IP Office Technical Bulletin

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Title: General Availability (GA) of IP Office Release 6.0

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Region: GLOBAL

General Availability (GA) of IP Office Release 6.0 Software

Avaya is delighted to announce the launch and availability of IP Office Release 6.0 and the IP500v2 control unit. IP Office is Avaya's Small and Medium Enterprise (SME) solution designed as a global solution for customers supporting up to 384 extensions, 192 analog lines, 192/240 T1/E1 lines and 128 VoIP trunk lines.

1 Product Overview

Avaya IP Office Release 6.0 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 6.0 will be supported on the following control units:

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

Note: *If upgrading to Release 6.0 from a previous release an upgrade license is required. New installations using the IP500v2 platform will not require this license.*

IP Office Release 6.0 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 has been removed.*

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

Please note that the following applications are not supported with IP Office Release 6.0 and will not connect to the system.

Compact Contact Center (CCC)
Delta Server
Call Status
Conferencing Center

IP Office Release 6.0 is the entry-level software required to support the following new hardware:

- IP Office IP500v2 Control Unit
- Combination Card [1]
- ETR 6 Card [2]
- 1400 Series Digital Telephones
- 1600 “Localized” Series IP Telephones
- 9600 Series IP Telephones [3]
- ETR PARTNER® Telephones [4]
- 3910/3920 Wireless Telephones [5]
- DECT R4 Compact Base Station [6]

1) The Combination card is only supported in the IP500v2 control unit. A maximum of two cards are supported.

2) The ETR 6 Card is only supported in the IP500v2 control unit and only when it is running in Essential Edition – PARTNER® Version mode. A maximum of three cards are supported.

3) 9600 series IP telephones are only supported on the IP500 and IP500v2 control units.

4) ETR PARTNER® phones are only supported on the ETR 6 card, in the IP500v2 control unit and only when it is running in Essential Edition – PARTNER® Version mode.

5) Avaya 3910/3920 Wireless telephones are only supported on the ETR 6 card, in the IP500v2 control unit and only when it is running in Essential Edition – PARTNER® Version mode.

6) The DECT R4 Compact Base Station will also be supported with the 1Q 2010 IP Office Release 5 service pack.

As well as increased reliability through improvements to the core system software, IP Office Release 6.0 also supports the following new features:

IP Office Core Software: Version 6.0.7

- IP Office Essential Edition - PARTNER® Version (IP500v2 only)
- Simplified Manager (Essential Edition PARTNER® Version management)
- Licensing Changes
- 1000 User SCN (NOT Essential Edition - PARTNER® Version)
- New Telephone Features (1400/1600/9600 only)
- SIP Trunking Enhancements
- ISDN Redirecting Number Enhancement (IP500/IP500v2 only)
- Send Calling Party Name T1 PRI DMS 100
- Mobility Callback (IP500/IP500v2 only)
- Persistent SMDR storage (IP500/IP500v2 only)
- IP Office Essential Edition - Messaging – Direct Extension Dialing

IP Office Preferred Edition (formerly VoiceMail Pro): Version 6.0.22

- Exchange 2007 Integration Enhancements
- House Keeping and Data Backup
- Text to Speech Announcements
- Graceful shutdown
- Client Enhancements
- Enhanced User Personal Configuration Options
- Small Community Network Resilience Enhancements*
- Distributed Voice Messaging*

Note: *The Small Community Network resilience enhancements and Distributed Voice Messaging features are not supported at GA but will be available during the first half of 2010.*

one-X Portal for IP Office: Version 6.0.21

- User Profiles
- Telecommuter Mode
- Call Pickup
- Improved Integration with Preferred Edition
- Instant Messaging
- Audio Conferencing
- Personal Conference Bridge Number
- Personal Directory Groups

Customer Call Reporter: Version 1.2.2.78

- Wallboard
- Scheduler Enhancements
- Historic Reporting Enhancements
- Installation Changes
- CCR Licensing

DECT R4: Version 3.3.11

- Directory Integration

End of Support for IP Office Release 4.2

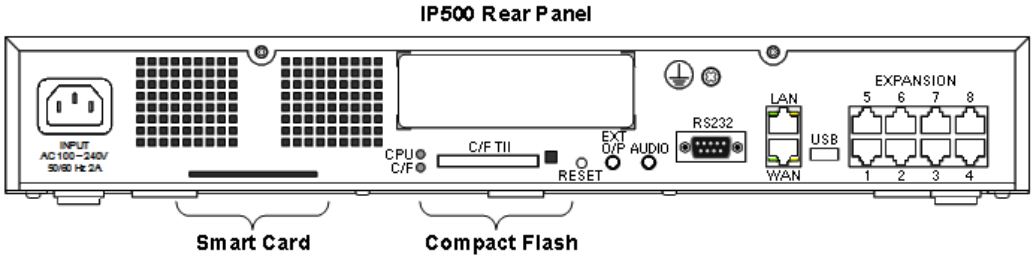
With the GA of IP Office Release 6.0, systems running Release 4.2 will no longer be supported in most instances. However, critical issues found in Release 4.2 can be logged if found on platforms that do not support Release 5 or later (i.e. Small Office Edition systems).

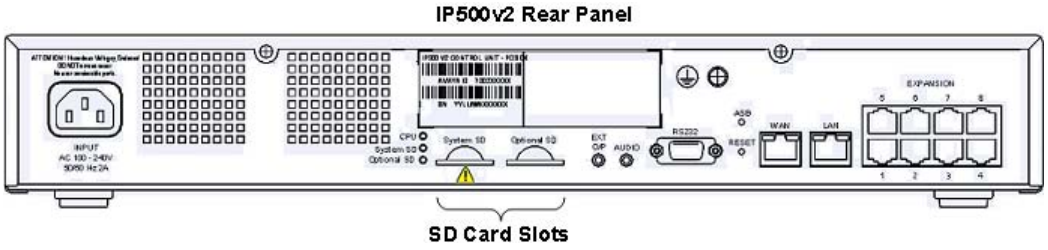
2 IP Office Hardware

The following new hardware is supported with IP Office Release 6.0 software:

2.1 IP Office IP500v2 Control Unit

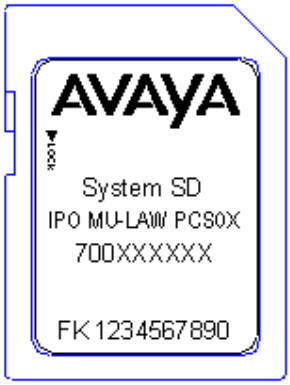
IP Office Release 6.0 introduces a new Control Unit the IP500v2. The IP500v2 is based on the current IP500 control unit and looks identical from the front. The main differences are at the rear of the unit where two Secure Digital (SD) card slots are present. These replace the current smart card and compact flash card slots that exist on the IP500.





The SD card slots are known as the System SD slot and the Optional SD slot.

The System SD card is supplied by Avaya (in 3 variants: IP Office a-law, IP Office mu-law, PARTNER® Version) and has a unique Feature Key number and is used for:



- Validating licenses
- Setting the companding law
- Setting the default system operation to PARTNER® Version or IP Office mode

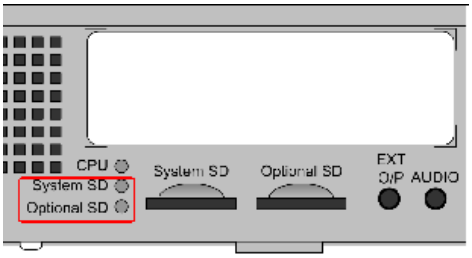
The System SD card also acts as storage for Essential Edition Messaging, with two ports of Voicemail being enabled as standard. Additional increments of two ports can be added through licensing, up to a total of 6 Voicemail ports. It is also used to hold the system configuration, core and expansion module binaries and phone firmware.

The System SD card allows for faster implementation times since it can be pre-loaded with the correct software, configuration and licenses before installation and then sent to site where it can simply be inserted in to the system to complete installation with no management software required. On start up the system will compare its System SD card contents with what it currently has loaded, and if it is different it will upgrade the system software, load the configuration and load the licenses.

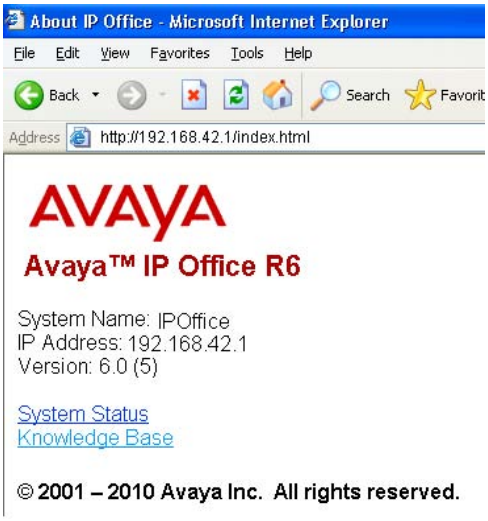
The System SD card is watermarked and should not be reformatted outside of the IP Office Manager application. To format using Manager you can plug the card into a card reader and then use Manager: File>Advanced>Format IP Office SD Card.

Note: *If you use standard windows formatting to reformat the SD card you will lose the watermark and no longer be able to use the card as a System SD card.*

Each card slot has an associated LED and should not be removed whilst the LED is on or flashing. You must use the card or system shutdown commands in Manager or SSA to shutdown the card correctly.



The System SD card also contains the System Status Application (SSA) software. If you point your web browser at the IP address of the IP Office you will see the following screen:



Note: *If you have enabled the "Avaya HTTP Clients Only" setting in the configuration then you will not be able to access this landing page.*

The Optional SD card can be any SDHC specification card of 4GB or larger and is used to provide additional storage, e.g. for System Logs, to provide an upgrade mechanism and act as a backup medium, e.g. for backing up the System SD card. If necessary, the system may run from the Optional SD card, but the System SD card must still be present to provide license validation.

The IP500v2 control unit supports:

- All existing IP500 plug in cards
- All existing IP500 expansion modules.
- The same IP400 expansion modules as the existing IP500
- ETR6 Base Card *
- Combination Base Card *

Note: *The ETR6 and Combination base cards are being introduced with IP Office Release 6.0. They will only be supported in the IP500v2 Control Unit.*

2.2 Combination Card

Note: *Supported in IP500v2 only, maximum of 2 cards supported.*

IP Office Release 6.0 also provides support for a new base card, the Combination card. This card will provide a mix of digital phone (6 ports), analog phone (2 ports) and trunk connections and also provide 10 voice compression channels in a single base card. This will allow for a more competitively priced solution in the sub 20 user space. The Combination base card will only be supported in the IP500v2 and will be supported in IP Office and Essential Edition - PARTNER® Version mode. If the base card is inserted into an IP500 then there will be a visual indication (red solid LED) to show that it is an unsupported card. Manager and SSA will also flag this as an error.

The Combination card will only be sold in bundles with an appropriate trunk card pre-fitted. In the first release only the ATM4 and BRI4 trunk cards will be supported. The combination card with ATM4 will be available globally, the combination card with the BRI4 will be available in APAC and EMEA.

IP Office Release 6.0 will prevent the card from running if a trunk card is not fitted or if the wrong type of trunk card is fitted, e.g. PRI or BRI8. The absence of a trunk daughter card and consequent disabling of the Combination card will be highlighted by a warning in Manager and SSA and an LED error indication on the card.

The Combination card provides 12 RJ45 ports providing the following connectivity:

- Port 1-6 Digital Station (DS) Ports
- Port 7-8 Analog Phone (POTS) Ports*
- Port 9-12 Trunk Ports

Note: *The POTS ports on the Combination card can be used as a Loudspeaker Paging port. Port 8 can also act as a Power Fail port if an ATM4 daughter card is fitted.*

A maximum of 2 Combination cards, installed in any slot, are supported in the IP500v2. If more than 2 cards are installed a warning will be displayed in Manager and SSA and an LED error indication on the card. If this does happen only the 2 cards in the lowest numbered slots will operate.

The restriction of a maximum of 3 DS8 cards in a chassis still applies, but this is in addition to the new Combination card, so the maximum number of DS ports supported in an IP500v2 chassis is now 30 (3 x DS8 plus 1 Combination card).

The restriction of up to 2 VCM32 or VCM64 cards in an IP500v2 chassis still applies but this is in addition to any Combination cards, so the maximum supported number of VCM channels in an IP500v2 system will now be 148 (2 x VCM64 plus two Combination cards).

2.3 ETR 6 Card

Note: *Supported in IP500v2 / Essential Edition - PARTNER® Version only, maximum of 3 cards supported.*

IP Office Release 6.0 provides support for another new base card, the ETR 6. This provides support of the PARTNER® system ETR (Enhanced Tip and Ring) telephones and allows existing PARTNER® customers to migrate to an IP Office solution. The ETR 6 base card will only be supported in the IP500v2 and only when it is running in PARTNER® Version mode. If the base card is inserted into an IP500 or into an IP500v2 running in IP Office mode then there will be a visual indication (red solid LED) to show that it is an unsupported card. Manager and SSA will also flag this as an error.

The ETR 6 card provides 12 RJ45 ports providing the following connectivity:

- Port 1-6 ETR Ports
- Port 7-8 Power Fail Analog Phone Ports*
- Port 9-12 Trunk Ports (ATM4 or Universal PRI single, ordered optionally)

Note: *The Power Fail Analog Phone ports are wired together and provide one circuit only in the event of power fail if an ATM4 daughter card is fitted.*

The ETR 6 card supports the following PARTNER® ACS telephones:

PARTNER ACS “Refreshed” telephones:

- 6 Button display telephone
- 18 Button display telephone
- 34 Button display telephone
- 3910 Wireless telephone
- 3920 Wireless telephone

PARTNER ACS “Euro” telephones:

- 6 Button non-display telephone
- 18 button non-display telephone
- 18 Button display telephone
- 34 Button display telephone

Note: *For the 34 button telephones there is a limit of 4 per system and a maximum of 2 per ETR card.*

A maximum of 3 ETR6 cards, installed in any slot, will be supported in the IP500v2 allowing a maximum of 18 ETR telephones to be connected. Each port will support a single ETR telephone, and can also support a POTS analog telephone or other device such as a fax machine (maximum 1 REN) which can be directly connected to the port or through the auxiliary port on an ETR telephone. Support of the 34 button display telephone is limited to a maximum of 4 per chassis and 2 per card.

Message waiting indication for the POTS device is only supported using the Avaya 51V stepped method. The high voltage method is not supported (phones with neon lights), this is consistent with PARTNER® ACS.

Caller ID on POTS is implemented using the Bellcore 202 method.

ETR ports will not support use as a Loudspeaker Paging port. Standard IP Office POTS ports can be used for paging. Modems connected to ETR ports will be supported to a minimum of V.32.

The maximum distance supported for ETR telephones is 1000 feet or 305m, the maximum distance supported for POTS devices is 3000 feet or 915m.

2.4 1400 Series Digital Telephones

Note: *The 1400 series telephones will be available for volume shipments in April 2010.*

The 1400 series digital telephones are a new range supported on IP Office, Communications Manager (support to be announced separately at a later date) and Integral 55. The 1400 series telephones are very similar in appearance and function to the 1600 IP telephones, offering the option of many buttons with LEDs for ease of use. All 1400 phones have graphical displays with white backlight for easy readability. These phones work in IP Office and PARTNER® Version modes.

The 1400 series telephones being introduced with IP Office Release 6.0 are:

- 1403 (IP Office only, not CM or I55)
- 1408
- 1416*
- DBM32**

Note: **The 1416 telephones support the connection of up to 3 DBM32 modules, each with 32 buttons. The BM32 for the 1600 telephones is identical in appearance but is not compatible with the 1400 telephones.*

*** DBM32 modules are expected to be available during the first half of 2010.*

2.5 1600 “Localized” Series IP Telephones

Note: *NOT supported in Essential Edition - PARTNER® Version*

A new range of 1600 series IP telephones (1603-I / 1603SW-I / 1608-I / 1616-I) are being introduced to replace the current 1600 series telephones that are offered today. The main difference in these is the introduction of a new graphical display with a white backlight for improved readability, the same as the 1400 series telephones.

These new telephones are referred to as “localized” IP telephones and the existing 1600 series IP telephones are referred to as “Global” IP telephones.

IP Office Release 6.0 will also continue to support the existing global range of 1600 series IP phones, however the 1600-I phones will eventually completely replace those. Currently the 1600-I phones are not supported on IP Office Release 5.

Note: *These phones will only work when the system is in IP Office mode, IP Phones are not supported in PARTNER® Version.*

2.6 9600 Series IP Telephones

Note: *Supported in IP500 / IP500v2 only. NOT supported in Essential Edition - PARTNER® Version*

IP Office Release 6.0 provides support for selected 9600 series IP telephones. Only the H.323 versions of these telephones will be supported, the SIP versions of the 9600 series telephones will not be supported on IP Office. Avaya 9600 Series IP Telephones have a clear and elegant interface with high resolution graphical displays making it easier to read contextual menus, prompts and instructions.

IP Office Release 6.0 will support the following 9600 series telephones:

- 9620L
- 9620C
- 9630G
- 9640
- 9640G
- 9650
- 9650C
- SBM24 (only supported with the 930G/9640/9650/9650C)*
- 9600 Gigabit Ethernet Adapter (all phones except 9620L)
- 9600 Bluetooth Adapter (all phones except 9620L)

Note: *The 9630G/9640/9640G/9650/9650C telephones will support the connection of up to 3 SBM24 modules.*

The 9630G and 9650 have a QVGA quality gray-scale pixel based backlit display. The 9620C, 9640/9640G and 9650C have a ¼ VGA color display. The 9620L has a 320*160 pixel gray-scale backlit display; the 9620C has a color display with the same resolution.

9600 phones can run as VPN phones for secure remote installation, without requiring a special software load. No license is required for VPN operation other than the standard Avaya IP Endpoint license.

9600 phones support the IP Office phone menu including advanced visual voice as well as centralized call log and personal directory.

The following 9600 series IP telephone features are not supported on IP Office:

- USB
- Voice activated dialing
- Wideband codec

Note: These phones will only work when the system is in IP Office mode; IP Phones are not supported in Essential Edition - PARTNER® Version. They are also not supported on the IP406v2 or IP412.

9620L, 9620C IP Telephones

Avaya 9620 IP Telephones are specifically designed for the everyday telephone user – those who typically rely on multiple communications tools such as e-mail and instant messaging (IM), yet still require a high quality and intuitive telephone for voice communications.

The Avaya one-X Deskphone family features an intuitive user interface which helps to make users proficient and confident in performing common telephone tasks such as setting up a conference call or completing a transfer.

With access to up to 12 lines and features through a simple-to-navigate interface, plus a clear view of 3 lines/features with LED indicators, the phones are efficient to use while the small footprint consumes minimal space on a users desktop.

Display/Housing:
· 7-line 3.45 inch backlight graphical display with 320*160 pixels. 9620L: grayscale; 9620C: color
· Flip stand/dual position

Fixed Buttons: 10 plus Four-way navigation cluster button.
· Volume up/down (separate volume levels for the handset, speaker, and ringer), Mute button (LED),Speaker button(LED), Headset button(LED).
· Avaya Menu button (browser, options and settings access), Telephony application button – to return to main telephone screen.
· Contacts button, Call log button (LED), quick-access voicemail Message button with LED and corner message indicator.

Programmable/Contextual buttons:
· 4 contextual softkey buttons. Context sensitive allocation of e.g. Hold, Transfer, Conference and more.
· 12 virtual Appearance/Feature buttons, accessible through navigation cluster. Three features shown at a time with three LED indicators.

Key Labels: Icons and text used on fixed feature keys.

Features Menu: Mobile cell phone style menu with access to most often used features like Call Forwarding, Park, Settings etc. On screen status indication for activated features like call forwarding.

Speakerphone: Full duplex speakerphone

Hearing Aid Compatible: Yes - ergonomic hearing aid compatible handset also supporting TTY acoustic coupler.

Message Waiting Indicator: Yes - also useable as a ringing call alert indicator.

Personalized Ring Patterns: Yes - 8.

Headset Socket: Yes.

Embedded Applications:
· Centralized call log and contact application, fully transparent between 9600 phones, 1408/1416/1608/1616 phones and one-X Portal for IP Office.
· Contacts application (up to 100 entries) and Call Log (Missed, Incoming, Outgoing, up to 30 calls).
Please note that the Voice-dial feature of the 9600 phones on Communication Manager is not available on IP Office.
· Access to company directory and Visual Voice applications on IP Office.

Mounting: Desk or wall mountable with optional wall mount adapter.

Supports the following languages: English, French, Spanish, German, Italian, Dutch, Portuguese and Russian (English and 4 of the other languages at once in the phone).

Note: While it may be possible to download other languages onto the phone, this will not be applicable to any data sent by IP Office. If you switch the language to any other not listed above, the display will show a mixture of the new language and the user's default language on IP Office.

9630G, 9640 and 9640G IP Telephones

Smart, sleek, stylish and highly functional, the Avaya 9640 IP Telephone is designed for the Essential user, those for whom the telephone is essential in order to perform their jobs. Workers who conduct much of their business on the phone — moving from one call to the next. The Avaya 9630/9640 delivers advanced communications capabilities — a brilliant high resolution pixel based color display on the 9640/9640G, an integrated WML application interface, one-touch access to Avaya IP Office mobility or forwarding features — in a solution designed for those who are absolutely dependent on voice communications for their business operations.

With built in Gigabit Ethernet on the 9630G and 9640G, these phones are also perfectly suited for customers with very high requirements for data network performance.

Display/Housing:
· 9-line 3.8 inch backlight graphical display with 320*240 pixels. 9650: grayscale; 9650C: color
· Flip stand/dual position

Fixed Buttons: 11 plus Four-way navigation cluster button.
· Volume up/down (separate volume levels for the handset, speaker, and ringer), Mute button (LED),Speaker button(LED), Headset button(LED).
· Avaya Menu button (browser, options and settings access), Telephony application button – to return to main telephone screen. Call Forward shortcut button
· Contacts button, Call log button (LED), quick-access voicemail Message button with LED and corner message indicator.

Programmable/Contextual buttons: 10.
· 4 contextual softkey buttons. Context sensitive allocation of e.g. Hold, Transfer, Conference and more.
· 6 self-labeled Appearance/Feature buttons with LED for access to 24 administered Appearances/Features at the side of the display.

Key Labels: Icons and text used on fixed feature keys.

Features Menu: Mobile cell phone style menu with access to most often used features like Call Forwarding, Park, Settings etc. On screen status indication for activated features like call forwarding.

Speakerphone: Full duplex speakerphone

Hearing Aid Compatible: Yes - ergonomic hearing aid compatible handset also supporting TTY acoustic coupler.

Message Waiting Indicator: Yes - also useable as a ringing call alert indicator.

Personalized Ring Patterns: Yes - 8.

Headset Socket: Yes.

Embedded Applications:
· Centralized call log and contact application, fully transparent between 9600 phones, 1408/1416/1608/1616 phones and one-X Portal for IP Office.
· Contacts application (up to 100 entries) and Call Log (Missed, Incoming, Outgoing, up to 30 calls).
Please note that the Voice-dial feature of the 9600 phones on Communication Manager is not available on IP Office.
· Access to company directory and Visual Voice applications on IP Office.

Mounting: Desk or wall mountable with optional wall mount adapter.

Supports the following languages: English, French, Spanish, German, Italian, Dutch, Portuguese and Russian (English and 4 of the other languages at once in the phone).

Note: While it may be possible to download other languages onto the phone, this will not be applicable to any data sent by IP Office. If you switch the language to any not listed above, the display will show a mixed of the new language and the user's default language on IP Office.

9650 and 9650C IP Telephones

Smart, sleek, stylish and highly functional, the Avaya 9650 IP Telephone is designed for the Essential and Navigator user profiles who need one touch access to features and bridged appearances of team/Partner buttons. Building receptionists, executive admin staff are examples of Navigator users who answer incoming calls, transfer customers to other extensions and monitor several bridged appearances throughout the day.

This is the ideal solution for receptionists, executive assistants, Contact Center Agents as well as Knowledge workers that need quick access to partner and team features.

Display/Housing:
· 9-line 3.8 inch backlight graphical display with 320*240 pixels. 9650: grayscale; 9650C: color
· Flip stand/dual position

Fixed Buttons: 10 plus Four-way navigation cluster button.
· Volume up/down (separate volume levels for the handset, speaker, and ringer), Mute button (LED), Speaker button(LED), Headset button(LED).
· Avaya Menu button (browser, options and settings access), Telephony application button – to return to main telephone screen.
· Contacts button, Call log button (LED), quick-access voicemail Message button with LED and corner message indicator.

Programmable/Contextual buttons: 15.
· Four contextual softkey buttons. Context sensitive allocation of e.g. Hold, Transfer, Conference and more.
· 3 self-labeled Appearance/Feature buttons with LED for access to 24 administered Appearances/Features at the side of the display.
· 8 Additional self-labeled Appearance/Feature buttons with LED with direct access to 2*8 Appearances/Features.

Key Labels: Icons and text used on fixed feature keys.

Features Menu: Mobile cell phone style menu with access to most often used features like Call Forwarding, Park, Settings etc. On screen status indication for activated features like call forwarding.

Speakerphone: Full duplex speakerphone

Hearing Aid Compatible: Yes - ergonomic hearing aid compatible handset also supporting TTY acoustic coupler.

Message Waiting Indicator: Yes - also useable as a ringing call alert indicator.

Personalized Ring Patterns: Yes - 8.

Headset Socket: Yes.

Embedded Applications:
· Centralized call log and contact application, fully transparent between 9600 phones, 1408/1416/1608/1616 phones and one-X Portal for IP Office.
· Contacts application (up to 100 entries) and Call Log (Missed, Incoming, Outgoing, up to 30 calls).
Please note that the Voice-dial feature of the 9600 phones on Communication Manager is not available on IP Office.
· Access to company directory and Visual Voice applications on IP Office.

Mounting: Desk or wall mountable with optional wall mount adapter.

Supports the following languages: English, French, Spanish, German, Italian, Dutch, Portuguese and Russian (English and 4 of the other languages at once in the phone).

Note: While it may be possible to download other languages onto the phone, this will not be applicable to any data sent by IP Office. If you switch the language to any not listed above, the display will show a mixture of the new language and the user's default language on IP Office.

SBM24 Adapter Module

The SBM24 Expansion Module is an optional device that extends the number of call appearances and Feature buttons available on the telephone. The SBM24 Expansion Module is supported on the 9630G, 9640, 9640G, 9650 & 9650C IP telephones and provides 24 additional programmable buttons - to be configured as call appearances, bridged appearances, or feature keys. A maximum of three SBM24 modules are supported on a single phone.

24 programmable call appearance/feature keys.
· Two backlight LCD screens for button labels. Buttons automatically labeled from the system (no paper labels).
· Connects directly to the associated phone.
· No separate power supply needed. However, using more than one button module changes the power class of the phone to Class 3.
The total number of SBM24 button modules supported on one IP Office system is 42, subject to the total system limits. Please refer to System Button Module Support for the total system limitations of button modules on IP Office.

A Button Module adapter (Ferrite adapter), is required when SBM24 button expansion module is used with the 9630G or 9640/9640G IP telephones.

Gigabit Ethernet Adapter

The 9600 Gigabit Ethernet Adapter can be fully integrated into a wedge stand of the 9600 phones and provides a transparent upgrade option if customers decide to install Gigabit Ethernet at a later date. It is fully integrated into the management interface of the 9600 phones. It requires a separate "Wedge stand".

Bluetooth Adapter

The Bluetooth Adapter for 9600 Series IP Telephones allows Bluetooth communications devices such as headsets to interoperate with 9600 Series telephones. The adapter allows highly mobile workers to use the same Bluetooth headset with their 9600 Series Deskphone as they use with their cell phone.

The Avaya ABT-35H Bluetooth headset when used with the Bluetooth Adapter supports a wireless range of up to 30 feet from the Deskphone and provides simple call control (answer, hang-up) directly from the headset. For users while at their desk, the Bluetooth Adapter allows the freedom of hands free communication along with investment protection for those already utilizing Bluetooth headsets while on the go.

It supports the Avaya ABT-35 as well as many other third party Bluetooth Headsets. The Bluetooth Adapter can be used with all 9600 phones except the 9620L. It requires a separate "Wedge stand".

2.7 ETR Telephones

Note: *Supported on ETR 6 Card / IP500v2 / Essential Edition - PARTNER® Version only*

IP Office Release 6.0 supports the following PARTNER® ACS telephones on the ETR 6 base card when the system is running in PARTNER® Version mode.

PARTNER ACS “Refreshed” telephones:

- 6 Button display telephone
- 18 Button display telephone
- 34 Button display telephone
- 3910 Wireless telephone
- 3920 Wireless telephone

PARTNER ACS “Euro” telephones:

- 6 Button non- display telephone
- 18 button non-display telephone
- 18 Button display telephone
- 34 Button display telephone

Note: *These phones will only work when the system is in Essential Edition - PARTNER® Version mode. The ETR 6 base card and ETR telephones are not supported when the system is in IP Office mode. The ETR 6 card is only supported in the IP500v2. For the 34 button telephones there is a limit of 4 per system and a maximum of 2 per ETR card.*

2.8 DECT R4 Compact Base Station

Note: *NOT supported in Essential Edition - PARTNER® Version*

IP Office Release 6.0 supports lower cost/functionality base-stations for the IP Office DECT R4 system released with IP Office 5.0. The compact base station, which is exactly the same size as the current DECT R4 base station, supports a maximum of 4 channels per base station with a maximum of 5 base stations being supported per IP Office.

Standard and Compact base stations may be mixed as long as no more than 5 compact base stations are installed.

IP Office Release 6.0 will also support directory integration from the base station of the DECT solution without requiring an AIWS server. This applies to both the standard and compact base stations.

Note: *DECT R4 Compact is only supported when the system is in IP Office mode, it is not supported with Essential Edition - PARTNER® Version.*

2.9 Extended Support of Button Modules on IP Office

The total number of buttons on associated DSS units (EU24, BM32, DBM32, SBM24, DSS 4450, T3 DSS) has been increased to a total of 1024 buttons per system, supporting e.g. 32 BM32/DBM32 modules. Some restrictions on the total number of certain types of button modules still apply.

BM32: *(1616)*
Up to 32 DBM32s supported (32 buttons each, up to 1024 buttons).

DSS4450: *(4412D+, 4424D+)*
Up to 8 DSS4450 modules per system (maximum of 2 per DS expansion module, including the
IP406v2) (50 buttons each, max 400 buttons).

DBM32: *(1416)*
Up to 32 DBM32s supported (32 buttons each, up to 1024 buttons).

EU24/EU24BL: *(4620, 4621, 5420, 5620, 5621/4620, 4621, 5620, 5621)*
Up to 8 EU24s per system (24 buttons each, max 196 buttons).

SBM24: *(9630G, 9640, 9640G, 9650, 9650C)*
Up to 42 SBM24s per system (24 buttons each, up to 1008 buttons).

T3 Series: *(All T3 phones)*
Up to 30 T3 DSS modules supported. (36 buttons each, up to 1080 buttons). T3 DSS buttons are not included in the overall system limit for additional button module buttons.

XM24: *(6416+, 6424+)*
Up to 10 XM24 units per system (maximum of 2 per DS expansion module, including the IP406v2) with a maximum of 1 per 6416+/6424+ phone (24 buttons each giving a maximum of 240 XM24 buttons).

The maximum number of additional button module buttons per system is 1024 (regardless of if the buttons are configured for use or not). Only exception is use of T3 DSS only where, if no other button modules are used, a total of 1080 buttons is supported:

- 12 x BM32 + 5 x EU24 + 20 x SBM24 is allowed (984 buttons in total).
- 15 x T3 DSS + 15 x BM32 is allowed (1020 buttons in total).
- but NOT 8 x BM32 + 10 x EU24 (because it exceeds the maximum number of 8 EU24s, even though total buttons < 1024).
- and NOT 14 x BM32 + 5 x EU24 + 20 x SBM24 (total 1048 buttons, this exceeds the maximum 1024 buttons per system).

2.10 Avaya IP Office Telephone Support

IP Office Release 6.0 will support the following telephones (pending regional availability, not all telephones available for new sales).

- 2400/5400 series digital telephones
- 1400 series digital telephones
- 4600/5600 series IP telephones
- 1600 series IP telephones
- 1600-I series IP telephones
- 9620L/9620C/9630G/9640/9640G/9650/9650C IP telephones
- 4406, 4412, 4424D+ telephones (DS expansion modules only, no longer sold new)
- 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 (WiFi, 3616/3620/3626 no longer sold new)
- 3810 (900MHz)
- ETR6/ETR6D/ETR18/ETR18D/ETR34D/3910/3920 telephones (Essential Edition – PARTNER® Version only)
- Analog phones
- Associated DSS units (EU24, BM32, DBM32, SBM24, DSS 4450, T3 DSS)

2.11 New Material Codes for IP Office Release 6.0 Hardware

The following table lists the new material codes for the hardware that is being introduced/supported with IP Office Release 6.0:

Material Code	Description
700476005	IPO IP500V2 CONTROL UNIT
700476013	IPO IP500V2 COMBINATION CARD ATM
700476021	IPO IP500V2 COMBINATION CARD BRI
700476039	IPO IP500V2 EXTN CARD ETR 6
700479702	IPO SYSTEM SD CARD A-LAW
700479710	IPO SYSTEM SD CARD MU-LAW
700479728	IPO SYSTEM SD CARD PARTNER
700480080	DECT IP RBS COMPACT IP OFFICE
700469927	1403 TELSET FOR IPO
700469851	1408 TELSET FOR CM/IE/IPO UpN
700469869	1416 TELSET FOR CM/IE/IPO UpN
700469968	BUTTON MOD FOR 1400 SERIES
700476849	IP PHONE 1603-I BLK
700458524	IP PHONE 1603SW-I BLK
700458532	IP PHONE 1608-I BLK
700458540	IP PHONE 1616-I BLK
700461197	IP PHONE 9620L CHARCOAL GRY
700461205	IP PHONE 9620C CHARCOAL GRY
700383938	IP PHONE 9650 GRY 9650D01A
700461213	IP PHONE 9650C CHARCOAL GRY
700405673	IP PHONE 9630G GRY 9630GD01A
700383920	IP PHONE 9640 GRY 9640D01A
700419195	IP PHONE 9640G GRY
700462518	9600 SBM24 BUTTON MOD GRY
700461239	IP PHONE 9620L W/O FCPLT
700461247	IP PHONE 9620C W/O FCPLT
700408602	IP PHONE 9630G W/O FCPLT
700408610	IP PHONE 9640 W/O FCPLT
700429095	IP PHONE 9640G W/O FCPLT
700408628	IP PHONE 9650 W/O FCPLT
700461254	IP PHONE 9650C W/O FCPLT
700462526	IP PHONE SBM24 BUTTON MOD W/O FCPLT
700383789	Bluetooth Adapter for 9600 Series
700383888	Wedge stand for 9620
700382492	Wedge stand for 9630/40/50
700383771	IP PHONE 9600 GIGABIT ETHERNET ADPTR

3 IP Office Core Software Enhancements

3.1 Avaya IP Office Essential Edition - PARTNER® Version

Note: *Supported on IP500v2 only*

IP Office Essential Edition - PARTNER® Version is a new mode of operation, only supported on the IP500v2, that is being introduced in IP Office Release 6.0. It is enabled by purchasing the appropriate System SD card. By using the new ETR 6 base card as part of the solution customers will be able to reuse their existing PARTNER® ETR telephones if migrating to an IP Office platform.

PARTNER® Version replicates much of the functionality of the PARTNER® ACS system including an extensive Telephony User Interface for administration. The IP Office Manager has been modified to provide a simplified interface when managing a system in Essential Edition - PARTNER® Version mode.

PARTNER® Version is a stand-alone system, and will support a maximum of one 16 port expansion module. PARTNER® Version will not support networking (Small Community Networking) or IP telephones.

PARTNER® Version is designed to work directly out of the box without any programming or administration, allowing users to immediately make and place calls via analog trunks.

PARTNER® Version provides the end user a similar desktop experience as the PARTNER® ACS: e.g. 2 dedicated intercom buttons, key system functionality, Caller ID, etc. This helps minimize training requirements and provides the end user with the same comfort level they have come to expect with the PARTNER® ACS system.

PARTNER® Version provides the bulk of the feature set that the current PARTNER® ACS R8 supports; plus integrates enhanced features from the IP Office Essential Edition platform, such as Mobile Twinning, Visual Voice Mail, 64-party Conferencing, SIP Trunking, Full PRI/T1 and more.

PARTNER® Version provides the same 2-digit extension numbering plan as PARTNER® ACS.

PARTNER® Version supports all “Refreshed” and “Euro” versions of PARTNER® ACS telephones. PARTNER® Version also introduces a new line of digital telephones, the 1400 series, which will work in conjunction with the PARTNER® ETR sets.

Also supported are the current 3910 and 3920 wireless products, as well as most analog devices (fax, wireless headsets, analog phones). The original MLS series telephones and older wireless telephones are not supported.

PARTNER® Version includes Essential Edition Embedded Voicemail, providing a 2-port Automated Attendant (upgradeable to 6-ports) and voicemail coverage activated for all telephones. The system also includes 3 SIP licenses and 3 mobility licenses.

PARTNER® Version will support the following base cards, daughter cards and expansion modules:

Component	Maximum Supported
Phone 8 Base Card	4
Phone 2 Base Card	4
DS8 Base Card	3
ETR6 Base Card	3
Combination Base Card	2
ATM4U Daughter Card	4
PRI1 Universal Daughter Card	1
IP500 Phone16 Expansion Module	1
IP500 Digital Station 16 Expansion Module	1

Based on these configurations the PARTNER® Version system will scale to the following capacities (may not be attained simultaneously):

48 Stations:

- Up to 18 ETR Stations
- Up to 46 Digital Stations
- Up to 48 Analog Stations

56 Trunks:

- Up to 12 analog CO lines
- 1 PRI/T1 (24 lines)
- Up to 20 SIP lines

System programming is accomplished in 2 ways:

Telephone User Interface (TUI) from extension 10 or extension 11; and uses the same administration codes as the PARTNER® ACS system. Button Programming Features use the same codes as the PARTNER® ACS system.

TUI programming is an alternate way to do System Administration and Programming on the PARTNER® Version. It is based on the PARTNER® ACS TUI, but with improvements.

Using TUI System Administration on ETR and 14xx telephones, the buttons under the display become soft-keys which are used to navigate and change the administration items. There are also shortcut codes which enable the administrator to jump to particular administration items.

Centralized Programming is similar to System Administration, but it is used to administer buttons and other options for any phone in the system.

Individual Programming continues to be supported, allowing the station user to program buttons on their telephones.

Graphical User Interface (GUI) – PARTNER® Version provides a “Simplified Manager” mode in the IP Office Manager software; allowing you to program the system in less than 30 minutes.

Changing from Essential Edition - PARTNER® Version to IP Office Mode

If a customer wishes to change from Essential Edition - PARTNER® Version to IP Office mode then this can be accomplished easily by using the Simplified Manager application. After the change, if no valid configuration is present the system will come up with the defaults for the current mode. There will be no attempt to convert the existing configuration. If the system has any ETR 6 base cards installed then these will stop working when the system is switched into IP Office mode.

System Features

Essential Edition - PARTNER® Version provides the majority of the feature set that the PARTNER ACS R8 supports. A listing of the built-in features is detailed on the following pages.

911 Emergency Dialing
911 Emergency Dialing can be made from any extension, regardless of any restrictions.

Abbreviated Ringing (#305)
This feature activates or deactivates Abbreviated Ringing at the system telephone at a specific extension. When you are on a call and Abbreviated Ringing is Active, any incoming call rings only once. The green light next to the line button flashes until the call is answered or the caller hangs up (or for a transferred call, until the call returns to the transfer return extension).

This feature prevents incoming calls from distracting you when you are busy on another call. To allow calls to ring repeatedly, set Abbreviated Ringing to Not Active.

Absent Text Messaging (FEATURE 28)
This feature allows you to post a message (such as “Do Not Disturb” or “Away from desk”) on the display of your PARTNER ETR or DS system phone. When another extension calls your extension, your active Absent Text Message appears on the caller’s display.

The system provides 15 pre-defined messages, plus 2 that may be customized by each user. Pre-defined messages include:

Back Soon	Back Tomorrow	Do Not Disturb
On Vacation	Out to Lunch	At Home
Away from Desk	Be Right Back	Please Call
On Holiday until	Meeting until	Don’t disturb until
With visitors until	With cust. til	At lunch until

Account Code Entry (FEATURE 12)
Account Code Entry (ACE) is used to enter an account code (up to 15 digits) for an incoming or outgoing call currently being handled.

The Account code is printed in the SMDR record for the call for the customer to use for cost tracking purposes.

There are two different ways the feature can be used:

- Voluntary — Account Code Entry is optional.
- Forced – Account Code Entry is mandatory before dialing on an idle CO facility.

Forced account codes are validated against a list of pre-defined account codes which is created via the Manager application.

Allowed Lists(#407) / Disallowed Lists (#404)

Allowed Lists Use this feature to specify telephone numbers that users can dial regardless of other dialing restrictions, as long as they have access to an outside line. For example, if you restrict an entire category of calls through Disallowed Phone Number Lists (#404), you can permit calls to a specific number in that category by placing that number on an Allowed Phone Number List.

You can create up to eight Allowed Phone Number Lists of up to 10 telephone numbers each.

Disallowed Lists allow you to specify telephone numbers that users cannot dial. For example, you may want to prevent calls to a specific telephone number or to categories of numbers, such as international numbers.

You can create up to eight Disallowed Phone Number Lists of up to 10 telephone numbers each.

Allowed and Disallowed Phone Numbers can be up to 12 digits long and may include 0–9, #, *, and 'hold' (a wildcard character, displayed as "?").

Attendant Positions
Extension 10 and Extension 11 serve as system attendant positions. TUI System Administration can only be done at these extensions. Night Service and Unlock capabilities can only be initialized from Extension 10. When a user dials 0, Extension 10 will ring.

Auto Dialing
Enables you to dial outside numbers, extension numbers, feature codes or account codes with a single touch by pressing a programmed button.

3 types of auto dial are supported:

- Auto Dial – Outside
- Auto Dial – ICM (internal extension)
- Auto Dial – ICM Page (voice signaling)

Automatic Callback
When calling an extension that is busy, press the Auto Callback softkey and the system will call you when the extension becomes free.

Automatic Daylight Savings Time (#126)
This feature automatically updates the system clock for annual Daylight Savings Time and Standard Time changes. Turn this feature off if you live in an area without daylight savings time.

Automatic Extension Privacy (#304)
Prevents other extensions with access to the same line from joining a call from their extension. Use this feature for extensions connected to a modem, fax, or any device whose function can be disrupted by someone trying to join it.

Automatic Line Selection
Specifies the order in which the system selects an available line (intercom or outside) when a user at the extension lifts the handset or presses SPEAKER to make a call without first selecting a specific line button.

For extensions with analog telephones, set Automatic Line Selection to intercom first. This enables analog telephones to access system features, including intercom calling. When users lift the handsets on analog telephones, they hear intercom dial tone. To access an outside line, they must dial 9.

Note: *Analog phones connected to an ETR phone hear CO dial tone and Analog phones connected to POTS ports hear intercom dial tone when they go off hook.*

Automatic Voice Mail Coverage (#310)
Use this feature to program a button to turn VMS Cover on and off, routing unanswered intercom and transferred calls for a users' extension to the Call Answer Service of the voice messaging system after the specified number of rings (factory setting is three rings).

Backup Failure Alarm
When an Automatic Backup fails, a 'Backup Failed' message is displayed on Extension 10 and 11.

Backup Programming – Automatic (#123)
The PARTNER® Version does a nightly backup to the system SD card. No manual intervention or external PC is required.

Backup Programming - Manual (#124)
The administrator may do a manual system programming backup to the system SD card via the TUI.

Bridging (Joining Calls)
A user can bridge (join) on any active call by pressing the associated CO line button (assuming Privacy is not active on the originating extension.) When a call is bridged, the red and green LED's will alternate on all telephones which are bridged on the call.

Call Coverage (FEATURE 20)
This feature is used to redirect all Intercom, Transferred, DID and outside calls from a user's extension to another extension.

When Call Coverage is activated, covered calls are routed to the covering extension after a specified number of rings.

Call Forwarding (FEATURE 11)
The Call Forwarding feature provides a means of redirecting Intercom (ICOM), CO, and Transfer calls from one extension to another specified extension.

Call Log - Digital Sets
Call Log is a fixed button on digital telephones and provides a visual record of calls made and received, including unanswered calls. Details are store for both users (maximum 30 entries) and hunt groups (maximum 10 entries). The method of operation varies according to the phone type but in all cases the call records can be used for return calls.

Call Logs - ETR sets (#317)
Use this feature to program extensions to log answered Caller ID calls so that calls answered at an individual extension can be viewed in the Caller ID Call Log. Once the feature is activated for an extension, when a user activates the Caller ID Call Logging and Dialing (F23), all Caller ID calls that are answered by that extension are logged.

Up to 3 extensions can be designated to log all calls, both answered and unanswered, regardless of where they terminated in the system.

Call Pickup (INTERCOM 6xx)
This feature is used to answer any intercom, transferred or outside call ringing at another specified extension.

Call Timer Display
Displays the duration of the call.

Call Waiting (#316)
Call waiting applies to analog, single line telephones, and applies to the following types of calls:

- Intercom calls
- Transferred calls
- Forwarded calls
- Covered calls
- Outside calls (if the extension has a line appearance of the line)
- Voice Signaled calls

Caller ID Features
The following Caller ID features are supported:

- **Caller ID Inspect (FEATURE 17)** Allows you to view Caller ID information for a second call, without disconnecting the current call or putting it on hold. Use this feature to inspect ringing, active, or held calls.

- **Caller ID Name Display (FEATURE 16 – ETR phones only)** When an incoming call is ringing at a display telephone, both the name and number appear on the user's display. The Number is displayed on the top line, and the Name on the 2nd line. In some cases users may wish to see the Caller ID Name on the top line and the Caller ID Number on the 2nd line. This feature provides this capability.

Cell Phone Connect
See Mobile Twinning.

CO Disconnect Time (Programmable) (#203)
Use this feature to change the hold disconnect time for an outside line. Different telephone companies use different length signals. The length of the signal is called the CO or Hold Disconnect time.

Conference Drop
On 1400 digital telephones, you may choose which specific party you want to drop from a conference call. On ETR and analog telephones, the last party added to the conference call is automatically dropped.

Conferencing
PARTNER® Version supports up to 64-party conferencing. A maximum of 2 analog lines can be in any conference call. Conference calls may consist of combinations of analog lines, digital lines (PRI/T1, SIP), analog phones, ETR phones or digital phones.

Contact Closure Support for 2 Devices (FEATURE 41 and FEATURE 42)
The IP500 v2 Control Unit has a Contact Closure control jack on the back panel. This is a 3.5 mm stereo jack that can drive two external adjunct relays.

This feature allows users at eligible extensions to control an external adjunct by entering feature codes at their system or standard telephone. The port can control a variety of devices, the most common being door strikes, night bells, lighting, or voice recording systems.

Copy Settings (#399)
This feature allows a System Administrator to copy extension-specific programming from one extension to another. When the Extension Copy operation is performed, the destination extension will have the same properties as the originating extension.

Extension Copy can only be performed from an Attendant Extension in System Programming Mode or from the Windows Manager PC application.

Dial Code Features
- 'F' represents the FEATURE button.
- 'I' represents the INTERCOM button.
- 'XX' represents a system extension number.
- 'PP' represents a Personal Speed Dial code.
- 'G' represents a single-digit group number.
- 'LL' represents a system line number.

Dial Code	Feature
F01	Do Not Disturb
F03	Recall
F04	Save Number Redial
F05	Last Number Redial
F06	Conference Drop
F07	Privacy
F11	XX,XX Call Forwarding/Call Follow-Me
F11	XX,*PP Cell Phone Connect
F11	XX,PP Remote Call Forwarding
F12	Account Code Entry
F14	Voice Mailbox Transfer
F15	VMS Cover
F16	Caller ID Name Display
F17	Caller ID Inspect
F20	XX,XX Call Coverage
F21	Station Lock
F22	Station Unlock
F23	Caller ID Call Logging and Dialing
F26	Hot Dial
F28	Absent Text Message
F41	Contact Closure 1
F42	Contact Closure 2
F59	System Release

Direct Inward Dialing (DID)
Use this feature to have Direct Inward Dialing (DID) numbers on a PRI / T1 or SIP line routed automatically to a specific extension, hunt group, or calling group. When a Direct Inward Dialing (DID) call arrives, the PARTNER® system “collects” the digits from the T1 service provider (that is, the number that was dialed) and matches the number to a DID Mapping Table you created. If the incoming number matches an entry in the DID Mapping table, the call rings at the matching extension or hunt group.

Direct Line Pickup (INTERCOM 68LL, INTERCOM 8LL)
With the Direct Line Pickup features, you can access a ringing or held call or a call in progress. You can also select a line to use that is not assigned to your telephone.

The Direct Line Pickup features are:

- Direct Line Pickup—Active Line
- Direct Line Pickup—Idle Line

Display Language
All system telephones offer the choice of 3 display language:

- US English
- Canadian French
- Latin American Spanish

Each individual extension may select their own language.

Distinctive / Differentiated Ringing (#308)
PARTNER® Version provides different ring patterns for external calls, internal calls and transferred calls.

Do Not Disturb (FEATURE 01)
Use this feature to be able to press a programmed button to prevent incoming calls for the extension from ringing (lights still flash). When Do Not Disturb is on, outside callers hear ringing while inside callers hear a busy signal (if there is no voicemail). Calls are immediately sent to your voice mailbox.

Doorphones
A doorphone is usually placed near an entrance, to screen visitors. You can connect up to two doorphones to the system.

Emergency Phone Number List (#406)
Create a list of up to 10 telephone numbers that all users can dial regardless of dialing restrictions, provided they have access to an outside line. Typical list entries include fire, police, and emergency services numbers. police, and emergency services numbers.

Extension Hunt Groups (#505)
Use this feature to assign any number of extensions to a Hunt Group. When extensions are in a Hunt Group, an incoming call searches or “hunts” for the longest idle extension that is available. The system supports up to six Hunt Groups

Extension Name Display
The user’s name and extension number is displayed on the display telephone.

External Hotline (#311)
When a user lifts the handset of an external hotline, a predetermined outside number is dialed automatically. The external number might be, for example, a frequently called service bureau. The external hotline must be a single line telephone, not a system telephone, and should not have a dial pad.

Fax Machine Extensions (#601)
Allows you to connect your fax machine to your system, allowing you to share network facilities.

Group Calling - Ring / Page (INTERCOM 7G / INTERCOM *7G)
Use this feature to ring, page, or transfer calls simultaneously to all of the extensions in any one of four Calling Groups. (G represents a Calling Group number from 1–4.) When paging, you hear a beep and can begin speaking. Your voice is heard on the speakers of all idle system telephones in the selected Calling Group. The first extension to answer the call (on ETR sets - by picking up the handset or pressing SPEAKER, on 14xx sets - or by pressing the ANSWER softkey on the display) is connected to the caller.

You can group page employees for general announcements; this is an inexpensive alternative to a paging
system.

Group Hunting - Ring / Page (INTERCOM 77G / INTERCOM *77G)
Use this feature to distribute call volume among extensions in a group, to off-load call activity from a single user. When an intercom or transferred call is placed to a Hunt

Group (where G is a Hunt Group number from 1–6), the system rings or voice-signals the first available extension in the group, passing over busy extensions (or those with Do Not Disturb on) in a circular hunt. If a ringing call is not answered within three rings, the call moves to the next available extension, and so on, until the call is answered or the caller hangs up.

For a voice-signaled call, only the first available extension is signaled; if that extension does not answer, the call does not keep hunting. After an extension receives a Hunt Group call, the next call to the Hunt Group will not ring or voice-signal that extension first unless all other extensions in the group are busy or do not answer.

Group Pickup (INTERCOM 66G)
Use this feature to answer any outside, intercom, or transferred call ringing at an extension in a Pickup Group (where G is a Pickup Group number from 1–4). When a call rings at an extension that is in a Pickup Group, you can answer the call at any extension without knowing which extension or line is ringing and without being in the same Pickup Group. The system can have up to four Pickup Groups.

Hold Reminder
If a call is left on hold for longer than a specified period, the extension will ring back, alerting you that the call is still on hold. This ring continues to sound until the held call is retrieved, or until the caller hangs up.

Hot Dial
This feature enables a user to start dialing a number without first pressing the speaker button or going off-hook. Hot Dial can only be turned off on ETR telephones.

Hotline
This feature allows designated extensions to automatically place an intercom call to another designated extension, when it goes off-hook.

Intercom Dial Codes

- 'I' represents the INTERCOM button.
- 'XX' represents a system extension number.
- 'G' represents a single-digit group number.
- 'LL' represents a system line number.

Intercom Dial Code	Feature
I6XX	Call Pickup
I66G	Group Pickup
I68LL	Direct Line Pickup – Active Line
I70	Loudspeaker Paging
I*70	Simultaneous Paging
I7G	Group Calling – Ring
I*7G	Group Calling – Page
I77G	Group Hunting – Ring
I*77G	Group Hunting – Voice Signal
I8LL	Direct Line Pickup – Idle Line

Intercom Dial Tone (#309)
Use this feature to determine the type of dial tone that the system provides at an extension. It may be necessary to change this setting to Machine (outside line) dial tone for an autodialing device, such as a fax or modem, that has trouble making calls. For example, if you have a modem that checks for outside line dial tone before dialing, use this procedure to change from Regular to Machine dial tone.

Intercom Manual Dialing
Users can make an internal call to another extension by ringing it (dialing it's extension number) or voice signaling the extension by prefixing the extension number with a *.

Line Coverage Extension (#208)
Use this feature to identify an extension as the "owner" of a specific outside line, so the extension can activate Call Coverage or VMS Cover for that line.

Line Ringing Options
This feature specifies the options: Immediate Ring, Delayed Ring or No Ring for incoming calls on a line appearance programmed on the extension.

Log All Calls
Up to 3 extensions can be designated to log all calls, both answered and unanswered, regardless of where they terminated in the system.

Music on Hold (#602)
Connect an audio source to the Music-On-Hold port on the processor module so that a caller hears music when placed on hold when the Music-On-Hold feature has been activated.

Network Time Synchronization (#128)
Use this feature to specify whether you want to synchronize the system clock with the network time delivered by your service provider over Caller ID. If Network Time Synchronization is set to Active, the system will use the network time provided by your service provider; if it is set to Not Active, the system will use the time provided by its system clock.

Night Service
When Night Service is on and a call comes in, all extensions in the Night Service Group ring immediately, regardless of normal Line Ringing settings.

Paging Features (INTERCOM 70 and INTERCOM *70)
- **Loudspeaker Paging** - Users can initiate a call (either manually dialed or intercom auto dialed) to an extension connected to an analog station port that connects them to an external loudspeaker paging system.
- **Simultaneous Paging** is a feature that pages both the external loudspeaker paging system (via the paging port) as well as voice signals all of the extensions in Calling Group 1.

Personal Line Termination
Allows you to assign a line / lines to specific extension / extensions.

Personalized Station Ringing (#323)
Allows you to assign a personalized ring pattern to an extension

POTS (Plain Old Telephone Set) Support
POTS (single line analog telephones) are supported on the system. They are connected to an analog station port on the Combo module; a port on the Phone-2, Phone-8 or Phone-16 module, or to an ETR port.

Privacy (FEATURE 07)
Privacy prevents other users from joining on your call.

Recall (FEATURE 03)
Use this feature to send a timed switch hook flash over the telephone line to “recall” a new dial tone or to access certain central office features, such as Call Waiting or 3-way calling.

Redialing Features
There are 2 features which offer redial functionality:

- Last Number Redial (FEATURE 05): Last Number Redial redials all digits dialed on the last outside call except account codes.
- Save Number Redial (FEATURE 04): Saves into temporary memory the last outside number (up to 28 digits) dialed from a system telephone. Use this feature to save a number before you hang up on a busy or unanswered call. Once saved, the number can be redialed at any time. The number stays in memory until a different one is saved.

Remote Call Forwarding / Mobile Twinning
There are two types of Remote Call Forwarding supported on PARTNER® Version:

- **Remote Call Forwarding (FEATURE 11)** allows a user to send internal and external calls destined for a particular extension to an external phone number (off the system) over an analog trunk. Remote Call Forwarding will not alert the internal extension, but immediately sends the call out to the predefined number.

- **Mobile Twinning** allows a user to send internal and external calls destined for a particular extension to an external phone. The call simultaneously rings at the internal extension and receives the extension’s coverage (including Voicemail) treatment if the call has not been answered. This feature is only available on systems that have digital trunks (SIP, PRI/T1).

Ring on Transfer (#119)
This is used to specify what the caller hears while they are being transferred. If Ring on Transfer is set to Active, callers hear ringing while they are being transferred; if it is set to Not Active callers hear beeps or Music on Hold if this is activated and a music source is connected to the system.

Ringing Line Preference
When an extension is on-hook and the extension is ringing, the user simply goes off-hook to answer the call and is automatically connected to the ringing call. If more than one call is alerting, the system automatically connects the user to the longest ringing call.

Speed Dial Options
PARTNER® Version supports the following types of Speed Dialing:

- **Personal Speed Dial** (FEATURE 80 – 99) Each user can store up to 20 frequently-dialed numbers in their extension. Personal Speed Dial numbers can be dialed quickly by pressing FEATURE (or # at intercom dial tone on a single-line telephone) and the two-digit speed dial code from 80 through 99. The Personal Speed Dial numbers programmed for a particular extension are for use only at that extension.

- **System Speed Dial** (FEATURE 600 – 699) System Speed Dialing is a shared list of up to 100 frequently-dialed numbers up to 28 characters in length. All users on the system may dial a System Speed Dial number by pressing FEATURE (or # at intercom dial tone on a single-line telephone) and the three digit speed dial code, from 600 through 699

Station Lock / Unlock (FEATURE 21)
Station Lock helps to prevent unauthorized people from making outside calls from extensions. Users enter a four-digit code on their telephone dialpad to “lock” your extension. To unlock the phone, the user enters the identical code.

Station Unlock (FEATURE 22)
This feature allows the attendant to force the unlocking of a station.

Station Message Detail Reporting (SMDR)
SMDR is a call reporting feature that provides records of call activity. It is commonly used in many types of business, including Legal, Contact Centers, Sales and Real Estate. Call reporting information allows users to:

- Detect any unauthorized calls.
- Bill clients or projects.
- Bill back by department.
- Reduce telephone costs by identifying the need to change telecommunications services
- Print Caller ID information.

The output is generally sent to a PC running an optional Call Accounting software package.

System Groups
PARTNER® Version supports the following types of groups:

- **Pickup Groups** (4 Groups) (INTERCOM 6xx) When a call rings at an extension assigned to a Pickup Group, a user at any other extension in the system can answer the ringing call by dialing the Pickup Group code. The Pickup Group feature helps when a user needs to answer calls on lines or pools not assigned to his or her telephone.

- **Calling Groups** (4 Groups) (INTERCOM 7x / *7x) A Calling Group is a group of extensions that can be called at the same time. Any user in the system can ring or page all extensions in a Calling Group at the same time or

transfer a call to a Calling Group. The first extension to pick up the call is connected to the caller. A typical use of this feature is to have callers ring into a Calling Group of sales representatives, or to create a “Page All” group.

- **Hunt Groups** (6 Groups) (INTERCOM 77G / *77G) When extensions are in a Hunt Group, an incoming call searches or “hunts” for the first available extension.

- **Night Service Group** (1 Group) (#504) When Night Service is activated and a call comes in, all extensions assigned to the Night Service Group ring immediately, regardless of normal Line Ringing settings.

System Password (#403)
Allows you to define a four-digit password that users can enter from PARTNER telephones to override dialing restrictions (if the extension has access to an outside line) or turn Night Service on and off.

Transfer / Transfer Return
You can transfer calls to an extension or group using the TRANSFER button on a system telephone or the switch-hook on a single-line telephone.

Transferred calls return to the originating extension if they are not answered. You also can program the number of times a transferred call rings before it returns to an extension.

Visual Voice Messaging
Supported on 1400 digital telephones only. Allows the user to manage their voice mail messages via their display. Features supported include:

- access new /old / saved messages.
- next and previous message.
- fast forward and rewind.
- pause message.
- save, delete and copy message to other users of the system.
- change default greeting.
- change password.

VMS Cover (FEATURE 15)
Allows you to turn VMS Cover on and off, routing unanswered intercom and transferred calls for users extensions to the Call Answer Service of the voice messaging system after the specified number of rings

VMS Cover Ring Interval (#321)
Allows you to define the number of times a call rings before it is sent to a user's voice mailbox.

VMS Hunt Delay (#506)
Allows you to determine when outside calls should be answered by the Automated Attendant of the voice messaging system. You can set the system for any number of rings, 0–6. Assigning more rings gives the operator an opportunity to answer calls before they go to the Automated Attendant.

VMS Hunt Delay is programmable on a per-line basis. In addition, you can program this feature so that calls can be handled one way during the day and a different way when the system is in Night Service.

VMS Hunt Schedule Interval (#507)
Allows you to determine when outside lines are covered by the Automated Attendant. Options include all the time, only during Day operation (Night Service is off), only during Night operation (Night Service is on) or Never.

Voice Mailbox Transfer - Direct (FEATURE 14)
Allows you to transfer a call directly into a user's voice mailbox without ringing their extension.

Voice Messaging
The PARTNER® Version has Embedded Messaging included in the system. All extensions are automatically assigned a mailbox. Embedded Messaging also provides:

- A 2-port, single level Automated Attendant
- Separate Morning, Afternoon, Evening and Out of Hours menu greetings with time profiles
- Dial by Name capability
- 15 hours of storage
- Up to 3 minutes per message
- Fast Forward, Rewind, Replay and Skip capability
- Voice Mail to e-mail capability
- Remote message retrieval
- Visual Messaging (on Digital telephones only)

System Programming Features

Code	Feature	Code	Feature
#101	System Date	**#321**	VMS Cover Rings
#103	System Time	**#322**	Remote Call Forwarding
#104	Number of Lines	**#323**	Personalized Station Ringing
#105	Transfer Return Rings	**#399**	Copy Settings
#107	Recall Timer Duration	**#401**	Outgoing Call Restriction
#109	Outside Conference Denial	**#402**	Toll Call Prefix
#119	Ring on Transfer	**#403**	System Password
#123	Backup Programming – Automatic	**#405**	Disallowed List Assignments
#124	Backup Programming – Manual	**#406**	Emergency Phone Number List
#125	Restore Programming	**#407**	Allowed Phone Number Lists
#126	Automatic Daylight/Standard Times	**#408**	Allowed List Assignments
#127	Hold Reminder Tone	**#409**	Forced Account Code List
#128	Network Time Synchronization	**#501**	Pickup Group Extensions
#203	Hold Disconnect Time	**#502**	Calling Group Extensions
#205	Direct Extension Dial Lines	**#503**	Night Service Button
#206	Group Call Distribution	**#504**	Night Service Group Extensions
#208	Line Coverage Extension	**#505**	Hunt Group Extensions
#301	Line Assignment	**#506**	VMS Hunt Delay
#303	Display Language	**#507**	VMS Hunt Schedule
#304	Automatic Extension Privacy	**#601**	Fax Machine Extensions
#305	Abbreviated Ringing	**#602**	Music On Hold
#306	Transfer Return Extension	**#603**	Hotline
#307	Forced Account Code Entry	**#604**	Doorphone Extension 1
#308	Distinctive Ring	**#605**	Doorphone Extension 2
#309	Intercom Dial Tone	**#606**	Doorphone Alert Extensions
#310	Automatic VMS Cover	**#610**	SMDR Output Format
#311	External Hotline	**#612**	Contact Closure Group
#316	Call Waiting	**#613**	Contact Closure Operation Type
#317	Caller ID Log Answered Calls	**#617**	Loudspeaker Paging Line
#318	Caller ID Call Log Line Association	**#728**	System Reset Programming Saved
#320	Call Coverage Rings	**#730**	Remote Administration Password

Application Programming Interface (TAPI)

Application Programming Interface (TAPI) allows new and existing server-based applications from Avaya's Developer Connection (DevConnect) program to work with and enhance the operation of the PARTNER® Version system. The current requirement is to support a screen pop type application called "Pickup IP" which is also supported on IP Office.

The TAPI API only supports 1st Party TAPI applications.

Availability

The Essential Edition - PARTNER® Version will be 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 to be supported are US English (default), Latin American Spanish and Canadian French – the same languages as currently supported on the PARTNER® ACS.

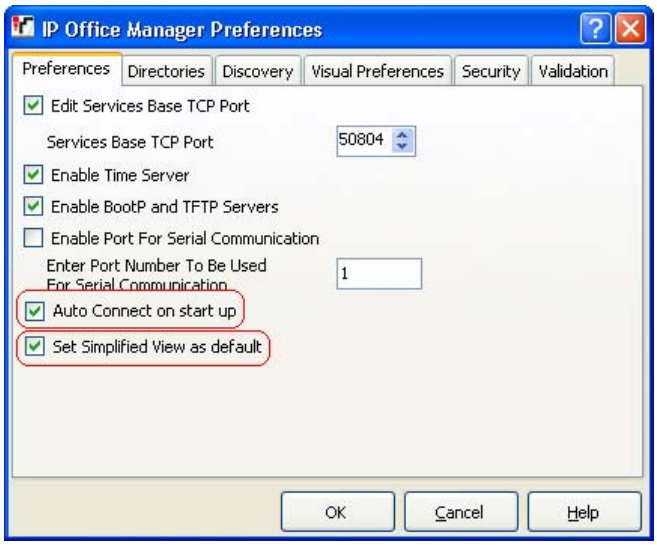
3.2 Simplified Manager

Note: *Supported in Essential Edition - PARTNER® Version only*

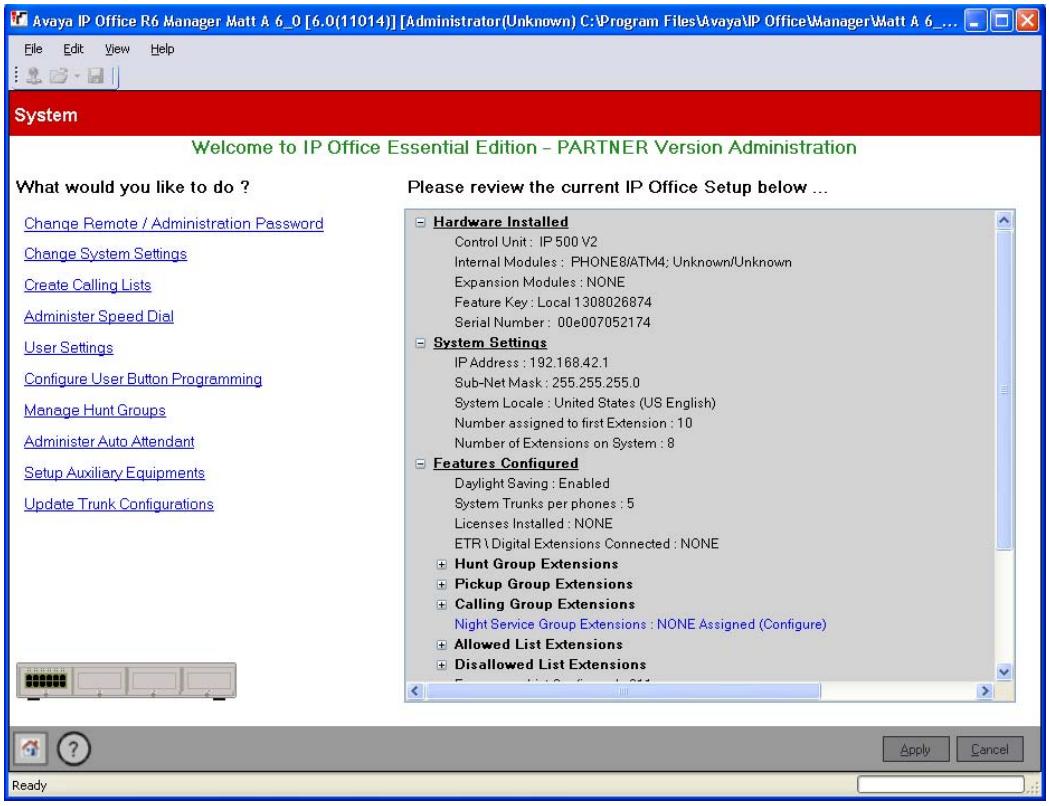
A new mode of operation has been provided within the IP Office Manager, known as Simplified Manager, which is used to administer the system when it is running in Essential Edition - PARTNER® Version.

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 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:



By using the 10 System Hyperlinks, you are able to quickly and easily administer the PARTNER® Version system in a short time. Further details can be found in the IP Office Essential Edition - PARTNER® Version Manager documentation guide posted on the Avaya Support website.

3.3 Licensing Changes

Today there are many different purchasable license material codes for an IP Office solution, including user licenses, port/channel licenses, system licenses, and mixed licenses. In order to simplify the selling of applications the total number of licenses / material codes has been reduced significantly.

With IP Office Release 6.0 there will be only two types of application licenses – System Edition licenses and User Profile licenses. The System Edition licenses enable features and applications system-wide and the User Profile licenses enable features and applications for single users.

The System Edition Licenses available with IP Office Release 6.0 are as follows:

Essential Edition: Introduced with IP Office Release 5, this has no explicit license. It is the normal IP Office mode of operation plus Embedded Messaging functionality. In Release 6.0 this will continue to be the mode of operation for the IP406v2 and IP500, but the IP500v2 will have two Essential Edition Messaging ports included, with

additional ports able to be added through the purchase of optional licenses, up to a total of 6 ports.

Preferred Edition: This license adds a 4-port VoiceMail Pro system to the IP Office system together with Meet Me Conferencing and Basic Call Recording capabilities. The existing VoiceMail Pro 4 additional ports license can be purchased to add increments of 4 ports to the Preferred Edition license, up to the maximum supported by the platform (e.g. 40 ports on IP500).

Note: *The Conference Meet Me feature is only supported on systems with the Preferred Edition license.*

Advanced Edition: The Advanced Edition license can be added to the Preferred Edition license to enable customer service enhancements. It contains the Customer Call Reporter application (for real time and historic reporting), the 3rd party Database Interface (IVR) with Text-to-Speech, Visual Basic Scripting and ContactStore (for search and replay of call recordings).

Note: *A Preferred Edition license is a prerequisite for the Advanced Edition license to be valid.*

There are a number of different User Profile licenses available:

Basic User: By default a user inherits standard IP Office capabilities including a voicemail mailbox. No license is required to be a Basic User. In a default configuration every user is a Basic user.

Office Worker: This license will enable the following functionality for the user:

- one-X Portal for IP Office (with Telecommuter mode disabled)
- Unified Messaging Service (web and email access to voicemail)

Note 1: *No specific System Edition license is required as a prerequisite for this license to be valid. However, the Unified Messaging Service (UMS) function will not work without the Preferred Edition license.*

Note 2: *For migration purposes, any existing Phone Manager Pro license can enable the Office Worker functionality (instead of Phone Manager Pro).*

Mobile Worker: This license will enable the following functionality for the user:

- Mobility including Mobile Twinning, Mobile Call Control and one-X Mobile Client support
- Text-to-Speech (TTS)

Note: *No specific System Edition license is required as a prerequisite for this license to be valid. However, the TTS function will not work without the Preferred Edition license.*

Teleworker: This license will enable the following functionality for the user:

- one-X Portal for IP Office
- Unified Messaging Service (web and email access to voicemail)
- IP Office Video Softphone

Note: *No specific System Edition license is required as a prerequisite for this license to be valid. However, the UMS function will not work without the Preferred Edition license.*

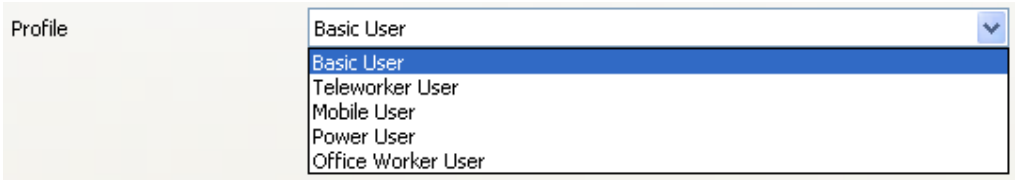
Power User: This license will enable the following functionality for the user:

- Mobility including Mobile Twinning, Mobile Call Control and one-X Mobile Client support
- one-X Portal for IP Office
- IP Office Video Softphone
- Unified Messaging Service (web and email access to voicemail)
- Text-to-Speech (TTS)

Note: *No specific System Edition license is required as a prerequisite for this license to be valid. However, the UMS and TTS functions will not work without the Preferred Edition license.*

The above profiles are mutually exclusive, only one may be selected per user.

The user profile is assigned to the user in Manager from the main User form. Before the profile can be set for a user the license needs to be added to the configuration. Once this is done a drop down box labeled “Profile” will show a list of the user profiles that are available on the system. The number of users that can be configured for each profile other than Basic User is controlled by the user licenses present in the configuration.



When a profile is allocated to a user the relevant check boxes, to enable the allowed features, within the user’s configuration are automatically enabled. Some of these options are visible on the main User form, others will be on different tabs, like the mobile options for instance. Individual features can be de-selected but that does not make them available for allocation to another user.

e.g. A user could have their profile set to Power User but you might want to stop them from using the mobility features. In this case you simply assign the profile and then go to the Mobility tab and deselect the Mobility Features option.

Loss of TTS Email Reading Functionality: In releases prior to 6.0 there was no special user license/user entitlement for e-mail reading, in Release 6.0 a User has to be a Power or Mobile User to have this feature. This means that sites upgrading to

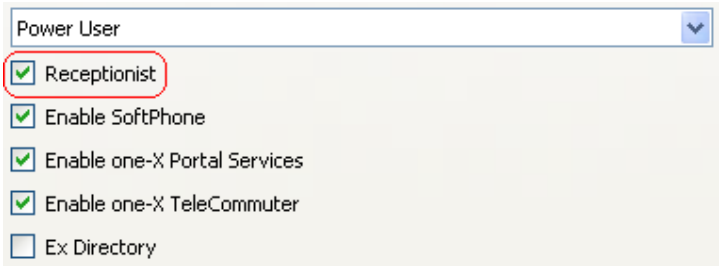
6.0 who had this facility before will no longer has access to this until the relevant profile licenses are added to the configuration and assigned to users.

In addition to the above user profiles, the user may have one of the licenses below.

Receptionist: This license will enable the following functionality for the user:

- Functionality found in application previously known as SoftConsole

Note 1: *When a system is upgraded to Release 6.0 the license that was previously called SoftConsole is renamed to Receptionist. No users are able to run the SoftConsole application until it has been explicitly set in the IP Office configuration that they can do this. On the main user form, just underneath the profile drop down, is a checkbox option called Receptionist. Selecting this option allows the user to run the SoftConsole application. Up to 4 users can be configured as receptionists if sufficient licenses are available.*



Note 2: *No specific System Package license is required as a prerequisite for this license to be valid.*

Customer Service Agent: This license will enable the following functionality for the user:

- CCR Agent

This license is concurrent and is allocated on a first-come, first-served basis.

Note: *An Advanced Edition system license or legacy CCR SUP license or* a CCC Supervisor and CCR CCC Upgrade license *is required as a prerequisite for this license to be valid.*

Customer Service Supervisor: This license will enable the following functionality for the user:

- CCR Supervisor

This license is concurrent and is allocated on a first-come, first-served basis.

Note: *An Advanced Edition system license or legacy CCR SUP license or* a CCC Supervisor and CCR CCC Upgrade license *is required as a prerequisite for this license to be valid.*

Distributed User Licensing: All of the new IP Office User Profile licenses (except Customer Service Agent and Customer Service Supervisor) will follow the user when hot-desking to another IP Office system in the Small Community Network (SCN), so that there is no need to purchase multiple licenses for the same user.

This also helps greatly in failover scenarios where users have to fallback to an alternative IP Office system and want to continue to access their personal productivity applications (improves resilience of user applications).

Avaya IP Telephone licensing: IP500 and IP500v2 ONLY
In IP Office Release 6.0 all Avaya IP telephones will need to be licensed (this includes desktop phones like 5600, 1600 or 9600 IP phones as well as wireless phones like 3600 and 3700 series). Digital phones will not be licensed. This applies to IP500 and IP500v2 only. IP406v2 and IP412 systems will not require IP Telephone licenses for Avaya IP Phones.

Removal of VCM channel licenses: In order to reduce product complexity, all VCM licenses will also be removed. VCM 32 and VCM 64 cards will always deliver 32 and 64 channels respectively (subject to existing codec restrictions). Each VCM 32 and VCM 64 card comes with 12 Avaya IP endpoint licenses enabled. Once the system is upgraded the license will show in the IP Office configuration as a “VCM Channel Migration License”.

Note: *The Combination card does not come with any Avaya IP endpoint licenses.*

Upgrade of IP500 systems from earlier releases: IP500 customers upgrading from earlier releases to Release 6.0 will get a smooth migration where each VCM channel license they already have will be automatically converted into 3 Avaya IP Phone licenses.

Example: A customer already has a VCM32 card installed and a VCM license for 28 channels. When the system is upgraded to Release 6.0 the 4 free VCM channels that are provided for the VCM32/64 cards are converted into 12 Avaya IP endpoint licenses (4*3). The additional 28 channels that they licensed are converted into 84 Avaya IP Endpoint licenses (28*3), giving a total of 96 Avaya IP endpoint licenses.

If the VCM licenses are ever removed from the configuration then the Avaya IP endpoint licenses available will be reduced, leaving only 12 licenses.

When there are insufficient Avaya IP endpoint licenses any IP phone that can not get a license will show “No License Available” on its display. This applies to all telephone types expect DECT.

If you are using a legacy VCM on a carrier card then all channels available on the VCM card are converted to Avaya IP Endpoint licenses. For example if you have a VCM16 card this will give you 48 Avaya IP Endpoint licenses (16*3).

If after upgrading to release 6.0 you do not have enough Avaya IP endpoint licenses available through this conversion process then you will need to purchase further licenses.

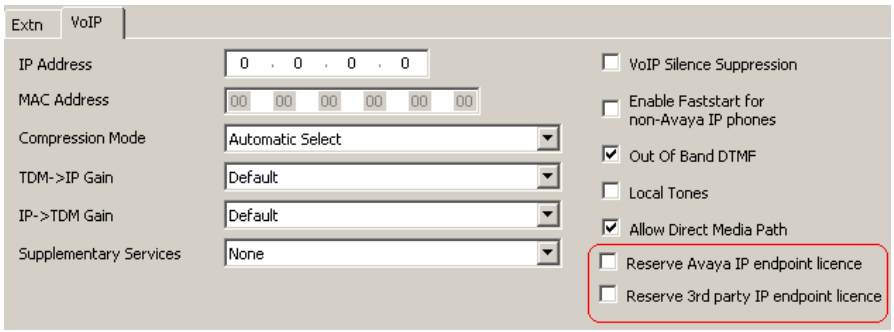
IP Endpoint License Reservation: For IP Office Release 6.0, Avaya IP phones require an Avaya IP Endpoint license in order to connect to the IP Office. Normally licenses are issued in the order that devices connect. License Reservation allows an extension to be pre-licensed before the device has connected. Licenses can be reserved for H323, IP DECT and SIP extensions.

Every IP Endpoint will register with the system, IP Office will then check if the device has a license reserved, if it has the phone will be allocated that license and will connect to the system and can be used as normal. If the device does not have a reserved license IP Office then checks the availability of non-reserved IP Endpoint licenses. Again if a licence can be allocated then the phone will be allocated that license and will connect and can be used as normal. If there are no licenses available then the phone will stay registered but will display “No License Available” on the display and can only be used to make emergency calls that match any dial emergency short codes.

Note: *The IP Endpoint license is allocated to the Extension, not to the User.*

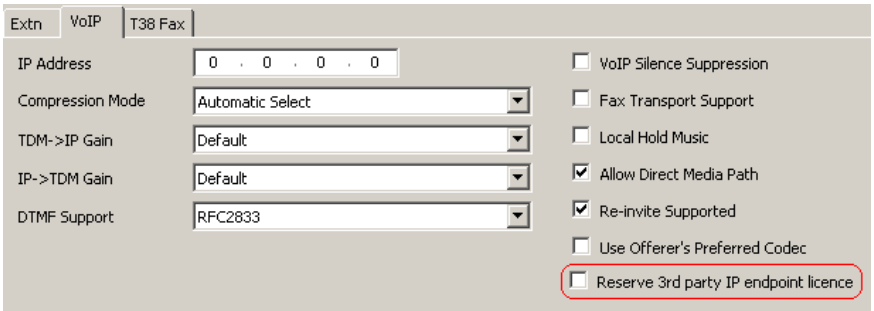
For H323 extensions there are 2 new checkboxes (default unchecked, can be checked/unchecked independently):

- Reserve Avaya IP endpoint license
- Reserve 3rd party IP endpoint license



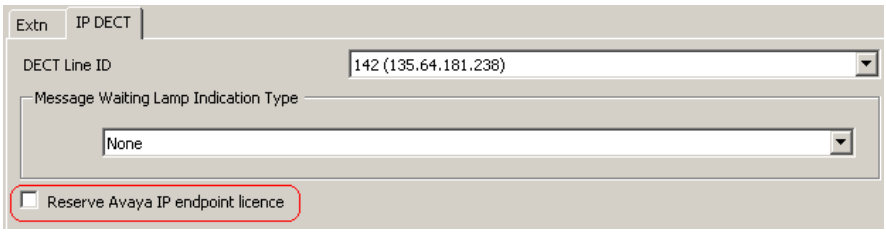
For SIP extensions, there is 1 new checkbox (default unchecked):

- Reserve 3rd party IP endpoint license



For IP DECT extensions there is 1 new checkbox (default unchecked):

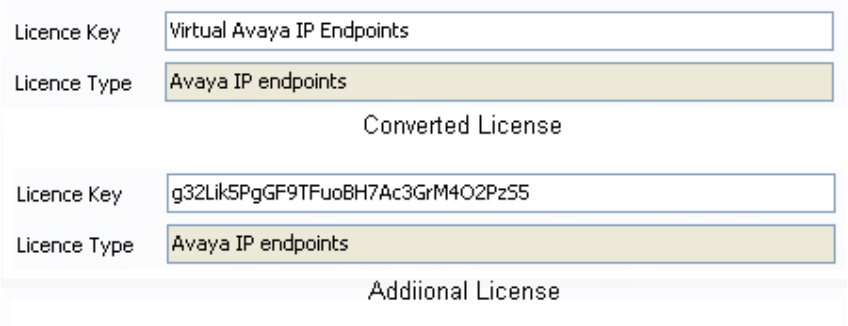
- Reserve Avaya IP endpoint license



Manager will restrict how many licenses can be reserved, the same way it already does for Mobility licenses. It will display a warning message if the number of reserved Avaya or 3rd party IP Endpoints exceeds the number of corresponding licenses present on the system.

Note: *Changing values of these fields will require a reboot.*

Converted Avaya IP Endpoint Licenses: Licenses which have been derived through a conversion from the VCM channel license, or a conversion of legacy VCM channels, will show in the IP Office configuration as "Virtual Avaya IP Endpoints" in the license key field, whereas those licenses that have been purchased as extras will show the license key string in that field.



Note: *Third party IP phones, e.g. SIP devices, do not require an Avaya IP phone license. The third party IP Endpoint license is all that is required.*

Emergency Services calls on unlicensed phones: An unlicensed phone can make emergency calls on an outside line using the Dial Emergency short code. An unlicensed phone will not be able to receive any calls.

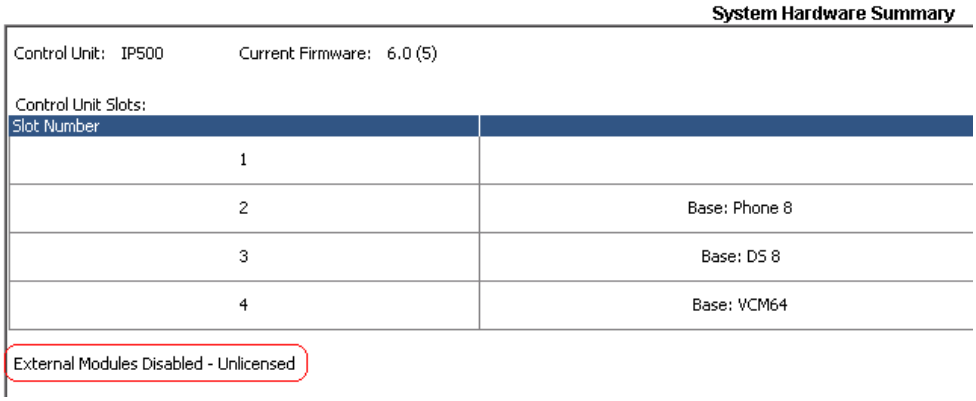
IP Phone licenses on backup gatekeepers: When an IP Phone re-registers to a secondary IP Office on the failure of the primary control unit, the second system will allow it to operate indefinitely as a "guest", but only until the system resets. Licenses will never be consumed for a guest IP phone.

Removal of VPN license: In IP Office Release 6.0 the existing VPN licenses will no longer be required. This is applicable to all platforms. VPN phones (4600/5600/9600 series IP Phones) will be able to register either without licenses (IP400 platforms) or using Avaya IP phone licenses only (IP500 platforms).

Upgrade License: From Release 6 onwards IP Office needs a valid software license to run. For new installations using the IP500v2 control unit no license is required, this also applies to Essential Edition - PARTNER® Version systems. To upgrade an older IP Office system to Release 6 a software upgrade license is required. This comes in three variants:

- Standard upgrade to Release 6.0 (IP406v2/IP412/IP500)
- Upgrade to Release 6.0 for small sites up to 32 extensions (IP500 only)*
- Multiple upgrades for several future releases (up to and including Release 10.x)

Note: *The small site upgrade license is only for use on single box IP500 systems and has a limit of 32 extensions. Expansion modules are NOT SUPPORTED with this license. This can be seen in SSA, see below:*



System Hardware Summary

Control Unit: IP500 Current Firmware: 6.0 (5)

Control Unit Slots:

Slot Number	
1	
2	Base: Phone 8
3	Base: DS 8
4	Base: VCM64

External Modules Disabled - Unlicensed

Once the limit of 32 extensions is reached no further extensions will be permitted to operate.

Extensions are allocated each time the system is re-booted and in the order of:

- *TDM (analogue/digital) on Base Cards starting at the first (left most when looking at the front)*
- *IP Extensions which are explicitly configured*
- *Any auto created IP extensions (including IP DECT). Once the limit is reached auto create is disabled.*

The Avaya IP Office Video Softphone will not count against the extension count and is only permitted as long as the user has a base extension that is operational on the system.

Other License States: When upgrading to Release 6.0 you may notice licenses in states other than valid and invalid. It is possible that you will also see Obsolete and Dormant.

- An Obsolete license is one that is no longer required, for example the VPN Phone license.

- A Dormant license is one that is valid but has a dependency that is either missing or invalid. For example if you had CCC Agent licenses in a 5.0 system these would become Dormant if there was no CCC CCR Upgrade License present when you upgraded to Release 6.0.

Trial Licenses: All new licenses will have a 60-day trial license introduced. All User Profile licenses will enable 5 users except for the following:

- Receptionist – enables 2
- Customer Service Supervisor – enables 1

A "Try & Buy" 60-day trial license for 5 Avaya IP phones will be offered. When the license expires the phone(s) will continue to work until a system reset or configuration merge.

3.4 1000 User SCN

IP Office Release 6.0 will increase the supported number of users on a Small Community Network from 500 to 1000 users. In mixed networks (e.g. with Release 5 systems in the network), the limit will remain at 500 users.

3.5 New Telephone Features

Note: *Applies to 1400/1600/9600 only*
En-bloc Dialing: IP Office Release 6.0 supports en-bloc-dialing on the 1400, 1600 and 9600 series telephones. This is supported in PARTNER® Version (1400 only) and IP Office mode.

If set up, via the telephone's own menu (Features/Call Settings), when dialing a number without lifting the handset, the phone does not automatically go off-hook (in speakerphone mode) but just displays the dialed digits in the display. A user is able to compose the dialed digits, e.g. delete or insert numbers. After they have completed dialing they have different options to make the call:

- Lift the handset
- Press the headset or speakerphone button
- Press the "OK" button
- Press an idle Call Appearance button
- Press an idle Line Appearance button
- Press an idle Bridged Appearance button
- Press the soft key labeled "Call"

Call Park Button Enhancement: The operation of the Call Park button has been enhanced to display details of parked calls. This allows the user to select whether to un-park a call or to cancel the action.

Phone Inactivity Timers - Auto Return and Auto Lock: The phones now support two inactivity timers. These can be adjusted by the user through the telephone's menus. For 9600 Series phones these are in addition to the phone screen saver options.

- **Auto Return:** When this timer expires, the phone will exit any menu and return to it normal idle phone display.
- **Auto Lock:** When this timer expires, the phone will be automatically locked.

Name Display (1400 and 1600 Series only)*:* Through the telephone's menus the user can select to have their name displayed rather than the extension number when the phone is idle.

Visual Voice: The operation of visual voice has been enhanced. After selecting a messages category (New, Old or Saved) playback no longer starts automatically. Instead the user can scroll through a listing of the messages and select which message to play or delete.

3.6 SIP Trunking Enhancements

The following SIP Trunking enhancements have been delivered in IP Office Release 6.0.

SIP Trunk Prefixes: The SIP line settings now include fields for defining the prefixes to be added to the caller's number. This is similar to the prefix fields used by other external line types supported by IP Office. On systems where a prefix is required to make external calls, adding the prefix to the number received on incoming calls allows the call records shown in call logs and other areas to be used for return calls.

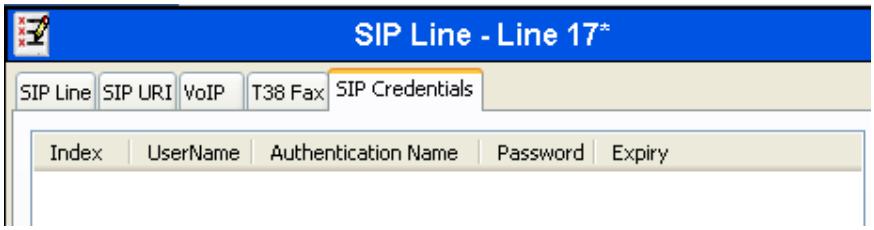
SIP URI Parameters for Hunt Groups and Voicemail: Currently if there is a SIP trunk on the system that has at least one channel configured to use User Data, the SIP tab in Users is enabled. In IP Office Release 6.0 the same tab has been added to Hunt Groups. IP Office Release 6.0 also adds the same fields to the System | Voicemail form. “Use User Data” has also been renamed to “Use Internal Data”.

Support for SIP session timers: IP Office Release 6.0 implements SIP session timers, as per rfc4028. SIP session timers are a keep alive mechanism for SIP sessions. Periodically session refresh requests are sent to keep the session alive. If a session refresh request is not received before the session timer interval passes, the session is considered terminated.

IP Office Release 6.0 adds support for Session-Expires header field and Min-SE header field using UPDATE request. IP Office does not use this method, but it will respond appropriately if the other side requests this. IP Office will dynamically detect whether the trunk (or phone) supports the UPDATE method, and if it does, it will use UPDATE instead of re-INVITE in order to refresh the session. IP Office will not support Min-SE values less then 1 minute.

Multiple account credentials per SIP trunk – IP Office Release 6.0 will support multiple account credentials per SIP trunk. Previously it was only possible to add two sets of ITSP authentication parameters per SIP trunk, now it is possible to add up to a maximum of 30 authentication parameters (name/password) per SIP trunk.

To achieve this, SIP credentials have been moved from the main SIP Line tab to a separate SIP Credentials tab.



SIP trunk Out of Service check – IP Office Release 6.0 is able to detect whether a SIP trunk is in service. Checking that SIP trunks are in service ensures that outgoing call routing is not delayed waiting for response on a SIP trunk that is not currently useable.

- For both UDP and TCP trunks, the OPTIONS message will be sent at least every 5 minutes. If no reply is received the trunk will be taken out of service.
- For TCP trunks, breakage of the TCP connection will cause the trunk to go out of service (not applicable to SES trunks).
- For trunks using DNS, if the IP address is not resolved or the DNS resolution has expired, the trunk is taken out of service.

Note: *Trunks that are out of service will be retried every 2 minutes.*

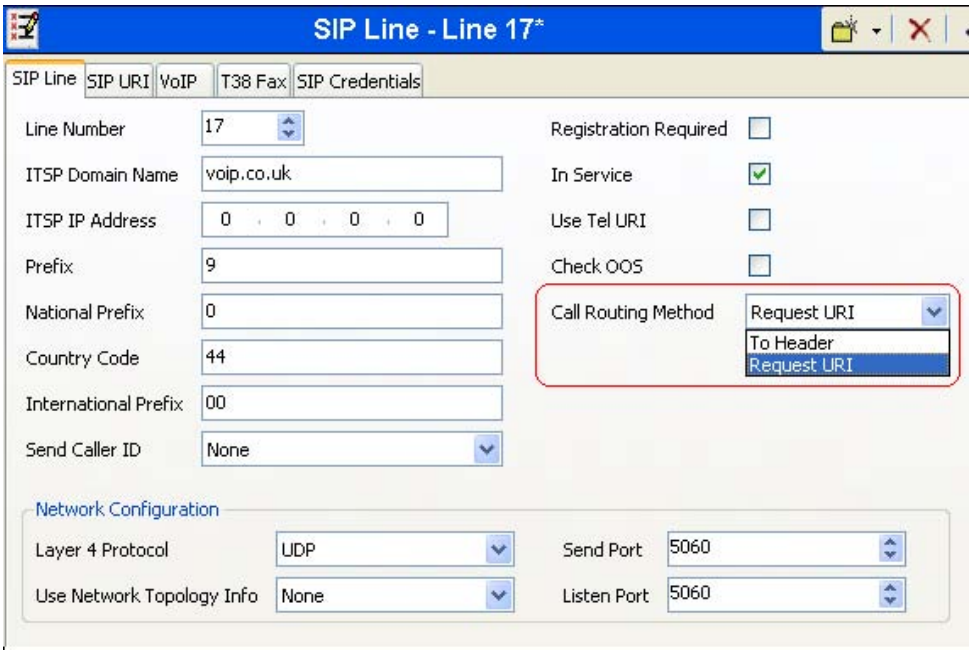
The Out of Service check is configurable on each SIP trunk, in case the SIP trunk does not support the OPTIONS message for example. A new check box on the SIP line tab “Check OOS” is used to enable this feature. For a configuration that has been upgraded to IP Office Release 6.0 from a previous release this option will be off by default. For any new SIP lines added the default setting will be on.

SIP Trunk Registration using DNS: IP Office Release 6.0 allows the configuration of a SIP line using a domain name as an alternative to using a fixed IP address. This relies on the IP Office system being configured correctly to be able to connect to a domain name server.

SIP Diversion Header: IP Office Release 6.0 now includes Diversion Header as one of the Send Caller ID options.

Note: *If the system twinning settings (Send original calling party information for mobile twinning or Calling party information for mobile twinning) are set then the feature will not work. This also applies if Remote Party ID or P Asserted ID are selected, this is a change from how these 2 options interacted with the system twinning settings in Release 5. This should be taken into consideration when upgrading to Release 6.0 if these features were previously being used.*

Call Routing Method: IP Office Release 6.0 allows each SIP trunk to be configured to route the incoming call using the Request URI or the string part of the To Header. This setting is configured using a new drop down menu option on the SIP line tab.



Most service providers send the same information in the Request URI and the To Header, but there are some that send different information. If you have a problem routing your incoming SIP calls then you can check what is being sent using the sysmon application. Prior to IP Office Release 6.0 the number in the To Header was used to route the call.

The following is an example of an incoming SIP call:

SIP Rx: UDP 193.203.210.38:5060 -> 192.168.42.85:5060
INVITE sip:987654@192.168.42.85:5060;transport=UDP SIP/2.0
Via: SIP/2.0/UDP 193.203.210.38:5060;branch=z9hG4bK-sYn-0-9bff6b128daf
Max-Forwards: 8
Contact: <sip:193.203.210.38:5060>
From: <sip:01707299900@voip.co.uk;user=dialstring>;tag=25dd0408cd
To: <sip:01131234567@voip.co.uk;user=dialstring>
Call-ID: 43d7c106-0bf9-11df-a0a2-3993852fea92
CSeq: 1726121442 INVITE
Expires: 120
Allow: INVITE, ACK, OPTIONS, BYE, CANCEL
Content-Type: application/sdp
Content-Length: 556

In the example above the number being presented in the Request URI, which is the line starting “INVITE”, is 987654. However the number being presented in the “To Header”, which is the line starting “To:” is 01131234567.

For a configuration that has been upgraded to IP Office Release 6.0 from a previous release this option will set to “To Header” by default. For any new SIP lines added the default setting will be “Request URI”.

3.7 ISDN Redirecting Number Enhancement

Note: *Only supported on IP500 / IP500v2*

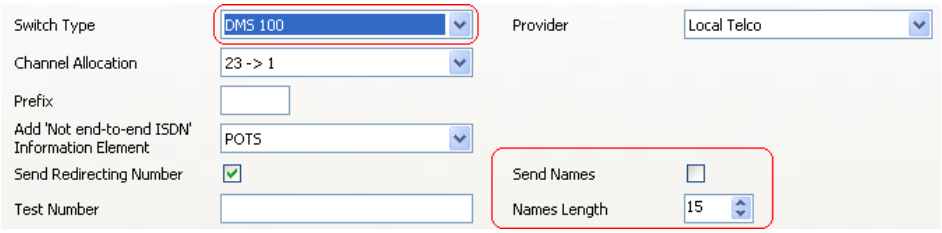
This option, when enabled, populates the ISDN “Redirecting Number” field when transferring off-switch. (Service provider will need to support this).

Where supported, on twinned calls the caller ID of the original call is passed through to the twinning destination. This option is only used for twinned calls, not for forwarded calls. The feature is enabled via the “Send Redirecting Number” checkbox on the Line form.

Note: *If the system twinning settings (Send original calling party information for mobile twinning or Calling party information for mobile twinning) are set then the feature will not work.*

3.8 Send Calling Party Name T1 PRI DMS 100

A new feature has been added in IP Office Release 6.0 for a T1 PRI line when the switch type is set to DMS 100. When the switch type is set to DMS 100 2 additional fields are displayed on the line form.



If send names is checked the name of the calling party is sent using the Display Information Element.

If checked IP Office will send the user's name, unless the user is set Ex Directory. The user's name will come from User Full Name, with Username as a fallback.

The length of the name being sent can also be controlled with 15 characters being the maximum length.

For Mobile Twinning calls, where the system has been configured to forward incoming CallerID IP Office will also forward any received name, as long as it has been received by the time the Setup is sent for the mobile twin call (some US PRIs deliver the incoming name in a Facility message after the Setup).

It is also possible to use a short code character, z or Z, to set the User name, similar to the existing “s” for calling party number. The character z must be immediately followed by a quoted string, using double quotes. E.g. z“My Name”

3.9 Mobility Callback

Note: *Only supported on IP500 / IP500v2*

The mobility callback feature, being introduced with IP Office Release 6.0, provides the ability to call-back the mobility user after the user calls a pre-configured DID/DDI number and then clears the call.

A new “Mobile Callback” checkbox has been introduced in the Mobility Features tab for Users to enable use of this feature.

A new FNE service (FNE33) is used for mobility callback. When a mobility user calls the FNE33 service for mobility callback the call is not answered but the line is signaled to indicate that the incoming call is ringing. The IP Office system will then wait for the user to drop the call. If the incoming call is dropped within 30 seconds, callback will be initiated after 5 seconds using the user’s Twinned Mobile Number.

If the incoming call is not dropped after 30 seconds, the incoming call will be cleared and IP Office will exit the FNE service operation. If the call is answered within the User's No Answer time (but outside of the User's Mobile Answer Guard time), IP Office with proceed with the FNE service operation. If the callback fails on targeting (e.g. line/user busy) or the callback is answered within the Mobile Answer Guard time, or has not been answered before the User's No Answer time, the callback will be cleared and IP Office will exit the FNE service operation. No re-targeting will be applied.

If the callback call is answered then dial tone is presented and the mobility user is then able to dial an internal or external number. If no dialed digit is received within 30 seconds the call will be cleared.

3.10 Persistent SMDR storage

Note: *Only supported on IP500 / IP500v2*

In IP Office Release 6.0 the operation of SMDR on IP500 and IP500v2 control units has been enhanced to store any buffered SMDR records (max 3000) during any controlled IP Office power downs or reboots. The current buffer is also stored at approximately midnight and midday.

3.11 IP Office Essential Edition Messaging – Direct Extension Dialing

IP Office Release 6.0 allows direct extension dialing from an Essential Edition Embedded Auto Attendant. This allows the caller to immediately dial the required extension number without first entering a menu option. This is enabled in Manager via a checkbox “Direct Dial-By-Number” on the Auto Attendant tab. This is disabled by default for backwards compatibility.

When this option is checked it affects the operation of any key presses in the auto attendant menu set to use the “DialByNumber” action.

- If checked, the key press for the action is included in any following digits dialed by the caller for IP Office extension matching. For example, if 2 is set in the actions to DialByNumber, a caller can dial 201 for extension 201.

- If not checked, the key press for the action is not included in any following digits dialed by the caller for IP Office extension matching. For example, if 2 is set in the actions to DialByNumber, a caller must dial 2 and then 201 for extension 201.

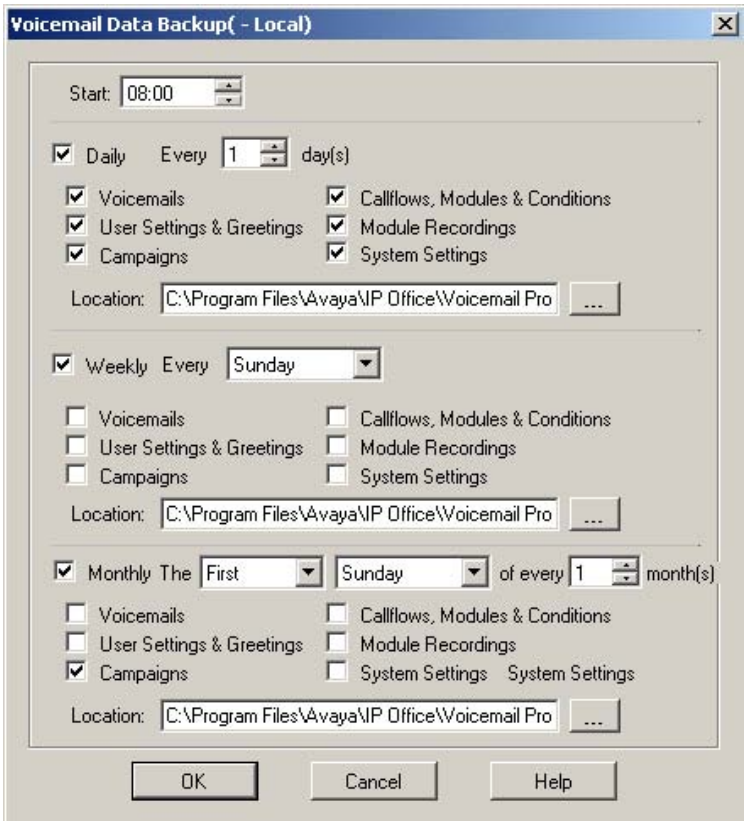
4 IP Office Preferred Edition (formerly VoiceMail Pro)

4.1 Exchange 2007 Integration Enhancements

When VoiceMail Pro is configured so that Exchange 2007 is used as the mail store then the Web Services Interface now provides the ability to forward existing messages in to the Exchange mail store. Such messages are formatted as 'voicemail' in contrast to emails, which enables Exchange to handle them differently, for example showing a voicemail symbol in the inbox view instead of the standard email symbol.

4.2 House Keeping and Data Backup

For increased resilience, Preferred Edition housekeeping has been supplemented with the ability to provision VoiceMail Pro data backups. This allows an administrator to schedule daily, weekly and/or monthly Data backups, providing the ability for the scope of the data to be backed up to be selected. Once data has been backed up the administrator is then able to choose between backup sets to be restored back to the VoiceMail Pro server, if so desired.

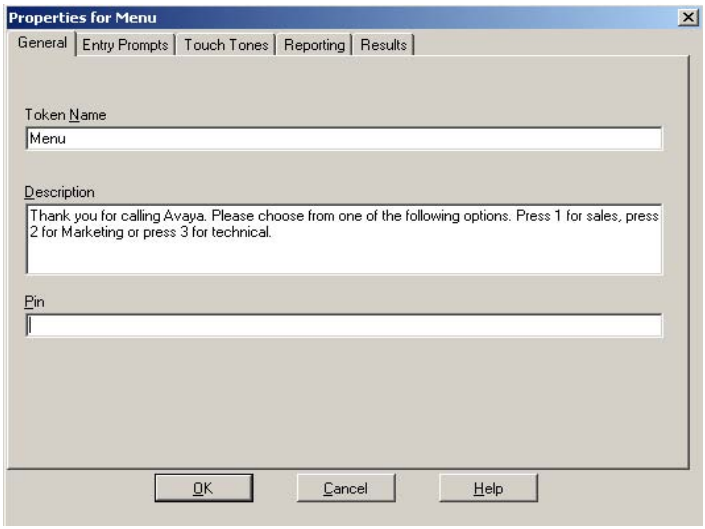


In addition, the housekeeping capability will introduce a separate duration for new and old call recordings as well as deleted messages before they are deleted from the system. Messages can be automatically marked for deletion and moved to the deleted state from their current states based upon the housekeeping settings. The configured duration for deleted messages will dictate how long these messages remain on the system before being permanently deleted; the default setting is 24 Hours.

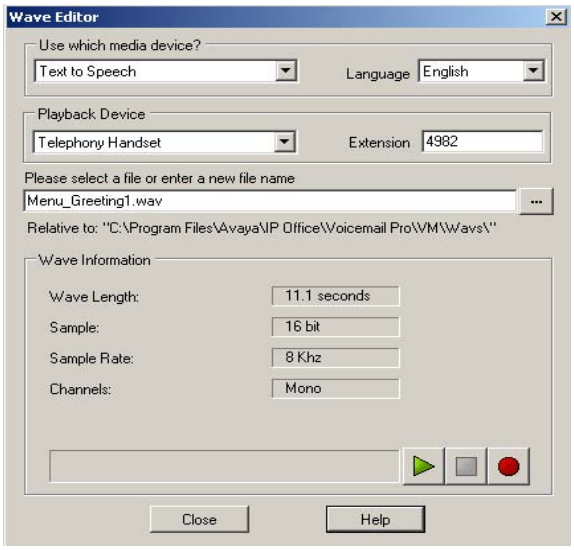
4.3 Text to Speech Announcements

VoiceMail Pro Release 6.0 will supplement the announcement / prompt recording capability provided for system administrators constructing Callflows / auto attendants, with the ability for them to be generated via the systems Text To Speech capability, if licensed. This will allow call flows to be generated with a consistent 'voice artist' and reduce call flow generation costs.

This implementation gives the administrator the ability to select a TTS option that will utilize the Actions Description property to generate the announcement to play when the action is reached in the call flow. The description field text is used to generate a wav file by the TTS engine.



Once you have constructed the text of the announcement you then click on the Entry Prompts tab and click on the ✚ icon to open the wave editor. From the "Use which media device?" drop down at the top of the form select Text to Speech. Enter a new file name and then click on the record button. Once the greeting has been recorded the Wave Information fields will be filled in.



If you want to listen to the announcement that has been created you need to close down the wave editor, then re-open it and select Telephony Handset in from the “Use which media device?” drop down.

Once generated the wav file is set as the prompt to play whenever the action is reached in the call flow. If however the description text field is changed from when the TTS generated wav file was created, then a new file will need to be generated.

4.4 Graceful shutdown

In the current releases of VoiceMail Pro, an administrator is not provided with a capability of disabling new calls from being presented to voicemail, whilst waiting for currently active calls to terminate. Basically, short of rerouting all calls from the IP Office, if calls continually come into the system the administrator will at some stage be forced to shutdown the VoiceMail Pro whilst calls are active.

This new feature allows the administrator to request that no more calls be placed to the VoiceMail Pro, allowing all active sessions to complete before the VoiceMail Pro shutdown is initiated.

4.5 Client Enhancements

A number of VoiceMail Pro client enhancements have been added in this release.

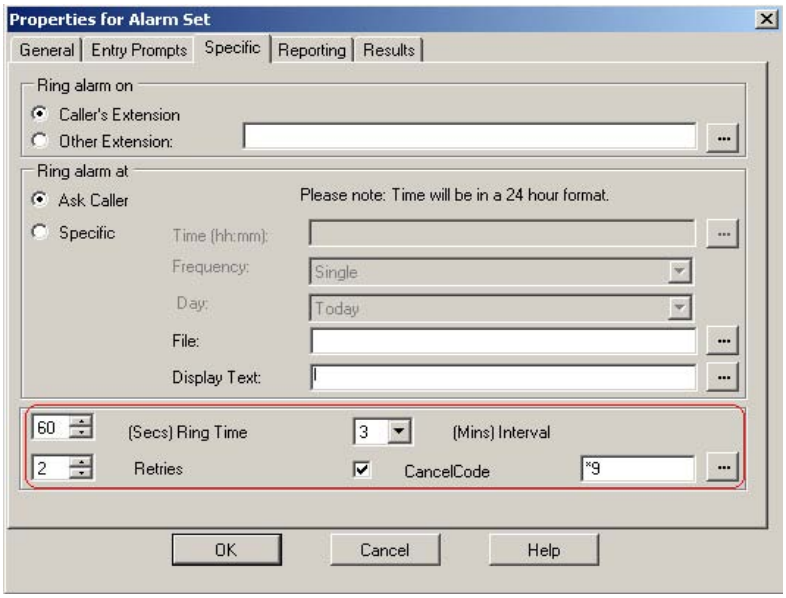
The Alarm Action specific tab has been modified to accommodate additional parameters:

- **Ring Time**: This is how long the Alarm will ring the target number before giving up. The units will be in Seconds, with a range from 5 to 120 seconds. The Default value will be 60 Seconds.

- **Retries**: This is the number of retries that will be made at calling the number. Range: 0 to 10. Default value ‘0’.

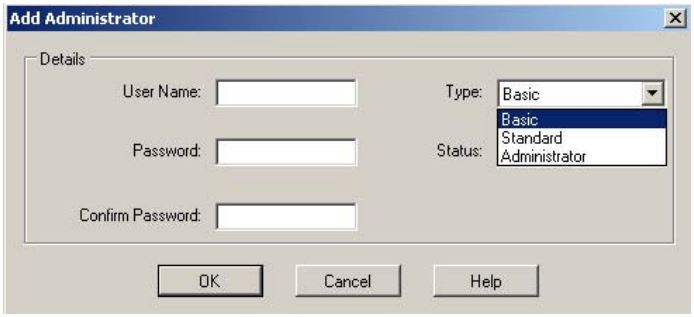
- **Interval**: When retries is set at ‘0’, this is grayed out. This is the time after the previous alarm call finished before the next attempt. Range 1 to 15 Minutes, default value of ‘1’.

- **Cancel Code**: When this is disabled the action of answering the call will cancel the Alarm. When checked a DTMF key or sequence of keys, will be required to cancel the alarm and retries.



Note: *When adding an Alarm Set Action by a right click add via the VoiceMail Pro Client Alarm Queues, the same additional parameters will be presented.*

When using the VoiceMail Pro Client interface to create Administrative users, a new Administrative type of “Basic” has been added.



A Basic level Administrator will be able to:

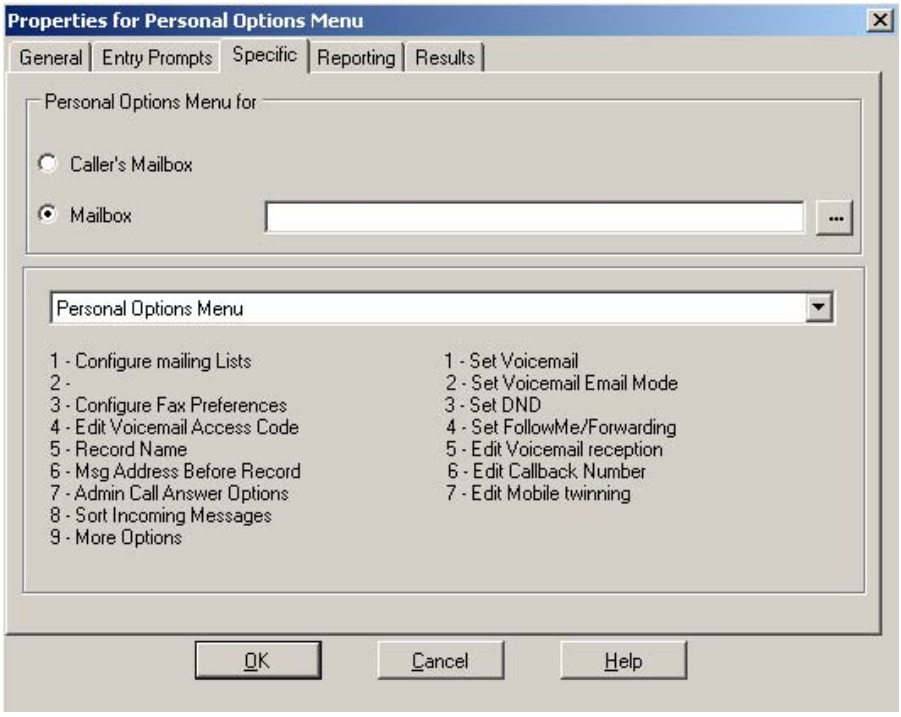
- View the user mailboxes and reset.
- View the Alarms list, and add/remove/modify Alarms from the Alarm List.
- View the User Variable list.
- View Callflows, but not modify them.
- View system Settings but not modify them.
- View Callflows but not modify them.

4.6 Enhanced User Personal Configuration Options

The enhancements in VoiceMail Pro Release 6.0 allow all user configuration options to be configured via the Intuity mode Personal Options Menu, and / or the Play Configuration Menu.

The 'Play Configuration Menu' Action has been renamed to the 'Personal Options Menu' and for backwards compatibility supports the current mode of operation with legacy call flows. If a new call flow is constructed using the R6.0 VoiceMail Pro client, then by default the new capabilities are enabled.

The system administrator can also choose between the legacy mode or new enhanced mode using a dropdown list box in the specific tab of the action.



The additional options allow the user to:

- Activate / Deactivate Voicemail
- Configure the Voicemail to email mode
- Activate / Deactivate Do Not Disturb Feature
- Configure the Follow-Me / Forwarding Options
- Configure the Voicemail reception number
- Configure the Callback Options
- Configure the Mobile Twinning Options

4.7 Small Community Network Resilience Enhancements

Note: *This feature is not supported at GA but will be available during the first half of 2010.*

To reinforce the resilient Small Community Networking (SCN) capabilities introduced in Release 5, VoiceMail Pro will become part of the resilience offering shortly after Release 6.0 becomes available.

There are three configurations that will be supported:

- A single Preferred Edition (VoiceMail Pro) server provisioning a Primary & Backup IP Office.
- Two Preferred Edition servers: Primary Preferred Edition plus Primary IP Office with a Backup Preferred Edition & Backup IP Office.
- Multiple Preferred Edition servers: Primary Preferred Edition plus Primary IP Office, a Backup Preferred Edition plus Backup IP Office with additional Distributed Preferred Editions on an SCN.

Note: *The first configuration was delivered in Release 5, the second and third configurations are being delivered in Release 6.*

4.8 Distributed Voice Messaging (Preferred Edition)

Note: *This feature is not supported at GA but will be available during the first half of 2010.*

There has been a growing requirement for the SCN solutions that are on offer today to support multiple voice messaging systems, distributed on the SCN. This will allow local Preferred Edition resources to be utilized, providing the ability for Voice port usage to be distributed as required by the system topology across sites, minimize bandwidth required across the WAN and provide a level of resilience which is not currently available.

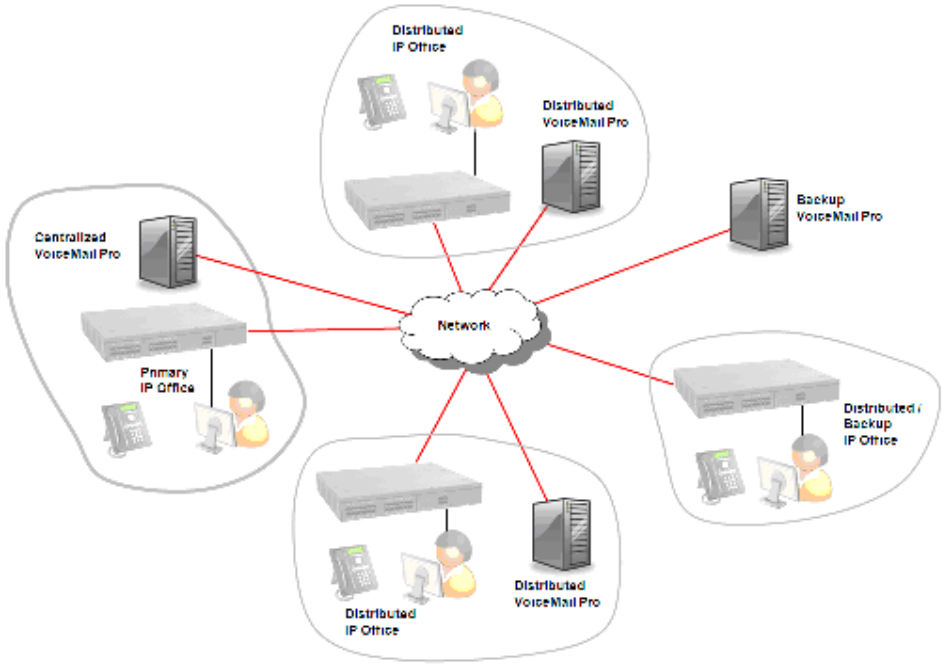
The SCN arrangement, as at present will support a single centralized Preferred Edition associated with an IP Office with the voicemail option set to ‘Voicemail Pro’. This centralized Voicemail system will provide the Media store for all of the Voicemail messages left on the system, as well as all the Media services required on the SCN to IP Office sites not supported by a local Preferred Edition.

Therefore, while Voice resources will be available locally to IP Offices providing local capabilities for queuing announcements, auto attendants, call recording and leaving of voicemails, the retrieval of voicemails will require access to the centralized Preferred Edition server.

In order to support multiple voicemails on an SCN, the IP Office Manager and core software will support an additional voicemail option of ‘Distributed Voicemail’, which will be the mode selected for the IP Offices supporting a local Preferred Edition. The availability of local voice messaging resources, such as the number of voicemail ports and / or TTS ports, will be determined by local licensing.

IP Office supporting a local Voicemail will utilize the locally licensed resources to provided local services such as: Auto Attendant, Queue Announcements, Still

Queued Announcements, Conferencing Announcements, Leaving Voicemails, Voice Recording and Campaigns. Retrieval of messages will be serviced by the centralized Preferred Edition solution as at present.



Note: *All Preferred Edition messaging servers on the network (whether centralized, distributed or back up) will require a valid Preferred Edition license.*

5 one-X Portal for IP Office

Avaya one-X Portal for IP Office Release 6.0 will introduce a number of enhancements to those already delivered in Release 5.

5.1 User Profiles

one-X Portal for IP Office will allow the user to create, edit and delete profiles. Each profile will specify the user routing options: There will be three standard profiles to match the most common routing options: Office (default), Home and Mobile.

The user will need to configure the relevant parameters (e.g. Twinning or Telecommuter number) to make the profiles work. Additional profiles can also be added. The user can switch between profiles without having to log out of the application.

There are also times when a users' profile might show as "Detected". If your settings on the telephone system do not match any of your profiles, one-X Portal will show you as being set to a profile called Detected. This profile will contain your current settings read from the phone system.

The Detected profile can reappear even after you have selected an active profile. When you select a profile to be active, that profile's settings are applied to your settings on the telephone system. However many of those settings can also be changed using other methods such as dialing short codes. If you do that, your

telephone system settings no longer match your profile and so you are changed back to the Detected profile.

5.2 Telecommuter Mode (Teleworker and Power User only)

Telecommuter mode allows you to make and receive calls using a phone at a remote location with all calls being started by and going via the telephone system. It requires you to also have a data connection from that location over which you can connect a web browser to your company's one-X Portal server. This mode is not available to all users.

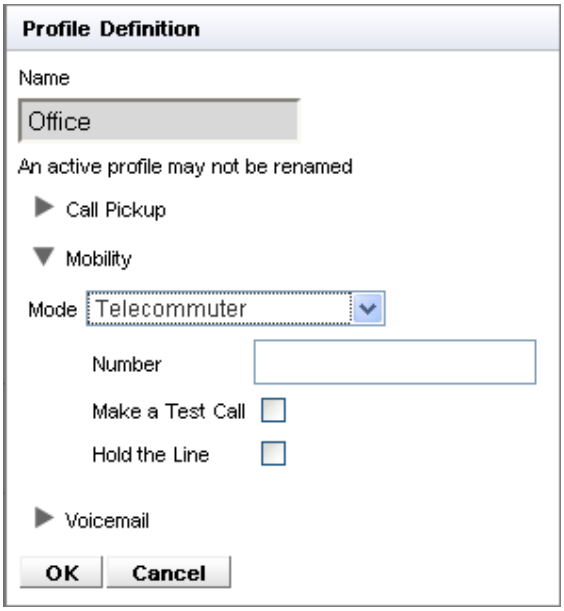
You make calls using one-X Portal to dial the number. The phone system will make a call to your configured telecommuter number and when answered it will then make a call to the number dialed as if it had been dialed from your normal internal extension. The numbers that you can call will still be subject to your normal dialing restrictions if any.

Incoming calls to your normal extension are redirected to the telecommuter number and accompanying call details are displayed in one-X Portal.

While telecommuter mode is selected, you are logged out of your normal phone extension. When you end telecommuter mode, your extension number is returned back to your normal phone extension if you have one and it is not in use by another hot desking user. If you don't have a normal extension or it is not available, you are treated as being logged off from the phone system and your calls will go to voicemail if available or receive busy.

The phone at the telecommuter number location is assumed to be a single line phone. If call waiting is enabled in your IP Office configuration it can be used for a single additional call when an existing call via the IP Office is connected.

Telecommuter mode is selected via your profile settings, under the mobility section.



The Telecommuter number that you enter must be a number that can be dialed from the phone system, including access code if required.

If you select the “Make a Test Call” option then when you make this profile active, the telephone system will make a call to the telecommuter number.

If you select the “Hold the Line” option then this controls how you use the connection from the phone system to the telecommuter number.

If Not Selected:

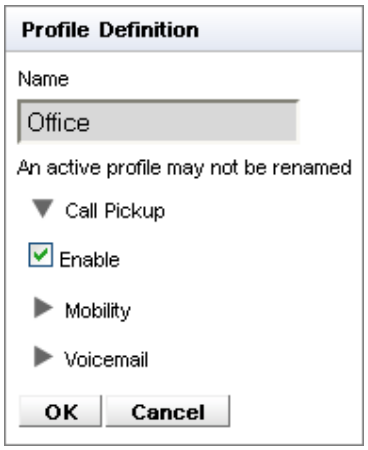
- In this mode, the phone system only calls the telecommuter number when required, that is when making or receiving a call for you. For example, when you make a call using one-X Portal, the phone system will call the telecommuter number and only when answered will it connect and make a call to the number you dialed in the one-X Portal. You can end the call through either using the one-X Portal or simply replacing the handset.
- In this mode, it is possible for the telecommuter number to also receive and make non-telecommuter calls. You can recognize telecommuter calls as they will be displayed in one-X Portal.
- Calls will not succeed if the phone system detects that the wrong type of trunk is being used (analog loop start or analog loop start emulation).
- Page calls are not redirected to the telecommuter number. Intercom and dial direct calls are redirected but as normal calls.

If Selected:

- In this mode, once you make the profile active, the phone system will make a call to the telecommuter number. When you answer you should leave the connection open (off-hook). You should then use the one-X Portal to make, answer and control calls without replacing the handset between calls.
- This option should be used whenever analog loop start trunks or analog loop start emulation trunks are involved or you suspect they are involved. This may include cellular phones connected to cellular gateways on analog trunks.
- In this mode, if the continuous call is ended while there are parked or held calls, whether a new call is established when using one-X Portal to un-hold/un-park the calls may not always work.

5.3 Call Pickup

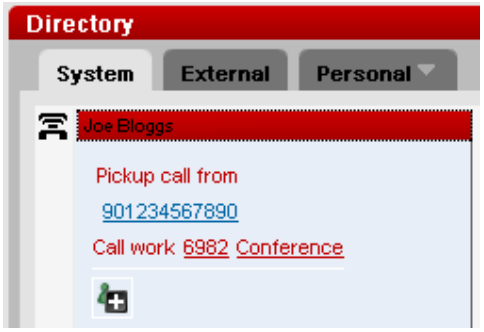
Each one-X Portal user has an active profile that includes a Call Pickup setting. When this setting is enabled, other one-X Portal users can see when you have a call waiting to be answered and can pickup the calls if required.



Note: *This is not applied to all calls waiting to be answered, for example it is not used for hunt group calls.*

When a user has enabled call pickup, their name is shown on a red background in the one-X Portal directories when they have calls waiting that can be answered using one-X Portal call pickup.

Click on the contact to display their details including information about the calls waiting to be answered. The call can then be picked up by clicking on the number link.

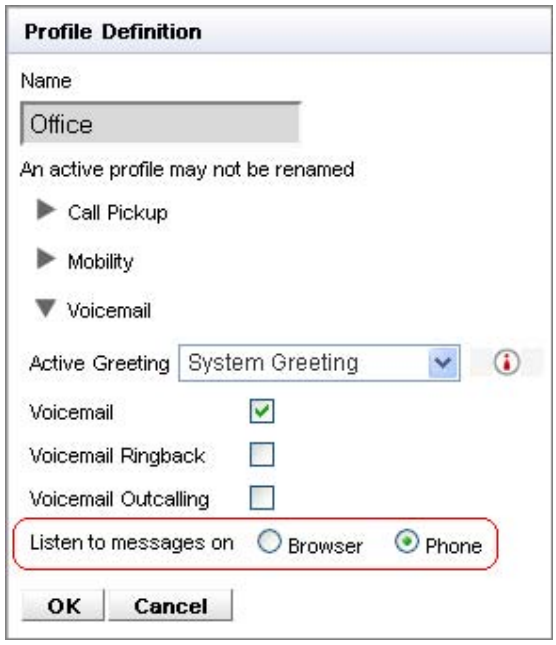


5.4 Improved Integration with Preferred Edition

one-X Portal for IP Office does not currently communicate directly with the Preferred Edition sever, but routes communication through IP Office. Avaya one-X Portal for IP Office Release 6.0 provides direct integration with Preferred Edition. This integration with the Preferred Edition server provides an interface for the user to do administrative as well as user level tasks to manage and personalize their voicemail.

Voicemail Playback - The user will be able to listen to voicemails on their PC using voicemail media player control buttons which are already embedded in the voicemail gadget. The user can toggle whether voicemails are played on the PC or the deskphone by changing a setting in their profile.

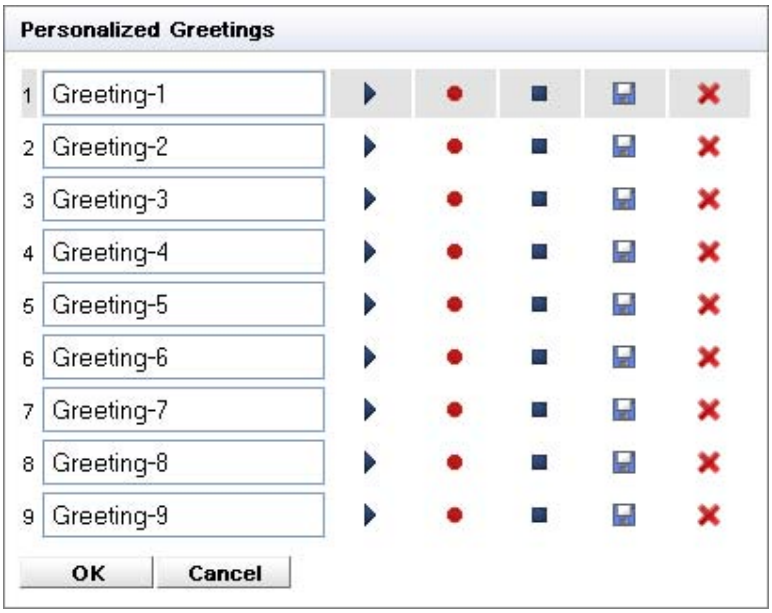
Note: *To be able to listen to voicemail messages on the PC the client PC must be able to resolve the VoiceMail Pro's Fully Qualified Domain Name (FQDN) in order to connect and play the voicemail .wav recording. It will not work if DNS is not correctly configured.*



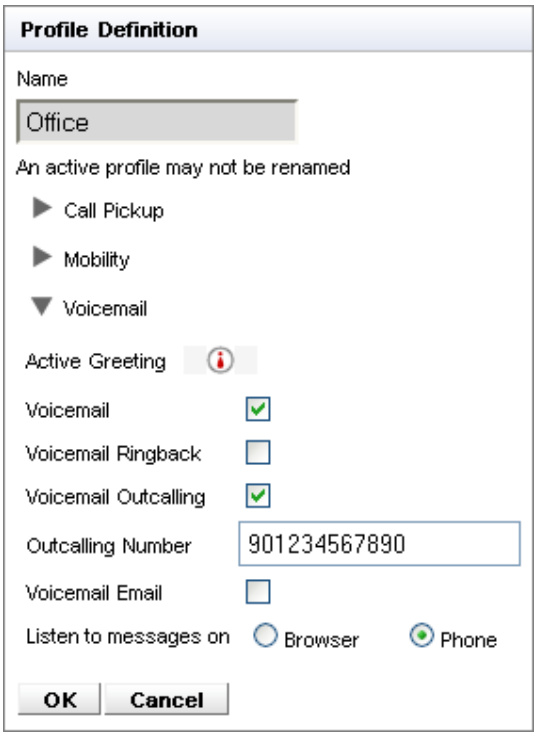
The current setting is shown in the Voicemail gadget by either a telephone handset icon or a headphones icon.

Personalized Greetings - The user will also be able to use one-X Portal for IP Office to do the following:

- Record up to 9 personalized Greetings
- Save and Name personalized Greetings
- Set Default Greeting



Outcalling - A simple interface is provided to configure this feature. Users will be able to configure this feature via their one-X Portal for IP Office profile settings. Once the Voicemail Outcalling option is selected the Outcalling Number field becomes available for configuration of the destination number.



For simplicity, the number of retries and the retry times are system dependent – i.e. the user will not be able to modify them, these are set using the Preferred Edition client.

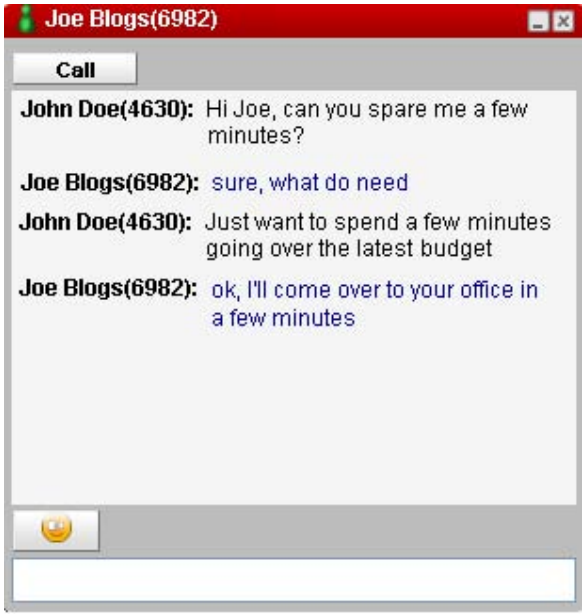
User Overrides - When voicemail is setup for a user, the administrator sets up default values for certain features. One-X Portal for IP Office will allow the user to make the following changes:

- Change Voicemail code
- Change Voicemail email options: On, Off, Copy, Forward, Alert
- Turn Voicemail On or Off
- Enable or Disable Voicemail Ringback

5.5 Instant Messaging

Opening a contact from the Directory Gadget, in any of the directory tabs, allows a user to choose to either call the contact if available to receive a call in any form or to send an Instant Message (Chat with) to the contact if the contact is able and available to receive IMs.

To start the chat session you simply click on the users name link next to “Chat with”, this will then bring up the chat window. Messages are then exchanged as per standard IM clients.



The Instant Message Gadget also has a “Call” button to allow the IM conversation to be escalated to a voice call.

5.6 Audio Conferencing

The Audio Conferencing feature in one-X Portal for IP Office provides a user with a way of creating ad-hoc conferences. The call info gadget provides conferencing related information in a conferencing tab.

The one-X Portal for IP Office will provide the following features for any ad-hoc or meet me conferences set up within an IP Office network:

- View participants
- Add/drop conference participants on the fly
- Mute/un-mute selected participants (conference initiator only)
- Mute all participants (conference initiator only)
- Individual participants can mute themselves
- Record conference (IP Office Preferred Edition required). An announcement that the conference is being recorded will be supported due to legal requirements.

The following screenshots show some of the features listed above in more detail.

Clicking on an individual entry in the list of participants allows you to either mute/unmute or drop that party.

If you wish to mute all participants in the conference you simply click on the MuteAll button. This then changes to UnmuteAll. Any muted participants are clearly shown in the list of participants.

If a participant drops from the conference, either as a result of you dropping them or if they hang-up themselves, then their appearance in the list of participants changes to reflect this.

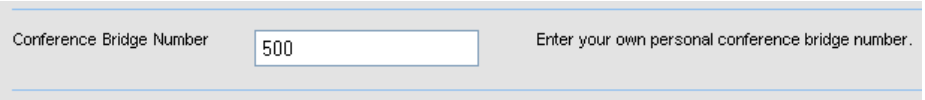
5.7 Personal Conference Bridge Number

For conferences that you start, the one-X Portal conference display provides you with functions to mute and drop other parties or to mute all parties.

Each conference hosted by the IP Office is given an ID and your system administrator can configure methods for accessing or starting a conference with a particular ID. The owner of a personal conference bridge has the same privileges as the conference initiator in an ad-hoc conference such as mute-all, un-mute all, drop etc.

To setup your personal conference bridge number:

- Select the configure tab
- Locate the Conference Bridge Number field



- Enter the name or number of the conference meet me that you want to control.
- Click on Save.

Note: *No two users can have the same personal conference bridge number.*

5.8 Personal Directory Groups

The Directory Gadget supports the concept of Groups in order to limit a set of contacts to a precise list the user chooses to make members of a particular group. Contacts from any directory source can be added into the groups.

Groups are added by clicking on the arrow in the right hand corner of the Personal tab, or once other groups have been created they will also have the arrow and will allow further groups to created from their tabs. Once a group has been created it can also be renamed or deleted.

5.9 Language Support

one-X Portal for IP Office user interface is now available in the following languages:

- English
- French
- German
- Spanish
- Italian
- Dutch
- Portuguese
- Russian

5.10 Maximum Supported Client Sessions

In the previous Release 5 version of one-X Portal for IP Office the maximum supported number of concurrent client sessions was 100. In Release 6.0 this has been increased to 500 concurrent sessions.

5.11 Minimum Supported Version of one-X Portal for IP Office

When upgrading to IP Office Release 6.0 core software the original one-X Portal 5.0 GA version (5.0.25) will not work with this release. The minimum installed version of one-X Portal must be either the 4Q09 GA build (5.0.30) or preferably the 6.0 version of one-X Portal.

6 Customer Call Reporter

IP Office Customer Call Reporter (CCR) version 1.2, being introduced with IP Office Release 6.0, introduces a number of enhancements to those already delivered in version 1.1.

6.1 Wallboard

A new browser based Wallboard is introduced with CCR 1.2. In addition to creating supervisor accounts, the administrator can also add wallboard accounts. When a browser is logged in using a wallboard account, the browser can be used to display queue statistics for any queues plus other information such as messages sent or scheduled by IP Office Customer Call Reporter supervisors. This is displayed in a style intended for display on large screen monitors being used as wallboards.

Each Wallboard can be fully customized with a logo and title allowing the customer to use their own branding for the Wallboard. IP Office Customer Call Reporter supports up to 30 wallboards. However the maximum number of wallboards that can be logged in at any time is controlled by the number of available Supervisor licenses (each license enables a simultaneous Supervisor login and Wallboard login).

Microsoft Silverlight

IP Office Customer Call Reporter wallboard views use Silverlight. When logging in at a computer without Silverlight installed, if the computer has access to the internet, you will be prompted to install Silverlight. If the computer does not have access to the internet, Silverlight must be installed manually for the wallboard view to run. Full details of Silverlight and the browsers on which it is supported can be obtained at http://www.microsoft.com/silverlight

To ensure that messages scroll smoothly, especially when a large number of rapidly changing statistics are being displayed, it is recommended that the wallboard PC uses a dedicated graphics card rather than integrated graphics provided on the motherboard of many computers. The recommended minimum specification is a DirectX 9.0c or above compatible graphics card for GPU hardware acceleration via DirectDraw with 1GB or greater video memory. Support for 60Hz or greater refresh rate at the chosen resolution, for both card and monitor.

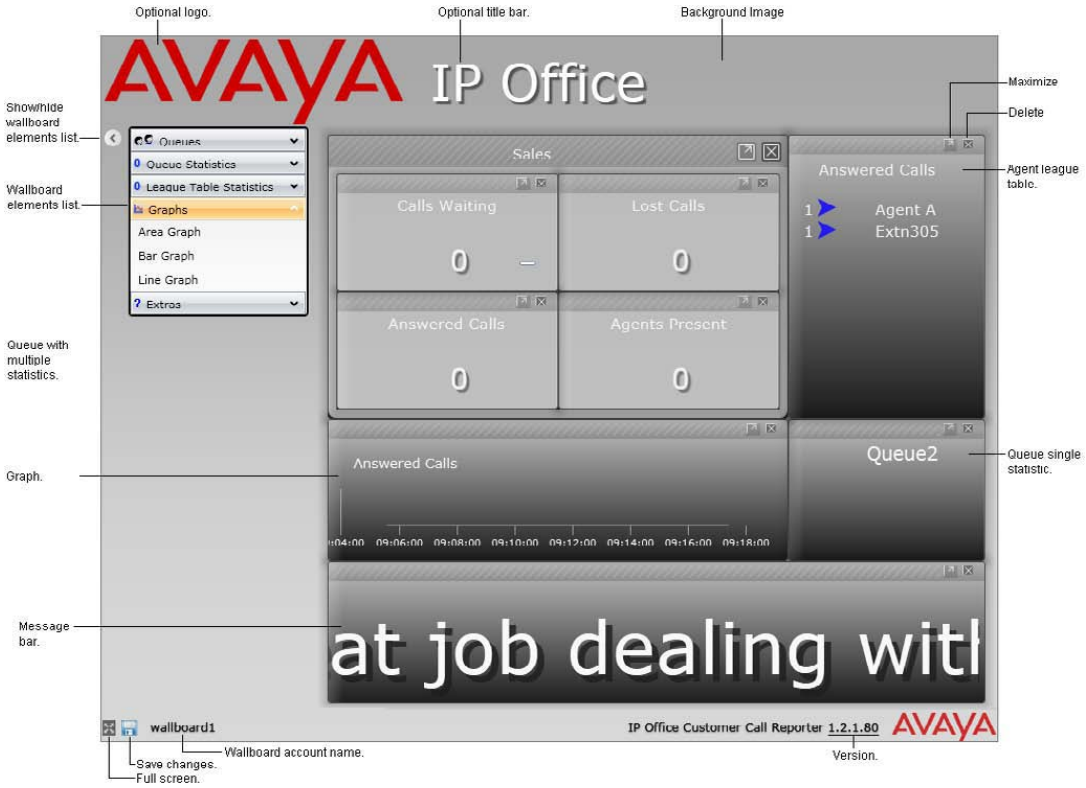
Before running a wallboard on a computer, it is recommended to switch off the computers screen saver and any monitor power saving mode. Unlike other IP Office Customer Call Reporter login account, a wallboard login does not automatically expire after the Session Expiration Minutes set by the IP Office Customer Call Reporter administrator.

On initial access the wallboard will be in design mode, displaying the configuration accordion that contains:

- Queues
- Queue Statistics
- League Table Statistics
- Graphs
- Extras – Message Bar, Logo & Custom Title

The accordion expand/collapse control is only available once a valid entry has been configured.

The wallboard display is edited directly through the browser in order to add or remove the elements required.



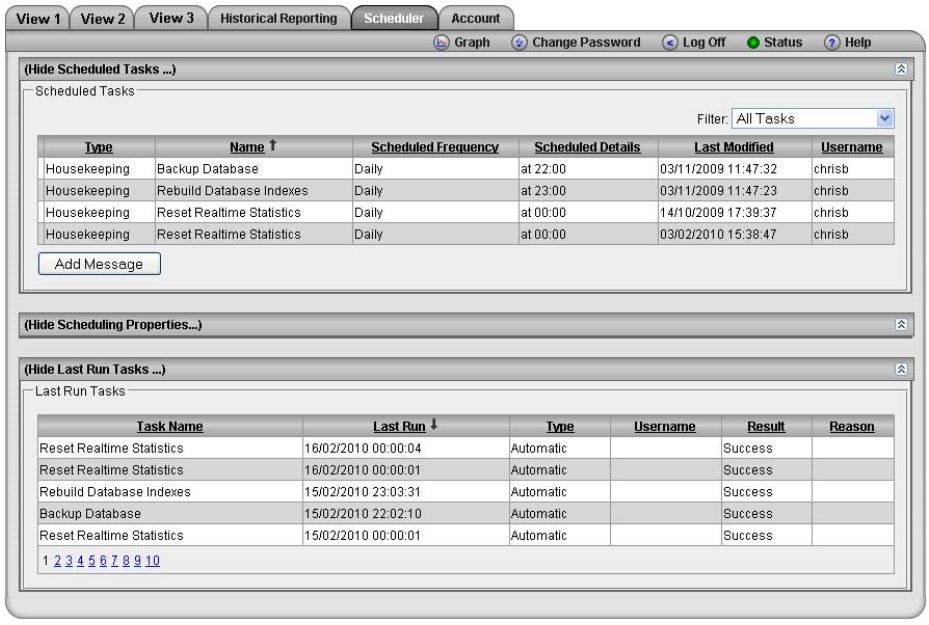
The wallboard is configured by dragging items from the accordion to the right hand of the screen, in the same manner that a supervisor creates views. Dragging a queue onto the screen creates a container into which statistics can be dropped to create the display object. The same queue can be dropped multiple times to create additional containers.

Each wallboard can contain the following different types of elements:

- **Logo:** You can display a logo image at the top of the wallboard.
- **Title:** You can display a title at the top of the wallboard.
- **Queue Statistics:** Statistics for any IP Office Customer Call Reporter queues can be added to a wallboard. These can be grouped in sets or added individually. Individual warning and alarm settings can be applied to each statistic added.
- **Message Bar:** Message bars can be added to scroll messages across the wallboard display. The messages are scheduled and sent to the wallboard by IP Office Customer Call Reporter supervisors.
- **League Table:** League tables of the top and bottom performing agents in a queue can be added to the wallboard.
- **Graphs:** Selected queue statistics can be displayed in graphs.
- **Background Image:** The image shown in the wallboard background can be customized.

6.2 Scheduler Enhancements

The current Report Scheduler tab has been renamed ‘Scheduler’, to reflect its more general purpose. The Scheduler tab can be used to view and edit a range of scheduled tasks. It also shows the status of recent scheduled tasks run by IP Office Customer Call Reporter.



The different types of tasks that can be scheduled are:

- **Reports:** When a report is created and scheduled as Automatic, its details appear in the Scheduled Tasks list. The Scheduler tab is then used to set the reports schedule.
- **Wallboard messages:** Messages to appear on IP Office Customer Call Reporter wallboards can be added from the Scheduler tab.
- **Housekeeping:** Various IP Office Customer Call Reporter housekeeping tasks can be scheduled through the Scheduler tab, these are as follows:
    - **Reset Realtime Statistics:** Reset the statistics shown in all supervisor, agent and wallboard views.
    - **Reset Web Services:** By default this task is already scheduled and occurs at 02:30.
    - **Rebuild Database Indexes:** By default this task is already scheduled and occurs at 23:00. Re-indexing the database allows reports to run faster, however during the actual re-indexing the response of IP Office Customer Call Reporter is slowed.
    - **Backup Database:** The backup database is placed into the default MS-SQL backups folder with a date and time prefix to the file.

6.3 Historic Reporting Enhancements

The following changes have been made to the historical report templates:

IP Office Connection Loss Reporting: If the data analyzer component of the CCR is unable to connect to the IP Office system, it will record details in the CCR database. Any reports run that cover the same period as a loss of connection will include details of the connectionless period in the report.

Call Summary Report Template Changes: In the Call Summary Report template, the Total Calls column is now labeled “Call Interactions” and does not include any No Answer calls. An additional column, labeled “Customer Calls”, which is the number of unique calls has been added.

Call Details Report Template Change: In the Call Details Report template, the summary sections now include totals for Customer Calls and Call Interactions.

6.4 Installation Changes

The IP Office switch discovery and selection is no longer part of the CCR software installation. Instead the IP Office switch details can be edited manually by the CCR administrator after software installation.

Note: *This also applies to software upgrades from earlier versions of CCR software. Once the upgrade is complete you must login with the Administrator account and edit the IP Office switch details.*

6.5 CCR Licensing

To support IP Office Customer Call Reporter, the IP Office Release 6.0 system must have Preferred Edition and Advanced Edition licenses. For a system being upgraded to IP Office Release 6, IP Office Customer Call Reporter can alternately be enabled by a legacy CCR Sup license or a CCC Supervisor and CCR CCC Upgrade license.

License	Function
Customer Service Supervisor Profile	The presence of at least one Customer Service Supervisor Profile license is required to use IP Office Customer Call Reporter. The license is available in 1, 5, 10, 15 and 25 supervisor versions. Multiple licenses can be added to achieve the number of supervisors required up to 30 supervisors. Each license instance enables 1 supervisor account login and also 1 wallboard account login.
Customer Service Agent Profile	Enables support for IP Office Customer Call Reporter agents. The license is available in 1, 5, 10, 20, 50 and 75 agent versions. Multiple licenses can be added to achieve the number of agents required up to the maximum of 150 agents. One license instance is consumed for each IP Office Customer Call Reporter agent currently logged in on the IP Office system.
CCR CCC UPG	This license allows existing CCC licenses to be used for IP Office Customer Call Reporter. The CCC licenses are used as follows: • **CCC Server -** Enables 1 supervisor, 1 wallboard and 5 agents. • **CCC Supervisors -** Enables the equivalent number of supervisors and the equivalent number of wallboards. • **CCC Agents -** Enables the equivalent number of agents.

6.6 IP Office Core Software Support

This version of Customer Call Reporter is supported with IP Office Release 5 and Release 6.0 core software.

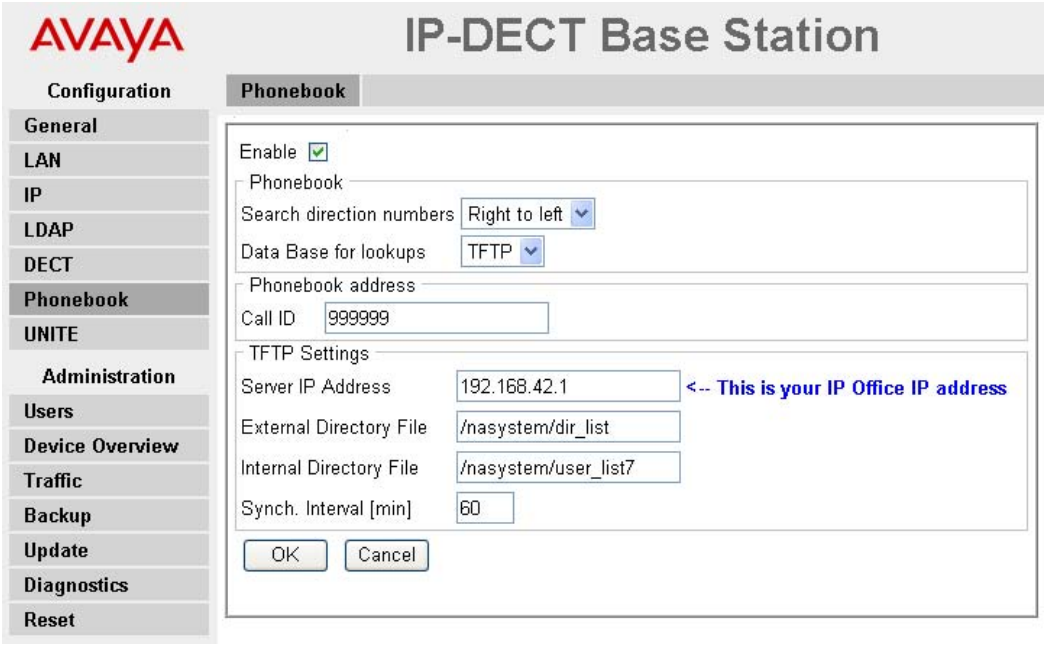
7 DECT R4

7.1 Directory Integration

IP Office Release 6.0 now supports directory integration from the base station of the DECT solution, for both standard and compact base stations, without requiring an AIWS server.

Note: *An AIWS is still required for over the air firmware upgrades to the telephones.*

This is enabled in the base stations by first upgrading them to the version of DECT R4 software supplied as part of Release 6.0, version 3.3.11, and secondly by making a configuration change on the Master base station once it has been upgraded.



8 Virtual Server Support

All server-based applications are now able to run in a virtual server environment.

IP Office server applications have been tested with VMware Server (free version), Microsoft Virtual Server 2005 R2 and Microsoft Windows Server Hyper-V environments.

This means that a customer can now run their own server applications together with Avaya IP Office server applications on one physical server. The virtualization will provide more reliability and security as installing on the same PC without virtualization.

9 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	✓	✓	✓	✓	✓	✓	✓	✓	✓
ContactStore	✗	✗	✓	✗	✗	✗	✓	✓	✗
one-X Portal for IP Office	✗	✗	✗	✗	✗	✗	✓	✓	✓
Customer Call Reporter	✗	✗	✗	✗	✗	✗	✓	✓	✓
SoftConsole	✓	✗	✓	✓	✓	✓	✗	✗	✗
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 6.0, 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.

Application	Windows				Mac
	Internet Explorer	Firefox	Opera	Safari	Safari
Preferred Edition UMS	✓ 7+	✓ 3+	✓ 2+	✓ 3.2+	✓ 3.2+
One-X Portal for IP Office	✓ 7+	✓ 3+	✓ 2+	✓ 3.2+	✓ 3.2+
Customer Call Reporter	✓ 7+	✓ 3+	✓ 2+	✓ 3.2+	✓ 3.2+
ContactStore	✓ 7+	✗	✗	✗	✗

10 Resolved in IP Office Release 6.0

IP Office Release 6.0 software has parity with the IP Office 5.0 Q1 2010 maintenance release, as well as including the following additional fixes.

One-X Portal for IP Office

CQ Number	Description of Issue
CQ83572	One-X Portal displays Ex-Directory Users in System Directory

For details of the issues resolved in the IP Office 5.0 Q1 2010 maintenance release please refer to Technical Bulletin 113.

11 Known Issues

The following is a list of issues that exist in this release of IP Office 6.0 software. These will be addressed in a future release of software.

Core Software – IP Office Mode

CQ Number	Description of Issue
CQ97056	Twinned 9650s : call originated from Secondary not in Call Log
CQ92896	Reset required for the update of list presented in Dial by Name feature
CQ91809	Busy is not received when a DECT user(has no VM and the DECT system is busy) is called
CQ96780	Call to *17 (VM Collect) is in call log - it shouldn't be
CQ92890	SIP Endpoints - Unsupervised Transfer Cannot Be Completed For Innovaphone
CQ92665	The entry in Call Log of SoftConsole from a SIP anonymous call does not show Unknown
CQ91274	Call recording: Bridge on to the call which is recording will truncate the warning message heard by the user who intruded.
CQ92600	SNMP doesn't show the phone type for F1603L phone.
CQ88637	Account /Auth codes are stored in Redial / Call log for 56xx/46xx
CQ96216	External calls assisted transferred by VM to a user with HFANS enabled are auto answered

Core Software – Essential Edition - PARTNER® Version

CQ Number	Description of Issue
CQ95970	If the user pauses during dialing the display will show "ssss" in place of digits dialed
CQ92608	Call log button does not work at call forward - intercom and T1 line
CQ96812	User Information under Status softkey for Forward/Coverage should use Number instead of Name
CQ92764	Loudspeaker Paging Extensions should not forward the calls
CQ87273	DTMF tones not heard on POTS phone when using LND, SSD, or personal speed dial
CQ88759	Cannot place the modem (76) in a hunt group.

Manager Application

CQ Number	Description of Issue
CQ85570	The access shortcode to an auto attendant works using the auto attendand's name only.
CQ92766	Some items under Firewall are not translated to French
CQ96987	Manager fails to connect to IP Office if subnet broadcast is used in Unit/Broadcast Address textbox
CQ96946	Unable to add new user using keyboard in IPOffice Manager Application
CQ92593	Error Pane incomplete warning info - user short code
CQ92727	Cannot send two CSV files in a row using the same session of Manager (Send Config option is disabled)
CQ92577	Help for Analogue extension needs updating

Simplified Manager Application

CQ Number	Description of Issue
CQ92680	Manager - User Setup - Advanced Settings - Account Entry not available on POTS users
CQ92707	Call Pickup Button should not be programmed more than once from manager with the same extension
CQ92720	The Manager should not allow me to program 2 different buttons with Pickup Group P01
CQ92848	Pressing the question mark will not start the help when PC is localized to Canada
CQ92846	Manager should allow me to configure call forward button with no origin or destination
CQ92860	When programming call forward the origin should be only the extension from which it is programmed.
CQ91861	Unable to configure the List Memberships and the Group Memberships

Preferred Edition

CQ Number	Description of Issue
CQ97190	Changing email settings using TUI do not give a warning message

One-X Portal for IP Office

CQ Number	Description of Issue
CQ97254	Conference owner doesn't have ownership rights when 2 users are added to Conference & both are put on hold
CQ97304	Call Pickup doesn't work if no CLI available
CQ91877	Doc: Firewall on Windows 2008 R2 64bit is blocking the One-X Web Client.
CQ97318	One-X Portal LDAP Providers - only one LDAP source possible
CQ96810	Call Pickup: Error is displayed on call gadget when users on call logout & re-login to 1XP
CQ96929	Call Intrude - Incorrect Display if SCN user intrudes on a local User

Customer Call Reporter

CQ Number	Description of Issue
CQ97264	CCR - Call Waiting & Current Wait Time are shown incorrectly when Park is used by any user on the system.
CQ96600	CSR - Hold time is included in Average Answer Time
CQ96054	TR: "Available" and "Queue Disabled" events are displayed for deleted Hunt Groups.
CQ91205	Help File Notes \ Real Time Sample for "Queue Call Picked Up by Another Agent 1" incorrect
CQ91272	Server 2003 always indicates the RTCE service has failed to start when machine is rebooted

CQ96158	RealTime: Marker Info (Number) not updated for initial caller after a supervised transfer
CQ96680	The warning message when erasing a WB message is not translated in supported languages.
CQ79010	CDR:Answered status after transferred is not displayed for parked calls.
CQ96945	Call Summary Report DDI - Lost call not shown if the call is lost before being presented to an agent
CQ96953	Italian Translation - Incorrect spelling for "Database" on the Administrator "Preference Details"
CQ91267	A problem when exporting a report to excel format
CQ88865	Remember me on this computer not translated
CQ96502	Scheduler: The maintenance task “Update database statistics” is not documented in help.

12 Technical Notes

Before any upgrades commence the IP Office Release 6.0 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 6.0 will be supported on the following control units:

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

Note: *If upgrading to Release 6.0 from a previous release an upgrade license is required. New installations using the IP500v2 platform will not require this license.*

IP Office Release 6.0 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 has been 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 6.0. Failure to do this will result in your system being left in a non-operational state.

12.1 Core Software Upgrade Summary

The table below shows the necessary steps that must be taken to upgrade your IP Office system to Release 6.0. 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 6.0
IP412	4.0 / 4.1 / 4.2 / 5.0	Load 6.0	--
IP500	4.0 / 4.1 / 4.2 / 5.0	Load 6.0	--
All modules	4.0 / 4.1 / 4.2 / 5.0	Load 6.0	--

12.2 Software Upgrade License Installation

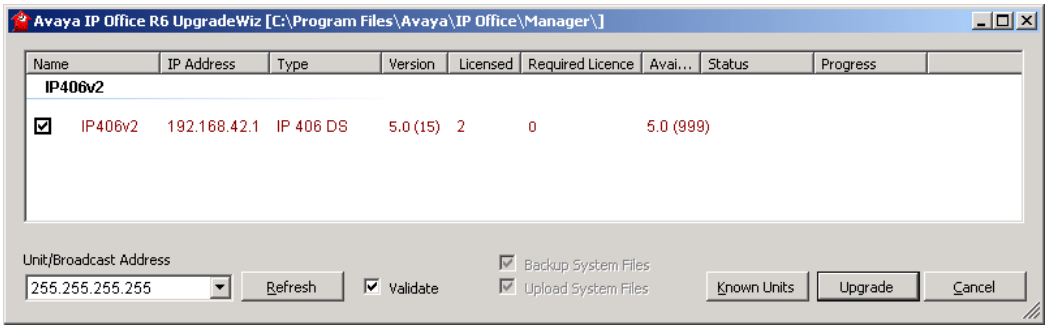
To make for a smoother upgrade process it is recommended that you install your IP Office Release 6.0 Software Upgrade license before you proceed any further. Although the key may not be recognized immediately by the system, dependant on

the current software version you have, it will be recognized when you come to upgrade your system.

12.3 Core Software Upgrade Instructions

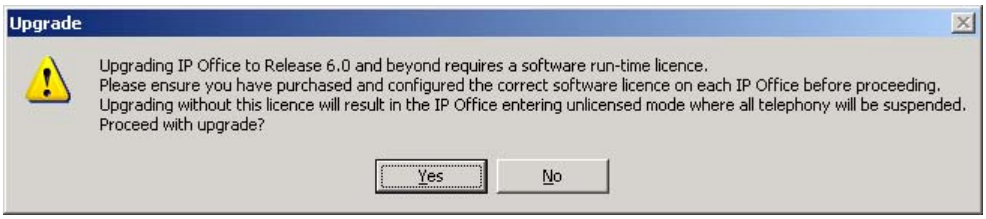
If you do not need to upgrade your loader (IP406v2 systems only) please go to step 11 otherwise follow the instructions below:

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 6.0.

Note: *If you have not installed the IP Office Release 6 Software Upgrade license the upgrade wizard will present you with the following warning:*



12.4 Unit Compatibility – Expansion Unit Interoperability

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

12.5 Phone Firmware Support

The table below lists the phone firmware versions that are supported by IP Office Release 6.0.

Phone 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 (R3)
9620 & 9630 & 9640 & 9650 App	3.1 (R3)
16xx App	1.2200
16xx Boot	1.2200
1616 Button Module	1.0.9

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.*

12.6 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 6.0. 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	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.

12.7 Upgrading from VoiceMail Pro 4.0 or later

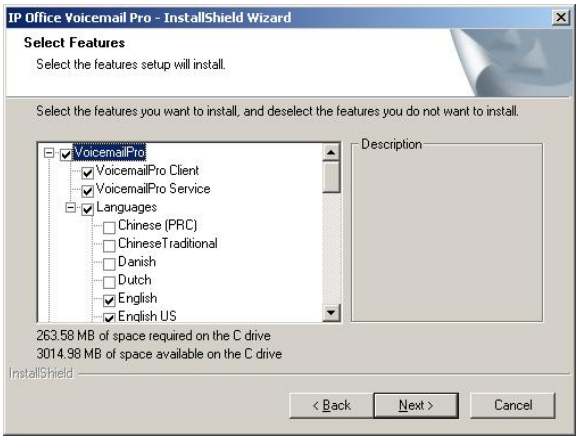
1. Export the VoiceMail Pro Database.

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.

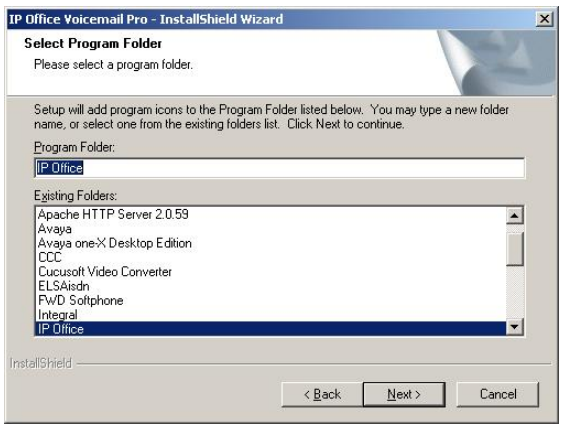
10. A prompt will appear informing you that there is an older version of VoiceMail Pro installed and will offer a major upgrade. A major upgrade looks very similar to a new installation.

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

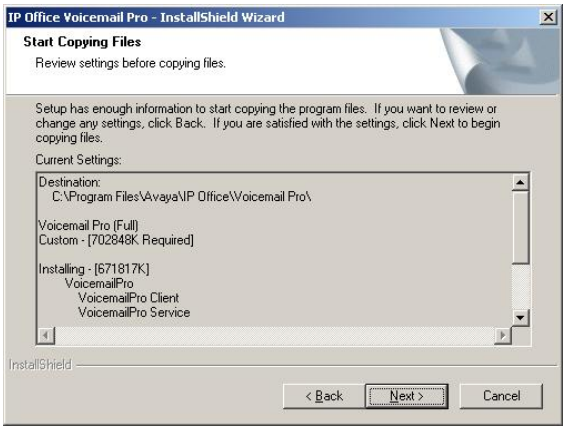


13. At the Service Account name screen enter your service account details and then click on Next.

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

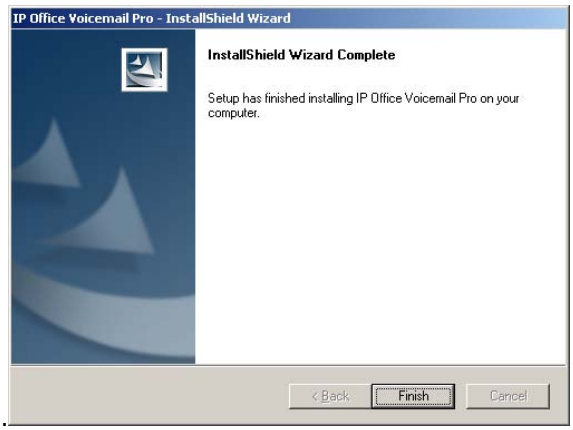


15. At the Start Copying files screen click on Next.



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

17. Finally click on Finish to complete the installation



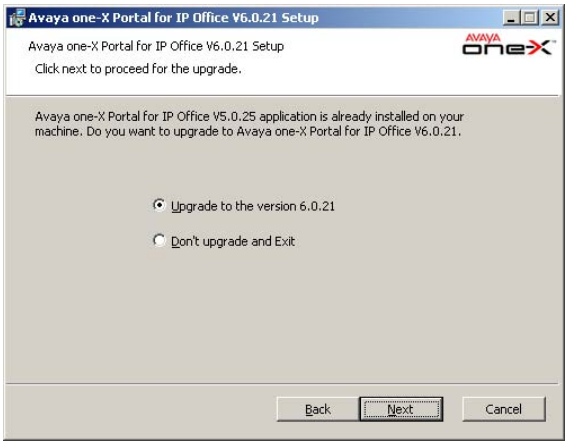
12.8 one-X Portal for IP Office Upgrade Summary

The table below shows the necessary steps that must be taken to upgrade one-X Portal for IP Office.

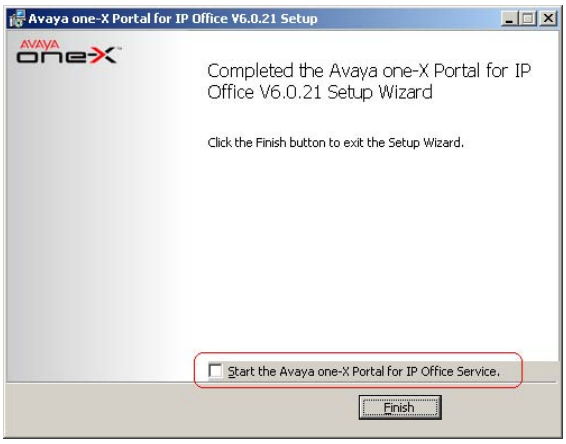
Product	Current Release	Upgrade Step
one-X Portal	5.0	Upgrade Installation Available

12.9 Upgrading from one-X Portal for IP Office 5.0

1. Insert the new one-X Portal for IP Office CD. If the setup does not start automatically, right click the CD drive & select **AutoRun**. Alternatively run one-Xportal.msi
2. At the Welcome screen click **Next**.
3. At the Upgrade screen make sure the upgrade option is selected and click **Next.**



4. At the Destination folder screen click **Next.**
5. At the Ready to install screen click **Next.**
6. When the upgrade has completed select the “Start the Avaya one-X Portal for IP Office Service” checkbox and click on **Finish.**



12.10 Customer Call Reporter Software Upgrade Summary

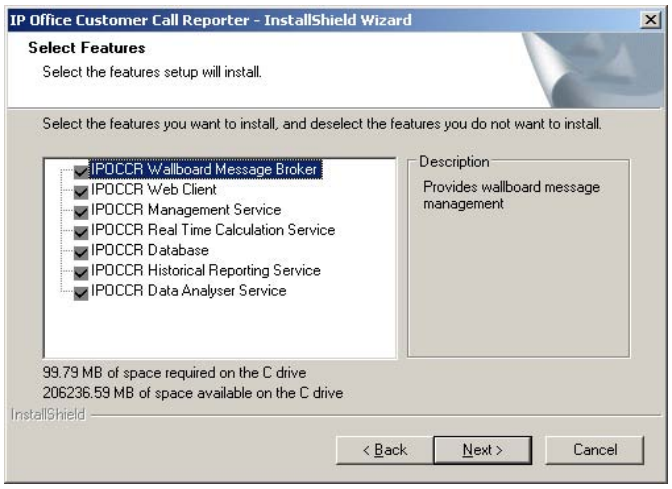
The table below shows the necessary steps that must be taken to upgrade your Customer Call Reporter to Release 1.2.

Product	Current Release	Upgrade Step
Customer Call Reporter	1.0	Upgrade Installation Available
Customer Call Reporter	1.1	Upgrade Installation Available

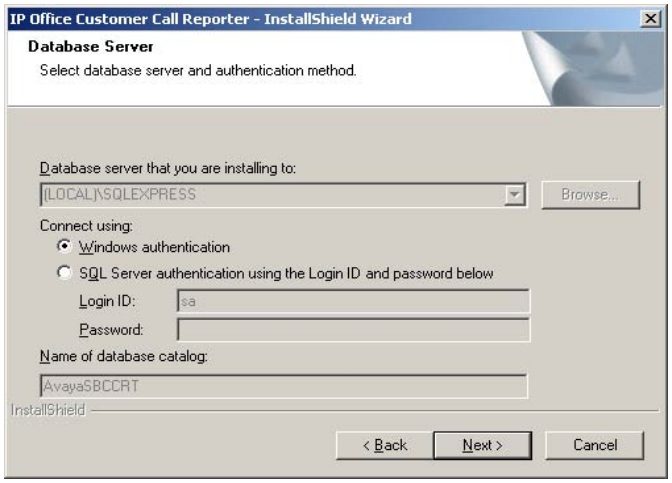
12.11 Upgrading from Customer Call Reporter 1.x

1. Insert the new Customer Call Reporter CD. If the setup does not start automatically, right click the CD drive & select **AutoRun**. Alternatively run IPOCCRsetup.exe
2. A prompt will appear informing you that there is an older version of Customer Call Reporter installed. Click on **Yes**.

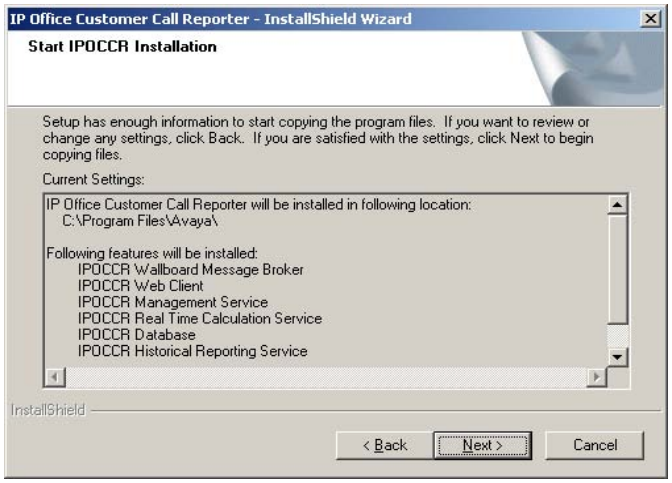
3. At the Select Features screen click on **Next**.



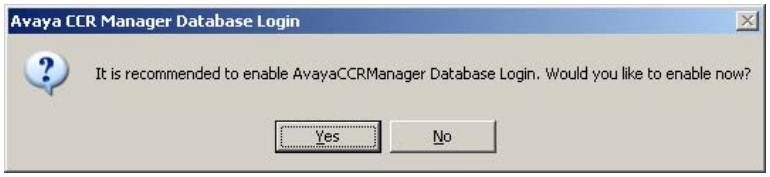
4. At the Database Server screen make sure the details are correct and click on **Next.**



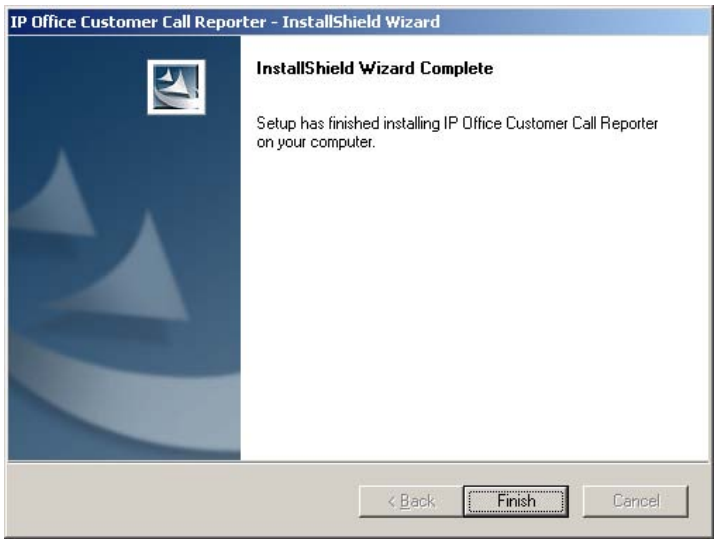
5. At the Start IPOCCR Installation screen click on **Next.**



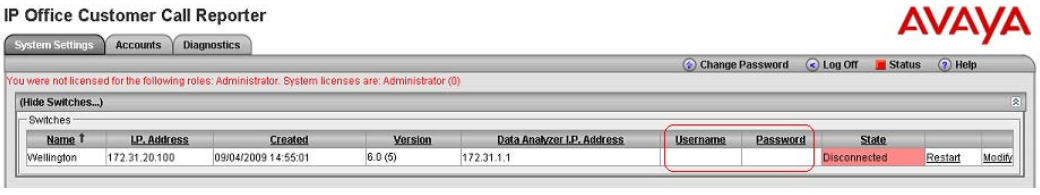
6. At the Avaya CCR Manager Database Login screen click on **Yes**. If you do not click on Yes then Supervisors will be unable to cancel report generation once it has been started.



7. When the installation is complete the InstallShield Wizard Complete window opens. Click on **Finish**.



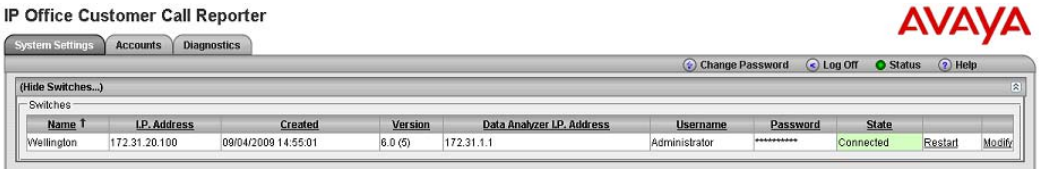
8. To complete the installation it is necessary to logon using the CCR Administrator account and re-enter the IP Office username and password to allow the CCR to connect to your IP Office system.



9. Click on **Modify** to update the details and after these have been entered click on **Update**.

10. Finally click on **Restart**, and then click on **OK** to restart the service.

11. You will be logged out of the Administrator screen. When you log back in the CCR should now be connected to your IP Office System.



Note: *After upgrading your CCR the Status indication displayed may be Red. If this is the case please stop the services, clear the Windows Application Event log on the PC and then restart the services.*

13 Assistance

13.1 Documentation

IP Office Release 6.0 Documentation can be found on http://support.avaya.com

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

13.2 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 no longer available with Release 6.0. The USER/ADMIN SET and the VOICEMAIL PRO will be together on the DVD.

The following DVD is available with Release 6.0 of IP Office:

Material Code	Description
700480569	IPO DVD R6 USER/ADMIN SET

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 6.0 binaries for the core platform and DVD images of IP Office Release 6.0 applications will be available on the Avaya Support website by February 26th 2010.

Upgrades to IP Office Release 6.0 require the purchase of a valid Release 6.0 upgrade license. However, the software images may be downloaded without restriction:

1. Go to http://support.avaya.com
2. Click “Downloads” under “Resource Library”
3. Select “IP Office” under “Download by Product Name”
4. Select “6.0” under “Select a Release”

13.3 IP Office Release 6.0 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 phone firmware binaries. Optional SD cards can be obtained from other sources.

The following System SD cards are available with Release 6.0 of IP Office:

Material Code	Description
700479702	IPO IP500V2 SYSTEM SD CARD A-LAW
700479710	IPO IP500V2 SYSTEM SD CARD MU-LAW
700479728	IPO IP500V2 SYSTEM SD CARD PARTNER

13.4 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

New and updated Classes available with IP Office Release 6.0

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 6.0 through the addition of a new IP Office Product Delta course that covers the major enhancements and customer benefits associated with Release 6.0. Please find the new or updated courses below:

Course Title/ Description	Modality	Duration
ATU01207WEN IP Office Technical Delta Release 6	WBT	3 hrs
ATU01207AEN IP Office Technical Delta Release 6 Assessment	Assessment	0.5 hrs
ATC00916WEN IP Office Technical Overview	WBT	6 hrs
ATC01220IEN IP Office Basic Technical Implementation Workshop.	ILT	24 hrs
ATC01203AEN Assessment for APIA Credential	Assessment	1 hr
ATA01221IEN IP Office Advanced Technical Implementation Workshop	ILT	16 hrs
ATI00484IEN IP Office Advanced Applications	ILT	40 hrs
3000.1 IP Office Proctored Exam for ACSS Credential	Exam	2 hrs

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Avaya SMEC New Product Introduction

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