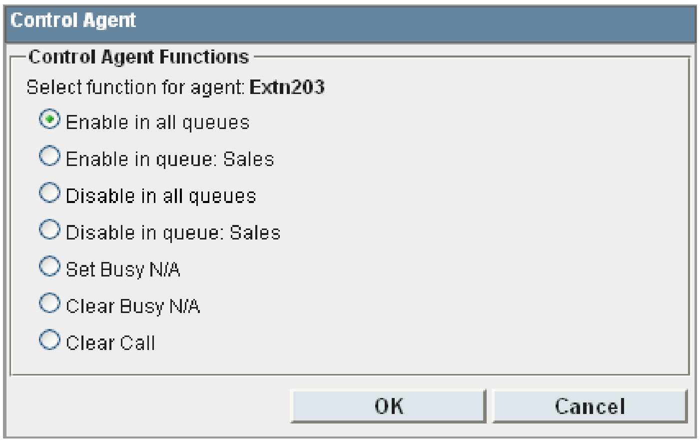


12.4. Force Agent State

Supervisors can now force a change to an agent's status. For example log an agent out or enable/disable an agent’s queue membership. The IP Office Customer Call Reporter administrator configures which supervisors have this function. This feature requires the IP Office Customer Call Reporter server to have access to a one-X Portal for IP Office server.

Supervisors with this option enabled are able to click on an agent name in a view and select from a list of actions that change the state of that agent, for example to force the agent to login or logout.

Note: This option requires the IP Office Customer Call Reporter to be configured with details of the one-X Portal for IP Office server.



A supervisor can set the state of an Agent telephone. This feature requires a connection via CCR Administration to a one-X Portal server (Windows or Linux) in order to enable agent states. Clicking on an Agent name displays a window with the following options:

- Enable in all queues - the agent is logged in and enabled in all queues to which they belong – SIP phones are not logged in automatically.
- Enable in queue - the agent (if already logged in) is enabled in the currently selected queue. Note this action does not log the agent in if currently logged out
- Disable in all queues - the agent's membership of all queues to which they belong is disabled and the agent is logged out
- Disable in queue - the agent's membership of the currently selected queue is disabled
- Set Busy N/A - the agent's status is set to busy not available with the reason code of automatic
- Clear Busy N/A - the agent is taken out of busy not available state
- Clear Call - Clears the agent's current connected call. It does not affect calls parked or held by the agent
- If the agent is in a conference call, it clears just the agent's connection to the conference

12.5. One-X Portal for IP Office Integration

The IP Office Customer Call Reporter option used by supervisors to control agent status uses the one-X Portal for IP Office to send commands to the IP Office system. To configure this requires access to the one-X Portal for IP Office administrator settings.

Enabling the one-X Portal for IP Office User Account

1. Login to the one-X Portal for IP Office using the administrator login.

2. Select **Configuration** and then **Users**.

3. Select one of the user records and click **Edit**. The user record used is unimportant other than use one that is less likely to be deleted as the result of user adds moves and changes. Typically use one of the IP Office Customer Call Reporter supervisors if they are also IP Office users.

4. 

5. Change the **User Role Configuration** from ***User*** to ***User Manager***.

6. Click **Save**.

7. Select the checkbox next to the user record and then click **Put Selected**.

Enter the one-X Portal for IP Office Account Details in IP Office Customer Call Reporter

1. Login to IP Office Customer Call Reporter as the administrator.

2. Select the **System Settings** tab and expand the **Preferences** section if necessary.

3. In the one-X Portal for IP Office section enter the address details of the one-X Portal for IP Office server and the name and password of the user enabled as a one-X Portal for IP Office ***User Manager*** in the one-X Portal for IP Office configuration.

Enabling a Supervisor
In order to use this feature, the IP Office Customer Call Reporter supervisor must have the **Control Agent** setting in their account details enabled.

12.6. New Statistics

There are some new real-time statistics added for Queues and Agents:

- Talk-time Total (external only)
- Talk-time Average (external only)
- Talk-time Inbound
- Talk-time Inbound average
- Talk-time Outbound
- Talk-time Outbound average
- Talk-time Internal
- Agent Productivity Factor: % of calls an agent has handled within service level prescribed

12.7. Report Templates

The following changes have been made for historical reporting.

- The **Call Summary Report** now includes **Average Answer Time** when reporting on agents (previously this value was blank when reporting on an agent or agents).

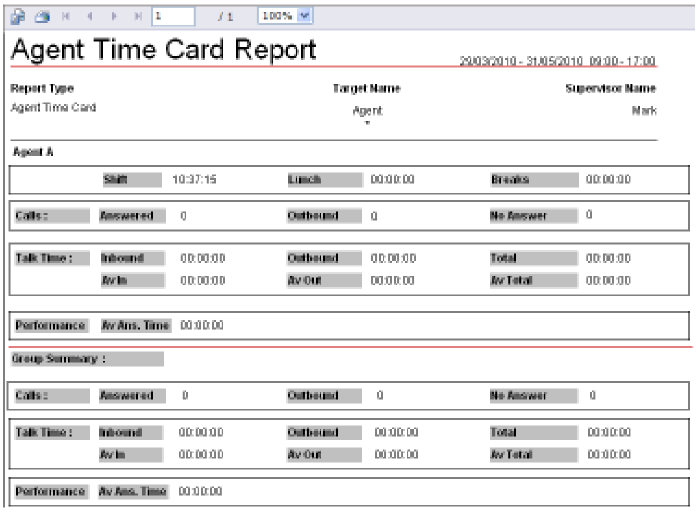
- A new template called the **Agent Report Card** template has been added. It provides historical reporting on the Talk Time statistics.

New Agent Time Card Report

This new report summarizes agent performance including:
- Agent Name
- Date
- Shift Start Time-End Time
- Total Hours Logged On
- Lunch
- Start-End of Busy
- Breaks
- Start-End time of all other Busy Not Available codes
- Calls Answered
- Outbound & No Answer

- Talk Time: Inbound, Outbound, Total, Average In, Average Out, Average Total
- Average Answer Time

Note: The report does not include internal calls.



Agent Time Card Report

29/03/2010 - 31/05/2010 09:00 - 17:00

Report Type	Target Name	Supervisor Name
Agent Time Card	Agent	Mark

Agent A

	Shift	10:37:15	Lunch	00:00:00	Breaks	00:00:00
Calls :	Answered	0	Outbound	0	No Answer	0
Talk Time :	Inbound	00:00:00	Outbound	00:00:00	Total	00:00:00
	Av In	00:00:00	Av Out	00:00:00	Av Total	00:00:00
Performance	Av Ans. Time	00:00:00				

Group Summary :

Calls :	Answered	0	Outbound	0	No Answer	0
Talk Time :	Inbound	00:00:00	Outbound	00:00:00	Total	00:00:00
	Av In	00:00:00	Av Out	00:00:00	Av Total	00:00:00
Performance	Av Ans. Time	00:00:00				

Average Call Answer Time added to Call Summary Report

This new statistic gives the sum of the answer time for queue calls answered divided by the number of queue calls answered.

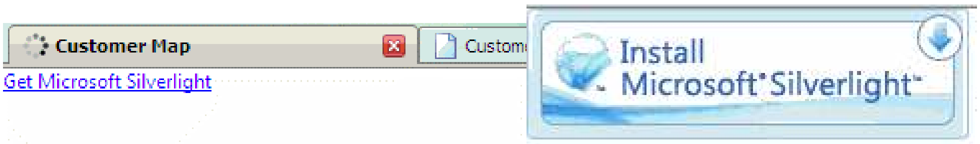
12.8. Message Color

When scheduling a wallboard message, supervisors can now select the color for the message.

12.9. Customer Map

Supervisors can display and configure a map that will plot calls based on caller ID numbers. The map can be used to show a combination of historical and real-time calls and can be overlaid onto Google or Yahoo maps.

The customer map shows the location of callers based on the caller's number. When a caller's location has been identified, a pin is placed on the map at that location. The color of the pin changes with the volume of calls that match that same location. When the caller cannot be resolved beyond just a country, a coloured button is used instead of a pin. Hovering your cursor over a pin or button displays details of the location and the number of callers from that location. The customer map can use a number of different map backgrounds. These vary by map provider and map type.



The IP Office Customer Call Reporter Customer Map function uses Silverlight, which must be manually installed. When logged in at a computer with Internet connectivity, click the Customer Map icon. If the computer does not have Silverlight installed, the user will be prompted to install it.

Once Silverlight is installed successfully you must restart the browser for the Plug-In to activate.   Login to CCR again, click the Customer Map icon

12.10. CCR Manuals and Guides

For more information on CCR, refer to the following manuals and guides:

IP Office – Customer Call Reporter 6.1 Installation
IP Office – Customer Call Reporter 6.1 User Guide
IP Office 6.1 – one-X Portal for IP Office Installation
IP Office Release 6.1 – Manager 8.1

13. IP Office Application DVD – CentOS 5.5 Linux incorporating one-X Portal and Voicemail Pro Preferred Edition

IP Office Applications Server – a single install image that installs a *CentOS 5.5* Linux O/S, Voicemail Pro and one-X Portal onto a PC. This will simplify the overall end user applications installation experience and reduce the overall solution costs by removing the need for Microsoft Windows Server licenses.

Therefore, One DVD Install of IP office Application Server installs:

- Core services
- Voicemail pro
- One-X Portal server
- Linux & Tomcat

"Linux" is a registered trademark of Linus Torvalds

Application Server Minimum PC Requirements:

PC solution running Voicemail Pro, one-X Portal

	Minimum Specification	Recommended Specification
Processor	Intel Core 2 duo	Intel Pentium Quad Core 2.4GHz or AMD Athlon 64 4000 or equivalent
RAM Memory	2GB	4GB
Hard Disk Space	30GB	30GB

The IP Office Application Server is not currently supported on Virtual Servers

Operating System: The IP Office Application Server installation installs its own operating system (CentOS 5.5), replacing any existing operating system on the PC.

Drives: DVD Drive or other bootable source for the operating system installation

Other requirements:

- The server PC must be configurable to boot from DVD in order to overwrite any existing OS. This may require access to the BIOS in order to change the boot order if the PC is supplied with an operating system already installed.

- The IP Office Application Server is intended to operate as a headless server, i.e. without requiring any keyboard, video and mouse (KVM) connections after initial installation. All configuration and user access is done remotely from other PCs.

13.1. Preparing for installation

Server Applications
During the installation process, you can select which IP Office server applications are installed.

Voicemail Pro
If selected, the same information is required as for a Windows based installation of the Voicemail Pro server. Refer to the IP Office Voicemail Pro Installation and Maintenance manual.

One-X Portal for IP Office
If selected, the same information is required as for a Windows based installation of the one-X Portal for IP Office application. For example, IP address of IP Office system, LDAP server information and voicemail server address (if other than the IP Office Application Server address). Refer to the Installation manual.

Server IP Address Settings
The IP Office Application Server can support both IPv4 and or IPv6 addressing, obtained through either DHCP or static addressing.

Hostname
A hostname helps simplify access to the server and the applications it provides rather than requiring users to use the IP address.
.
Time zone
The time zone in which the server is located and whether the sever uses UTC or local time.

Root Password
This password is used for configuration access to the server.

13.2. Client PC

The IP Office Application Server is designed and intended for remote configuration and management. It is not managed directly from the server. Therefore a client PC with a web browser on the same network as the server PC is required for initial configuration. If Voicemail Pro server is one of the selected server applications then the client PC must be a Windows based PC.

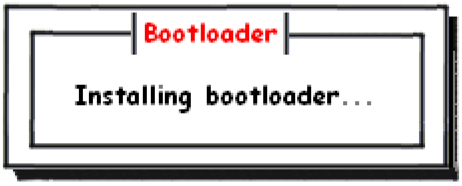
IP Office Voicemail Pro Client uses the following:
Username: Administrator & Password: Administrator

The installer will boot the PC from the Applications Installation DVD to perform an installation. The installation process will be text based and user prompts are kept to a minimum. A customised CentOS kick start script will perform a server reformat and installation of CentOS with the required Linux components to support Voicemail Pro, one-X Portal. Also installed will be the web based service control pages (webControl) and a watchdog utility that takes care of ensuring that the web service is running.

At boot, the install script will check to see if the system already has the Application server installed. If it does then the installer will be offered the option to upgrade/downgrade/re-install or carry out a clean installation. For a new install the installation process will reformat the hard disk to install Linux. There will be no options to allow the installer to configure partitioning or make the server multi O/S.

13.3. Installation

The installation is achieved by creating the DVD, and booting the server with the DVD in the drive. It is recommended that you follow the full instructions and details in the *Application Server Installation & Maintenance Manual*. You then just follow the installation instructions for locale settings, Network Information, enter your "root" password and make a note of it for reference later, accept the license agreement and the installer will complete. The Server will ask you to remove the DVD from the drive then reboot the PC.



Please refer to the IP Office Release 6.1 – *IP Office Application Server Installation and Maintenance Manual* for full details.

13.4. Software

The IP Office Application Server software is provided as a DVD set orderable from Avaya. The first DVD is the one referred to by the installation processes and other information in this documentation. The second DVD is a copy of the Open Source Software (OSS) used by the IP Office Application Server as required by the license terms of that OSS software.

For installers and maintainers, access to a program that supports SFTP/SSH, for example SSH Secure Shell, is useful but not obligatory.

13.5. Licences

The IP Office Application Server software is installed subject to agreement with the Avaya software license terms.

Go to http://support.avaya.com/licenseinfo - On this webpage, the sections below connect to copies of the licenses on the Avaya support web site

- **Avaya Global Software License Terms - English**
- **Avaya Global Software License Terms - Portuguese**
- **Avaya Global Software License Terms - Spanish**

- **Avaya Global Software License Terms - French**
- **Avaya Global Software License Terms - German**
- **Avaya Global Software License Terms - Russian**

13.6. Administration – Webcontrol Client

To administer the Server, the "**WebControl**" browser client facility is used. You need another PC with a browser on the same network:

Enter in your browser the URL http://IPAddress:7070

When the login Page is displayed enter the login:

- *Username : webcontrol*
- *Password : web*

The home page is opened as shown above, which serves as a general health check, also as a Services stop/start and a landing page for the other tabs.

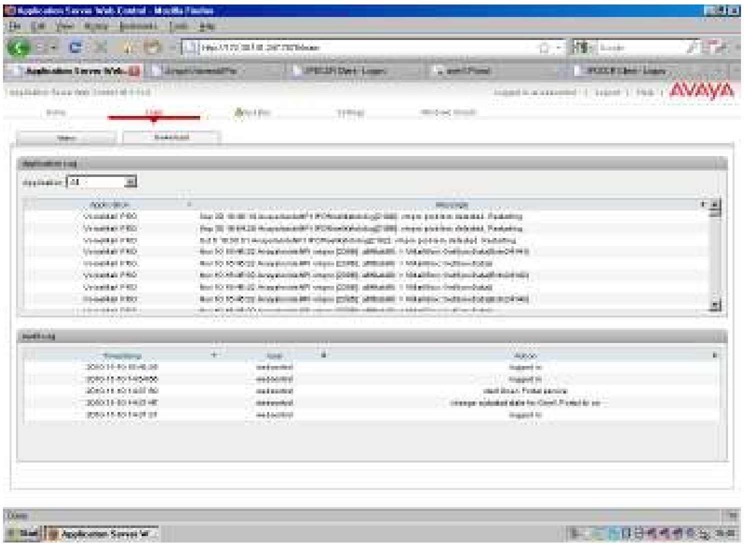
Webcontrol Home Page

The Home page primarily gives status information about the Server PC:

- Operating System Version
- Up Time
- Average CPU Load
- Memory Usage
- Disk Usage
- Control panel to check, stop and start services for one-X Portal and Voicemail Pro.
- Notifications Panel providing a summary of the most recent log messages

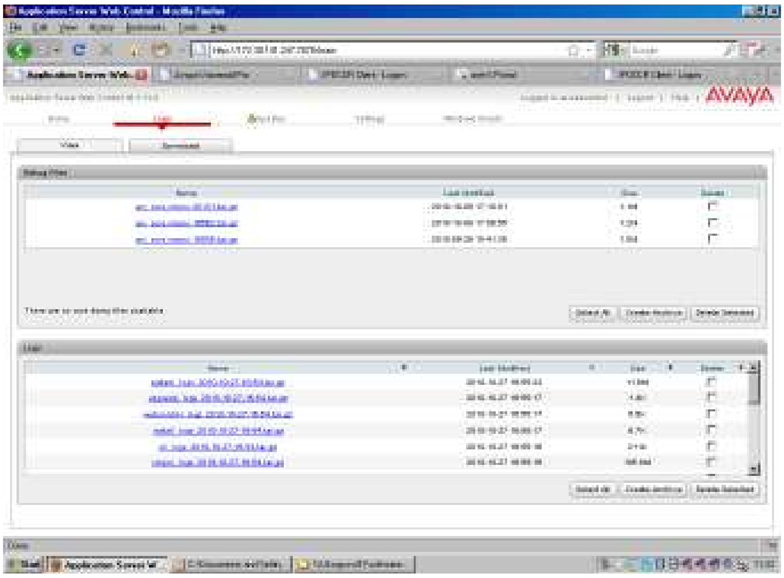
Webcontrol Logs

The IP Office Application Server maintains log files and audit trails for each of the applications it supports. The log records can be viewed through the web browser interface or downloaded for viewing elsewhere.



Application Log
This table lists the log records for a selected server application supported by the IP Office Application Server; the **Application** drop-down is used to select which records are shown. Clicking on a column header sorts the records using that column. The records shown are all those generated since the last time the log files were archived using the **Create Archive** command on the Logs | Download page.

Audit Log
This table lists the actions performed by users logged in through the IP Office Application Server's web browser interface. Clicking on a column header sorts the records using that column.



Debug Files
This gives the option to create/delete/download archives with debug information (core dumps). The table displays the file name, date and size for each debug archive created. Each archive can be downloaded by clicking on the archive file name link.

Logs

This area manages (create/delete/download) archives containing log files. The table displays the file name, data and size for each logs archive created. Each archive can be downloaded by clicking on the archive file name link.

- **Select All –** Select all debug/log archives shown in the list
- **Delete Selected –** All the selected archives will be deleted
- **Create Archive –** Create new archives from the current log files/debug information

Webcontrol Updates

This page displays the different versions of IP Office Application Server operating system files and application files available in the file repository (the file repository locations are configured through the Settings / General page.

Change Version – Clicking the "Change Version" button will open a popup that will allow user to choose any existing version to upgrade or downgrade existing versions of the service.

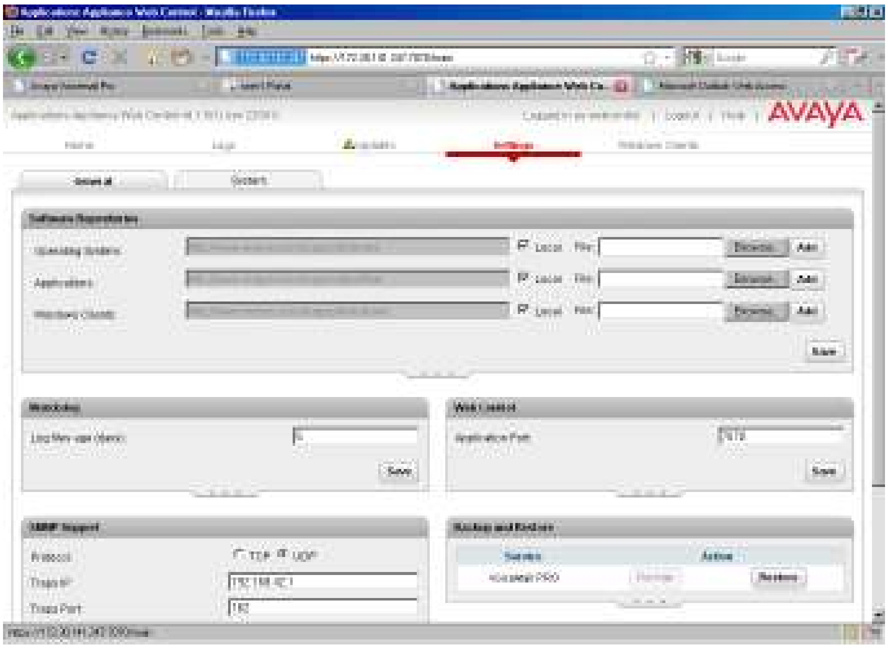
Update – If the user chooses to update a running service a confirmation dialog will be displayed to inform that existing service must be stopped before proceeding with the update.

Check Now – Triggers a new check for available updates for the installed applications.

The behaviour of the **Change Version**, **Update** and **Update All** buttons in this panel depend on the following of factors:

- The controls are not useable unless appropriate update files are available in the applications software repository. This also affects the availability of the **Install** button option.
- Some applications do not support being upgraded or downgraded without first being uninstalled. The controls are not active for those applications until the application is uninstalled (and appropriate update files are available as above).
  - Voicemail Pro supports the options without having to first be uninstalled.
  - For one-X Portal for IP Office, the buttons are only available after using **Uninstall** to remove the existing one-X Portal for IP Office.

Webcontrol Settings / General



Software Repositories
To configure the location of Avaya applications and operating system software repositories. There are two types of software repositories:

- **Remote –** Repository is hosted on a remote machine
- **Local –** Repository is hosted on the local machine

Operating System – Software repository for operating system and updates

Applications – Software repository for Avaya applications such as one-X Portal and Voicemail Pro

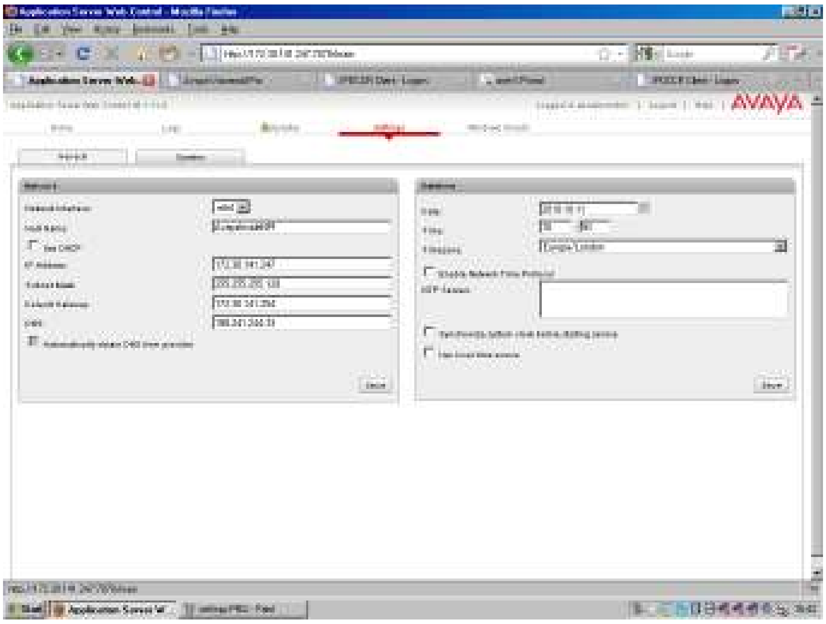
Windows Clients – Software repository for Avaya Windows client applications such as the Voicemail Pro client.

Watchdog – Settings for the Avaya services watchdog
Log Files Age – Age in days for the Avaya applications log

SNMP Support
Protocol – SNMP Transport Protocol
Traps IP – Traps receiver IP Address
Traps Port – Traps receiver port
Device ID – SNMP Device Identifier
System Description – Value of sysDescr MIB
System Location – Value of sysLocation MIB
Enable SNMP – Enable SNMP service checkbox

Backup and Restore - These controls allow you to backup and restore the application settings being used by the selected IP Office applications. Note that this is a simple backup and restore of current application settings. For complete backup and restore functions including user information and files, the separate backup and restore functions of the applications should be used.

Settings / System



Network

Network Interface - If the server PC has multiple ethernet interfaces, this drop down allows selection of which of the interfaces is currently being configured by the web form.

Host name - Sets the host name that the IP Office Application Server should use. This setting requires the local network to support a DNS server.

Use DHCP - If selected, the IP address, subnet mask and default gateway information is obtained by the server making DHCP requests. The related fields are greyed out and cannot be set manually

IP Address - Displays the IP address set for the server. If DHCP is not being used, the field can be edited to change the setting.

Subnet Mask - Displays the subnet mask applied to the IP address. If DHCP is not being used, the field can be edited to change the setting.

Default Gateway - Displays the default gateway settings for routing. If DHCP is not being used, the field can be edited to change the setting

DNS - Enter the address of the primary DNS server. This option is greyed out if the address of the DNS server is set to be obtained from the DHCP server (see below).

Automatically obtain DNS from provider - Obtain the DNS server details from the DHCP server

Date Time

These settings are used to set or obtain a UTC date and time value for use by the IP Office Application Server and services.

Date - Shows the current UTC date being used by the server. If **Enable Network Time Protocol** is selected, this is the date obtained from the NTP server and cannot be manually changed.

Time - Shows the current UTC time being used by the server. If **Enable Network Time Protocol** is selected, this is the time obtained from the NTP server and cannot be manually changed.

Time zone - In some instances the time displayed or used by a function needs to be the local time rather than UTC time. The **Time zone** field is used to determine the appropriate offset that should be applied to the UTC time above.

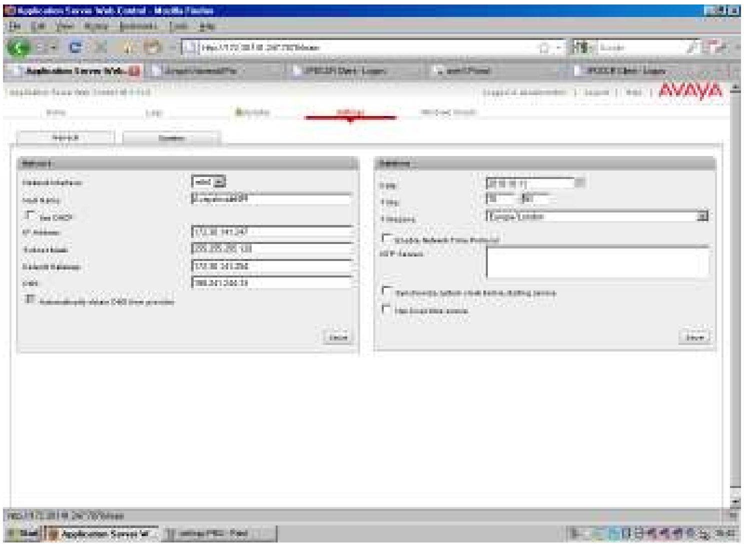
Enable network Time Protocol
This field is used to enter the IP address of an NTP server or servers which should be used when Enable Network Time Protocol is selected. Enter each address as a separate line. The network administrator or ISP may have an NTP server for this purpose. A list of publicly accessible NTP servers is available at http://support.ntp.org/bin/view/Servers/WebHome however it is your responsibility to make sure you are aware of the usage policy for any servers you choose. Choosing several unrelated NTP servers is recommended in case one of the servers you are using becomes unreachable or its clock is unreliable. The operating system uses the responses it receives from the servers to determine which are reliable.

An IP Office Release 6.1+ system can also use NTP to obtain its system time. Using the same servers for the IP Office Application Server and IP Office system is recommended

Synchronize system clock before starting service - When using NTP, the time obtained by the operating system is used to gradually change the server PCs hardware clock time. If this option is selected, an immediate update of the server PC clock to match the NTP obtained time is forced.

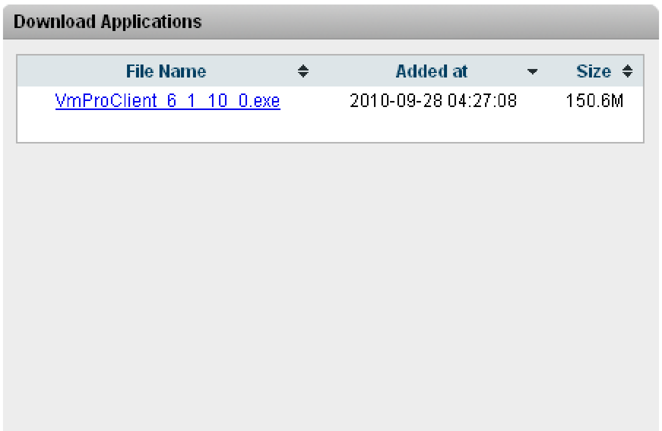
Use Local Time Source - When using NTP, the time obtained by the operating system is used to gradually change the server PCs hardware clock time. If this option is selected, the PCs hardware clock time is used as the current time rather than the NTP time.

Webcontrol Windows Clients



This page is used to download files for use on the local PC. For example the Voicemail Pro client used to administer the Voicemail Pro server application.

The file repository location is configured through the Settings | General page.



13.7. Upgrading Applications

The IP Office application services hosted by the IP Office Application Server can be upgraded without having to reinstall or upgrade the whole server. This is done using files either uploaded to the server (local) or downloaded by the server from an HTTP folder (remote repository), *see File Repositories*. Once an .rpm file (Application upgrade file) or files are available, the IP Office Application Server web configuration pages will list the available versions and allow switching between versions or simple upgrading to the latest version.

Warning

Before upgrading or changing the version of any installed application or operating system components, you must ensure that you have read the relevant Avaya Technical Bulletins for the IP Office Application Server. The Technical Bulletins will detail supported versions of software and known issues or additional actions required for upgrading.

Loading Application Files onto the Server

This method uploads the .rpm file for an application onto the server PC. The files can then be used to update the IP Office applications. The alternative is to use files loaded into a remote software repository.

1. Login to the IP Office Application Server's web configuration pages
2. Select the Settings menu and then the General sub-menu
3. Check that the Local checkbox for Applications is selected.
4. Click on the Browse button and browse to the location of the file that you want to load and select the file. The file name should now be listed in the File field.
5. Click Add. The server will now start uploading the file. The progress of the upload is displayed at the bottom of the browser window.
6. Repeat the process for any other files required.

Example: The following steps can be taken to upgrade to a new version Voice Mail Pro or one-X Portal:

- Open Web Control page and go to the “Settings” tab.
- For the Applications repository select browse and select the VMPRO “rpm” file. Then select Add.
- Navigate to the Updates tab. The ‘Latest Available’ should show as its upgrade version i.e. 6.1.15 (if not then click the Check Now button.)
- Click the ‘Update’ button. If voicemail is running then you may get a prompt to stop it.
- After upgrade you will need to go to the Home tab and start the Voicemail Pro service.
- The 6.1.15 VMPRO Client can be uploaded in same way to make it downloadable from the Windows Clients tab.

13.8. File Repositories

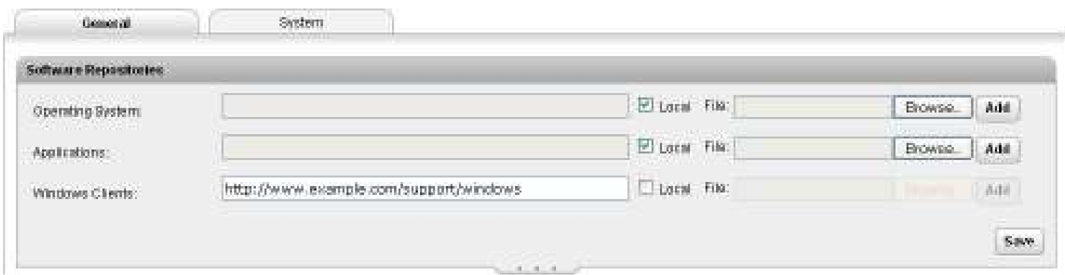
The Updates and Web Client menus use files stored in the configured file repositories. Each repository can be either a set of files uploaded to the server or the URL of a remote folder on an HTTP server. You can add files to these repositories without affecting the existing operation of the server. However, when the application or operating system repositories contain later versions of the files than those currently installed, an icon is displayed on the Updates menu.

Source Files
Update files may be made available individually in response to particular issues or to support new IP Office releases. The files are also included on the IP Office Application Server DVD. Files can be extracted from a DVD .ISO image using an application such as WinZip.

Warning
Before upgrading or changing the version of any installed application or operating system components, you must ensure that you have read the Avaya Technical Bulletins for the IP Office Application Server. The Technical Bulletins will detail supported versions of software and known issues or additional actions required for upgrading.

Upgrade Files	File Type	DVD/.ISO Folder
Voicemail Pro Application *	.rpm	\AVAYA\VMPro
one-X Portal for IP Office Application	.rpm	\AVAYA\ONEX
Windows Client Files	.exe	\AVAYA\THICK_CL
Operation System Files CentOS	.rpm	\CENTOS

*** Voicemail Pro**
Each version of the Voicemail Pro server application is split into separate .rpm files for the server and each of the prompt languages it supports. Unless advised otherwise you should copy or upload the full set of files to the file repository.



Setting the Repository Locations
The IP Office Application Server can use either remote or local software repositories to store software update files. Separate repositories are configured for operating system updates, IP Office application installation files and Windows client files.

The files uploaded or present in the file repositories are used by the Updates and Windows Clients menus.

Repository
If the Local option is not selected, this field is used to set the URL of a remote HTTP file repository. Note that each repository must be different; the same URL must not be used for multiple repositories.

Local
This checkbox is used to set whether the file repository used is local (files stored on the IP Office Application Server or remote (a folder on a HTTP web server specified in the Repository field).

File / Browse / Add
If the Local option is selected, this field and adjacent buttons can be used to browse to a specific update file. When the file is located and selected, click Add to upload the file to the file store on the IP Office Application Server.

Uploading Local Files
The processes below can be used to upload files to the server if it is being used as a repository for that type of file.

Refer to: ***IP Office Application Server Installation and Maintenance Manual***

Uploading Application Files
This method uploads the .rpm file for an application onto the server PC. The files can then be used to update the IP Office applications. The alternative is to use files loaded into a remote software repository.

1. Login to the IP Office Application Server's web configuration pages.
2. Select the Settings menu and then the General sub-menu.
3. Check that the Local checkbox for Applications is selected.
4. Click on the Browse button and browse to the location of the file that you want to load and select the file. The file name should now be listed in the File field.
5. Click Add. The server will now start uploading the file. The progress of the upload is displayed at the bottom of the browser window.
7. Repeat the process for any other files required.

Voicemail Pro
Each version of the Voicemail Pro server application is split into separate .rpm files for the server and each of the prompt languages it supports. Unless advised otherwise you should copy or upload the full set of files to the file repository.

Uploading Operating System Files
This method uploads the .rpm file for an application onto the server PC. The files can then be used to update the IP Office applications. The alternative is to use files loaded into a remote software repository.

1. Login to the IP Office Application Server's web configuration pages.
2. Select the Settings menu and then the General sub-menu.
3. Check that the Local checkbox for Operating System is selected.
4. Click on the Browse button and browse to the location of the file that you want to load and select the file. The file name should now be listed in the File field.

5. Click Add. The server will now start uploading the file. The progress of the upload is displayed at the bottom of the browser window.
6. Repeat the process for any other files required.

Uploading Windows Client Files
This method uploads the .rpm file for an application onto the server PC. The files can then be used to update the IP Office applications. The alternative is to use files loaded into a remote software repository.

1. Login to the IP Office Application Server's web configuration pages.
2. Select the Settings menu and then the General sub-menu.
3. Check that the Local checkbox for Windows Clients is selected.
4. Click on the Browse button and browse to the location of the file that you want to load and select the file. The file name should now be listed in the File field.
5. Click Add. The server will now start uploading the file. The progress of the upload is displayed at the bottom of the browser window.
6. Repeat the process for any other files required.

13.9. Creating Remote Software Repositories

Alternatively to using local files uploaded to the server for updates, the server can be configured to display the versions of files available for use in remote file folders hosted on an HTTP server.

Creating an Application Update Repository
As an alternative to uploading individual .rpm files to the IP Office Application Server, sets of .rpm files can be stored in folders on an HTTP server. The IP Office Application Server is then given the URL of the folders and checks that location for possible updates.

1. Create a folder on the web server for the remote file repository. For example a folder called *Applications*. If the folder is a sub-folder of the existing web site it will be available to browse as part of that website's URL, i.e. if the folder is a sub-folder of *wwwroot*. If the folder is on a separate path, then it must be mapped to the web server URL path, the process for this will depend on the HTTP server being used.
2. The folder directory must be available to browse. For example, in IIS right - click on the folder, select Properties and ensure that Directory Browse option is selected.
3. Copy the .rpm files from their source into the folder. From another PC, test that you can browse to the URL of the folder and that the list of files in the folder is displayed.
4. Login to the IP Office Application Server web configuration pages.
5. Select Settings and then General.
6. Uncheck the Local checkbox for Applications. Enter the URL of the HTTP server folder into the preceding field.
7. Click Save.
8. Select Updates.
9. If the IP Office Application Server is able to access the HTTP folder, the details of versions available will now reflect this available in that folder. The

message *repository error* indicates that the IP Office Application Server was not able to connect to the folder or not able to list the files in the folder.

Creating an Windows Client Repository
The process is the similar to that shown above for application .rpm files. However a separate folder on the HTTP server must be used and the files placed in it are the .exe files used for installing the Windows applications.

Creating an Operating System Repository
The repository for operating system updates is different from those used for application updates and Windows clients. It must be a YUM repository, details of how to setup and configure a YUM repository will depend on the version of Linux being used on the HTTP server. Each time an .rpm file is added, deleted or changed, the directory must be updated using the createrepo *<folder_path>* command.
In order to host the repository on a Windows web server, the folder must be setup and maintained on a Linux server where the createrepo command can be used and the folder then copied to the Windows server.

13.10. SSH Clients

Remote file transfer and command line can be achieved with SSH Clients, some of which are free downloads from the internet. When installed on a Windows PC, use them to remotely administer using Linux command line and file maintenance tasks if required.

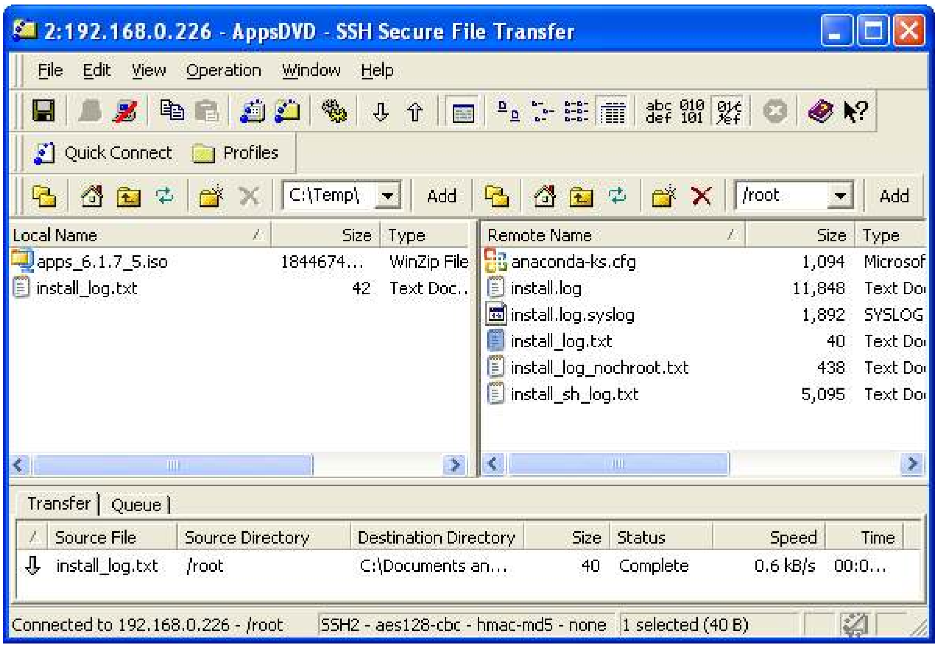
On most SSH clients, just click connect, enter the IP Address of the Applications Server, and enter the login credentials:

Username : root
Password : Whatever root password entered at server installation

Command Line SSH clients will display the same interface that you would see on a directly connected server VDU, e.g.:

```
Last login: Tue Aug 31 14:16:23 2010 from 172.30.141.248
[root@AvayaInsideNPI ~]# ls
anaconda-ks.cfg          install.log.syslog  upgrade.log
install.log              install_log.txt     upgrade_sh_log.txt
install_log_nochroot.txt install_sh_log.txt
[root@AvayaInsideNPI ~]#
```

SSH clients can also provide file transfer functionality similar to a traditional windows interface:



13.11. Client Information

One-X Portal administration and implementation is the same as the Windows variant via a web browser i.e.:

User Client: http://IPAddress:8080/inyama/inyama.html
Administration: http://IPAddress:8080/inyama/inyama.html?admin=true

Voicemail Pro Administration is via a Remote Client - this can be downloaded via the WebControl page in the "Windows Clients" section.

The Voicemail Pro Client Login is:

Username : Administrator :
Password : Administrator

Webcontrol Client Login is via a Web Browser http://IPAddress:7070

Username ; webcontrol
Password : web

13.12. Voicemail Pro Backup and Restore

Voicemail Pro Backup restore facilities are the same on the Windows and Linux versions using the Voicemail Pro client. Please note:

- A Windows Voicemail Pro cannot be backed up then restored to a Linux Voicemail Pro
- A Linux Voicemail Pro cannot be backed up and restored to a Windows Server

This will be addressed in a future release of Voicemail Pro

13.13. One-X Portal for IP Office Backup and Restore

one-X Portal for IP Office 6.1 supports a new set of menus for the backup and, if necessary, restoration of one-X Portal for IP Office configuration settings. These allow backup and restoration using the one-X Portal for IP Office server, an FTP server or your own browser PC as the destination for the backup files.

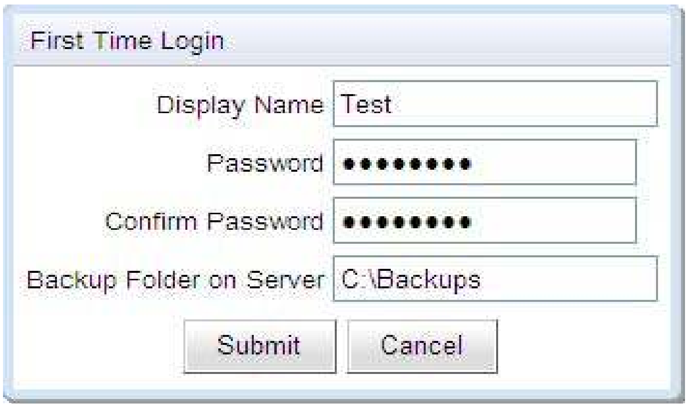
The menus are also intended to allow backup and restoration between an old and a new installation of one-X Portal for IP Office on a new server. Access to the advanced backup and restore menus is controlled by a separate user and password from other administrator access.

Superuser login

Enter the browser address: http://<server>/induna/induna.html

At the login menu, enter the name Superuser and enter the associated password.

- If this is the first login, use the default password ***MyFirstLogin1_0***. After logging in you will be prompted to enter a new password for the ***Superuser*** account plus additional information.

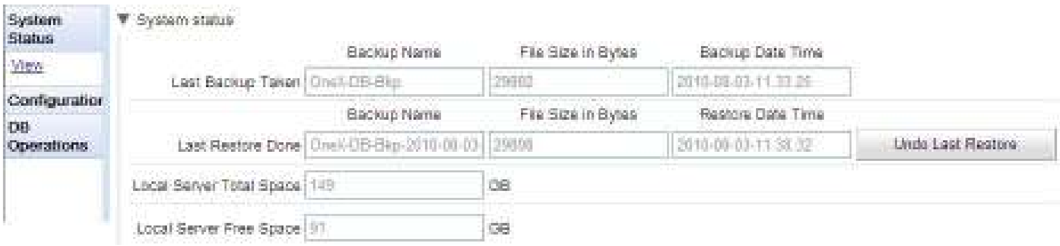


Display Name
Enter a name for display in the one-X Portal for IP Office menus.

Password/Confirm Password
Enter a password that will be used for future ***Superuser*** access.

Backup Folder

This is the path to be used for backup and restore operations on the one-X Portal for IP Office server. Note: Even if backing up and restoring to and from an FTP or local PC folder, this server folder is still used for temporary file storage.



This menu gives a summary of the previous usage of the Superuser menus. It also allows the rollback of the last previous restore operation.

13.14. IP Office Applications Server Manuals and Guides

For more information on the IP Office Applications Server, please refer to the following manuals and guides:

IP Office Release 6.1 – IP Office Application Server Installation and Maintenance
IP Office 6.1 – one-X Portal for IP Office Installation Manual
IP Office 6.1 – one-X Portal for IP Office User Guide
IP Office Release 6.1 – Voicemail Pro Installation and Maintenance

2 Windows Operating System Support

The following table gives a summary of the Server & Client Operating Systems (OS) on which various IP Office applications are tested and supported for IP Office Release 6.0.

IP Office Application	Client Systems						Server Systems		
	XP Pro		Vista		Windows 7		2003	2008	
	32	64	32	64	32	64	32	32	64
Preferred Edition Server	✓	✗	✓	✗	✓	✓	✓	✓	✓
… plus UMS	✗	✗	✗	✗	✗	✗	✓	✓	✓
… plus campaigns	✗	✗	✗	✗	✗	✗	✓	✓	✓
Preferred Edition Client	✓	✓	✓	✓	✓	✓	✓	✓	✓
Contact Store	✗	✗	✓	✗	✗	✗	✓	✓	✗
one-X Portal for IP Office	✗	✗	✗	✗	✗	✗	✓	✓	✓
Customer Call Reporter	✗	✗	✗	✗	✗	✗	✓	✓	✓

Soft Console	✓	✗	✓	✓	✓	✓	✗	✗	✗
IP Office Manager	✓	✓	✓	✓	✓	✓	✓	✓	✓
System Monitor	✓	✓	✓	✓	✓	✓	✓	✓	✓
System Status Application	✓	✓	✓	✓	✓	✓	✓	✓	✓
TAPI – 1st Party	✓	✓	✓	✓	✓	✓	✓	✓	✓
TAPI – 3rd Party	✓	✓	✓	✓	✓	✓	✓	✓	✓
Phone Manager Lite/Pro	✓	✗	✓	✗	✓	✓	✗	✗	✗
Phone Manager PC Softphone	✓	✗	✓	✗	✓	✓	✗	✗	✗

- Vista support is only on Business, Enterprise and Ultimate versions.
- Windows 7 support is only on Professional, Enterprise and Ultimate versions.

Virtual Server Support

For IP Office Release 6.1, all applications supported on Windows server operating systems are supported while running on the following virtual servers:

- VMware Server (free version)
- Microsoft Virtual Server 2005 R2
- Microsoft Server Hyper-V

Browser Application Support

The following applications are accessed using web browsers. The table below details the browsers tested by Avaya.

	Internet Explorer	Firefox	Opera	Chrome	Mac Safari
Preferred Edition UMS	✓ 7+	✓ 3+	✓ 2+	✓	✓ 3.2+
One-X Portal for IP Office	✓ 7+	✓ 3+	✗	✗	✓ 3.2+
Customer Call Reporter	✓ 7+	✓ 3+	✓ 2+	✓	√ 3.2+
Contact Store	✓ 7+	✗	✗	✗	✗

14. Fixed and Known Issues in IP Office Release 6.1

Pre-6.1 issues resolved in 6.1

CQ Number	Description of Issue
CQ120498	System reset when mobile twin set to internal user also configured for mobile twinning
CQ99649	Avaya DS 1400 Phone displays incorrect French Logon - seen as 'onnexion' - should be 'connexion'
CQ106134	16xx and 96xx IP Phones - In band tones not mapped through when redialing a busy number
CQ40395	Display name information element sent is incorrect for DMS 100 switch type
CQ40507	Bridged Appearance Button does not work correctly
CQ40450	IP500 - V5.0.15 + Dial Anonymous on T3 no longer sends the FULL NAME on internal calls
CQ43448	VM $CID and SMDR Call ID NOT the same - VM $CID Help Text states that it should be.
CQ43514	SMDR: Fields 12 and 13 appear incorrect when withholding outgoing CLI using short code
CQ43801	30 seconds of silence before SIP endpoint plays out busy tone after dialing a busy destination over ISDN
CQ104211	SSA - IP500v2 System Hardware Summary - Button translations (French)
CQ44835	Missed call incorrectly indicated on forwarding 1600 phone when fwd unconditional set and forward target user busy
CQ40459	Viewing UMS users via the UMS license disables UMS feature from power users
CQ45811	Russian Cyrillic not displayed correctly using 96xx telephones.
CQ40436	Out of Office Greeting will not play when the time profile has a gap in coverage time
CQ105395	SMDR - incorrect fields (Party1Device) and (Party1Name) when using System Short-code with the W option.
CQ99774	Duplicate of CQ96477 - but on IPO v6.0.8. EVM greetings/messages are distorted.
CQ104629	Call to Extn (with Announcements enabled) with FwdNoAns from SIP phone cuts off after No Answer Time.

Known Issues

The following is a list of issues that exist in this release of IP Office 6.1 software. These will be addressed in a future release of software.

Core Software – IP Office Mode

CQ Number	Description of Issue
CQ112991	11xx/12xx phones: MWI not lit if twinning pair receives message and the Nortel phone is secondary extension
CQ046225	9608 - Call waiting does not show CLI of Caller across SCN
CQ113082	Record inbound for hunt group does not work for internal calls
CQ109672	Embedded Voicemail Recording a greeting from voicemail cuts the end with 1-2 seconds
CQ109678	Erase to default will start an IP Office Law instead of Partner
CQ109681	The free space on the SD card is displayed as being bigger than its capacity, in some case
CQ040499	Can't manage SD card files (Method Not Allowed) file transfer fails
CQ120925	Transfer (REFER) not working to Siemens Gigaset 470IP
CQ040269	IPO Manager Application not working with XP Professional x64 Edition OS.
CQ109092	Nortel Phones 11xx/12xx: Speaker volume fluctuates when dialling *17
CQ109847	IP/VPN phones register but have blank LCD displays
CQ107969	BM32 button doesn't work after login on another phone
CQ110662	Nortel Phones 11xx/12xx: Option not translated Prefs>1. Display>1. Display Settings - all languages
CQ109744	Default button label for Voicemail On shows up as "unknown" for 3641/3645 Spectralink phone
CQ109753	Call recording does not work when hot-desk user logs onto another SCN IPO
CQ109774	14xx handset operating in half duplex.
CQ109908	1416 phone beeps when call on second CA is answered if listening to dial tone on CA1
CQ109909	IPO sends Voice messages notification to one-X Portal after some period of time after deletion

CQ110660	Nortel Phone 1140: Arabic language incorrect displayed and menu in English
CQ112515	1040 does not correctly adjust the transmitted bandwidth during a video call
CQ107151	Call to Hunt Group - Queue message not played and call receives busy in twinning scenario
CQ111463	When blind transfer over SIP Trunk the caller and connected ID is wrong.
CQ108163	Incorrect SMDR Info from a call transferred to an external number – no call time info
CQ112621	T1 Line in Admin Out Of Service still receives incoming calls
CQ46125	Error in Manager - Nortel 11xx & 12xx Phones causes 3rd party endpoint licenses exceeded
CQ113162	Several Danish translated strings are missing, strange strings appear in place.
CQ107508	Conference participants are not updated with Mute information.
CQ 41006	IPO SIP trunk fails to register with certain DNS Servers due to DNS query format error

Core Software – Essential Edition - PARTNER Version

CQ Number	Description of Issue
CQ112589	Norstar Mode: User with call on hold can not receive secondary call
CQ046183	Voice announced transfer can not be automatically answered from a user set to no ring for the line.
CQ110268	Doc: DTMF Breakout not properly presented in Help
CQ045906	Getting "Backup Failed" alarms on daily basis after upgrading from 6.0.8 to 6.1011012
CQ112548	VMS Coverage Rings not displayed correctly from 1400 TUI on Norstar Mode
CQ112585	Nortel Mode: Transfer Return to Programmable Extension when both ICM busy
CQ112553	Wrong number displayed at caller after call is transferred on Norstar Version
CQ110749	Transfer return extension dropdown box displayed in front of label “Transfer return extension"
CQ110900	The screen is not updated when the recall feature is activated

Manager Application

CQ Number	Description of Issue
CQ043744	ENM - Creating a new SIP Extension in a template gives a warning message

VoiceMail Pro (Preferred Edition)

CQ Number	Description of Issue
CQ120934	Binary Mime not supported on Yahoo SMTP servers, Exchange cannot convert VMPRO message.
CQ109409	It is not possible to browse for prompts in VMPro client if the VMPro server is on Linux
CQ045692	Web voicemail: Forward button issue unless you refresh the page on IE8

15. Technical Notes

Before any upgrades commence the IP Office Release 6.1 Administration suite must be installed. Administration suite upgrades are supported from version 4.2, any version prior to this must be removed first before the Administration suite can be installed.

IP Office Release 6.1 will be supported on the following control units:

- IP406v2 (64Mb PCS 8 and later), IP412, IP500, IP500v2

Note: *If upgrading to Release 6.1 from a previous release an upgrade license is required.*

IP Office Release 6.1 will NOT be supported on the following control units:

- IP401, IP403, IP406 (v1), IP406v2 (16Mb PCS7 or earlier), Small Office Edition

Note: *The WAN3 and WAN3 10/100 expansion modules were supported on the IP406v2 and IP412 platforms in Release 5. In Release 6.0 this support was removed.*

If you are not sure how to identify that you have a supported IP406v2 system then please refer to Technical Bulletin 109 for further details.

IMPORTANT INFORMATION – IP406v2 Loader Upgrade

If you have a suitable IP406v2 system you must upgrade the loader before attempting to upgrade to IP Office Release 6.1. Failure to do this will result in your system being left in a non-operational state.

2.1 Core Software Upgrade Summary

The table below shows the necessary steps that must be taken to upgrade your IP Office system to Release 6.1. Once you have identified the steps involved please proceed to section 12.2. If running a software version older than 4.0 then please refer to Technical Bulletin 109.

Platform	Current Release	Upgrade Step 1	Upgrade Step 2
IP406v2	4.0 / 4.1 / 4.2 / 5.0	5.0.999 Loader	Load 6.1
IP412	4.0 / 4.1 / 4.2 / 5.0	Load 6.1	--
IP500	4.0 / 4.1 / 4.2 / 5.0	Load 6.1	--
All modules	4.0 / 4.1 / 4.2 / 5.0	Load 6.1	--

2.2 Software Upgrade License Installation

To make for a smoother upgrade process it is recommended that you install your IP Office Release 6.1 Software Upgrade license before you proceed any further. Although the key may not be recognized immediately by the system, dependant on the current software version you have, it will be recognized when you come to upgrade your system.

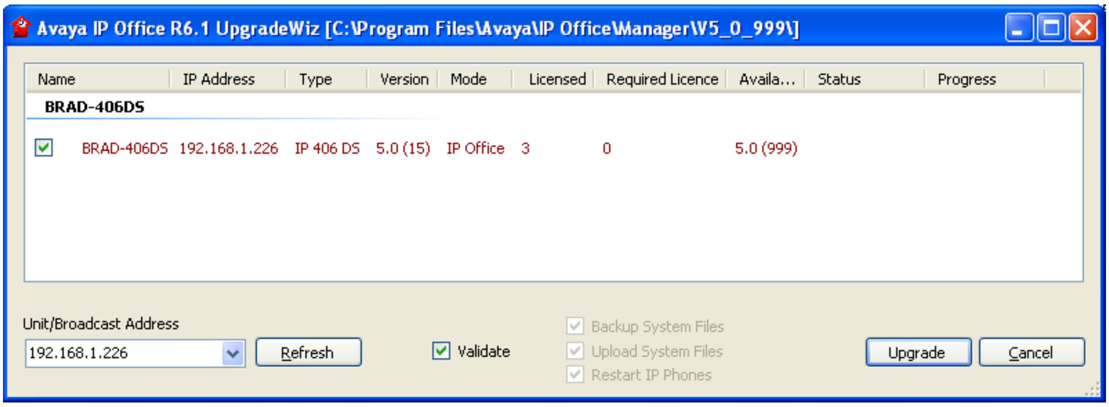
2.3 Core Software Upgrade Instructions

If you do not need to upgrade your loader (IP406v2 systems only) please go to step 11 otherwise follow the instructions below:

1. Install the Admin Suite as normal.
2. Open the Manager application.
3. Before starting any upgrades ensure that you have received and made a backup copy of the latest IP Office configuration. If for any reason the

upgrade fails, the current configuration may be erased, so a backup copy is essential.

4. From the file menu go to Change Working Directory and change the Binary Directory to C:\Program Files\Avaya\IP Office\Manager\V5_0_999.
5. In Manager select File | Advanced | Upgrade. This will start the UpgradeWiz application.
6. After a few seconds the upgrade wizard should show the units found.
7. A window similar to the one below is displayed. The list shows the current software levels of the units and the level of the appropriate bin file that is available in the Manager/Binary working directories.



8. The current version and available versions are displayed. Tick the check box under Name if it is not already ticked then click on Upgrade.
9. After clicking on Yes the upgrade process will begin, follow any on screen prompts. When the upgrade wizard informs you that all units have been upgraded click on OK and close down the upgrade wizard.
10. Make sure that the Manager Binary Directory is set back to C:\Program Files\Avaya\IP Office\Manager.
11. Now follow steps 5-9 again to upgrade your system to IP Office Release 6.1.

Note: IP Office 6.1 Requires a software Upgrade license key (Value 3)

15.1. Unit Compatibility – Expansion Unit Interoperability

All expansion units must be upgraded or downgraded to match the CPU software.

15.2. Phone Firmware Support

The table below lists the telephone firmware versions that are supported by IP Office Release 6.1

Telephone Type	Firmware Version
4610SW, 4620SW, 4621SW, 5610SW, 5620SW, 5621SW	2.9.1 (SP1)
4625	2.9.1(SP1)
4620 (Not 4620SW)	2.3

4601, 4602D, 4602SW, 5601, 5602D, 5602SW	2.3
4601+, 4602+, 5601+, 5602+	2.9.1 (SP1)
4610SW, 4620SW, 4621SW, 5610SW, 5620SW & 5621SW VPN	2.3.252
2410, 2420, 5410, 5420	6
1408,1416	R1 build 49
1403	R1 build 39
9620,9630,9640,9650 Boot Code	3.1.1
9620 & 9630 & 9640 & 9650 App	3.1.1
16xx App	1.3
16xx Boot	1.3
1616 Button Module	1.0.9
9608 & 9611 & 9621 & 9641 Kernel – Not supported	V0r04_V0r15
9608 & 9611 Application – Not supported	S9608_11HALBR6_1r15_V4r52
9621 & 9641 Application – Not supported	S9621_41HALBR6_1r15_V4r52
2410, 2420, 5410, 5420	6
9500 Application – Not supported	R09
9500 Boot – Not supported	R06
1120, 1140, 12x0 SIP Phone Firmware	04.00.03.00

Note: *When upgrading 5410 telephones to the new R6 firmware version make sure that you have added the NoUser source number* ***ALLOW_5410_UPGRADES*** *to your configuration.*

15.3. Preferred Edition Software Upgrade Summary

The table below shows the necessary steps that must be taken to upgrade your Preferred Edition Server (formerly Voicemail Pro) to Release 6.1. If running a software version older than 4.0 then please refer to Technical Bulletin 109.

Product	Current Release	Upgrade Step
Voicemail Pro	4.0 / 4.1 / 4.2 / 5.0 / 6.0	Upgrade Installation Available

It is important that the settings of an existing Voicemail Pro are exported before any upgrade. Although folders that contain prompts and messages are not affected by the upgrade process it is good practice to make a backup just in case something goes wrong.

15.4. Upgrading from Voicemail Pro 4.0 or later

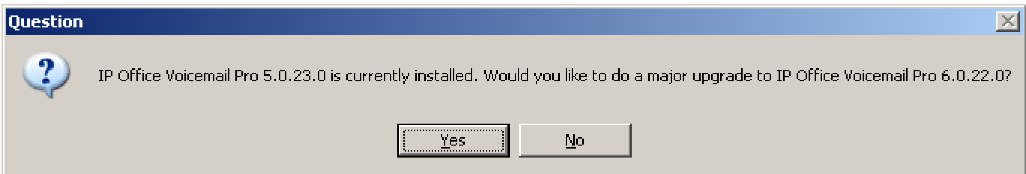
1. Export the Voicemail Pro Database.

Before upgrading Voicemail Pro, you should create a backup copy of the call flow database. This will contain any customizations made to the default call flow.

1. Start the Voicemail Pro GUI.
2. From the File menu, select the option Import or Export.
3. Select the option Export callflows and click **Next**.
4. Enter a file path and file name ending in .mdb, e.g. C:\temp\backup.mdb.

5. Click **Next**.
6. Click **Finish** to start the export then click **Close** to complete the export procedure.
7. Close the program.
8. Insert the new Voicemail Pro CD.   If the setup does not start automatically, right click the CD drive & select **AutoRun**. Alternatively run setup.exe
9. At the language prompt, make your selection & Press OK.

10. A prompt will appear informing you that there is an older version of Voicemail Pro installed and will offer a major upgrade. A major upgrade looks very similar to a new installation.

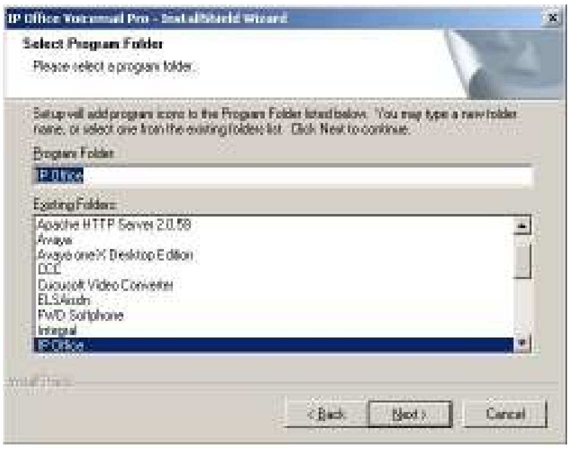


11. Select Yes.

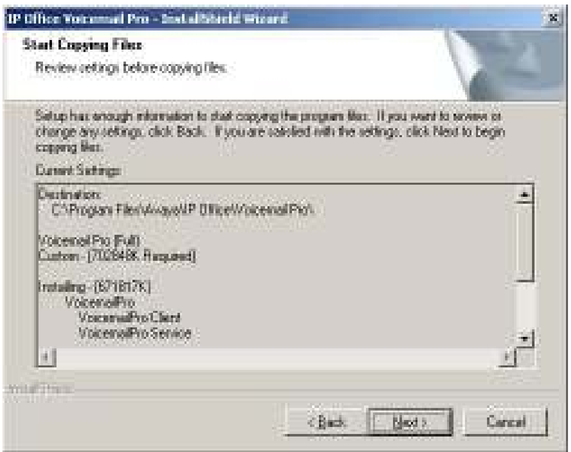
12. At the Select Features screen, make sure that the components you already have installed are selected then click on Next.

13. At the Service Account name screen enter your service account details and then click on Next.

14. At the Select Program Folder screen click on Next.

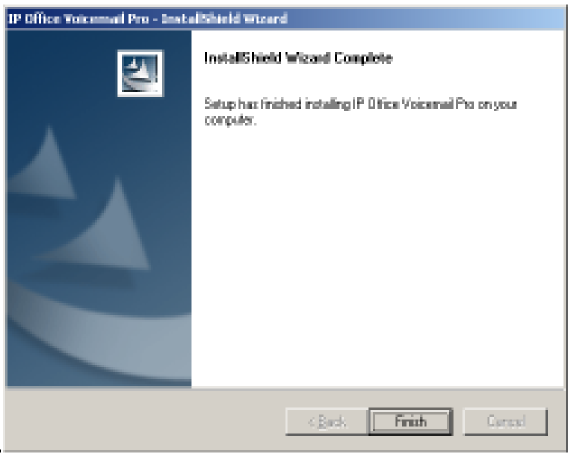


15. At the Start Copying files screen click on Next.



16. The new version of IP Office software will now be installed.

17. Finally click on Finish to complete the installation


IP Office Voicemail Pro - InstallShield Wizard
InstallShield Wizard Complete
Setup has finished installing IP Office Voicemail Pro on your computer.
< Back
Finish
Cancel

.

16. Assistance

16.1. Documentation

IP Office Release 6.1 Documentation can be found on http://support.avaya.com as .pdf copies or as a complete CD image.

1. Select FIND DOCUMENTATION and DOWNLOADS by PRODUCT NAME.
2. Select IP Office.
3. Select the Software release required – 6.1.
4. Select the Documentation Categories required.

16.2. Software

Avaya will supply DVD media to Avaya Authorized Distributors that have a current contract with Avaya. Avaya will not supply DVDs directly to reseller Partners. Partners are required to order DVD media from their respective Avaya Authorized Distributors.

Separate CDs are not available in Release 6.1 the USER/ADMIN SET and the VOICEMAIL PRO will be together on the DVD. The new IP Office Applications Server DVD is available as below.

The following DVDs are available with Release 6.1 of IP Office:

Material Code	Description
700500949	IPO 6.1 USER/ADMIN SET DVD
700501078	IPO R6.1 APPL SRVR DVD

The following License keys are required to upgrade your system to Release 6.1 for IP Office and IP Office Small System Upgrades.

Material Code	Description
262694	IPO LIC UPG R6.1
262695	IPO LIC UPG R6.1 SML

Note: *It may be acceptable to duplicate this media but your contract with Avaya needs to be reviewed in the first instance. If permitted, copies may then be made which must contain an Avaya Proprietary Notice on the DVD.*

Web Availability

The IP Office Release 6.1 binaries for the core platform and DVD images of IP Office Release 6.1 applications will be available on the Avaya Support website by November 29th 2010.

Upgrades to IP Office Release 6.1 require the purchase of a valid Release 6.1 upgrade license. However, the software images may be downloaded without restriction:

1. Go to http://support.avaya.com
2. Click “Downloads”
3. Enter “IP Office” in the “Enter Product Name” search box
4. Select “IP Office” from the drop down list of results

5. Select the software that you want to download from the list of Release 6.1 downloads

16.3. IP Office Release 6.1 System SD Cards

Avaya will supply SD card media to Avaya Authorized Distributors that have a current contract with Avaya. Avaya will not supply SD cards directly to reseller Partners. Partners are required to order SD card media from their respective Avaya Authorized Distributors.

System SD cards supplied by Avaya contain all the system software required for the IP500v2, including expansion module and telephone firmware binaries. Optional SD cards can be obtained from other sources.

The following new System SD card is available with Release 6.1 of IP Office for the MENA region only:

Material Code	Description
700500948	IPO IP500 R6.1 SD CARD NORSTAR

16.4. IP Office Authorization and Avaya University Training

Avaya Product Authorization is designed to ensure our Avaya Channel Partners have the capabilities and skills to successfully design, sell, and implement Avaya products and solutions to exceed customer expectations.

Product Authorization requirements may be found at:

- http://www.avaya-learning.com

16.5. New and updated Classes available with IP Office Release 6.1

Training is one component that must be fulfilled prior to being an Authorized Avaya Channel Partner. The Avaya University IP Office Technical curriculum is updated to reflect IP Office Release 6.1 through the addition of a new IP Office Product Delta course that covers the major enhancements and customer benefits associated with Release 6.1. Please find the new or updated courses below:

Avaya University: **http://avaya-learning.com**

Issued by:
Avaya SMEC New Product Introduction

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