IP Office 3.1

Product Description

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1. Introduction

Avaya IP Office Family

The Avaya IP Office Family is the latest advancement in converged voice and data technology from Avaya. IP Office combines high-end voice and data applications normally reserved for large enterprises with easy to use tools that allow the smallest of businesses to deliver cutting edge customer service.

Customer Relations Management, Computer Integrated Telephony, Voicemail, Remote LAN Access, high-speed Internet Access and a full range of other communications tools have all been integrated into this cost effective platform making it the one tool required to meet all the communications needs of the small to medium enterprise.

The Avaya IP Office family is designed to solve the complex communications challenges of the Home Office, Small Office and Medium Enterprise with simple yet powerful communications tools.

This document forms no part of a contract, the specification of the Avaya IP Office family is subject to change without notice. Not all components and features documented are available in all territories refer to Appendix A or your Avaya Representative for further details. This document should be read in conjunction with any issued technical bulletins and/or product offer announcements.

What's New in IP Office 3.1

For those already familiar with IP Office, this page lists the new features introduced in IP Office 3.1. This is not a exhaustive list, it covers just the major changes.

IP Office Core 3.1 Software

- Extension twinning.
- Buffered Call Detail Record (CDR) output in the same format as ACM.
- Message Waiting Indication on analog phones via line polarity reversal.
- Support for Cyrillic character set on most 2400, 4600, 5400, 5600 and 6400 Series phones.
- QSIG enhancements for call status indication.

New Phones

Support for new telephones and telephone accessories:

- Avaya 4621.
- Avaya 3701 and 3711 IP DECT telephones.
- EU24BL DSS module.
- Avaya T3 Compact, T3 Classic and T3 Comfort digital phones.
- T3 Headset Link.
- T3 DSS module.

IP Telephony improvements

- Increased duration of echo cancellation from 25ms to 64ms on newly introduced VCM modules - VCM 4/8/16/24.
- Fallback from IP trunks to private leased lines or, via IP Office Least Cost Routing, to PSTN trunks.

Operating System Support

- Compatibility of IP Office applications with Service Pack 1 of Microsoft Windows 2003 Server.

IP Office Management Software

- IP Office Manager Enhancements.

Avaya IP DECT Mobility Solution

- New mobility solution supports up to 120 IP DECT telephones and 32 base stations.

Voice Communication Solution

IP Office offers full voice functionality with a comprehensive list of features and benefits for the small or mid-size business, including:

- **Full PBX features**
  Caller ID, Call Forwarding, Conference Calling, Voice Messaging and more.
- **Trunk Interfaces**
  A variety of network trunk interfaces, including E1, T1, PRI, ISDN, analog loop start and analog ground start for comprehensive network connectivity. Not all trunk types are available in all territories, please check for local availability.
- **Extensions**
  Support for a range of extensions, from 2 to 360 that provide sophisticated voice performance for new and growing businesses.
- **Telephones**
  A variety of telephones including analog, digital and IP hard and soft phones (wired and wireless) that provide the appropriate desktop or device phone for every need.
- **Advanced Call Routing**
  Incoming calls are directed to the best available person or messaging service, according to the company's unique criteria.
- **Alternate Call Routing**
  Ensures reliable handling of calls by selecting from analog, digital or VoIP trunks.
- **QSIG Networking**
  Standards-based multi-site networking to interoperate with other PABXs.

Converged Voice Communications Solution

For converged communications, the IP Office acts as an IP telephony server:

- Integrated H.323 Gatekeeper and Gateway.
- Quality of Service (QoS) support through DiffServ for routing.
- 64ms Echo cancellation.

Data Communication Solution

For offices with basic data networking needs, IP Office can provide a complete data communications and networking solution:

- **Internet Access**
  Firewall protected, leased line or dial-up connectivity via PRI, T1 or WAN port: high-speed dialed access, direct leased line connections for high usage and Web site hosting, integral security, and efficient access to information and a larger business presence via the Web.
- **Routing**
  Integral Static or Dynamic (RIP I/II) routing for both Internet and Branch-to-Branch solutions.
- **Security**
  NAT (Network Address Translation) and built in firewall to protect your internal network. IPSec support allows secure data transmission across public IP Networks using 3DES encryption.
- **DHCP**
  Automatic IP address allocation for local and remotely attached PCs.
- **Remote Access Server**
  Access to local LAN servers via optional two-channel V90 modem or digital trunks: individual firewall security, access control per user, and standards-based security enable remote workers.
- **LAN Hub/Switching**
  The Avaya IP Office – Small Office Edition has a 4 port Ethernet switch (Layer 2) plus a fifth Ethernet WAN port (Layer 3). The IP406 V2 offers an 8 port Ethernet switch (Layer 2). The IP412 offers 2 switched Ethernet ports (Layer 3).
- **LDAP client support**
  For standards based directory synchronization for Phone Manager.

Applications Platform

IP Office provides big business benefits and enhanced productivity for small and mid-size businesses with a full compliment of sophisticated applications, including:

- **Voicemail**
  Incoming callers never reach an empty office. With Call Forwarding, Dial-by-Directory, the ability to retrieve phone messages via the Phone Manager Pro PC Softphone, and more.
- **Auto-Attendant**
  Simplify service for customers with this easy-to-use graphical interface; the ability to construct customized automated services means callers can efficiently navigate the system, and reach the right person, without the assistance of an operator. Available with Voicemail Pro, Avaya IP Office – Small Office Edition PCMCIA voice mail and Avaya IP406 V2 Compact Flash voice mail.
- **Integrated Messaging**
  Voice messages can be automatically forwarded to an SMTP email server or MAPI compliant email client and with Integrated Messaging Pro also synchronized with a Microsoft Exchange email server.
- **Interactive Voice Response (IVR) and Text to Speech**
  Create automated customized systems allowing callers to interact with business information, for example, Account enquiry systems, Automated ordering systems, Ticket purchasing systems, PIN number checking, Remote time sheet management, etc. Enhance theses systems by using Text To Speech to read information back to callers
- **SoftConsole**
  Graphical User Interface (GUI) for attendants via a PC-based console for call handling and physical phone for the speech path; an easy way to learn and use sophisticated tools in a comfortable environment.
- **Phone Manager**
  A powerful desktop application for the IP Office, available in Lite, Professional, and IP Softphone versions to allow you to control and manage phone calls from your Windows desktop.
- **Open CTI interfaces**
  TAPI-compliant out of the box. IP Office integrates easily with popular contact management applications such as Outlook, ACT!, GoldMine and Maximizer. Sophisticated custom applications can be rapidly developed and deployed with our full software development kit.
- **Compact Business Center**
  Report on overall system performance and basic call center functionality for up to three workgroups with quality of service reports, selected group reports, simple plug-and-play installation, and more.
- **Compact Contact Center**
  The formal Multi-Media Contact Center option, with a full customer management toolset including real time agent, system, group management, standard and custom reporting, real time tracking and analysis, options for agent connection, and remote agent support, wallboards for installations of up to 75 agents.
- **Queue Manager and Campaign Manager**
  Powerful voice and IVR applications for the Contact Center that facilitate agent and traffic management for better productivity and customer service.

Management Tools

The full IP Office solution (phone system, router/firewall/DHCP server, voicemail and other applications) is easily managed through the IP Office Manager.

The IP Office Manager is a Windows PC software application that connects to the IP Office system using TCP/IP. It can be on the same LAN as the IP Office, remote on the WAN, or connected via the Remote Access Server with either a Telephone Adaptor, Router or the optional Internal Modem Card.

A suite of IP Office Wizards can also be used to configure systems and to manage user adds, moves and changes. These wizards make setting up and using the IP Office very easy – Business Partners can use them or allow clients to access them for basic moves, adds and changes.

Scalable Platform

The "all-in-one" IP Office Family — servers, media modules and cards for connectivity and preloaded applications — give small and mid-size enterprises the options they want to meet today's communications needs and plans for the future.

- **Avaya IP Office - Small Office Edition**
  The IP Office - Small Office Edition is a compact platform specifically designed to meet the needs of very small businesses and home offices. In a single unit, it can provide a PABX with Auto Attendant and Voicemail, Broadband Access, Wireless Access Point (WiFi) and VPN tunneling. Voice Compression is included as standard to support IP Extensions or provide IP Trunks back to a head office. The IP Office - Small Office Edition is available in the following configurations:-
  - 4 Analog trunks, 4 analog extensions, 8 digital stations and 3 VoIP resources.
  - 4 Analog trunks, 4 analog extensions, 8 digital stations and 16 VoIP resources.
- **Avaya IP Office IP406 V2**
  Supports 6 Expansion Modules providing a combination of up to 190 analog and digital extensions, with capacity for 8 analog trunks or 2 digital trunks (up to 72 T1 channels or 90 E1 channels). 8 Digital Station ports (DS), 2 analog phone ports, a compact flash memory card socket for optional embedded voicemail. Additional analog trunks can be added using IP400 Analog 16 modules. Features include up to 30 optional voice compression channels, 8 Ethernet port switch (Layer 2), a 9-pin serial port and the trunk interfaces can include a dual-PRI trunk card in slot A and 24 data channels. Data channels are used for Routing, RAS and Voicemail applications. An Internal Modem Card can be added to answer up to 12 V.90 analog modem calls.
- **Avaya IP Office IP412**
  Supports 12 Expansion Modules providing a combination of up to 360 analog and digital extensions, with capacity for 8 analog trunks or 4 digital trunks (up to 96 T1 channels or 120 E1 channels). Additional analog trunks can be added using IP400 Analog 16 modules. Features include 60 optional voice compression channels, 2 independently Switched LAN ports, and 100 data channels. Data channels are used for Routing, RAS and Voicemail applications. An Internal Modem Card can be added to answer up to 12 V.90 analog modem calls.

Telephone Options

IP Office supports multiple telephone solutions, giving the small and mid-size business maximum flexibility to choose according to their current and future needs:

- **IP Telephones**
  IP Office's integral H.323 Server supports Avaya 5600 IP telephones, selected Avaya 4600 Series H.232 IP telephones, 3600 Wireless VoIP handsets and Phone Manager running in Phone Manager Pro PC Softphone mode.
- **Digital Telephones**
  IP Office Digital Station 16 or 30 Modules support the Avaya 5400 Series of digital phones and Avaya T3 Series telephones. The IP Office Digital Station modules also support existing selected MERLIN MAGIX 4400 Series phones and all 6400/2400 Series phones except the 6402.
- **Analog Phones**
  IP Office Phone 8, 16 or 30 Modules support standard analog phones, faxes and modems, with support for calling line identification.
- **Wireless Telephones**
  Avaya IP DECT base stations can be added to support the new Avaya IP DECT 3701 and 3711 telephones. The IP Office Digital Station modules support the Avaya 3810 telephone.

Application Licensing

IP Office is an application platform, which provides free-of-charge applications, including Phone Manager Lite, Voicemail Lite and CTI interfaces (details are given within the price list). These free-of-charge applications can be upgraded to provide enhanced functionality.

All chargeable applications are enabled by the use of a license key. For these applications to work, a license service either inside the IP Office control unit or optionally on a PC (running Microsoft Windows 2000, XP Professional or 2003) must be used. When using a PC as the license service, the PC must be on the same LAN segment as the IP Office control unit and should be permanently switched on.

Licensing is achieved by the use of a physical feature key dongle, which plugs into the License Service PC's parallel port, USB port or to the serial port of the Small Office Edition, IP406 V2 and IP412 control units. This feature key contains a serial number which is used to validate licenses. The Feature Key serial number must be provided with any order for charged applications.

All license keys are stored on the IP Office switch. This means that if the hardware fails - a rare event - full functionality can quickly be restored by installing a replacement system unit and restoring the previous configuration, and feature key, significantly increasing the resilience of the system. In cases where the Feature Key fails or is changed for one of a different type, a replacement set of license keys is required to match the serial number of the new key.

The license keys are periodically validated against the License Service. If the PC is not running, or the Feature Key (dongle) is unplugged (or otherwise unavailable), then the licenses will be invalidated (and therefore the applications will not work, or will provide 'Lite' functionality) until the system can re-validate them. A short grace period is provided, to ensure that transient network problems do not affect the level of service provided by the system. As soon as the connection between the IP Office system unit and the License Service is restored, the licenses become valid again.

2. IP Office - Small Office Edition Platform

IP Office - Small Office Edition Overview

IP Office - Small Office Edition supports the applications and functionality of the IP Office product range (refer to the relevant sections for further detail). This section details those aspects unique to the IP Office – Small Office Edition.

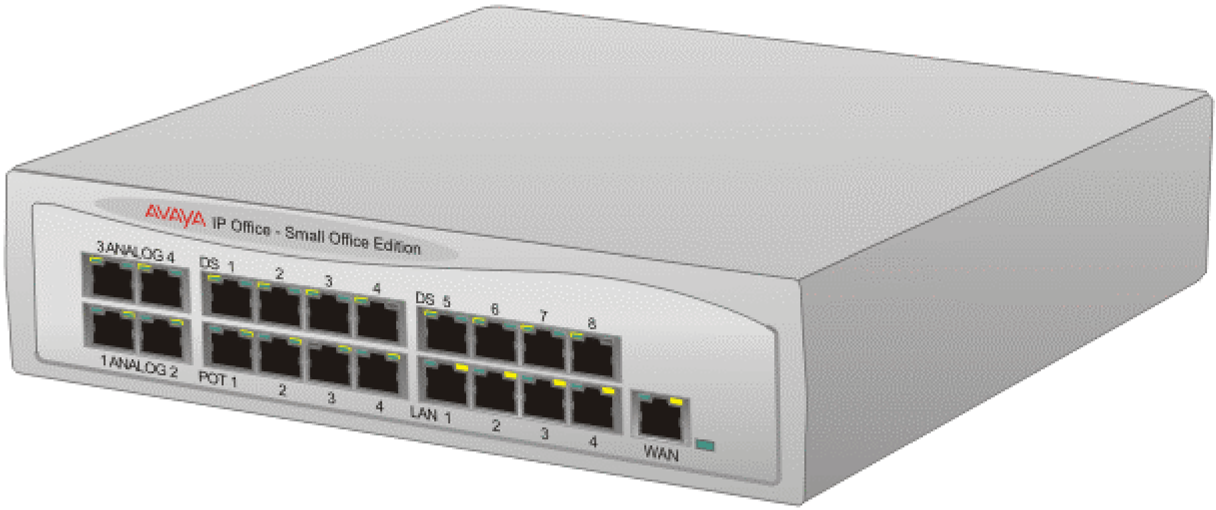
The IP Office - Small Office Edition is a compact fixed configuration unit that is available in several variants (availability may vary by territory) which provide different mixes of Analog trunks, Analog extensions, Digital extensions and Voice over IP (VoIP) capacity. Dependant on the model chosen, up to a maximum of 28 extensions can be supported (4 Analog, 8 Digital and 16 IP).

All IP Office - Small Office Edition's variants have a four-port Ethernet Switch (Layer 2) and a dedicated switched Ethernet WAN port (Layer 3), making the system ideal for connection to local area networks and broadband wide area network services such as ADSL and Cable. With Voice over IP as standard and optional IPSec security, the system can be quickly configured to provide secure voice and data networking from remote offices or branch locations back to a head office over a broadband connection.

The IP Office - Small Office Edition includes a WAN option slot on the rear of the unit which can be used to support other network connection types such as V35, V24, X21 and T1 leased lines.

The back of the unit also features a twin PCMCIA socket that can support a plug-in voice memory card for use with the embedded voicemail function, and a Wireless LAN card when using the system as an Access Point.

As well as supporting the external license key server to enable licensed applications, IP Office - Small Office Edition also supports a serial port dongle. This can be plugged directly into the unit removing the need for an external PC for license verification.



The pre-defined configurations supported in IP Office 3.1 are detailed in the following table.

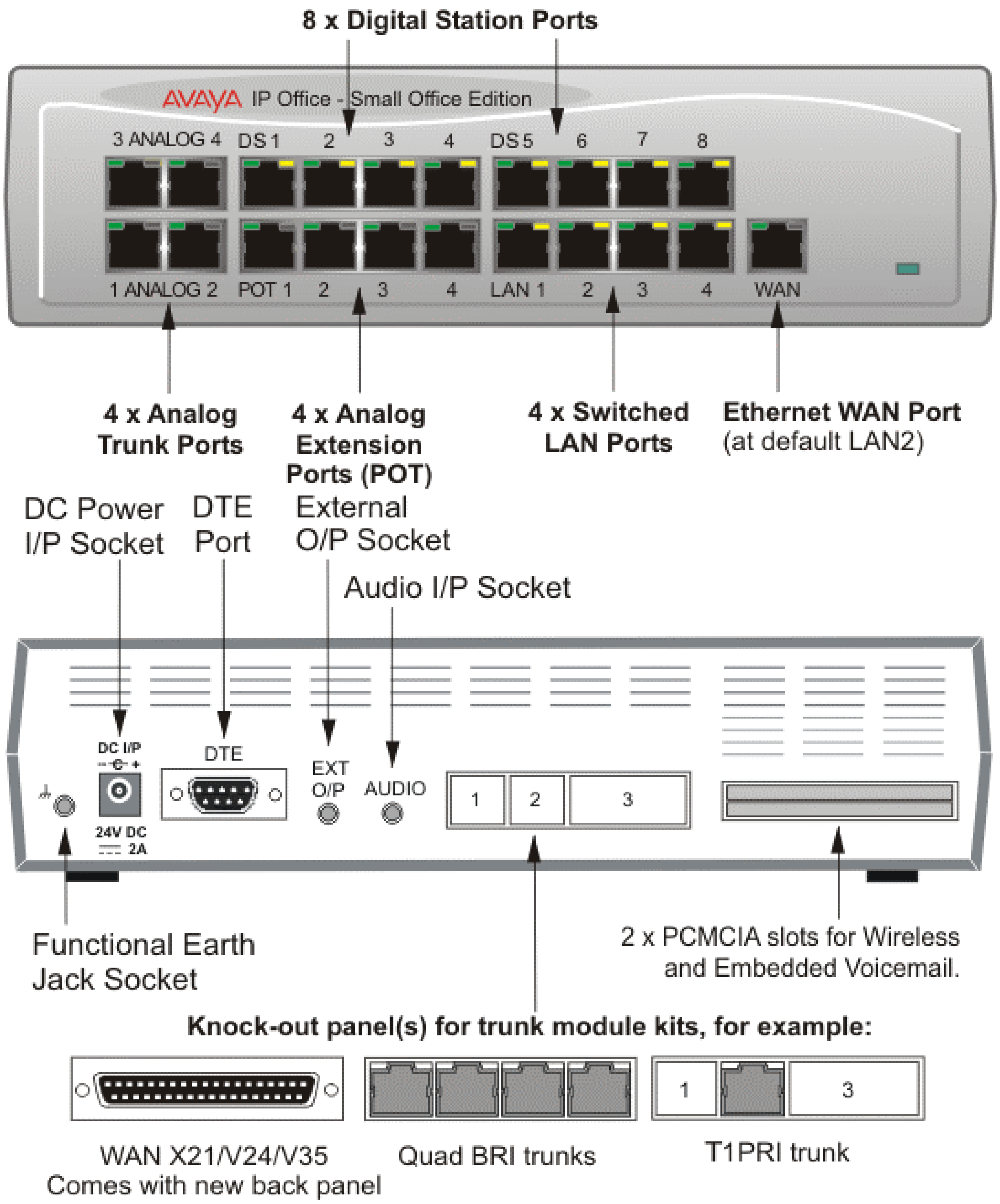
IP Office - Small Office Edition	Analog Trunks	Analog Extensions	Digital Stations	IP Extensions	VoIP Channels
4T+4A+8DS (3 VoIP)	4	4	8	16	3
4T+4A+8DS (16 VoIP)	4	4	8	16	16

- During power fail, Analog port 2 is connected to POT port 1.

IP Office - Small Office Edition 4T+4A+8DS (3 VoIP)

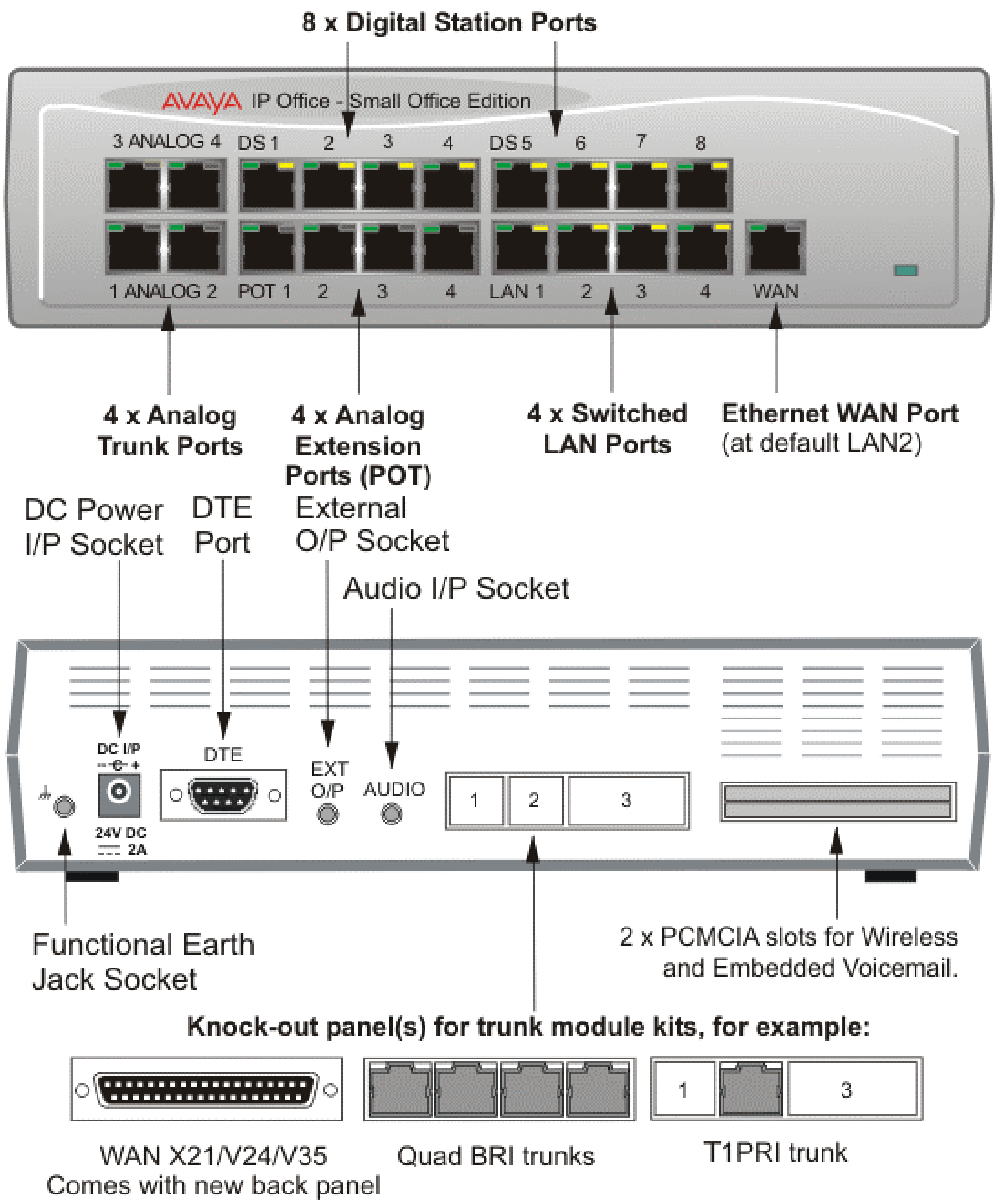
The IP Office - Small Office Edition 4T+4A+8DS (3 VoIP) provides:

- Four Analog Loop Start Trunks (Caller ID enabled).
- Four Analog extension (POT) ports. During power fail, analog trunk port 2 is connected to analog extension port 1.
- Eight Digital Station (DS) ports for selected 2400, 5400, 6400 and T3 Series phones plus 3810 wireless (US) phones.
- Three VoIP Codecs (G.723.1, G.711 and G.729a) and 48ms echo cancellation.
- 4 Switched Ethernet ports (Layer 2).
- Dedicated Switched Ethernet WAN port (Layer 3).
- 2 x PCMCIA Slots for Wireless and Memory card support.
- WAN Slot for Optional WAN card (V35/V24/X.21, BRI or T1 PRI).
- DTE port.
- Audio port for external music on hold source.
- Two relay switch port for door entry systems (External O/P socket).



IP Office - Small Office Edition 4T+4A+8DS (16 VoIP)

Specification as per IP Office - Small Office Edition 4T+4A+8DS(3 VoIP) except with 16 VoIP resources as standard.



Avaya IP Office - Enterprise Edition 4T+4A+8DS (16 VoIP)

Specification as per IP Office - Small Office Edition 4T+4A+8DS(16 VoIP).

This variant is uniquely positioned for use in small autonomous branch offices of large enterprises. It targets price-sensitive customers with limited application needs and no initial requirement for centralized administration.

As such IP Office – Enterprise Edition has the same functionality as Small Office Edition. However as part of the Avaya Large Communications Systems portfolio, Enterprise Edition is eligible for a future migration offers for integration into a centrally managed system, based on Avaya Communication Manager and MultiVantage applications.

G150 Media Gateway

Based on a similar architecture and form factor to Small Office Edition, G150 Media Gateway targets price-sensitive customers that do require the benefits of Avaya Communication Manager, basic applications support and local survivability for small branch offices with 2-12 users.

G150 is available in 4 user and 12 user variants and complements the Avaya range of Media Gateways, including G250, G350 and G700.

As part of the Large Communications Systems portfolio, G150 relies on centralized call processing, management and control. This is provided by an Avaya Media Server, such as S8300, S8500 or S8700, running Avaya Communication Manager software. As such G150 is not designed for standalone or autonomous use as it relies on an IP-based network connection to the Media Server for telephony and communication services.

IP Office – Enterprise Edition and G150 Media Gateway are available to suitably accredited Avaya BusinessPartners only. For further details, contact your local Avaya representative or visit www.avaya.com.

IP Office - Small Office Edition WAN Expansion Interfaces

All IP Office - Small Office Edition variants provide an expansion slot for an optional WAN interface of the following types (check locally for availability). Each of these interface cards are now described in more detail.

IP400 WAN Expansion

The IP400 WAN Expansion card provides a single WAN connection (X21, V24 or V35 via a 37-way D Type socket). Line speeds up to and including 2Mbps are supported on the interface. The carrier providing the line dictates the actual operating speed, i.e. in some territories the maximum speed may be 1.544M.

IP400 Office BRI Card

The BRI trunk card provides 4 European Basic Rate ISDN S/T-Bus interfaces (8 trunks).

Details of the supported supplementary services on BRI interfaces are given in the 'Public and Private Voice Networks' section.

- Not available in all territories, check for availability.

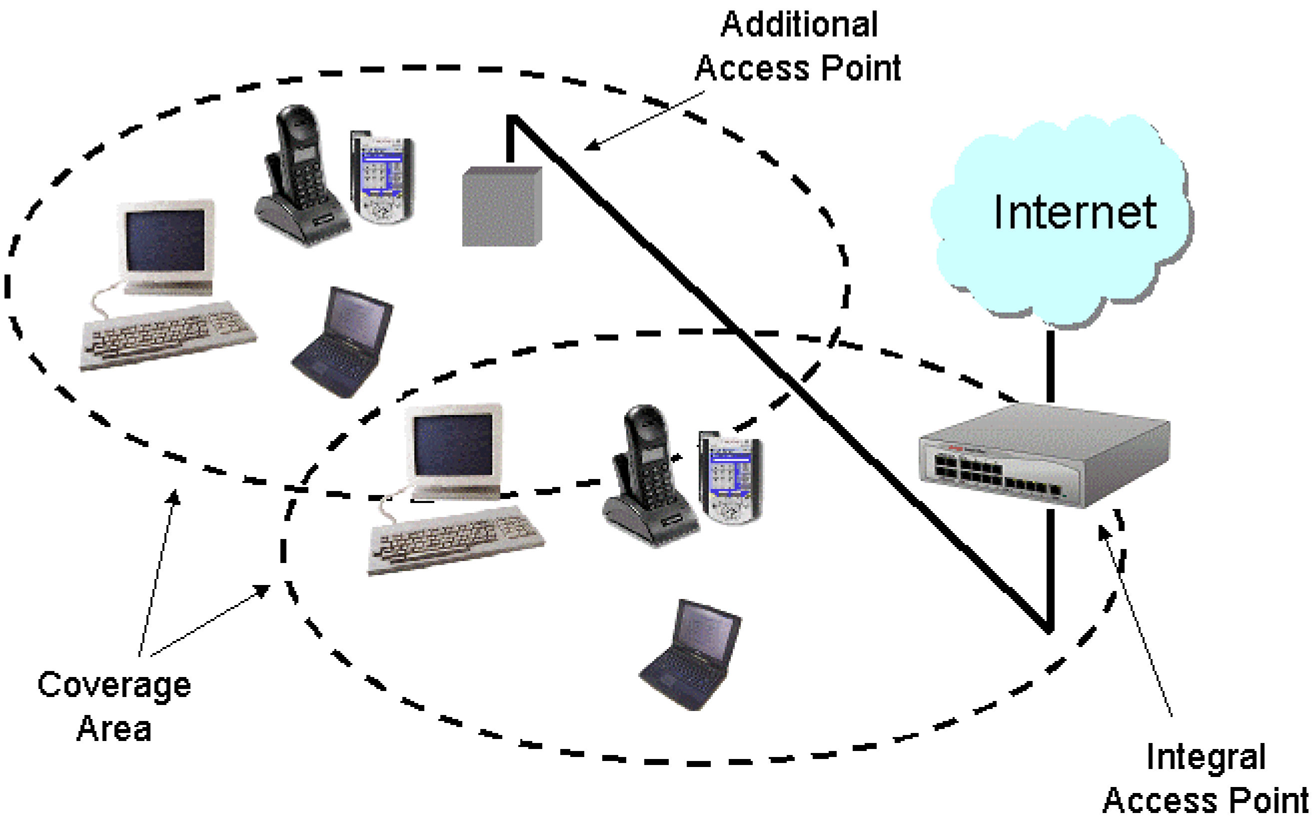
IP400 Office T1 PRI Card

The IP400 Office T1 PRI card provides a single primary rate trunk interface for supporting voice services and fractional leased lines, providing up to 256K bandwidth on IP and Frame Relay services.

- Not available in all territories, check for availability.

Optional Wireless Access Point

All IP Office - Small Office Edition platforms can be configured to become Wireless LAN access points. An Access Point acts as a Hub in a wireless network providing connectivity between devices in the vicinity. In ideal conditions a range of up to 550M (1750 ft) is achievable although this range will be decreased if walls and other obstacles are present. This is used where local conditions impair coverage and additional Access Points are needed to cover the black spots.



The IP Office - Small Office Edition wireless network can be secured against intruders using either the Wired Equivalent Privacy (WEP) or RC4. WEP uses 64 bit encryption key and RC4 uses a 128 bit encryption key. Only devices with a matching security key can participate in the network.

IP Office - Small Office Edition complies to the IEEE 802.11 and IEEE 802.11b standards meeting the Wireless Ethernet Compatibility Alliance (WECA) Wireless Fidelity Wi-Fi requirements for interoperability.

Summary

- 2.4 GHz to 2.5 GHz band (Scientific Medical and Industrial (SMI) band).
- Automatic fallback 11Mbps, 5.5Mbps, 2Mbps or 1Mbps.
- IEEE 802.11 and IEEE 802.11b Compliance.
- Wireless Fidelity Wi-Fi Compliance.
- Interoperable with other 802.11b compliant devices.
- WEP or RC4 security.
- Range up to 550M (1750ft).

Range (meters/ft)	11Mbps	5.5Mbps	2Mbps	1Mbps
Open	160m/252ft	270m/885ft	400m/1300ft	550m/1750ft
Semi-Open	50m/165ft	70m/230ft	90m/300ft	115m/375ft
Closed	25m/80ft	35m/115ft	40m/130ft	50m/165ft
Receiver Sensitivity dBm	-82	-87	-91	-94
Delay Spread (at FER of <1%)	65ns	225ns	400ns	500ns

For wireless operation, IP Office - Small Office Edition must be fitted with a Wireless LAN card and the Wireless LAN Access Point license key. Alternatively, a 3rd party wireless access point can be connected directly to one of the LAN ports.

Optional Embedded Voicemail and Auto-Attendant

Entry-level voicemail and auto attendant applications are available using the Avaya Voice Memory Card (64MB Flash memory) in one of the PCMCIA slots on the rear of the Small Office Edition. This provides small locations with an effective embedded messaging solution without the need for an external PC. No license is required to enable embedded voicemail and or auto attendant.

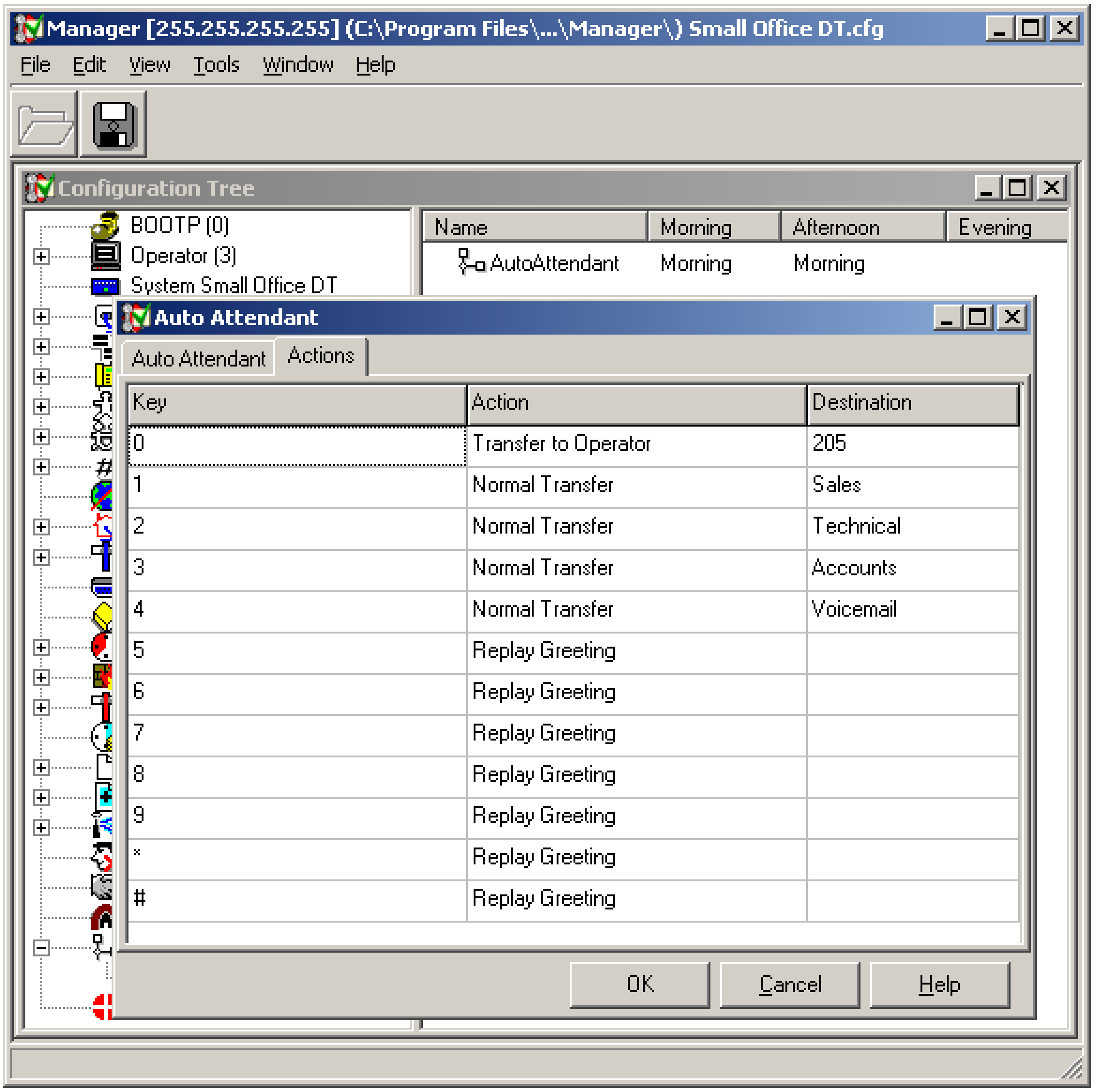
The embedded voicemail supports up to 10 hours of compressed message storage. The number of available voicemail ports (to support simultaneous calls to voicemail) is determined by the available number of voice compression channels up to a maximum of 10.

Personalized greetings and PIN-code access can be simply enabled for each mailbox by the mailbox users. Inactivity timeout and return to operator options ensure efficient message handling. Mailbox users can also access their mailboxes when out of office using a simple remote login sequence.

Up to four independent auto-attendants can be configured on the platform. The choice of which auto-attendant is to answer a call can be made on any of the criteria on the Incoming Call Routing form such as called number, calling number and time of day.

Each auto-attendant has a single menu of 12 items (0...9, *, #) that a caller can select from to either be transferred to a predefined number or replay the greeting. The greeting for the menu is controlled by time profiles to allow three alternative messages to be played i.e. Morning, Afternoon and Evening.

- Note: The Small Office Edition and IP406 embedded voicemail memory cards are not interchangeable. In addition embedded voicemail is only supported on Avaya supplied memory cards.



3. Platform Overview

IP Office Overview

This section provides an introduction to the main components of the IP Office platform - the IP406 V2 and IP412.

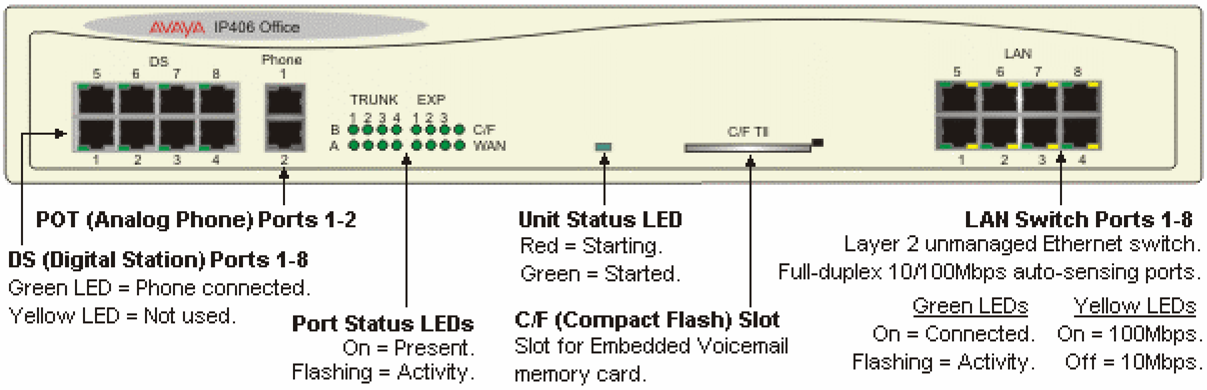
Avaya IP Office IP406 V2 Control Unit

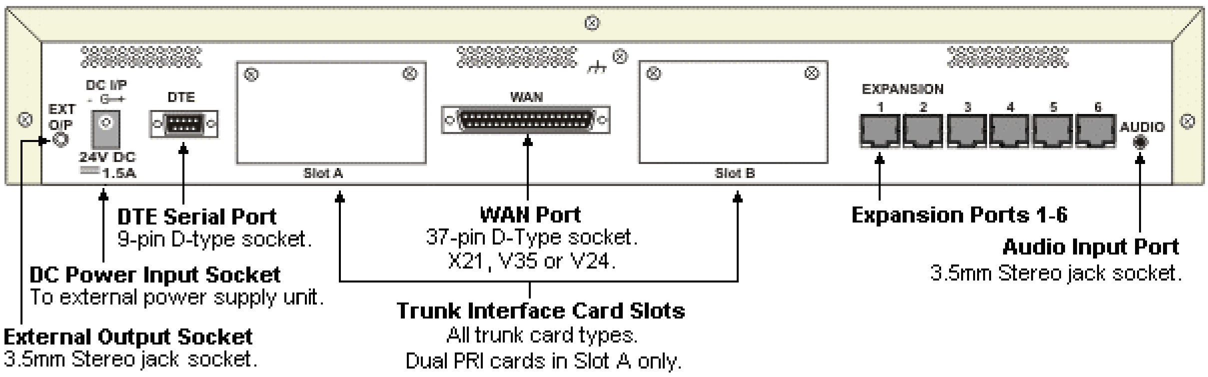
The IP406 V2 control unit is a stackable unit with an optional 19" rack mounting kit. The IP406 V2 includes:

- Eight Digital Station (DS) ports for selected 2400, 5400, 6400 and T3 Series phones plus 3810 wireless (US) phones.
- Two Analog telephone ports.
- Eight 10/100 Mbps LAN Switched ports (Layer-2, unmanaged).
- Support for optional embedded voicemail/auto-attendant (Compact Flash card)
- 9-pin DTE Port (for maintenance or serial dongle connection for PC-less licensing).
- X.21/V35 WAN interface.
- Support for 6 Expansion Modules.
- External output port containing two switches for door entry systems.
- Audio port for external music on hold source.
- 40 Data channels *(Maximum 20 useable for Voicemail Pro)*.
  A data channel is used whenever a call is made from the IP network to an exchange line (Central Office). For example, four people surfing the Internet will use a single data channel since they all share the same line to the ISP. Two people remotely accessing the Office LAN from home will use two data channels since they have dialed in on separate lines. IP extensions do not use data channels.

Optional internal upgrades allow for the support of up 12 x V.90 modem calls and a 4, 8, 16, 24 or 30-channel Voice Compression Module (VCM). The VCM module supports 4, 8, 16, 24 or 30 simultaneous Voice over IP sessions. These can be used for either providing networking between sites over a Wide Area Network or supporting IP Telephones and Soft phones. Typically an IP extension only uses a compression module channel when starting a call and while on a call to a non-IP extension/line. It is possible to configure more IP extensions than the capacity of the installed VCM.

Through support of up to six external Expansion Modules, IP406 V2 can be enhanced to support a mixture of analog, digital or IP phones to maximum of 190 phones.





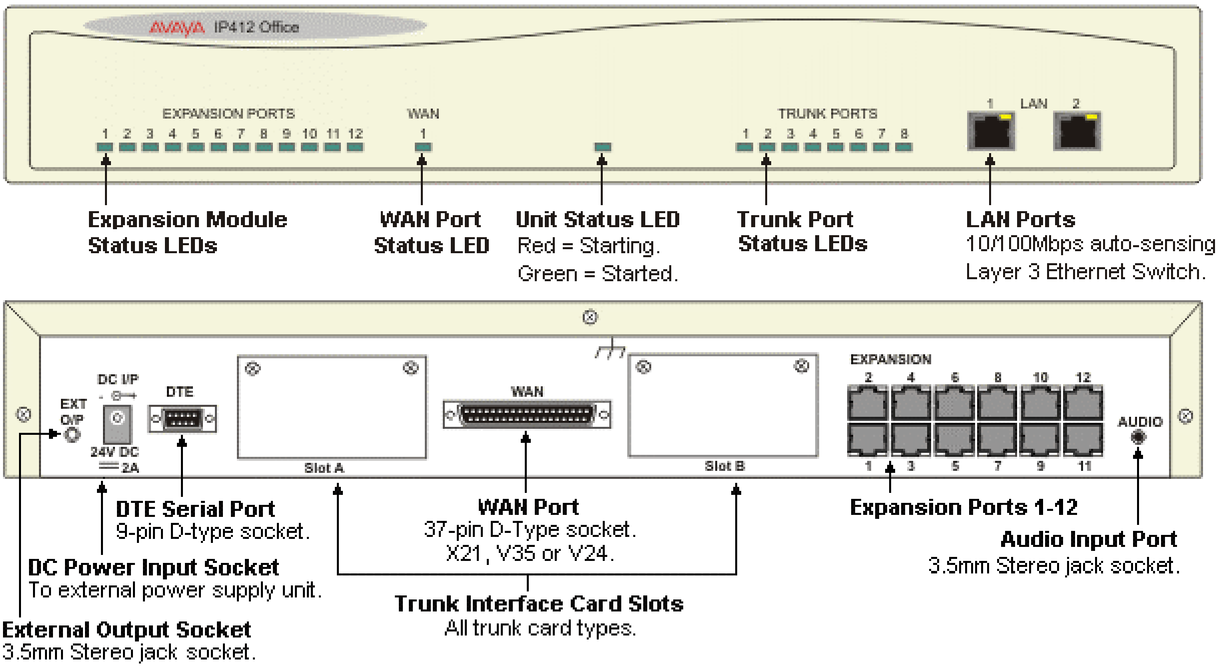
Avaya IP Office IP412 Control Unit

With a more powerful call processing engine and greater internal data transfer capability the IP412 is the most suitable of the IP Office range for meeting the needs of the small contact center or businesses with a CRM focus. The IP412 differs from the IP406 Office by providing a greater trunk expansion capability of up to four PRI trunks. The IP412 is a stackable unit with an optional 19" rack mounting kit. The IP412 includes:

- Two 10/100 switched Ethernet ports (Layer 3).
- DTE Port.
- X.21/V35 WAN interface.
- Support for 12 Expansion Modules (360 extensions maximum).
- External output port containing two switches for door entry systems.
- Audio port for external music on hold source.
- 100 Data channels *(Maximum 30 useable for Voicemail Pro)*.
  Note: A data channel is used whenever a call is made from the IP network to an exchange line (Central Office). For example, four people surfing the Internet will use a single data channel since they all share the same line to the ISP. Two people remotely accessing the Office LAN from home will use two data channels since they have dialed in on separate lines. IP extensions do not use data channels.

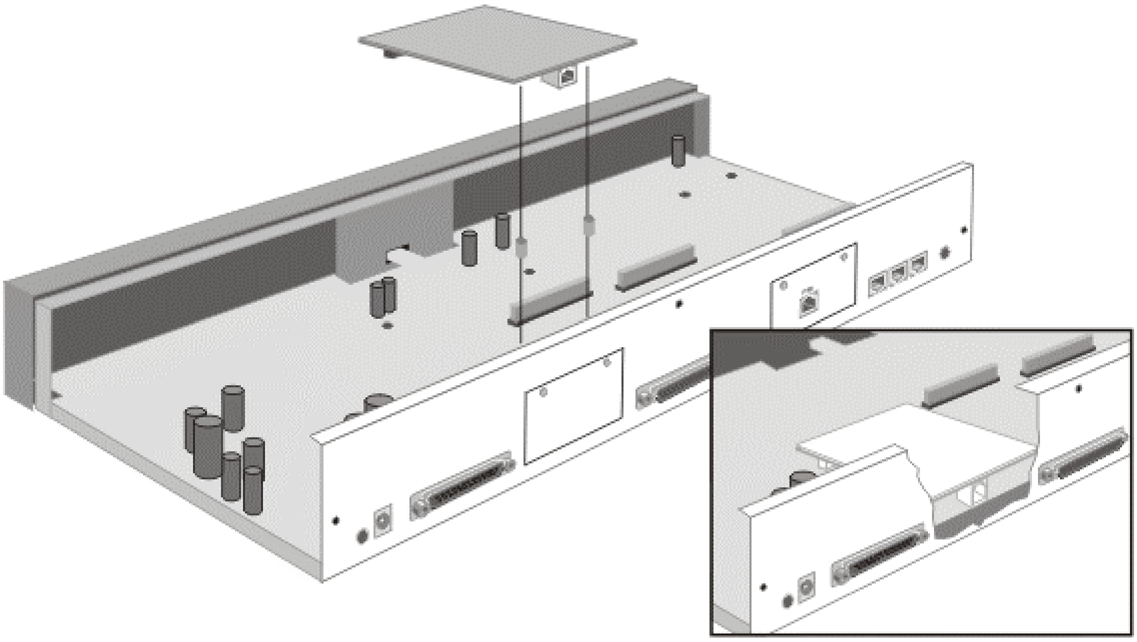
Optional internal upgrades allow for the support of up to 12 x V.90 modem calls and two 4, 8, 16, 24 or 30 channel Voice Compression Modules (VCM). Fitted with two 30 channel voice compression modules, 60 simultaneous voice over IP sessions can be supported. These can be used for either providing networking between sites over a wide area network or supporting IP Telephones and Soft phones. Typically an IP extension only uses a compression module channel when starting a call and while on a call to a non-IP extension/line. It is possible to configure more IP extensions than the capacity of the installed VCM.

The IP412 can be expanded by 12 Expansion Modules, however this is restricted to a maximum capacity of 360 analog, digital or IP phones.



Trunk Interface Cards

Trunk interface cards are rear mounted to provide flexible trunk connectivity for the IP406 V2 and IP412 platforms. The IP406 V2 and IP412 support two trunk interface cards. Dual PRI cards can be used in either slot of IP412 control unit and only Slot A of IP406 control unit.



There are seven trunk interface cards (Not available in all territories).

- IP400 Office Quad BRI.
- IP400 Office PRI E1.
- IP400 Office Dual PRI E1 *(Supported in either slot of an IP412 and slot A of an IP406)*.
- IP400 Office E1R2MFC.
- IP400 Office Dual E1R2MFC *(Supported in either slot of an IP412 and slot A of an IP406)*.
- IP400 Office PRI T1.
- IP400 Office Dual PRI T1 *(Supported in either slot of an IP412 and slot A of an IP406)*.
- IP400 Office Universal Quad Analog Trunk (LS).

IP400 Office BRI Card

The BRI trunk card provides 4 Basic Rate ISDN S/T-Bus interfaces (8 trunks).

Details of the supported supplementary services on BRI interfaces are given in the 'Public and Private Voice Networks' section.

IP400 Office PRI Cards (T1/E1/E1R2)

Available in single and dual versions the IP400 Office PRI card provides single and dual primary rate trunk interfaces respectively. The PRI is available as either T1, E1 or E1R2MFC depending on the market. The dual version is only supported on the IP412 and in slot A of the IP406.

Details of the supported supplementary services and protocols for each PRI are given in the 'Public and Private Voice Networks' section.

T1 trunk cards incorporate an integrated CSU/DSU, eliminating the need for an external unit. The CSU function allows the trunk to be put in loop-back mode for testing purposes. This can be set manually, using the monitor application, or automatically from a Central Office sending a Line Loop Back (LLB) pattern. The DSU function allows the T1 trunk to be shared between data and voice services.

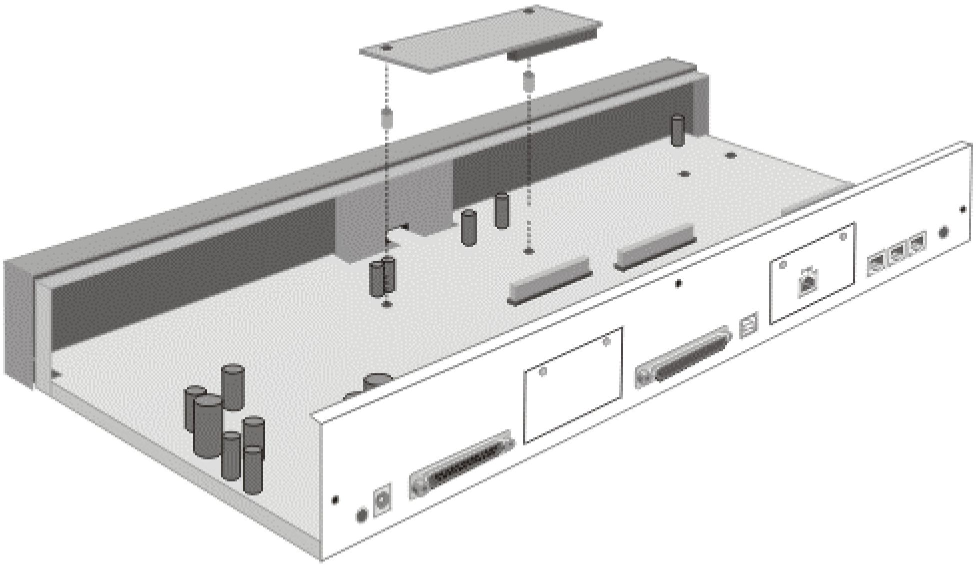
IP400 Office Universal Quad Analog Trunk (LS) Card

Provides four Analog trunk ports. These are 2-wire loop start interfaces including support for caller ID. With IP Office R3.1 and later, this module supports optional 16ms echo cancellation. This card is available in all territories where the IP Office is sold.

Ground start analog trunks are supported via the IP Office Analog Trunk 16 Expansion Module.

Internal Daughter Cards

Internal Daughter Cards are fitted inside the IP406 V2 and IP412 control units.



IP400 Office VC Module – 4/8/16/24/30

The Voice Compression Module (VCM) is used for Voice over IP (VoIP) applications in the IP406 and IP412 control units. Five VCM variants are available supporting 4, 8, 16, 24 and 30 channels of compression. On IP Office - Small Office Edition systems, either 3 or 16 VCM/VoIP channels are pre-built.

The IP406 supports a single VCM while the IP412 can have any two VCMs installed.

The echo cancellation capabilities of the VCM's vary. The VCM 4, 8, 16 and 24 offer 64ms of echo cancellation. while the VCM 30 offers 25ms.

IP400 Internal Modem Card

An internal modem card with 12 modems can be installed in both the IP406 and IP412 to provide dial-up capacity that is better matched to remote access requirements of customers. The Internal Modem card allows up to 12 simultaneous V.90 (56kbps) analog modem calls into the IP Office.

External Expansion Modules

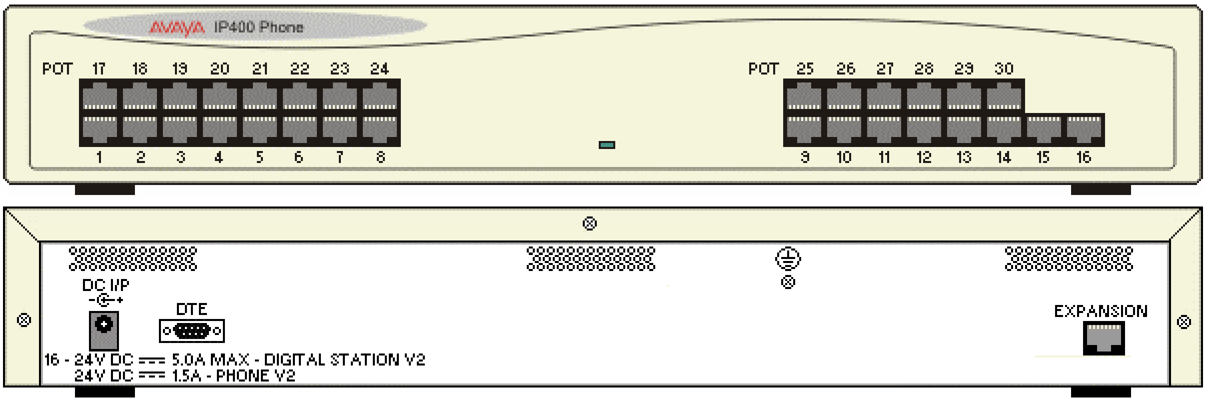
External Expansion Modules

Except for the WAN3 10/100 module, the IP406 V2 supports six expansion modules and the IP412 supports twelve expansion modules. The WAN3 10/100 differs in that it connects to the control unit via a LAN port rather than one of the rear expansion ports. A maximum of 2 WAN3 10/100 modules are supported on the IP406 V2 and IP412 - they are not supported on the IP Office - Small Office Edition.

- **IP400 Office Phone Expansion Module**
  Available in three variants for 8, 16 or 30 extensions.
- **IP400 Office Digital Station Expansion Module**
  Available in two variants for 16 or 30 extensions.
- **IP400 Office So8 Expansion Module**
  Not available in some territories.
- **IP400 Office WAN 3 Expansion Module**
- **IP400 Office Analog Trunk 16 Expansion Module**

IP400 Office Phone V2 Module

This module provides additional two-wire analog telephone interfaces (POT ports). The IP400 Office Phone V2 module is available in 3 versions, giving 8, 16 or 30 extensions. Telephones can be located up to 1km from the control unit - for cabling details please see the IP Office Installation Manual. For extensions located "out-of-the-building" additional line protection will be needed.



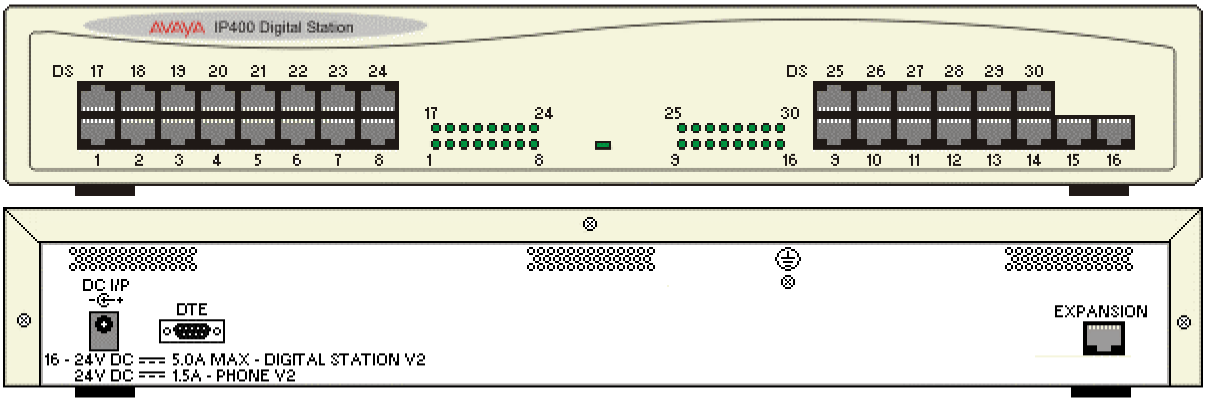
- The Phone V2 expansion module supports several different methods of message waiting indication. They are: 81V, 101V, 51V Stepped (pulsed) and Line Reversal.

IP Office Digital Station V2 Module

This expansion module provides additional Digital Station (DS) ports for supported 2400, 5400 and T3 Series phones plus 3810 wireless phones. The IP400 Office Digital Station V2 module is available in 2 versions; 16 or 30 extensions.

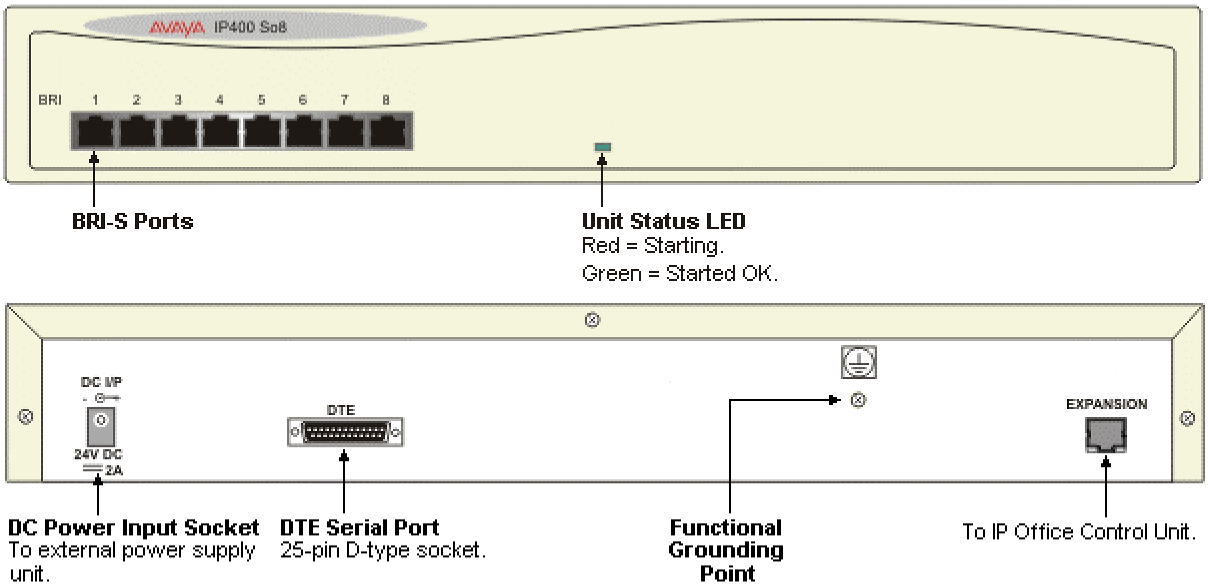
Telephones can be located up to 1km from the control unit - for cabling details please see the IP Office Installation Manual. For extensions located "out-of-the-building" additional line protection will be needed.

- It is not possible to mix T3 Series and any other digital phone type on the same installation, although T3 and 5600 Series IP phones can be mixed.
- A maximum of eight EU24/EU24BL units are supported on any Digital Station module. Note that if EU24 or EU24BL units are being used with IP phones, as maximum of eight are supported on the whole system.



IP400 Office So8 Module

The IP400 Office So8 module provides 8 S-Bus interfaces for Basic Rate ISDN devices.



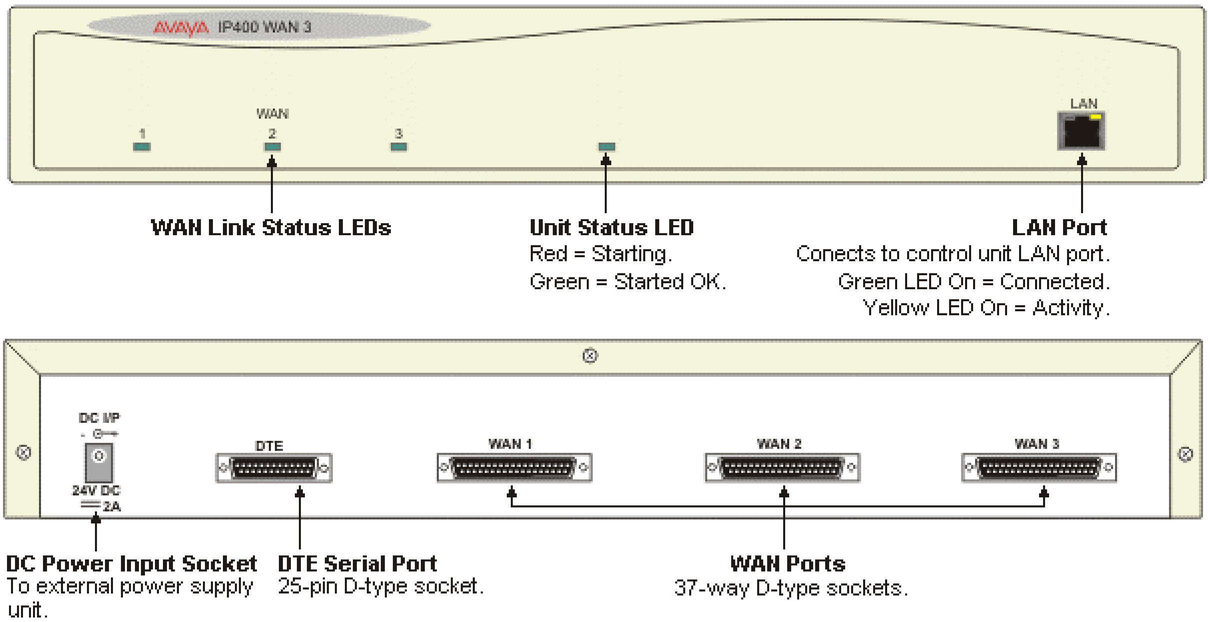
The IP Office So8 expansion module supports both point-to-point and point-to-multipoint connections. A maximum of 10 terminal endpoints identifiers (TEIs) are supported.

IP400 Office WAN3 10/100

The IP400 Office WAN 3 10/100 module provides three WAN connections (X21, V35 or V24 via a 37way D Type socket and using an appropriate connector cable). Line speeds up to and including 2Mbps are supporting on each interface, the carrier providing the line dictates the actual operating speed i.e. in some territories the maximum speed may be 1.544M. These interfaces are identical to the single connection provided as standard on the IP406 and IP412 platforms.

The IP400 Office WAN3 10/100 may be connected to the IP406 and IP412 platforms to provide additional WAN ports. Each platform, except Small Office Edition, can support up to two WAN3 10/100 modules.

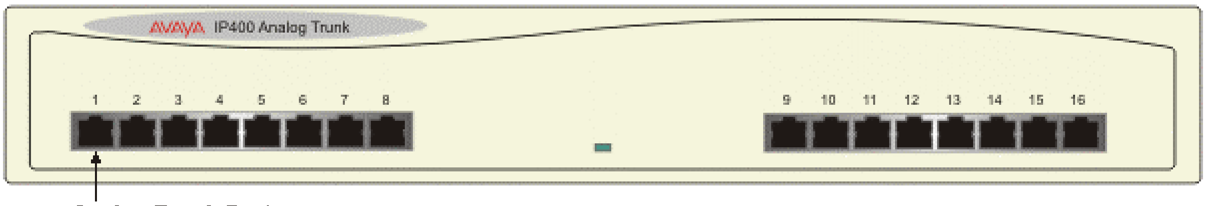
The IP400 Office WAN3 connects to the base unit through the Local Area Network via a 10/100Mbps connection and does not use an expansion port on the control unit.



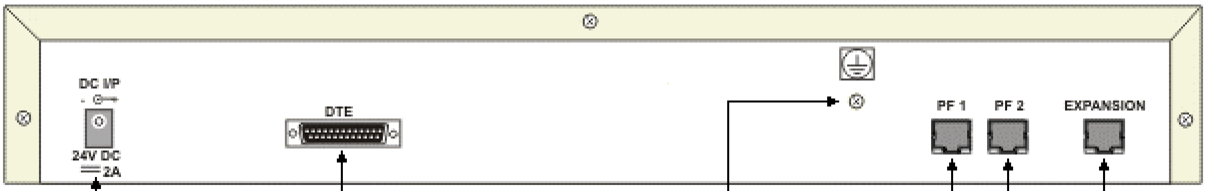
IP400 Office Analog Trunk 16

This expansion modules provides an additional sixteen Loop Start or Ground Start two-wire analog trunks. The first two trunks on the module are automatically switched to power fail sockets on the rear of the unit in the event of power being interrupted.

- **Note:** Ground start trunks are not available in all territories.



Analog Trunk Ports
Individually switchable to loop or ground start.



DC Power Input Socket
To external power supply unit.

DTE Serial Port
25-pin D-type socket.

Functional Grounding Point

Power Fail Analog Phone Sockets
During power fail:
PF1 = Analog Trunk 1.
PF2 = Analog Trunk 2.

To IP Office Control Unit.

4. Telephones

Introduction to IP Office Telephones

All Avaya telephones are designed to ensure that features and functions are easily accessible to the user ensuring that, through ease of use, the full benefits of the system are delivered to the desktop.

With release 3.1, Avaya is introducing a new series of DECT telephones – the 3701 and 3711 handsets. Additionally, a new series of telephones is being introduced in EMEA – the Avaya T3 Upn series of phones.

The telephones listed below are the preferred and premier range of telephones for use on the IP Office. These telephones are sold worldwide in every country that the IP Office is available. This telephone range consists of both digital and IP telephones.

IP Office worldwide digital phones:

- 5402 Telephone.
- 5410 Telephone.
- 5420 Telephone.

IP Office worldwide H.323 IP phones:

- 5601 IP Telephone.
- 5602SW IP Telephone.
- 5610 IP Telephone.
- 5620 IP Telephone.

In addition to the telephones above, the IP Office supports a wide range of phones as listed below. However, note that some of those phones are only available in certain countries and regions.

North America and CALA

- 4406D Telephone.
- 4412D Telephone.
- 4424D Telephone.
- 4450 DSS Unit.
- 9040 Wireless Telephone.*
- 3810 Wireless Telephone.

EMEA and APAC

- 20DT DECT Telephone.
- T3 Compact (Upn).
- T3 Comfort (Upn).
- T3 Classic (Upn).
- 3701 IP DECT Wireless Handset.
- 3711 IP DECT Wireless Handset.
- Interquartz Gemini 9281-AV, 9330-AV and 9335-AV analog telephones.

Phones supported worldwide in addition to 5400 Series.

- 2402 Telephone.
- 2410 Telephone.
- 2420 Telephone.
- 6408D Telephone.
- 6416D Telephone.
- 6424D Telephone.*
- XM24 DSS Unit.
- EU24/EU24BL DSS Unit.
- Analog Telephones**.

H.323 IP phones supported worldwide in addition to the 5600 Series.

- 4601 IP Telephone.
- 4602 IP Telephone.*
- 4602SW IP Telephone.
- 4606 IP Telephone.*
- 4610 IP Telephone.
- 4612 IP Telephone.*
- 4620 IP Telephone.
- 4624 IP Telephone.*
- 3616 Executive Wireless (WiFi) Phone.
- 3626 Ruggedized Wireless (WiFi) Phone.

- For maximum cabling distances please refer to the IP Office Installation Manual.
- IP Office does not support SIP telephones in R3.1 or earlier releases.
- Those phones that support handsfree operation are intended for individual use only, not for group and conference room operation.

*These phones are no longer available as new from Avaya but are still supported by Avaya IP Office R3.1.

**Avaya does not guarantee that all analog phones will work in every region, however most analog phones will work on the IP Office.

20DT - DECT Telephone

This telephone is the preferred telephone for the Avaya IP Office non-IP DECT systems. It can be used with the Avaya IP DECT system – however, it will only provide basic Generic Access Profile (GAP) functionality when used with the Avaya IP DECT system.

IP Office 20DT DECT telephone features include:

- 2 Independent User Profiles for ringer/volume settings.
- 36 Character LCD display.
- 5 Display Icons.
- Intuitive Keys for driving the display.
- Keys for volume control & off-hook.
- Vibrating ringer.
- 10 Number Redial Store.
- 80 Number Local Phone Book.
- Keypad lock.
- 9 tone ringer options.
- Headset connection.
- Automatic Answer Option (used with headsets).
- 10 hours talk time and 90 hours stand by time.
- Lightweight, weighing less than 130gms.
- Pocket size (dimension 143mm x 48mm x 26mm).
- Connects to the IP Office via a non-IP DECT base station which connects using POT ports. See IP Office DECT.

Option handset accessories include:-

- A desktop charger.
- An adapter cord for use with headsets.
- Handset cover including a robust belt clip.
- Heavy-duty belt clip.

2402D Telephone

Features:

- **Used on:** Avaya IP Office and or Avaya Communication Manager.
- **Connect to:** Digital Station (DS) port.
- **Mounting:** Desk or wall mountable.
- **Adjustable Desk Stand:** No.
- **Display:** 2 lines x 24 characters.
  (IP Office offers only limited support for the 2402. The display on the 2402 is not supported on IP Office. If full display functionality is needed Avaya recommends the 5402.)
- **Fixed Feature Buttons:** 10 - Conference, Transfer, Drop, Redial, Speaker, Messages, Hold, Mute, Volume Up, Volume Down.
- **Programmable Feature Buttons:** 2 plus an additional 12 programmable feature keys can be accessed via the FEATURE key (not suitable for call appearance features).
- **Key Labels:** Icons used on fixed feature keys. Display labels and icons used on 2 programmable feature keys.
- **Speakerphone:** Listen-only handsfree speaker (no microphone).
- **Hearing Aid Compatible:** Yes.
- **Message Waiting Indicator:** Yes - also used as ringing call alert indicator.
- **Personalized Ring Patterns:** Yes - 8.
- **Headset Socket:** No.
- **Embedded Applications:** None.
- **Upgradeable Firmware:** No.
- **Optional Add-Ons:** None.
- **Color:** Multi-grey.
- **Power Supply:** From phone system.

2410D Telephone

Features:

- **Used on:** Avaya IP Office and or Avaya Communication Manager.
- **Connect to:** Digital Station (DS) port.
- **Mounting:** Desk or wall mountable.
- **Adjustable Desk Stand:** Yes - Supplied with phone.
- **Display:** 5 lines x 29 characters (168 x 80 pixel 4-grayscale).
- **Fixed Feature Buttons:** 11 - Conference, Headset, Transfer, Drop, Redial, Speaker, Messages, Hold, Mute, Volume Up, Volume Down.
- **Programmable Feature Buttons:** 12 - in 2 switchable display pages of 6 matching the 6 physical display buttons.
- **Key Labels:** Icons used on fixed feature keys. Display labels and icons used on 2 programmable feature keys.
- **Speakerphone:** Two-way handsfree speaker and microphone.
- **Hearing Aid Compatible:** Yes.
- **Message Waiting Indicator:** Yes - also used as ringing call alert indicator.
- **Personalized Ring Patterns:** Yes - 8.
- **Headset Socket:** Yes.
- **Embedded Applications:** Speed Dial List (48) and Call Log (Missed, Incoming, Outgoing).
- **Upgradeable Firmware:** Yes.
- **Optional Add-Ons:** None.
- **Color:** Multi-grey.
- **Power Supply:** From phone system.

2420D Telephone

The 2420D supports the following features:

Features:

- **Used on:** Avaya IP Office and or Avaya Communication Manager.
- **Connect to:** Digital Station (DS) port.
- **Mounting:** Desk or wall mountable.
- **Adjustable Desk Stand:** Yes - Supplied with phone.
- **Display:** 7 lines x 29 characters.
- **Fixed Feature Buttons:** 11 - Conference, Headset, Transfer, Drop, Redial, Speaker, Messages, Hold, Mute, Volume Up, Volume Down.
- **Programmable Feature Buttons:** 24 - arranged in 3 switchable display pages of 8 matching the 8 physical display buttons..
- **Key Labels:** Icons used on fixed feature keys. Display labels and icons used on 2 programmable feature keys.
- **Speakerphone:** Two-way handsfree speaker and microphone.
- **Hearing Aid Compatible:** Yes.
- **Message Waiting Indicator:** Yes - also used as ringing call alert indicator.
- **Personalized Ring Patterns:** Yes - 8.
- **Headset Socket:** Yes.
- **Embedded Applications:** Speed Dial List (104) and Call Log (Missed, Incoming, Outgoing).
- **Upgradeable Firmware:** Yes.
- **Optional Add-Ons:** EU24 - requires phone to use an individual power supply unit (1151B1/B2).
- **Color:** Multi-grey.
- **Power Supply:** From phone system.

3616 Executive Wireless Telephone

The Avaya 3616 IP Wireless Telephone is a WiFi (802.11b) telephone that runs using H.323.

The 3616 supports the following features:

- Lightweight innovative design .
- Simple to use.
- 802.11b standard-compatible.
- Radio Frequency 2.4000 – 2.835 GHz (SMI).
- Transmission type Direct Sequence Spread Spectrum (DSSS).
- FCC certification Part 15.247.
- Management of telephones via DHCP and TFTP.
- Voice encoding G711.
- Transmit Power 100mw peak, <10mW average.
- Wired Equivalent Privacy (WEP), 40bit and 128 bit.
- 2x16 character alphanumeric, plus status indicators.
- 4 hours talk time and 80 hours standby.

Note: 3616 supports both R1.0 and R2.0 firmware on the set itself. However, as of R3.1 of IP Office, only 3616 phone R1.0 firmware is supported.

3626 Ruggedized Wireless Telephone

The Avaya 3626 Wireless Telephone is a WiFi standard (802.11b) telephone that runs using H.323.

The 3626 supports all of the features of 3616 with the following differences:

- Designed for industrial environments.
- Ruggedized durable design.
- Push-to-talk (walkie-talkie) feature for broadcast communications between employees.

Note: 3626 supports both R1.0 and R2.0 firmware on the set itself. However, as of R3.1 of IP Office, only 3626 phone R1.0 firmware is supported.

3701 IP DECT Telephone

This handset is supported on the Avaya IP DECT system only.

- Listen-only handsfree speaker.
- SOS Emergency key for speed dialling an emergency number.
- Information key that can be used for:
    - Phone number lists and voice mail indication.
    - Information and speaker key flash when active.
- 50 phone book entries in every handset, independent of the system phone book.
- 10 possible ring tones with temporary mute.
- 4-level signal strength display.
- Speaker and handset volume, 3-levels and mute capability.
- Manual and automatic key lock (1 minute timer).
- Temporary ring tone muting.
- Silent charging.
- 12 menu languages: Czech, Danish, Dutch, English, Finnish, French, German, Italian, Norwegian, Portuguese, Spanish and Swedish. However, in the Czech and Norwegian language mode some menu items may appear in the English language.
- Illuminated 3-line graphic display (96 x 33 pixels), variable 3-level contrast.
- Stand-by time: up to 200 hours.
- Talk time: up to 20 hours.
- Charge time: max. 6 hours for empty batteries.
- Weight: 138 grammes including 3 AAA (NiMH) batteries.
- Dimensions (Height x Width X Depth): 146 x 55 x 28 mm.

Optional telephone accessories include:

- Desktop charger.
- An adapter cord for use with headsets.
- Heavy-duty belt clip.

3711 IP DECT Telephone

This telephone is supported on the Avaya IP DECT system only.

The 3711 phone supports the same features as the 3701 IP DECT handset but with the following differences:

- Full hands-free speakerphone operation.
- Headset connection (2.5 mm jack).
- Vibrating alarm.
- Personal phone book with 100 entries, independent of the system phone book.
- Voice Mail indication.
- Choice from 30 ring tones.
- Speaker and handset volume, 7-levels and mute capability.
- Automatic call pick-up using a headset.
- 10 menu languages: Danish, Dutch, English, Finnish, French, German, Italian, Portuguese, Spanish and Swedish.
- Illuminated 5-line graphic display, (96 x 60 pixels), variable 7-level contrast.

Optional handset accessories include:

- Desktop charger.
- An adapter cord for use with headsets.
- Heavy-duty belt clip.

3810 Wireless HandsetTelephone

The 3810 supports the following features:

- 2-line, 32 character Handset Liquid Crystal Display (LCD).
- 4 displayed operation modes indicating Talk, Ringer On/Off, Battery Low, and Message Waiting.
- Single button access to fixed features – Hold, Transfer, Conference, and Redial.
- 4 programmable buttons to access features on the PBX.
- 10 channels, supporting up to 10 simultaneous conversations.
- Headset jack.
- Ringer and Handset volume control.
- User selectable ring type.
- Vibrate alert.
- Base Unit and Charger Unit.
- Base station connects to an IP Office DS (Digital Station) port.

The Avaya 3810 Wireless Telephone is a digital telephone designed to work with IP Office (minimum release 2.0) by connecting to a DS port. It offers the mobility inherent in a wireless telephone plus access to a number of features and functionality of the connected communications system. The Avaya 3810 wireless telephone uses 900 MHz digital technology allowing a maximum range of 160 feet from the base station.

A maximum of 3 to 5 Avaya 3810 wireless handsets can be connected to the same IP Office in any given area.

4406D Telephone

The 4406 supports the following features:

- 6 Programmable call appearance/feature keys with twin lamps.
- 8 Fixed Feature Keys: Speaker, Mute, Hold, Volume Up & Down, Conference, Transfer, Redial.
- 2 x 16 Character Display.
- Message waiting indicator.
- Two-way handsfree speaker phone.
- Hearing aid compatible.
- Optional wall mounting/desk stand.
- Connects to an IP Office DS (Digital Station) port.

Note that this telephone does not support integrated directory access on the IP Office. This phone does not support personalized ringing.

4412D Telephone

The 4412 supports all of the features of the 4406 with the following differences:

- 12 Programmable call appearance/feature keys with twin lamps.
- 12 Programmable feature keys without lamps (ie. not suitable for call appearance features).
- 4 Display Navigation Keys, right of the display: Menu, Previous (<), Next (>), & Exit.
- 4 Display Soft Keys below the Display.
- 8 Fixed Feature Keys: Speaker, Mute, Conference, Transfer, Redial, Hold, Volume Up/Down.
- DSS port to support 2 DSS4450 adjuncts; Auxiliary power required.
- 2x24 Character Display.
- Two-way handsfree speaker phone.
- Optional wall mounting/desk stand.
- Connects to an IP Office DS (Digital Station) port.

This phone does not support personalized ringing.

4424D Telephone

The 4424D supports all of the features of the 4406 with the following differences:

- 24 Programmable call appearance/feature keys with twin lamps.
- 8 Fixed Feature Keys: Speaker, Mute, Conference, Transfer, Redial, Hold, Volume Up & Down.
- 4 Display Soft Keys below the Display.
- 4 Display Navigation Keys, right of the display: Menu, Previous (<), Next (>), & Exit.
- DSS port to support 2 DSS4450 adjuncts. Auxiliary power required.
- 2 x 24 character display.
- Connects to an IP Office DS (Digital Station) port.

Note: A maximum of sizteen 4424D+ telephones are supported on any single IP Office Digital Station 30 V2 expansion module.

This phone does not support personalized ringing.

DSS4450 Unit

The DSS4450 works in association with the 4412D+ and 4424D+ telephones, each of which can support up to two DSS4450 adjuncts.

Auxiliary power is required. Each DSS4450 provides an additional 60 programmable keys with single red lamps except for the bottom two rows which have green lamps. The DSS4450 requires a power supply unit for the phone, power supply socket and must be used with the cables supplied.

Each IP Office DS module, including control units with integral DS ports, supports a maximum of two EU24, XM24 and/or 4450 units only.

4601 IP Hardphone

The 4601 supports the following features:

Features:

- **Used on:** Avaya IP Office and or Avaya Communication Manager.
- **Connect to:** LAN using H.323 VoIP.
- **Mounting:** Desk or wall mountable.
- **Adjustable Desk Stand:** No.
- **Display:** None.
- **Fixed Feature Buttons:** 8 - Conference, Transfer, Drop, Redial, Messages, Hold, Volume Up, Volume Down.
- **Programmable Feature Buttons:** 2 with single colour indicator lamps.
- **Key Labels:** Icons used on fixed feature keys. None on programmable feature keys.
- **Speakerphone:** No.
- **Hearing Aid Compatible:** Yes.
- **Message Waiting Indicator:** Yes - also useable a ringing call alert indicator.
- **Personalized Ring Patterns:** No.
- **Headset Socket:** No.
- **Embedded Applications:** None.
- **Upgradeable Firmware:** Yes.
- **Optional Add-Ons:** None.
- **Color:** Multi-grey.
- **Power Supply:** IEEE 802.11af Power over Ethernet (PoE) or individual power supply unit (1151B1/B2).

IP Features:

- **Codecs:** G.711, G.729a/b.
- **QoS Options:** UDP Port Selection, DiffServ and 802.1p/B (VLAN)
- **SNMP Support:** Yes.
- **IP Address Assignment:** Static or dynamic IP address assignment.
- **Ethernet Ports:** Single 10/100 BaseT Ethernet port.

4602SW IP Telephone

Features:

- **Used on:** Avaya IP Office and or Avaya Communication Manager.
- **Connect to:** LAN using H.323 VoIP.
- **Mounting:** Desk or wall mountable.
- **Adjustable Desk Stand:** No.
- **Display:** 2 Lines x 24 Character.
- **Fixed Feature Buttons:** 9 - Conference, Transfer, Drop, Redial, Speaker, Hold, Mute, Volume Up, Volume Down.
- **Programmable Feature Buttons:** 2 with adjacent display icons and labels.
- **Key Labels:** Icons used on fixed feature keys. Display used for programmable feature keys.
- **Speakerphone:** Listen-only handsfree speaker (no microphone).
- **Hearing Aid Compatible:** Yes.
- **Message Waiting Indicator:** Yes - also used as ringing call alert indicator.
- **Personalized Ring Patterns:** No.
- **Headset Socket:** No.
- **Embedded Applications:** None.
- **Upgradeable Firmware:** Yes.
- **Optional Add-Ons:** None.
- **Color:** Multi-grey.
- **Power Supply:** IEEE 802.11af Power over Ethernet (PoE) or individual power supply unit (1151B1/B2).

IP Features:

- **Codecs:** G.711, G.729a/b.
- **QoS Options:** UDP Port Selection, DiffServ and 802.1p/B (VLAN)
- **SNMP Support:** Yes.
- **IP Address Assignment:** Static or dynamic IP address assignment.
- **Ethernet Ports:** Two port full-duplex 10/100 BaseT Ethernet switch for PC pass-through connection.
    - Auto-negotiation provided separately for each port.
    - 802.3 Flow Control.
    - Phone has priority over PC port at all times.

4610 IP Hardphone

Features:

- **Used on:** Avaya IP Office and or Avaya Communication Manager.
- **Connect to:** LAN using H.323 VoIP.
- **Mounting:** Desk or wall mountable.
- **Adjustable Desk Stand:** Yes - included with phone.
- **Display:** 5 line x 29 character (168 x 80 pixel 4-grayscale).
- **Fixed Feature Buttons:** 10 - Conference, Headset, Transfer, Drop, Redial, Speaker, Hold, Mute, Volume Up, Volume Down.
- **Programmable Feature Buttons:** 2 with adjacent display icons and labels.
- **Key Labels:** Icons used on fixed feature keys. Display used for programmable feature keys.
- **Speakerphone:** Two-way handsfree speaker and microphone.
- **Hearing Aid Compatible:** Yes.
- **Message Waiting Indicator:** Yes - also used as ringing call alert indicator.
- **Personalized Ring Patterns:** Yes - 8.
- **Headset Socket:** Yes.
- **Embedded Applications:** Speed Dial List (48), Call Log (Missed, Incoming, Outgoing) and WAP WML browser.
- **Upgradeable Firmware:** Yes.
- **Optional Add-Ons:** None.
- **Color:** Multi-grey.
- **Power Supply:** IEEE 802.11af Power over Ethernet (PoE) or individual power supply unit (1151B1/B2).

IP Features:

- **Codecs:** G.711, G.729a/b.
- **QoS Options:** UDP Port Selection, DiffServ and 802.1p/b (VLAN)
- **SNMP Support:** Yes.
- **IP Address Assignment:** Static or dynamic IP address assignment.
- **Ethernet Ports:** Two port full-duplex 10/100 BaseT Ethernet switch for PC pass-through connection.
    - Auto-negotiation provided separately for each port.
    - 802.3 Flow Control.
    - Phone has priority over PC port at all times.

4621 IP Telephone

Features:

- **Used on:** Avaya IP Office and or Avaya Communication Manager.
- **Connect to:** LAN using H.323 VoIP.
- **Mounting:** Desk or wall mountable.
- **Adjustable Desk Stand:** Yes - included with phone.
- **Display:** 7 line x 29 character (168 x 132 pixels). Display backlight supported with local power (not PoE).
- **Fixed Feature Buttons:** 10 - Conference, Headset, Transfer, Drop, Redial, Speaker, Hold, Mute, Volume Up, Volume Down.
- **Programmable Feature Buttons:** 24 - arranged in 2 switchable display pages of 12 matching the 12 physical display buttons.
- **Key Labels:** Icons used on fixed feature keys. Display used for programmable feature keys.
- **Speakerphone:** Two-way handsfree speaker and microphone.
- **Hearing Aid Compatible:** Yes.
- **Message Waiting Indicator:** Yes - also used as ringing call alert indicator.
- **Personalized Ring Patterns:** Yes - 8.
- **Headset Socket:** Yes.
- **Embedded Applications:** Speed Dial List (108), Call Log (Missed, Incoming, Outgoing) and WAP WML browser.
- **Upgradeable Firmware:** Yes.
- **Optional Add-Ons:** EU24/EU24BL - requires phone to use an individual power supply unit (1151B1/B2).
- **Color:** Multi-grey.
- **Power Supply:** IEEE 802.11af Power over Ethernet (PoE) or individual power supply unit (1151B1/B2).

IP Features:

- **Codecs:** G.711, G.729a/b.
- **QoS Options:** UDP Port Selection, DiffServ and 802.1p/b (VLAN)
- **SNMP Support:** Yes.
- **IP Address Assignment:** Static or dynamic IP address assignment.
- **Ethernet Ports:** Two port full-duplex 10/100 BaseT Ethernet switch for PC pass-through connection.
    - Auto-negotiation provided separately for each port.
    - 802.3 Flow Control.
    - Phone has priority over PC port at all times.

5402 Telephone

Features:

- **Used on:** Avaya IP Office only.
- **Connect to:** Digital Station (DS) port.
- **Mounting:** Desk or wall mountable.
- **Adjustable Desk Stand:** No.
- **Display:** 2 lines x 24 characters.
- **Fixed Feature Buttons:** 10 - Conference, Transfer, Drop, Redial, Speaker, Messages, Hold, Mute, Volume Up, Volume Down.
- **Programmable Feature Buttons:** 2 plus an additional 12 programmable feature keys can be accessed via the FEATURE key (not suitable for call appearance features).
- **Key Labels:** Icons used on fixed feature keys. Display labels and icons used on 2 programmable feature keys.
- **Speakerphone:** Listen-only handsfree speaker (no microphone).
- **Hearing Aid Compatible:** Yes.
- **Message Waiting Indicator:** Yes - also used as ringing call alert indicator.
- **Personalized Ring Patterns:** Yes - 8.
- **Headset Socket:** No.
- **Embedded Applications:** None.
- **Upgradeable Firmware:** No.
- **Optional Add-Ons:** None.
- **Color:** Multi-grey.
- **Power Supply:** From phone system.

5410 Telephone

The 5410 supports the following features:

- **Features:**
    - **Used on:** Avaya IP Office only.
    - **Connect to:** Digital Station (DS) port.
    - **Mounting:** Desk or wall mountable.
    - **Adjustable Desk Stand:** Yes - Supplied with phone.
    - **Display:** 5 lines x 29 characters (168 x 80 pixel 4-grayscale).
    - **Fixed Feature Buttons:** 11 - Conference, Headset, Transfer, Drop, Redial, Speaker, Messages, Hold, Mute, Volume Up, Volume Down.
    - **Programmable Feature Buttons:** 12 - in 2 switchable display pages of 6 matching the 6 physical display buttons.
    - **Key Labels:** Icons used on fixed feature keys. Display labels and icons used on 2 programmable feature keys.
    - **Speakerphone:** Two-way handsfree speaker and microphone.
    - **Hearing Aid Compatible:** Yes.
    - **Message Waiting Indicator:** Yes - also used as ringing call alert indicator.
    - **Personalized Ring Patterns:** Yes - 8.
    - **Headset Socket:** Yes.
    - **Embedded Applications:** Speed Dial List (48) and Call Log (Missed, Incoming, Outgoing).
    - **Upgradeable Firmware:** Yes.
    - **Optional Add-Ons:** None.
    - **Color:** Multi-grey.
    - **Power Supply:** From phone system.

5420 Telephone

Features:

- **Used on:** Avaya IP Office only.
- **Connect to:** Digital Station (DS) port.
- **Mounting:** Desk or wall mountable.
- **Adjustable Desk Stand:** Yes - Supplied with phone.
- **Display:** 7 lines x 29 characters.
- **Fixed Feature Buttons:** 11 - Conference, Headset, Transfer, Drop, Redial, Speaker, Messages, Hold, Mute, Volume Up, Volume Down.
- **Programmable Feature Buttons:** 24 - arranged in 3 switchable display pages of 8 matching the 8 physical display buttons..
- **Key Labels:** Icons used on fixed feature keys. Display labels and icons used on 2 programmable feature keys.
- **Speakerphone:** Two-way handsfree speaker and microphone.
- **Hearing Aid Compatible:** Yes.
- **Message Waiting Indicator:** Yes - also used as ringing call alert indicator.
- **Personalized Ring Patterns:** Yes - 8.
- **Headset Socket:** Yes.
- **Embedded Applications:** Speed Dial List (104) and Call Log (Missed, Incoming, Outgoing).
- **Upgradeable Firmware:** Yes.
- **Optional Add-Ons:** EU24 - requires phone to use an individual power supply unit (1151B1/B2).
- **Color:** Multi-grey.
- **Power Supply:** From phone system.

5601 IP Hardphone

The 5601 supports the following features:

Features:

- **Used on:** Avaya IP Office only.
- **Connect to:** LAN using H.323 VoIP.
- **Mounting:** Desk or wall mountable.
- **Adjustable Desk Stand:** No.
- **Display:** None.
- **Fixed Feature Buttons:** 8 - Conference, Transfer, Drop, Redial, Messages, Hold, Volume Up, Volume Down.
- **Programmable Feature Buttons:** 2 with single colour indicator lamps.
- **Key Labels:** Icons used on fixed feature keys. None on programmable feature keys.
- **Speakerphone:** No.
- **Hearing Aid Compatible:** Yes.
- **Message Waiting Indicator:** Yes - also used as ringing call alert indicator.
- **Personalized Ring Patterns:** No.
- **Headset Socket:** No.
- **Embedded Applications:** None.
- **Upgradeable Firmware:** Yes.
- **Optional Add-Ons:** None.
- **Color:** Multi-grey.
- **Power Supply:** IEEE 802.11af Power over Ethernet (PoE) or individual power supply unit (1151B1/B2).

IP Features:

- **Codecs:** G.711, G.729a/b.
- **QoS Options:** UDP Port Selection, DiffServ and 802.1p/B (VLAN)
- **SNMP Support:** Yes.
- **IP Address Assignment:** Static or dynamic IP address assignment.
- **Ethernet Ports:** Single 10/100 BaseT Ethernet port.

5602 IP Hardphone

The 5602SW supports the following features:

- **Features:**
    - **Used on:** Avaya IP Office only.
    - **Connect to:** LAN using H.323 VoIP.
    - **Mounting:** Desk or wall mountable.
    - **Adjustable Desk Stand:** No.
    - **Display:** 2 Lines x 24 Character.
    - **Fixed Feature Buttons:** 9 - Conference, Transfer, Drop, Redial, Speaker, Hold, Mute, Volume Up, Volume Down.
    - **Programmable Feature Buttons:** 2 with adjacent display icons and labels.
    - **Key Labels:** Icons used on fixed feature keys. Display used for programmable feature keys.
    - **Speakerphone:** Listen-only handsfree speaker (no microphone).
    - **Hearing Aid Compatible:** Yes.
    - **Message Waiting Indicator:** Yes - also used as ringing call alert indicator.
    - **Personalized Ring Patterns:** No.
    - **Headset Socket:** No.
    - **Embedded Applications:** None.
    - **Upgradeable Firmware:** Yes.
    - **Optional Add-Ons:** None.
    - **Color:** Multi-grey.
    - **Power Supply:** IEEE 802.11af Power over Ethernet (PoE) or individual power supply unit (1151B1/B2).

- **IP Features:**
    - **Codecs:** G.711, G.729a/b.
    - **QoS Options:** UDP Port Selection, DiffServ and 802.1p/B (VLAN)
    - **SNMP Support:** Yes.
    - **IP Address Assignment:** Static or dynamic IP address assignment.
    - **Ethernet Ports:** Two port full-duplex 10/100 BaseT Ethernet switch for PC pass-through connection.
        - Auto-negotiation provided separately for each port.
        - 802.3 Flow Control.
        - Phone has priority over PC port at all times.

5610 IP Hardphone

Features:

- **Used on:** Avaya IP Office only.
- **Connect to:** LAN using H.323 VoIP.
- **Mounting:** Desk or wall mountable.
- **Adjustable Desk Stand:** Yes - included with phone.
- **Display:** 5 line x 29 character (168 x 80 pixel 4-grayscale).
- **Fixed Feature Buttons:** 10 - Conference, Headset, Transfer, Drop, Redial, Speaker, Hold, Mute, Volume Up, Volume Down.
- **Programmable Feature Buttons:** 2 with adjacent display icons and labels.
- **Key Labels:** Icons used on fixed feature keys. Display used for programmable feature keys.
- **Speakerphone:** Two-way handsfree speaker and microphone.
- **Hearing Aid Compatible:** Yes.
- **Message Waiting Indicator:** Yes - also used as ringing call alert indicator.
- **Personalized Ring Patterns:** Yes - 8.
- **Headset Socket:** Yes.
- **Embedded Applications:** Speed Dial List (48), Call Log (Missed, Incoming, Outgoing) and WAP WML browser.
- **Upgradeable Firmware:** Yes.
- **Optional Add-Ons:** None.
- **Color:** Multi-grey.
- **Power Supply:** IEEE 802.11af Power over Ethernet (PoE) or individual power supply unit (1151B1/B2).

IP Features:

- **Codecs:** G.711, G.729a/b.
- **QoS Options:** UDP Port Selection, DiffServ and 802.1p/b (VLAN)
- **SNMP Support:** Yes.
- **IP Address Assignment:** Static or dynamic IP address assignment.
- **Ethernet Ports:** Two port full-duplex 10/100 BaseT Ethernet switch for PC pass-through connection.
  - Auto-negotiation provided separately for each port.
  - 802.3 Flow Control.
  - Phone has priority over PC port at all times.

5620 IP Hardphone

Features:

- **Used on:** Avaya IP Office only.
- **Connect to:** LAN using H.323 VoIP.
- **Mounting:** Desk or wall mountable.
- **Adjustable Desk Stand:** Yes - included with phone.
- **Display:** 7 line x 29 character (168 x 132 pixels). Display backlight supported with local power (not PoE).
- **Fixed Feature Buttons:** 10 - Conference, Headset, Transfer, Drop, Redial, Speaker, Hold, Mute, Volume Up, Volume Down.
- **Programmable Feature Buttons:** 24 - arranged in 2 switchable display pages of 12 matching the 12 physical display buttons.
- **Key Labels:** Icons used on fixed feature keys. Display used for programmable feature keys.
- **Speakerphone:** Two-way handsfree speaker and microphone.
- **Hearing Aid Compatible:** Yes.
- **Message Waiting Indicator:** Yes - also used as ringing call alert indicator.
- **Personalized Ring Patterns:** Yes - 8.
- **Headset Socket:** Yes.
- **Embedded Applications:** Speed Dial List (108), Call Log (Missed, Incoming, Outgoing) and WAP WML browser.
- **Upgradeable Firmware:** Yes.
- **Optional Add-Ons:** EU24/EU24BL - requires phone to use an individual power supply unit (1151B1/B2).
- **Color:** Multi-grey.
- **Power Supply:** IEEE 802.11af Power over Ethernet (PoE) or individual power supply unit (1151B1/B2).

IP Features:

- **Codecs:** G.711, G.729a/b.
- **QoS Options:** UDP Port Selection, DiffServ and 802.1p/b (VLAN)
- **SNMP Support:** Yes.
- **IP Address Assignment:** Static or dynamic IP address assignment.
- **Ethernet Ports:** Two port full-duplex 10/100 BaseT Ethernet switch for PC pass-through connection.
  - Auto-negotiation provided separately for each port.
  - 802.3 Flow Control.
  - Phone has priority over PC port at all times.

6408D Telephone

The 6408D supports the following features:

- Desk/wall-mount.
- Administrable handsfree operation.
- 2 line x 24 character display.
- Speakerphone.
- User administration.
- Time/day default.
- Adjustable display.
- Ringer volume and tone.
- 8 Programmable call appearance/feature keys with twin lamps.
- 8 Fixed feature keys: Speaker, Mute, Conference, Transfer, Redial, Hold, Volume Up & Down.
- Connects to an IP Office DS (Digital Station) port.

6416D Telephone

The 6416 supports the following features:

- Desk/wall-mount.
- Administrable handsfree operation.
- 2 line x 24 character display.
- Speakerphone.
- Expansion module capable.
- User administration.
- Time/day default.
- Adjustable display.
- Ringer volume and tone.
- 16 Programmable call appearance/feature keys with twin lamps.
- 8 Fixed feature keys: Speaker, Mute, Conference, Transfer, Redial, Hold, Volume Up & Down.
- Connects to an IP Office DS (Digital Station) port.

6424D Telephone

The 6424 supports the following features:

- Desk/wall-mount.
- Administrable handsfree operation.
- 2 line x 24 character display.
- Speakerphone.
- Expansion module capable.
- User administration.
- Time/day default.
- Adjustable display.
- Ringer volume and tone.
- 24 Programmable call appearance/feature keys with twin lamps.
- 8 Fixed feature keys: Speaker, Mute, Conference, Transfer, Redial, Hold, Volume Up & Down.
- Connects to an IP Office DS (Digital Station) port.

EU24BL

The EU24BL is an add-on unit that works in association with a 4621/5620 and 4620 phone only. This module will not work in conjunction with any digital phones. The EU24BL provides an additional 24 programmable buttons with associated display label and status icons. Only one EU24BL per phone. The EU24BL provides a backlit display.

Although the EU24BL Expansion Module supports an additional 24 Call Appearance/Feature buttons, it displays only the button labels for one column of 12 buttons at a time. A dotted line separates the left column from the right column. When you view the labels and icons for the left column, the icons for any active or selected right column features display to the right of the dotted line. To view the column not currently displayed, press the Alternate Display button. You can alternately press any Call Appearance/Feature button on the column not currently displayed. Doing so displays that column and selects the line/feature associated with the button you pressed.

Each IP Office DS module, including control units with integrated DS ports, supports a maximum of eight EU24 or eight EU24BL or two XM24 or two 4450 units only. If using an EU24 with only IP telephones the IP Office will support up to 8 EU24 modules per IP Office switch.

- 24 Programmable call appearance/feature keys.
- Automatically labeled from the system (no paper labels).
- Connects directly to the associated phone.
- Requires a 1151B1 or 1151B2 power supply unit for the phone including for IP phones using Power over Ethernet (PoE).
- Use the EU24BL with these Avaya telephones:
    - 4621SW IP Telephone

XM24

The XM24 is an add-on unit that works in association with a 6416 or 6424 display phone and provides an additional 24 programmable call appearance/feature keys with twin lamps. Only one XM24 per phone.

Each IP Office DS module, including control units with integral DS ports, supports a maximum of two EU24, XM24 and/or 4450 units only.

- Connects directly to the associated phone.
- Requires a power supply unit (1151B1) for the phone, power supply socket and must be used with the cables supplied.

Analog Telephones / POTS

As well as providing a lower cost alternative to system specific telephones, analog telephones can still deliver a high degree of functionality. They are particularly appropriate in applications where users are using Computer Telephony (CT) for a high proportion of call control.

Analog telephones that are compatible with caller display functionality can display the telephone number of the calling party if available. Simple programming of IP Office can convert that numeric display in to the company name associated with that number.

Feature activation by analog telephones is via short codes. IP Office is pre-programmed with a default set of short codes but these can be changed to mimic a legacy telephone system as required.

Avaya would like to stress that although most analog phones will work on IP Office - Avaya cannot guarantee that all analog phones in every region of the world will work on the IP Office.

- Analog phones connects to IP Office via POT ports.

Interquartz Gemini Phones

Avaya have tested the new generation Interquartz Gemini analog phones with IP Office to ensure that set and system are compatible. The Interquartz Gemini range offers a whole host of benefits to business users and home-workers including Caller ID, hands-free and headset compatibility and are built to the highest standards, using high-impact plastic casing, double injection moulded buttons and non-slip rubber feet.

The Gemini phones offer good value for money without compromising on quality. Their stylish new design and rugged build quality are sure to make it a popular choice for buyers on a limited budget. For information contact Interquartz at avaya-enquiries@interquartz.co.uk.

Basic telephone 9330-AV

- Visual Message Waiting Indication.
- Locking mute button with LED indicator.
- Last number redial.
- Recall button.
- Ringer volume adjust.
- Ringer indicator light.
- Wall mountable - no additional bracket required.
- Hearing aid compatible.

CLI Feature phone 9335-AV

- All features of 9330-AV.
- Caller ID with 80 memories (shows date, time & new/repeat/answered/unanswered calls) .
- Large 3 line LCD display.
- 100 name and number personal directory.
- 20 lockable direct access memories.
- Full hands-free working.
- Headset port.
- Data port.
- Switchable Time Break Recall 100 / 200 / 300 / 600 ms.
- Call timer.
- Alphanumeric keypad.
- Last number redial with 5 memories.

Hotel Phone 9281-AV

- Removable inlay card for personalized logo printing.
- Triple standard message waiting light (high voltage, reverse polarity and voltage drop).
- 10 non-volatile memories.
- Ringer indicator light.
- Ringer volume and pitch adjustment.
- Last number redial & Recall button.
- Hearing aid compatible.
- Wall mountable – no additional bracket required.
- ELR/TBR switchable.
- MF Only.

T3 Series Phones

T3 Compact

T3 Series phones are available within the EMEA region. They cannot be mixed with 2400, 4400, 5400 and 6400 Series telephone. They can be mixed with 5600 Series and analog telephones. The IP Office only supports T3 Upn phones, it does not support T3 ISND (Ipn) and T3 IP phones.

- Easy to operate – with interactive user prompting.
- Available in Graphite grey or polar white.
- Single line 24 character LCD display.
- 1 Link port for connection to optional T3 DSS or Headset unit.
- 8 Pre-programmed keys with printed text label.
- Volume control.
- Navigation Cursor control.
- Alpha entry Alphanumeric via dialing keypad.
- Call signaling via LED and/or loudspeaker.

T3 Classic

T3 Series phones are available within the EMEA region. They cannot be mixed with 2400, 4400, 5400 and 6400 Series telephone. They can be mixed with 5600 Series and analog telephones. The IP Office only supports T3 Upn phones, it does not support T3 ISND (Ipn) and T3 IP phones.

- Easy to operate – with interactive user prompting.
- Available in Graphite grey or polar white.
- 4 Line x 26 character hinged LCD display.
- 2 Link ports for connection to optional Headset unit and or T3 DSS units.
- 11 Pre-programmed keys with printed text labels.
- Volume control.
- 4 Programmable key with associated display labels.
- Navigation Cursor control.
- Text entry via dialing keypad.
- Call signaling via LED and/or loudspeaker.

T3 Comfort

T3 Series phones are available within the EMEA region. They cannot be mixed with 2400, 4400, 5400 and 6400 Series telephone. They can be mixed with 5600 Series and analog telephones. The IP Office only supports T3 Upn phones, it does not support T3 ISND (Ipn) and T3 IP phones.

- Two colors: Graphite grey and polar white
- Two versions: one with a pre-printed alphanumeric keyboard and one with a blank keyboard with three different self-adhesive templates
- LCD display with 17 lines of 40 characters that can be adjusted for optimum viewing angle
- 2 accessory expansion ports (e.g. Headset Link)
- Support for Direct Station Select (DSS) module
- 5 dedicated feature keys
- 2 dedicated feature keys for volume control
- 12 programmable feature keys, 6 with LED indicators
- 10 programmable Softkeys
- 16 Partner/Line keys
- Navigation Cursor control
- Integrated keyboard
- Handsfree speakerphone

Mobility Solutions

Avaya Mobility Solutions

IP Office supports the following wireless solutions:

- DECT in the EMEA and APAC regions.
- Avaya 3810 in the North American market.
- Avaya 3616 and 3626 VoIP Wi-Fi Solution offered worldwide.

There is little doubt that in the business environment of today, telecommunications are a valuable source of competitive advantage. It is clear that improved internal and external communications leads to increased organizational efficiency, enhanced customer relationships and hence increased profitability. The primary objective of IP Office's in-building Mobility Solutions is to improve communication with staff who, because of the function they perform, are mobile within the workplace.

Using wireless technology such individuals may be instantly contactable, with many obvious benefits;

- The wireless telephone is carried in the pocket, so users are not tied to the desk in order to remain in contact.
- Users may be contacted instantly to ensure fast, accurate decision making and immediate response to problems.
- The risk of endless telephone tag and missed, inaccurate, or old messages is negated.

Almost all organizational activities and staff functions can benefit from wireless communication, but those that will benefit most include:

- **Organizational Activities**
    - Manufacturing and Production.
    - Warehousing.
    - Healthcare.
    - Retail.
    - Hotels and Hospitality.
    - Support Services.
    - Management.

- **Staff Functions**
    - Maintenance Personnel.
    - Production and Warehouse Supervisors.
    - IT Support and Building Services Support.
    - Key Managers.
    - Security.
    - Guest Phones.
    - Sales Teams.

Avaya 3810

Avaya's 3810 wireless handset delivers the benefits and accessibility of a wireless phone with all the power and functionality of a wired desk telephone.

The Avaya 3810 is an in-building wireless system that provides a mobility solution up to a 160 feet range from the Digital Base Module depending on environment. The peak power for this unit is 60mW for transmissions.

Avaya's 3810 solution integrates fully with IP Office. The base station connects directly to the Digital Station port on IP Office, users have the same call-handling flexibility and control that they have with their desk telephones, combined with the mobility of a wireless system.

A built-in headset connection for true "hands-free" mobility and increased productivity is included with all Avaya 3810 handsets

With the system's 2-line display capability, users can set priorities and handle calls more effectively helping to improve productivity and customer service. Caller ID and message waiting notification are readily available, so that employees will be accessible anytime, anywhere.

The Avaya 3810 uses the 902 to 928 MHz ISM (Industrial, Scientific, and Medical) band. Unlike some other in-building wireless systems, there are no airtime charges, and no license is required. This handset uses digital radio technology and spread-spectrum frequency hopping to provide extremely secure wireless communications.

The Avaya 3810 Wireless Telephone is a digital telephone designed to work with IP Office (minimum release 2.0). It offers the mobility inherent in a wireless telephone plus access to a number of features and functionality of the connected communications system.

The Avaya 3810 wireless telephone uses 900 MHz digital technology allowing a maximum range of 160 feet from the base station.

A maximum of 5 Avaya 3810 wireless handsets can be connected to the same PBX, Site Planning rules do apply, please refer to installation guide available from the following web site: http://www.avaya.com/support and then select

- Product Documentation
- Telephone Devices and User Agents

Full documentation is also contained within the package.

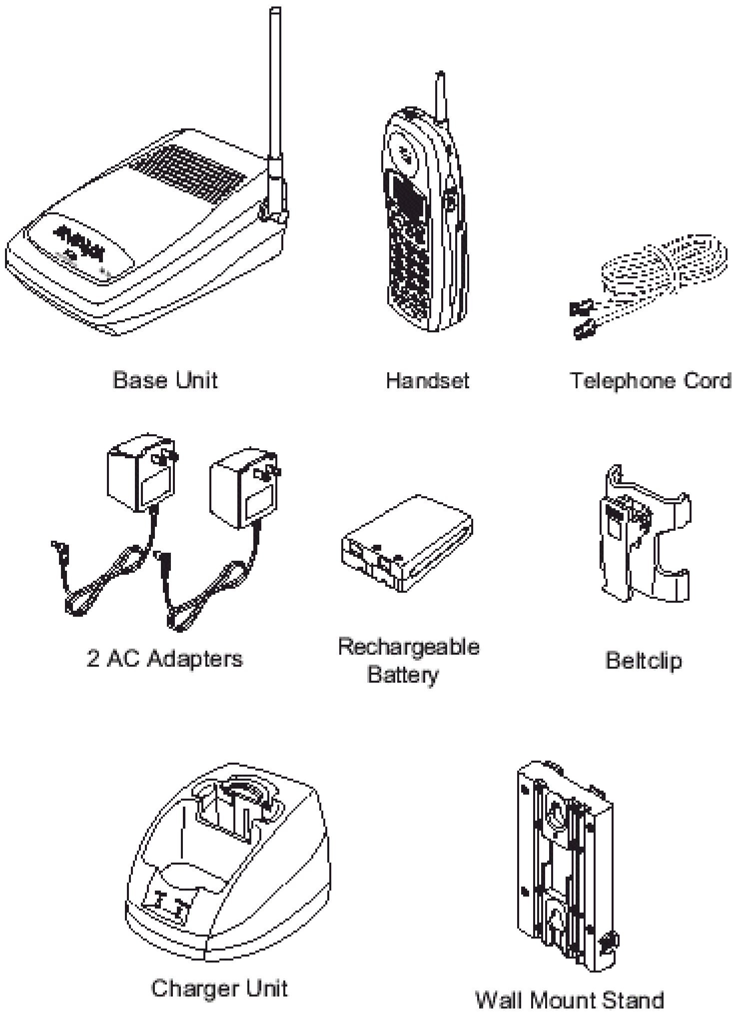
The 3810 provides the following features:

- 2-line, 32 character Handset Liquid Crystal Display (LCD).
- 4 displayed operation modes indicating Talk, Ringer On/Off, Battery Low, and Message Waiting.
- Single button access to fixed features – Hold, Transfer, Conference, and Redial.
- 4 programmable buttons to access features on the PBX.
- 10 channels, supporting up to 10 simultaneous conversations.
- Headset jack.
- Ringer and Handset volume control.
- User selectable ring type.
- Vibrate alert.
- Base Unit and Charger Unit.

Hardware

The Avaya 3810 is ordered as a single unit containing:

- Base Unit.
- Handset.
- Telephone Cord.
- Base Unit Power Supply Adapter.
- Charging Stand Power Supply Adapter.
- Rechargeable Battery.
- Belt Clip.
- Charging Stand.
- User & Installation Guide.
- Wall Plate Adapter.



- Voice Quality - The 3810 handset:
    - Provides full-duplex voice transmissions, using ADPCM (Adaptive Differential Pulse Code Modulation) to provide the digital encoding
    - Applies a highly sophisticated "companding" feature to transmissions, which helps cancel out background noise (also known as "white noise")
- 20 Number Memory - The 3810 can store up to 20 phone numbers for quick and easy speed dialling
- Vibrator Alert - The 3810 handset provides a vibrator alert feature as an alternative, or in addition to, an audible ringing tone.
- Redial Button - A fixed Redial button is provided to facilitate repeated calling attempts.

Avaya 3810 Voice Telephone Attributes

Each 3810 includes one standard battery and a charging cradle and power supply that connect to standard AC power. The charging cradle will charge the handset battery and an optional spare battery pack that may reside in the cradle's spare battery garage.

Battery Charging

Each Avaya 3810 handset comes with its own charging cradle. The charging cradle will charge both a battery in a handset and an optional spare battery (if purchased) in the charger's spare battery compartment. Because the charger is upright, the phone display is clearly visible when sitting at your desk; so incoming calls can be visually screened.

With the fast-charging battery capability built into both the handset cradle charger and the spare charger, batteries charge fully in only 6 hours. With each charge:

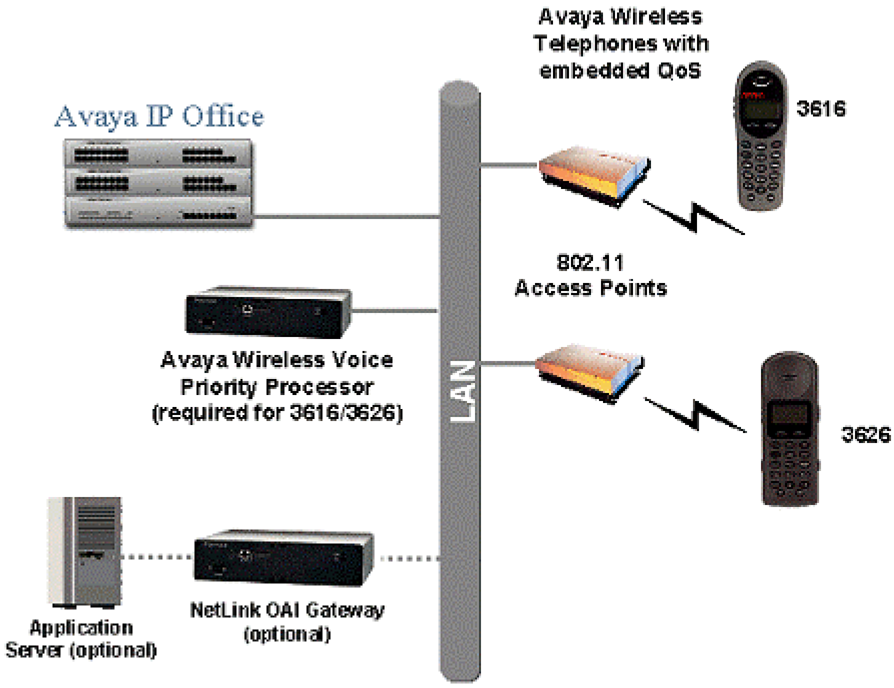
- Batteries are discharged and recharged, which eliminates the memory effect that reduces battery life (the spare is automatically reconditioned; the handset battery is reconditioned in the cradle if manually selected).
- Users get 10 hours of talk time, and 4 days of standby time.
- Users can continue to screen calls because the upright position keeps the display clearly visible.
- In addition, an optional extended-use battery provides 12 hours of talk time and 8 days of standby time. The unit supports a hot swap option – in the middle of a call a user can take out an old battery and put in the new charged battery as long as the swap is accomplished in 20 seconds or less.

Overview of Wireless VoIP

The Avaya IP Wireless solution offers an advanced Voice over IP (VoIP) client for wireless networks. This solution allows SME's to take advantage of the cost savings and simplified management of a converged voice and data infrastructure.

Both the Avaya 3616 and 3626 phones are optimized for Avaya IP telephony and emulate the wired 4606 IP Telephone. They work in conjunction with the Avaya Voice Priority Processor to ensure voice quality over Wireless LANs. They are field upgradeable through an integrated TFTP client, so telephones can be updated with new protocols, features, and capabilities as they become available.

Based on global standards for wireless LAN's, the Avaya IP Wireless Telephone Solution simplifies network infrastructure by enabling voice traffic to be carried along with data traffic over the same wireless network. Both the 3616 and 3626 telephones are available for direct sequence 802.11b Wi-Fi networks. The SpectraLink Voice Priority (SVP) quality of service protocol is simple to implement and reduces packet queuing delays for voice traffic. SpectraLink Voice Priority-enabled access points are available from the leading providers of enterprise wireless networks such as Avaya.



Users can have a choice of an executive or rugged workplace telephone and all the productivity benefits of their desk telephone in this next generation of wireless telephone solutions.

SpectraLink Voice Priority (SVP) Compliance Matrix

When using this solution, certain wireless access points must be used to ensure QoS for the voice conversation.

Manufacturer	Make / Model	SVP Certified	Field Verification	Calls per Access Point
Cisco	Aironet 340	10.13	11.03, 11.07, 1.10t	6
Cisco	Aironet 350	11.03	11.07, 11.01t	7
Cisco	Aironet 4500 & 4800 Turbo DS	8.12 & 8.24	8.55	5
Proxim	Orinoco AP1000	D3.78S6 3.83	7.4a	5
Telxon	802 DS & 802 DS 11	8.12	8.24	5
Avaya	AP-1, AP-2		3.83, 3.92	6
Avaya	AP-3		1.4 (v222)	7
Cisco	Aironet 1200		11.40t	7
Enterasys	Roamabout AP2000		V6.02	6
Intermec	Mobile LAN Access 2100, 2101, 2102		1.51 or later	6
LXE	6250 Access Point		3.83	6
Proxim	AP 2000		7.4, 1.3	6
Symbol	Spectrum 24 DS		2.21-23, 2.51-21, 3.50-18	6
Teklogix	9150 Wireless Gateway		E301R, J041	4

1. Alvarion BreezeNET Pro 11 Series software version 4.4.5 is not compatible with Avaya wireless telephones.
2. Cisco Aironet 350 software version 11.21T is not compatible with Avaya wireless telephones.
3. Direct Sequence (DS) Avaya wireless telephones support up to 11Mbps data rates.

Benefits

- Supports the 802.11b standard for Wi-Fi networks converging voice and data over a single network.
- A lightweight, executive 3616 handset with a form factor similar to a cell phone.
- Seamless integration with IP Office.
- Excellent voice quality on converged wireless networks.
- Lightweight, durable handset specifically designed for workplace use.
- Improved display, battery life, processor power all with lower costs.
- Multitude of accessories are available:
    - Dual Charger (full charge accomplished in approximately one and a half hours).
    - Quick Charger (full charge accomplished in approximately one and a half hours).
    - Belt Clip.
    - Nylon Pouch.
    - Carrying case with Lanyard.
    - Handsfree Pouch.
    - Noise canceling headset.
    - Over the ear headset.

SpectraLink Voice Priority (SVP)

To enhance voice quality over the wireless network, SpectraLink has developed a Quality of Service (QoS) mechanism that is implemented in the wireless telephone and access point.

Avaya IP Wireless Telephony Solution (AWTS) Open Application Interface (OAI) Gateway

The AWTS Open Application Interface (OAI) Gateway enables third- party software applications to communicate with the Avaya IP Wireless Telephones. This serves as a two-way messaging device. Many companies provide applications that interface to your in-house paging systems, email, and client-server messaging. Other vendors with complementary systems such as nurse call, telemetry, alarm, and control system manufacturers are currently developing applications to interface with the Avaya IP Wireless Telephone solution.

DECT

IP Office DECT

Based on the Digital Enhanced Wireless Telecommunications (DECT) standard, IP Office's DECT wireless systems support the Generic Access Profile (GAP) standard designed to allow interoperability of handsets supplied by various suppliers. Delivering on site mobility for staff on the move, DECT is a digital solution designed to integrate with IP400 Office to provide roaming extensions on both IP Office and alternative vendors' PBXs.

There are three DECT options on IP Office:

- **Compact DECT Control Unit**
  Connects to the IP Office via analog extension ports with a maximum of 8 DECT telephones.
- **DECT Control Unit (DCU)**
  Connects to the IP Office via analog extension ports with a maximum of 128 DECT telephones.
- **Avaya IP DECT**
  A solution which consists of multiple DECT bases stations connected to the IP Office via IP trunks. Note that Avaya IP DECT equipment is not compatible with the Avaya DECT equipment above, with the exception of the 20DT telephone which does provide basic functionality when used with Avaya IP DECT system.

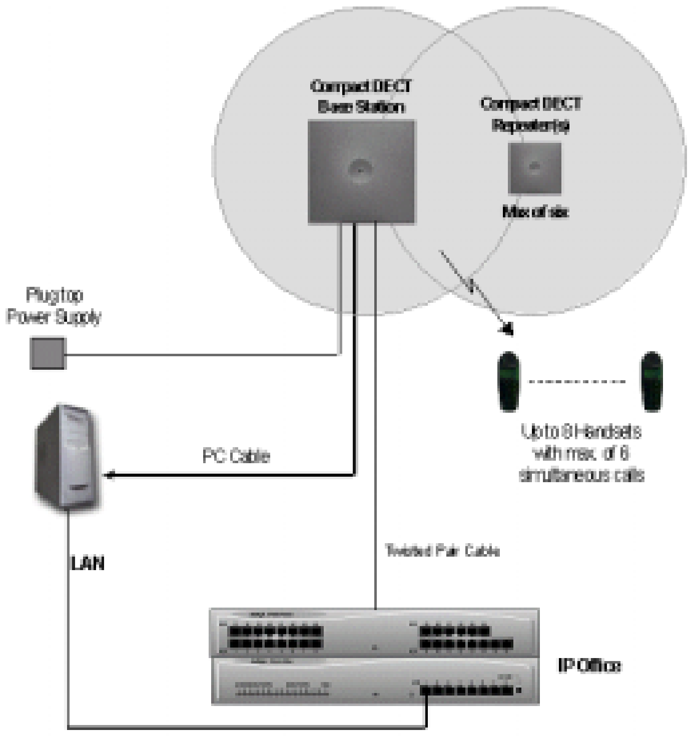
Compact DECT Control Unit

The Compact DECT Control Unit (CU) is a wall mounted central equipment unit that both incorporates a base station and provides the control functions and interfaces to the IP Office system (or alternative PABX). The Compact DECT CU solution supports a maximum of 8 wireless handsets and 7 DECT Base Stations (DBS) [1 Main base station plus 6 repeaters]. The Compact DECT CU is connected to the IP Office control cabinet by 2 wire analog extension ports and to a PC via a V24 control link enabling enhanced feature integration. The V24 control link enables the IP Office system to offer sophisticated features on the DECT handsets thanks to the intelligent LAN connection.

When connected to IP Office, the Compact DECT CU offers unique integrated features and continues to provide many of the functions associated with fixed IP Office digital phones without confining users to their desks.

The Compact DECT CU can be deployed up to 300m from the IP Office system providing coverage of up to 600 meters, depending on building construction and local environment. The average radius coverage within buildings is approximately 50m to 60m. The installation of the Compact DECT CU is very straightforward and simply requires a connection to local power and the associated IP Office.

In an area with a requirement for high wireless densities the Compact DECT CU should be located centrally with Repeater Base Stations being used to extend the coverage area over the site.



Compact DECT integration to IP Office¶

Extending Compact DECT Coverage

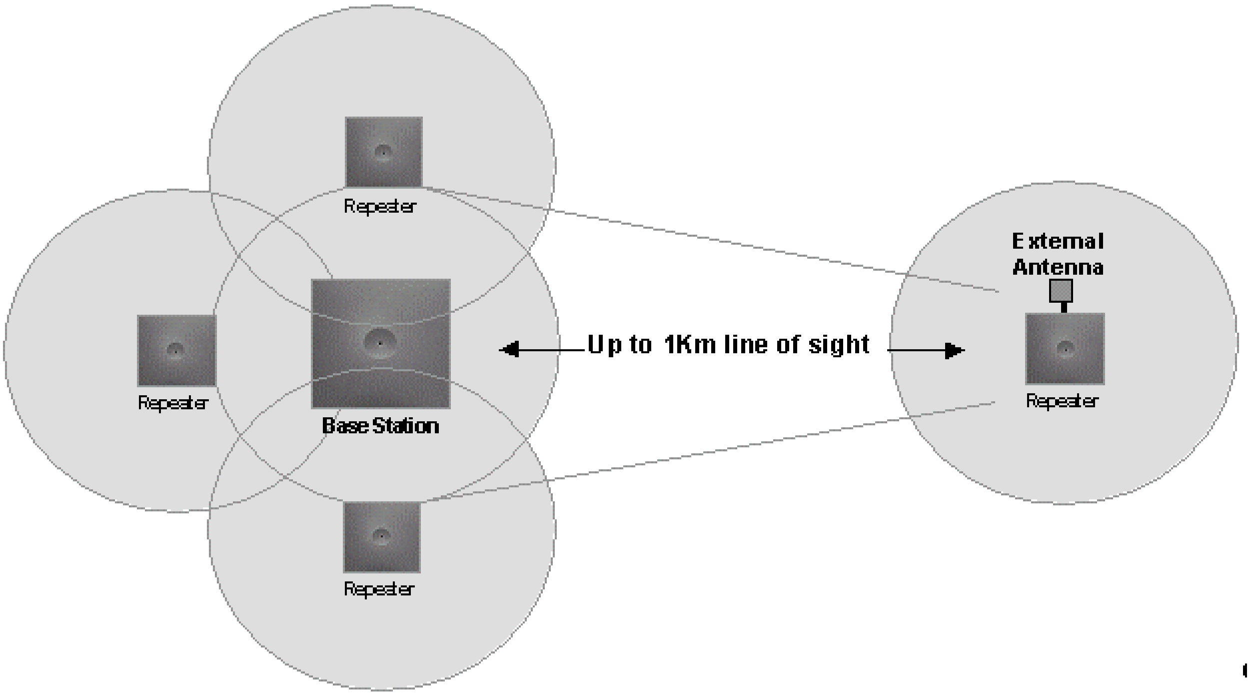
The coverage offered by the Compact DECT CU can be supplemented by up to 6 DECT Repeater Base Stations.

The Repeater Base Station is able to offer an extension to the area serviced by IP Office's DECT system in a simple and cost effective manner without the need to lay more cables.

Both Compact DECT CU and Repeater Base Station designs are very compact and may be installed out of sight within false ceilings. Local power is required for the Repeater Base Stations.

As with the standard DECT Base Station, the Repeater Base Station offers seamless handover and roaming enabling users to move freely between cells during calls over an extended area.

Each Repeater Base Station has a capacity of two simultaneous calls while the main Central Compact Base Station supports 6 simultaneous calls.



Compact DECT control Unit and Repeater Base Stations

DECT Control Unit

Analog connection DECT for IP Office. The Compact DECT solution provides smaller businesses with a highly functional entry-level wireless solution. However, there are many organizations with requirements for larger and more scalable solutions.

The DECT Control Unit (DCU) is a wall mounted central equipment unit providing interfaces for a maximum of 128 handsets and 32 base stations meeting the requirements of larger customers. This unit is being phased out in favor of the Avaya IP DECT solution presented in the next section. The DCU is mains powered and is always supplied pre-configured with a power supply unit and intelligent motherboard. The DCU motherboard contains the central processor function and provides interfaces for the connection of 8 DECT Base Stations (DBS) as standard but can be extended, see below for details.

The installation of a maximum of 8 DECT Analogue Boards (DAB's) and a single Expansion Board (DEB), supporting a further 8 DECT Base Stations (DBS's) is provided by the DCU motherboard. To support 128 wireless handsets and 32 base stations, two DCU's are linked using two Link Cards.

Each wireless handset that is to be registered for use on the system requires a two wire analogue connection through the host communications platform.

- **DECT Analogue Board (DAB)**
  The DAB boards contain 8 circuits for connection to two wire ports on the host communications platform. These support MF or pulse dialing and Time Break Recall (hook flash)
- **DECT Expansion Board (DEB)**
  The DEB is an optional board fitted within the DCU providing support for a further 8 DECT Base Stations (DBS) per cabinet, increasing the maximum number of supportable base stations to 16.
- **DECT Base Stations (DBS)**
  The radio base stations can be up to 2km's cable distance from the DCU and have coverage of up to 600 metres, depending on building construction and local environment. The average radius coverage within buildings is approximately 50m to 60m.
  - Single 2 wire connections are required to each Base Station (using standard telephone cabling or CAT5) making the installation very straightforward. Connection to each base station is from a standard secondary socket.
  - As the base station takes power from the DCU, there is no need to provide local power to base stations, again simplifying installation.
  - Each Base Station has a capacity of 4 simultaneous calls and, in areas of high traffic concentrations, such as restaurants and small offices, multiple base stations may be deployed to a maximum of 3, with further bank of 3 situated a minimum of 20 metres away.
  - The Base Station design is very compact and discrete (dimensions Length x Height x Depth = 100x100x36 mm) and maybe installed out of sight within false ceilings.
  - Seamless handover and roaming is supported by all Base Stations allowing users to move freely between cells during calls, based on effective deployment.

Avaya IP DECT

This solution is only available in EMEA and APAC regions. The IP DECT solution provides businesses with a highly functional wireless solution with the ability to scale to support large numbers of users. This system also supports users in different offices connected via a WAN. The Avaya IP DECT solution radio fixed part (RFP) or base station connects to the IP Office using an IP protocol based on H.323.

The Avaya IP DECT solution supports up to 120 handsets and 32 base stations. Each base station can be powered over the LAN using the Power over Ethernet (PoE) standard. Each indoor base station can also optionally be connected to main power via an external power adaptor. Each outdoor base station can only be powered using PoE - no individual power supplies are available to power the outdoor IP DECT base station.

This system supports the 3701 and 3711 handsets. The 20DT handset is also supported, but with minimal functionality – a special version of Message Waiting Indication is supported, but other features are not. For example: Access to the system directory will not work on the 20DT handset when used with this new solution. Avaya recommends that for new deployments, for full feature functionality the 3711 handset be used with the IP DECT solution.

Each Base station has the following features:

- Frequency range: 1,880 – 1,900 GHz.
- 10 carrier frequencies (1,728 MHz spacing) with 12 time slots each.
- Maximum transmission power of 10 mW.
- 8 simultaneous Voice and up to 12 Signaling Channels.
- Codec G.711, G.723, G.729 for base station IP trunk connection.
- Handover
  While in motion, the handset performs continuous measurements to determine which IP DECT base station has the strongest signal. The one that can be best received is defined as the active Base station. To prevent the handset from rapidly switching back and forth between two base stations that are equally well received, threshold values are used. Handover between base stations occurs seamlessly whether a call is active or not.
- DECT Networking
  An IP DECT telephone can travel from one office to another which is connected over a wide area network (WAN) link and make and take calls. In this scenario the main IP DECT controller remains at one "headquarters" location.

Given the degree of integration available to wireless users with DECT, there are a variety of means by which calls can be routed to wireless handsets:

- **DDI/DID**
  Since each wireless handset is an extension on the IP Office system calls may be routed directly using a DDI/DID number.
- **Transfer**
  Calls may be transferred to DECT extensions by operators or other extension users and DECT extension users may transfer callers to any other extension user.
- **Hunt Group compatibility**
  Wireless handsets may be programmed as members of groups and answer calls in the same manner as any other extension within that group.
- **Group working**
  Wireless handsets may be programmed as members of groups and attract calls in the same manner as any other extension within that group. DECT handsets must NOT be configured into collective groups.
- **Divert destination**
  Users may initiate any or all diverts from an Avaya desk phone to a wireless handset.
- **Twinning**
  Added in IP Office 3.1, twinning allows calls to a user main extension number to alert at both that extension and a secondary extension. Though not restricted to DECT, this feature is aimed primarily at users who have both a desk phone and a wireless extension. Calls from the secondary twinned extension are presented as if from the users main extension. Presentation of call waiting and busy is based on whether either of the twinned extensions is in use.

DECT Comparison

Feature	Compact DECT	DECT Control Unit	IP DECT
Maximum handsets	8	128	120
Maximum base-stations	1	32	32
Total base-stations/repeaters	7	64	32
Maximum simultaneous calls	6	128	100*

*May be limited by the available VCM voice compression channels for calls to non-IP destinations.

DECT Licenses

Compact DECT and DECT DCU Systems

For these systems, the following additional features are available through the use of a CTI DECT licence and Avaya IP Office DECT Integration software running on a PC. The license is entered into the IP Office configuration

- Available with 20DT sets only.
- Desktop and Mobile Handset Twinning - The desktop and DECT handset can be synchronized to logically act as a single unit. Calls presented to the desktop phone will simultaneously be presented to the DECT handset. When either device is busy any further calls presented will receive busy tone or be rerouted to the relevant divert on busy destination which may be Voice Mail if configured. The integration of the devices extends beyond status information to incorporate more detailed feature integration including the simultaneous presentation of voice mail indication.
- Other advanced features - Wireless users require many of the standard, as well as advanced, functions available to users of Desktop handsets. All telephone users, including mobile, have access to the system codes on IP Office and are therefore able to benefit accordingly. However, IP Office offers a number of enhanced features in conjunction with both wireless options detailed above. The variety of features addresses the needs of even the most sophisticated user. These enhanced functions include:
- CLI/ANI Presentation or associated name
- Voicemail Message Waiting Indication
- Intuitive Voice Mail Access
- Call Waiting Indication
- Presentation of Calling/Called Party Identity
- Access to both Internal & External Directories for simplified dialing
- Parallel ringing, vibration support and user definable ring cadence with a fixed phone (twinning).

Avaya IP DECT System

For this system, a license is necessary for even basic functionality. This license is called the Avaya IP Office IP DECT Mobility Manager license. This license is entered through the main base station (ADMM) and is NOT entered through the IP Office System manager. A feature key server is NOT necessary to enable the IP DECT functionality.

No separate PC or software is required with this system.

The Avaya IP DECT system is sold in bundles that are supplied complete with the licenses for new installs. For new installations the bundles will be the only license related items that should be purchased, along with the requisite numbers of telephones and base stations.

Additional upgrade licenses are available for systems that need to expand their current coverage or capacity.

5. Features

Telephony Functions & Call Handling

IP Office provides a comprehensive telephony feature set to enable a fast, courteous and efficient response to a telephone call .

Features such as CallerID display and call tagging allow employees to see who is calling and why before they pick the handset up. Client information can even be 'popped-up' on the user's PC.

For those who are not tied to a desk, Wireless handsets offer mobility around the office. For those out of the office, be it on the road or working from home, comprehensive and easy to use call forwarding facilities, PC Softphone and remote access server software allow them to remain in telephone contact and access centralized resources at all times.

Incoming calls can be efficiently handled using either Direct Dialling (DDI/DID) or dedicated operators. For out of hours calls or times when you just can't take calls, IP Office provides voicemail and optional Auto-Attendant services.

Basic Call Handling

Basic Call Handling

Description

The most fundamental set of features for a telephone system cover:

- Answering a call.
- Making a call.
- Putting a call on hold.
- Answering a second call.
- Transferring a call.
- Conferencing multiple parties.

CallerID

Feature

- Display of the caller’s number on incoming calls.
- Sending of number for return calls on outgoing external calls.

Benefit

- Confirmation and recognition of who is calling.
- Storage of CallerID numbers for return calls.
- Directory name matching to CallerID numbers.
- Screen-Popping customer records in compatible applications.

Description

Where supplied by the local exchange, the IP Office can receive and use the callers CallerID. The CallerID is passed through to the answering phone or application and is included in any call log or history supported by the phone or application. If the CallerID matches a number in the IP Office's Directory, the matching directory name is shown.

Where Phone Manager Pro, or the IP Office TAPI service provider, and certain applications are in use it is possible to have an automatic query performed on the supplied CallerID to have the caller’s record in front of the user before the call is answered.

For outgoing calls the IP Office can insert a custom CallerID or set a flag to have Caller ID withheld. For users with a DDI number routed to their extension, that DDI is also used as their CallerID for outgoing calls. Alternatively short codes can be used to specify the CallerID that should be sent with outgoing calls.

Note that the sending and receiving of CallerID is subject to the local exchange supporting that service. The line provider may also restrict which numbers can be used for outgoing CallerID.

Hold

A call may be placed on hold with optional Hold music. A held call cannot be forgotten as it is presented back to the extension after a timeout set by the systems administrator.

See also **Call Park**.

Toggle Calls

Toggle Calls cycles round each call that the user has On Hold locally within the system.

Hold Call Waiting

Hold Call Waiting is a compound feature combining hold and answer and provides a convenient way to hold an existing call and answer a waiting call through a single button press.

Hold Music (Music on Hold)

The system supports either internal or external music on hold. The internal source uses a WAV file of up to 30 seconds length. WAV files are a widely used PC file format making it simple to change the music to meet the customers needs. External music on hold sources connect to the 3.5mm Audio socket on all IP Office Base Modules.

Park

As an alternative to placing a call on hold a call can be parked on the system to be picked by another user.

The call park facility is available through the user's telephone, Phone Manager, Windows Operator Console and Busy Lamp Field applications. Calls are Parked against a ‘slot number’ which can for example be announced over a paging system so the person the call is for can go to any phone and access the call using the park slot number.

For convenience Phone Manager has 4 pre-defined park buttons. On digital phones with DSS/BLF keys it is possible to program Park keys that will indicate when there is a call in a particular park slot and allow calls to be parked or retrieved.

There is a system configurable timeout that determines how long a call may remain parked before it is re-presented to the extension that originally parked the call.

Automatic Callback

Feature

- When calling an extension that is busy, set the system to call you when the extension becomes free. This feature is also called "Ringback When Free".
- When calling an extension that just rings, set the system to call you when the extension is next used. This feature is also called "Ringback When Next Used".

Benefit

- Carry on with other work and let the system initiate a call for you when the extension becomes available.

Description

Depending on the type of phone a user has, the feature is accessed by dialing a digit while listening to internal busy tone, selecting an option from an interactive menu or a programmed DSS/BLF key. Call back when free can also be activated from Phone Manager.

You can also set a ringback when free or a ringback when next used using a short code. This is useful as it doesn't require a call attempt.

Note that a user can only have one automatic callback set at any time.

Direct Inward Dialing (DID)

This relies on all or part of the incoming dialed number being supplied to the IP Office from the local exchange, this can then be used to route the call to individual phones or groups of phones. Typically used to reduce the load on a reception position by giving members of staff or departments individual numbers so they can be called directly. For convenience it is common to have the extension or group number the same as the digits supplied from the network but IP Office can convert the number by, for example, taking three digits from the network and adding a leading ‘4’, or by constructing a table that maps received digits to the internal number.

Transfer

Call Transfer allows another party to be placed on hold and transferred to another destination number.

If the phone is put down before the destination has answered, the original caller will be automatically transferred. This is called an Unsupervised or Blind Transfer.

A transferee can alternatively wait for the destination to be answered before hanging up to complete the transfer. This is called a Supervised Transfer.

Unless restricted by the system administrator, the IP Office makes no differentiation between internal or external calls (other than ringing sequence) so it's just as easy to transfer a call to a mobile/cellular telephone as to extension 201.

Advanced Call Handling

Advanced Call Handling

Description

In larger businesses or businesses with greater reliance on the telephone for internal and external communications some of the more advanced features will improve efficiency and customer service. Features like Pick-Up which permit users to take a call for a colleague who is temporarily away from their desk, of Absence Text which can quickly give information to internal callers about a persons availability.

Absence Text

Feature

- Display a text message on the user's phone and IP Office Phone Manager application.
- Display the same message on other internal phones and IP Office applications when calling the user.

Benefit

- Inform other internal users of your current status and likely availability.

Description

Any user can set Absence Text on their phone, even users of standard analog phones, but it can only be displayed on certain phones or Phone Manager and Soft Console. Most supported feature phones give the option of adding some text, for example message number 7 is "With visitors until" and on these phones the user can enter the time, or even a completely custom message.

When a user has an absence message set, call processing is not affected to the user has the choice of also using features like Do Not Disturb or Forward on No Answer as appropriate. Phones that support the interactive setting of Absence Text will also display it on the users own phone for the benefit of people who come to their desk.

Call Tagging

Feature

- Display a text message on the user's phone, or Phone Manager application, when a call is presented to it.

Benefit

- Provide additional information about the call.

Description

This feature is used to provide additional information about the call to the targeted user, before they answer it. This feature may be used when transferring a call from Phone Manager or Soft Console to giver caller info if the user doing the transfer is not able to announce the call. It is also possible to add a tag to a call using CTI and Voice Mail pro, see later section. On some phones, displaying the Tag may mean that it is not possible to display the usual call source and target information.

Reclaim Call

Feature

- The ability to recover, or reclaim, the last call that was at your phone but is now ringing or is connected elsewhere.

Benefit

- Saves the time of letting the caller leave a message, you dial in a listen to it, and try to contact them back.

Description

This is a special version of the Acquire Call feature, that only applies to the last call at your extension, very useful when you get to your phone just as the call goes to Voice Mail or if you have transferred a call but want to get it back.

Hunt Group Enable/Disable

Feature

- The ability for a user to temporarily suspend their membership of Hunt Groups.

Benefit

- A user may need to temporarily join or leave individual hunt groups, for example to cover a peak of calls without changing the system programming.

Description

A team supervisor or administrator may not usually take calls for a team but at times of high traffic they may join the group to take calls and when the peak is over leave the group to resume their regular tasks. To use this feature the User must be configured as a member of the Hunt Group by the systems administrator, it is not possible for a User to arbitrarily join a Hunt Group that they have not been administered as a member.

Call Waiting

A User may not want people calling them to receive busy tone if they are already on another call, but have the call receive ring tone and have some kind of alert that there is a call waiting. The user can then decide to finish or hold the current call and answer the one that is waiting. The amount of information that is available about the call that is waiting depends on the type of phone the user has and if they are using Phone Manager.

As Call waiting tone can be disruptive it is possible to turn the feature on or off and even suspend it for a single call – useful for conference calls.

Do Not Disturb (DND)

This is the ability to temporarily stop incoming calls to a user's telephone. It will prevent the user from receiving Hunt Group calls and give direct callers either Busy or Voicemail if available. This feature can be enabled/disabled from the phone or via the Phone Manager application.

It is possible to have some numbers that can bypass the DND setting, the numbers can be internal or external. For example a manager might have their PA's extension number and their Directors extension and mobile/cell number on the DND Exceptions list.

Flexible Dial Plan

IP Office has a very flexible numbering scheme for extensions, hunt groups and feature commands. While the system has default numbering for feature codes and extensions they can all be re-defined, in default extensions and hunt groups have 3 digit numbers starting at 200 but these can be freely changed to up to 9 digits. There is also a default set feature access codes but these can also be changed so if IP Office is replacing a system where DND is accessed by dialling *21 it is possible to change the IP Office Short Codes from the *08 and *09 Defaults for DND on/off.

Where required, IP Office can support Secondary Dial Tone, though this limits some functionality like LCR. IP Office can also be configured to work without line access digits, by analysing digits as they are dialled and determining if they are for an internal number or should be sent out on a line – this is valuable in SOHO installations where users will not necessarily be used to dialling '9' for a line.

Paging

Almost all of the feature phones supported on the IP Office have loud speakers, these can be used to broadcast messages to an office without having to install a separate PC or paging system. Since Release 2.1 IP Office supports paging to IP Hardphones as well as digital extensions. Paging can be to individual phones or groups of phones.

Analogue extension ports can be configured for connection to external overhead paging systems, usually through an adapter, such a port can be included in a paging group to permit mixed phone and overhead paging.

Some feature phones are able to answer a page by pressing a key while the page is going on, this terminates the page and turns it into a normal call.

Intrude
The Call Intrude feature allows a user, if permission is given, to join an existing conversation whether this is an internal or external call.

A user with the "Can Intrude" option can join a call on any extension on the system however, a User with "Cannot be Intruded" setting would prevent others from joining their call.

Inclusion
This capability enables selected parties to intrude on calls that are already in progress. The intruding party intrudes on the existing call and all parties hear a tone. The speech path is enabled between the intruding party and the called extension, the other party is forced onto hold and will not hear the conversation. On completion of the intrusion the called party speech path is reconnected to the original connected party. The feature is enabled or disabled on a per user basis through the Manager application.

Hot Desking
Hot Desking allows a number of users to use the same extension. Each user logs in as themselves so they can access their own Voicemail and other facilities. For example, sales personnel who visit the office infrequently can be provided with telephony and Voicemail services without being permanently assigned a physical extension.

Relay On/Off/Pulse
IP Office is fitted with two independent switch outputs for controlling external equipment such as door entry systems. Control of these switches is via allotted handsets allowing the switches to be opened, closed or pulsed as required. Control of switches is also accessible via Phone Manager Pro, SoftConsole and Voicemail Pro.

Pickup
Call Pickup allows a user to answer a call presented to another extension. Types of call pickup include:

- Pick up any call ringing on another extension.
- Pick up a Hunt Group call ringing on another extension. The user must be a member of that Hunt Group.
- Pick up a ringing call at a specified Extension.
- Pick up any call ringing on another extension that is a member of the Hunt group specified.

Call Recording
Where IP Office is installed with IP Office Voice Mail it is possible to record a call to a user's mailbox, this is useful when a caller is going to give detailed information like an address or phone number.

Where call recording is required for Quality Assurance it is possible to set the IP Office to automatically record a percentage of calls for later review.

Twinning
Twinning allows two local extensions to operate together as a single telephone, typically used in scenarios like workshops or warehouses where team supervisors may have a desk with a fixed phone but also have a Mobile or wireless extension. When a call is presented to the primary phone the secondary will also ring, when a call is made from the secondary phone it will appear to have come from the primary phone. Other users of the system need not know that the supervisor has two different phones.

Twinning is currently not supported in the US.

Key and Lamp Operation

Key and Lamp Operation

IP Office offers a full range of Key and Lamp features on Avaya feature phones. These features include; Line Appearance, Call Appearance, Bridged Appearance and Call Coverage. As the features require a phone with buttons and LED's or LCD displays the features are only supported on certain phones in the vast range of phone types supported on the IP Office.

Appearance Buttons

Feature

- Use the programmable buttons available on Avaya phones to represent individual calls.
- Answer, make and join calls by pressing the appropriate appearance buttons.

Benefits

- Indication of calls connected and calls waiting.
- Handling of multiple calls from a single phone.

Description

Many Avaya phones supported by IP Office have programmable buttons. These buttons can be assigned to appearance functions that allow the handling of calls. These functions are:

- **Line Appearance Buttons**
  Used to indicate make and answer calls on a specific external trunk.
- **Call Appearance Buttons**
  Used to handle multiple incoming and outgoing calls from a user's extension.
- **Bridged Appearance Buttons**
  Used to match the call appearance buttons on a colleagues extension.
- **Call Coverage Buttons**
  Used to indicate unanswered calls ringing at a colleagues extension.

Line Appearance

A Line Appearance is a representation of a line on the system, the indicator tracks the activity on the Line. Only external calls can be answered or made on Line Appearances. All types of PSTN trunks - Analog, Primary Rate and Basic Rate can be assigned to Line Appearances. IP trunks can NOT function with Line Appearances.

Call Appearance Buttons

Feature

- Uses a programmable button on the phone to represent an incoming or outgoing call.
- Separate buttons are used to represent each simultaneous call that the user can make or answer.
- Where possible, the status of the calls (ringing, connected, held) is indicated by the button lamp or display.

Benefit

- Call appearances allow a single user to make, answer and switch between multiple calls by pressing the appropriate call appearance button for each call.

Description

On Avaya IP Office phones that have programmable buttons, those buttons can be set as call appearance buttons. The number of call appearance buttons set for a user determines the number of simultaneous calls they can make and answer.

The use of call appearance buttons overrides the IP Office's call waiting features. These will only apply should the user log on at a phone that doesn't support call appearance buttons.

The use of call appearance buttons also changes when an extension user is seen as busy and returns busy tone to further calls. For calls routed to the users extension number, they will only return busy tone when all their call appearance buttons are in use.

When call appearance buttons are used, a minimum of three call appearance buttons is recommended where possible (some phones are restricted to two call appearance buttons by the number or design of their programmable buttons).

Bridged Appearance Buttons

Feature

- Allow the user to have an appearance button that matches another user's call appearance button.

Benefit

- Answer and make calls on behalf of the other user.
- Visual indication of when the other user has calls ringing, held and connected.
- Join and exchange calls using the paired call appearance and bridged appearance buttons.

Description

A bridged appearance button matches the activity of one of another user's call appearance button. For example, when the call appearance shows a ringing call, the bridged appearance button will also show the ringing call and can be used to answer that call.

Similarly, if the bridged appearance button is used to make a call, the call activity is shown on the matching call appearance button. The call appearance button user can join or takeover the call using their call appearance button.

Thus bridged appearance buttons allow paired 'manager/secretary' style operation between two users.

Bridged appearance button are only supported for users who also have call appearance buttons.

Call Coverage

Feature

- Allow unanswered calls to alert at other user extensions and be answered there before being forwarded or going to voicemail.

Benefit

- Provide users the opportunity to answer colleague's unanswered calls before they go to voice mail.

Description

When a user has an unanswered call ringing, after a configurable delay, the call will also start alerting on any call coverage buttons associated with the user on other extensions. The call can then be answered by pressing the call coverage button. If still unanswered the call is forward or goes to voicemail as normal.

The time a call rings before also alerting on any associated call coverage buttons can be adjusted for each user.

Outbound Call Handling

Outbound Call Handling Features

Description

Every business needs to make calls, but depending on the type of business these calls may need to be recorded against a project or client, Account Codes provides a solution for this. Alternatively a business may have several sites linked via a private network but certain users, like customer services agents, may need to be able to call colleagues in other offices even when the network is busy, while other users can wait for a line to come free, Least Cost Routes can automatically translate the internal number to a DDI/DID call over the public network while other users wait.

Account Codes

Feature

- Associate an account code with a call.
- Validate account codes used against list stored by the IP Office.
- Include the account code used with call log details.

Benefit

- Through the call records, group calls by account code for the purpose of call costing and tracking.
- Restrict outgoing calls by requiring users to enter a valid account code.

Description

The IP Office stores a list of valid account code numbers. When making a call or during the call, the user can enter the account code they want associated with that call. The IP Office will check the account code against its list of valid codes and request the user to re-enter the code if it is not valid.

Individual users can be set to Forced Account Code operation. They are then required to enter a valid account code before making external calls.

With IP Office Short Codes it is possible to flag certain numbers or call types as requiring a valid account code before permitting the call to proceed, for example long distance or international numbers.

For incoming calls, the CLID can be used to match it with an account code from the IP Office's list of valid codes.

Analog phone users can only enter account codes before making a call or in response to an audible system prompt to enter a code when making the call.

Account codes can also be entered through the IP Office Phone Manager application, a system wide setting, determines whether Phone Manager will display a list of account codes from which users can select the code required or will hide the account code list.

In all the cases above, the account code entered is included with the call details in the IP Office's call record output.

Dial Emergency

Part of IP Office Short Codes, permits certain numbers to be dialled regardless of call barring, or a phone being locked.

Call Barring

It is possible to bar or allow calls to certain numbers such as international numbers or premium rate numbers for individual users or on a system wide basis.

Feature

- Restrict the dialing of specific numbers or types of numbers.
- Restrict certain users from dialing specific numbers or types of numbers.

Description

The IP Office supports call barring at many levels. Individual users can be barred form making any external calls. Short codes can be used at the system or individual user level to block the external routing of specific numbers or types of numbers. Typically the barring short codes are set to return busy tone, however they could route the call to an alternate number or to a Voicemail service that returns a 'barred dialing message'.

For users, the short codes can be placed in a User Restrictions set. Users are then configured for which User Restriction set applies for their calls.

In addition to barring the dialling of certain numbers, the IP Office can be set to bar the forwarding or diverting of calls to external numbers.

Least Cost Routes

By configuring a Least Cost Route calls may be routed via an alternative carrier. Time profiles can also be used to allow customers to take advantage of cheaper rates at specific times.

Multiple carriers are supported. For example, local calls are to go through one carrier between specific hours and international calls through an alternative carrier. Carrier selection using 2-stage call set up via in-band DTMF is possible.

Maximum Call Length

Allows the system to control the maximum duration of a call based on the dialed number. This could be used for controlling calls to cellular networks or data calls made over the public network.

PIN Restricted Calling

See Account Codes.

Forwarding

Forwarding

This is the ability to forward a user's calls to another extension or external number such as a Mobile/Cell Phone.

Calls can be forwarded in a number of scenarios and if not answered at the forward destination, will go to IP Office Voice Mail if voicemail is enabled for the user. There are two separate forward destinations, one for forwarding on busy and no answer and one for forward unconditional. Once the numbers have been set the user can easily set the required forwards to be active or not as and when required without having to re-enter the numbers.

If the user is a member of a hunt group, some types of Hunt Group calls can also follow forward unconditional.

Forward on Busy

If enabled, this forward will be triggered when the user is busy and another call comes for them, this does not include calls for a hunt group that they are a member of. A user is normally considered to be busy when they are on a call but depending on call waiting settings and key & lamp features this may not be the case.

Forward on No Answer

This forward is triggered if a call has been ringing for a user but they haven't answered it within the configured answer time, this includes calls that have been indicating call waiting if enabled.

Forward Unconditional

This makes all calls for the user go to the forward unconditional number, for example when away from their department IT support staff can have their calls forwarded to wireless, mobile or cell phones.

Call forwarding is processed after Do Not Disturb and Follow-Me.

Forward Hunt Group

Calls for a hunt group that the user is a member of can also follow forward unconditional if it is set, the hunt group must be set for either linear or rotary ring type, if the call is not answered at the forward destination it will not go to the users voice mail if unanswered but follow the hunt group call handling. This can be particularly useful in a sales environment where a number of people may be out of the office on Mobile/Cell Phones and still participate in the hunt group as if in the office.

Follow Me

Follow-Me is similar to Forwarding except that the destination can only be an extension on the same IP Office as the user making use of the feature. Follow-Me is typically used when a user is going to be working away from their desk, for example in a workshop. All the call settings the user has on their main phone will apply to calls that follow the follow-me feature, including forward on busy or no answer.

Follow-Me can be set either from the users usual phone – Follow-Me To – or from the phone where they want calls to go to – Follow-Me Here. Several people can have their phones forwarded to a follow-me destination and if the phone has a display it will indicate who the call is for.

Feature Phones

Programmable Buttons

As well as the usual digit keys, Avaya feature phones have dedicated function buttons like Mute, Volume, Hold, Conference and Transfer. In addition, on many Avaya feature phones, some of these keys can be programmed with a range of selected special functions. Originally used for calling other extensions on the system (Direct Station Select or DSS), these can be used for options from speed dialing numbers to controlling features such as Do Not Disturb.

For optimal operation many features also require an indicator to show whether a feature is active, see BLF Lamps.

Button programming is usually managed by the system administrator as part of the system configuration though some phones allow the user to reprogram some buttons and functions.

Busy Lamp Field (BLF) Indicators

Feature

- Status lamps or icons which indicate the status of a programmable buttons associated feature or function.

Benefit

- Indication of when a button or associated feature is active.

Description

Many Avaya IP Office phones have programmable buttons which can be assigned to various features. When those buttons include some form of BLF lamp, the button can also be used to indicate when the feature is active. This may be through the use of coloured LED lamps or different display icons and backgrounds (depending on the phone model).

For example, a button associated with another user will indicate when that user is active on a call. A button associated with a group will indicate when the group has calls waiting to be answered.

The speed dial icons within the IP Office Phone Manager and SoftConsole applications also act as BLF's. When the icons are associated with internal users, the icons will change to indicate the current status of the users.

Call History

Feature

- Storage of called and calling number details within the user's phone and/or IP Office application.

Description

Many IP Office phones keep a record of calls made and received, including unanswered calls. The method of operation varies according to the phone type but in all cases the call records can be used for return calls.

The IP Office Phone Manager application also maintains a call history record of the users last 100 calls. Phone Manager Lite can display call history for All calls and Missed calls only. Phone Manager Pro can display call histories for All calls, Missed calls, calls In and calls Out. Entries in the call history can be used for return calls, sorted and added to the Phone Managers local directory or speed dials

Language

Feature Phone menus and displays are available in many languages and usually the system default setting will be applicable to all phones, however it is possible to have language set on an extension by extension basis, this will also change the language of menus for IP Office Voice Mail.

Directory

The Directory is a list of up to 1000 numbers and associated names stored centrally in the system. A Directory Entry can be used to label an incoming call on a caller display telephone or on a PC application. The Directory also gives a system wide list of frequently used numbers for speed dialling via Phone Manager or a feature phone with a suitable display.

For example "Head Office" can be displayed when a known CLI/ANI is received. A user can also select "Head Office" in the Directory List in Phone Manager or on the display phone Directory to speed dial this number. IP Office's Directory is LDAP (Lightweight Directory Access Protocol) compliant which allows it to be synchronized with the information on any LDAP server. A maximum of 500 records can be retrieved by this method.

Self-Administration

The systems administrator may give the ability to change some of the phone settings, button programming for example, to certain users of the system, the range of changes that the user can make depends on the phone type.

On Hook Dialling

Most Feature Phones alloy the user to make calls by just dialling the number on the keypad, without having to lift the handset or pressing a speaker button. Usually the call progress can be monitored using the speaker in the phone, on phones that support hands free the whole conversation can be had without having to lift the handset.

Inbound Call Handling

Inbound Call Handling

Efficient handling of incoming calls is vital to many businesses, IP Office offers several features to provide versatile call processing including PC based applications Phone Manager and Soft Console covered elsewhere, and a standards based TAPI interface for 3rd party applications.

Incoming Call Routing

Traditionally incoming calls used to be presented to an Operator who then decided where to pass the call. The IP Office supports intelligent call routing capable of making routing decisions based on a number of criteria.

The system currently supports routing based on;

- presentation digits from the exchange such as DDI/DID or ISDN MSN.
- the calling parties telephone number or CallerID (This could even be part of the number received such as an area code).
- ISDN sub-address.
- service type i.e. Voice Call, Data Call, etc.

It is even possible to look for multiple criteria so, for instance, a DDI/DID call to a sales group could be handled differently depending on which part of the country the call is originating from.

Each incoming Call Route also supports a secondary destination 'Night Service' that can provide alternative routing for an incoming call based on 'time of day' and 'day of week' criteria.

Calls that cannot be routed to the configured destination are re-routed to a user defined 'Fall Back' destination. This can be particularly useful where calls are normally answered by an auto-attendant and a network fault occurs.

Where multiple call routes are set up to the same destination, a Priority level can be associated with the call. This priority level is used to determine a calls queue position in place of simple arrival time.

- **Note**: Calls that are ringing a free extension are not considered queuing and are not effected by a high priority call joining a queue.

Hunt Groups

A Hunt Group is a collection of users, typically users handling similar types of calls, e.g. a sales department. An incoming caller wishing to speak to Sales can ring one number but the call can be answered by any number of extensions that are members of the Hunt Group.

Four modes of call presentation are supported on IP Office;

- **Hunt mode / Linear mode:**
  One extension at a time sequentially always starting at the top of the list.
- **Group mode:**
  All extensions in the Hunt Group simultaneously.
- **Rotary mode / Circular mode:**
  Start with extension next in list to extension that was answered the last Hunt Group call.
- **Idle mode / Most Idle mode:**
  Start with extension that has been free for the longest time.

If all extensions in the Hunt Group are busy or not answered, another Hunt Group, called an Overflow Group, can be used to take the calls. An overflow time can be set to stipulate how long a call will queue before being passed to the Overflow Group.

Outside normal operation a hunt group can be put into two special modes; Night Service and Out of service.

In Night Service calls are presented to a Night Service Group. This can be controlled automatically by setting a time profile which defines the hours of operation of the main group or manually using a handset feature code.

The Out of Service mode is controlled manually from a handset. While in this mode calls are presented to the Out of Service group

Voicemail can also be used in conjunction with Hunt Groups to take all group related messages, play an announcement when the Hunt Group is in Night Service or Out of Service mode and give announcements while a call is held in a queue.

The voicemail functionality is being enhanced in R3.0 – a broadcast option provided. This feature will alter the operation so that the message notification will only be turned off for each hunt group member when they retrieve the message.

Night Service

When a Hunt Group is in Night Service mode the Hunt Group is temporarily disabled.

Callers to this Hunt Group will receive the busy tone or, if Voicemail is operational, played the Out of Hours greeting.

Alternatively a Night Service Fallback group can be used to provide cover, e.g. pass calls to a manned extension or an external number, e.g. a mobile.

A Hunt Group can be switched to Night Service mode by a user dialing the appropriate short code – by any extension or by specific users.

Time Profiles

Time Profiles can be used to stipulate when a Conference Bridge, Service, Hunt Group, Least Cost Route or a user's Dial In facility are operational. For example, a time profile can be used to route Hunt Group calls to a manned extension or voicemail outside of office hours, or be used to apply different Least Cost Routes at varying times to take advantage of cheaper rates. Multiple Time Entries can be created so that a Time Profile can be used to stipulate specific hours in the day e.g. 09:00-12:00 and 13:00-17:00. Outside of a Time Profile, voice calls would be re-routed according to the configuration but any currently connected calls at the time the Time Profile changes would not get cut off as the change only affects the routing. Data calls will get cut off as the time profile goes out of service but a new data call will start immediately if specified.

Queuing

Queuing allows calls to a Hunt Group to be held in a queue when all extensions in the Extension List are busy. When an extension becomes free the queued call is then presented.

While queuing, if Voicemail is operational, the caller will be played the Queue messages for this Hunt Group.

Contact Center Features

Contact Center Features

Contact Centers have specific needs for reporting on how calls are handled and these are covered in Chapter 13. The basic handling of the telephony requirements for a Call Center is a standard part of IP Office from Call Queuing to agents logging on and selecting the groups that they service.

Login

A contact center agent function, login is required before the agent is able to make or receive calls from that phone. A login idle period can be specified which will dictate how long an extension can be idle before the user is automatically logged off, ensuring that an extension is not left logged in and unattended.

Monitor Calls

A user can monitor, that is listen but not be heard, to other calls. This feature is not available by default, it must be specifically enabled in the system configuration. An option exists to have a beep tone indicate when monitoring is in use.

- **Support:** All digital and analog phone types including IP phones can be used to monitor. However IP phones cannot be monitored.

Acquire Call

Feature

- Takeover a call currently connected at another extension. This feature is also known as "Call Steal".

Benefit

- Assist a colleague who indicates they want you to take the call.

Description

The Acquire Call function can be setup as a special short code or programmed against a button on extension with programmable buttons. Use of the feature is subject to the IP Office's intrusion control settings, ie. the user acquiring the call must be set to be able to intrude and the user whose call is being acquired must be set to can be intruded.

Miscellaneous Features

Conference Calls

Calls can be placed on hold and a conference created using either the phone or desktop applications. Additional conference members may be added up to a maximum number of 64 members.

- The IP Office - Small Office Edition supports 24 conference parties with a maximum of 8 parties in any single conference.
- The IP406 V2 systems can support multiple conference calls totalling up to 64 parties. For example one conference of 64 calls or 21 conferences of 3 calls each.
- The IP412 has two 64-party conference bridges giving any combination from 2 x 64-party conferences to 42 x 3-party capacity.

Notes: Only two analog trunks are permitted in any single conference.

Meet-Me Conference

Also known as a conference bridge, this facility allows users to dial into a pre-configured conference set up by the system administrator.

Voicemail Pro provides a pre-configured facility to allow callers to be routed to a conference. This provides added security through the use of Passwords and time/date checking. *For more information on conferencing, refer to Chapter 12.*

- **Notes:** Only two analog trunks are permitted in any single conference.

Dial On Pickup

Automatically dials a specified extension when the phone is taken off hook. This facility is commonly used in unmanned reception areas to allow visitors to easily gain assistance.

External Control Port

The IP Office Base Module has two electronic switches, similar to relays, which can be either normally open, normally closed, pulsed open or pulsed closed.

These ports can be used for several purposes, for example as an addition to an electronic door release, rather than installing a set of release buttons at staff positions a single connection to the IP Office and either a dialled sequence or a feature button can open the door.

The External Control Port switches are used to trigger/control purpose built door release equipment which is supplied by a third party. All that needs to be done is to wire the trigger/control output of the third party device to the appropriate External Control port pins. The relay switch action is activated by use of a short code, Phone Manager, SoftConsole or Voicemail Pro action.

E911

A USA specific service. Upon connection to emergency services IP400 provides calling party information to an external line interface unit. The external unit carries out a number to text translation and forwards this to the emergency services bureau so that the originating location of the call can be unambiguously identified.

6. IP Telephony

Introduction to IP Telephony

As previously described IP Office can provide support of traditional analog and digital telephones in any mix creating a traditional PABX or Telephone system. Through the support of IP Phones, combined with the systems inherent gatekeeper and gateway functionality, IP Office can provide a 100% IP telephony solution or a hybrid of both the traditional and IP worlds.

With a conventional telephone system you plug your analog or digital telephone into an extension socket connected to your PBX or Key System. With IP Telephony you connect your telephone to your IP PBX via the LAN. There are two basic types of IP phones:

- A physical phone, which looks very similar to a standard telephone (IP Hard Phone)
- A software application (Phone Manager PC Softphone) which runs on the user's PC, allowing them to use either a headset/microphone.

IP telephony has the advantage of allowing extensions to be deployed both locally and remotely through the use of routers or VPN services.

When making use of IP Extensions, quality of service should not be ignored. In situations where more than eight IP extensions are required or where LAN Bandwidth is limited, a quality of service capable LAN switch, such as the Avaya P130, should be used. Also with IP hardphones there is need for Power over Ethernet (PoE) or “midspan power” to be provided to the phones – a list of Avaya approved POE options is available at the end of this section.

Gateways, Gatekeepers and H.323 - Technology Overview

H.323 Architecture is comprised of four logical components

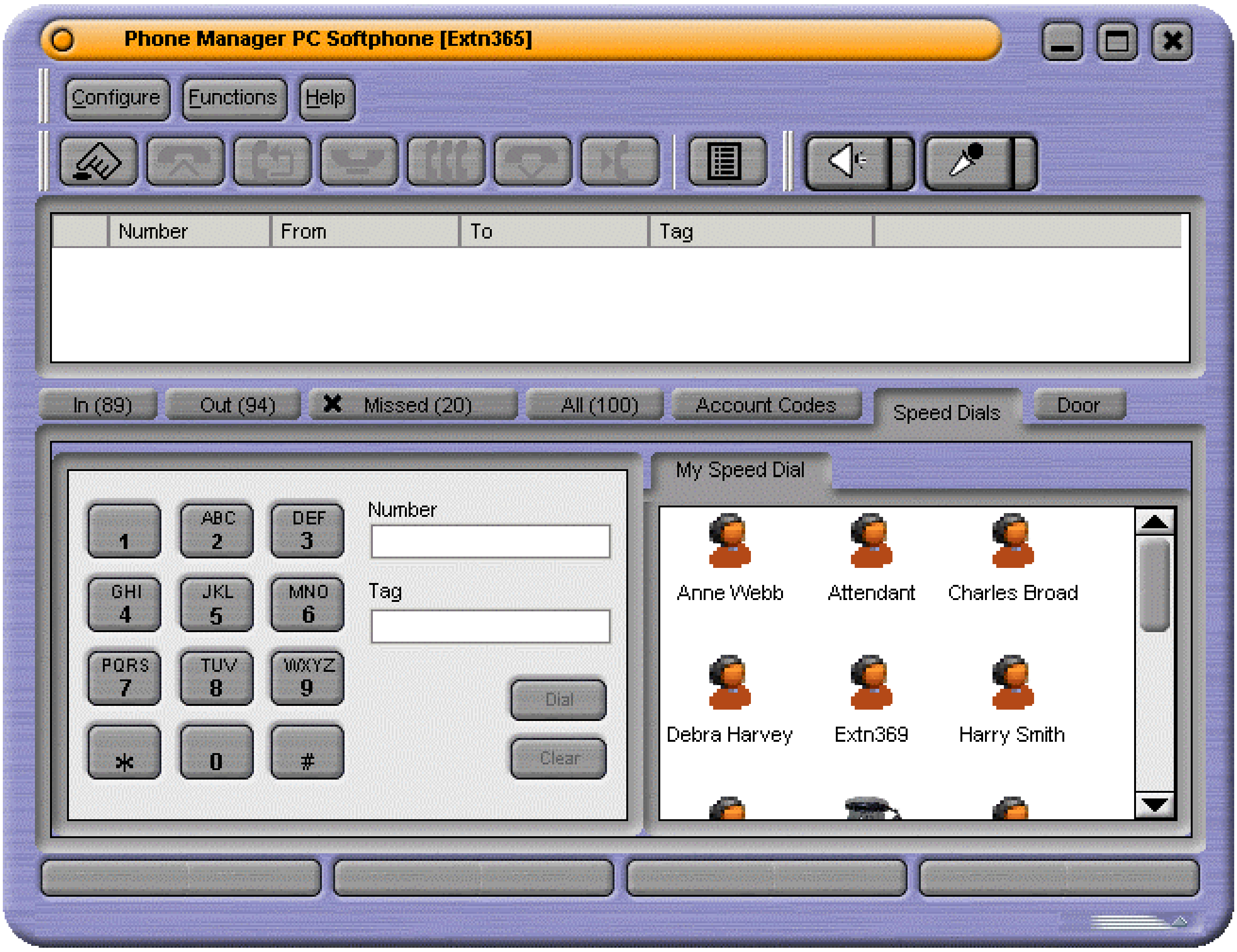
- Telephones are H.323 devices that can support Audio, Video and Data calls in any combination
- Gateways provide media translation to allow calls to be made to non-H.323 devices, for instance an analog telephone or the public network
- Multipoint Connection Units (MCU) facilitate multipoint conferences
- Gatekeepers control the call processing for all of the above

These four devices types are grouped together in what is known as an H.323 zone (a zone is analogous to a PABX). Each zone has a single Gatekeeper that can be considered as the brains of the system dealing with call distribution, call control and the management of resources. On power-up, telephones, Gateways and MCU's make a registration request against a Gatekeeper who authenticates, accepts or rejects their request to become a member of the zone. Once accepted, a telephone wishing to make a call sends a call set-up message to the Gatekeeper who will then send an alert to the called party or if the call is to a non-H.323 telephone establish the call via a Gateway.

The design of IP Telephony systems has been driven with open standards in mind. IP Phones, Gateways and Gatekeepers all support the H.323 standard and it is this that allows devices from different manufacturers to work together. IP Office has the integral Gateway and Gatekeeper functionality required to provide a fully functional IP Telephony environment.

IP Softphone (Phone Manager PC Softphone)

IP Office's Phone Manager Pro can be configured to operate as an IP Softphone – 'Phone Manager PC Softphone' by use of license keys.



Phone Manager PC Softphone has the same GUI as Phone Manager Pro, but add the ability to make and receive phone calls and control call volume from the PC. Like Phone Manager Pro, Phone Manager PC Softphone communicates with the IP Office system unit via the wired or wireless LAN. The difference is that there is no physical telephone and conversation actually takes place via the PC's audio interface such as a sound card or USB headset.

Phone Manager PC Softphone has the significant advantage for mobile users with remote access to the LAN, providing 'a phone within their laptop' with all the features available as in the office.

Notes:

- Phone Manager PC Softphone requires on a multi-media PC running Windows 2000 and Windows XP operating systems with speakers and microphone installed (USB headset, USB handset or soundcard).
- The minimum PC specification is a Pentium 400 MHz (700 MHz recommended) or above with 128MB RAM minimum
- Phone Manager PC Softphone supports QoS in the form of DiffServ for both Windows XP and Windows 2000 when used in soft phone mode.
- When Phone Manager PC Softphone is used over a wireless LAN, no more than 3 simultaneous calls can be supported per access point.

IP Telephony Features

- **Gatekeeper**
  The IP Office gatekeeper allows the registration of up to 180 IP extensions on the IP406 V2, 360 IP extensions on the IP412; less the number of traditional analog and digital telephones already configured on the system.
- **Gateway**
  The IP Office must fitted with a Voice Compression Module to enable IP telephony. The Voice Compression Module provides the H.323 gateway function that allows IP extensions to make calls to other non-IP devices. The maximum number of simultaneous calls is limited by the number of channels available on the Voice Compression Module.
- **Silence Suppression**
  Silence suppression is a technique used to make the best use of available bandwidth. Silence suppression works by sending descriptions of the background noise, rather than the actual noise itself, during gaps in conversation thereby reducing the packet size needed. Background noise is very important during a telephone call. Without noise the call will feel very unnatural and give a perception of poor quality.
- **Compression**
  IP Office supports a wide range of voice compression standards including G.711, G.723.1 and G.729a. The method of compression can be either automatically established on a call-by-call basis or be configured on an individual extension basis.
- **Fast Start**
  When supported by an IP extension, this facility reduces the protocol overhead allowing an audio path to be established more quickly.
- **Out of Band DTMF**
  When configured, IP Office will generate DTMF tones on behalf of an IP extension. This is useful when navigating external voicemail systems and Auto-Attendants.
- **Direct Media Path**
  Direct Media Path allows the speech path between two IP extensions (after call setup) to be routed directly to each other. This allows the IP Office system to free voice compression resources allowing them to be used in the most efficient way.
- **Auto-Create Extensions**
  This facility allows IP Office to automatically create an extension entry for new extensions added onto the local area network. In cases where the local area network is not secure this facility can be disabled.
- **Fax Transport**
  Fax Transport allows fax calls to be routed over VoIP trunks between IP Office systems using a proprietary IP Office transport protocol. T.38 is not supported.

Power Options for IP Telephones

IEEE, the standards body governing Power over Ethernet (PoE), has ratified P802.3af, "DTE Power via MDI". The standard is more commonly known as IEEE Power over Ethernet. With the ratification of this standard, Avaya has confirmed the support of the standard in its current range of IP telephones.

With Power over Ethernet, both power and data are carried over one CAT 5 Ethernet cable. Deploying IP telephones utilizing Power over Ethernet eliminates the need for local power supplies, AC adapters and cables, and allowing power to be provided from the wiring closet/switch room where it can be easily connected to a UPS system.

There are several power options, in addition to IEEE Power over Ethernet, available to customers to power their Avaya IP telephones.

Avaya 1151B1 Individual Power Supply

The Avaya 1151B1 individual power supply is a "desktop"; single output 48-volt Direct Current (DC), 20-watt power supply. The power supply can operate within a wide range of Alternating Current (AC) input voltages: 90 - 264 Volts Alternating Current (VAC), 47-63 Hz. This power supply replaces the MSP-1, KS-22911-L1/2, 329A, 353A and the 1151A1/A2 DC power supplies and the 2012D AC transformer. The 1151B1 power unit has a green Light Emitting Diode (LED) that shows the unit has power on the PHONE jack, pins 7&8 when AC power is applied.

This item is available in three different Price Element Codes (PECs) as follows:

1. 1151B1 power supply: material code 700227242
2. 1151B1 with a Category (CAT) 5 Cable for Internet Protocol Telephones: material code 175707
3. 1151B1 with a CAT 3 Cable for Digital Communications Protocol (DCP) Telephones: material code 175706

1151B1 local power supply

Avaya 1151B2 Individual Power Supply with Backup

The Avaya 1151B2 power supply is a "desktop ", single output 48-volt DC, 20-watt power supply with battery holdover. The power supply can operate within a wide range of AC input voltages: 90 - 264 VAC, 47-63 Hz. When AC power fails the battery will provide 15 minutes of holdover at full load (20 watts) or 8 hours at light load (2 watts). The 1151B2 power unit has two LED's: a green LED that shows the unit has power on the PHONE jack, pins 7&8 when AC power is applied. A YELLOW LED that shows the unit is charging the battery when illuminated. The yellow LED is off when the battery is fully charged. The GREEN LED blinking indicates the unit is running on battery power. This item is available in three different PEC codes as follows:

a. 1151B2 power supply: material code 700237019
b. 1151B2 with a CAT 5 Cable for IP Telephones: material code 177086
c. 1151B2 with a CAT 3 Cable for DCP Telephones: material code 177087

1151B2 power supply with battery backup

Avaya Mid-Span Power Distribution Units

The Avaya Mid-Span Power Distribution Units. These are power devices, specially designed for IP-telephony, providing power over Ethernet (PoE) for up to 24 IP telephones or wireless LAN (WLAN) access points. The Mid Span Power units are designed to mount in a 19-inch rack with the data equipment or they can be stacked up to four units high using the optional rubber feet. It is 1U in height (1.75 inches) and has up to twenty-four RJ45 data input jacks on the bottom row and twenty-four data and power output RJ45 jacks. The Mid-Span Power units provide a maximum of 200 Watts or a peak of 16.8 watts per port. The Mid Span Power units can also be called a PDU (Powered Data Unit) POE device. Power over the LAN will simplify the installation and support of IP telephones for our customers, enhancing acceptance of the technology. Data is unaffected if power is disrupted and if the device does not require power (an example is a laptop connected to the unit.) The Material Codes for the Mid-Span power units are:

- 700180433 - Mid Span 24 Port AC LAN Hub IP Phones
- 700253099 - Mid Span 24 Port AC LAN Hub IP Phones SNMP
- 700250525 - Mid Span 12 Port AC LAN Hub IP Phones
- 700253107 - Mid Span 12 Port AC LAN Hub IP Phones SNMP
- 700253115 - Mid Span 6 Port AC LAN Hub IP Phones
- 700253123 - Mid Span 6 Port AC LAN Hub IP Phones SNMP

Mid-Span power supply

Avaya C460 Multilayer Modular Switch.

The C460 can provide power to up to 196 IP phones.

The Avaya C460 features a compact modular six-slot chassis with the following main characteristics:

- Four I/O slots and two Supervisor slots
- Fully redundant architecture (including switching fabric, supervisor modules and PSUs)
- Power over Ethernet (PoE) support with the FE ports
- High density – up to 192 FE PoE ports and 48 GE ports
- Fabric switching throughput of 64Gbps at Layer 2 and 48Mpps routing at Layer 3
- Policy and QoS mechanisms
- Full router functionality
- Wire-speed Layer 3 forwarding on all ports
- Optimal use of physical chassis size (10U)
- 300W or 1000W (for PoE support) power supplies

The C460 full redundancy (supervisor and fabric, power supply, link and port interfaces, router processor, and fans), high port density and powerful Layer 2 and Layer 3 wire-speed switching engine make it suitable for robust network infrastructure. The C460 offers advanced management and monitoring capabilities using complete GUI tools, including the SMON and Any-layer SMON applications based on the Avaya Integrated Management suite.

C460's available I/O modules:

- 48 10/100 PoE port Inline Power module
- 48 10/100 PoE port Inline Power + 2 GBIC (SFP) Gigabit Ethernet port module
- 12 GBIC (SFP) ports Gigabit Ethernet module
- 48 10/100 port Ethernet module
- 48 10/100 port Ethernet + 2 GBIC (SFP) Gigabit Ethernet port module

The C460 extends Avaya Convergence solutions to the network edge by providing advanced network capabilities, including Quality of Service (QoS), high performance, advanced power management, security and manageability. Designing a converged network infrastructure using this highly resilient, modular, high performance solution ensures a lifespan of the network, which will reduce cost of ownership and improve return on investment

With its flexible configuration options and high-capacity performance, the C460 can also be deployed as a distribution layer switch or as the network backbone for small to medium enterprises looking for a reliable modular solution.

For enterprises deploying Avaya Communications Manager for mission-critical call center and large-scale campus environments, the C460 offers an ideal IP Telephony platform that combines fault tolerance, network responsiveness for business continuity, and integrated management and monitoring for converged networks.

The following table includes the PoE items of the C460:

Avaya C460	Description	Material Code
C460ML-PWR-CFG	C460 Switch PoE basic configuration (SPV, PSU, fan)	700281603
M4648ML-T-PWR	C460 Multi-layer 48 x 10/100BaseT (RJ-45) Inline Power module	700281587
M4648ML-T-2G-PWR	C460 Multi-layer 48 x 10/100BaseT (RJ-45) + 2 x GBIC (SFP) Inline Power module	700281579
MPS4610-AC	C460 1000w Power Supply (AC) Inline Power Support	700281595

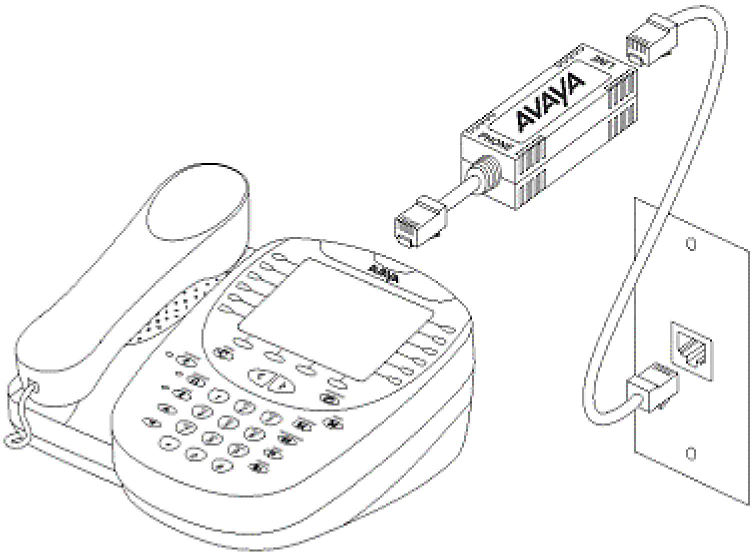
Avaya IP Phone Power Adapter

Despite the ratification of IEEE 802.3af-2003 and the support of the standard by vendors, some customers may utilize a legacy power scheme supported by Cisco switches. The following power adapter is specifically for Avaya IP Telephones and can be used to power theses telephones from specific Catalyst power blades (Catalyst is a registered trademark of Cisco Systems, Inc.).

The Avaya IP Phone Power Adapter (material code 700259369) was tested and will work with the following:

- Catalyst 6000 Inline Power 10/100 BaseT Switching Module - (WS-X6348-RJ45V).
- Catalyst 4000 Inline Power 10/100 BaseT switching module - (WS-X4148-RJ45V).

More detail on implementation of IP Power options is covered in the IP Office IP Phone Installation manual.



Avaya IP Phone Adapter

Power Consumption

Measured in Watts using an IEEE 802.3af power supply at 48V.

Phone	Typical	Worst Case	IEEE 802.3af
4601, 4602, 5601, 5602	3.5W	4.6W	Class 2
4602SW, 5602SW	4.1W	5.0W	Class 2
4606, 4612, 4624	5.0W	6.4W	Class 0
4610SW, 5610	4.0W	6.0W	Class 2
4620	7.7W	9.9W	Class 3
4620SW, 4621SW, 5620	5.9W	8.0W	Class 3

Typical is measured off-hook sample size 1. Worst Case is analytical. Except the 4601, 4602, 5601 and 5602 all telephones had a PC attached at 100Mbps. The EU24 adds less than 1W to the 4620, 4620SW and 5620 numbers.

VoIP FAQ

What is Quality of Service?

Quality of Service (QoS) is a measure of the performance of a network that reflects the availability of network service and the quality of network transmissions. The term itself refers to a number of networking technologies and techniques and does not necessarily restrict itself to any single protocol or standard.

There are a number of measures that can be taken on the LAN and WAN to make them 'good enough' to carry voice traffic. Some of these are the implementation of standards based QoS protocols while are simply a matter of network architecture and good network management practices.

The term 'good enough' is intentional. Every customer will have different expectations and different budgets to work to. Some will be willing to upgrade their networks to use the best possible equipment and practices. To others the additional expense may be viewed as unnecessary.

Examples of standards based Quality of Service protocols include DiffServ and 802.1p/q.

What are the Symptoms of Poor Speech?

Poor speech quality manifests itself in three distinct ways.

- Echo.
- Delays.
- Warble, that is 'in a cave' or underwater like sound.
- Clipping.
- Distortion.
- Disappearing speech, some words or parts of words missing.
- No speech or speech in one direction only.

What Causes Poor Speech Quality?

Poor speech quality can be caused by a number of different problems that can occur in most networking environments

- VoIP conversion delays.
- Network delays.
- Dropped traffic.
- Traffic being received at varying speeds.
- Inappropriate conversions from speech to VoIP.
- Multiple VoIP conversions.

These causes can be generalized into three specific areas, network induced delays which magnify the effect of echo inherent in VoIP networks, unreliability in the network which can introduce large latencies or inconsistencies in the transmitted VoIP traffic and the actual process of converting speech to VoIP traffic and vice versa.

How Do I Minimize Delay Induced Echo?

Delay in a network originates from a number of different sources and phenomena.

A primary source of delay is the process of converting speech to VoIP traffic. The IP Office supports a number of standards based encoding methods to allow the optimum trade off between quality and bandwidth to be made (see What Bandwidth Do I Require for Each Voice Call?). IP Office incorporates integral echo cancellation to minimize the effect of echo introduced in the VoIP conversion process.

Another source of delay comes from data and voice traffic queuing at the ports of switches, routers, gateways and or bridges that make up the network. It is possible that the traffic queuing at a port is minimal and no action needs to be taken. This would be the case if the available bandwidth far exceeded the demand. To overcome queuing bottlenecks in the network, IP Office prioritizes voice traffic using a standard known as DiffServ. This marks each IP packet carrying voice with a flag so that routers, etc. can force packets containing voice to the front of the transmission queue. An alternative method of prioritization that can be used by switches and routers, with an equally satisfactory result, is to look at what protocol is being used and prioritize this. All voice traffic is carried using two easily identifiable protocols, RTP and RTCP. Both methods are equally good, choose whichever method is the most cost effective and easiest to implement and manage.

A similar source of delay can be attributed to specific network nodes that convert from one network medium to another. For example T1 trunk lines may be carried across a high speed DSL like connection and converting from the high speed link back to T1 in the access gateway takes time to perform. Any VoIP traffic being carried through this link is therefore subject to the delay introduced by this conversion step. The delay may be minimized by ensuring that an appropriate QoS mechanism is enabled in the gateway to prioritize the VoIP traffic. IP Office incorporates integral echo cancellation to help minimize the effect of this kind of delay introduced by the network.

Delay can also be introduced as a consequent of collisions occurring on particular segments of the LAN. Collisions result when two devices on a shared switch port or segment try to transmit simultaneously. This causes all devices to stop transmitting for a period of time. This is a normal characteristic of many older Ethernet networks and, if occasional, may pass unnoticed. The more devices sharing a switch port, and the busier they are, the greater the opportunity for collisions. This is simply resolved by reducing the number of devices on each port, or by dedicating a port to each VoIP device. If you are just using VoIP to link two IP Offices together, it's well worth dedicating a port to each IP Office and router at either end of the link as the cost implications are likely to be minimal. In this regard it is important to dimension a network to cope with existing traffic demands as well as any future increases in traffic carrying capability.

How Do I Minimize Warble and Clipping?

Warble, clipping and some distortion quality problems are symptoms of variable delay and or packet loss. Variability in the delays of traffic is called jitter. Jitter and packet loss may be the result of switches and routers that are either faulty or working outside their design intentions.

IP Office provides jitter buffers to compensate for a moderate amount of jitter found in networks. Voice traffic is quite tolerant of small amounts of packet loss so in most cases this may be ignored. Where packet loss is excessive (greater than 2% say) the cause should be established and fixed. This could be due to a fault or simply an over worked device discarding packets. Significant packet loss can cause perceptible losses in speech, to the extent that no speech may be heard either in one or both directions.

How Do I Minimize Distortion?

Each time speech is converted into a digital signal and back again, tiny difference from the original creeps in. The more times this happens on a single call, the bigger those differences can become. These differences can become perceptible as distortion.

Ideally, the path speech takes should only require one 'analog to digital to analog' conversion and this will be the case in many instances. Exceptions to this occur when making calls to mobile telephones or voice mail systems where the analog to digital to analog conversion may occur twice (once on IP Office and once on the mobile network, etc).

Different encoding methods will have different effects. IP Office supports a range of encoding methods to allow you to choose the one with the right quality versus bandwidth for your network. In general multiple conversions should be minimized wherever possible.

What Benefits Do I Get From Using IP Office To Provide My WAN?

IP Office allows you to intelligently manage the bandwidth over any directly connected WAN link. Using IP Office, it is possible to guarantee bandwidth for data as well as voice traffic. When no voice traffic is present, the free bandwidth is made available for data. Through the use of silence suppression, data can even borrow the gaps in conversations for additional throughput.

What Bandwidth Do I Require for Each Voice Call?

The bandwidth used varies depending on the compression method chosen. IP Office supports a wide range of compression standards, including the most popular G.723.1 and G.729a. These will occupy approximately 10K and 13K of bandwidth respectively.

Use the following chart to choose the most appropriate compression algorithm for your available bandwidth.

Audio Codec	RTP Voice Data Payload	Packets per Second	LAN (bps)	% Overhead LAN	WAN (bps)	% Overhead WAN	Algorithmic Delay (milli-seconds)
G.723.1	24 Bytes	33.33	20,800	225%	9,867	54%	80
G.729a	20 Bytes	50	29,600	270%	13,200	65%	40
G.711 (64K)	160 Bytes	50	85,600	34%	69,200	8%	20

What Delay is Acceptable?

Try to keep the overall end-to-end delay to 150 milliseconds or below.

An idea of the delay inherent in the network can be measured by carrying out a ping test and dividing the result by two. IP Office has built in echo cancellation to maximize speech quality.

What is the Perfect Network?

For those customers who are willing to upgrade their infrstructure, the best preactice network would have every device on the LANconnected through its own dedicated port on a DiffServ capable layer 3 switch. Connections to the WAN should be through devices that support DiffServ, for instance IP Office.

How Many Simultaneous Calls Can I Get Down My Link?

The following chart illustrates the theoretical maximum number of simultaneous voice calls that can be delivered over a WAN for a given link speed. This does not take into account any bandwidth that may be required for data traffic between sites or the physical limit of VoIP calls for the specific version of IP Office in use.

The number of simultaneous voice calls can be in excess of the capabilities of the individual platform, where the calls transit the switch as data traffic. In this situation compression resources are not used but obviously must be catered for in the overall bandwidth provision.

Compression	G.723.1 (6K3)	G.729a (8K)	G.711 (64K)
Algorithmic Delay (seconds)	0.08	0.04	0.02
Number of Calls			
- 64Kbps Link	6	4	0
- 128Kbps Link	12	9	1
- 256Kbps Link	25	19	3
- 512Kbps Link	51	38	7
- 1Mbps Link	103	77	14
- 2Mbps Link	207	155	29

What is the Maximum Number of Simultaneous VoIP Calls?

Each IP Office can be fitted with an optional Voice Compression Module (VCM) to support VoIP connections.

- The IP406 can be fitted with a single module offering up to 30 simultaneous calls.
- The IP412 is capable of supporting two modules of all types, allowing up to 60 simultaneous calls.

Does the IP Office Support Fax over IP ?

The IP Office has a proprietary method for carrying Fax traffic on a VoIP call. IP Office does not currently support the T.38 Fax standard. IP Office supports Fax speeds up to 14.4 Kbps. The bandwidth requirements for a Fax call will initially be as per the specified or negotiated compression method and then the bandwidth requirement will change to accommodate the Fax data. The Fax bandwidth will vary depending on the speed with which the Fax devices are communicating and the type of link, at 14.4 Kbps the bandwidth requirement will be approximately 27 Kbps on the LAN or 19 Kbps on a Point to Point WAN link with header compression enabled.

Network Assessment

With IP Office, optimum network configurations can support VoIP with a perceived voice quality equivalent to that of the Public Switched Telephone Network (PSTN). However, not every network is able to take advantage of VoIP transmissions. It is important to distinguish between basic compliance with the minimal VoIP standards and validated support for QoS which is needed to run VoIP applications over a data network.

With the exception of standalone configurations where IP phones connect directly connected to the ports on IP Office, Avaya now requires that all customers formally audit their networks for IP telephony readiness before attempting to install any VoIP application.

A network assessment should normally include:

- Physical inventory of all equipment inclusive of the current version of code, and configurations as needed.
- An accurate and complete network topology for all networked sites, inclusive of IP addressing and physical/logical connections.
- An evaluation of the network's topology to check that the design is both sound and reasonable.
- Measurement of packet loss, jitter and delay over the course of multiple days and measured on a per minute basis. A graphical representation of the data is the preferred output method.
- Examination of QoS/Class of Service (CoS) parameters in place in the network.
- Summary of findings and possible actions to correct problems.

The assessment should leave you confident that the implemented network will have the capacity for the foreseen data and voice traffic, and can support H.323, DHCP, TFTP, and jitter buffers in H.323 applications.

With this in mind, if you require support during or after an IP Office VoIP installation, a copy of your network assessment documentation will be requested by your support channel.

For more details about available tools, resources and services to enable you to audit your network for VoIP readiness, please contact your local Avaya representative.

VoIP Standards Supported

IP Office supports the following protocols and standards:

- H.323 V2 (1998), Packet-based multimedia communications systems.
- Q.931, ISDN user-network interface layer 3 specification for basic call control.
- H.225.0 (1998), Call signaling protocols and media stream packetization for packet-based multimedia communication systems.
- H.245 (1998), Control protocol for multimedia communication.
- Audio CODECs:
    - G.711 A-law/U-law.
    - G.723.1 MP-MLQ.
    - G.729 Annex A – CS-ACELP.
- Silence Suppression.
- Fax Relay (IP Office to IP Office Fax Transport over IP).
- Local End Echo Cancellation 25ms.
- Out of band DTMF.
- Jitter buffer, 5 frames of jitter buffer.
- Internet Standards/Specification (in addition to TCP/UDP/IP).
    - RFC 1889 – RTP/RTCP, Real Time and Real Time Control Protocol.
    - RFC 2507, 2508, 2509 – Header Compression.
    - RFC 2474 – DiffServ, Type of Service field configurable.
    - RFC 1990 - PPP Fragmentation.
    - RFC 1490 - Encapsulation for Frame Relay.
    - RFC 2686 - Multiclass Extensions to Multilink PPP.

7. Public and Private Voice Networks

Public and Private Voice Networks

IP Office supports a wide variety of voice and data networking options. This section describes traditional circuit switched options to packetized voice solutions such as Voice over IP and Voice over Frame Relay.

Traditional Private Voice Networking

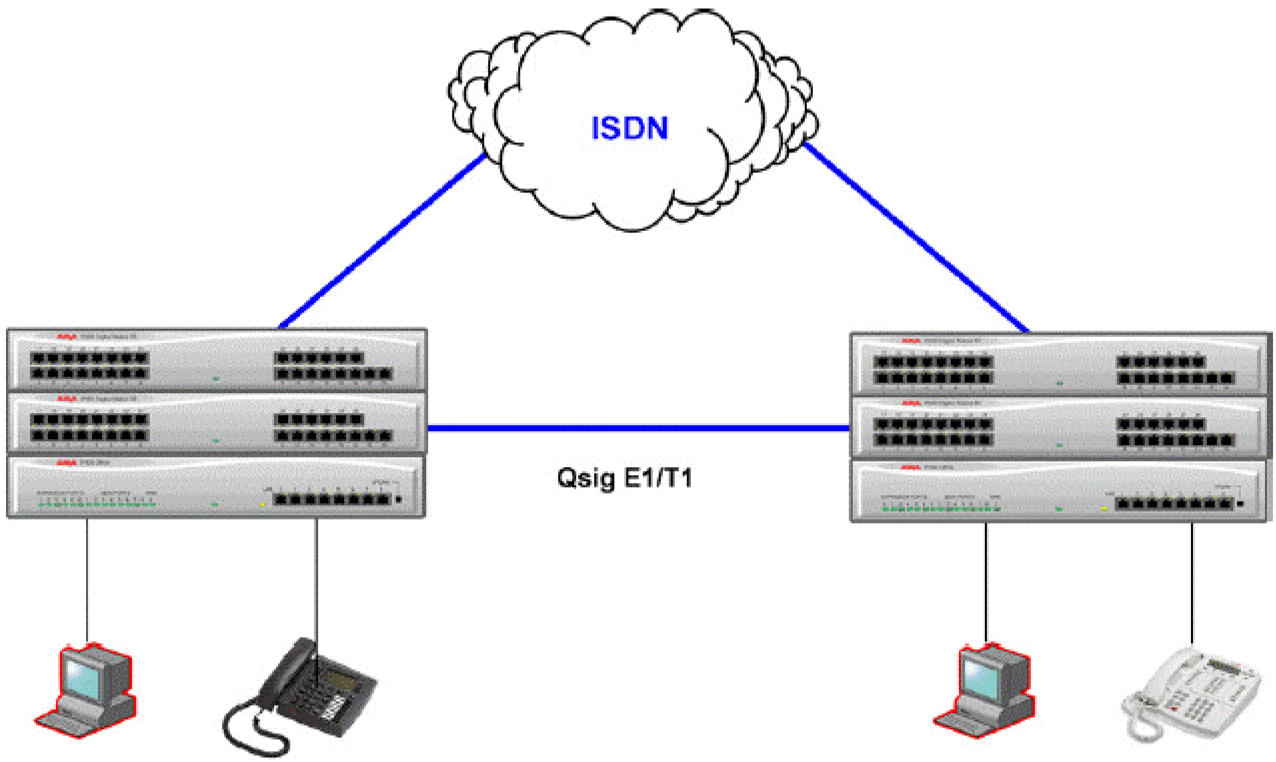
Private voice networks can be constructed utilizing structured leased line circuits (E1 or T1) or alternatively by establishing permanently connected 'B' channels between IP Office systems. Each channel within the Primary Rate interface can provide a single voice or 64K/56K data call.

If leased line circuits are used within a private networking scenario, these PRI interfaces are typically configured in software to employ QSIG signaling between sites.

QSIG provides feature transparency between PBXs and is the favored signaling standard within multiple vendor and international voice networks. The PRI module terminates a QSIG connection with a 120 ohm RJ45 interface.

QSIG provides the following additional supplementary services across this network:

- **Simple Telephony Call/Basic Call:** ETS300 171/172.
- **Circuit Switched Data Call/Basic Call:** ETS300 171/172.
- **Called/Calling Line ID Presentation:** ETS300 173.
- **Called/Calling Name Presentation:** (SS-CNIP, SS-CONP, SS-CNIR) ETS300 237/238.
- **Message Waiting:** (SS-MWI) EN301 260/255.
- **Transfer:** (SS-CT) ETS 300 260/261.



Traditional Voice Networking

Trunk/Line Types Supported

The IP Office platform supports a range of trunks and signaling modes for connection to the public telephone network (Central Office). Some of these lines are only available in certain territories; please check with your distributor for local availability.

ISDN Primary Rate (ETSI CTR4) - IP400 Office PRI E1

ISDN Primary Rate provides 30 x 64K speech channels over an E1 circuit. Signaling Conforms to the ETSI Q.931 standard with Cyclic Redundancy error Checking (CRC).

The following supplementary services are supported:

- Calling Line Identification Presentation (CLIP) Provides the telephone number of the incoming call to the IP Office.
- Calling Line Identification Restriction (CLIR) Inhibits the telephone number of the IP Office being presented on an outbound call.
- Connected Line Identification Restriction (COLR) Inhibits the COLP service.
- Direct Dialing In (DDI) Where the exchange provides the last x digits of the dialed number on an incoming call. This allows IP Office to route the call to different users or services.
- Sub-addressing Allows the transmission/reception of up to 20 digits, additional to any DDI/DID or CLIP information, for call routing and identification purposes.

ISDN Basic Rate (ETSI CTR3) - IP400 Quad BRI

ISDN Basic rate provides 2 x 64K speech channels using Q.931 signaling and CRC error checking. Both point to point and point to multipoint operation is supported. Multipoint lines allow multiple devices to share the same line, however point-to-point is the preferred mode.

Basic rate supports all the services that are supported on the primary rate version with the addition of

- Multiple Subscriber Number This service is usually mutually exclusive with the DDI/DID service and provides up to 10 numbers for routing purposes, very similar to DDI/DID.

North American T1 - IP400 Office PRI T1

T1 Primary Rate provides up to 24 56K channels over a 1.54M circuit. Each channel of the T1 trunk can be independently configured (channelized) to support the following signaling emulations with handshake types of immediate, delay or wink.

- Loop-Start
- Ground-Start
- E&M Tie Line
- E&M DID
- E&M Switched 56K
- DID - Channels configured for DID/DDI support incoming calls only. The carrier or Central Office will provide the last x digits that were dialed to be used for call routing.
- Wink-Start

IP Office T1 trunks support both DNIS and ANI services, where available from the central office.

- Dialed Number Identification String (DNIS) Provides a string of digits to the IP Office depending on the number dialed by the incoming caller. This string can then be used to route callers to individual extensions, groups or services.
- Automatic Number Identification (ANI) Provides IP Office with a number identifying who the caller is. This may then be used for routing or computer telephony applications.

T1 trunk cards incorporate an integral CSU/DSU, eliminating the need for an external unit. The CSU function allows the trunk to be put in loop-back mode for testing purposes. This can be set manually, using the monitor application, or automatically from a Central Office sending a Line Loop Back (LLB) pattern. The DSU function allows the T1 trunk to be shared between data and voice services.

North American Primary Rate Interface - IP400 Office PRI T1

IP Office supports Primary Rate trunks on 5ESS or DMS100 central office switches provided by AT&T, Sprint, WorldCom and other Local Telcos. Channels can be pre-configured for the supported services or negotiated on a call-by-call basis.

Special Services can be configured to route calls to local operators or pre-subscribed carriers for both national and international calls (SSS). Alternate carriers can also be selected through the configuration of IP Offices Transit Network Selection (TNS) tables.

IP Office also supports the Calling Name and Number service over Primary Rate Trunks (NI2).

Analog Trunks (Loop Start / Ground Start)

- **Loop Start**
  Loop start trunks are available on the IP Office as 4-port plug in cards for the control unit or as a stackable 16-port expansion module. The first two trunks on the expansion module are automatically switched to power fail sockets in the event of power being interrupted. They conform to the TIA/EIA-646-B standard. The loop start trunks also support incoming caller line identification (ICLID) conforming to GR-188-CORE and GR-31-CORE standards. IP Office can use this information to route calls or provide it to computer applications to display additional information about the caller.
- **Ground Start**
  Not available in all territories. Ground Start trunks are available as a stackable 16-port expansion module. The first two trunks on the module are automatically switched to power fail socket in the event of power being interrupted. They conform to ANSI T1.401 and TIA/EIA-646-B standards.

PRI E1R2

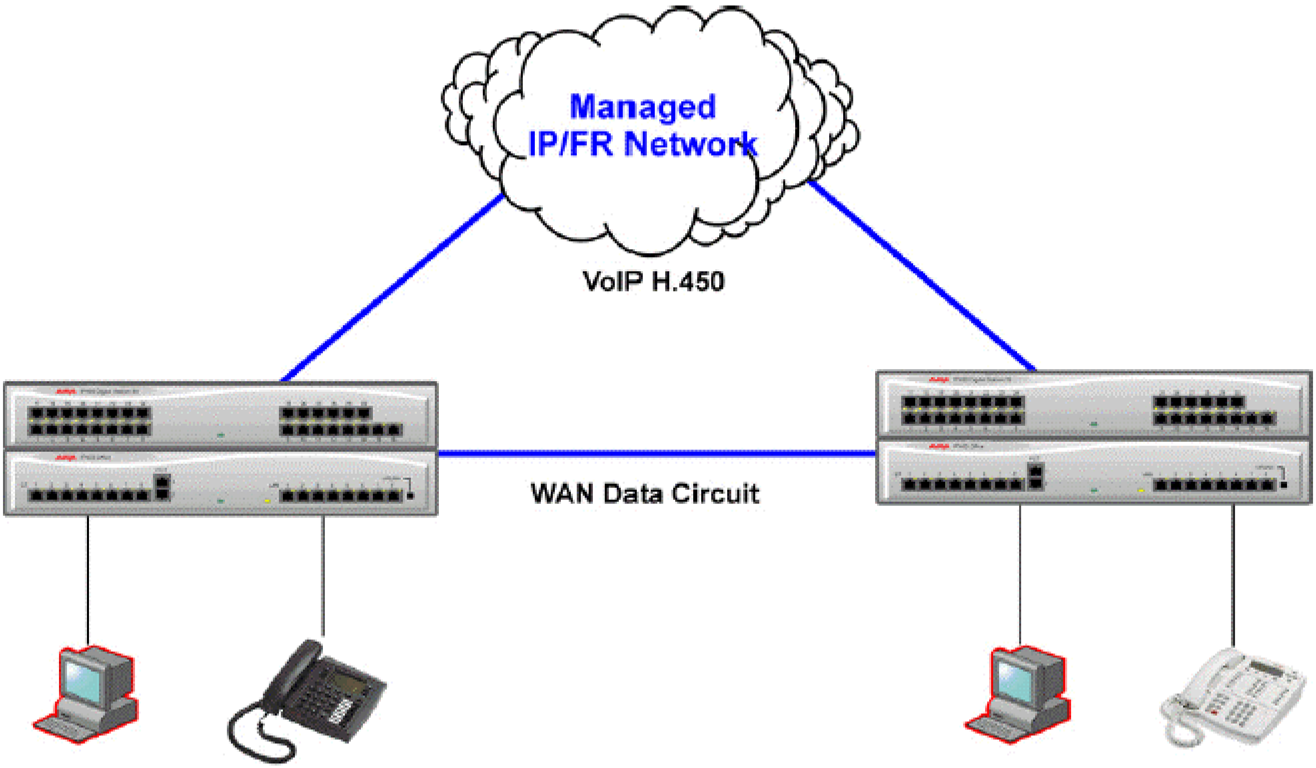
The IP400 Office PRI 30 E1R2 card is available in two versions supporting either RJ45 or coax network connections. Each card provides 30 or 60 channels that can be configured for MFC, Pulse or DTMF dialing dependent on the requirements of the network.

Packet Based Voice Networking

IP Office supports a wide variety of voice and data networking options from traditional public circuit switched networks and structured leased circuits to a multitude of packetized voice solutions. This section describes the options available for businesses that are ready to embrace packetized voice solutions such as Voice over IP (VoIP).

Packet based voice networking between IP Office sites can be achieved in a number of ways:

- VoIP over an unstructured private circuit.
- VoIP over a managed IP VPN.
- VoIP over a managed Frame Relay network.
- VoIP across the LAN.
- VoIP across the public network.



VoIP networking across IP network or WAN

VoIP over an Unstructured Private Circuit

Private voice networks can be constructed making use of available unstructured data circuits (X.21, V.35) at speeds of up to 2 Mbps.

These data circuits are accessed via IP Offices equipped with an optional Voice Compression Module (VCM), providing from 4 to 60 VoIP calls. This approach can realize significant savings by allowing compressed VoIP calls to be interleaved with data on any leased circuit with spare bandwidth.

Where multiple sites exist, the addition of the IP400 Office WAN3 10/100 Module allows larger networks to be designed. Two modules can be supported on a single system providing a total of 7 leased lines.

VoIP over a Managed Frame Relay Network

Frame Relay is a high-speed, packet switching WAN protocol that enables the interconnection of geographically dispersed LANs. Frame relay is usually offered as a service by a public network provider. However, some private organizations can also own and manage their own Frame Relay networks.

Frame Relay is a connection-oriented protocol, which means that it relies on an existing end-to-end path between devices connected across the network. It implements these connections using Permanent Virtual Circuits (PVCs).

Like a leased circuit, a PVC is a logical path that connects two devices. This path between the source and destination point is a dedicated connection, so the PVC is always available to the connected devices. However, unlike a leased circuit many PVCs can coexist on a single access bearer which allows devices to share the bandwidth of a given transmission line.

Voice over a managed Frame Relay network is similar to Voice over a managed IP network except that the access interface is usually an unstructured leased circuit via IP Office's WAN port. IP Office employs a Frame Relay Assembler Disassembler (FRAD) to allow voice and data traffic to be formatted and framed for a Frame Relay network.

VoIP over a Managed IP VPN

Even though the IP Office operates as a traditional 'circuit switched telephone system' utilizing standard analog and digital handsets, the inclusion of an integrated Voice over IP (VoIP) Gateway allows significant cost savings to be realized by converging voice and data onto a single managed IP VPN.

A managed IP network or IP VPN is a private network of routers managed and partitioned by a single network service provider who assigns IP addresses and manages the network. Because of this the network service provider can guarantee throughput levels, minimize latency and ensure transmission speeds to give greater quality of service supported by a contracted service level agreement.

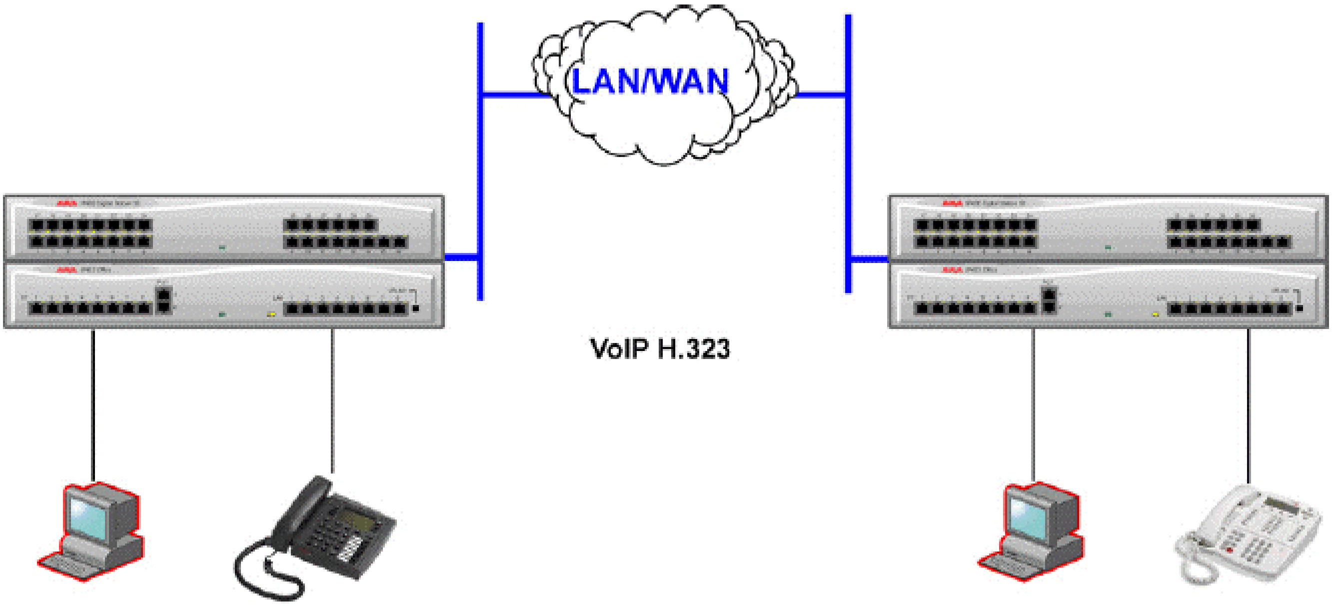
IP VPNs have some distinct advantages over Frame Relay networks: access bandwidth need not be pre-allocated between sites like Frame Relay's PVCs, they are generally cheaper and their global reach is normally greater. Access to the IP VPN is via one of IP Office's LAN ports.

- **Note:** Avaya do not recommend networking IP Office systems over a public, unmanaged, IP VPN where service levels cannot be guaranteed by the provider.

VoIP across the LAN

In a factory or campus environment, voice calls can also be linked utilizing 10/100 Mbps LAN connections, which can be copper or fiber. This is again facilitated by the optional Voice Compression Module (VCM).

In order to avoid bandwidth issues VoIP across the LAN will require some form of bandwidth management.



VoIP networking across the LAN

VoIP across the Public Network

Traditional circuit switched telephony over the public telephone network is restricted in the level of feature support that can be offered. By deploying VoIP over T1/E1/PRI, IP Office is unique in realizing the benefits of Q.931 and H.450 supplementary service support.

Details of Q.931 and H.450 feature support is given below within 'Supplementary services within IP networks'.

Supplementary Services within IP Networks

Supplementary services within an IP environment are provided via Q.931 and H.450. IP Office provides the same rich services as enjoyed within a traditional network environment. Our standards based approach allows interoperability within mixed vendor networks.

Features supported by H.450 on IP Office to IP Office IP trunk links are listed below. On IP trunks to non-IP Office systems the Supplementary Service will depend on those also supported by the non-IP Office system:

- Basic call set up (voice).
- Call Hold.
- Call Transfer.
- Called/Calling Name.
- Called/Calling Number.

Small Community Networking

When connecting IP Offices together over IP or Packet based networks, Small Community Networking enhances feature transparency. These networks can support up to a maximum of 500 users across 16 sites. The following additional features are available.

- Busy Lamp Field.
- Camp-on.
- Call Back When Free.
- Paging.
- Call Pick-up.
- Centralized Voice Mail (Voicemail Pro).
  Support for mailboxes, call recording, dial by name and auto attendants. Queuing on remote systems is not supported.
- Internal Directory.
- Absent Text Message.
- Anti-Tromboning.

If larger networks are required QSIG can be used to link multiple Small Community Networks together. Functionality between the communities is governed by the QSIG feature set.

Generic Networking Features

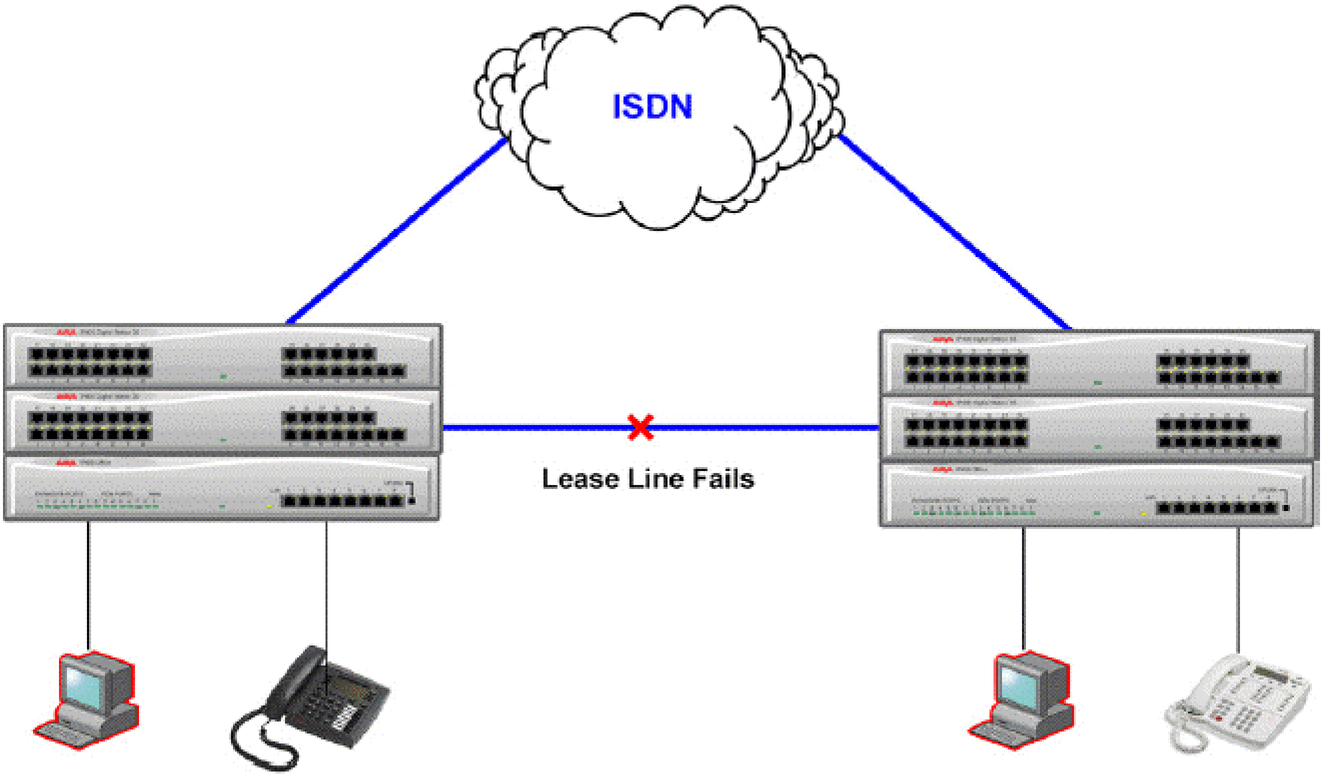
Least Cost Routing (LCR)

By configuring a Least Cost Route calls may be routed via an alternative carrier. Time profiles can also be used to allow customers to take advantage of cheaper rates at specific times.

Multiple carriers are also supported. For example, if local calls and international calls are to go through one carrier between specific hours, all calls to a specific country through an alternative carrier and all other calls via a third carrier. Carrier selection using 2 stage call set up via in-band DTMF is possible.

Alternate Call Routing (ACR)

Alternate Call Routing allows calls to be placed via an alternative route should the primary route fail or be unavailable through congestion etc. ACR is compatible with LCR and VoIP and can be configured to 'take' data channels for voice calls while preserving the data call, albeit with reduced bandwidth.

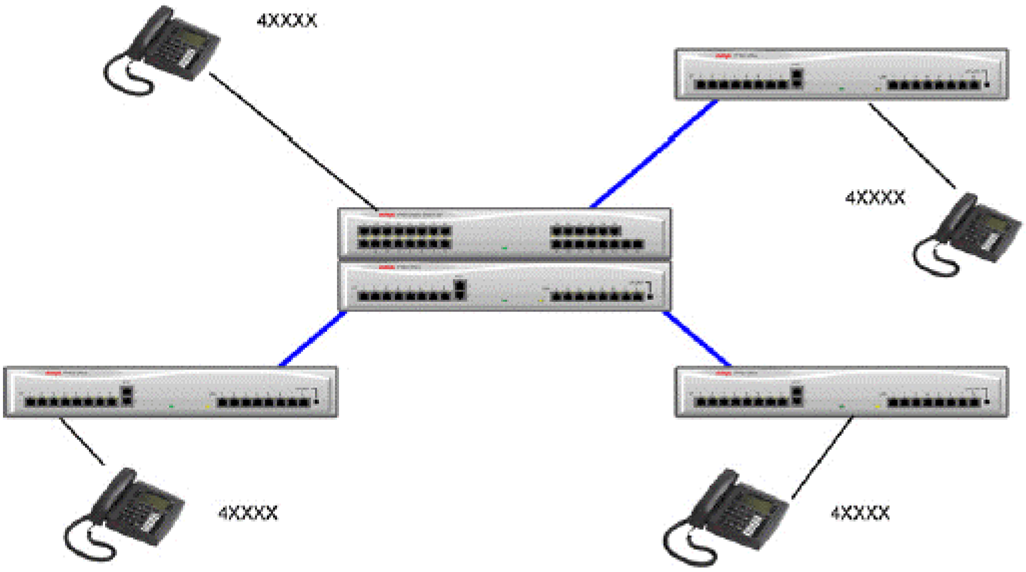


Alternate Call Routing

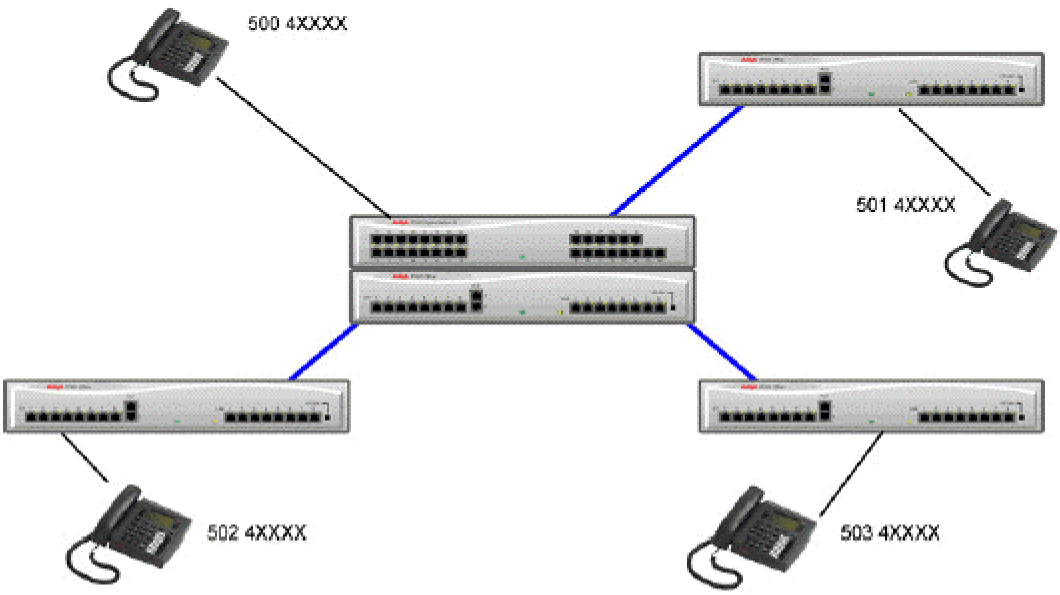
Network Numbering Schemes

IP Office employs fully flexible network numbering options. Dialed digits can be manipulated seamlessly to add and delete digits, access codes etc. in order to fit into any numbering scheme. Two types of numbering schemes are commonly deployed - 'Linked Numbering' and 'Node Numbering' schemes. In linked numbering schemes each site within the network has a unique range of extension numbers and users simply dial the extension number of the called party. Often, linked numbering schemes are used in very small networks (< 5 sites) with less than 500 extensions. With node numbering schemes each site is given a node ID and this is prefixed by the user when dialing extensions at other sites. In this way extension numbers can be replicated across sites while still appearing unique across the network. Node numbering schemes are common in larger networks. Linked numbering schemes and node numbering schemes are sometimes both employed within the same network with node numbering employed at the large offices and linked numbering employed at clusters of satellite offices.

The following figures depict these two types of numbering schemes.



Linked Numbering Scheme



Node Numbering Scheme

8. LAN / WAN Services

LAN / WAN Services

Computers in an office communicate via the LAN (Local Area Network). This at its simplest may be a length of coax cable connecting all the computers, or by twisted pair cables going into a central hub unit. The IP406 V2 supports an 8 port Layer 2 Ethernet switch. The Small Office Edition supports a 4 port Layer 2 ethernet switch with a fifth Ethernet port as a Layer 3 switch. The IP412 support a 2 port Layer 3 Ethernet Switching.

When computers communicate they do not care where the destination is. Their task is simply to pass the packet to the next machine and then forget about it. Where the destination is on another network, the router is the "gateway" to the rest of the world and its job is to cope with that traffic. The router alleviates the need to establish and hold the call for the duration of a remote communication session by automatically establishing a connection only when data is to be passed. Routers may be connected together using WAN (Wide Area Network) links that could be point-to-point leased lines, managed IP networks, Frame Relay networks or exchange lines (Central Office). IP Office platforms support all of these types of network connections.

The IP Office has a Wide Area Network (WAN) port that can be connected to a digital leased line service using either X.21 or V.35 interface at speeds up to 2048kbps. Point-to-Point protocol (PPP) is used over this link. The data within the call uses the Point-to-Point Protocol (PPP) which is used by the vast majority of manufacturers for linking routers. PPP support is essential if it is not the same manufacturer's equipment at each end of the link.

Exchange lines (Central Office) can also be used in the event of failure of the WAN link or to provide alternate or top up bandwidth on demand.

All IP Office platforms have an integral router with support for bandwidth on demand that allows the negotiation of extra bandwidth dynamically over time. IP Office initiates extra calls between sites only when there is data to be sent or sufficient data to warrant additional channels. It then drops the extra channels when they are no longer needed. The calls are made automatically, without the users being aware of when calls begin or end. The rules for making calls, how long to keep calls up etc, are configurable within IP Office.

It is possible to have several different routing destinations or paths active at any time linking the office to other offices and the Internet simultaneously.

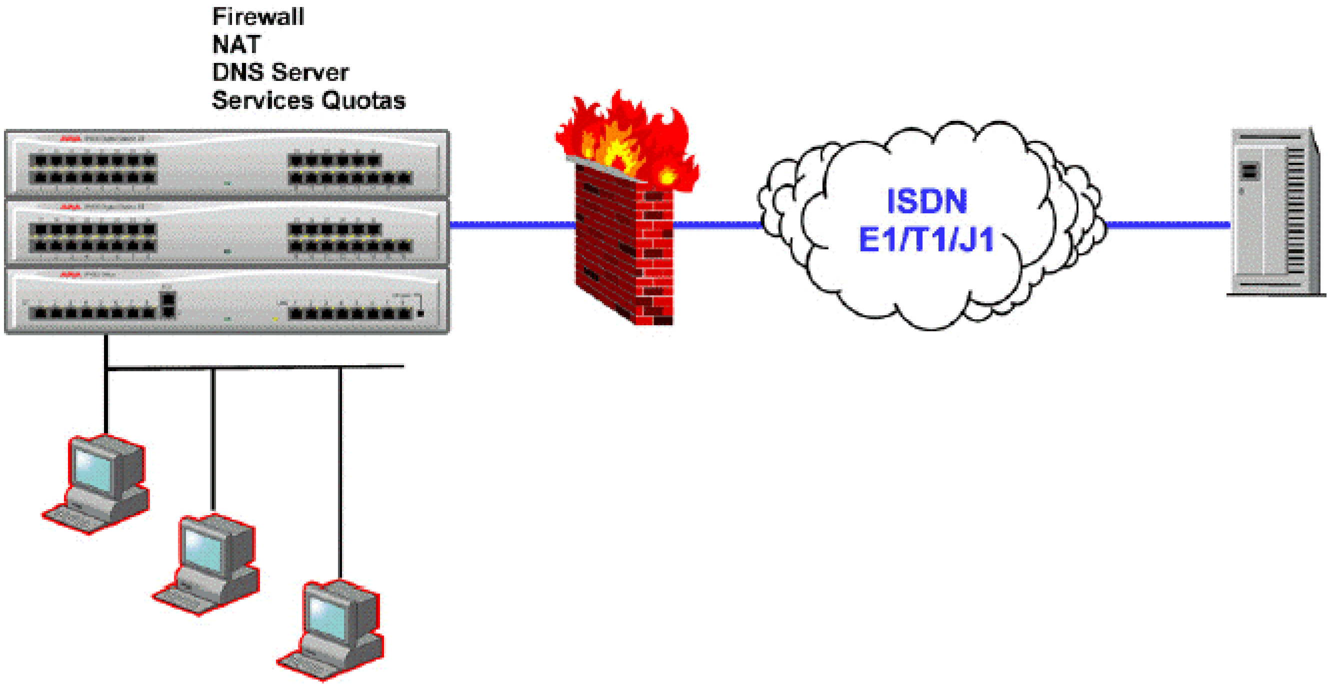
Internet Access

While the telephone is still the number one business communication tool, Internet access is becoming increasingly important for business-to-business communications. The ability to send and receive email, is now considered mandatory when dealing with many suppliers and customers, while access to the World Wide Web for e-commerce applications and information has become vital.

All IP Office systems provide shared, secure, high-speed access to the Internet via exchange lines (Central Office) or digital leased line services.

Internet security concerns are addressed through the provision of an integral firewall. This removes the need for an expensive, standalone, software solution tying up another PC. The firewall can be configured to cater for a variety of situations and will allow customers to control who can access external resources, and when. This isolates your private networks from the Internet, thereby ensuring that your network remains beyond the reach of hackers, while configurable service quotas can be set against the service to ensure that it is not abused. Service Quotas place a time limit on outgoing calls to a particular IP Service. This avoids excessive call charges either because of user abuse e.g. excessive surfing or when something changes on your network and call frequency increases unintentionally.

Each service can be configured with an alternative fall back service. For example, you may wish to connect to your ISP during working hours and at other times take advantage of varying call charges from an alternative ISP. You could, therefore, set up one service to connect during peak times and another to act as fallback during the cheaper period.



Internet Access

Remote Access Features

IP Office's integral firewall, service quotas and timebands all apply to remote access calls. Remote access security is supplemented by CHAP (encrypted passwords) to verify the end users (preferred) or PAP which does not support encryption. Timebands can control the hours within which the remote access service is available.

A "trusted location" can be set. These are locations that the System will allow either data access, e.g. a user dialing in from home, or access to voicemail without a voicemail code for a user collecting their voicemail messages from a mobile. The trusted location is also the location the Voicemail Server will call to inform the user of a new message.

Conversely a "specified location" can be set which restricts remote access from only that location, this specified location can also be a designated dial back number thereby minimizing the threat of unauthorized remote access.

IP Office systems can also incorporate remote access dial back services so that if a user always remotely accesses the office from a single location e.g. their home, then after logon verification the system will disconnect their call and dial them back. In addition to the added level of security dial back provides it can also be an excellent method of consolidating remote access charges onto the central office telephone bill instead of employing expensive free phone services.

In addition to remote access from Telephone Adaptors, an optional V.90 56Kbps modem module can be added to provide dial-in/dial-out to/from users equipped with analog modems. Also as standard, all ATM4 trunk cards and Small Office Editions analog trunk ports support switching of the first analog trunk to an integral V.32 modem for remote maintenance.

LAN to LAN Routing

Gone are the days when an office could exist in isolation without a requirement to transfer bandwidth hungry LAN traffic. Whether it's a requirement to share resources such as email servers, file servers and internet gateways, or seamlessly transport data between sites or network to and from their customers and suppliers, all businesses now have a need for data routing and this is why each IP Office platform offers IP routing as standard.

Embedding a router within IP Office removes the costs, complexity and additional points of failure of external WAN multiplexors by allowing data and voice traffic to converge and share the network resources of IP Office. These network resources can range from dial up ISDN connections, point-to-point leased circuits, managed IP networks or Frame Relay as IP Office supports all these types of network connections.

The IP Office has a Wide Area Network (WAN) port that can be connected to a digital leased line service using either X.21 or V.35 interface at speeds up to 2048kbps. Point-to-Point protocol (PPP) is used over this link. The data within the call uses the Point-to-Point Protocol (PPP) which is used by the vast majority of manufacturers for linking routers. PPP support is essential if it is not the same manufacturer's equipment at each end of the link. Exchange lines (Central Office) can also be used in the event of failure of the WAN link or to provide alternate or top up bandwidth on demand.

All IP Office platforms have an integral router with support for bandwidth on demand that allows the negotiation of extra bandwidth dynamically over time. IP Office initiates extra calls between sites only when there is data to be sent or sufficient data to warrant additional channels. It then drops the extra channels when they are no longer needed. The calls are made automatically, without the users being aware of when calls begin or end. The rules for making calls, how long to keep calls up etc, are configurable within IP Office.

It is possible to have several different routing destinations or paths active at any time linking the office to other offices and the Internet simultaneously.

Data Networking Features

Integral 10/100 Mbit Layer 2 Ethernet Switch

- **IP Office - Small Office Edition & IP406 V2 Only.**

All the IP Office - Small Office Edition platforms provide a four port Layer 2 Ethernet Switch. The IP406 V2 provides an 8 port Layer 2 Ethernet switch.

Each port auto-senses its operational speed, 10M or 100M. In addition to the four port layer 2 switch, IP Office - Small Office Edition has a fifth Ethernet port (labeled WAN) with its own IP Address (LAN2) intended for connecting to external xDSL or Cable Modems. This fifth port is a Layer 3 switch to the other four ports.

Integral 10/100 Mbit Layer 3 Ethernet Switch

- **IP Office - Small Office Edition & IP412 Only.**

The IP412 supports a two-port Layer 3 Ethernet switch. Both of these switched ports have their own IP addresses (LAN1 and LAN2). In order for traffic to pass from one port to the other a route is configured in the system's routing tables.

Additionally, it is possible to set up a firewall between the two LAN segments. The IP Office - Small Office Edition offers similar functionality between its four port Layer 2 Ethernet switch and its Layer 3 Ethernet WAN port.

Layer 3 switching is particularly useful in situations where it is desirable to have a 'trusted' and 'unsecured' network, where the 'unsecured' network is uncontrolled and carries public traffic on it.

DHCP Server

IP Office can manage your IP Network for you through its integral DHCP Server. IP Office can be configured to hold a pool of IP addresses for users on the Local Area Network. When a user powers up their PC, the system will allocate them an IP address for the duration of their session. The DHCP server also provides the user's PC with the address of the Domain Name Service (DNS) server and the Windows Name Service (WINS) server. Alternatively, for customers who have a separate DHCP Server, IP Office can be configured to obtain its address from that server or be set with its own dedicated static address. The IP Office - Small Office Edition & the IP412 have two independently controlled DHCP servers, one dedicated to each of the Layer 3 switched LANs.

Leased Line Support

All platforms are capable of connecting to leased line services. Six physical types of Leased Line are supported X.21, V.35 and V.24, via the WAN port, or E1/T1 and Basic Rate via the trunk interfaces on the base unit. The X.21, V35 and V24 are externally clocked and can operate at any speed up to and including 2M. E1/T1 trunks can be configured to operate in a fractional mode for 'point to multi-point' applications i.e. a single 2M interface could be treated as 3 x 512K and 8 x 64K going to 11 different locations. When using T1 as a Leased Line it is possible to use the same circuit for switched circuit services. Not all types of leased line are available in all territories, check for availability.

Dial-Up Circuit Support

Where the amount of traffic does not justify the cost of a dedicated leased line, the system can provide data connectivity via dial-up circuits using its E1/T1 or Basic Rate trunks. Where speeds greater than a single channel are required (64K/56K), additional channels can be added to the call as and when they are needed.

Point-to-Point Protocol (PPP)

PPP is an industry standard Wide Area Networking Protocol, that allows inter-working with a wide range of 3rd party routers. PPP is used over dial-up or leased line circuits where a single channel is used to connect the two locations together. e.g. A single channel maybe a 64K channel on a dial-up circuit or a 256K leased line etc.

Multi-Link Point-to-Point Protocol (ML-PPP)

IP Office supports Multi-Link PPP allowing additional calls to be made where bandwidth greater than a single channel is required. The maximum number of channels available to data can be set on a service-by-service basis. When the available bandwidth reaches a user defined limit additional channels can be automatically added. Similarly, when traffic falls then the number of channels in use can be automatically reduced. If there is no data traffic on any of the channels in use then all lines can be cleared. Since most carriers have a minimum charge for calls, the period that a channel has to be idle before clearing is configurable. Through these mechanisms call costs can be effectively controlled while ensuring that bandwidth is available as and when it is needed.

Frame Relay

Frame relay is a wide area networking protocol loosely based on ideas borrowed from the X.25 protocol. Individual network connections are multiplexed over a common medium by the use of Permanent Virtual Circuits (PVC). This allows a single Leased Line to provide connectivity to a number of different locations. Frame relay is currently implemented in IP Office as a CPE or 'router end' protocol over WAN connections. IP Office supports both PPP and RFC1490 encapsulation with fragmentation of large data packets to provide voice quality of service.

Service Quotas

IP Office allows a user to define the maximum number of minutes that a service, such as Internet Access, is available for. This is the sum total of calls made and does not include periods of inactivity. Once the quota has been used the service is no longer available. The quota can be either automatically refreshed daily, weekly or monthly or manually refreshed by dialing a secure feature code on a handset.

Time Profiles

Time profiles set the operational hours and days of a service. For example this would allow a customer to make Internet Access available to staff only during lunch times. Using time profiles it is also possible to define an alternative service to operate outside the operational hours of the main service. This may be used to take advantage of alternative tariffs at off peak periods. Switching to this fallback service can also be controlled manually by dialing a secure short code from a handset. This can be particularly useful in allowing quick restoration of service in the event of an ISP failure.

Bump Call

If a data call is using more than a single channel, this facility allows the system to reallocate a line to a voice call when all other lines are busy. If the data call is only using a single line the call cannot be bumped.

Password Authentication Protocol (PAP)

PAP is a method of authenticating the remote end of a connection using unencrypted passwords.

Challenge Handshake Authentication Protocol (CHAP)

Challenge Handshake Authentication Protocol allows an incoming data call to be authenticated using encrypted passwords. The system also provides the option to periodically reaffirm the authenticity of the caller during the data call.

Data Header Compression

IP Header Compression (IPHC) reduces the header size of the data packet to gain bandwidth efficiency over Wide Area Networks.

Data Compression
IP Office supports both Microsoft Point to Point Compression and Stac Lemple Ziv to provide greater throughput on slow speed wide area network links.

Bandwidth Allocation Control Protocol (BACP)
Bandwidth Allocation Control Protocol allows the negotiation with the other party of the data call to request additional calls to be made to improve data throughput.

Callback
Three types of call back are supported

- **LCP (Link Control Protocol)**
  After authentication the incoming call is dropped and an outgoing call is made to a predefined number to re-establish the link.
- **Callback CP (Microsoft's Callback Control Protocol)**
  After authentication from both ends, the incoming call is dropped and an outgoing call to a predefined number made to re-establish the link.
- **Extended CBCP (Extended Callback Control Protocol)**
  Similar to Callback CP however, the Microsoft application at the remote end will prompt for a telephone number. An outgoing call will then be made to that number to re-establish the link.

Domain Name Service (DNS) Proxy
Domain Name Service servers provide the translation of familiar names such as www.avaya.com to the IP address required in order to establish a connection. IP Office provides this service to PCs on the network by proxy.

Network Address Translation (NAT)
NAT is a mechanism that allows you to use a different IP address to that of your internal network while connected to an external party or service. When connecting to the Internet, ISPs typically want a customer to use an IP address they have allocated. Using NAT this is easily accommodated, eradicating the need for the customer to change their network numbering scheme.

Typically, a company maps its internal network addresses to a global external IP address and unmaps the global IP address on incoming packets back into internal IP addresses. This helps ensure security since each outgoing or incoming request must go through a translation process. This also offers the opportunity to qualify or authenticate the request or match it to a previous request. NAT also conserves the number of global IP addresses that a company needs.

Proxy Address Resolution Protocol (ARP)
Support for Proxy Address Resolution Protocol allows IP Office to respond on behalf of the IP address of a device connected to it when receiving an ARP request.

Auto Connect
If a service is idle, that is no one is using the Internet, Auto Connect allows the IP Office to periodically connect to a service. This is ideal for mail polling to retrieve email from an Internet Service Provider. An 'Auto Connect Time Profile' controls the time period during which automatic calls are made, for example not at weekends or during the middle of the night.

Firewall
The integrated firewall provides an easy point and click configuration allowing the filtering of the most common IP protocols including File Transfer Protocol (FTP) and Internet browsing (HTTP). Each protocol passing through the firewall can be restricted/allowed access in four different ways:

- **Drop**
  No sessions via this protocol will be allowed through the wall
- **In**
  An incoming session can "punch a hole" in the wall to allow traffic in both directions
- **Out**
  An outgoing session can "punch a hole" in the wall to allow traffic in both directions
- **Bothway**
  An incoming or outgoing sessions can "punch a hole" in the wall to allow traffic in both directions.

In cases where a protocol is not supported by default, the firewall can be customized to control packets based on their content.

IP Office allows the configuration of as many firewalls as needed. This permits different security regulations to be applied to individual dial-in users and data services.

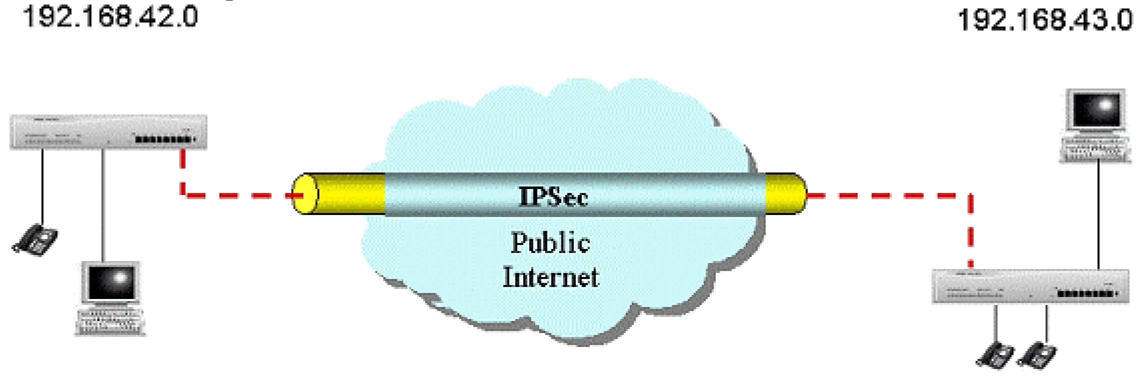
Light-Weight Directory Access Protocol (LDAP)
IP Office supports LDAP directory synchronization. This allows the telephone number Directory (names and telephone numbers) held in the main unit to be synchronized with the information on an LDAP server (limited to 500 entries). Although targeted for interoperation with 'Windows 2000 Server Active Directory', the feature is sufficiently configurable to interoperate with any server that supports LDAP version 2 or higher.

Remote Access Server (RAS)
IP Office provides RAS functionality allowing external users to dial in to the local area network from modems, telephone adaptors and routers. Several of the previously described features and services can be applied to the dial-in users to create a powerful Remote Access Server. Dial-in users can be authenticated using either PAP or CHAP. Once authenticated the DHCP server can automatically assign the user an IP address to use while connected to the LAN. Individual time profiles and firewalls can be applied to the user restricting what they have access to and when they have access. For further security and accounting ease, IP Office can automatically call a user back. This keeps the cost of the telephone call on the company telephone bill removing the need to process individual expense claims.

Transaction Packet Assembler Dissembler (TPAD)
TPAD is a lightweight version of the X.25 protocol used in the retail market for transaction processing. Through faster transaction processing a retailer can reduce the floor limit of credit authorizations and benefit from lower transaction charges. A PDQ or credit card "swipe" telephone can utilize the digital trunks, via the DTE port or the USB on the rear of the IP Office. Since the link between the main unit and the transaction authenticator is digital no modems are required at either end.

Routing Information Protocol (RIP)
RIP is a distance vector protocol that allows routers to determine the shortest route to a destination network. It does this by measuring the number of intermediary routers that need to be traversed to reach the destination network. If more that one route exists to the same destination the shortest route is used. If a fault occurs on the shortest route it will be remarked as being infinite and any alternative route will become the new shortest route. This behavior can be used to add resilience into a data network. Where a customer has an existing data network comprising of third party routers, IP Office added to the network can provide back up using its routing and dial-up capability. RIP enabled routers share their knowledge of the network with each other by advertising and listening to routing table changes. IP Office Supports both the RIP I and RIP II standards.

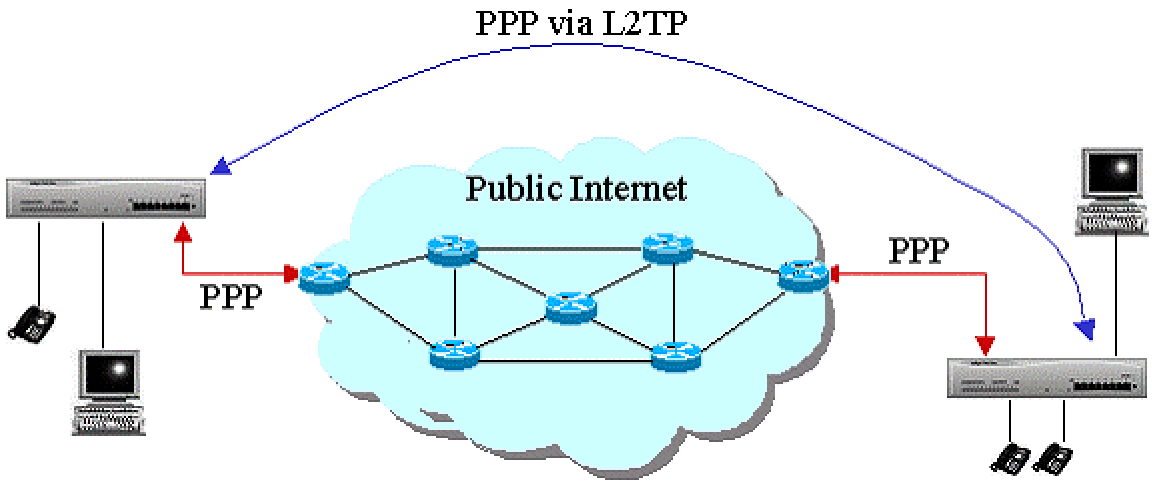
VPN: IPSec Tunneling



IPSec tunnels allow a company to pass data between locations over unsecured IP networks such as the public internet. The corporate data is secured using 3DES encryption making it unintelligible to other parties that might be 'eaves dropping' on the traffic. Tunneling can be applied to link offices together or provide workers access to the office over the internet. All Platforms support up to a total of 256K worth of encrypted traffic to multiple locations. Initially, inter-working is supported only between IP Offices that are connected either directly on a WAN port or via the LAN using a 3rd Party router. IPSec is enabled on IP Office through a License Key.

Note: Check with Avaya for supported scenarios and 3rd party devices.

VPN: Layer 2 Tunneling Protocol

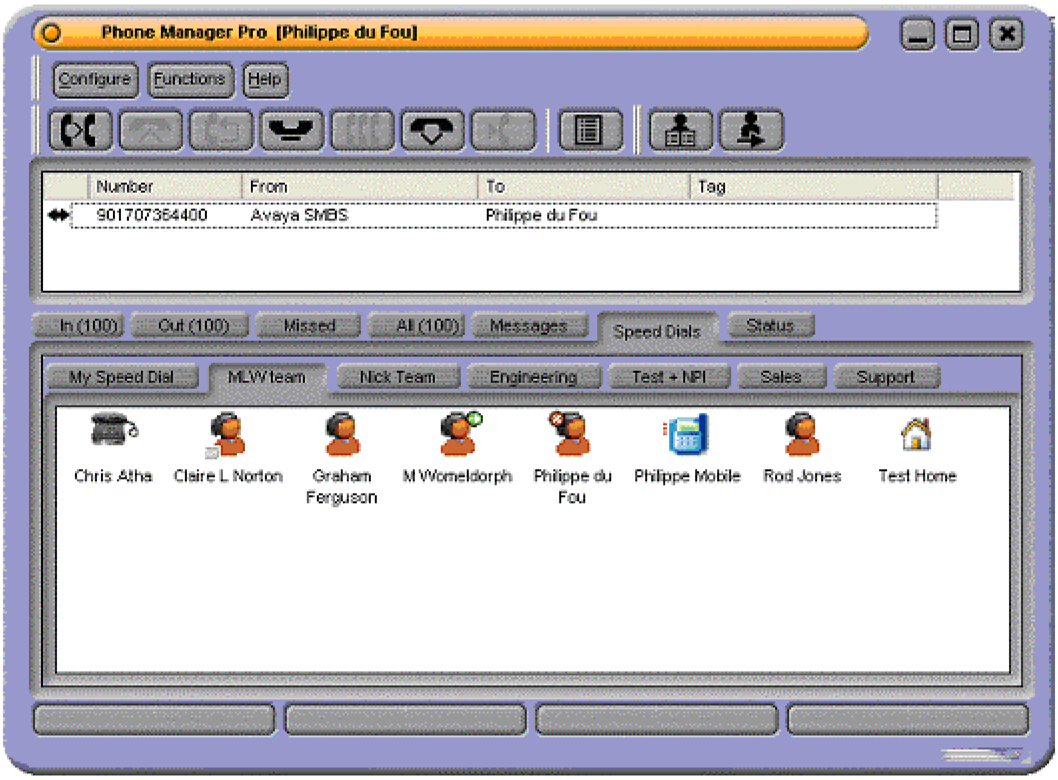


PPP authentication using PAP or CHAP takes place between directly connected routers only. When using a public IP Network to connect sites this authentication takes place between the customers router and the service provide router that it is connected to. In some circumstances it is desirable to authenticate between the customer owned routers, jumping over all the intermediary routers of the service provide network. Layer 2 Tunneling Protocol allow this to happen by facilitating a two stage authentication, firstly with the service provider router then the customer router on the remote network.

9. Phone Manager

Phone Manager

The Phone Manager application offers control of the user's telephone from their PC. Phone Manager is available in three versions; Phone Manager Lite, Phone Manager Pro and Phone Manager PC Softphone VoIP mode).



For Phone Manager Lite and Phone Manager Pro, this is similar to the PC-based IP Soft-phone except that the conversation actually takes place via a standard telephone telephone rather than the PC's soundcard. The Phone Manager PC Softphone adds PC-based telephony via a sound card or USB headset/handset to the product. See Section 6, IP Telephony for details.

Phone Manager Lite

Phone Manager Lite allows employees to take control of their phone and calls from their networked PC. It can be used with an analog telephone, a digital phone or IP phone for the speech path.

Caller ID/Name Presentation

CLI/ANI is presented as standard so you can see who's calling you before you even answer. The caller's phone number and name (if known to IP Office) are clearly shown on your PC, allowing you to have a good idea what the call's about before you take it. Also shown is information on the actual number dialed, this could be your own Direct Dial In (DDI/DID) number, or a specific department within your organization, e.g. switchboard, sales, support or administration. This feature allows you to answer accordingly and gives you the flexibility to participate in multiple groups, particularly important for small businesses. The same information is also displayed should a second call come-in, allowing you to easily switch between calls or allow the second call to go to voicemail. You can choose to have the information pop-up on your PC automatically as soon as a call comes in, when you answer the call, or it can be instigated manually via a click of your mouse.

Call History

Phone Manager's call history keeps a record of all your received, outgoing and missed calls. Double-clicking on any record calls that number. It even alerts you when you've received a new voicemail and presents unread voicemails so they can be simply retrieved by a click of a button.

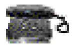
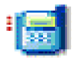
Desktop PC Telephony Controls

Phone Manager offers telephony buttons to activate standard functions such as Answer, Transfer, Hold, Account codes and Conference so you do not need to remember any specific feature codes. You can also elect to forward your calls and easily edit the forward destination using Phone Manager.

Calls can be easily parked using "drag & drop" functionality. Four Call Park areas, which can be shared between users and operators, or within a department, further add to the ease with which the entire call handling process is streamlined with Phone Manager.

Personal Productivity & Collaboration

Phone Manager features a Busy Lamp Field and Direct Station Select. This allows users to customize the application to reflect the status of their department, immediate colleagues or the whole company as desired. The Direct Station Select allows you to dial regularly used internal and external numbers via a single-click. A single Direct Station Select icon allows you to dial their work, mobile/cell phone and home numbers. The Busy Lamp Field feature allows you to see at a glance, who's available to take a call, who's already on a call and who's placed their phone on Do Not Disturb.

Internal User	Work Phone	Mobile/Cell Phone	Home Phone
			

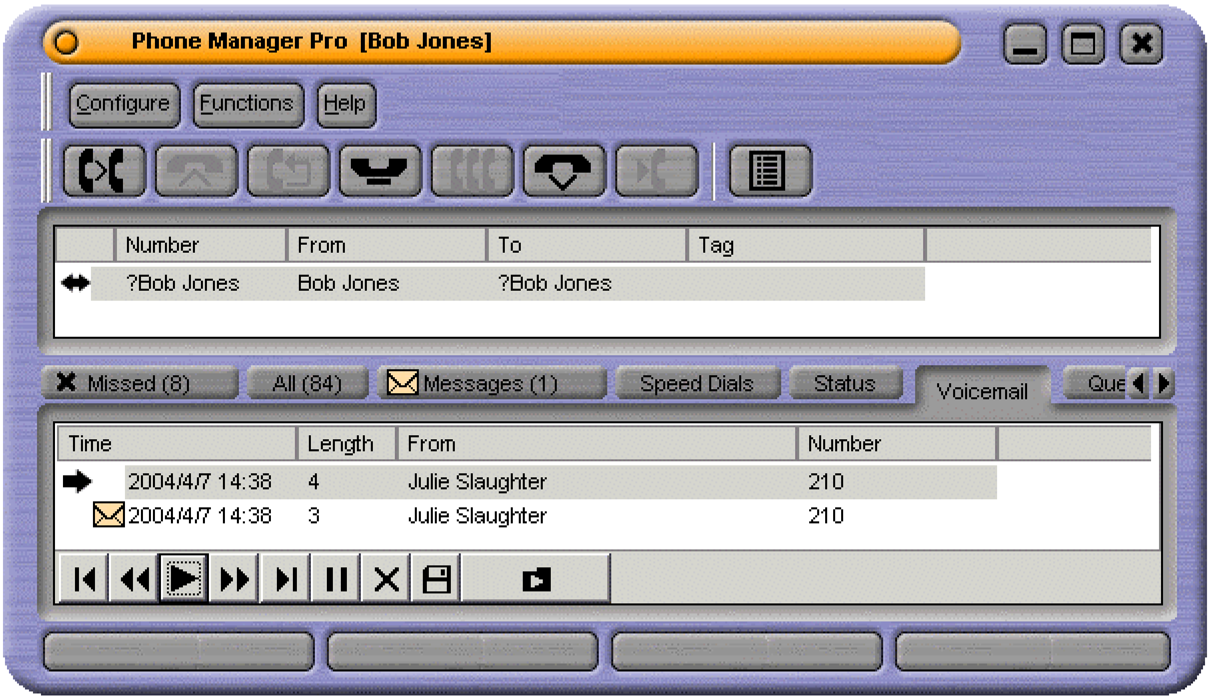
Where Microsoft Live Communications Server (LCS) is also available, Phone Manager users can view colleague's presence (online, away, offline) as well as send Instant Messages (IM). For example you could send an IM to alert a colleague that an important call is waiting for them even though they're busy on a call.

Phone Manager also offers Conferencing Center toolbar buttons that allow users to book a conference or join a web conference. Note: The booking feature is only available if permission is specified by your system administrator and Conferencing Center has been installed (see the Conferencing Center section for further details).

Phone Manager Pro

Phone Manager Pro builds upon Phone Manager Lite by offering the following additional features:

- **Integration with Contact Management packages** (such as Outlook, GoldMine, ACT! and Maximizer) to facilitate screen popping of the contact details of an incoming caller, dialing from the contact record with a simple mouse click and simple creation of new contact records with auto-insertion of the telephone number while on a call.
- **Voicemail box control** with Voicemail Pro, in either Intuity or IP Office modes, which allows you to play, rewind, fast forward, save or delete your voice messages. It also allows Pro users to configure their Personal Distribution Lists (Intuity mode).

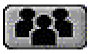
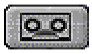


- **Personal phone number directory** which allows further personalization and improves productivity:
    - **Name matching:** If the Caller ID is recognized in the local PC directory, the caller's name can be displayed. Up to 1000 entries.
    - **Simple incoming call scripting:** Scripts can be displayed based on the Caller ID or the dialed number (DID/DDI) to remind users of a specific greeting or sales pitch to use.
    - **Distinctive ringing:** Allows the configuring of distinct ringing on a per caller basis. PC sound files can be associated with incoming callers' numbers and then played through the PC speakers when a call is received from that number. This allows you to easily differentiate calls from important customers and clients, and those from unknown callers.

- **Compact mode** minimizes the screen space required to run the Phone Manager application. While in compact mode, a notification slider alerts of new calls and allows the user to view the caller ID or associated caller's name and answer the call. Users can easily switch between standard and compact modes.

- **Agent Mode operation** which allows the user to perform contact center functionality without needing a specially designed contact center telephone, for example one with dedicated keys such as log on/off. You can also easily activate Account codes (during or before the call) through the "Account Codes" tab which allows the user to tag the call with an alphanumeric account code via a single-click. Agent-mode users can set their phone on "Busy" or "Wrap-Up" and select which hunt group they are member of via simple button clicks.

Busy Wrap Up Select Group Membership Busy Not Available Start Call Recording Stop Call Recording

- **Queue monitoring** allows you to monitor the number of calls waiting on up to 2 queues.
- **Door entry control** allows you to remotely activate the two electric switches connected to the IP Office.
- **Time on call** shows call duration.
- Separate tabs for **Incoming, Outgoing and Missed Calls**.
- **Multiple Speed dial tabs** to allow users to group speed-dial/Busy Lamp Field icons by department or location e.g. Sales or Support. Up to 10 tabs, each of up to 100 icons.

Phone Manager Feature Comparison

Feature	Phone Manager Lite	Phone Manager Pro and PC SoftPhone
Inbound/outbound call handling.	Yes	Yes
Phone call control.	Yes	Yes
Configure phone preferences.	Yes	Yes
Configure keyboard short cuts.	Yes	Yes
CLI (ANI) / Name display.	Yes	Yes
Speed dial management.	Yes - 15 icons maximum.	Yes - 100 icons maximum per tab.
Busy Lamp Field (via speed dial).	Yes - 15 icons maximum.	Yes
Multiple Speed Dial tabs (to group Busy Lamp Field icons)	–	Yes - 10 tabs maximum.
Microsoft Live Communications Server (LCS) Integration	Yes	Yes
View internal users' presence via LCS	Yes	Yes
Send Instant Messages (IM) to internal users via LCS	Yes	Yes
Compact mode	–	Yes
Local Phone Directory.	–	Yes - 100 entries maximum.
Call history log – in, out, missed, messages.	Yes	Yes
Separated incoming/outgoing call log.	–	Yes
Collect new voicemail messages.	Yes	Yes
Voicemail box control (Intuity and IP Office modes).	–	Yes
Personal Distribution List set up (Intuity mode)	–	Yes
Incoming call scripting.	–	Yes
Time on call.	–	Yes
Door opening control.	–	Yes
Queue monitoring.	–	Yes - 2 Queues
Conference Control Display.	Yes	Yes
Conferencing Center action buttons	Yes	Yes
'Screen pop' contacts (Outlook, Goldmine, ACT! and Maximizer).	–	Yes
Simple Outlook contact record creation.	–	Yes
Agent Mode.	–	Yes
Distinctive Ringing (WAV file).	–	Yes
Advice of Charge Indication *(this feature is only supported in Greece & Germany).*	–	Yes
Post Connect dial (sending DTMF while connected to another party).	Yes	Yes
VoIP mode (to run as an IP softphone)	–	Option

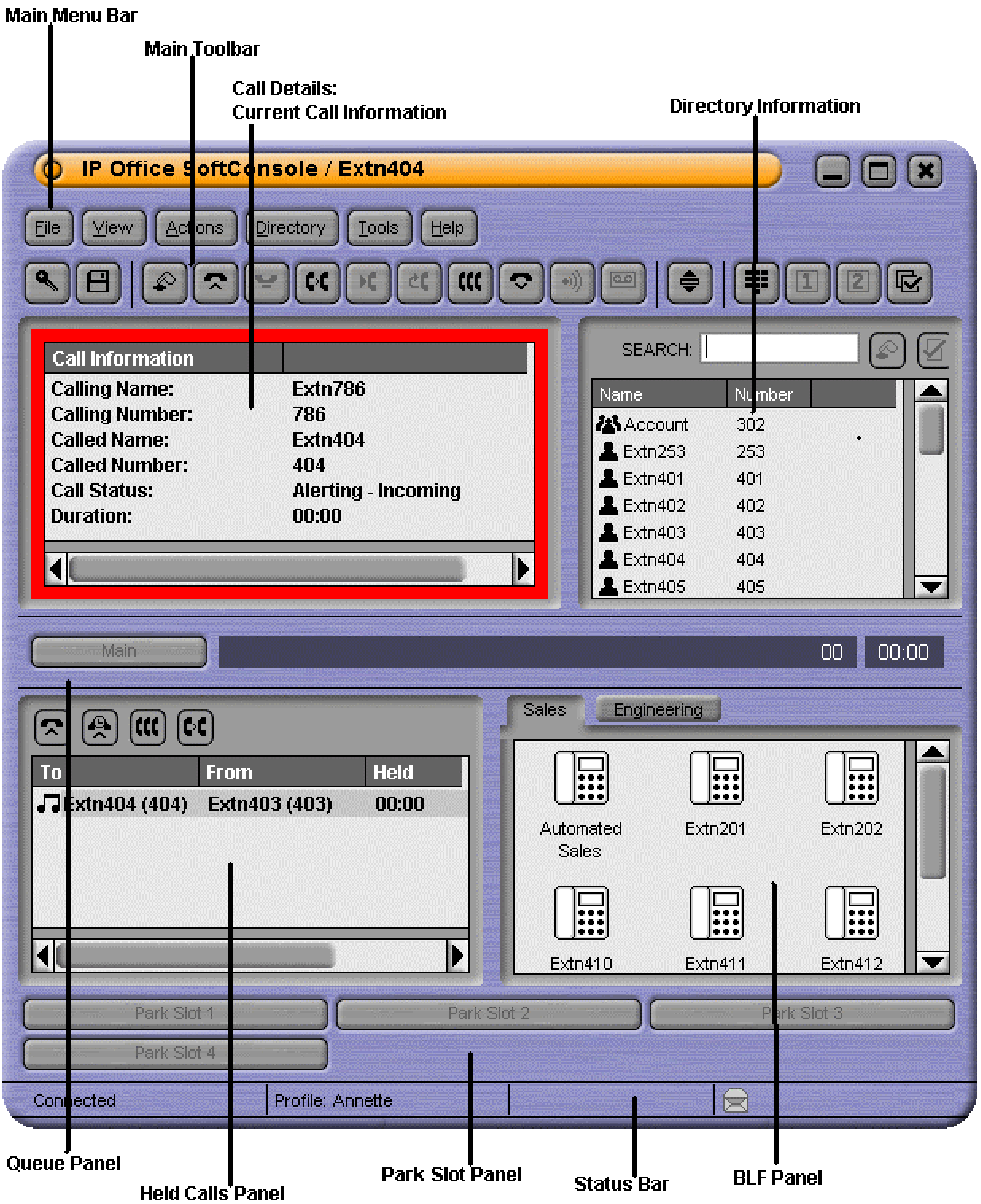
Phone Manager System Requirements

- Any IP Office system and supported telephones. Hands-free operation is only supported on suitable Avaya DS and IP phones.
- Ethernet attached PC running Windows operating system in conjunction with TCP/IP Networking.
  - Windows XP Professional (SP2).
  - Windows 2000 Professional (SP4).
- Phone Manager Lite/Pro: Minimum Pentium 266Mhz or above with 64MB RAM and 50Mb of free disk space (sound card if audio features required)
- Phone Manager Pro PC Softphone: Requires a VoIP license in addition to the Phone Manager Pro user license.
- Optional screen popping integration for Microsoft Outlook 2000/2003/XP, Act! 6.0 and 2005, Maximizer 7.5 and 8.0 Enterprise, Goldmine 6.0 and 6.7.
- Instant Messaging options require the network to have a Microsoft Live Communication Server (LCS) with both a server license and client licenses for each user. Phone Manager was tested against LCS 2003 and 2005. No license is required in the IP Office.

10. SoftConsole

SoftConsole

SoftConsole is the PC based Windows Operator Console for IP Office. SoftConsole has been specifically designed to benefit businesses through improved operator service by providing the operator with the correct information to prioritize call handling and give the appropriate response to the caller. At the same time, the operator can maintain visibility of the number and type of calls waiting and so ensure that clients are greeted in a professional manner, enhancing the image of the company.



SoftConsole has been designed to be easy to use, while offering a look and feel, which will appeal to experienced and novice operators alike.

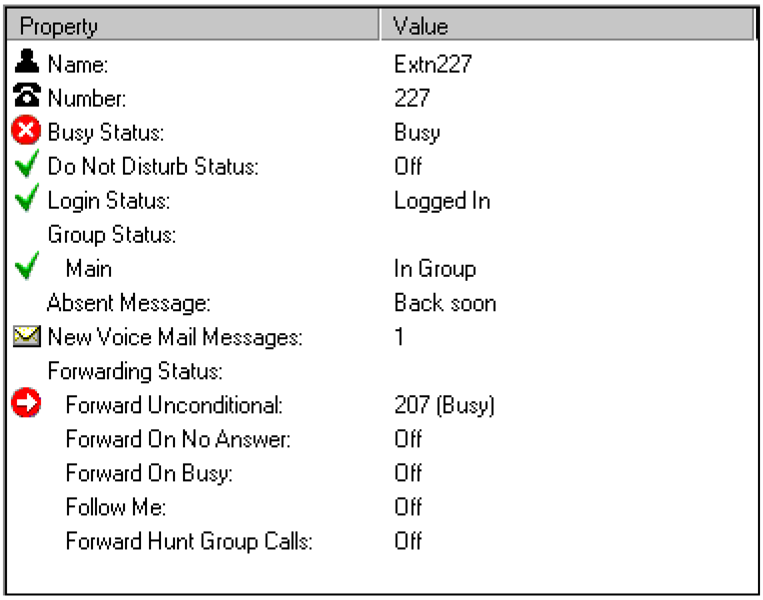
The SoftConsole is divided into the following areas:

- **Main Menu Bar**
  Commands & actions are available through menus. Some items are only accessible when the right conditions occur e.g. when a call is received. The following items available are:
  - Login.
  - Save Profile.
  - New call.
  - Answer call.
  - Hold call.
  - Transfer call.
  - Transfer complete.
  - Reattempt transfer.
  - Conference.
  - Hang up.
  - Page.
  - Record call.
  - Compact view.
  - Dial Pad.
  - Access conference room 1.
  - Access conference room 2.
  - Options.

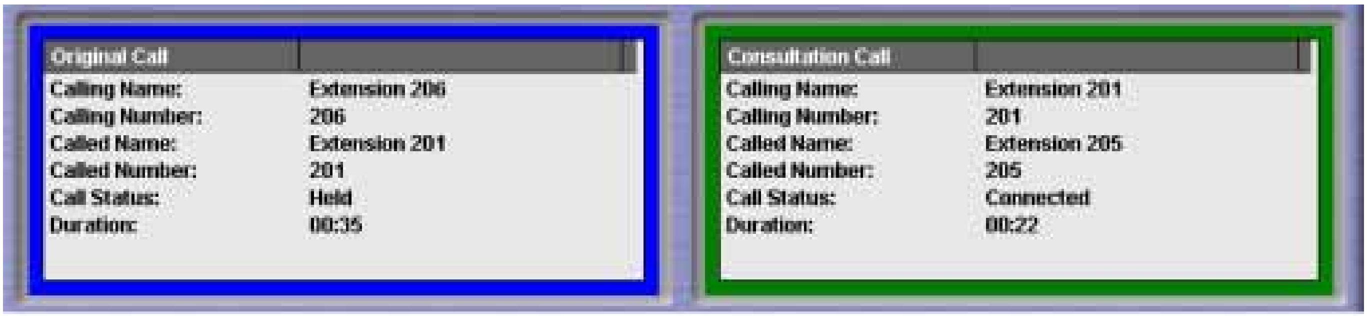
- **Call Details Panel**
  The call details panel on the left shows details of the current call which will include the following information:
  - **Calling Name**
    The system directory name associated with the calling number.
  - **Calling Number**
    The telephone number of the call originator.
  - **Called Name**
    The system user name or hunt group name associated with the called number.
  - **Called Number**
    The extension number the incoming call has been routed to by the system.
  - **Call Status**
    States the progress of a call. The border around the call status panel changes color to indicate the status of the call.
  - **Call Duration**
    The length of time that the has been in the state as indicated by the Call Status
  - **Notes**
    This area displays notes or information about the call i.e. when a call has been returned as there was no answer from the extension it was transferred to. If annotation is attached to the call, details are shown in the Notes area.

  If a new call arrives, the call details panel will display the calls waiting to alert the user and allow answering of the call based on the Caller ID.

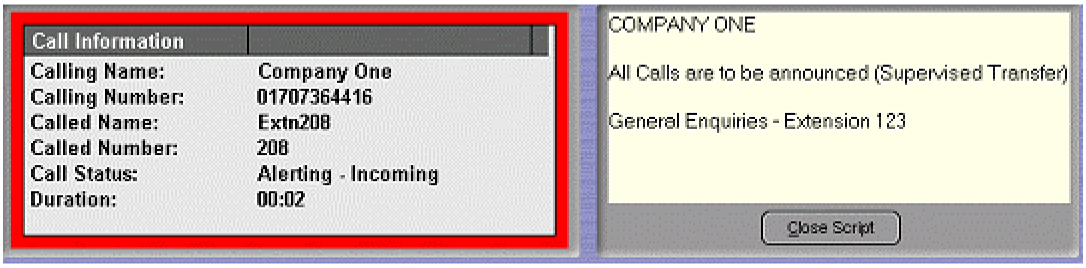
- **Directory Panel**
  The directory panel on the right shows information on following:
  - **Directory entries**
    Including IP Office users, hunt groups and external directory user (non IP Office user)
  - **Single directory entry details**
    Including IP Office users, Hunt Groups and external directory user (non IP Office user).



  - **Details about a consultation call**
    When operator wishes to carry out a supervised transfer.

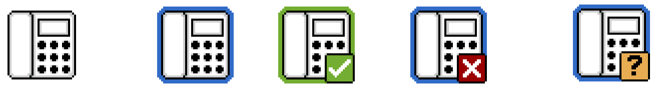


  - **Script**
    When a script has been configured for either the calling or called number, the script is displayed in this panel. For example, an operator may be answering calls on behalf of more than one company. To ensure the call is answered with the correct company name a script file can be created with the company name details. The script is displayed whenever a call is received for that company.

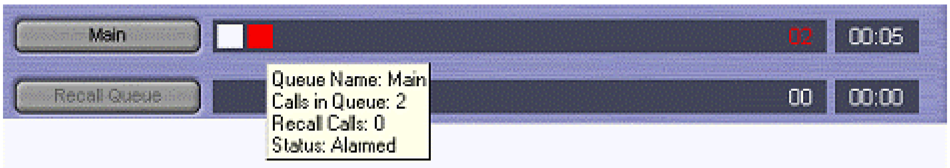


- **Conferencing**
  Within SoftConsole, calls can be conferenced when held, or a conference can be created through the two conference rooms:
  - **Conference Held Calls**
    An operator can conference calls that are in the Held Panel. All calls in the Held Panel will be conferenced.
  - **Conference Room**
    An operator can configure up to two conference rooms including details on who is hosting the conference plus the ability to send out invites to conference participants (automatic invites can be generated in conjunction with Voicemail Pro, see Conferencing Center for more details). Participant status as depicted by icons in conference room:

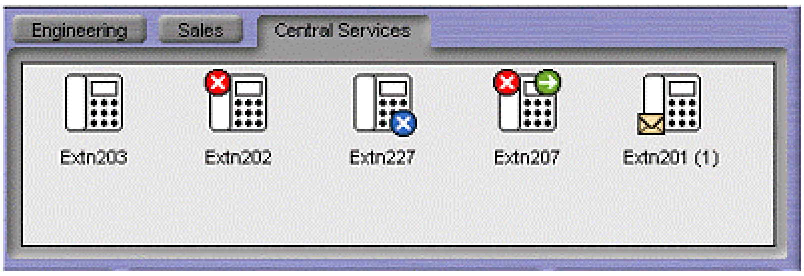
Not Invited. Invited. Joined. Declined. Unavailable.



- **Queue Panel**
  The queue panel displays graphical information via means of a dynamically updated bar graph, on the number and the status of external calls held in a particular queue. Up to 8 Call queues can be configured and labeled to reflect incoming calls for specific Hunt Groups.



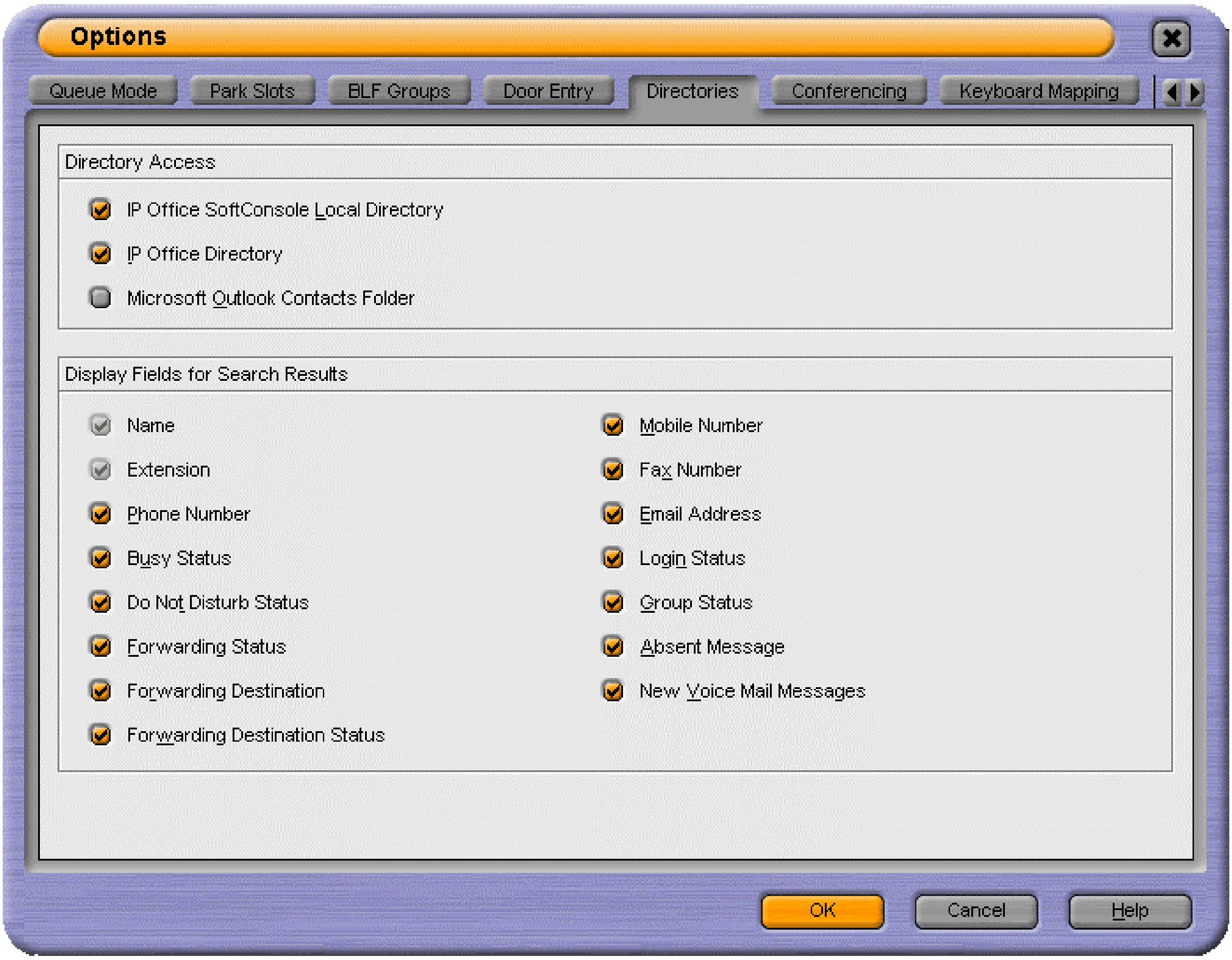
- **Held Calls Panel**
  The held call panel enables the operator to manage all calls held at the operator station. These will appear as a list in panel. The operator can perform the following the functions: Answer the highlighted held call, Answer the longest held call, Conference held calls (see conferencing section above) or Transfer held call
- **BLF Panel (Busy Lamp Field Panel)**
  The BLF panel displays icons to indicate the status of selected users. Each Icon provides information on individual users such as: Unread 'User' voicemail messages, User status information e.g. Busy, DND and Forwarded or Tabs can be configured to indicate different groups of BLF icons.



- **Park Slot Panel**
  The park slot panel can contain up to 16 system-wide park slots with specific Park ID's for each slot.
- **Status Bar**
  Shows current status of the system. The bar is divided into four sections that display: The current connection status, The current Profile name, Information messages e.g. alarm conditions and The number of new voice mail messages for the operator.

SoftConsole Options

SoftConsole has many configurable options available to the operator to personalize the look and feel. The Operator can tailor the usability specifically to each their personal preferences. The following configuration options are available:



- **Incoming Calls**
  This tab enables the operator to manage the local SoftConsole directory by creating, editing and deleting entries from the selected directory. Also the operator is able to associate a script or media file with each specific entry.
- **Queue Mode**
  This tab enables the operator to configure the queue window with up to 8 hunt group queues, which will include a recall queue. Queues can be created, edited and deleted while also providing the operator with the additional benefit of positioning them in the queue window in order of operator preference. Management by exception is employed to monitor queue status by enabling the operator to configure various alarm thresholds such as the Number of calls in queue and Longest waiting call time. Note: a media file can be associated with an alarm.
- **Park Slots**
  This tab enables the operator to configure, which park slots are accessible on a system wide basis up to a maximum of 16. The operator is also able to assign which key sequences are used to access each park slot and where they appear within the park slot panel.
- **BLF Groups**
  This tab enables the operator to create and edit BLF groups.
- **Door Entry**
  This tab enables the operator to configure up to two door entries.
- **Directories**
  This tab enables the operator to configure access to the following directories: SoftConsole local directory,

IP Office directory and Microsoft Outlook contacts. Secondly the operator is able to configure, which fields will be displayed for each individual directory entry.

- **Conferencing**
  This tab enables the operator to set up the names of the two conference rooms. This name will appear on the telephone displays of users in the conference room (maximum of 10 characters).
- **Keyboard Mapping**
  This tab enables the operator to assign short cut keys for SoftConsole functions.
- **Keyboard Actions**
  This tab enables the operator to specify the default action when alphabetic or numeric characters are entered.
    - Alphabetic Keystrokes: Begin directory search or Open call annotation window
    - Numeric Keystrokes: Begin directory search or Open pop-up dial pad
- **Appearance**
  This tab enables the operator to change the appearance of SoftConsole fonts, skins and the call information window color.
- **Save**
  This tab enables the operator to save the changes made to the configuration of SoftConsole either automatically or manually.

SoftConsole Administration

SoftConsole has an administration mode that enables the operator to configure the following settings:

- **Change and create templates**
  SoftConsole comes with three predefined templates, which can be altered. Or new templates can be created.
- **Control panel views**
  The BLF panel, held calls panel and park slot panel can be disabled or enabled for viewing purposes only when the operator accesses the viewing menu.
- **Change the Administrator password**
- **Edit operator profiles**
  Each operator can have a personalize profile, which can be configured by the administrator.
- **Specify the maximum length of call notes**
  IP Office supports numerous different endpoints. These have differing display sizes, which means the administrator is able to tailor the call notes field according to the endpoints utilized.

SoftConsole Telephone Requirements

SoftConsole requires an IP Office extension to provide the speech path. SoftConsole has been tested and is certified to work with all Avaya digital and IP phones that are listed in chapter 4, with the following exceptions:

- **Wireless Telephones**
  Avaya does not recommend using the SoftConsole application in conjunction with any of the Avaya wireless telephones.
- **Analog Telephones**
  Analogue single line telephones can be used with SoftConsole, however calls have to be answered manually, as IP Office cannot control the physical status of the hook switch on an analogue phone. This prevents useful headset operation.

SoftConsole PC Requirements

- IP Office switch software release 2.0 or higher
- Ethernet attached PC running the following operating systems, in conjunction with TCP/IP Networking:
  - Windows XP Professional (SP2).
  - Windows 2000 Professional (SP4).
- Minimum recommend PC specification:
  - Pentium II processor 400Mhz or Celeron 2 533Mhz or Athlon B 650Mhz.
  - 64MB RAM or higher as specified by the Windows version.
  - 1Gb of free disk space plus sound card if audio features are required.
- A maximum of four SoftConsole applications can be run per system. A license controls the number of simultaneous SoftConsole users.

11. Voicemail

Voicemail

Voicemail is one of the many applications provided to increase business efficiency and improve client handling. Voicemail provides the equivalent of a telephone answering machine on every employee's desk, indeed, voicemail facilities can be allocated to remote employees even though they may not have a desk or telephone in the main office.

Voicemail allows callers to leave messages for you when you are out of the office, away from your desk or engaged on another telephone call. Voicemail messages can be retrieved either locally or remotely via any telephone (you will be prompted for a PIN number if you are using any telephone other than your allocated extension or a trusted location e.g. your mobile telephone).

Alternatively, you can forward your voicemail to your email and collect it via your PC. This approach allows you to use your PC to display your two different types of messages. It also frees your telephone for incoming calls while using your PC to playback your voicemail. You can then also forward your voicemail, just like any email. For full integration with Microsoft Exchange server and control of voicemails from your client PC, please see Integrated Messaging Pro (described later in this section).

Voicemail, when used in conjunction with Phone Manager application, ensures that you will never miss a customer call again, even when the caller decides not to leave you a voicemail message. In this case the caller's number will be left on the Caller Display of your telephone and/or your PC screen allowing you to dial them back upon your return.

All IP Office systems have been specifically designed to give a business a competitive edge, by providing a total communications system. To this end a Voicemail application is provided as standard on all IP Office systems.

Five voicemail modes of operation are available:

- **Voicemail Lite**
- **Voicemail Pro**
- **Embedded Voicemail with Auto Attendant** ***(IP406 V2 and Small Office Edition only)***
- **Voicemail Pro Networked Messaging with other voicemail systems**
- **Centralized Intuity Audix / Modular Messaging Voicemail**

Voicemail Lite is the standard voicemail application provided with all IP Office platforms. Voicemail Pro builds on the features and facilities offered by Voicemail Lite and can be tailored to meet the individual needs of a business by adding applications such as auto-attendant, call recording and advanced Call Queuing.

Both Voicemail Lite and Voicemail Pro applications can reside on a Windows 2000, 2003 or XP Professional PC. Communication between IP Office and this 'Voicemail server' is via IP over a LAN connection. No specific hardware is required – not even a PC sound card.

If a PC cannot be designated as a voicemail server or you prefer to save space with an all-in-one-box solution then Embedded Voicemail is the preferred option. This application is provided on an Avaya supplied memory card, specially formatted for Embedded Voicemail and with multi-lingual prompts pre-installed. On the Small Office Edition, Embedded Voicemail uses voice compression channels to compress stored messages and prompts.

The voicemail server is multi-lingual and can offer different prompts depending on the user's preferred language, independently of the other internal users' set-ups. Similarly, external callers can hear prompts in their own language depending on their incoming call route (e.g. based on CLI/ANI or DDI/DID). This is very useful to multinational companies or in multi-lingual markets.

Feature Summary

For further details refer to Voicemail Feature Comparison at the end of this section.

Feature	Embedded Voicemail	Voicemail Lite	Voicemail pro
Supported IP Office Systems	Small Office and IP406 V2	PC Based - All IP Office systems.	
Mailboxes	Limited by available storage capacity only.		
Message Storage Capacity	Small Office = up to 10 hours. IP406V2 = up to 15 hours.	1MB per minute up to hard disk capacity.	
Maximum Simultaneous Calls	Small Office VCC3 = 3 Small Office VCC16 = 10 IP406 V2 = 4	4.	Requires licenses: Small Office = 10 IP406 V1/V2 = 20.
Centralized operation.	No.	No.	Yes.
Queue Announcements	No.	Yes.	Yes.
Auto Attendant	Yes.	No.	Yes.
Call Recording	No.	No.	Yes.
Intuity Emulation	No.	No.	Yes.

Centralized Voicemail

Where IP Office is deployed in Definity/Multi-Vantage/ACM Environments it may be desirable to utilize the voicemail system, whether Intuity Audix or Modular Messaging, to provide voicemail services to IP Office users.

Connectivity must be either an E1 or T1 circuit or an IP trunk running QSIG services. In addition to the IP Office license Key (Centralized VM with ACM RFA) that enables this service, further license keys may be required on the Definity/Multi-Vantage/ACM system.

Embedded Voicemail

(IP406 V2 and IP Office - Small Office Edition only)

In environments like retail or home office where no space for a PC is available or you do not want to keep a PC running all the time (notably to avoid the risk of someone accidentally switching off the PC thereby taking away the voicemail service), Embedded Voicemail may be the preferred option to enable an entry-level voicemail service.

On the IP Office Small Office Edition and the IP406 V2, Embedded Voicemail can be installed to provide an entry-level voicemail service. With the Small Office Edition this solution does require a voice compression channel for each call that it handles. See the Voicemail Feature Comparison table for functional details and the IP Office - Small Office Edition section for further details.

The maximum number of messages stored is only restricted by Embedded Voicemail cards storage capacity (currently around 10 hours on the Small Office Edition and 15 hours on the IP406 V2).

The features available include:

- No License Key required.
- 4 independent Auto Attendants (AA).
- 3 time profiles per AA.
- Up to 12 menu items per AA.
- Auto Attendant - automatic time out to fallback location: Default set to 8 seconds.
- 3 Port voicemail as standard on Small Office Edition (10 ports with 16VC variants of SOE), 4 port voicemail for IP406 V2.
- Up to 10 hours storage on SOE, 15 hours message storage on the IP406V2.
- Default local access from each extension.
- Define system-wide short code for VoicemailCollect (e.g. *17).
- Set access code/PIN for each extension for secure access.
- Configurable record time: Default value 2 minutes, maximum value 3 minutes.
- Access codes: Minimum of 4 characters to be set.
- Multiple languages stored on the Flash Memory card.
- Help menus (via *4). Greetings & Mailbox Navigation.
- Voicemail Breakout: Press *0 at any time to return to the operator.

Voicemail Lite

IP Office's standard Voicemail application can handle up to 4 simultaneous calls. When enabled Voicemail Lite automatically answers your telephone when you are not available to take a call. Personal greetings can be recorded, providing confirmation that the intended recipient will actually receive the message.

Messages may be played as a continuous loop. This allows information to be heard, but no message to be left. The caller may press a key on the telephone at any time to be transferred to a pre-determined number, usually the receptionist or secretary.

When voicemail messages have been left, the number of new messages waiting will be displayed on the Phone Manager application and/or the telephone caller display panel if used. Voicemail Lite can also periodically ring the extension to deliver any new messages. When voicemail messages are left they are time & date stamped and the caller's number recorded. Once listened to, old messages are deleted 24 hours after being left. Alternatively they may be saved permanently.

Voicemail can be collected remotely by dialing the Voicemail Lite server. Using the security inherent in all IP Office systems, if the number the user is dialing from is "recognized" (home number or Mobile/Cell Phone for example), they will automatically receive their voicemail as if collecting it from their office extension. This is especially useful when collecting your voicemail while on the move, using your Mobile/Cell Phone in hands-free mode. If the source number is not recognized, users will be prompted for a mailbox number and a PIN code for that mailbox, before they can collect voicemail. Users have the ability to set and change their own PIN codes.

Where a voicemail needs to be copied to others, Voicemail Lite provides many options:

- Voicemails can be simply forwarded to another mailbox, or group of mailboxes
- Recipients can append their comment to the voicemail before forwarding to another mailbox(es).
- Alternatively voicemails can be forwarded as emails.

Voicemail Pro

Voicemail Pro builds on the features and facilities offered by Voicemail Lite, can be tailored to meet the individual needs of a business and can scale up to 30 simultaneous calls if required.

Voicemail Pro allows message handling for individuals or groups, provides information to callers, assisting the operator during periods of heavy call activity and more by including a powerful voice processing system and an easy to use graphical user interface - the Voicemail Pro client. Voicemail messages can be integrated into a user's email box and dealt with as any email message. Through the use of Text To Speech facilities users can be provided with access to their voice and email messages through the telephone while in the office or remotely when away from the office.

The Voicemail Pro client application also allows far more to be achieved than just guiding a user to the group or extension they require. It allows Voicemail Pro to dial back users, internally or externally, as soon as a voicemail message is left for them. It provides security, by prompting for a PIN code should a user wish to change their Forwarding or Follow Me number from an external telephone.

A single Voicemail Pro server (PC-based) can provide voicemail services to multiple IP Office systems over the LAN, WAN or a Frame Relay network. This is referred to as 'Centralized Voicemail' and can reduce costs, while facilitating communication between IP Office sites.

Other uses for Voicemail Pro include Whisper Announce which prompts callers for information (usually their name) which is recorded and passed on to the user's extension (if free), allowing them to choose to accept the call or not. This is particularly useful on "CLI/ANI withheld" numbers - usually calls from telesales companies where somebody is trying to sell you something. Voicemail Pro will not intrude onto busy extensions.

Assisted Transfer allows transfer of a call to a destination, but allows the call to return to Voicemail Pro automatically for other options should the called party be engaged, or not answer within a pre-determined time.

By testing conditions (such as whether out-of-hours), calls can be routed depending on system or user-definable criteria. Conditions are constructed from a set of basic elements. These elements can be combined within a single condition to create complex rules. For example, the Week Planner can be used to define the company's standard working hours, and then combined with the calendar to define exception days such as public holidays / vacation.

Voicemail Pro also offers the concept of modules. Modules allow you to create sequences of actions that you want to share between a number of different call routing scenarios. These modules can be used to create a library of vertical voicemail applications or just easy dissemination to other IP Office voicemail sites, thanks to its import and export functionality.

Voicemail Pro can also trigger external actions such as activating the external relays on the IP Office. For example, remotely checking the status of the office heating and then turning it on from your Mobile/Cell Phone on your drive in to work.

Voicemail Pro provides the ability to allow a caller to select the language in which they require the system to respond in.

Finally, a Speaking Clock, that takes its time from the Voicemail server, is built into Voicemail Pro to minimize call charges.

In summary Voicemail Pro adds:

- Voicemail Pro client graphical user interface.
- Customizable voicemail services for individual business requirements.
- Personal Numbering.
- Broadcast group messages.
- Audiotex and Auto-Attendant services (including dial by name).
- Sophisticated Queue Announcement facilities.
- Conditions (e.g. test if 'out of hours').
- Automatic and On Demand Call Recording with an optional Search and Play utility if using IP Office ContactStore.
- Voice Forms/Questionnaire Mailboxes (Campaign Manager).
- Personal distribution lists.
- Access to Database information for building Interactive Voice Response (IVR) systems.
- Tag information retrieved from a database to a call and deliver it with the call to an agent.
- Visual Basic (VB) Script support to allow the configuration of the Voice system through VB Scripts rather than Voicemail Pro call flows.
- Extended Personal Greetings to customize the information presented to a caller based upon the availability of a user.
- Text To Speech facilities to allow emails to be read out over the telephone and/or for database information to be read to a caller in 14 languages.
- Housekeeping facilities for the management of messages.
- Automatic detection and routing of Fax calls within Auto Attendants and within a subscriber's voicemail box.
- Forwarding of voicemail messages to Email systems via SMTP.
- Support for a range of the Intuity telephone user interface features in Intuity emulation mode.
- Recording of system prompts through the telephone handset or using multimedia facilities on a PC.
- Speaking Clock.
- 22 supported prompt languages: Chinese (Mandarin), Danish, Dutch, English (UK), English (US), Finnish, French (France), French (Canadian), German, Greek, Hungarian, Japanese, Italian, Korean, Norwegian, Polish, Portuguese (European), Portuguese (Brazilian), Russian, Spanish (Castilian), Spanish (Latin American), Swedish.
- Centralized voicemail within a multi-site IP Office environment.
- Networked Messaging with other Avaya voicemail systems.
- Capacity of up to 30 ports (depending on IP Office Control Unit).

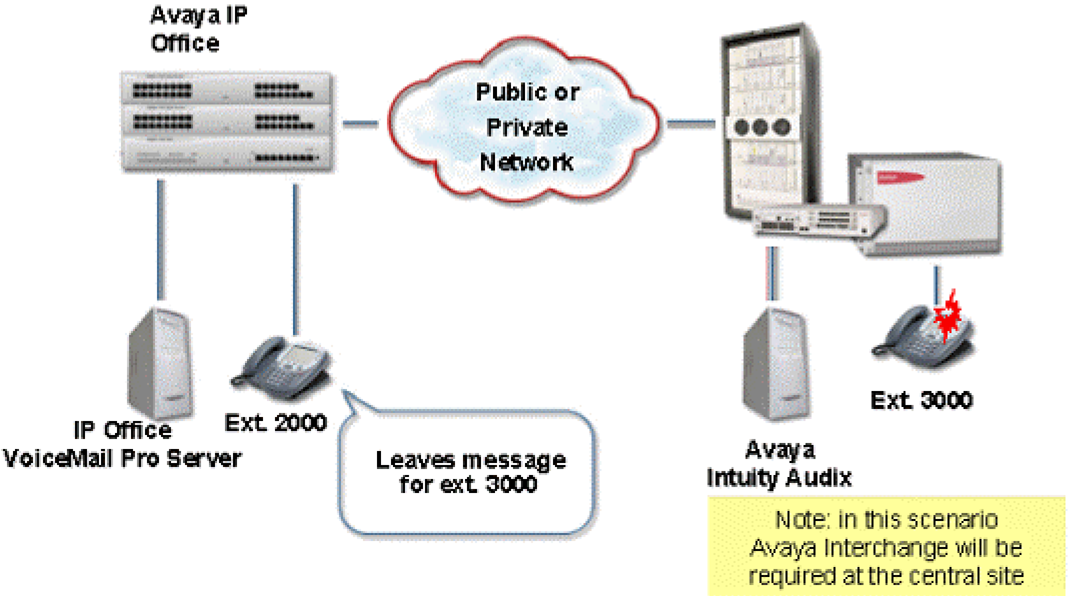
Further details on some of the Voicemail Pro client functionality listed above are described later in this section. Further information on Queue Announcements, Call Recording and Campaign Manager can be found in Compact Contact Center (CCC).

Networked Messaging

Increasingly organizations are operating a number of different voicemail systems across a number of sites. In this situation it is important to be able to provide integrated operation between voicemail systems so that messages can be passed between systems and delivered to a user's mailbox seamlessly. This is achieved by IP Office Voicemail Pro being licensed to support Networked Messaging.

The Networked Messaging Solution defines a common set of features to allow inter-working between Avaya voicemail systems. In Intuity mode, while listening to or having listened to a message, the user can select the option to forward the message to another mailbox, the mailbox entered can be any mailbox number on the local system or any mailbox on a remote Avaya system.

The IP Office Networked Messaging facility will allow configuration of up to 2000 remote mailboxes on each Voicemail Pro server and will operate with other IP Office systems supporting this feature, the Avaya Interchange and Avaya S3210 servers.



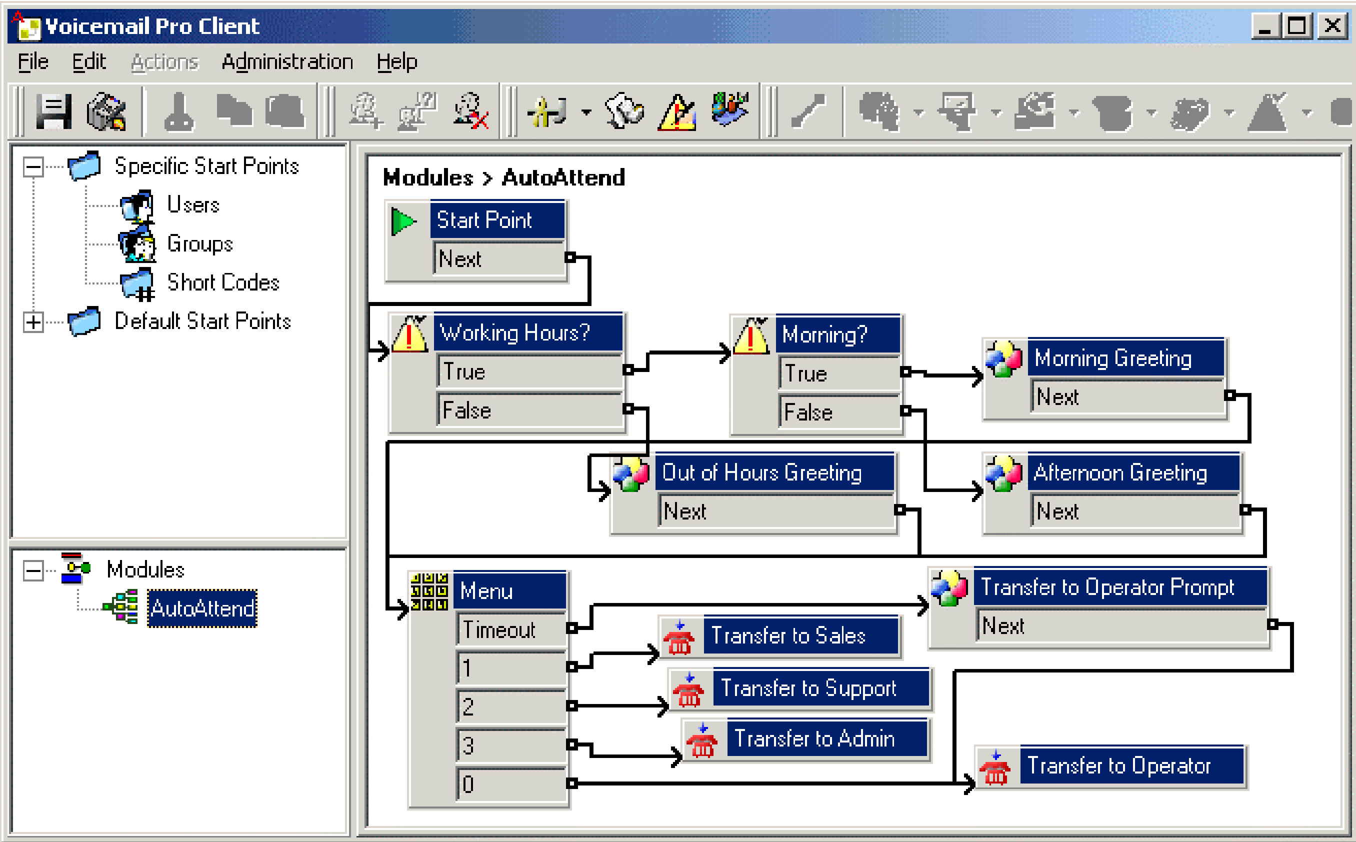
Auto Attendant

In addition to its advanced voicemail facilities, Voicemail Pro provides an easy-to-use, multi-level configuration tool 'the Voicemail Pro Manager' which allows network managers and system administrators to construct an interactive system, based upon DTMF telephone key entry.

At its most basic, this allows an Auto-Attendant system to be built and configured to suit the way the company operates in the best interests of staff efficiency and customer service, be that on its own, or as a back-up for the regular operator when call volumes are high. Voicemail Pro also offers the ability to enter the name of the person via DTMF key entry, after which the auto-attendant offers the caller a possible name that matches or if there is more than one, a selection list is provided and allows the caller to select which one they wish to call.

As an example, Voicemail Pro can be used to build an Auto-Attendant that prompts callers to "enter 1 for sales, 2 for support, 3 for admin, or 0 for the operator" allowing them to be transferred to the appropriate department without operator intervention. Alternatively, a list of personnel and their extension numbers could be listed, allowing the caller to directly access the person they want. For larger companies it could be department number listed first, followed by the list of employee extensions within the department.

The latter two examples are ideal where company telephone operation has changed from a central operator only based system to Direct Dialing In (DDI/DID), allowing callers to "learn" the required extension number from the prompting of Voicemail Pro, and then in future dial the extension number directly. Auto-Attendant operation is also ideal where multiple languages are required, for example "Dial 1 for English, 2 for German, 3 for French, ...".



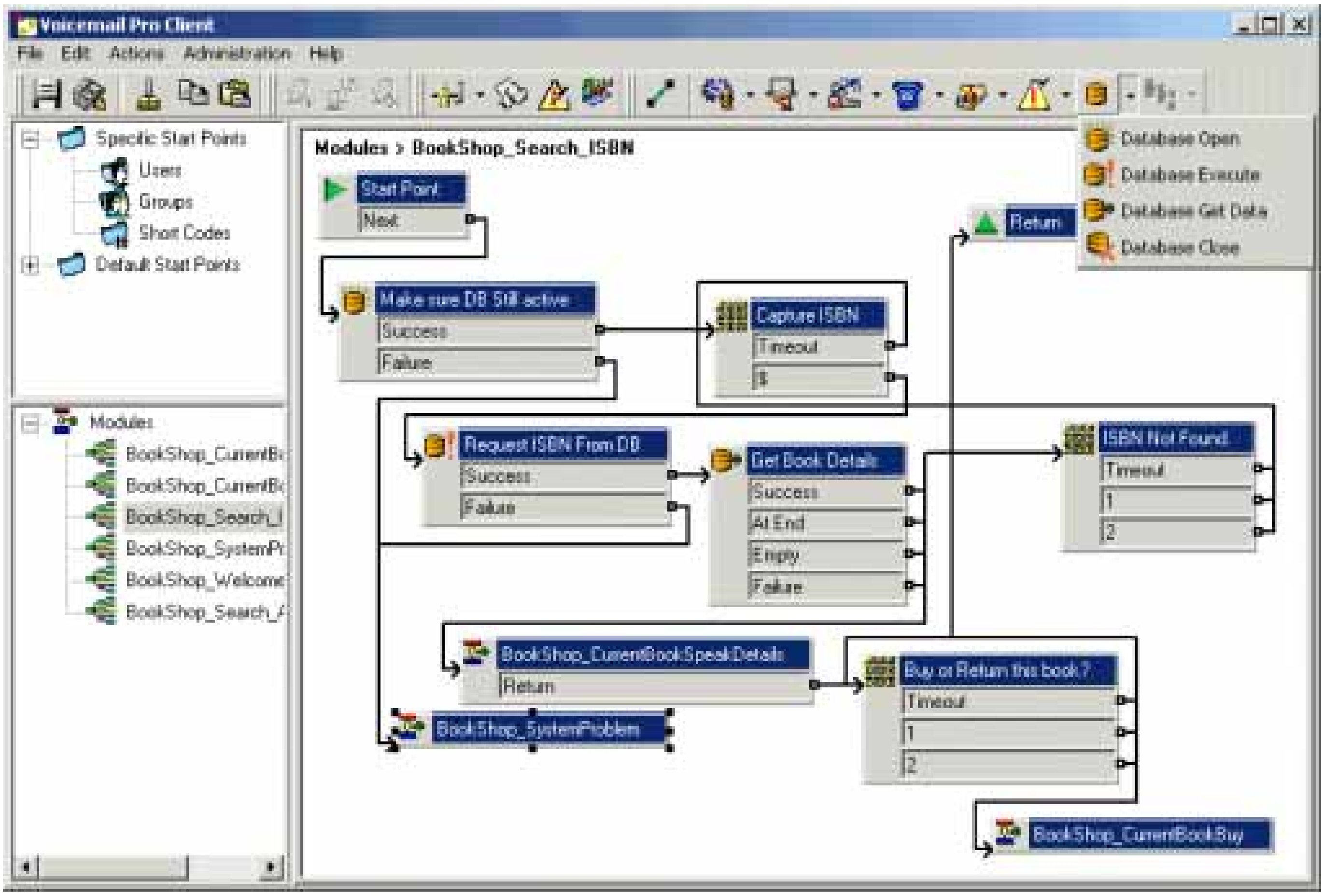
Auto-Attendant created using Voicemail Pro Manager

Accessing Database Information within Call Flows (IVR)

The Voicemail Pro Manager provides the ability to construct powerful interactive systems based upon DTMF telephone key entry. This is achieved by using the flexibility provided from the built in call flow actions. As a caller passes through any part of a defined call flow the system is capable of interacting with most third party databases through the use of the standards based ADO interface (ActiveX data objects). The system is capable of retrieving information from a database and writing information into databases. The result of this is that powerful Interactive Voice Response systems (IVR) can be designed to specifically meet the requirements of the business and the customer experience that is required.

Example interactive systems that can be built as a result of these facilities include: Information Bulletin Boards, order taking and order processing systems, front end systems to Help Desks/Support Desks, Contact Centers, secure access to information through PIN checking, survey systems, remote time sheet management, etc.

The ability to interact with Database information is enabled through the purchase of the IPO LIC - IP400 3rd PRTY IVR RFA license key. The entry of this key will enable the operation of four new Database Action Icons within the Voicemail Pro Manager GUI.



Example Call Flow Utilizing Database Actions

The new database actions that are provided through the Voicemail Pro Client are:

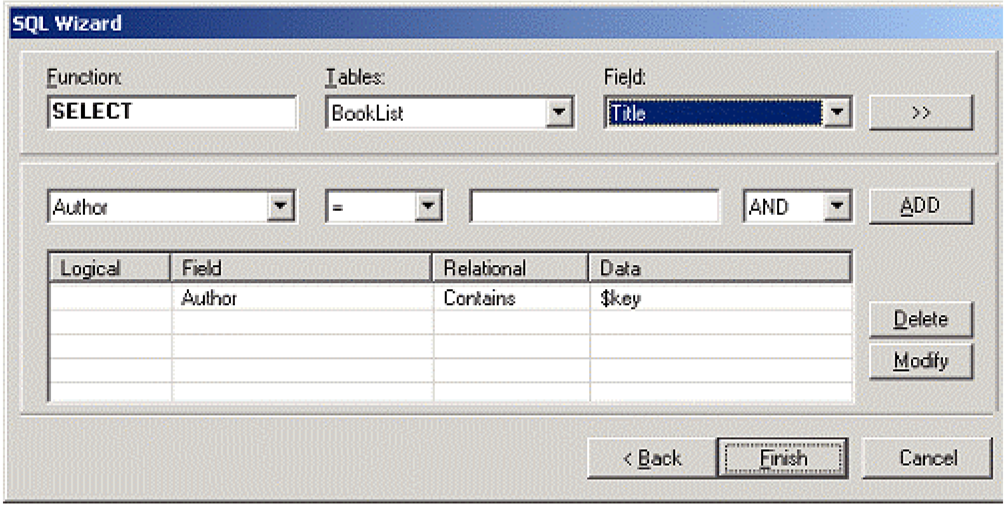
- **Database Open** – Opens a link to the required database. Multiple databases can be accessed during a call but only one database can be opened at one time.
- **Database Execute** – Provides the ability to enter a query on the opened database. The query can 'Select' data from the open database or can 'Insert' data into the database.
- **Database Get Data** – Provides access to the data that has been retrieved from a database through the Database Execute action. The user can retrieve the next item, previous item, first item in the list or the last item in the list.
- **Database Close** – This action will close the current database. If the database is open when a call terminates then the database will be automatically closed.

As with other Voicemail Pro call flow actions the new database actions include the ability to communicate with the Avaya Compact Contact Center for reporting purposes.

Access to ADO compliant databases is achieved through the use of database drivers. As standard the installation of the Voicemail Pro software will include the installation of the Microsoft Data Access Components (MDAC) version 2.5 service pack 3 to provide access to most database systems. Any database not included within this list can be added to the system.

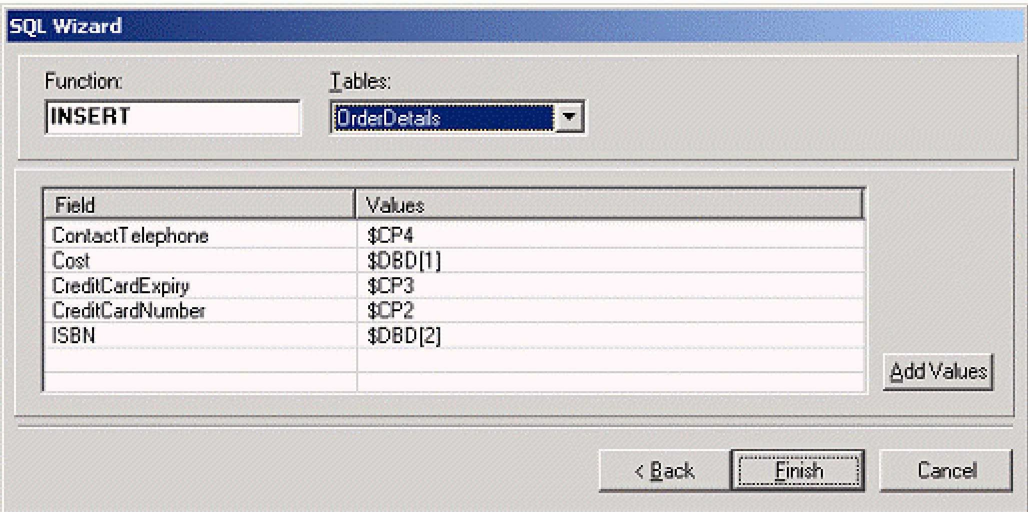
Interaction with the opened database is achieved through the use of Structured Query Language scripts (SQL). An administrator can enter SQL script directly into the 'specific' section of the Database Execute action. For administrators that are not familiar with SQL scripts a script can be automatically created through the use of a SQL Query Builder Wizard. The wizard will allow the administrator to create the SQL script by simply selecting options from drop down menus, e.g.:

When 'Selecting' information from a database:



In the example above the system will find the 'Title' filed entries within the 'Booklist' table where the 'Author' field contains the string held within the $key field ($key is the last DTMF entry made by the caller through their telephone handset, DTMF entries can be numeric or alpha-numeric through multi presses of the telephone keypad).

Alternatively, information can be 'Inserted' into a database:



In the example above the fields within the 'OrderDetails' table will be updated with the information held within the defined system variables, i.e.:

ContactTelephone will be updated with the current contents of the $CP4 variable, Cost will be updated with the current contents of the $DBD[1] variable, etc.

The information retrieved from a database can be assigned to any system or user defined variable and can be used at other points within a Callflow. For example any information held within a system variable can be passed to an extension through the use of the Callflow action ‘Assisted Transfer’. When an assisted transfer is completed any information assigned to the transfer will be passed to the extension and will be displayed on the telephones display. Alternatively, this information could be used by third party applications through the use of the TAPI interface for popping applications on a PC screen.

To further enhance the new database facilities two additional actions have been added to the Voicemail Pro Client. These actions are 'Alphanumeric Collection' and a 'Speak Text' action.

The Alphanumeric Collection action allows the system to collect characters as well as numbers through the telephone handset. A user can select the character they require through multiple presses of a key on the keypad. This is a similar operation to text entry on a mobile phone or the letter collection facility on Intuity Audix, e.g.: letter K is generated from pressing the 5 key twice. As a key is pressed the system will read back the letter that is selected.

The Speak action allows the use of Text To Speech facilities to play information back to a caller.

Using Text To Speech (TTS) Facilities within a Callflow

To further enhance the database facility, Text To Speech facilities can also be licensed. TTS facilities can enhance the callers experience by allowing the system to read back to them any information that has been captured from a database. In the examples above, a Book Shop system, the caller dials into the system and is asked for an ISBN number or for the Author's name of the book they require. The caller enters an Authors name through the telephone keypad and the system locates the title of the book from the database. As well as finding the title, as in the wizard above, the system could also look up the Author of the book and whether there were any books in stock. By using TTS, the system could now respond to the call:

" The book, Lord Of The Rings, costing $6.99, written by J R R Tolkien is in stock".

Offering to allow the caller to order this book by entering contact and credit card details could now enhance this system further.

The TTS facility is provided as a license through the use of either the IPO LIC-IP400 Avaya TTS RFA 1 or IPO LIC - IP400 3rd PARTY TTS SPPRT RFA 1.

Each license purchased provides a single use of a TTS engine, multiple engines can be licensed on each Voicemail Pro system. For example, a four port Voicemail may have two TTS licenses enabled, these two TTS engines will be used by all four Voicemail ports on a first come first served basis. At any instance in time only two callers can use the TTS facility in this example. Purchasing additional licenses will increase the number of TTS engines available.

The Avaya TTS RFA1 utilizes the Avaya TTS engine. In addition to this license the TTS software media pack IPO CD - IPO AVAYA TTS CD SET will also need to be ordered, this is a 5 CD set containing the TTS engine software and all supported languages.

The 3rd Party TTS SPPRT RFA provides the Voicemail system with a SAPI 5 interface for use with another suppliers TTS engine. When using this license, Voicemail Pro will look for a pre-installed SAPI 5 compliant TTS engine on the Voicemail Pro server and utilize this for the delivery of TTS facilities. Once again a TTS RFA will be needed for each TTS engine that is required. For information, all Microsoft Server Operating systems are shipped with the Microsoft TTS engine included as part of the system. As a result this engine should be available for use by a customer as default. The Microsoft TTS engine will operate with Voicemail Pro.

The Avaya TTS engine currently includes 14 languages as default. During installation the Administrator can select which Languages they wish to have installed on the Voicemail Pro server. Once installed on the Voicemail Pro server the TTS engine can use any combination of these languages. The language used will be decided by the system or user localization that is configured. This means that multi language solutions can be easily provided, for example some users may have their emails read in US English and others have theirs read in Chinese. Within a call flow, information can also be read back to callers in different languages by using the 'System Prompt Language' to select the require language.

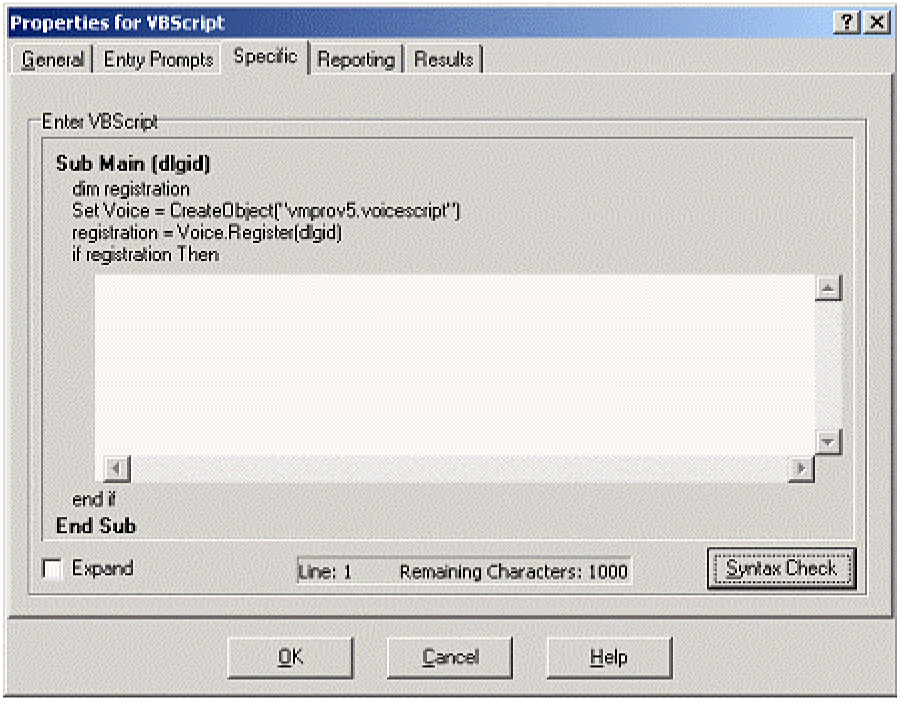
The languages currently supported by the Avaya TTS engine are:

- **Chinese (Mandarin)**
- **Dutch**
- **English (UK)**
- **English (US)**
- **French (Standard)**
- **German**
- **Japanese**
- **Italian**
- **Korean**
- **Norwegian**
- **Portuguese (Brazilian)**
- **Russian**
- **Spanish**
- **Spanish (Latin)**

Visual Basic (VB) Scripting

The Voicemail Pro Client call flow interface has been extended to allow an administrator to provide Visual Basic (VB) scripted logic that can be interpreted by the Voicemail Pro server. This ability allows system administrators to program the voice system via VB Scripts thus providing additional choice and flexibility in providing IVR applications.

The new VB script action contains a VB-Scripting parser (Syntax checker) to ensure the legitimacy of the administrator derived VB Script before it's incorporation.



Each VB script action used within a call flow can contain a maximum of 1000 characters, however a call flow may contain multiple VB script actions within it.

Personal Numbering

Contact-ability is all-important in winning and maintaining business. Voicemail Pro offers users the ability to remotely turn their voicemail on or off, set their email forwarding, edit their call forwarding and follow me numbers. Together these actions provide a comprehensive Personal Numbering service for the user who needs to remain in contact regardless of their physical location.

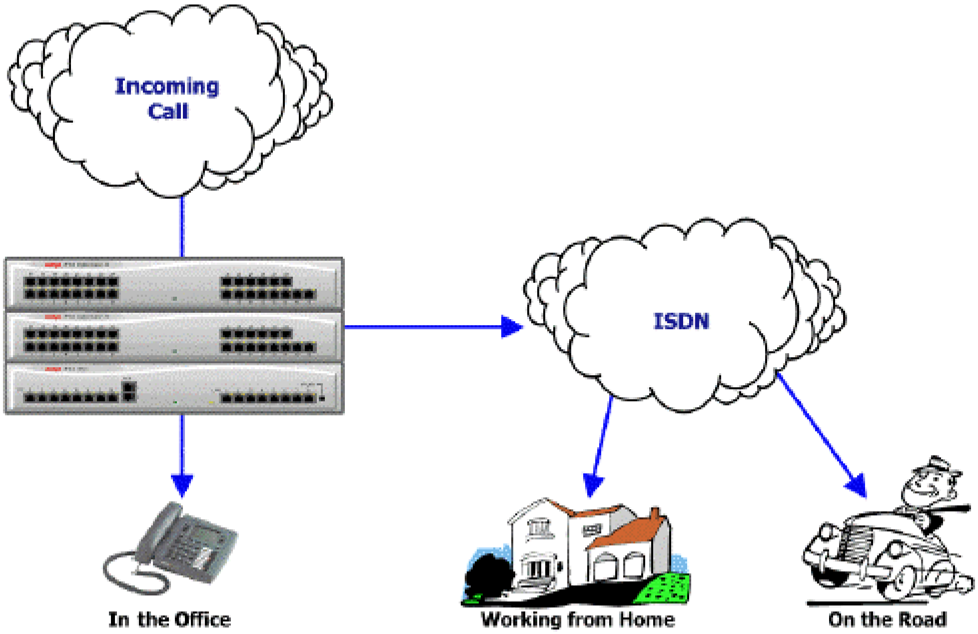


Diagram illustrating personal numbering

Extended Personal Greetings

In Intuity emulation mode, the Voicemail Pro system has the ability to hold a number of greetings within each users mailbox that can be played to a caller. In addition to the standard mailbox greetings, the extended personal greetings provide the ability to present the caller with a greeting that reflects where the call has come from (internal or external) or why the called party is unable to take the call. A mailbox user can configure the responses that are played back to the caller, based upon the reason the caller was routed to the Voicemail. The supported call states are:

- **Busy/Engaged -** The user is currently on a call and unable to accept a second call.
- **No Reply -** The user is away from the desk and unable to take a call.
- **Internal -** A greeting to be played to internal calls
- **External -** The greeting to be played to external callers
- **Out Of Hours -** The greeting played when a hunt group is operating 'out of hours'. Out of hours is defined with IP Office Manager.

A greeting can be recorded for each of the above conditions through the Telephone User Interface (TUI). If a recording is made for each condition the order of play back to a caller will be:

- 1st - Out of hours (Hunt group mailboxes only).
- 2nd - Busy/Engaged.
- 3rd - Internal/External greeting.
- 4th - No reply.

A mailbox owner will need to record greetings against these conditions to deliver the greeting that they wish to present to a caller.

Group Broadcast Messages

With Voicemail Pro, two modes of operation exist for the handling of hunt group messages. The method used is configured for the group through the IP Office Manager.

The first mode places the message in the hunt group mailbox and only informs those individuals configured for message waiting indication from that group. This is ideal for scenarios where only a few people such as a call center supervisor need to be initially aware of group messages. This is the default mode of operation. Any message waiting light lit by this is extinguished when the new hunt group message is accessed by a user.

In the second mode of operation the message not stored in the hunt group mailbox. Instead it is broadcast (copied and forwarded) to the individual mailboxes of all the hunt group member. This lights the individual messages waiting light of each user until they access their mailbox.

Personal Distribution Lists

Personal Distribution Lists are only available with Voicemail Pro when operating in Intuity mode. The feature provides the ability for a user to distribute a voicemail message to a list of recipients simultaneously. Lists can be configured by a voicemail box subscriber either through their voicemail box telephone user interface (TUI) or through the desktop PC application PhoneManager. This feature operates is similar in operation to the same feature available with the Avaya Intuity Audix.

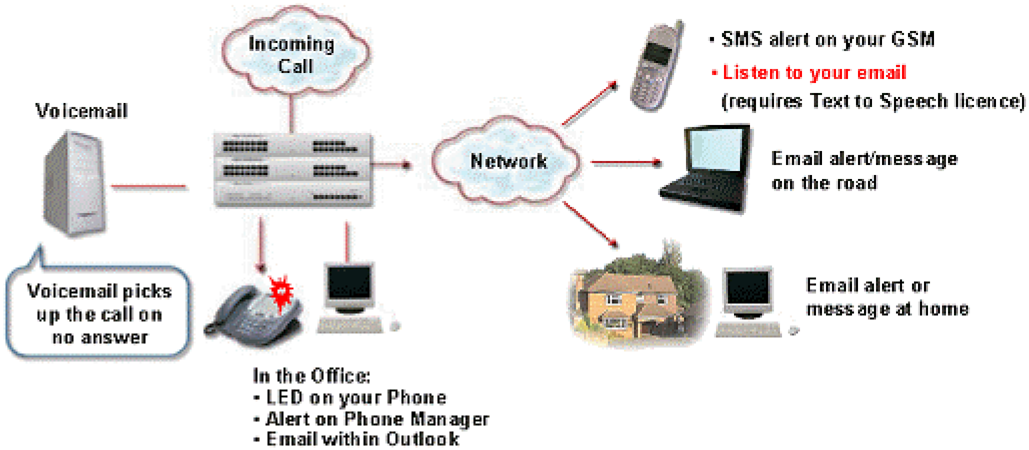
The features available to a voicemail box subscriber include:

- Create up to 20 lists with 360 members per list
- Mark a list as Private or Public, Private lists can not be accessed by any other voicemail subscriber. Public lists can be used by other subscribers but can not be edited.
- Public lists can be copied from one subscriber to another by adding the contents into a new list.
- Subscribers can 'Create' new lists, 'Scan' contents of an existing list or 'Modify' existing lists.
- List members can be added by using the station number or mailbox name (names are not supported for Voicemail Pro Networked Messaging mailboxes).
- Lists can include voicemail boxes that exist on other Avaya Voicemail systems that are available through Voicemail Pro Networked Messaging.
- Lists can be added together, duplicate members are automatically removed. This includes public lists owned by other voicemail subscribers.
- Mailing lists are accessible to the user at any 'send message' and 'forward message' option within the user's voicemail box.
- When displayed within Phone Manager Pro, distribution lists can have a list description added to it, this is only visible within Phone Manager Pro.

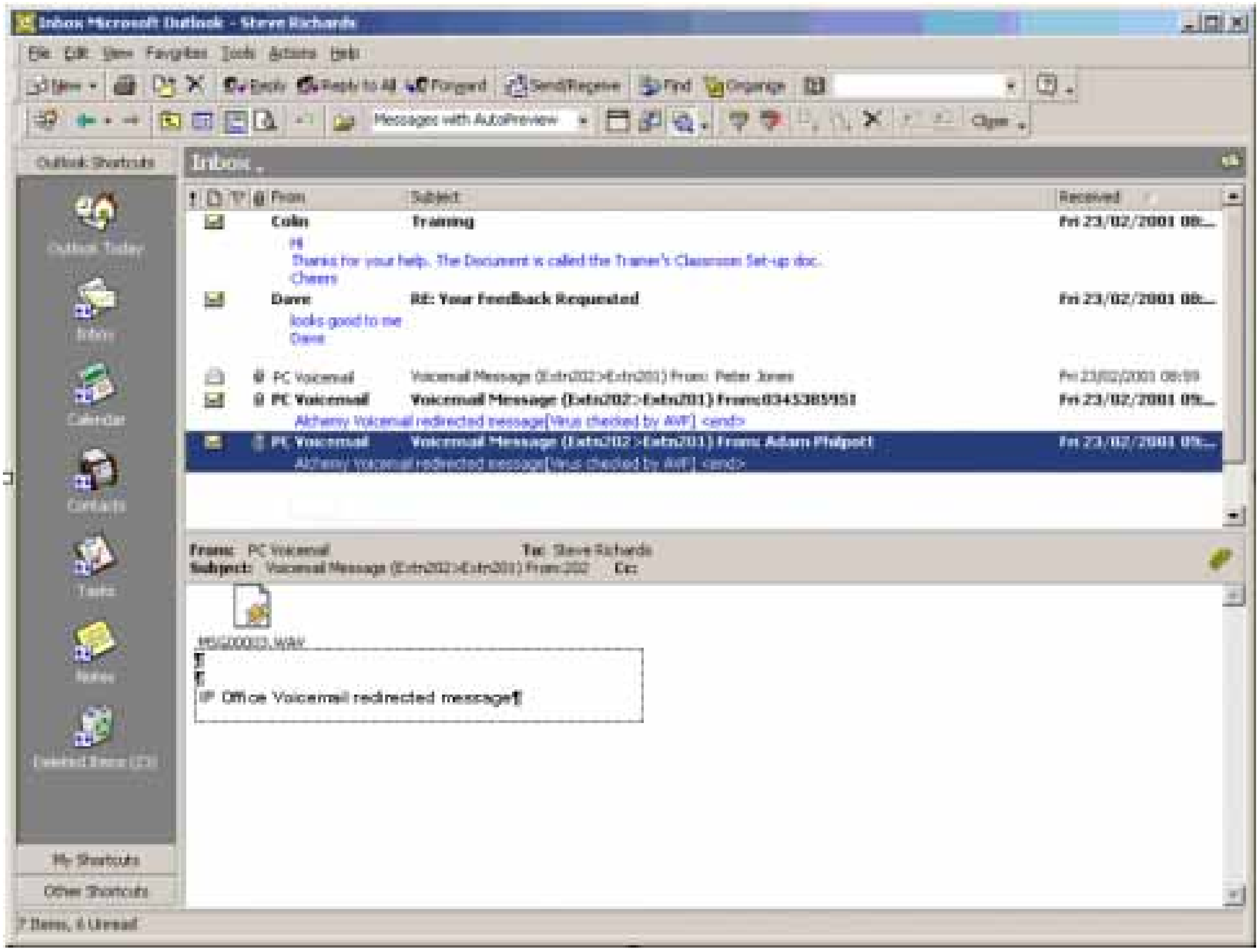
Interaction of Voicemail with Email Systems (Unified Mailbox) and Fax Systems

As standard, Voicemail Lite and Pro allow for a simple voicemail alert or the entire voicemail to be forwarded or copied to any MAPI or SMTP compliant Email application (Microsoft Outlook, Exchange, Lotus Notes, etc.) as a .WAV file attachment. This allows emails and voicemails to be unified and collected from a single source – the email client.

The simple alert option (which forwards only the time, date and caller's number information) has been designed for use with commercial Short Message System (SMS) services whereby this information can be forwarded to the display on a Mobile/Cell Phone or Pager when the user is away from the desk. This email notification, forwarding and copying, can be done for all voice messages or on individual selection and can be activated remotely. This is beneficial if you are working from home and have an email connection available.



Forwarding voicemail to email is one element of unified messaging and is particularly useful for group voicemail boxes as it allows a single voicemail message to be copied to the email of every member in that group.



Presentation of Voicemail to Email

While not directly supplying or supporting fax software, the same result can be achieved with fax to the desktop or client fax applications when using fax servers. This then allows an Email client (for example Microsoft Outlook) to be utilized as an easily affordable unified messaging solution. The many benefits of unified messaging include security (as faxes are sent to the users PC rather than on paper for everyone to see), ease-of-use and efficiency in terms of storage and retrieval of messages and the great gains that can be made in overall workforce efficiency and productivity.

To enhance the support of Third Party Fax solutions, Voicemail Pro supports the automatic detection of incoming fax calls. Traditionally a dedicated telephone number will be provided for all incoming fax calls. In addition to, or as an alternative to, the Voicemail Pro 'Menu' action or a subscriber's voicemail box (Intuity mode) can automatically detect any incoming fax calls and then direct the call to a predefined location. The benefit to a business or user is that only one number is required for either voice or fax calls.

The Voicemail Pro can store the default fax location for the automatic routing of fax calls. Alternatively, with fax tone detection at the voicemail box, each voicemail box can have a fax location number. If a voicemail box owner has set their own fax number, then that number is used instead of the default fax location. A voicemail box subscriber can set their own fax number through their mailbox menus.

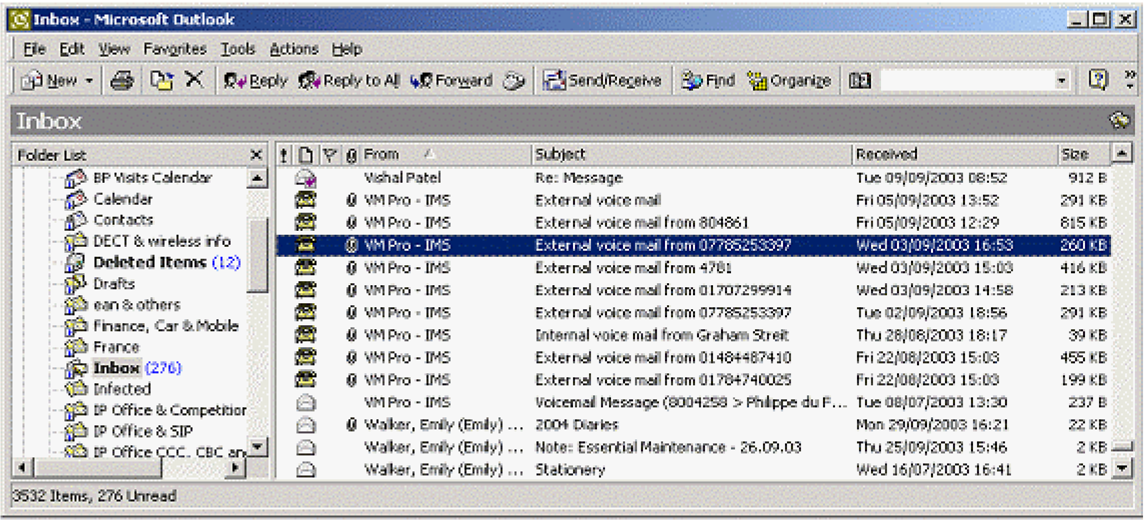
Most fax solutions can be used in conjunction with IP Office, however the following products have been tested and verified to operate in the above scenarios:

- **Equisys - Zetafax:**
  Zetafax for Networks provides versatile network fax software solutions for small businesses, corporate offices and distributed enterprise businesses. It enables employees to send and receive faxes at their desktop, without the need to print fax communications, take them to a fax machine and send them manually. Zetafax can be seamlessly integrated into market leading email systems like Exchange allowing users to send and receive faxes directly from their Outlook client. In addition Zetafax can be integrated with other existing applications, such as accounting or CRM systems, for fast, automated faxing from the desktop or back office. Zetafax for networks is already used by more than 60,000 customers worldwide.
    - Further product information available from www.equisys.com
- **Captaris - RightFax:**
  RightFax offers a broad, scalable product line that integrates with email, desktop, CRM, ERP, document management, imaging, archival, call center, copier/scanner systems, as well as host, legacy and mainframe applications–virtually all business applications.
    - Further product information available from www.captaris.com
- **Fenestrae – Faxination**
  Fenestrae Faxination Server for Microsoft Exchange integrates fax into email technology. Create faxes on your desktop and deliver them to your chosen fax machine at the click of a mouse.
    - Further product information available from www.fenestrae.com
- **GFI – GFI FaxMaker**
  GFI FAXmaker for Exchange/SMTP allows users to send and receive faxes and SMS/text messages directly from their email client It integrates with Active Directory and therefore does not require the administration of a separate fax user database. GFI FAXmaker integrates via the SMTP/POP3 protocol with Lotus Notes and any SMTP/POP3 server. .
    - Further product information available from www.gfi.com

Integrated Messaging Pro (Microsoft Exchange & Outlook only)

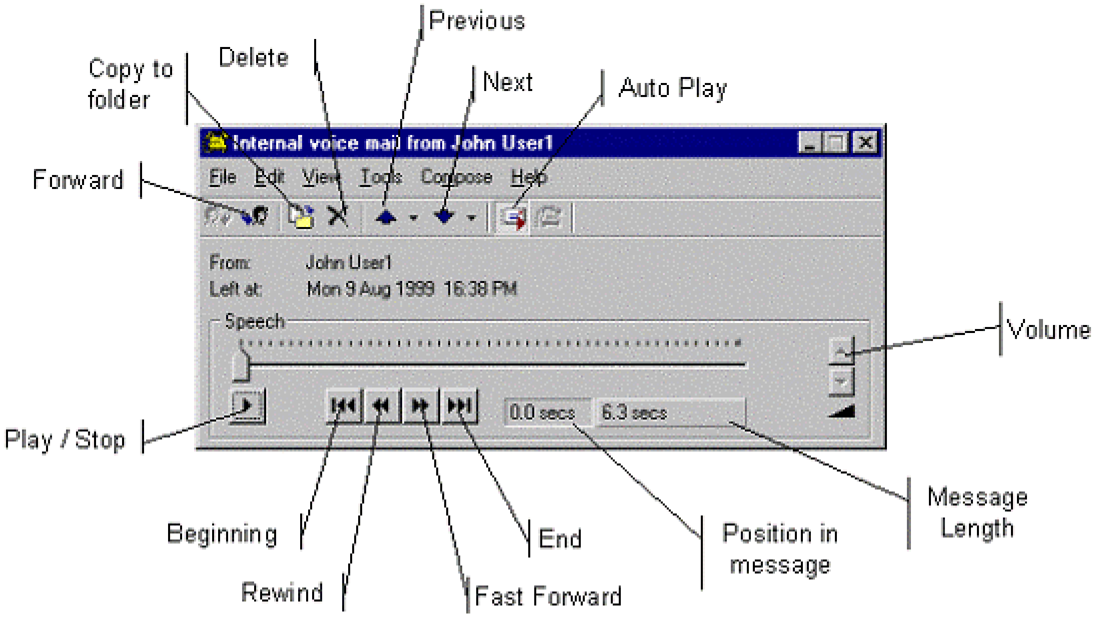
Integrated Messaging Pro allows increased productivity through easier management and prioritization of email and voicemail messages through one inbox. This optional application integrates IP Office Voicemail Pro and Microsoft Exchange email systems.

With Integrated Messaging Pro software installed on your PC you will find that your Voicemail messages will appear in your inbox along with your Email messages. A Voicemail message is shown with a telephone icon. To listen to the message open it by double clicking on it.



By keeping the voicemail messages on the Voicemail Server, bandwidth is kept to a minimum (each message is only a few hundred bytes rather than a few Megabytes) and therefore reduces the load on the computer network). When message files are transferred from the Voicemail server to the Email server using Integrated Messaging Pro the files are compressed using GSM compression to reduce the overhead on the network (approximately 1:11 compression of a .WAV file).

Users can listen to their voicemails either through their PC speakers, an associated desktop telephone, at home or on a Mobile/Cell Phone if diverts are set at the desktop. The latter option is useful when working from home or on the road as it avoids downloading large voicemail files for playback on a multimedia PC.



Integrated Messaging Pro user interface

The interface offers the following options to the user of Integrated Messaging Pro on IP Office:

- Playback via your handset, multimedia PC or Mobile/Cell Phone.
- Forward voicemails to other mailboxes.
- Delete.
- Answer in any order.
- Copy.
- Fast Forward.
- Rewind.
- Time and Date stamp.
- CLI/ANI information if external, or caller's name if internal.

When presented in Outlook, voicemails will appear similar to emails. Contained within the header message will be the caller's number information (if the CLI/ANI is available) or a name if the call is internal. If the name is not contained within the IP Office directory then the extension number will be shown.

With Integrated Messaging Pro, the email server and desktop telephone are synchronized i.e. deleting a voicemail will remove the relevant email notification and, vice versa, the red message waiting light on the desktop telephone telephone will disappear if a voice message is deleted within Outlook.

Within Intuity mode on Voicemail Pro voicemail messages can be marked as Private or Priority. Any Priority message received is shown with a red exclamation next to the telephone icon . A private message is indicated with a padlock shown in the toolbar when a message is opened.

Text To Speech (TTS) for Email Reading (Microsoft Exchange only)

In addition to providing a unified mailbox for voicemails, emails and Fax message, Voicemail Pro can also provide the ability to retrieve Voice and Email messages through the telephone. When operating in Intuity mode and with the system licensed for Text To Speech (TTS) facilities the user will be presented with a list of both Voicemail messages and Email messages. The emails can be read out over the telephone in any of the supported 14 languages, based upon the system or user localization settings. The benefit to the user is that their messages are now accessible while in and out of the office through any telephone.

When accessing messages through the telephone all new Voicemail messages will be presented to the mailbox owner before any new Email messages. When accessing an Email message the system refers to the message as "New message with text".

The TTS facility is provided as a license through the use of either the IPO LIC-IP400 Avaya TTS RFA 1 or IPO LIC - IP400 3rd PARTY TTS SPPRT RFA 1 (see TTS in Call Flows for a description of these two licenses and the TTS media pack).

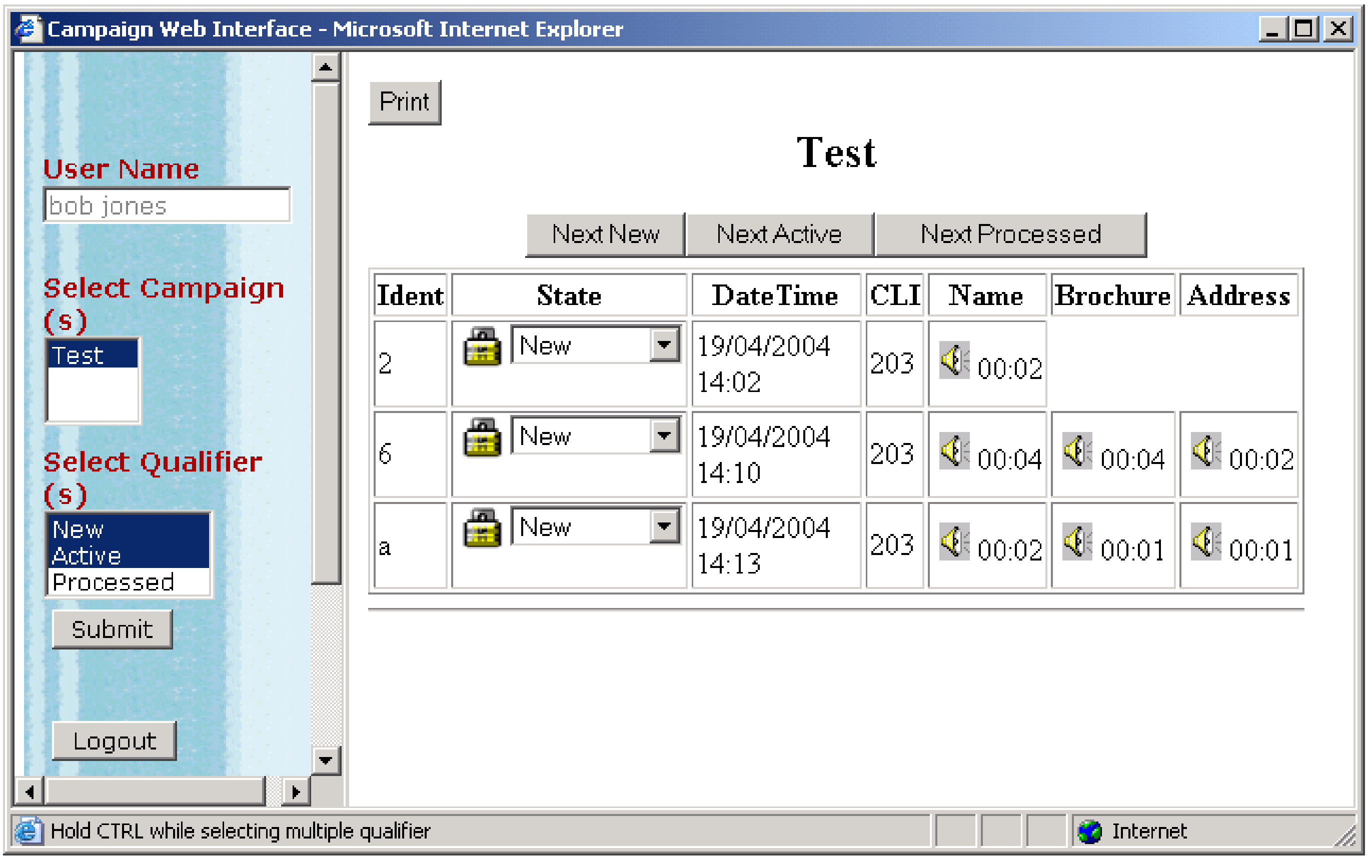
The differences between these two licenses are the language support and a difference in the Email reading capability. With the Avaya TTS RFA, a TTS engine is included with support for a minimum of 14 languages. All languages supported are included are made available as part of the license charge and TTS media pack (see TTS in Call Flows for a list of supported languages). Furthermore, when reading emails the Avaya TTS engine will analyzes (parse) the Email contents and only play out information that is appropriate, with the 3rd Party TTS RFA the capabilities offered will depend upon the TTS engine that is then utilized.

Configuring the reading of emails to users is a simple exercise. Firstly, TTS services will be loaded onto the Voicemail Pro server (the Avaya TTS media pack will install the Avaya TTS engine). Secondly a license key will need to be purchased and entered into IP Office manager. Thirdly, for each user who is wishes to utilize Email reading, the user's email address will need to be entered into the User profile details in IP Office Manager and the facility enabled through the email reading checkbox.

Campaign Manager

As part of Voicemail Pro, Campaign Manager enables the gathering of repetitive information (such as brochure requests) to be fully automated, leaving agents free to deal with other more complex calls which require human interaction. A definable sequence of recordings are played to the caller with time in between each recording to allow the capture of the caller's spoken answers and/or the caller's key presses via DTMF. At the end of the transaction the caller can be thanked and the completed transaction retrieved by an agent via a web interface or a short code.

Campaign Manager allows calls in queue to "break out" of the queue, or be directed in an "Overflow" situation to complete their transactions thereby increasing customer satisfaction by effecting an 'answer' to their call. This ensures that a minimum of customers give up when forced to wait in a queue or even worse, hear a recorded message stating that they are calling outside of "office hours".



Call Recording

Voicemail Pro also offers 'Recording Services' which allow the automatic/manual recording of calls for a variety of applications, such as for training purposes or to monitor abusive callers. As standard, recordings can be directed to the called extension's voicemail box or to any other mailbox for later retrieval. Alternatively, recordings can be stored in a central database for retrieval through a Web based browser by using ContactStore for IP Office.

The system administrator can select whether all calls are required to be automatically recorded or just a selection of calls. Alternatively, calls can be manually selected for recording. If for any reasons resources are not available then a recording may not be taken (for example all Voicemail Ports are busy).

Voicemail Pro provides a number of methods for triggering the recording of a call.

Most of the settings and controls for automatic voice recording are accessed through the IP Office Manager application. The proportion of incoming and/or outgoing calls that should be recorded and the time-period during which Voice Recording should operate can be selected.

- **User Recording:**
  The calls to and/or from a particular user can be automatically recorded. By default the recordings are placed in the user's mailbox.
- **Hunt Group Recording:**
  The calls to a particular hunt group can be automatically recorded. By default the recordings are placed in the hunt group's mailbox.
- **Account Code Recording:**
  An account code can be applied to a call by the user before it is made or during the call. One can also be applied automatically through CLI matching. Both incoming and outgoing calls which use a particular account code can be automatically recorded.
- **ICLID/CLI Recording:**
  Account codes can be assigned to a call by ICLID/CLI matching. This allows recording to be based on a ICLID/CLI match.
- **Time Profiles:**
  For each user, hunt group and/or account code, an IP Office time profile can be used to determine when auto-recording is used.

Call recording uses the conference facility and so is subject to the conference restrictions of the IP Office system. For some situations, it may be a requirement that call parties are advised that their call is about to be recorded. This is done by switching on the Play Advice on Call Recording option.

IP Office ContactStore

The standard Call Recording facilities provided with IP Office and Voicemail Pro can be extended further by using IP Office ContactStore. IP Office ContactStore stores and catalogs recordings so that they are easily accessible for later retrieval. Any recordings that you instruct Voicemail Pro to "send to the Voice Recording Library" are placed in a database.

IP Office ContactStore is provided with the Voicemail Pro software CD set and has an inbuilt 45 day trial license. A fully featured IP Office ContactStore system can be installed and used for 45 days from the creation of the first recording. After this time the system will stop taking recordings until a license is purchased and installed onto the IP Office.

IP Office ContactStore has a number of components, these are:

- An MSDE database into which details of all recorded calls are inserted.
- A browser-based call search and replay application.
- A browser-based system configuration and status monitoring application.
- Disk space management - Oldest recordings are automatically deleted as needed.
- Optional archive management - Recordings are automatically written to a DVD +RW drive.

To allow you to search for calls easily, the details of the recordings are stored within a MSDE database. It contains one record for each call recorded and additional records for each party on the call and the owner of the call. The information that is held for any recording is:

- A unique reference for the recording
- The start date and time
- The duration of the recording
- The name and number of the parties on the call—where this was available to IP Office (through ANI, CLI or DNIS) at the time of the call.
- The direction of the call (incoming, outgoing, or internal)
- The owner of the call recording
- The target or dialed number, which may be different from the number that actually took the call.

Recordings within IP Office ContactStore are stored as .WAV files. IP Office ContactStore uses the G.726 16kbps ADPCM compression standard, which provides the best compromise between storage capacity and CPU loading. IP Office ContactStore is designed to perform compression as a background task, which does not impact the systems ability to record, search or play other calls. It takes approximately 1 minute to compress a two hour recording. The compressed recordings are stored as 16kbps G.726 format, storage requirements are therefore 8MBs per hour of recording.

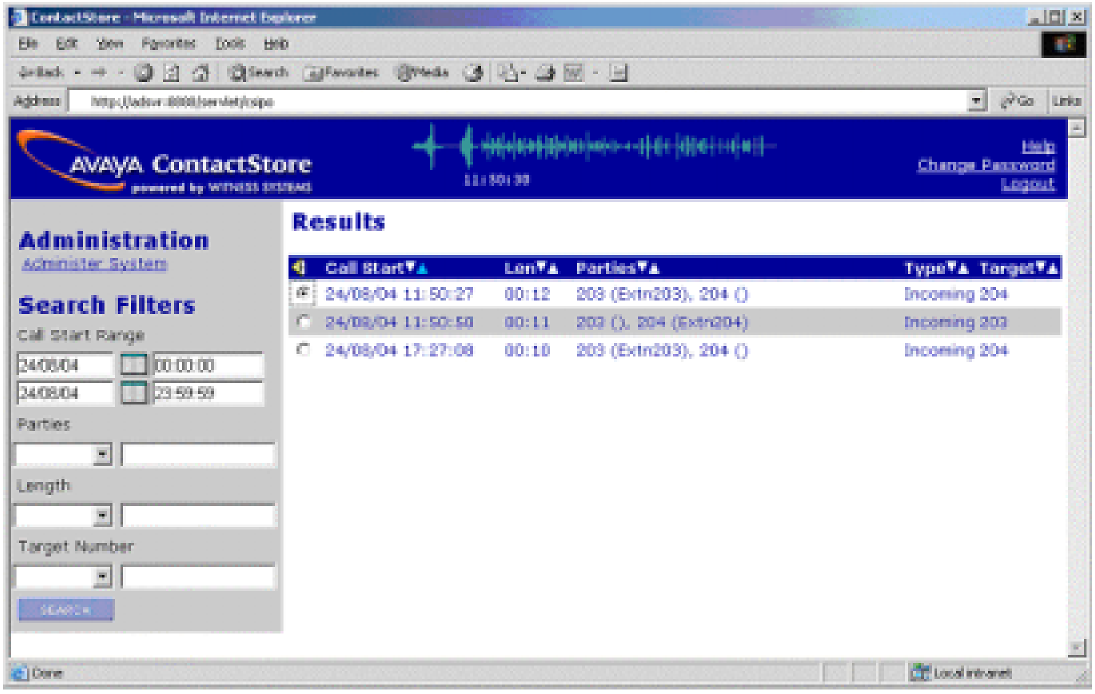
The IP Office ContactStore suite can be installed onto the same server as Voicemail Pro but must be loaded onto a separate partition. Alternatively, IP Office ContactStore can be installed on a separate drive within the same server or on a separate server. The minimum PC specification when Voicemail Pro and IP Office ContactStore are installed on the same server is detailed in the Voicemail System requirements later in this chapter.

IP Office ContactStore stores recorded calls with certain security in place. Access to recordings is strictly controlled according to the security constraints configured within the System Administration pages. Each recording has an owner; the call owner is the number of the extension that recorded the call. You can specify to which extensions each user has replay rights; the user can search for and replay all calls "owned" by those stations. Typically an individual may be given rights to replay calls owned by their extension number while managers may have rights to the extension numbers of all of their staff.

Any hard disk has limited storage capabilities. Once the available hard disk space is used, older recordings will be deleted, overwritten by newer recordings. To keep copies of recordings or to protect the recordings in the event of failure/theft/destruction of the hard disk on the recorder or to provide longer-term archive and replay capability, you can use a DVD +RW drive within the IP Office ContactStore server. With a DVD +RW drive installed in the ContactStore server, calls can be automatically archived. IP Office ContactStore requires the Nero DVD tools to write to the DVD drive. You must therefore use a drive that is supported by Nero. If Nero is not bundled with your DVD drive, you must install it separately. Single-sided 4.7GB DVD +RW media are supported.

The system will automatically generate alarms showing system warnings. Alarms are logged to IP Office ContactStore's database and held for a month before being purged. The administrator can define specific Email addresses for alarms to be automatically forwarded to. The email recipient could be a local system administrator, a manned help-desk and/or suppliers' support desks if you have a support agreement that includes this facility.

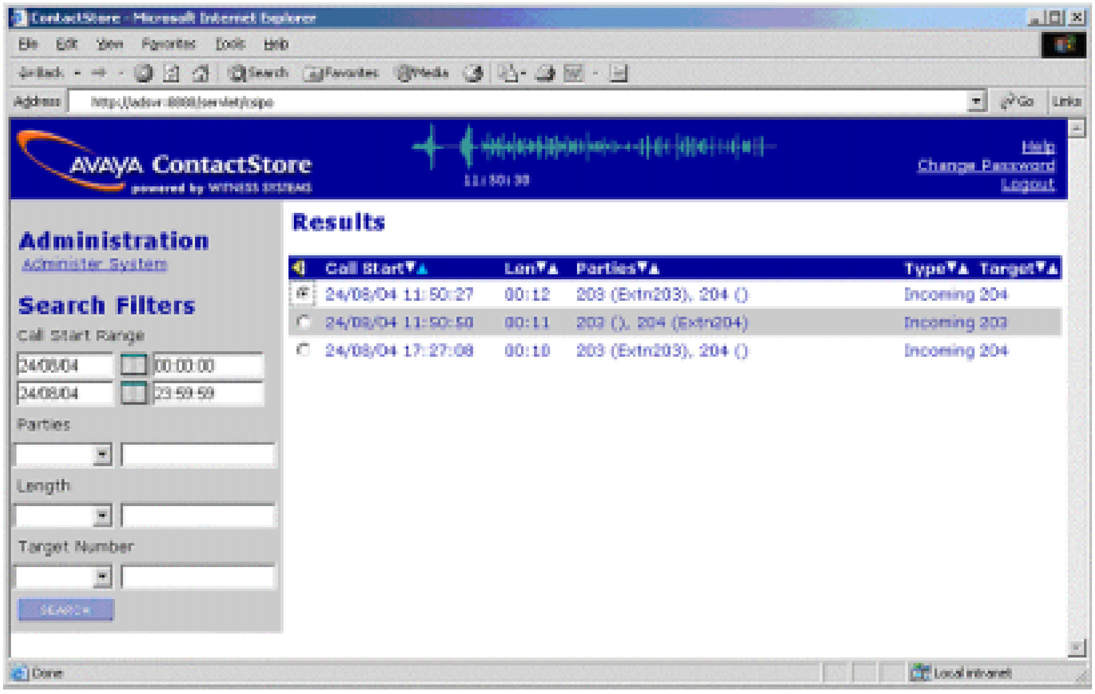
The system sends an email message each time an alarm occurs or is cleared. It also sends an email once per day as a "heartbeat" to let you know it is still operating. Failure to receive the daily heartbeat message should be investigated; it could indicate that the server has failed.



IP Office ContactStore allows replay of recordings by means of a browser-based application that is accessible with Internet Explorer (IE) V5.0 and higher. The Search and Replay facilities includes the following features:

- Personal security restrictions. The restrictions are applied as you log into the web server.
- Criteria-based search filter fields to perform specific searches.
- Replay controls. Use the replay controls to start, stop, pause, skip forward, skip backward, or to export the recording to a readily playable .wav file.
- Audio waveform display. The waveform presents a graphic representation of the audio content of the call. Use the waveform to avoid replaying static or silences, and to move easily to specific portions of a call.

The Search and Replay screen, shown below, provides filter fields that you can use to search for calls:



Voicemail Feature Comparison

Platform Support

	Embedded Voicemail	Voicemail Lite	Voicemail Pro
IP Office - Small Office Edition	Yes (uses in built VCM resources)	Yes	Yes
IP403	No	Yes	Yes
IP406 V1	No	Yes	Yes
IP406 V2	Yes (does not use VCM resources)	Yes	Yes
IP412	No	Yes	Yes

Capacities

Voicemail	Embedded Voicemail	Voicemail Lite	Voicemail Pro
Number of Mailboxes supported	No specific limit on IP Office - Small Office Edition or IP406 V2.	No Limit	No Limit
Maximum Number of Concurrent Calls (ports)	4 simultaneous calls on IP406 V2. From 1-10 simultaneous calls on IP Office - Small Office Edition depending up on available VCM resources	4 simultaneous calls on IP Office - Small Office Edition, IP403, IP406 V1/V2 and IP412	Up to 30 dependent on license & platform (IP Office - Small Office Edition=10, IP403=10, IP406 V1/V2 =20, IP412=30)
Recording Time	IP406 V2: Approximately 15 hours IP Office - Small Office Edition: 10 hours minimum	PC dependent (Requires 1MB per minute)	PC dependent (Requires 1MB per minute)

Features

	Embedded Voicemail	Voicemail Lite	Voicemail Pro
Runs as a service	No	No	Yes
Multi-lingual support	Yes	Yes	Yes
Voicemail for Individual users	Yes	Yes	Yes
Voicemail for Virtual users	Yes	Yes	Yes
Voicemail for Hunt Groups	Yes	Yes	Yes
Centralized Voicemail Services	No	No	Yes
Voicemail Ringback	Internal only	Internal only	Internal & external
Voicemail Help TUI	No	Yes	Yes
Message Waiting Indication	Yes	Yes	Yes
Integration with Phone Manager Pro	No	No	Yes
Personalized Greeting	Yes	Yes	Yes
Extended personal Greetings	No	No	Yes*
Continuous Loop Greeting	No	Yes	Yes
Forward to Email	No	Yes	Yes
Copy to Email	No	Yes	Yes
Listen To Email (Text To Speech)	No	No	Yes*
Send Email notification	No	Yes	Yes
Integrated Messaging & synchronization	No	No	Option
Save Message	Yes	Yes	Yes
Delete Message	Yes	Yes	Yes
Forward Message to another Mailbox	No	Yes	Yes
Forward to Multiple Mailboxes	No	Yes	Yes
Forward with a Header Message	No	Yes	Yes
Repeat Message	Yes	Yes	Yes
Rewind Message	No	Yes	Yes
Fast Forward Message	No	Yes	Yes
Pause Message	No	No	Yes
Skip Message	No	Yes	Yes
Set Message Priority	No	No	Yes*
Set automatic message deletion timeframe	No	No	Yes
Alphanumeric Data Collection	No	No	Yes*
Callers CLI/ANI, time & date announced	Yes	Yes	Yes
Call Back Sender (CLI/ANI)	No	Internal only	Yes
Remote Access to Mail Box	Yes**	Yes	Yes
User Definable PIN Code	Yes	Yes	Yes
Known CLI/ANI PIN Code By-Pass	Yes	Yes	Yes
Breakout to Reception	Internal only	Internal only	Internal & external

- *Intuity mode only.
- **Remote access can be provided via the embedded Auto Attendant on the Small Office Edition.

In-Queue Announcements

	Embedded Voicemail	Voicemail Lite	Voicemail Pro
Queue Entry Announcement	No	Yes	Yes
Queue Update Announcement	No	Yes	Yes
Queue Position Announcement	No	No	Yes
Estimated Time to Answer (ETA)	No	No	Yes
Exit Queue to alternative answer point	No	No	Yes

Auto-Attendant / Audiotex

	Embedded Voicemail	Voicemail Lite	Voicemail Pro
Multi-Level Tree Structure	Single tier option on IP Office - Small Office Edition and IP406 V2.	No	Yes
Message Announcements	No	No	Yes
Whisper Announce	No	No	Yes
Alarm Calls	No	No	Yes
Assisted Transfers	No	No	Yes

Other Features

	Embedded Voicemail	Voicemail Lite	Voicemail Pro
Call Recording	No	No	Yes
Test Conditions	No	No	Yes
Personal Numbering	No	No	Yes
Speaking Clock	No	No	Yes
Campaign Manager	No	No	Yes
Voicemail Pro Manager	No	No	Yes
Customized Voicemail	No	No	Yes
Intuity TUI emulation mode.	No	No	Yes
Forward Emails to External Systems (VPIM)	No	No	Yes
Third Party Database Access (IVR)	No	No	Yes
Text To Speech within call flows	No	No	Yes
Support for Visual Basic Scripts	No	No	Yes

IP Office Voicemail Pro Intuity Audix Emulation Features

Voicemail Box Feature	Intuity Feature support	Voicemail Pro support
Basic Commands		
*4 (or *H)	Help	Yes
*7 (or *R)	Return to main menu	Yes
*9 (or *W)	Wait	Yes
**6 (or **N)	Look up number/name	Yes
**9 (or **X)	Exit system	Yes
0 or *0	Transfer call to operator	Yes
*3 (or *D)	Delete	Yes
**8 (or **U)	Un-delete	Yes
**4 (or **H)	Hold message in category	Yes
*8 (or *T)	Transfer out	Yes
**7 (or **R)	Log in again	Yes
Options while listening to messages		
9	Increase speed	Not supported
8	Decrease speed	Not supported
4	Increase volume	Not supported
7	Decrease volume	Not supported
6	Skip forward	Yes
5	Skip backwards	Yes
*6	Skip to next message component	Yes
*5	Skip to previous message component	Yes
2 or (*2)	Rewind to start of message (skip to previous message)	Yes
3	Play back header after pressing 2	Yes
*1	Print fax or text	Available as an option but fax messages not currently supported
Options for addressing voicemails		
*2 (or *A)	Alternate between name and number addressing	Yes
*5 (or *L)	Use mailing list for addressing	Yes
Responding to a message		
0	Call the sender	Yes, provided Caller ID is provided.
1	Reply to the sender by voicemail	Yes
2	Forward with comment at beginning	Yes
3	Forward with comment at the end	Yes
4	Record and address a message	Yes
Main Feature Support		
1	Record/Send messages	Yes
2	Get messages	Yes
3	Create greetings	Yes
4	Outgoing and filed messages	Not supported
5	Personal Options	Support for options 1, 3-7.
6	Outcalling	Not supported.
7	Autoscan/Autoprint	Autoscan supported

PC Requirements

General Requirements

- If not already installed, an IP Office Feature Key Server must be installed. This can be installed onto the same PC as the Voicemail Pro Server.
- License for Voicemail Pro and any additional ports required. If Voicemail Pro server is installed without a license it will run for 2 hours and then shutdown.
- License for all aspects of Voicemail Pro being installed.
- IP Office Voicemail Pro CD.
- Installation on the same PC as being used for IP Office Manager is recommended.
- Switch off any PC and hard disk sleep, power down, suspend, hibernation modes.

PC Specification

The following configurations are supported.

- Windows 2000 with Service Pack 4 and Internet Explorer 5.5 or later.
- Windows XP Professional with Service Pack 2.
  Details of how to configure IP Office applications for operation with SP2 are contained within the IP Office Tech Tip Bulletin 49.
- Windows 2003.

The minimum recommended PC specification is as follows. Using a PC with a lower specification may degrade voicemail operation.

Applications	Minimum PC Resources	Intel Pentium	Intel Celeron	AMD	Notes
VM Pro	256MB RAM 2GB free disk space.	Any 1.4GHz.	Any 1.7GHz.	Any 1.4GHz.	To avoid replacing the server when adding new applications we recommend that a Pentium 4 2.8GHz (or equivalent) is used when possible.
VM Pro + IMS + Campaigns	512MB RAM	Pentium 4 2.8GHz.	Not tested	Athlon XP 3000+ All Athlon 64 chips.	
VM Pro + IVR + TTS	512MB RAM 20GB free disk space.	Pentium 4 2.8GHz	Not tested	Athlon XP 3000+ All Athlon 64 chips.	If the database being queried is located on the VM Pro server the query speed of the database will be affected by the amount of memory available. Please take into account the memory requirements of the database being queried.
VM Pro + ContactStore	512MB RAM 2GB free disk space.	Pentium 4 2.8GHz	Not tested	Athlon XP 3000+ All Athlon 64 chips .	ContactStore requires a separate disk or disk partition.
VM Pro + CCC	512MB RAM 10GB free disk space.	Pentium 4 2.8GHz	Not tested	Athlon XP 3000+ All Athlon 64 chips.	VM Pro and CCC can be run on the same server up to a maximum of 25 agents, 8 ports of VM Pro and on Windows server operating systems only.
VM Pro +CBC	512MB RAM 10GB free disk space.	Pentium 4 2.8GHz.	Not tested	Athlon XP 3000+ All Athlon 64 chips.	The client PC needs to be Pentium III, 800MHz with 128MB RAM minimum.
VM Pro + CCC + MMM	512MB RAM 10GB free disk space.	Pentium4 2.8GHz.	Not tested	Athlon XP 3000+ All Athlon 64 chips.	The database must be run with full SQL; MSDE is not supported with MMM.

1. Use of the **Large Fonts** setting is not supported. Use of this option may cause options on some screens to become inaccessible.
2. A 100Mbps network card is strongly recommended.
3. Free disk space requirements are also subject to the message storage required. See the "Disk Space" section below.
4. IMS and Web Campaigns options within Voicemail Pro are only supported on Windows Servers. Aspects of operation such as Voicemail to E-mail, Integrated Messaging Pro (IMS), Web Campaigns, etc, are subject to further requirements as listed in the following sections.

Network

The PC should be configured and tested for TCP/IP networking.

- We strongly recommend that the voicemail server PC is connected to the IP Office Control Unit directly or via a LAN switch.
    - If directly connected, changing the settings of the PC network card to match the IP Office control unit can resolve some issues. This should be done according to the PC or network card manufacturer's instructions. The options for IP Office LAN ports are:
        - **IP412:** Use LAN1 and half duplex.
        - **Small Office Edition and IP406 (V2):** Full duplex.
        - All IP Office LAN ports are 10Mbps/100Mbps auto sensing.
    - If not directly connected, using any of the above settings must be supported and matched by the intervening network equipment.
- The PC should have a fixed IP address. While PC's in a DHCP network may retain the same IP address between reboots this is **not** guaranteed.
- If the IP Office is acting as a DHCP server, it defaults to using 192.168.42.2 to 192.168.42.201 for DHCP clients. This leaves 192.168.42.202 to 192.168.42.254 for devices that require fixed IP addresses.

Disk Space

A compact or typical installation requires 500MB for the Voicemail Pro software. A full installation requires up to 2GB of disk space. However prompts and recorded messages consume an additional 1MB of disk space per minute.

- For Avaya IP Office - Small Office Edition, you can expect to require at least 200 minutes of message recording space, that is 200MB.
- For a busy environment you can expect to require at least 1,000 minutes of message recording space, that is 1GB.

Web Server Operation

If web browser access to campaigns is required, one of the following web servers must be installed on the server PC **before** Voicemail Pro. Note that both the Microsoft web server products run as services and require Voicemail Pro to also run as a service, that is on Windows 2000, 2003 or XP.

- Xitami Web Server.
- Microsoft IIS Web Server.
- Microsoft Personal Web Server.

Voicemail Email Connection

Voicemail Email operation is supported using either MAPI or SMTP. MAPI requires the Voicemail Pro server PC to have a MAPI compliant email client install. See Voicemail Email Integration.

If Text to Speech is installed, email text to speech is supported using MAPI.

In both cases above, full email sending from the server PC to users PC should be configured and tested before Voicemail Pro installation using the same PC user account under Voicemail Pro will be installed.

IMS Pro Connection

IMS requires the Voicemail server to use MAPI.

- Integrated Messaging Pro (IMS) is supported on Microsoft Exchange 5.5, 2000 and 2003. The R3.0GA release of Voicemail Pro does not support IMS operation with Outlook 2003 operating in cache mode. The R3.0 maintenance release will provide this support.
- An Exchange User account for user 'IMSAdmin' will be needed to as part of IMS installation.
- Must be a member of the same Domain as Voicemail Pro Server.
- A list equating Exchange User account names with voicemail box users.

ContactStore (VRL) Operation

The current IP Office Voice Recording Library (VRL) application is IP Office ContactStore. This application and its installation are documented separately. However:

- Avaya ContactStore for IP Office should be installed after Voicemail Pro has been installed and its operation verified.
- Avaya ContactStore for IP Office must use a separate hard disk partition for its message archiving from that used by Voicemail Pro for current mailbox messages. Use of a separate hard disk or installation onto a separate server PC are alternatives.
- The use of RAID 1 or RAID 5 are recommended.
- The use of a DVD recorder for long-term archiving is recommended.
- A figure of 7.2MB per hour of archived recordings is given.
- The archived messages held by IP Office ContactStore are accessed via web browser using the port address 8888. This port address is not configurable and so it is necessary to ensure that it does not conflict with any other web server service running on the same server PC.

12. Audio Conferencing

Why use Audio Conferencing?

A problem familiar to any organization is that of communicating effectively. As more and more people work from home or from dispersed locations, how do you ensure that your employees are planning and working together effectively, and regularly keeping in touch, when they are separated by time and distance? Also, many companies choose to sub-contract some services such as payroll, logistics or manufacturing to third-party suppliers. How do you ensure that you can act as one virtual enterprise? Audio conferencing provides a simple and effective solution.

Audio conferencing makes it easy to include key people wherever they are with minimum interruption from their work. It responds to business needs that every company faces:

- More meetings but less time available.
- Increasing pressure to be at two locations at once.
- Travel restrictions due to limited budget or risks (e.g. terrorism).

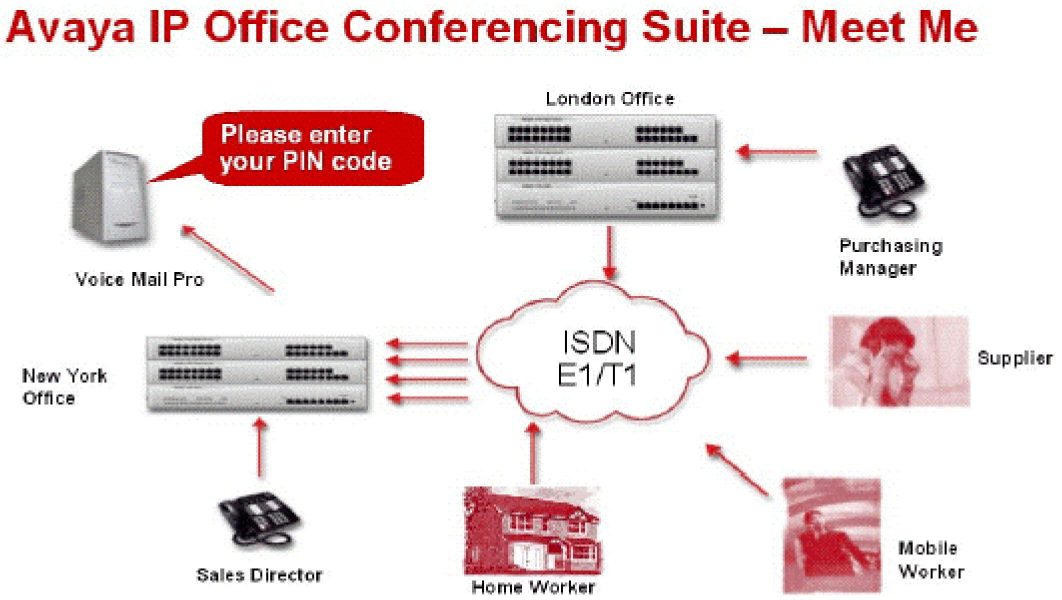
As a result of using conferencing, the benefits gained are:

- Reduction in travel, leading to lower costs and less wasted time.
- Increased worker productivity & personal security.
- More effective working practices, leading to shorter project times, and supporting dispersed organizations and complex supply chains.

Furthermore the Return On Investment (ROI) is very short as Meet Me conferencing is a built-in feature of IP Office. The typical ROI of just 4 to 6 months compared to Service Providers-based conferencing services based upon 2 hourly conferences with 5 participants per week.

IP Office Meet-Me Conferencing Solution

The built-in conferencing solution within IP Office enables multiple locations to participate in an audio conference. This allows on-site personnel as well as external parties (whether field-based engineers, sales staff on the road, customers or suppliers) to plan conference calls in advance or establish ad-hoc conference calls as and when required.



IP Office Voicemail Pro complements the built-in meet-me conference bridge facility on IP Office systems by adding guidance prompts as well as requesting PIN codes as participants enter the conference for security. For example, if conference calls are regularly scheduled, Voicemail Pro can have pre-programmed Call Flows for weekly conference calls e.g.: every Tuesday between 2pm and 5pm using PIN code 1234 is the weekly sales call, etc... Furthermore if multiple conference calls are scheduled, users can alternatively select which one they need to attend via a simple menu. Should users encounter any issues, calls can be automatically routed to the operator for assistance.

In addition (if CLI/ANI information is provided by the network), Voicemail Pro can allow CLI/ANI checks to be performed for further security.

Dial in prompts and PIN codes for security were added with Voicemail Pro Release 1.3. This facility simply requires the Voicemail Pro system license offering simultaneous access to 4 people i.e. 4 participants can type in the PIN or access a menu at the same time. Once they are on the conference bridge, Voicemail Pro ports are free for other users. If additional simultaneous accesses are required, further Voicemail Pro ports licenses (in steps of 2-port increments) can be added (up to 10 ports on IP Office - Small Office Edition and the IP403, 20 ports on the IP406 V1/V2 and 30 ports on the IP412).

If not already available, a PC will be required on which to install Voicemail Pro software (no voice cards required). This PC is connected to IP Office via the Local Area Network (LAN). Please refer to the Voicemail Pro section for the minimum PC specification.

IP Office Conferencing Capacity

The IP Office platforms provide maximum flexibility for conferencing. IP406 V2 systems can conference up to 64 parties at once or enable up to 21 three-party conferences, three 21-party conferences or any other equivalent combination.

IP412 systems can conference up to 42 three-party conferences or 2 x 64-party conferences. This capability is great for large briefings and, uniquely on IP Office, is made really powerful with the meet-me dial-in conference capability. This means you no longer have to rent expensive conference bridges from your service provider: instead IP Office can host the conference for you.

- Note: The term conference party refers to both internal and external callers.

Control Unit Conference Capabilities

The following tables show the maximum number of participants when calling via the different types of interface available on IP Office:

North America			
Maximum Participants	**Small Office Edition**	**IP406 V2**	**IP412**
T1/PRI-T1	3	64/64	96/92
IP	3	30	60
Internal users	3	64	2x64
Total max.	**3**	**64**	**2x64**

Rest of World			
Maximum Participants	**Small Office Edition**	**IP406 V2**	**IP412**
ISDN	3	640	120
IP	3	30	60
Internal users	3	64	2x64
Total max.	**3**	**64**	**2x64**

Notes:

1. **Analogue Trunk Restriction**
   In conferences that include external analog line calls, a maximum of two analog line calls are supported per conference.
2. **External participants**
   Each external caller requires a digital trunk/VoIP channel (for example 1 T1 allows 23/24 external parties, 1 E1 allows 30 parties and a VCM-20 allows 20 parties).
3. **Use of conference resources by other features**
   System features such as call intrusion, call recording and silent monitoring all use conference resources, as does automatic recording if enabled. When any of these features are active, the number of slots available for conference parties is reduced.
4. **The IP412 supports two 64-party conference banks**
   When a new conference is started, the bank with the most-free capacity is used for that conference. However once a conference is started on one conference bank, that conference cannot use any free capacity from the other conference bank (i.e. no more than 64 parties in any one conference).

IP Office Standard Conferencing Features

The IP Office provides the following features and benefits relating to conferencing:

- **No special conferencing equipment required**
  You only need an IP Office system unit with as many digital trunks/VoIP channels as external participants (as well as Voicemail Pro should PIN code/menu prompts be required).
- **Ease of use**
  Simply dial the direct number allocated to the conference bridge, type in the PIN if required and you have joined the conference! (PIN codes require Voicemail Pro).
- **Conference control from IP Office Phone Manager Lite and Pro**
  For ad-hoc conferences with a few participants, staff can easily set up immediate conferences by calling externally parties and bringing them to the conference bridge. Thanks to IP Office Phone Manager, the instigator of the conference can keep control: the CLI/ANI number (and the associated name if recognized) of each participant is displayed within the Conference tab of Phone Manager. If required, he/she can selectively hang-up a specific participant.
- **Phone commands**
  External participants can access pre-arranged conference calls from any touch-tone phone and enter PIN codes or select a menu (requires Voicemail Pro).
- **Customized greeting**
  Record a personalized greeting per conference (requires Voicemail Pro).
- **Conference entry/exit tones**
  Single beep on entry/double beep on exit
- **Conference call recording**
  Manual recording initiated by user on IP Office via Phone Manager, digital/IP display phone or a short code (requires Voicemail Pro)
- **Security**
  To prevent unauthorized access to the conference bridge, PIN codes, CLI/ANI number screening as well as time & date profiles can be set-up using IP Office Voicemail Pro.
- **Privacy**
  In cases where the security of calls is critical, in-house conferencing is the only way to ensure privacy.
- **Remote Management**
  enables a single person to manage the conferencing bridge facility from any location. Furthermore the full IP Office solution - phone system, voicemail, CTI server, router, firewall and DHCP server- can all be managed from a single management interface called IP Office Manager.

Conferencing Center

Introduction to IP Office Conferencing Center

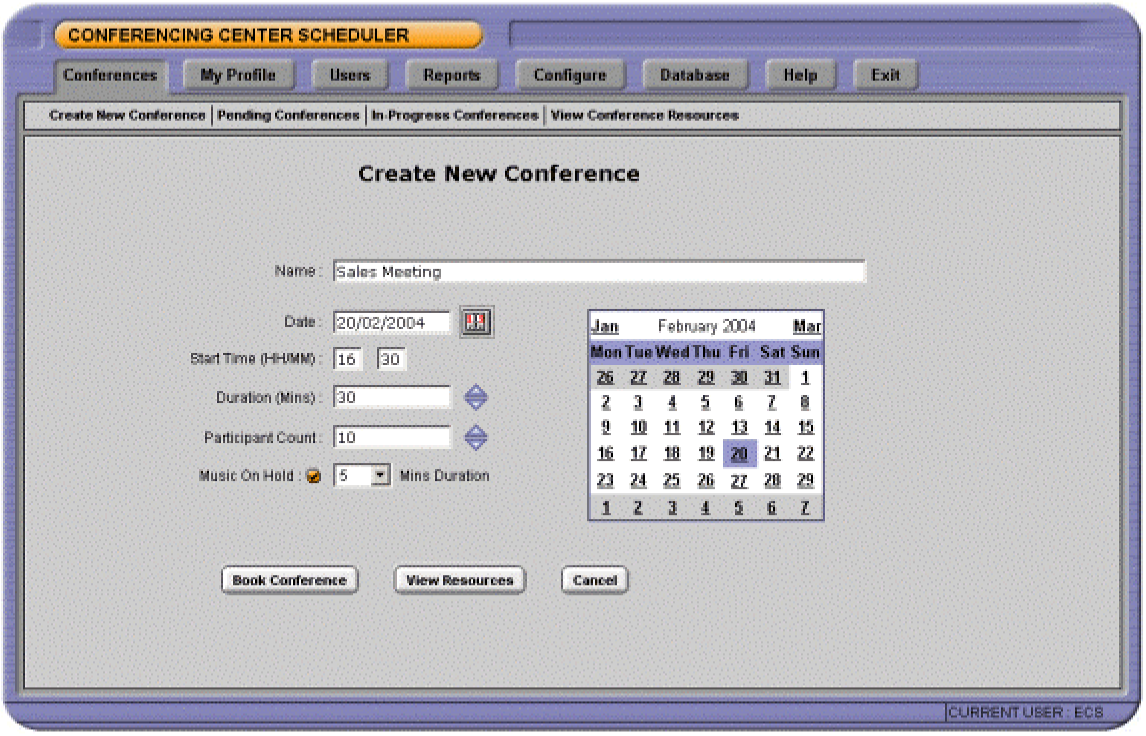
The integrated conferencing functionality on IP Office can be greatly enhanced by adding Conferencing Center. This optional application is a web-based software package that consists in two parts:

- a "Conferencing Center Scheduler" to book and reserve conferences.
- a "Conferencing Center web client" to complement an audio conference with a web interface.

The scheduler can be used independently of the web client. Conferencing Center also interacts with SoftConsole and Phone Manager.

Conferencing Center Scheduler

The Web Scheduler allows registered users to create and book conferences online using a web client interface. The Scheduler offers secure conferencing while being very easy to set up. Users simply enter the date, time, duration and the number of participants required. The conference is then created if the resources are available for that specific time. Once reserved, the conference resources are allocated to that conference call for the specified number of participants at the selected date and time. Additionally Music On Hold (if available on the system) can be played while waiting for a conference to start.



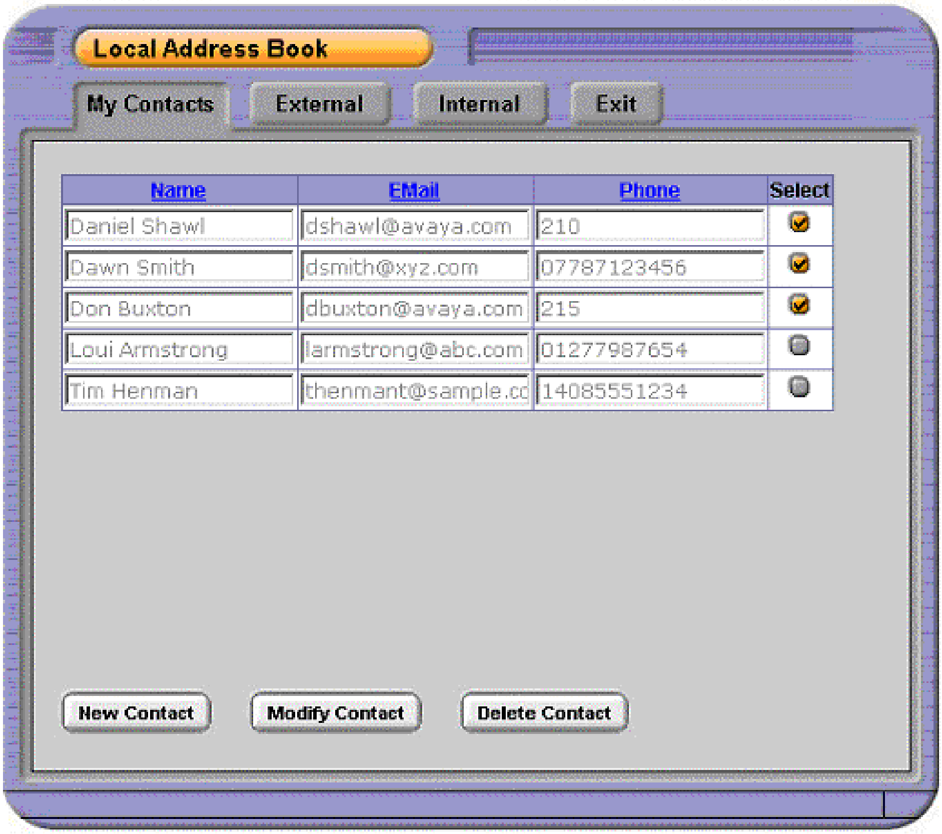
Access to the Web Scheduler requires a user to be set up by the administrator and have Internet Explorer (6.0 or above) installed on their PC. No other software is required. The System Administrator can set up an unlimited number of registered users on the Conferencing Center application. Once registered, users can review the system resources before booking a new conference, book a conference as well as list pending conferences they have previously set up.

The user setting up the conference can then add participant details including their email address and their telephone number. This allows email notification to all participants confirming the conference call details including the conference name, description, host contact details, bridge number, conference ID, their unique participant PIN code (if PIN checking has been selected) and the URL web address for the web client (if web support has been selected). Participants' details can be amended prior to the start of the conference.

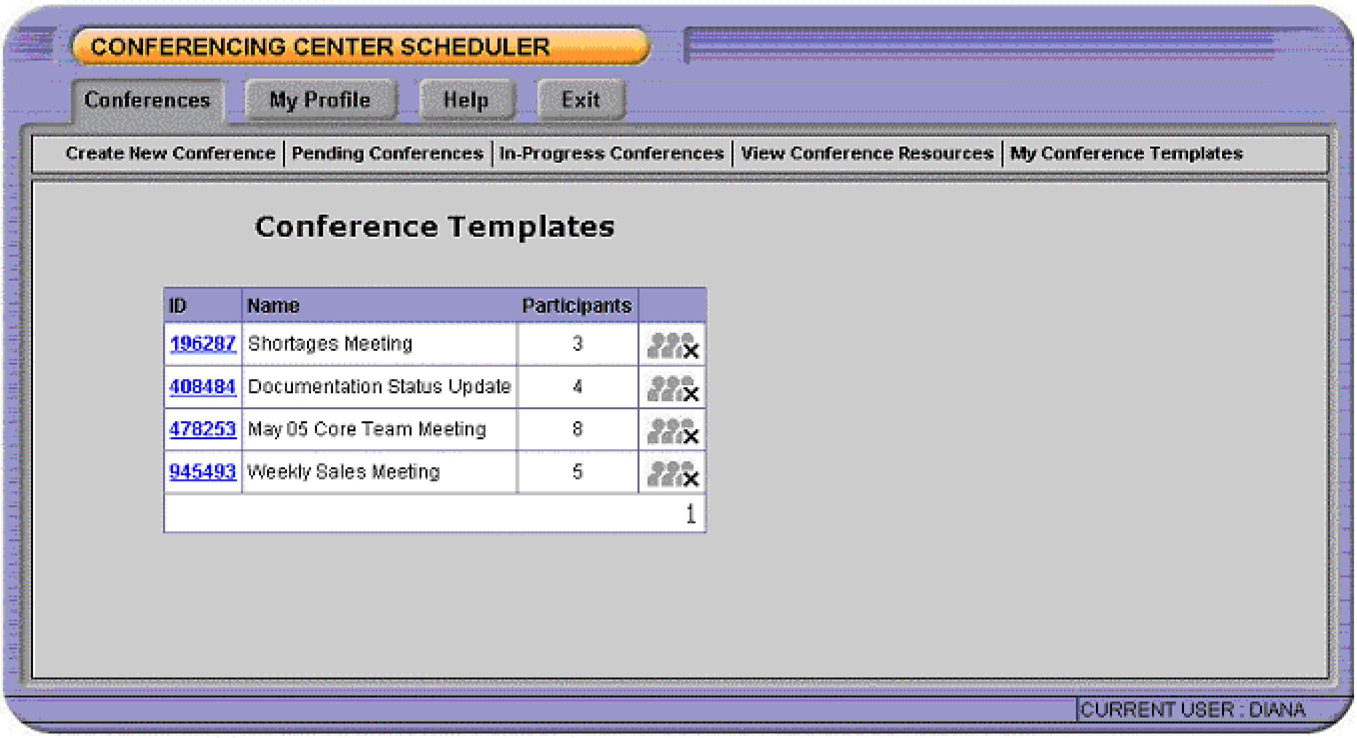
Voice Conferencing Notification (VCN) can also be activated for selected participants. This allows Voicemail Pro to dial out to participants when the conference is about to start and bring them to the conference bridge if they are available.

Advanced security is available by generating unique PIN numbers for every participant allowing them to be recognized by the system and displayed on the Conferencing Center Web client (if selected – see paragraph below). If announcements are required, Voicemail Pro can announce each participant by asking them for their name which is then announced to all participants already on the bridge. Similarly at the end of the conference, each participant leaving the conference will be announced.

A local address book facility is available to provide a convenient method of managing conference contacts and using these contacts when booking a conference. The address book can be accessed in two ways, either from the 'My Profile' tab or from the Add/Update Conference Participants process.

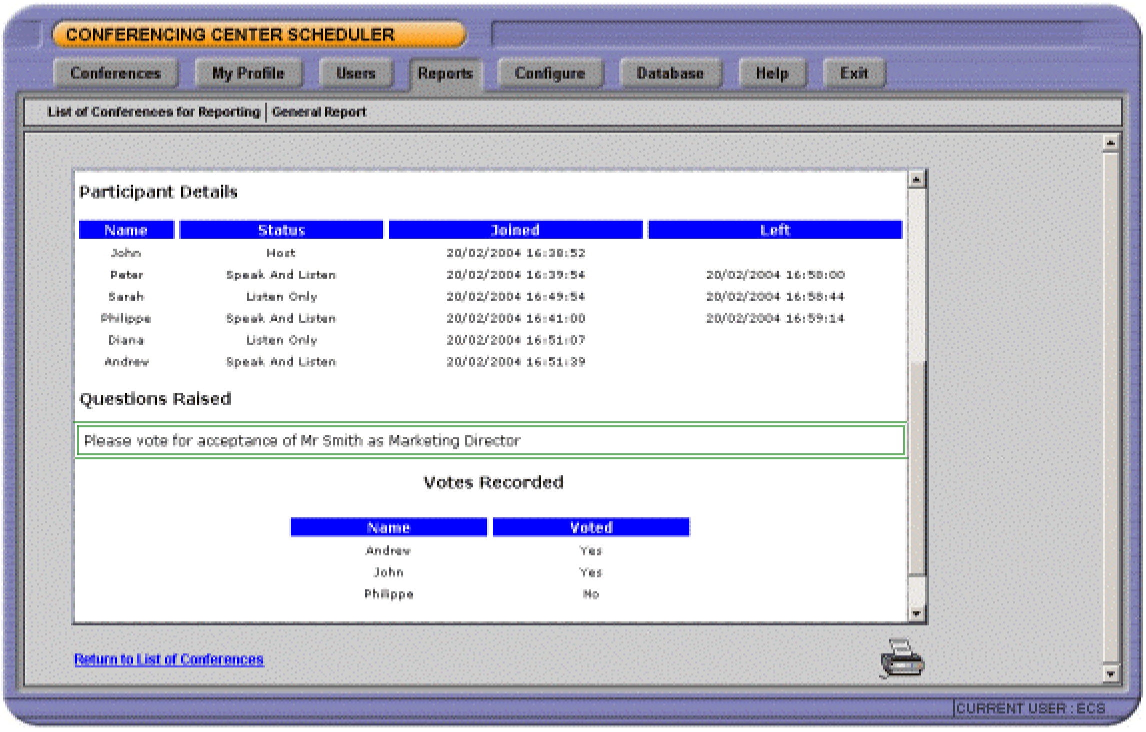


Conference templates can be used to book recurring conferences, all booking information including the conference ID and participants PINs are retained, except for the conference date. Using a conference template in this way can save re-entering of repetitive information thus saving time and effort. Once a template has been created they can be accessed via the 'My Conference Template' tab:



Conferencing Center Reporting

The System Administrator can generate reports regarding conference usage and individual conference reports. This will detail the conference name and ID, the start date and time, duration and number of participants. Individual reports can also be run listing participant details and when they joined/left the conference (if PIN codes were used). Additionally if voting was being used using the Conferencing Center Web Client, voting results for each participant would be shown for each question asked during the conference call.



In summary, the Conferencing Center Web Scheduler offers the following:

- Web-based booking tool to reserve conference resources (immediate or future).
- Ability to select “Listen-only” or “Speak & Listen” mode for each participant.
- Email notification to all participants.
- Voice Conference Notification (VCN) to dial out participants.
- Participants name announcements as they enter/leave the conference bridge.
- Unique computer-generated Conference ID for security.
- Unique PIN code for each participant for security and authentication.
- Web-based reports on conference usage and voting results.

Conferencing Center Web Client

To complement the audio-conference, the host has the ability to set up web support. This offers a web interface where the host and participants can not only see which participants have joined the conference but also whether they joined as audio-only or both audio and web. Privileges to a participant are limited to voting only, however a host to the conference has the ability to pose questions, modify participant contact details and whisper to a single participant connected into the conference.

Using the Web Client, the host can also modify participants status from listen-only mode to speak & listen mode or vice-versa. He can also mute all participants at once which is very useful if running staff briefing or product training sessions. Once the session is over, the host can then un-mute all parties when all participants will revert to their original status (listen-only or speak & listen) to allow a question & answer session for example. When in listen-only mode, participants can request the right to speak through their Web Client (raise hand function).

The host can also publish a document for review on the Web Client by all participants. This would need to be a document saved in html format (for example a PowerPoint presentation or an Excel spreadsheet) or simply a website URL. When presenting the document, the host has the ability to synchronize the document view to all participants (e.g. change slide) as long as he resides within the same domain as the Conferencing Center server (this is a Microsoft limitation).

Participants can be located anywhere on the Internet or across an extranet as long as they have access to the Web Server running the Conferencing Center application.

Access to the Conferencing Center Web Client simply requires the participant to have Internet Explorer (6.0 or above) installed on their PC. No download of the application is required. There can be as many web clients as there are participants on the conference call (that is up to 64 maximum per conference). For security, access to the Web Client also requires the participant to logon using the Conference ID and their unique PIN number. This offers security and allows the system to recognize who joined the conference and display its name on the right-hand side of the screen.

In summary, the Conferencing Center Web Client offers the following:

- Real-time view of participants status (Dialed in, Logged on to Web client, Speak & Listen, Listen Only).
- Ability for the host to change participant status in real-time.
- Ability for participants in listen-only mode to request the right to speak (raise hand function).
- Mute All / Un-Mute All facility for the host.
- Whisper facility for the host to have a private conversation with one of the participants.
- Viewing area for reviewing PowerPoint™ presentations (or any other document saved in html format).
- Questions & Voting facility.

SoftConsole Conferencing Center Integration

An operator equipped with the SoftConsole PC-based application can set up ad-hoc conferences via drag and drop if the participants are internal using the Busy Lamp Field (BLF). Voicemail Pro will then contact the participants and bring them to the conference. External participants are called by the operator and transferred to the conference. Using the SoftConsole application, the operator can transfer a call to an ad-hoc conference or a to a conference created via Conference Center Please refer to the SoftConsole section for more information.

Phone Manager Conferencing Center Integration

Phone Manager users can join a conference or book a conference via the Conference Center application simply by clicking on the relevant icons within Phone Manager. This will launch the Conferencing Center Web Client and the Conferencing Center Scheduler respectively. Note this feature is only available if permission is specified by your system administrator and if the Conferencing Center system is available.

System Requirements for Conferencing Center

Conferencing Center Server:

- Pentium 4 2.8GHz above with 512MB RAM running Windows 2000 Server or Windows 2003 Server (Windows XP Professional and Windows 2000 Professional could be used but would typically support only a maximum of 10 web clients).
- Microsoft Internet Information Service (IIS) installed capable of supporting as many web clients as required (please refer to Microsoft for licensing).
- 80MB of free disk space.

Conferencing Center Web client:

- Internet Explorer 6.0 or higher.
- No download required.

13. The Contact Center

IP Office Contact Center/CRM Solutions Overview

Avaya provides Customer Contact solutions that meet the needs of the small to medium business. From the smallest company that requires the aptly designed Compact Business Center, to the larger enterprises that need advanced routing and multimedia integration with the Customer Contact Center, Avaya provides a robust solution whether you have 5 staff or a formal contact center of 75 agents.

Here is a brief overview of the offerings for the IP Office communications platform:

- Compact Business Center
- Compact Contact Center

Compact Business Center

IP Office Compact Business Center is an entry-level management tool for small customer facing departments, typically handling anywhere from 2 to 15 agents. It provides graphs on real-time and historical information (up to 31 days) for up to three groups. It provides information on key performance indicators of the business - lost calls, trunks free, agents free and queuing time.

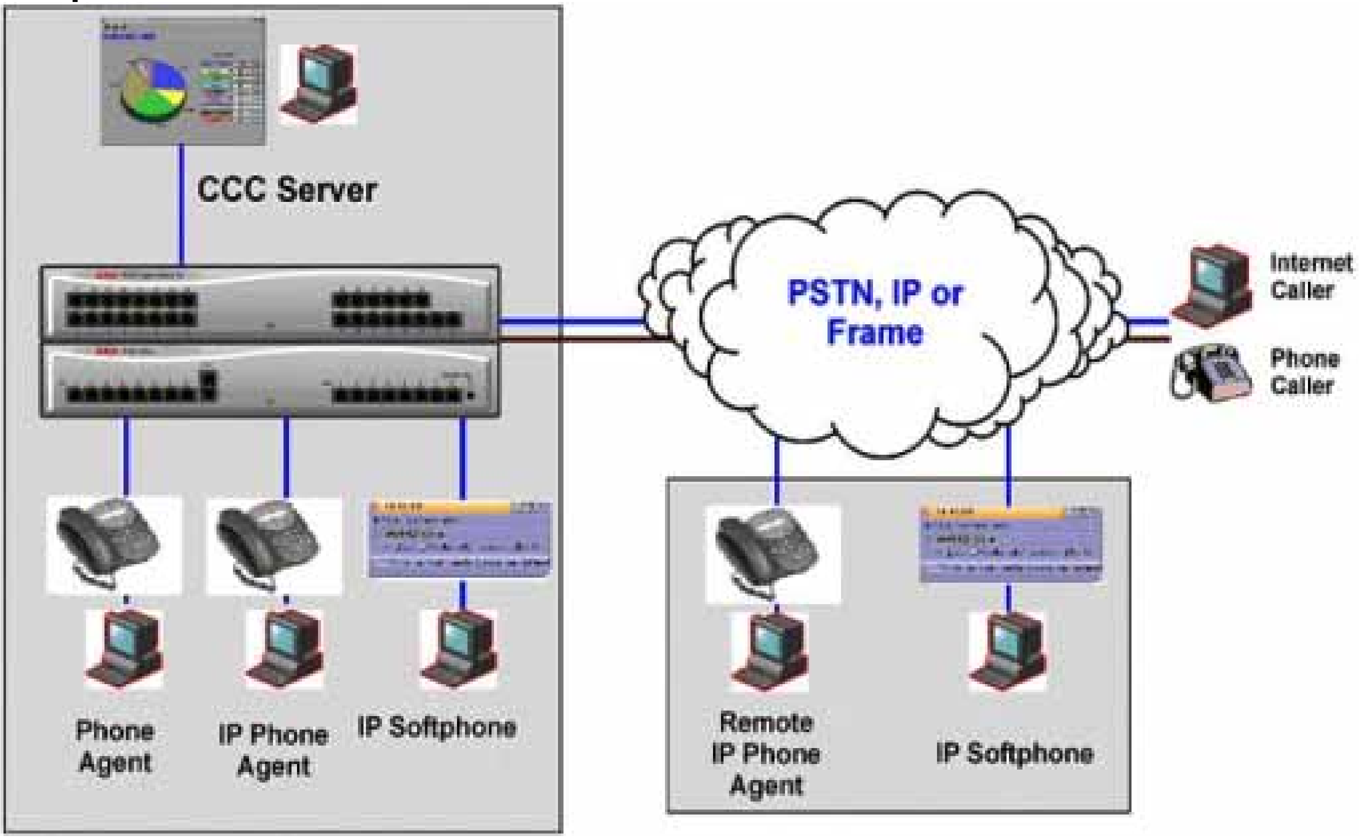
Compact Contact Center

IP Office Compact Contact Center is a highly modular contact center solution catering for all contact center sizes from 2 to 75 agents. The following modules are available as part of the CCC software application:

- **Compact Call Center (CCC) Server - Base System (IPO CCC SVR RFA)**
  Provides one supervisor position, with real-time information view, management by exception, plus historical reports for any aspect of the contact center. Up to 73 standard reports can viewed or printed. Also included are reporting capabilities for 5 agents and one license for a PC Wallboard (PCWB) application.
- **Agent & Site Management (Real Time)**
  - **Real Time Supervisor Monitoring (Call Center View – CCV)**
    As many as 21 supervisor positions (CCV) can be purchased in total for CCC (please note: MSDE installations can only be supported up to 5 supervisor positions). This provides a supervisor with the ability to monitor in real time the service being provided to callers. There are up to 12 separate real-time graphs that can be viewed by the supervisor. Alarms also appear in real time forcing the supervisor to acknowledge them as they occur.
  - **Phone Manger Pro: Agent Enabled**
    Provides agents with a PC CTI application where they can log in, join groups, and go into busy status when they are away from their desks for short periods. Provides a lower cost of ownership as proprietary handsets are not required. For IP users, Phone Manager PC Softphone can be used in agent mode as well, without the use of a handset. Please refer to the applications section for more information on Phone Manager Pro.
  - **Alarm Reporter**
    Alarm Reporter has been designed to enhance the exception management used by Call Center View (CCV). The Alarm Reporter enables the contact center supervisor to look back on the performance of the contact center, on a daily or weekly basis, by reporting on certain criteria predefined by the contact center supervisor.
- **Historical Reporting**
  The Compact Contact Center archives all call center interactions (telephony or multimedia) to a central database (MSDE or SQL). This provides a rich set of standard reports to the business, and the capability to create custom reports.
  - **CCC Reporter**
    The system can allow up to 20 separate Report Viewers within the contact center (for MSDE installations, up to 5 viewers are supported). Access to the standard reports is a thin client (new for version 5) application based on Crystal Reports. Up to 73 standard reports are available, with the ability to create up to 3 custom reports out of the box! (Requires additional software, see Custom reports below). Reports can also be exported to a variety of formats, including Excel, CSV, HTML, and PDF.
  - **Report Scheduler**
    All historical reports created within CCC can be scheduled for individual delivery to anyone via email. Reports can also be scheduled to multiple network printers.
  - **Custom Reports (new for CCC V5)**
    Starting with Compact Contact Center version 5, all reports are Crystal Reports™ based. This provides a much richer experience for the small to mid-market customer, and creates an environment where custom reporting is made more accessible. To create more than 3 custom reports requires the designer license (IPO CCC DESIGNER RFA) AND a compatible version of Crystal Reporting software (Crystal version 9).
  - **Microsoft CRM™ Integrated Reporting**
    As part of Avaya's continuing relationship with Microsoft Business Solutions with the creation of the IP Office Customer Management solution, Compact Contact Center version 5 will provide a selection of reports that integrate the information between the two systems. IP Office Customer Management is the combination of Avaya IP Office CCC and Microsoft CRM to integrate all contact points within a business in such a way that will transform it.

- **MultiMedia Module (MMM)**
  The MultiMedia Module (MMM) provides CCC with new routing schemes. It will also provide combined reporting for all interactions within the contact center. IP Office ensures that an organization can implement and measure a service level agreement against all aspects of the customer contact process – for example ensuring that all emails receive a reply within half an hour of them being sent. There are several queue types supported in the MMM; they are:
  - **Email Queues**
    A multimedia agent would receive emails from the queue and reply via MS-Outlook™ client interface. Emails queuing can be supported on POP3 or MS-Exchange™ servers.
  - **Chat Queues**
    Utilizes the Microsoft Chat™ interface to route requests for real-time chat between a customer and agent, providing that much needed response, in real time, for your online customers.
  - **Web Callback**
    For those online customers wanting a real person to speak to instead of a chat session, web callback enables them to leave their callback number, once that is sent to an agent, the system can automatically call back the customer, providing a savings in time and better customer satisfaction, with increased efficiency.
  - **Proactive List Dialing (Preview Dialing)**
    Import your direct marketing lists to Excel based datasheets and the Proactive List will create an outbound campaign for your agents. Agents can accept particular contacts (if desired) or have automatic acceptance. Result codes can determine if the call was successful or if it needs to be placed back in the queue for rescheduling.
- **Wallboards**
  - **Fixed Wallboards**
    Fixed scrolling wallboards enable key statistics and messages to be displayed for everyone in the call center to see. Supervisors can send ad-hoc messages to wallboards to broadcast important information, or to make announcements.
  - **PC Wallboards**
    PC-based wallboards allow individual agents to see their own individual statistics, those for their group, or for the whole contact center. Agents can customize their view so that information is presented in the way most useful to them. In additional, supervisors can set particular messages to appear on PC Wallboards, as a motivational or informational tool. Please refer to the CCC System Administration manual for a complete list of variables available.
- **3rd Party Integration**
  - **Workforce Management Interface**
    Compact Contact Center works with a number of workforce rostering packages, including Blue Pumpkin and Qmax. This module enables the interoperability of these packages with CCC.
  - **Microsoft™ TAPI Integration**
    By utilizing either the 1st party or 3rd party TAPI support on IP Office, you are enabling your business to speak to a wide range of supported software packages (e.g. ACT! Goldmine) that increase the productivity of your agents and the profitability of your contact center.

Compact Contact Center Overview



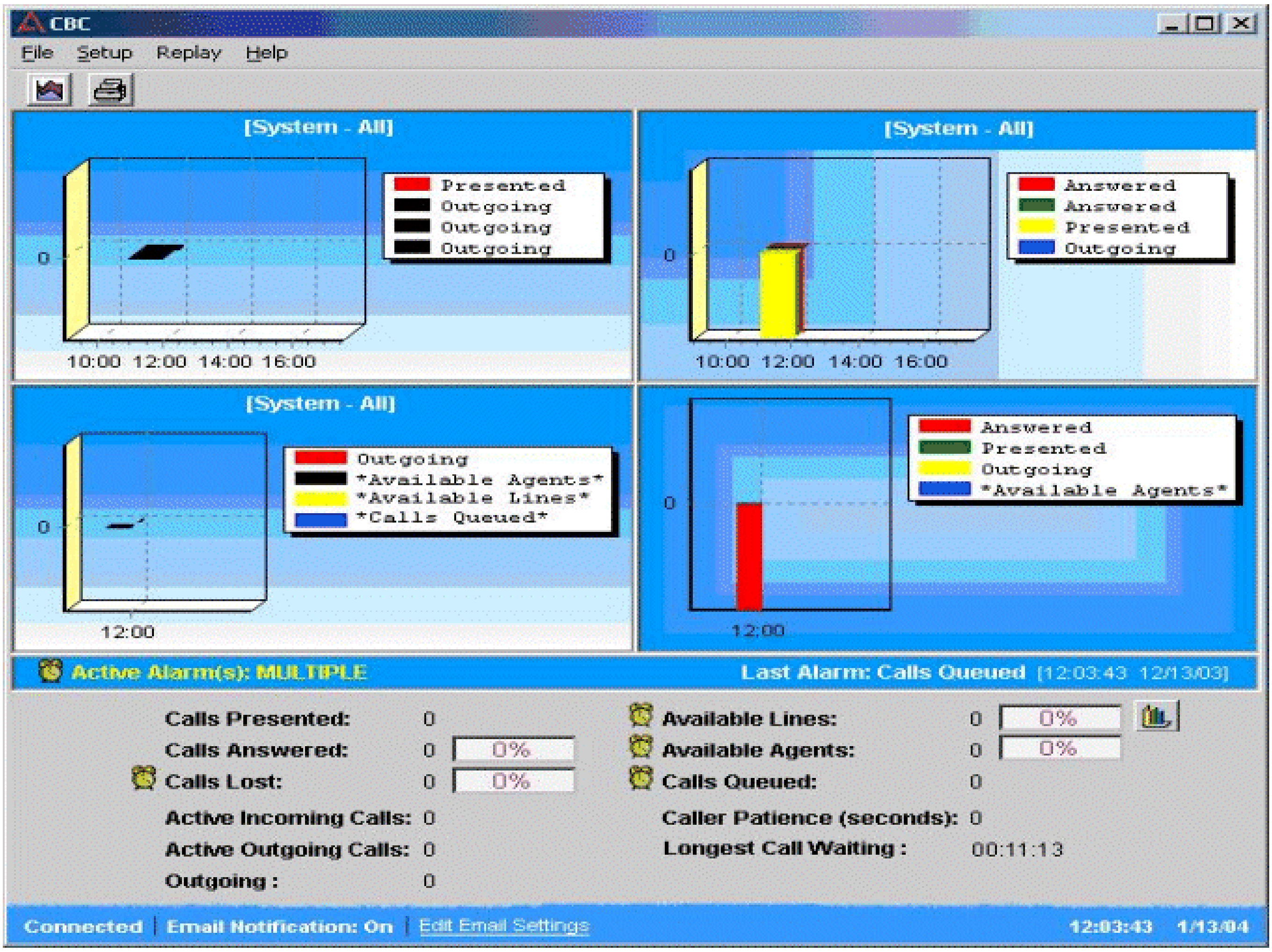
Compact Business Center

Compact Business Center

Compact Business Center is enabled by a license key (IP400 CBC RFA) and provides real time and historical analysis with export in CSV format to Excel or other reporting packages.

The system as part of a client/server relationship, the Delta Server (which is also the engine for IP Office SMDR and Compact Contact Center), is the engine that interfaces IP Office to CBC and associated applications, there are no license requirements on the Delta Server itself, just on the associated applications that run in conjunction with it. The client software (CBC) can be installed on any client workstation (as defined in the technical considerations section).

The Compact Business Center applications allow the user to create a maximum of 4 real time graphs, in any of 6 different graph types e.g. bar, pie, etc. These real time graphs display statistics for either the entire system or any three-departments/hunt groups.



Compact Business Center Example

CBC Real Time Information

In order to define the real time graphs the user may select 3 variables of their choice. The following variables are available:

- **Total Calls Presented**
- **Total Calls Answered**
- **Total Calls Lost**
- **Total Outgoing Answered**
- **Number of available 'Logged-on agents'**
- **Trunk Utilization**
- **Calls waiting**
- **Active incoming/outgoing Calls**
  The number of calls currently in progress across the entire system highlighting a snap shot view of call activity. This allows the user to have some insight into the balance between agent resource availability and call traffic load.
- **Caller satisfaction level**

It is possible to split these variables into two categories i.e. incoming and outgoing calls. These figures can be displayed permanently both in a numerical format and as a percentage of the total calls presented on the incoming side and all variables associated with outgoing. For example, outgoing answered as a percentage of the total outgoing calls made. A status bar provides a visual indication for each variable.

Historical analysis is provided by allowing the user to select the same variables, containing yesterday's data, so they can analyze the previous days performance against today's. Historical report capture can cover a maximum 31-day period. Data is stored in a CSV format enabling the export of the data into a reporting application that supports the CSV format e.g. Microsoft Excel. The advantage to the customer is the option to use the reporting package of their choice and not be restricted to one data mining report package.

Key Benefits

- **Lower TCO**
  Provides small businesses with robust contact center measurements produced in an easily understandable format.
- **Standards Based**
  Data is output to a CSV file format that is used by Microsoft Excel™. Customer can import format to other reporting applications.
- **Ease of Use**
  CBC's real-time charts are presented in an easily understandable graphical format, all information is contained in one single view, perfect for the small business.

CBC Alarms & Email Notification

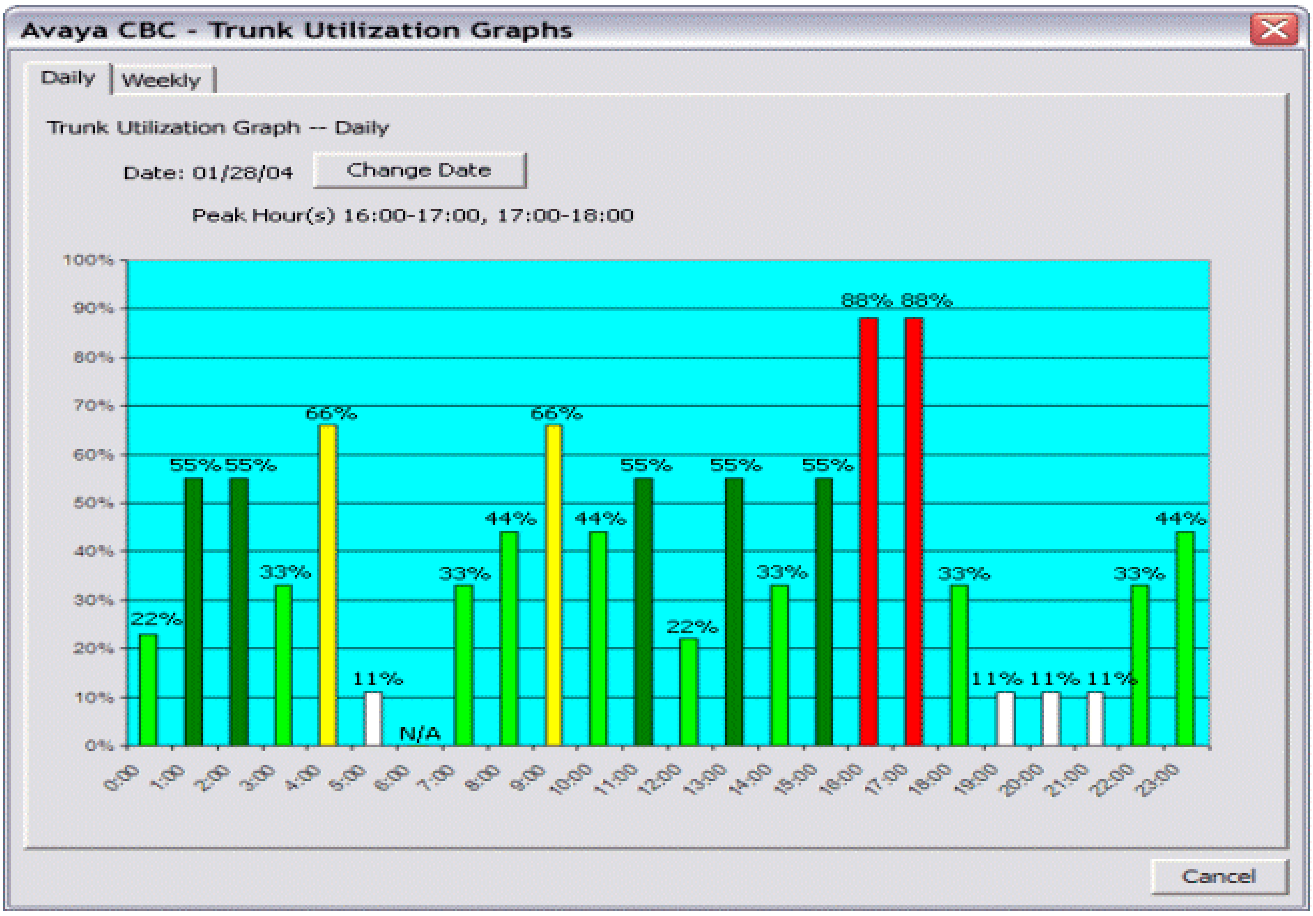
In order to provide those small enterprises with instant access to developing situations within their business, Compact Business Center V2.1 will provide alarms on the following pre-defined parameters:

- Lost Calls.
- Trunk Utilization (Available Lines).
- Calls Queued.
- Available Agents.

In addition to providing these alarms, CBC also provides email notification to key contacts in your business or installer, providing up to the minute status on the business. This feature is extremely useful for determining whether an increase in trunk capacity is needed, or if new employees are essential to future business.

Trunk Utilization Graph

Small Businesses quite commonly run on limited resources and budget, so it is absolutely critical for them to have precise data quickly before they make bigger investments in their communications architecture. The new Trunk Utilization Graph for CBC V2.1 is a perfect example of this principle. Previously, those businesses relied on guesswork and hearsay as to whether their incoming callers (and future customers) were experiencing busy signals, now with the Trunk Utilization Graph, a business can easily see when all their trunks are in use and what their busiest times of the day are. It even integrates with the email notification feature, so when all trunks in a business become unavailable, key people in the business know immediately.



Key Benefits of Compact Business Center

- **Lower TCO**
  Provides small businesses with robust contact center measurements produced in an easily understandable format.
- **Standards Based**
  Data is output to a CSV file format that is used by Microsoft Excel™. Customer can import format to other reporting applications.
- **Ease of Use**
  CBC's real-time charts are presented in an easily understandable graphical format, all information is contained in one single view, perfect for the small business.

Compact Contact Center

Compact Contact Center (CCC)

Compact Contact Center is a small to medium contact center solution that runs as a client/server application. It can support installations as small as 5 agents, up to a maximum of 75 agents. There are several integrated components within the CCC, they are:

- **Real Time (Management by Exception).**
  - **Call Center View.**
  - **Alarm Reporter.**
- **Historical Reporting.**
  - **CCC Reporter.**
    - **Standard Reports.**
  - **Report Scheduler.**
  - **Custom Reporting (Crystal Reports).**
  - **Microsoft CRM™ Integration.**
- **MultiMedia Module.**
- **Wallboard Server / Manager.**
  - **PC Wallboard.**
- **Workforce Management Interface.**
- **Queuing Announcements (Voice Mail Pro).**

The Compact Call Center is designed to provide a tightly integrated real time and historic reporting package and wallboard support for IP Office. CCC has been designed to allow customers to manage their contact center more effectively and improve the service they provide to their customers.

The product consists of a set of integrated modules, sharing a common database with IP Office. The benefit of this approach is that there is a single point of configuration, therefore the system is far easier to use and update than traditional call center management tools. The initialization of the multimedia component is designed around a common Microsoft Windows™ IIS server environment.

CCC is uniquely designed for those small and growing companies who want to meet the challenge of great customer service head on, while at the same time producing a manageable ROI back into the business. Giving small businesses the responsiveness to customer needs that was only previously available to large enterprises is what the CCC does best, whether you have an informal or formal contact center.

Call Center View

Supervisors in a contact center also serve as de facto human resource managers (especially in a small business), so accurate real-time data for managing agents is an imperative. Call Center View provides the customer with the combination of real time service monitoring and resource management, allowing a supervisor to balance and manage their resources (i.e. staffing levels against the traffic levels of incoming calls) and therefore improve customer service and reduce costs. Call Center View contains 18 real time screens showing all aspects of the Contact Center activity. Alarms may be set on up to 16 parameters per device, with three levels per alarms available, ensuring that a supervisor will automatically be informed should an exception occur, thus freeing the supervisor to continue with other, more productive activities.

CCV Supervisory Screens

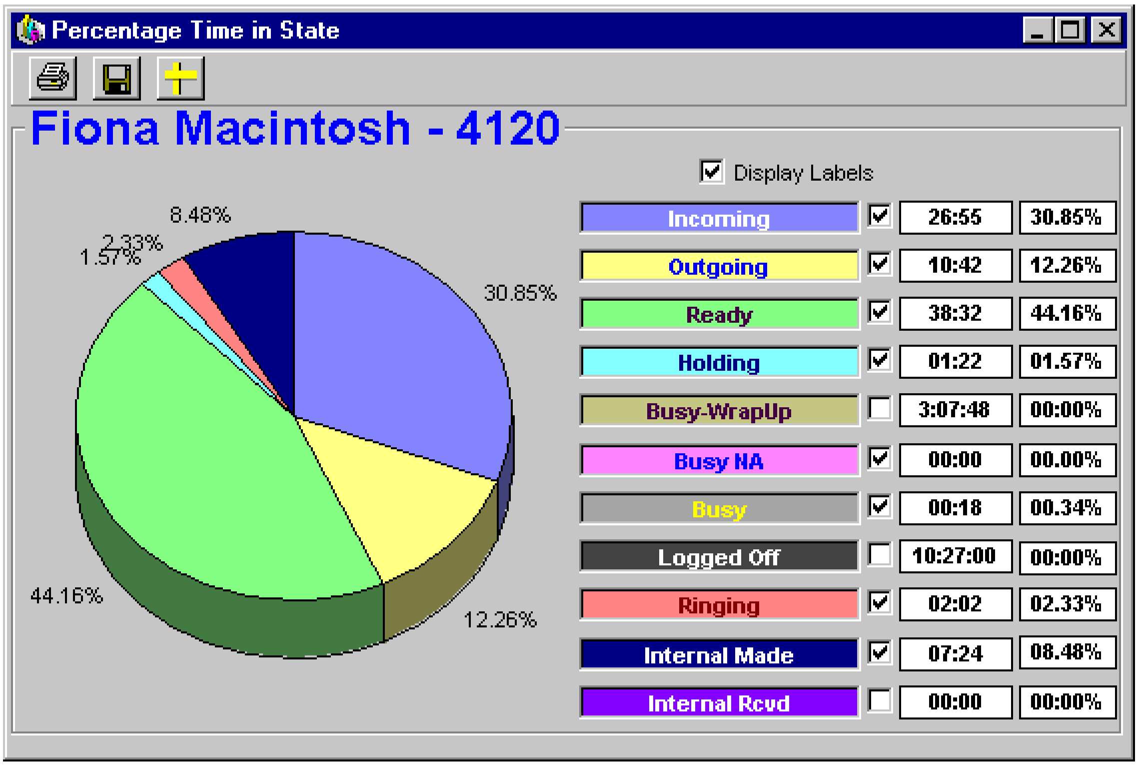
- Alarm Handling.
- BLF Details.
- Extension Activity.
- Callback Request.

Trunk Related Screens

- Trunk Group Monitor.
- Trunk Group Details.
- Real Time Status.
- Group Status (Percentage).
- Individual Trunk Details.

Agent and Queue Based Screens

- Group Monitor
- Agent Group Details
- Real Time Status
- Group Status (Percentage)
- Individual Agent Details
- Percentage Time in State
- Individual Group Details
- Queue Monitor
- Individual DDI/DID Details



Call Center View Real Time Example

Historical Reporting with Compact Contact Center

CCC Reporter

CCC Reporter provides in depth historical reporting on the customer facing department's activity. Report Manager provides up to 73 standard reports for measuring overall contact center call handling and individual/team performance. The data is retrieved from the archived database, which provides a source of data limited only by the hard disk space available (SQL only). These standard report templates may be formatted by the user top provide reports daily, weekly, monthly, or any defined time period. The standard reports can also be refined by individual, group, or trunk. CCC Version 5 now uses the Crystal Reports™ format, which provides ease of use and thin client operation for reporting.

Standard Reports List

1. Account Code Log by Agent Group (Graphical - All Media)
2. Account Code Log by Agent Group (Graphical)
3. Account Code Log by Agent Group
4. Account Code Log by DDI (Graphical).
5. Account Code Log by DDI.
6. Account Code Log by Pilot (Graphical)
7. Account Code Log by Pilot.
8. Account Code Log by Target (Graphical).
9. Account Code Log by Target.
10. Agent Activity Trace.
11. Agent Activity
12. Agent Callback Request.
13. Agent Group Busy Status.
14. Agent Group Graphical Summary (All Calls).
15. Agent Group Graphical Summary (All Media).
16. Agent Group Graphical Summary.
17. Agent Group Member Call Duration Report (All Calls).
18. Agent Group Member Duration (All Media).
19. Agent Group Member Duration.
20. Agent Group Tabular Summary (All Calls).
21. Agent Group Tabular Summary.
22. Agent Group Tabular.
23. Agent Individual (All Media).
24. Agent Individual.
25. Agent Tabular (All Media).
26. Agent Tabular.
27. Customer Tracking by Call Identifier.
28. Customer Tracking by CLI.
29. DDI Call Duration.
30. DDI Distribution by Target.
31. DDI Distribution
32. DDI Response
33. DDI Routing
34. DDI Summary.
35. External Transferred Account Code.
36. Incoming DDI Summary.
37. Incoming Duration Summary (All Media).
38. Incoming Duration Summary.
39. Incoming Pilot Summary.
40. Lost Call CLI.
41. Multi-Media Summary.
42. Outgoing Account Code Costing Log (All Media).
43. Outgoing Account Code Log (All Media).
44. Outgoing Account Code Log (Graphical).
45. Outgoing Account Code Log.
46. Outgoing Most Common Destination by Agent Group.
47. Pilot Call Duration.
48. Pilot Distribution by Target.
49. Pilot Distribution.
50. Pilot Response.
51. Pilot Routing.
52. Pilot Summary (All Calls).
53. Pilot Summary
54. Pilot Summary (All Media)
55. System Summary.
56. Target Graphical Summary (All Media).
57. Target Graphical Summary.
58. Target Member Duration (All Media).
59. Target Member Duration.
60. Transfer Call Tracking Detail by Agent.
61. Trunk Group Activity
62. Trunk Group Activity (All Media)
63. Trunk Group Busy.
64. Trunk Group Call Duration.
65. Trunk Group Response.
66. Trunk Group Summary.
67. VM Call Flow Monitor by Call Flow Name.
68. VM Call Flow Monitor by Topic.
69. VM Call Flow Monitor.
70. VM Summary
71. Proactive Report: By Agent
72. Proactive Report: By Campaign
73. Incoming Calls By Target Group
74. Plus 3 custom reports.
75. Plus 7 MS-CRM Reports.

Report Scheduler *(New Interface for Version 5)*

Report Scheduler allows reports to be scheduled to run at a specified date and time, or repeated at defined intervals. Supervisors can schedule reports to be delivered to various places within the contact center. Reports can also be delivered to multiple recipients via email in the following formats; PDF, CSV, XLS, RTF, RPT and Word format. Reports can even be scheduled for delivery to multiple printers within the network at the same time.

Custom Reporting (Crystal Reports) (New for CCC Version 5)

Custom Reporting allows the business to create site defined reports, or modify and change standard reports, providing total flexibility in the presentation of traffic and agent information. This capability is aimed at the contact center manager who requires a greater degree of flexibility to allow better informed decisions.

Compact Contact Center version 5 has now uses Crystal Reports as its reporting engine. This provides a much simpler experience for the user as now all reports are available over a thin client interface, no longer forcing the installation of large software components on a supervisor's desktop. This change has allowed CCC to modify how custom reports can be presented to a site.

Out of the box, all sites with CCC version 5 now have the ability to create 3 custom reports for their business, in effect; the CCC is delivering 3 free custom reports to a small business. For sites that desire to have more than 3 custom reports, a design license (IPO 400 CCC Designer RFA) is required.

Designing Reports Using Crystal Reports

One key difference with CCC version 5 from previous versions is the fact that the report designer software will no longer be carried by Avaya (The "Designer" license will still be utilized to allow customers to design three or more custom reports).

The implementation is designed to work with any Crystal Reports™ software package (using Crystal version 9). Crystal Reports is available in four different editions to meet the needs of application developers, IT professionals, and business users. The following is an overview of the types of Crystal products that can be used:

Application Development Solutions

- **Advanced Developer** – Web development and deployment bundle for integrating and deploying dynamic report creation and viewing capabilities into web applications.
- **Developer Edition** – For integrating report viewing, printing, and exporting capabilities into applications.

Report Design Solutions

- **Professional Edition** – For report creation and maintenance based on a large variety2 of data sources plus out-of-the-box web report delivery for workgroups.
- **Standard Edition** – For basic report design based on PC-based data sources.

The chart below illustrates some of the key feature differences between the various Crystal Reports 9 editions:

		S	P	D	A
Report Design	**DATA CONNECTIVITY**				
	PC-based and Microsoft® ODBC/OLE DB for MS Access and SQL Server	●	●	●	●
	XML		●	●	●
	OLAP		●	●	●
	Enterprise database servers (ODBC, native)		●	●	●
	Custom, user-defined data through JavaBeans™, ADO, .NET and COM			●	●
	DATA CONNECTIVITY				
	Visual report designer for rapid data access and formatting	●	●	●	●
	Customizable templates for faster, more consistent formatting	●	●	●	●
	Repository for reusing common report objects across multiple reports		●	●	●
	WEB REPORT DELIVERY				
	Crystal Enterprise Express for rapid web report delivery (Introductory workgroup offer)		●		
Application Development	**APPLICATION INTEGRATION**				
	Report viewing APIs (Java, .NET and COM SDKs)			●	●
	Report creation APIs (Java, .NET and COM SDKs) for end user report creation and modification at runtime				●
	Custom Java Tag Library for easy customization of the end user report viewing experience			●	●
	APPLICATION DEPLOYMENT				
	Crystal Reports Java, .NET and COM reporting components[3] for embedded report viewing, printing and exporting			●	●
	Crystal Enterprise Embedded for offloading report processing from Web Server				●

For more information on how to purchase Crystal Reports products, go to:

www.businessobjects.com/products/reporting/crystalreports

Crystal Reports Training

Training is available from a number of providers; the following is a sample list.

1. Learning Tree International - www.learningtree.com
2. World-Wide Source for Crystal Training - www.crystal-reports.com
3. Stafford Technology - www.crystaltraining.com

Microsoft CRM™ Reporting Integration New for CCC Version 5

Microsoft CRM™ was introduced in January 2003 and has quickly become the premier CRM application for the Small and Medium Enterprise (SME). Avaya and Microsoft are working together to provide a complete CRM, Communications, and Networking solution for any size of business.

In Compact Contact Center Version 5, in conjunction with the introduction of the IP Office Customer Management solution, Avaya has taken this integration one step further by integrating several Microsoft CRM reports with CCC. Supervisors who operate both systems can now drive any of the 73 CCC reports from the MS-CRM interface, and there are 7 combined reports that utilize both systems data to present a 360˚ view of the contact center. The 7 MS-CRM reports are listed below:

- Microsoft CRM Sales Reports
    - Opportunity Activity & Notes
    - Contact Activity & Notes
    - Account Activity & Notes
    - Contact Center Summary by State/Province
    - Contact Center Summary by Zip Code/Postal Code
- Microsoft CRM Service Reports
    - Account Activity & Notes
    - Account Service Report

MultiMedia Module

MultiMedia Module

The MultiMedia Module, which has been created in conjunction with the Compact Contact Center version 4 software offer, is an advanced contact center solution that enables companies and departments to manage multimedia contacts into and out of the organization. IM provides applications that manage Telephony, Web Chat, E-mail and Web Call Back communications. These robust applications convert any organization into a multi-channel, enterprise-wide customer contact center that will accept multimedia calls and route them to specified members of a group (or groups).

MMM furnishes the user interface and support modules for assigning **Group Members** to **specific** communications related responsibilities. MMM routes customer interactions to the right people, generates contacts lists, monitors both the system and individual performance and hence ensures that customers are entitled to individualized attention, no matter who they may encounter in your company.

MultiMedia Module Server Side Components

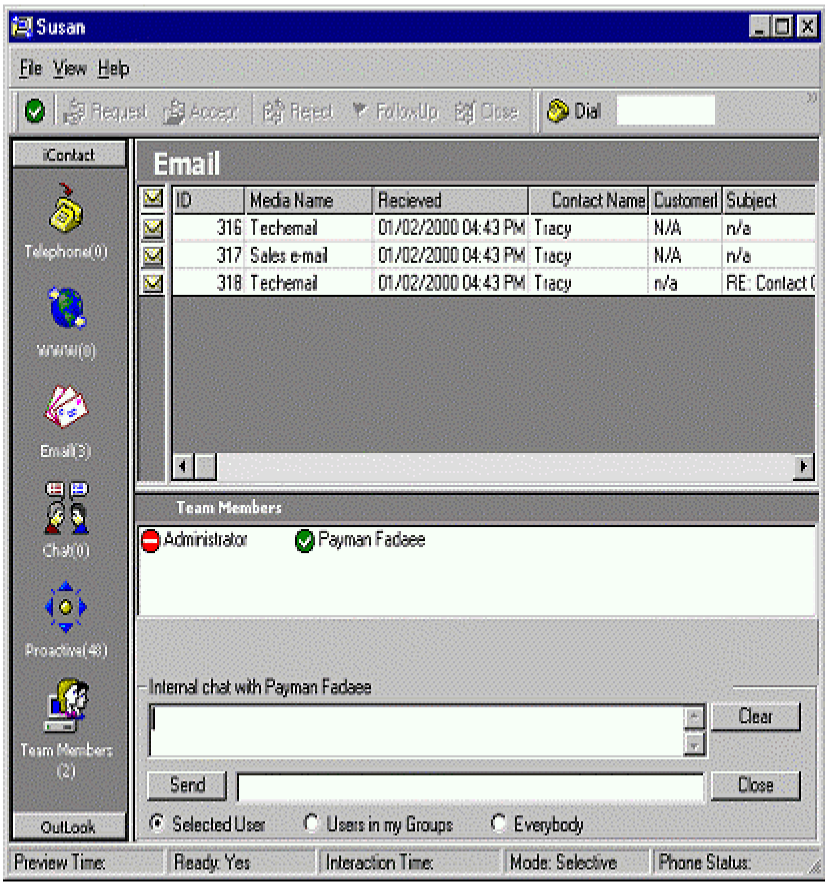
- **iServer**
  iServer consists of two parts. One is iService for Microsoft Transaction Server (MTS), and the other is a combination of different server components that run on the MTS.
- **iEmail**
  iEmail is responsible for forwarding incoming email messages to the E-mail queue or to the agent. iEmail also forwards Web Callback requests to the Web callback queue.
- **iPhone**
  iPhone is a service that applies telephony rules. Works with iServer to route incoming calls to available iContact users.
- **iChat**
  iChat is a service that forwards chat requests to the chat queues. It logs onto the chat server and creates the proper rooms based on the contents of the chat queues created by using the Resource Manager. Depending on the browsed page, the pop-up chat will log onto a defined chat room. iChat will detect the presence of the user in the chat room and log a chat request into the database.
- **Resource Manager**
  The Resource Manager administration module consists of components that enable you to add queues, define interaction results, and assign human resources to all from a single, unified console. Resource Manager has a user-friendly Microsoft Explorer look and feel interface.
- **Interaction Rules**
  With the simplicity of an Outlook Wizard look and feel,
- **Wizard**
  The Interaction Rules Wizard defines the rules for incoming contact treatment for telephone, email, chat and Web callback contacts, e.g. defines the route to the person(s) specified to answer the incoming contact.
- **Note:** The CCC Reporting module performs all MultiMedia Module reporting activities.

MultiMedia Module System Administration

- **Resource Manager**
  The Resource Manager administration module consists of components that enable you to add queues, define interaction results, and assign human resources to all from a single, unified console. Resource Manager has a user-friendly Microsoft Explorer look and feel interface.
- **Interaction Rules**
  With the simplicity of an Outlook Wizard look and feel, the Interaction Rules Wizard defines the rules for incoming contact treatment for telephone, E-mail, chat and Web callback contacts, e.g. defines the route to the person(s) specified to answer the incoming contact.
- **Proactive List Manager**
  The Proactive List Manager module facilitates the importing and assignment of outbound calling lists to Proactive Campaigns. It provides the administrator with the ability to manage outbound Proactive Campaign Lists. It furnishes the tools to create draft-calling lists, attach them to campaigns, and run the campaigns.

MMM Client Side Components

- **iContact**
  Converts the PC to an all-in-one communications and data tool and hence permits MMM users to prioritize and manage all interactions from one interface. An agent can see queued telephone calls, email messages, Web calls, and Web Chats and can communicate with group members from one centralized view.



MMM iContact Application

Wallboard Server/Client

Wallboard Manager

Two types of wallboards are available – traditional wall mounted units and PC based wallboards on the agent's PC desktop. Both types of wallboards are managed from Wallboard Manager/Wallboard Server.

Additional wallboard clients may be added and distributed across the LAN allowing additional supervisors access to create and schedule wallboard messages.

Traditional Wall Mounted Wallboards

CCC supports two physical wallboards (also known as reader boards or display boards); Spectrum (model 3214C, previously known as the 4120C) and the CCM WB/22. Both wallboards are 22 characters, tri-color, and two-line unit each. Up to 16 wallboards may be driven from the wallboard server. The Spectrum wallboard, when purchased as a Master Kit, will provide a communications module for use with the boards, which are then connected in serial. For those using the Wallboard/22, the communications card is shipped with a single cable able to drive the wallboards. Wall Mounted Wallboards are not available in all territories; please check with your Avaya representative for more information.

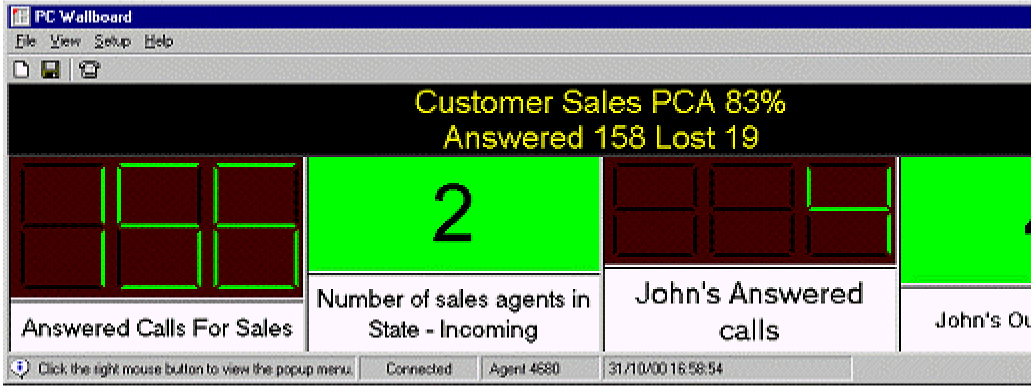
In addition to the physical Spectrum wall-mounted wallboard an IP Office license is required when being used with CCC. This IP Office license supports 4 x Spectrum wall-mounted wallboards. If more than 4 wall-mounted wallboards are required additional license keys must be purchased (each license key supports 4 wallboards at a time). A maximum of 16 wall-mounted wallboards can be supported.

Description	Short code	Material code
Wallboard/22	CCM WB/22	700040173
Wallboard Manager Communications card	CCM WB CC	700038854
Spectrum Wallboard by Avaya	5340-IW1	700289457
Spectrum Wallboard Master Kit US	Not Applicable	700289507
Spectrum Wallboard Master Kit Europe	5340-IKT	700289556
Wireless Keyboard (Remote Control)	5340-905	700289564
IP 400 CCC Wallboard 4 RFA License key required supporting 4 wallboards.	Not Applicable	176196

PC Wallboard

The PC Wallboard delivers traditional wallboard functionality to the contact center manager and contact center agent's desktop but with the additional benefit of each agent being able to configure and monitor a personalized view of the contact center via their own PC wallboard. Supervisors can provide one template for all users in order to standardize the view that agents obtain when starting PC Wallboard.

A CCC agent is able to split their PC Wallboard into twenty (20) different variables (refer to CCC System Administrator manual for details) that allow different measures of groups and agents in real-time. The data that is selected is identical to that of the physical wallboard. Examples of this are; Answered Calls, Longest Call Waiting, Agents logged in, and Lost Calls.



PC Wallboard Example

Queuing Announcements Within the Contact Center

Queuing Announcements Within the Contact Center

Voicemail Pro provides the ability to create a bespoke voicemail and interactive voice response solution which meets the specific business needs of a customer contact center.

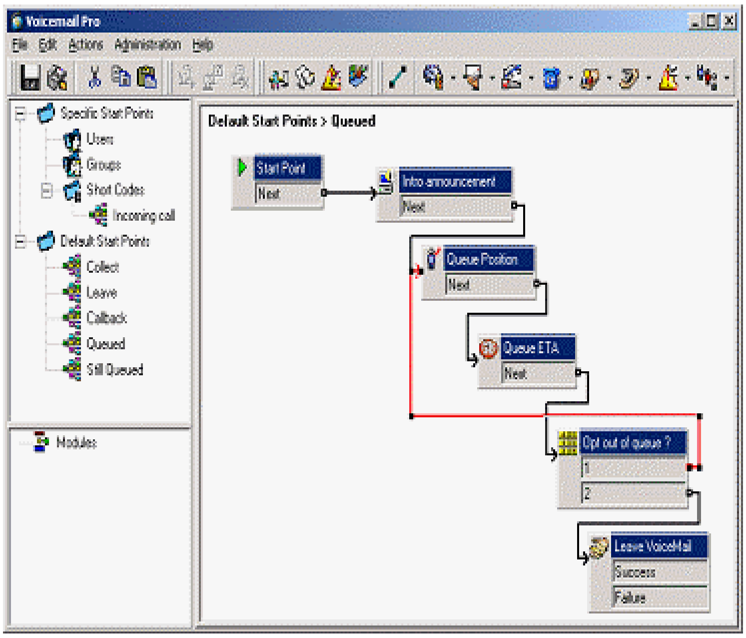
Thanks to a powerful graphical user interface, it can easily and rapidly create and modify call flows from simple announcements to multiple single digit tone menus, to provide Audiotex services and Automated Attendant functionality.

It can also be used to completely tailor the pre-connection call experience that a customer receives when making contact. In addition to the functionality provided by Voicemail Pro's call in-queue announcements, supervisors may create sophisticated queue and call routing plans with access to a host of features such as message taking, interview services, and the ability to play estimated time to answer or queue position information to customers.

Queue Announcements

The Voicemail Pro application provides Queue Handling facilities, allowing incoming Hunt Group calls to be automatically answered when department, group or individual telephones are busy. Customers entering a queue are played a message informing them of the situation and then hear hold music (internally generated or from an external source), while being regularly updated. Two unique messages may be recorded for each Hunt Group (queue entry and queue update message). Queue announcements can also provide position in queue and estimated time to answer to the caller.

Additionally, to suit the needs of the contact center, Voicemail Pro provides the caller with their position within the queue and an estimated time for their call to be answered. It furthermore gives the caller the option to opt out of the queue and leave a message at any time if desired.



Contact Center Queuing using Voicemail Pro

Auto-Attendant Operation (Advanced Call Flow)

In addition to its advanced voicemail facilities, Voicemail Pro provides an easy-to-use, multi-level configuration tool (Voicemail Pro Manager) which allows network managers and system administrators to construct an interactive system, based upon DTMF telephone key entry.

At its most basic, this allows an Auto-Attendant system to be built while more complex scenarios can be configured using telephony actions such as CLI/ANI routing or assisted call transfer. The call flow can be based on conditions such as week/time of day or user-defined variables. You can even set Voicemail Pro to send by email a voice recording previously collected or activate a door-entry relay.

- **Note:** Queue announcements are only available when Voicemail Pro is provisioned with 4 or more ports.

Campaign Manager

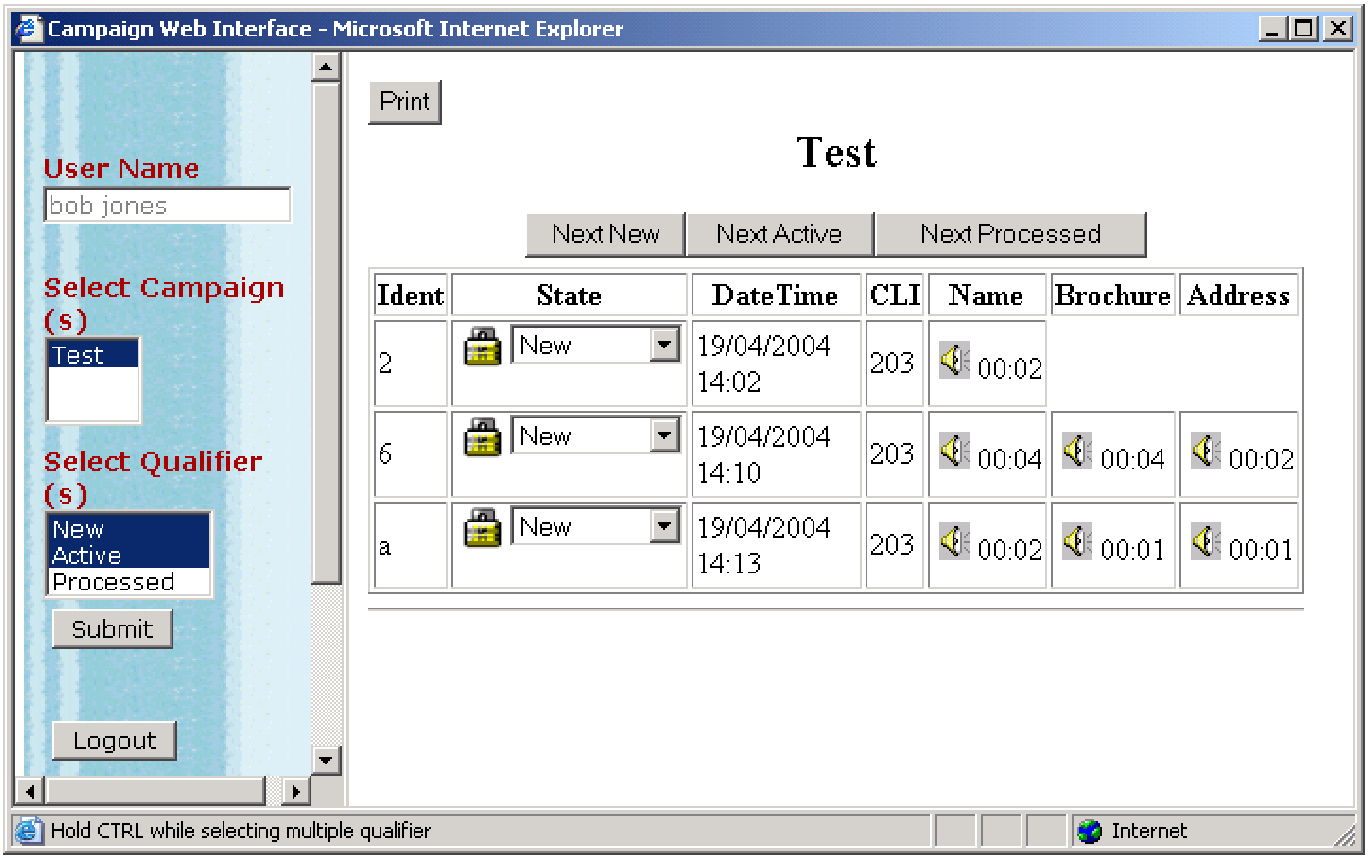
An Integral part of Voicemail Pro, this application enables repetitive information gathering (such as brochure requests) to be fully automated, leaving agents free to deal with other, more complex calls which require human interaction. Campaign Manager enables round-the-clock immediate answering, offering callers a series of clear and uncomplicated questions. Callers give their response either spoken or through the telephone keypad. Recordings can be interrupted by the caller at anytime by pressing a key on their phone. When agents are busy, overflow to Campaign Manager relieves congestion and pressure on agent groups.

An agent can collect the completed transaction via a web browser (see figure 9.6 below) or via a short code representing the park slot number of a particular campaign. This number can be pre-programmed under a DSS key and used by agents to access the campaign. If the DSS key incorporates a BLF lamp, that lamp is lit when new campaign messages have been left. Agents then transcribe the caller's answers into a database or other records.

Once a caller has entered information via a campaign message, the agents can then check these messages, with any responses left via touch-tone played back as voice.

The powerful Windows Graphical User Interface (GUI) of Voicemail Pro Manager makes customization and creation of questionnaires simple. Inbound call campaigns can be easily created and modified via a campaign wizard.

With Voicemail Pro, customers are able to "break out" of a queue, or be directed in an "Overflow" situation to complete their transactions via the Campaign Manager. This ensures that a minimum of customers give up when forced to wait in a queue and therefore maximizes revenue opportunities.



Campaign web client

Recording Services

Voicemail Pro also offers the ability to record calls for a variety of applications, such as for training purposes or to monitor abusive callers. Recording can be initiated manually by agents or automatically. Recorded conversations can be later recalled in the same manner as voicemail messages.

Automatic recordings can be set via the IP Office Manager for a selection/all calls based upon a variety of criteria such as dialed number, caller's CLI/ANI, agent extension number, inbound/outbound, account code, time of day, etc. For storage considerations, 1 minute of recording time is approximately 1MB of data on the target drive.

Manual recordings can be invoked by Phone Manager, the 'record' soft key on the display telephones or by placing the call on hold and invoking a short code.

- **Note:** Recording services is not suitable for applications where recording is a legal requirement.

IP Office Manager

In order to control a customer facing department or call center, the supervisor may need to change set-up parameters such as agent names, campaign names, call routing patterns or group memberships. The Manager application allows them to carry out these changes quickly and simply from any client PC connected to the Local Area Network (LAN). The application can operate on the same PC as the Call Center View, Wallboard and Report Clients, giving the supervisor all of the tools they need to manage the customer facing department or call center from one desktop. See Section 10 Common Management Utilities for further details.

Workforce Management Interface

A generic API is available to facilitate the integration of several workforce management packages, including those from QMAX Systems Limited and Blue Pumpkin Software, to the IP Office CCC server. Avaya only provides the interface license, the management software is priced, supplied, installed, configured and maintained by the supplier directly.

Compact Business/Contact Center Modules Summary

Feature	CBC	CCC
Real time screens	1	18
Real time graphs	4	By Group/Agent
Variables	3 of 13	N/A
Reporting period	24 hours	24 hours
Historical data	31 days	Hard Disk Dependant
Pre-defined reports	None	73
Call Center View	Not available	Included
Report Manager	Not available	Included
Wallboard Manager	Not available	Included
Networked Administrator	Not available	Included
Remote Management	Not available	Via RAS
System (Note: Both systems require Delta Server, see HW requirements).	Windows 2000 Windows XP	Windows 2000 Windows XP
PC Wallboard	Not available	Optional
Report Designer	Not available	Optional
WFM Interface	Not available	Optional
Agents	Not Applicable	75
Supervisor	Not Applicable	21

CCC/CBC Technical Specification

All CCC & CBC applications are based on industry standards and exploit the resilient Windows 2000/2003/XP operating systems and Microsoft's MSDE and SQL technology. Openness and data export are achieved through standard SQL tools and ODBC drivers, as well as a very powerful Report Designer module. This sections sets out the minimum recommended requirements for both the server and client platforms:

Compact Business Center (CBC)

Server PC (Delta Server):

- **Operating Systems Supported:**
    - Windows 2000 Server SP4 and later.
    - Windows XP Professional.
    - Windows 2000 Professional SP4 and later.
- **PC Specification:**
    - Pentium 4, 2.8GHz or equivalent.
    - 10GByte hard disk.
    - Minimum 256 MB of RAM.

Client PC:

- **Operating Systems Supported:**
    - Windows 2000 Server SP4 and later.
    - Windows XP Professional.
    - Windows 2000 Professional SP4 and later.
    - Windows NT Workstation SP6.
    - Windows 98.
- **PC Specification:**
    - Pentium III 800MHz or equivalent.
    - Minimum 128 MB of RAM.

Customer Contact Center (CCC)

Server PC (Delta Server):

Supporting a maximum of 5 Supervisor Positions:

- **Operating Systems Supported:**
    - Windows 2000 Server SP4 and later.
    - Windows 2003 Server.
- **PC Specification:**
    - Pentium 4, 2.8GHz or equivalent.
    - Minimum 10GByte hard disk.
    - Minimum 512 MB of RAM.

- **Voicemail Pro and CCC:**
  From IP Office 3.0 onwards this is a tested and supported option. It is subject to the PC and OS requirements of both applications, whichever is the higher plus the following:
    - Maximum CCC Agents: 20.
    - Maximum Voicemail Ports: 8.
    - Windows 2000/2003 Server OS's only.

Client PC:

Supervisor running CCV, Wallboard Manager, Report Manager

- **Operating Systems Supported:**
    - Windows XP Professional.
    - Windows 2000 Professional SP3 or higher.
    - Windows NT Workstation SP6.
    - Windows 98.
- **PC Specification:**
    - Pentium III 800MHz or equivalent.
    - 1 GB hard disk.
    - Minimum 128 MB of RAM.

Computer Telephony Integration

Computer Telephony Integration

Computer Telephony Integration (CTI) is about bridging the gap between the telephone system and business applications. On IP Office, this is achieved by use of the IP Office CTI Link, a CTI middleware product and Software Developers Kit.

On IP Office, CTI is delivered through adherence to open standards. This gives customers access to a wide range of third-party solutions, addressing vertical markets, and designed to meet their requirements. For developers, migrating their offering from other platforms to IP Office is quick and easy, and the advanced CTI features IP Office offers makes it easy to demonstrate full integration, and more business benefits.

IP Office provides two levels of CTI interoperability: CTI Link Lite, which is free of charge, provides all the functionality required to support the vast majority of applications, including screen-popping, and many third-party products.

CTI Link Pro provides enhanced functionality, including the ability to control more than one telephony device, and also provides advanced call center operation.

Because the network is integrated into the fabric of the IP Office system, all CTI is done through the LAN. On many other systems, CTI is delivered by a physical connection between each handset and computer (first party CTI). This introduces additional points of failure, as well as relying on non-standard interfaces and handsets. On IP Office, all devices can be used with CTI.

The Benefits of CTI

CTI delivers real business benefits in the following key areas: Reducing costs, increasing productivity and delivering better customer service.

Often helpdesks or contact centers are overloaded with phone calls which results in customers having to wait for an available agent and then answer a long list of trivial questions before the real purpose of the call is addressed. Sometimes callers are transferred to many different departments before reaching someone able to assist them. This type of service results not only in errors and inconsistencies in data entry and information relayed to a caller, but also to unhappy customers and lost time and profits.

Reducing Costs

Half the cost of running a call center, service center or helpdesk is tied up in labor, 40 per cent in communications charges and less than 10 per cent in equipment. Saving seconds on each call can make a big difference enabling agents to be more efficient, deliver a better service and dramatically reduce company overheads.

In a helpdesk or call center with a high volume of phone calls each day, it takes many agents to handle these calls efficiently. Callers may have to wait for an available agent, which increases costs to the customer and can be a potential loss of business, due to abandoned calls and unhappy customers.

With CTI, costs can be reduced through the following:

- Shortening the average length and duration of calls thereby maximizing the number of talk minutes per hour and reducing the required number of staff.
- Reducing/reducing telephone line requirements.
- By using CLI/ANI, automating the call-back of inbound abandoned calls, the warm leads, and outbound calls that were unanswered or received a busy signal.
- Professionalism improves the company image thereby increasing the volume of customer calls.

Increasing Productivity

By implementing CTI, organizations can reduce the average duration of each call, ensuring that a higher percentage of call time is spent productively. This extra time can be used to handle a larger call volume, without increasing staffing levels.

Delivering Better Customer Service

With CTI, customer service can be improved in the following ways:

- Offering a faster, more personalized service based on CLI/ANI, DDI/DID and voice processing input by minimizing time spent on the 'discovery' phase of the call.
- Providing a higher degree of accuracy of data entry.
- Retaining customer information on transfer (avoiding the need to request or repeat information when transferred to another agent).

Target Customers & Markets

Applications for CTI are quite broad, however eight major types of organization can be identified as the key targets for sales of CTI solutions.

- **Telemarketing Centers**
  These are call centers with many dedicated agents processing high volumes of calls each hour, both inbound and outbound.
- **Sales Departments**
  These are organizations with sales professionals such as computer software or insurance agencies. Agents have a finite list of customers with whom they work to create and maintain one-on-one relationships.
- **Service Centers/Helpdesks**
  Almost all companies have some type of customer service or helpdesk department. Some support external customers, others have an internal hotline or helpdesk arrangement. All are looking to improve their response time and overall quality of service.
- **Collection Agencies/Debt Recovery**
  These organizations consist of agents who spend many hours on the telephone. Here, saving even seconds from each call can increase both productivity and profits. Additionally, the application can provide comprehensive management reports.
- **Knowledge Workers**
  These people, like PC power users, rely heavily on their PC for information access and processing capabilities.
- **General Office Workers**
  This includes receptionists and secretaries who have phone monitoring and messaging responsibilities.

Computer Telephony Integration with IP Office 2.0

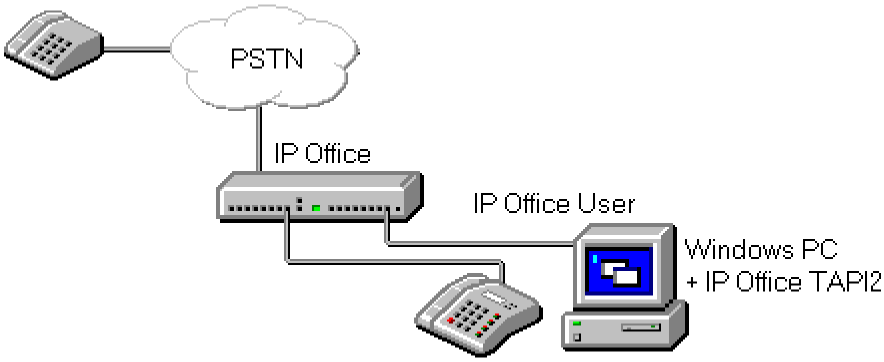
IP Office offers a significant CTI capability. Several interfaces are supported:

- **TAPILink Lite.**
- **TAPILink Pro.**
- **TAPI-WAV driver.**
- **DevLink Pro.**
- **IP Office SMDR.**
- **IP Office Software Development Kit.**
- **Microsoft™ CRM Integration Phase 1 (Screen Pop).**

- **TAPILink Lite**
  Provides first-party CTI support (defined below) for Microsoft TAPI 2.1 and TAPI 3.0, so each PC can control or monitor one handset device. The software components are shipped with the IP Office system on the User CD-Rom, and do not required a license key for use, and therefore no charge is made.
- **TAPILink Pro**
  Provides third-party CTI support (defined below) for TAPI 2.1 and 3.0. These components are identical to their Lite equivalent; the presence of the CTI Link Pro RFA license key (which can be purchased in the usual way for products) enables this additional functionality.
- **TAPI-WAV driver**
  Provides software-based support for voice processing. Purchasing the CTI Link Pro RFA license key also enables 4 ports of voice processing; additional ports can be purchased in 4 port increments. The TAPI-WAV driver is for use with TAPI 2.1 only; for TAPI 3.0, IP Office supports the Media Service Provider (MSP) interface, defined by Microsoft in TAPI 3.0.
- **Dev*Link* Pro**
  Provides a real-time event stream in addition to the SMDR interface provided in IP Office SMDR (see below). The real-time event stream takes the form of a call record, which is issued whenever the state of any endpoint of a call changes (typically there are two endpoints on a call, but for some circumstances, such as conference calls, intruded calls there may be more).
- **IP Office SMDR**
  Provides an interface to obtain SMDR events. A comma-separated record is issued for each call, when the call is completed. This interface is designed for call accounting and call billing applications, and replaces the previous DevLink Lite interface. IP Office SMDR is available free of charge, and distributed on the IP Office Admin CD-ROM.
- **Software Development Kit**
  Consists of a single CD-Rom, containing the developer documentation for TAPILink Lite, TAPILink Pro, DevLink Lite and DevLink pro, as well as pre-compiled programs for exploring TAPI 2.1 and 3.0. In addition, example source code is included, making it easy for developers to quickly become productive on these advanced CTI interfaces.

TAPILink Lite (1st Party TAPI Support)

TAPI*Link* Lite provides simple first-party CTI via Microsoft TAPI 2.1 and 3.0. Individual desktop PCs connected to the Local Area Network communicate with IP Office via an IP connection over the LAN. Each PC is capable of controlling one telephone device (see diagram below).



Microsoft TAPI 2.1 and 3.0 are specifications and developers interfaces for controlling and monitoring a telephony device. The specification requires that a certain amount of core functionality is implemented, and additionally defines a series of optional functionality that switch vendors may also implement.

TAPILink Pro (3rd Party TAPI Support)

TAPI*Link* Pro provides all of the features and functionality of TAPI*Link* Lite, but additionally provides third party CTI operation. This means that a single server can control and monitor any number of telephone devices.

In addition, TAPI*Link* Pro provides the ability to monitor and control groups. This allows an application to be notified when a call enters a queue, and can also redirect it to another location.

TAPILink Pro also supports additional TAPI functionality that is not available through TAPILink Lite. This functionality is supported through the LineGetLineDevStatus and LineDevSpecific calls. The additional features are:

- Agent login.
- Agent logout.
- Set and retrieve divert destination.
- Set and retrieve extended divert status (Forward All Calls, Forward on Busy, Forward on No Answer, Do not Disturb).
- Retrieving the extension locale (language).
- Set and clear the message waiting lamp.
- Enable and disable group membership.
- Generate and detect DTMF digits and tones (requires the TAPI-WAV driver).

Support for Developers

All IP Office CTI products can be sold via the normal channel. As with any other element of the IP Office product range, support for end-customers is via the reseller and distributor. Avaya does not provide support services directly to end-customers.

However, in recognition of the fact that not all resellers will have the ability to support a sophisticated CTI developer, Avaya also operate a third-party developer partner program, called the Developer Connection Program.

The Developer Connection Program ("DevConnect") is the Avaya developer partner programme, and is designed for third-party companies who are creating a product for sale, and who wish to receive technical support. Membership of the program is at the sole discretion of Avaya.

DeveloperConnect members pay an annual fee, for which they receive technical support directly from Avaya. In addition, Avaya will perform interoperability testing between IP Office and the member's product, and may also create opportunities for joint marketing, including exhibitions, use of Avaya's logo, and other benefits.

More information on the DeveloperConnect program can be found at: www.devconnectprogram.com.

Microsoft™ CRM Integration

Avaya and Microsoft Business Solutions have signed an agreement to create a packaged applications and hardware solution for all small and medium businesses with multiple customer touch points. This alliance will position IP Office as the Convergence platform of choice for customers of Microsoft CRM™.

IP Office support for Microsoft CRM has been divided into three phases, providing telephony integration (screen pop) is phase one while phases two and three deal with an integrated reporting module. This strategic alliance is aimed at small and medium businesses who need an interconnected workplace extending across business systems, communications infrastructure and web services. They also need a turn-key system, simple implementation, and an affordable price point.

For IP Office systems running software release 1.4 and above, the first phase of this program has been introduced. This includes the development of a TAPI based CTI application integrated with Microsoft™ CRM.

14. Common Management Utilities

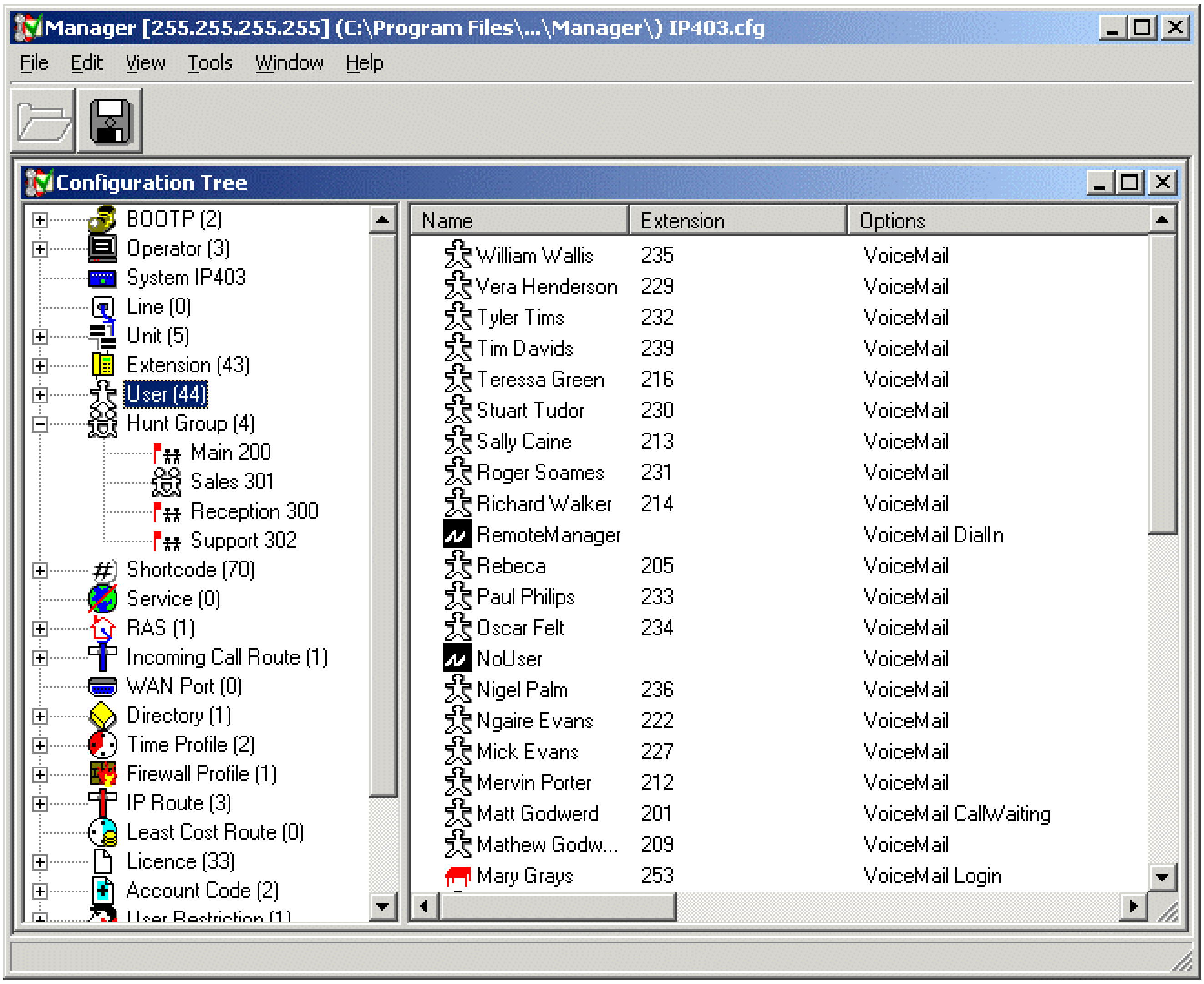
Introduction to IP Office Management Utilities

This section gives an overview of the management applications that are common to all IP Office platforms.

- **IP Office Manager**
  IP Office's main configuration tool.
- **Wizard**
  An installation and administration wizard.
- **Call Status**
  Displays current call activity.
- **IP Office SMDR**
  Outputs call detail records for off switch processing.
- **Monitor**
  A trace utility for trouble shooting.
- **SNMP**
  Alerts and alarms from IP Office systems to SNMP tools.

IP Office Manager

This application is IP Office's main configuration tool. Utilizing a Windows Graphical User Interface, Manager provides a familiar, intuitive interface for both installation configuration and subsequent moves and changes. As with all IP Office applications the Manager is multi-lingual. This, coupled with the ability to use the application both locally and remotely, it is possible for a customer with a global presence to manage any of their IP Offices from any country using their local language preference. Access to the Manager is protected by passwords and definable user rights. This provides a secure yet customizable application that allows it to operate according to the individual users level of expertise.



The IP Office Manager operates on a copy of the configuration held either locally or on a network drive. Configurations are prepared and reviewed 'off line' before committing to the IP Office. This has the benefit of ensuring a backup copy of the system configuration is always available for disaster recovery.

A number of other convenient features are provided by the manager including:

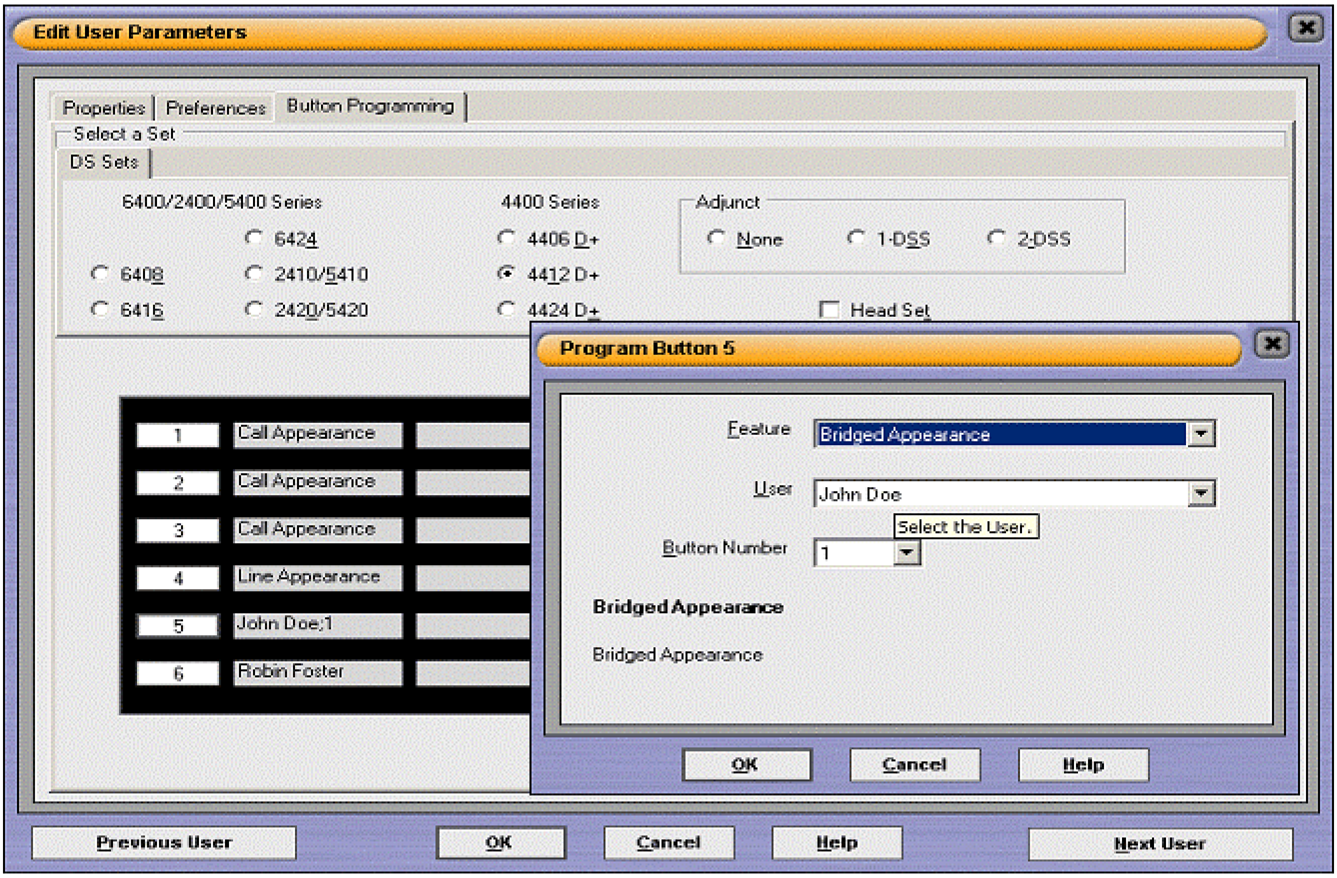
- Easy upgrades to the IP Office system software using the upgrade wizard.
    - Systems running 2.1 or later have the added benefit of being able to send software to a system and have it validated, before choosing to run it or stay with the old software. This facility is available locally via the LAN, or remotely over a VPN and Dial-up connections.
- Copying information such as the shortcode list from one IP Office to another.
- Importing and Exporting Directory information in CSV format to and from applications like Microsoft Excel and Word.

Installation and Administration Wizard

The IP Office Wizard is a Windows based application that has been designed to simplify the installation of IP Office in the most common configurations. The Wizard will reduce the time spent installing an IP Office and remove configuration errors. The application operates in two modes, 'Online' for configuring a live system or 'Offline' for designing or modifying a configuration of a system that is not physically present. When a configuration is created in the off line mode, the installation engineer simply loads the configuration file to the appropriate system when available.

The wizard incorporates error checking of the configuration. It detects a wide range of configuration conflicts through the error checking of individual entries and validity checks the configuration before saving to a system or as a configuration file on a PC hard drive. The wizard will automatically suggest corrections to any errors found. All IP Office configurations from version 1.4 can utilize the error checking capabilities of the Wizard to review existing configurations that may contain errors even if the configuration was not created using the Wizard.

The wizard also provides the ability to design user templates, complete with button programming, that can be subsequently applied across a range of extensions. Each new template can be saved to build up a library of templates for future configurations.

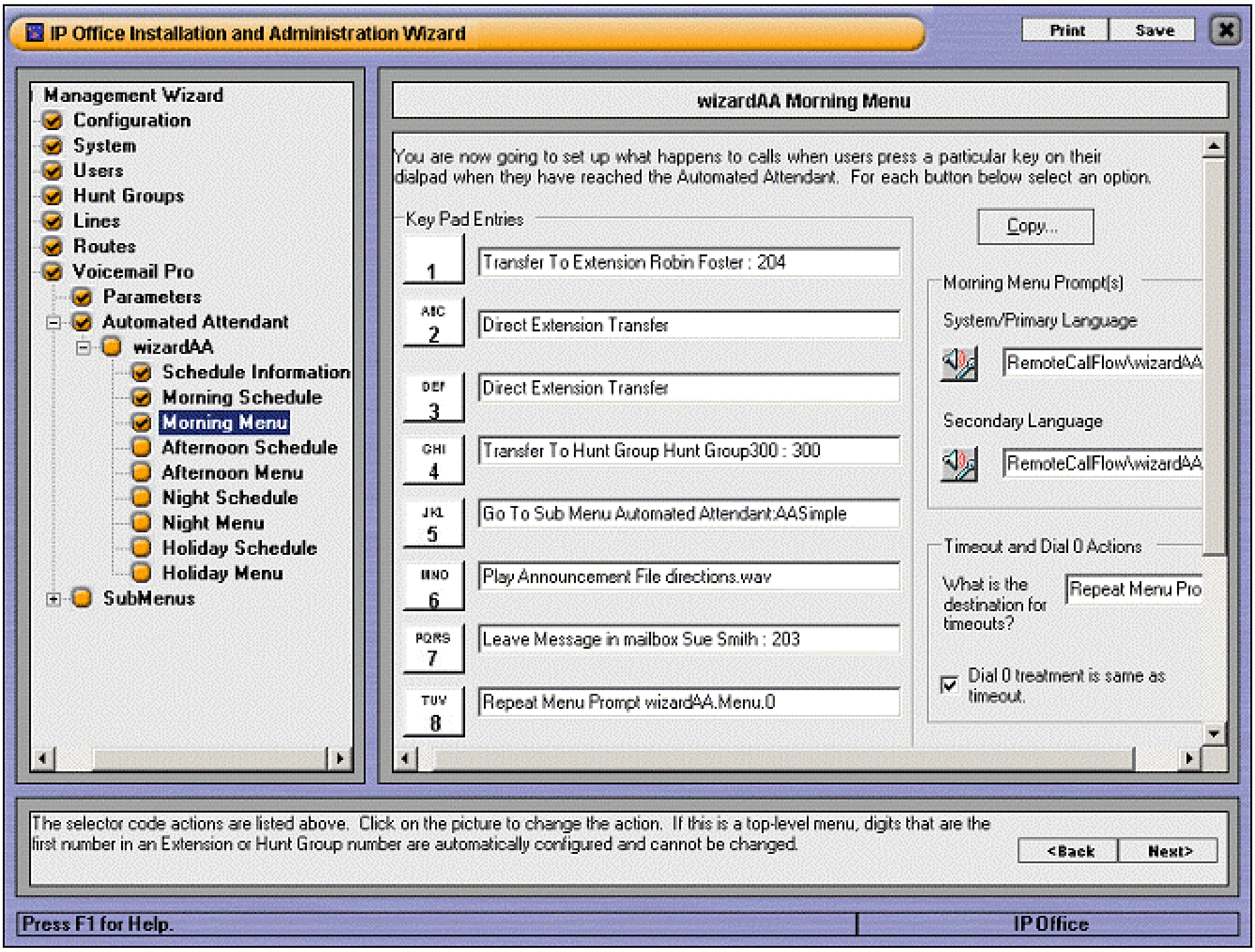


The IP Office Wizards provide simple tools to enable the quick and accurate setup of IP Office in both Key and PBX modes. The graphical interface allows the installer to easily assign functions, lines and coverage and bridge appearances to individual buttons. The standard user templates, including button programming, can be created and saved within the wizard for use on future IP Office installations.

The IP Office Wizard supports the configuration of the Embedded and Voicemail Pro voice mail platforms either offline or online. In offline mode, where the voice mail is not present, the configuration is saved to the PC and sent to the appropriate voicemail platform when it is available.

The IP Office Wizard Voicemail Pro configuration supports the setup of four two-tier auto attendants and actions including transfer, play directory list, leave message and repeat menu options and announcement. The recording of bilingual prompts and messages can be completed via the wizard through a set connected to the system or via the PC. The Wizard allows Day, Afternoon, Night and Holiday schedules to be defined and assigned on a system wide or per individual auto attendant basis.

The ability to print the IP Office configuration or part of the configuration is included as part of the IP Office Wizards.



The Wizard has three versions, IP Office - Small Office Edition Wizard, IP Office Installation and Maintenance Wizard and Moves Adds and Changes Wizard.

The IP Office – Small Office Edition and Installation and Administration Wizards are password protected to prevent unauthorized access to more sophisticated configuration items.

- **The IP Office - Small Office Edition Wizard**
  The IP Office - Small Office Edition Wizard is used with the range of IP Office Small Office Editions. The wizard simplifies the installation process for the Small Office Edition and the elements specific to the Small Office Edition such as the Integrated Wireless Access Point. The Small Office Edition Wizard walks an installer through an installation by asking a series of questions allowing the wizard to build the appropriate configuration in the background. The Small Office Edition Wizard is for use on single site/stand alone systems.
- **The IP Office Installation and Administration Wizard**
  The IP Office Installation and Administration Wizard is used with the entire range of IP Office systems. The wizard systematically guides the installer through the installation of the system and checks the configuration for any errors whist configuring the system. The Wizard provides integration with Voice Mail Pro for simplified voice mail access and the creation of start points and two tier Voice Mail Auto Attendants.

  The IP Office Installation and Administration Wizard simplifies the setup of remote / home workers. The IP Office system administration via the Wizard supports the configuration of the connection for the remote / home workers. The remote / home workers can be setup to use the IP Office Virtual Private Network (VPN) software or a non VPN high speed internet connection. Please note that the additional administration and configuration of the remotely connected device is required and the Quality of Service over Internet connections varies which could impact voice and connection quality.
- **The Moves Adds and Changes Wizard**
  The Moves Adds and Changes Wizard for Users and Hunt Groups is used by Business Partners and suitably trained end users to modify a subset of the system functions. This version of the wizard is not password protected but prevents unintentional system changes while providing a simple and intuitive interface for the most commonly accessed system management functions – users, and hunt groups and System Speed Dials.

Importing System Settings

The wizards allow items that have been created in Word or Excel (CSV file) to be imported into the configuration. The end user can complete customer specific information and the files can be uploaded into the IP Office quickly and check for errors. Other files such as license keys can also be loaded directly into the IP Office. These options reduce the need for post installation modifications, such as spelling mistakes in names, and reduce the time required to configure the IP Office. The following files can be created and loaded directly into IP Office Wizards:

- License Key Files.
- Users Name, Extension number and Group Membership.
- Hunt Groups Names, Number and Hunting Type.
- System Speed Dial.
- Feature code templates - to allow mimicry of other systems.
- Account Codes.
- Dial plan (for those cases where there are non-continuous ranges for users and Hunt Groups).
- Voicemail Pro Schedules for Day, Afternoon and/or Night.
- Voicemail Pro Holiday Schedules.

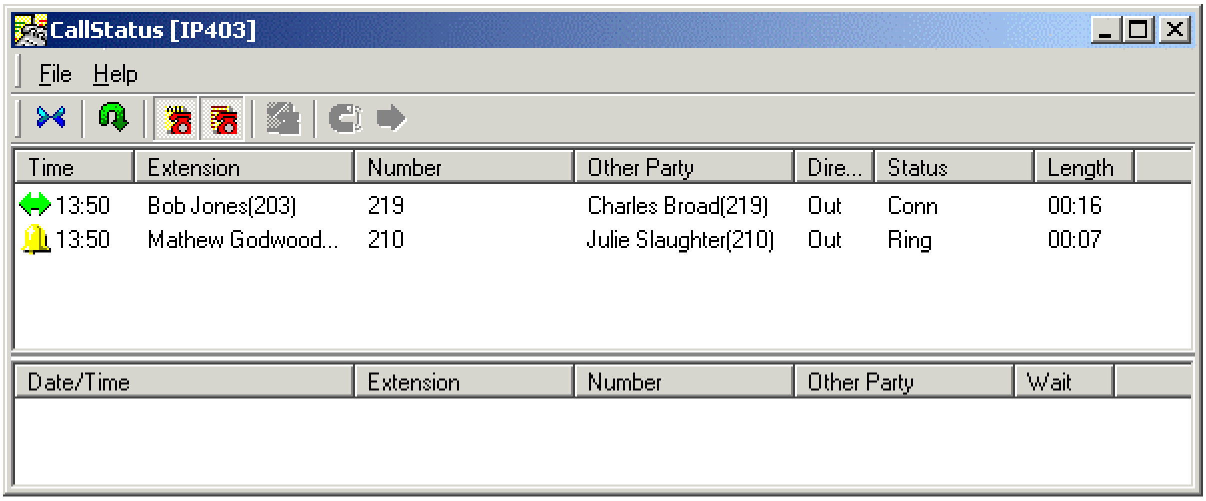
CSV File Format

The file formats are all comma separate with no text delimiters and no header.

- **System Speed Dial CSV File**
  Column 1: Speed Dial Name, Column 2: Speed Dial Number, Column 3: Telephone Number.
- **Short Codes CSV File**
  Column 1: Code, Column 2: Telephone Number, Column 3: Feature Name.
- **Users CSV File**
  Column 1: Full Name, Column 2: Extension Number, Column 3: Template Name, Columns 4 to 9: Hunt Group Extension Number.
- **Hunt Groups CSV File**
  Column 1: Hunt Group Name, Column 2: Extension Number, Column 3: Group Type.
- **Account Codes CSV File**
  Column 1: Account Code, Column 2: Caller ID.
- **Dial Plan CSV File**
  Column 1: Extension Number, Column 2: Type (Physical, Virtual, Hunt Group).
- **License Key CSV File** *(provided by Avaya as a CSV file)*
  Column 1: License Option, Column 2: License Key.
- **Voicemail Pro Day, Afternoon, Night Schedule CSV File**
  Column 1:Day, Column 2: Start Time, Column 3: End Time
- **Voicemail Pro Holiday Schedule CSV File**
  Column 1: Month (3 Characters only), Column 2: Date (2 digits)

Call Status

The Call Status utility allows a PC to display all telephone and data calls that are currently active on the IP Office system.



Call Status displays two window panes, one on top of the other, in one window. The top window pane is a status of all of the active telephone and data calls on the system, the lower pane will collect a list of incoming calls that have not been answered.

The Active Calls List displays the time the call was made, the extension either making or receiving the call, the number dialed or the received CLI/ANI, the party at the other end of the call, the direction of the call, the current status of the call (Idle, Ringing, Connected, Disconnected, Suspended, Resuming, Dialing, Queued, Parked, or Held) and the length of time the call has been active.

The Missed Calls List displays the date and time the call was received, the extension that was receiving the given call, the number received via CLI/ANI, the party at the other end of the call and the length of time the third party waited for an answer before hanging up.

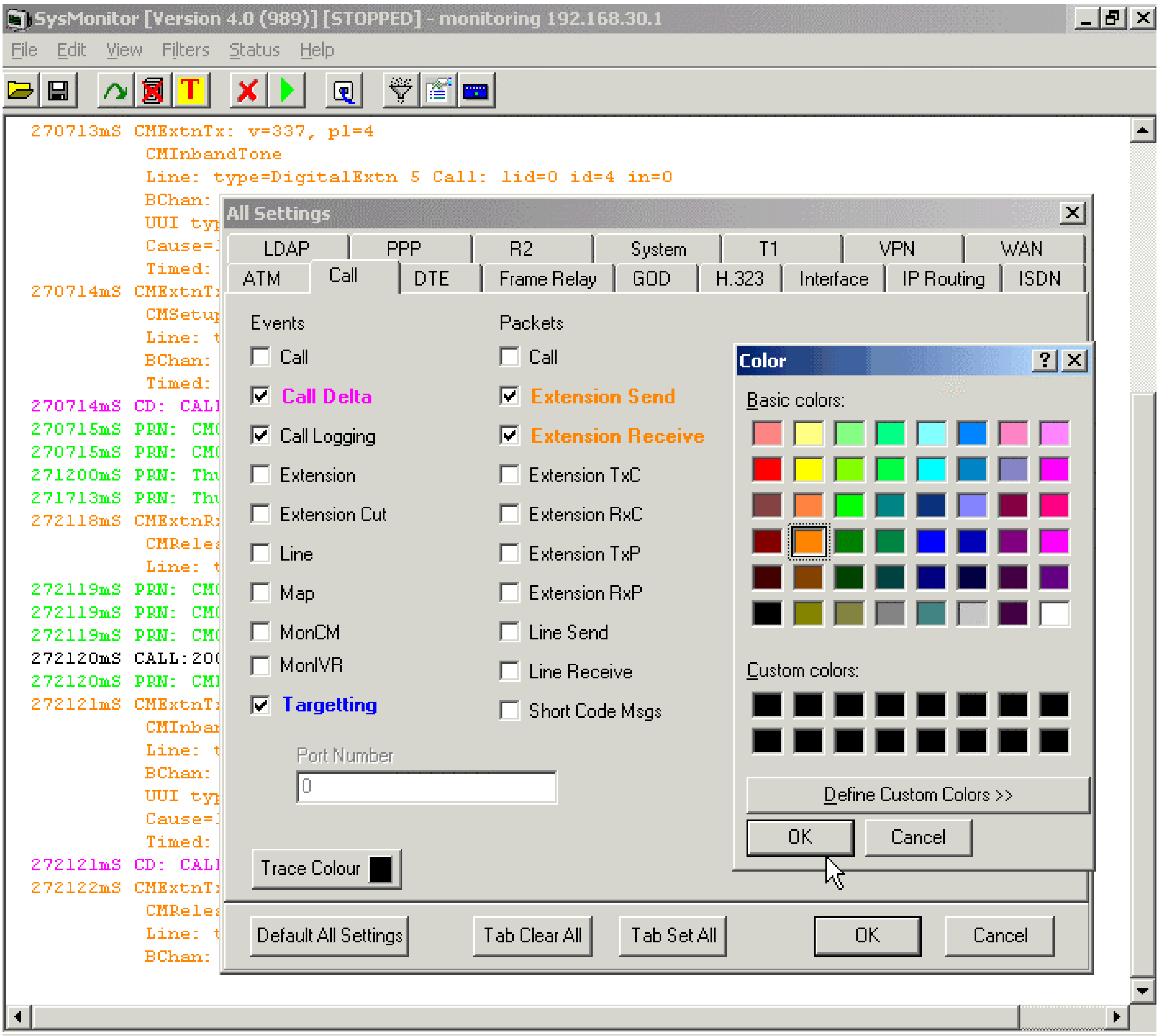
Call Status is extremely useful in establishing if there are any data calls in progress.

Monitor

Monitor is a real-time maintenance utility to assist with IP Office trouble-shooting. As the application connects to the IP Office over an IP Connection it can be used from both local (LAN) and remote locations (WAN).

A simple interface allows an engineer to select which protocols and interfaces are to be monitored and decoded. The trace can either be captured directly to screen or as a log file for later analysis.

Different protocols can be color coded to improve the clarity of large log files. In addition to monitoring, the application captures system alarms and will display a death log of the last 20 alarms that have occurred.



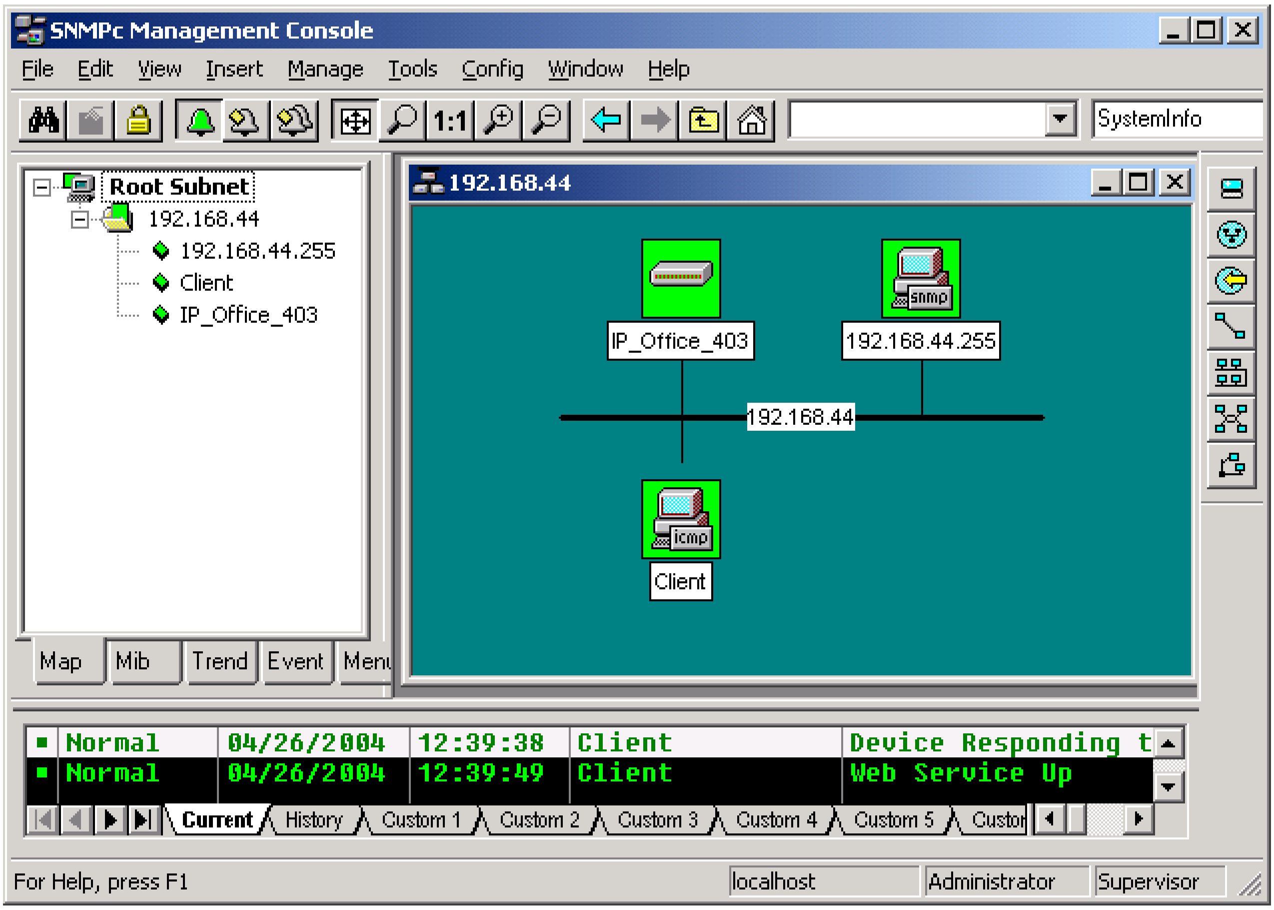
Simple Network Management Protocol (SNMP)

SNMP is a industry standard designed to allow the management of data equipment from different vendors using a single application known as a Network Manager. The Network Manager will periodically poll equipment to solicit a response, if no response is received an alarm is raised.

In addition to responding to polls, IP Office also monitors the state of its Extensions, Trunk cards, Expansion Modules (except WAN3 module) and Media cards, if an error is detected IP Office will notify the Network Manager.

The IP Office implementation allows for two separate Network Managers to be nominated. This permits both a customers Network Manager and a Maintainers Network Manager to be notified of the same alarm condition. IP Office has been tested against CastleRock's SNMPc-EE™ and HP's Network Node Manager (part of the OpenView application suite). Avaya's 'Integrated Management Suite' also uses HP's Network Node Manager.

IP Office 3.0 adds additional alarms for Voicemail Pro and Embedded Voicemail. For Embedded Voicemail the initial alarm is triggered when the memory card is 90% full, a critical alarm is given at 98% (99% on the IP406 V2) and a rest alarm given when the space returns to below 90% full. For Voicemail Pro these alarm levels are configurable through the Voicemail Pro client. The alarms are generated and sent by the IP Office control unit configured for SNMP.



IP Office SMDR

A call logger utility, IP Office SMDR, is included which allows the detail of all calls to be sent to a file on the PC.

Third party applications can then use this data to allocate costs to departments, analyze trunk capacity, report usage against account codes etc. The IP Office SMDR utility does not provide any reports or graphical analysis of telephone usage. For multi-site IP Office configurations, one IP Office SMDR application per site is required.

The following operating systems are supported: Windows 98 SE, Windows 2000, Windows NT 4 and Windows XP.

SMDR Diagnostics

Time Of Call Arrival	Call Duration	Ring Time	CLI	Dir.	DDI	DDI	Account Code	Internal	Call ID	More	P1 ID	P1 Name	P2 ID	P2 Name	Hold Time	Park Time
2004/10/19 07:47:07	00:00:00	0	211	O	215	215		1	6	0	E215	Extn215	E215	Extn215	0	0
2004/10/19 07:47:07	00:00:00	0		O				1	100	0	E-1	No Name			0	0
2004/10/19 07:46:56	00:00:10	0	215	I	215	215		0	6	0	V9551	CO Channel 1	E215	Extn215	0	0
2004/10/19 07:46:54	00:00:09	1	211	I	369	369		0	7	0	V9551	CO Channel 1	E211	Extn211	0	0
2004/10/19 07:46:56	00:00:07	0	211	I	9551	9551		0	7	0	V9551	CO Channel 1	E369	Extn369	0	0

A: Configurations

Configurations and Factory Build Options

This section provides information on the factory builds available for each of the base modules along with some example configurations.

Not all options are available at launch, please contact your Avaya representative for local, up to date, information.

Factory Configurations

Small Office Control Units

All Small Office Edition control units include twin PCMCIA slot for embedded voicemail and wireless access point options, four port Ethernet switch, single Ethernet WAN port and a slot for optional V24/V35/X21 or T1 WAN option modules.

- **Avaya IP Office Small Office Edition - 4T+4A+ 8DS (3 VC) US *(700350424)***
  Providing four US specification analog trunks, four analog extensions and eight Digital Station ports. Comes with three voice compression resources as standard for VoIP applications.
- **Avaya IP Office Small Office Edition - 4T+4A+8DS (3 VC) INT *(700280209)***
  Providing four analog trunks (not US), four analog extensions and eight Digital Stations. Comes with three voice compression resources as standard for VoIP applications.
- **Avaya IP Office Small Office Edition - 4T+4A+8DS (16 VC) US *(700350432)***
  Providing four US specification analog trunks, four analog extensions and eight Digital Station ports. Comes with sixteen voice compression resources as standard for VoIP applications.
- **Avaya IP Office Small Office Edition - 4T+4A+8DS (16 VC) INT *(700280217)***
  Providing four analog trunks (not US), four analog extensions and eight Digital Stations. Comes with sixteen voice compression resources as standard for VoIP applications.

Avaya IP Office - Small Office Edition Expansion Cards

- **Avaya IP Office Small Office Edition - WAN Expansion Kit *(700289713)***
  Optional card for connection to private circuits and network terminating devices with V.24, V.35 and X.21 interfaces.
- **Avaya IP Office Small Office Edition - 64MB Memory Card *(700289721)***
  64M PCMCIA format memory card for embedded auto-attendant and voicemail.
- **Avaya IP Office Small Office Edition - Wireless LAN Card *(700289739)***
  PCMCIA Wireless card providing IEEE 802.11b based Access Point functionality (also requires IP400 Access Point RFA license).

IP406 V2 Control Units

Includes: 8 x Digital Station ports, 2 x analog station (POTS) ports, 1 x compact flash slot for embedded voicemail option., 8-port Layer-2 LAN switch, 9-pin DTE serial port, 37-pin WAN port, Music-on-Hold audio input and 2-switch external door-relay control port. Internal expansion slots to support 1 x 12-port remote access modem module and 1 x Voice Compression Module (up to VCM30 for non-blocking IP/PRI applications). 6 x external expansion module ports to support additional analog trunks, WAN interfaces, digital or analog extensions. Includes 60W earthed external power supply. Regional power cord and software/documentation CD pack not included.

- **IP406 V2 Office Mu-Law *(700359946)***
  Base unit pre-configured for Mu-law companding and US locale settings. 2 x trunk module slots to support US T1 PRI and 4-port analog trunk cards.
- **IP406 Office V2 A-Law *(700343536)***
  Base unit pre-configured for A-law companding and multi-country locale settings. 2 x trunk module slots to support Euro-ISDN BRI, E1/PRI and 4-port analog trunk cards. software/documentation CD pack not included.

IP412 Control Units

Includes: 2-port Layer-2 LAN switch, 9-pin DTE serial port, 37-pin WAN port, Music-on-Hold audio input and 2-switch external door-relay control port. Internal expansion slots to support 1 x 12-port remote access modem module and 2 x Voice Compression Modules (including VCM24 and 30 for non-blocking IP/dual-PRI applications). 12 x external expansion module ports to support additional analog trunks, WAN interfaces, digital or analog extensions. Includes 40W external power supply. Regional power cord and software/documentation CD pack not included.

- **IP412 Office Mu-Law Base Unit (700350408)**
  Base unit pre-configured for Mu-law companding and US locale settings. 2 x trunk module slots to support T1/PRI and 4-port analog trunk modules.
- **IP412 Office A-Law Base Unit (700234479)**
  Base unit pre-configured for A-law companding and multi-country locale settings. 2 x trunk module slots to support Euro-ISDN BRI, E1/PRI and 4-port analog trunk modules.

IP Office External Expansion Modules

- **Phone 8 Module V2** ***(700359896)***
  Adds an additional 8 Plain Ordinary Telephone ports to IP406 and IP412 control units.
- **Phone 16 Module V2** ***(700359904)***
  Adds an additional 16 Plain Ordinary Telephone ports to IP406 and IP412 control units.
- **Phone 30 Module V2** ***(700359912)***
  Adds an additional 30 Plain Ordinary Telephone ports to IP406 and IP412 control units.
- **Digital Station 16 Module V2** ***(700359839)***
  Adds an additional 16 Digital Station ports to IP406 and IP412 control units.
- **Digital Station 30 Module V2** ***(700359847)***
  Adds an additional 30 Digital Station ports to IP406 and IP412 control units.
- **So8 Module** ***(700185077)***
  Provides 8 ISDN S-interface device lines to the desktop.
- **Analog Trunk 16 - North America only** ***(700211360)***
  Provides an additional 16 Analog trunks (loop start or ground start) and two power fail sockets.
- **Analog Trunk 16 EU** ***(700241680)***
  Provides an additional 16 Analog trunks (loop start) and two power fail sockets. European CTR21 specification.
- **Analog Trunk 16 NZ** ***(700241698)***
  Provides an additional 16 Analog trunks (loop start) and two power fail sockets. New Zealand specification.
- **WAN3 10/100 Module** ***(700262009)***
  Provides an additional three V.24/V.35/X.21 ports. This expansion module is connected to the IP406 and IP412 control unit using the LAN and does not impact on the maximum number of external expansion modules supported.

Voice Compression Modules

- **Voice Compression Module 4 *(700359854)***
  5 Channel Voice Compression module required for IP trunks and extensions. Includes 64ms echo cancellation.
- **Voice Compression Module 8 *(700359862)***
  8 Channel Voice Compression module required for IP trunks and extensions. Includes 64ms echo cancellation.
- **Voice Compression Module 16 *(700359870)***
  16 Channel Voice Compression module required for IP trunks and extensions. Includes 64ms echo cancellation.
- **Voice Compression Module 24 *(700359888)***
  24 Channel Voice Compression module required for IP trunks and extensions. Includes 64ms echo cancellation.
- **Voice Compression Module 30 *(700293939)***
  30 Channel Voice Compression module required for IP trunks and extensions. Includes 25ms echo cancellation.

Modems cards

- **IP400 Office Modem 12 *(700343452)***
  Internally fitted card allowing twelve simultaneous V.90 modem calls.

Trunk Interface Cards

- **IP400 Office BRI-8 (UNI) *(700262017)***
  Interface card for the Small Office Edition, IP406 and IP412 providing 4 x ISDN basic rate ports (8 lines).
- **IP400 Office PRI 30 E1 (1.4) *(700272461)***
  Interface card for the IP406 and IP412 providing 1 x ISDN Primary rate port (30 lines).
- **IP400 PRI 30 E1R2 RJ45 - CALA *(700241631)***
  Interface card for the IP406 and IP412 providing 1 x E1R2 Primary rate port (30 lines). RJ45 termination.
- **IP400 PRI 30 E1R2 COAX - CALA *(700241656)***
  Interface card for the IP406 and IP412 providing 1 x E1R2 Primary rate port (30 lines). Co-Ax termination.
- **IP400 Office Dual PRI E1 *(700185184)***
  Interface card for the IP406 and IP412 providing 2 x ISDN Primary rate ports (60 lines).
- **IP400 Office PRI T1 *(700185200)***
  Interface card for the IP406 and IP412 providing 1 x T1/PRI port (24 lines).
- **IP400 Office Dual PRI T1 *(700185218)***
  Interface card for the IP406 and IP412 providing 2 x T1/PRI (48 lines).
- **IP400 Office Quad Analog Trunk (Universal) *(700359938)***
  Interface card for the IP406 and IP412 providing 4 x Loop start analog trunks. Universal variant supports specifications for North America, Europe and New Zealand.
- **IP400 ANLG 4 EU (LS) *(700241672)***
  Interface card for the IP406 and IP412 providing 4 x Loop start analog trunks (European CTR21 specification).
- **IP400 ANLG 4 NZ (LS) *(700241706)***
  Interface card for the IP406 and IP412 providing 4 x Loop start analog trunks (New Zealand specification).

Spares

The following are orderable spares available from Avaya.

Item	Color	Material Code
Replacement Handset	Dark Grey	700203797
HDST HIP QD CORD- 4606/16/24/30 SETS		700212442
Amplified Handset	Dark Grey	700229735
Noisy Location Handset	Dark Grey	700229743
Push to Talk Handset	Dark Grey	700229727
Cat 5 Cable specific to 4620		700261613
6416 Stand	Grey	848219127
6416/24D&M Stand	White	848219119
Stand for 6402/6408 phones.	Grey	108933169
Stand for 6402/6408 phones.	White	108933177
24 Button expansion module for 5620/5420/4620/2420	Grey	700203656
Handset Cords 25ft	Dark Grey	700217417
IP PHONE MOD CORD 1 FT CAT5	–	408406932
IP PHONE MOD CORD 7 FT CAT5	–	408406957
IP PHONE MOD CORD 14 FT CAT5	–	408406940
1151B1 Power supply	–	700227242
1151B2 Power supply with battery backup	–	700237019
1151B1 Power supply and CAT 5 cable for use with IP terminals	–	175707
1151B1 Power supply and CAT 3 cable for use with digital terminals	–	175706
1151B2 Power supply with battery backup and CAT 5 cable for use with IP terminals	–	177086
1151B2 Power supply with battery backup and CAT 3 cable for use with digital terminals	–	177087
Power Cord INPUT 10A - European - 106336 CRD31	–	106336
Power Cord 98IN European 12013S	–	407786623
Power Cord 98IN United Kingdom 14012	–	407786599
IP PHONES Power 1152A1 Mid-Span	–	700180433

Country Availability

IP Office is available in the following countries. Please refer to your country price list for the availability of individual items.

- **Australia**
- **Belgium**
- **Brazil**
- **Canada**
- **Chile**
- **China**
- **Colombia**
- **Croatia**
- **Denmark**
- **Finland**
- **France**
- **Germany**
- **Hong Kong**
- **Hungary**
- **Iceland**
- **Ireland**
- **Italy**
- **Japan**
- **Korea**
- **Luxembourg**
- **Mexico**
- **Netherlands**
- **New Zealand**
- **Norway**
- **Peru**
- **Poland**
- **Portugal**
- **Russia**
- **Spain**
- **Sweden**
- **Switzerland**
- **Taiwan**
- **United Kingdom**
- **USA**

Sample Configurations

IP406 Office

Scenario 1:

A customer in Europe with sophisticated telephony requirements, needing 30 exchange lines and 80 digital extensions.

This configuration provides support for up to 98 Avaya digital extensions (18 spare for growth) and a single Primary Rate Euro-ISDN connection (30 channels) . If growth beyond 98 users or additional trunk capacity is anticipated, up to 3 more external expansion modules (another 90 extensions) and another trunk card (up to 60 additional channels) can be fitted. Typically, a business of this size has a data network that interconnects its users and provides access to business applications, front and back office systems as well as internet resources. The IP406 Office can be connected to this network through its integrated 8-port LAN switch. This provides all users with access to the business communications and personal productivity applications supported by IP Office.

Kit List

- 1 x IP406 Office DS control unit.
- 4 x Region specific power cords.
- 1 x PRI 30 E1 trunk card.
- 3 x Digital Station 30 external expansion modules.
- 80 x Avaya 5410 digital feature phones.

Scenario 2:

A business in the US requiring 32 analog telephones and one PRI (23+1D channels).

The IP406 Office with a single T1 PRI card and two Phone 16 external expansion modules provides the required line and extension capacity. The Phone Manager Lite application enhances the capabilities of each analog telephone, by enabling user to handle calls and control their extension settings through a PC-based interface. For future growth, the system can support a further 4 external expansion modules and one additional internal trunk card.

Kit List

- 1 x IP406 Office DS control unit.
- 2 x Region specific power cords.
- 1 x Single T1 PRI trunk card.
- 1 x IP400 Office Phone 16 external expansion module.

IP412

Scenario:

A US business requiring 180 display phones and 96 digital trunks with 20 analog lines for fallback purposes.

The configuration illustrates a IP412 providing 180 extensions and 96 digital trunks (4 x T1) and two IP400 Office Analog Trunk 16 modules offering capacity of up to 32 analog trunk lines . Factory shipped with a single Dual PRI T1 interface, the system is fitted with an extra trunk card in its spare slot to provide the additional 48 lines.

Kit List

- 1 x IP412 control unit.
- 9 x Region specific power cords.
- 2 x PRI 48 T1 trunk cards.
- 6 X IP400 Office Digital Station 30 external expansion modules.
- 2 x IP400 Office Analog Trunk 16 external expansion modules.
- 180 x Avaya 5410 digital phones.

Scenario 2:

A Business requiring 90 IP hardphones, 90 IP softphones and 60 lines.

An organization, based in Asia requires a LAN-based telephony system with 60 IP hardphones, 60 IP softphones and 60 external trunk lines for its main location and the ability to network with other sites using IP trunking.

This configuration illustrates an IP412 PRI 60 E1 fitted with two 30-channel Voice Compression Modules (VCMs). These two internally fitted cards allow up 60 simultaneous calls to external parties (IP extension calling a non-IP telephone or line). For IP to IP calls, VCM resources are only required for initial call set-up. Depending on the typical utilization of external trunks, a lower capacity VCM variant could be employed, as appropriate.

The IP Office softphone is 'Phone Manager Pro PC Softphone' which is an enhanced version of the standard Phone Manager Pro application enabled for each user using two License Keys as listed below.

Kit List

- 1 x IP412 control unit.
- 1 x PRI 60 E1 trunk card.
- x Region specific power cords.
- 2 x IP400 VCM 30 cards.
- 60 x 5610 IP phones.
- 1 x IP400 Phone Manager Pro RFA 50.
- 1 x IP400 Phone Manager Pro RFA 10.
- 1 x IP400 IPPro RFA 50.
- 1 x IP400 IPPro RFA 10.

B: TAPI Functions Supported by IP Office

TAPI 2.1 Functions Supported

TAPI*Link* Lite provides the following functionality for TAPI 2.1:

- lineAddToConference
- lineAnswer
- lineBlindtransfer
- lineCompleteTransfer
- lineConfigDialog
- lineClose
- lineDeallocateCall
- lineDial
- lineDrop
- lineGetAddressCaps
- lineGetAddressID
- lineGetAddressStatus
- lineGetAppPriority
- lineGetCallInfo
- lineGetCallStatus
- lineGetDevCaps
- lineGetID
- lineHold
- lineInitialiseEx
- lineMakeCall
- lineNegotiateTAPIVersion
- lineOpen
- linePark
- lineRedirect
- lineRemoveFromConference
- lineSetAppPriority
- lineSetAppSpecific
- lineSetCallPrivilege
- lineSetStatusMessages
- lineSetupTransfer
- lineShutdown
- lineSwapHold
- lineUnhold
- lineUnpark
- lineSetCallData
- lineDevSpecific
- lineGenerateDigits
- lineGenerateTone
- lineMonitorDigits
- lineMonitorTones

TAPI 3.0 functions supported

The following functions are supported using TAPI 3.0:

- **ITTAPI**
- Initialize
- Shutdown
- EnumerateAddresses
- RegisterCallNotifications
- Put_EventFilter

- **ITAddress**
- get_AddressName
- get_dialableAddress
- get_ServiceProviderName
- CreateCall

- **ITMediaSupport**
- get_MediaTypes

- **ITCallInfo**
- get_Address
- get_CallState
- get_CallInfoString
- SetCallInfoBuffer

- **ITBasicCallControl**
- Connect
- Answer
- Disconnect
- Hold
- SwapHold
- ParkDirect
- Unpark
- BlindTransfer
- Transfer

- **ITCallStateEvent**
- get_Cause
- get_State
- get_Call

- **ITCallNotificationEvent**
- get_Call

- **ITCallInfoChangeEvent**
- get_Call

- **ITCallHubEvent**
- get_Event
- get_Call

Notes:

- TAPI*Link* Lite can be used from C, C++ and Delphi. Visual Basic cannot directly use TAPI 2.1, but does support TAPI 3.0 without any third-party tools.
- TAPI*Link* Lite provides detailed information on telephony events, including the ability to screen-pop based on CLI and/or DDI.

Changes from previous versions of IP Office

TAPI Reserved Fields

TAPI fields that were previously reserved by IP Office for internal use have now been released for general use by developers. A full definition of theses fields are contained in the IP Office developers SDK CD. The following table shows the device specific data available via TAPI.

- Phone's extension number
- Forward on busy flag
- Forward on no answer flag
- Forward unconditional flag
- Forward hunt group flag
- Do not disturb flag
- Outgoing call bar flag
- Call waiting on flag
- Voicemail on flag
- Voicemail ring-back flag
- Number of voicemail messages
- Number of unread voicemail messages
- Outside call sequence number
- Inside call sequence number
- Ring back sequence number
- No answer timeout period
- Wrap up time period
- Can intrude flag
- Cannot be intruded upon flag
- X directory flag
- Force login flag
- Login code flag
- System phone flag
- Absent message id
- Absent message set flag
- Voicemail email mode
- User's extension number
- Users Locale
- Forward number
- Follow me number
- Absent text
- Do not disturb exception list
- Forward on busy number
- User's priority
- Number of groups the user is a member of
- Number of groups that the user is a member of that are currently outside their time profile
- Number of groups the user is currently disabled from
- Number of groups that the user is a member of that are currently out of service
- Number of groups that the user is a member of that are currently on night service

DevLink Reserved Fields

DevLink fields that were previously reserved by IP Office for internal use have now been released for general use by developers. A full definition of these fields is contained on the IP Office 2.0 developers SDK CD. The following table shows the device specific data available via DevLink. A "Y" in the column indicates that the field is already described in the DevLink manual.

#	Field Data (S Message)	#	Field Data (S Message)
1	A call id	26	Voicemail disallow
2	B call id	27	Sending complete
3	A state	28	Bc.tc,bc.tm
4	B state	29	Owner hunt group name
5	A connected	30	Original hunt group name
6	A is music	31	Original user name
7	B connected	32	Target hunt group name
8	B is music	33	Target user name
9	A name	34	Target RAS name
10	B name	35	Is internal call
11	B list (possible targets for the call)	36	Time stamp
12	A slot ,channel	37	Connected time
13	B slot , channel	38	Ring time
14	Called party presentation & type	39	Connected duration
15	Called party number	40	Ring duration
16	Calling party presentation & type	41	Locale
17	Calling party number	42	Park slot number
18	Called sub address	43	Call waiting
19	Calling sub address	44	Tag
20	Dialled party type	45	Transferring
21	Dialled party number	46	Sv active
22	Keypad type	47	Sv quota used
23	Keypad number	48	Sv quota time
24	Ring attempt count	49	Account code
25	Cause	50	Unique call identifier
#	**Field Data (D Message)**	**#**	**Field Data (A Message)**
1	A call id	1	A call id
2	B call id	2	B call id
3	Unique call identifier	3	Unique call identifier

C: Technical Specifications

General

Dimensions

Unit Dimensions (mm/inches)	Width	Height	Depth
IP406 V2, IP412 and all Expansion Modules	445mm/17.5"	71mm/2.8"	245mm/9.7"
IP Office - Small Office Edition	255mm/10.0"	76mm/3.0"	241mm/9.5"

- The recommended minimum clearance, front and rear, for the connection of cables and other devices is 75mm/3".

Environmental

- 0°C to +40°C (32°F to 104°F). 95% relative humidity, non-condensing.

Telephone Cable Lengths

The following table details the maximum cable lengths supported for the telephone range using AWG22, 24 and 26 cabling. These figures assume that standard twisted-pair telephone cable or CAT5 network cable is used.

Telephone	AWG22	AWG24 (~ 0.5mm Ø)	AWG26
2400/5400 Series	1.67km - 5500 feet	1.1km - 3500 feet	0.67km - 2200 feet
4406D	1km - 3280 feet	1km - 3280 feet	0.4km - 1310 feet
4412D	1km - 3280 feet	0.7km - 2295 feet	0.4km - 1310 feet
4424D	0.5km - 1640 feet	0.5km - 1640 feet	0.4km - 1310 feet
6400 Series	1km - 3280 feet	1km - 3280 feet	0.4km - 1310 feet
T3 Series	1km - 3280 feet	1km - 3280 feet	0.4km - 1310 feet
POTs	1km - 3280 feet	1km - 3280 feet	0.5km - 1640 feet

Weight

Unit	Weight
IP406 V2 Control Unit	3.0Kg/6.7lbs
IP412 Control Unit	3.0Kg/6.7lbs
IP Office - Small Office Edition	1.2Kg/2.6lbs
Analog 16 Module	2.9Kg/6.5lbs
DS 16 Module	3.0Kg/6.7lbs
DS 30 Module	3.5Kg/7.8lbs
WAN3 Module	2.8Kg/6.3lbs
So8 Module	2.8Kg/6.3lbs
Phone 8 Module	2.8Kg/6.3lbs
Phone 16 Module	2.9Kg/6.5lbs
Phone 30 Module	3.1Kg/6.94lbs

Heat Dissipation

Note that the above numbers are for reference only. For practical purposes, for example the calculation of heat dissipation, it is recommended to base environmental requirements (for example air cooling or UPS ratings) on the maximum input rating of the power supplies of the planned IP Office configuration, as follows.

In order to calculate the maximum, that is worst case, amount of heat that can be generated by an IP Office system, it is assumed that all input power is converted to heat; whether from the PSU itself, the system unit, expansion module and/or cabling.

Heat dissipation is normally measured in British Thermal Units (BTU's). A heat value expressed in Watts can be converted to BTU/hr by multiplying by 3.41297. As indicated above, you should use the maximum power input of 115 VA of each power supply to calculate this most accurately

Using the conversion factor:

- Heat Dissipation = 115 x 3.41297 = 392.5 BTU/hour.

The metric equivalent to BTU is a Joule where 1 BTU = 1,055 Joules.

This calculates the BTU value per power supply. The maximum BTU per system is therefore calculated, based on total number of power supplies installed in the system. For example, for a IP412, this would be 1 for the base unit and up to 12 for the expansion modules.

- IP412 Maximum Heat Dissipation = 13 x 392.5 = 5,103 BTU/hr.

Remember to budget for the power requirements of any additional devices that are to be co-located with the IP Office such as server PC's (voicemail, etc).

Power Supply

- **Input**
    - **Small Office Edition:** 2.5mm DC inlet socket. 24Vdc power input. Rating 24V DC, 1.8A maximum.
    - **All Other Units:** 2.5mm DC inlet socket. 24Vdc power input. Rating 24V DC, 2A maximum.

- **Power Supply Units:** All CE/UL/Dentori Safety Approved.
    - **Standard 40W Power Supply Unit** (All control and expansion units unless otherwise indicated) Supplied with the control or expansion unit. 40W PSU with integral lead to the unit. Connection to switched mains supply requires separately supplied country specific IEC 60320 C7 power cord (2-wire figure 8 connector).
        - Input: 100-240V AC, 50/60Hz, 81-115VA, 2A maximum.
        - Output: 24Vdc, 1.875A, output power 45W maximum.
    - **Small Office 45W Power Supply Unit**
      Supplied with the unit. 45W PSU with integral lead to control unit. Connection to switched mains supply requires separately supplied country specific IEC 60320 C13 power cord (3-wire earthed cold kettle lead).
        - Input: 100-240V AC, 50/60Hz, 81-115VA, 1.5A maximum.
        - Output: 24V DC, 1.875A, output power 45W maximum.
    - **IP406 V2 60W Power Supply Unit**
      Supplied with the control or expansion unit. 60W PSU with integral lead to the unit. Connection to switched mains supply requires separately supplied country specific IEC 60320 C13 power cord (3-wire earthed cold kettle lead).
        - Input: 100-240V AC, 50/60Hz, 81-115VA, 2.5A maximum.
        - Output: 24V DC, 1.5A, output power 60W maximum.

Interfaces

Interface	Information
DTE Port	• 25 way D-Type female connector, V.24/V.28. • 9 way D-type on IP412, IP406 V2 and IP Office - Small Office Edition.
ISDN Ports	**EU/JP Interfaces:** • **BRI:** RJ45 sockets. ETSI S/T Interface to CTR3 for Pan European Connection. • **PRI E1:** RJ45 socket. ETSI S/T Interface to CTR4 for Pan European Connection. • **PRI T1/J1:** RJ45 socket: FCC Part 68/JATE connection. **USA Interfaces:** • **PRI T1 Service:** Ground Start (GS) – Default, E&M, 56k data for 5ESS, 56/64/64 restricted for 4ESS. • **PRI ISDN Switch support:** 4ESS, 5ESS, DMS-100, DMS-250 (includes conformance to ANSI T1.607 & Bellcore Special Report SR4287, 1992). • **PRI ISDN Services:** AT&T Megacom 800, AT&T WATS (4ESS), AT&T SDS Accunet 56kB/s & 64kB/s (4ESS), AT&T Multiquest (4ESS).
Analog Trunk Ports	• **RJ45 sockets:** Loop start/Ground start (regional dependant)
Power Fail Ports	• **RJ45 sockets:** Telephone ports act as master sockets.
ISDN Data Rates	• **BRI:** B-channel 64kbps or 56kbps, D-channel 16kbps. • **PRI:** B-channel 64kbps or 56kbps, D-channel 64kbps.
Analog Phone Ports	• **RJ45 sockets:** EU - Telephone ports act as Master sockets. • **CLI Schemes:** DTMFA, DTMFC, DTMFD, FSK and UK20. • **REN:** 2. (External Bell via POT port: REN = 1) • **Off Hook Current:** 25mA. • **Ring Voltage:** 40V (nominal) RMS.
LAN	• RJ45 sockets. Auto-negotiating 10/100 BaseT Ethernet (10/100Mbps).
WAN	• Small Office Edition: RJ45 Ethernet socket. • All Other Control Units (optional on Small Office Edition): 37 way D-Type female sockets. X.21 interface to 2048k bps, V.35 interface to 2048Kbps and V.24 Interface to 19.2Kbps.
Audio	• 3.5mm Stereo Jack socket. Input impedance - 10k /channel. • Maximum AC signal – 200mV rms.
External Output Port	• 3.5mm Stereo Jack socket. Switching Capacity - 0.7A. • Maximum Voltage - 55V DC. On state resistance - 0.7. • Short circuit current - 1A. Reverse circuit current capacity - 1.4A.
Wireless Module	• Small Office Edition only. • 16bit Type II PCMCIA format PC card. • IEEE 802.11b WiFi.
Embedded Voice Memory	• Small Office Edition: 64MB Flash memory, 16bit Type II PCMCIA card. • IP406 V2: 512MB Compact Flash memory card.

Protocols

Protocol	RFC	Information
V120	-	A standard Rate Adaptation mechanism.
V110	-	A standard Rate Adaptation mechanism.
PPP	RFC1661	Point to Point Protocol.
LCP	RFC1570	Link Control Protocol.
MP	RFC1990	Multi-Link (Point to Point) Protocol.
IPCP	RFC1332	Internet Protocol Control Protocol.
PAP	RFC1334	Password Authentication Protocol.
RTP/RTCP	RFC1889	Real Time and Real Time Control Protocol.
CHAP	RFC1994	Challenge Handshake Authentication Protocol.
CCP	RFC1962	Compression Control Protocol.
STAC	RFC1974	STAC LZS Compression Protocol.
MPPC	RFC2118	Microsoft Point to Point Compression (Protocol).
BACP	RFC2125	Bandwidth Allocation Control Protocol.
UDP	RFC768	User Datagram Protocol.
IP	RFC791	Internet Protocol.
TCP	RFC793	Transmission Control Protocol.
DHCP	RFC1533	Dynamic Host Control Protocol.
NAT	RFC1631	Network Address Translation.
BOOTP	RFC951	Bootstrap Protocol.
TFTP	RFC1350	Trivial File Transfer Protocol.
NTP	RFC868	Network Time Protocol.
SNMPv1	RFC1157	Simple Network Management Protocol. (STD15)
	RFC1155	Structure and identification of management information for TCP/IP based internets. (STD16)
	RFC1212	Concise MIB Definitions. (STD16)
	RFC1215	A convention for defining traps for use with SNMP.
MIB-II	RFC1213	Managment Information base for network management of TCP/IP based internets: MIB-II. (STD17)
ENTITY MIB	RFC2737	Entity MIB (Version 2).
RIP	RFC1058	Routing Information Protocol.
	RFC2453	RIP Version 2. (STD56)
	RFC1722	RIP Version 2 Protocol Applicability Statement. (STD57)
IPSec	RFC2401	Security Architecture for the Internet Protocol.
	RFC2402	IP Authentication Header.
	RFC2403	The Use of HMAC-MD5-96 within ESP and AH.
	RFC2404	The Use of HMAC-SHA-1-96 within ESP and AH.
	RFC2405	The ESP DES-CBC Cipher Algorithm with Explicit IV.
	RFC2406	IP Encapsulation Security Payload. (ESP)
	RFC2407	The Internet IP Security Domain of Interpolation for ISAKMP.
	RFC2408	Internet Security Association and Key Management Protocol.
	RFC2409	The Internet Key Exchange.

	RFC2410	The NULL Encryption Algorithm and its Use with IPSec.
	RFC2411	IP Security Document Roadmap.
L2TP	RFC2661	Layer Two Tunneling Protocol "L2TP".
	RFC3193	Securing L2TP using IPSec.
Header Compression	RFC2507	IP Header Compression (IPHC).
	RFC2508	Compressing IP/UDP/RTP Headers for Low-Speed Serial Links.
	RFC2509	IP Header Compression over PPP.
DiffServ	RFC2474	Definition of the Differentiated Services Field (DS Field) in the IPv4 and IPv6 Headers.
PPP MP	RFC1990	The PPP Multilink Protocol (MP).
Frame Relay Encapsulation	RFC1490	Multi protocol Interconnect over Frame Relay.
ML-PPP	RFC2686	The Multi-Class Extension to Multi-Link PPP.

D: Software History

History

This section details the main features of previous IP Office releases. This is not a definitive list are changes and addition many also occur during interim maintenance releases.

- **IP Office 3.0.**
- **IP Office 2.1.**
- **IP Office 2.0.**
- **IP Office 1.4.**
- **IP Office 1.3.2.**
- **IP Office 1.3.**

IP Office 3.0

Note: IP Office 3.0 removes support for DT ports and therefore associated DT port phones (Avaya 2030, 2059, 20CC and 20DS). Small Office Edition control units with integral DT ports and Digital Terminal expansion modules are no longer supported. Other control units with integral DT ports are still supported but use of their DT ports is not.

New hardware capabilities:

- New Digital Phones - 5400 Series: 5402, 5410 and 5420.
- New IP Phones - 5600 Series: 5601, 5602, 5610 and 5620.
- Support for 2402 and 2410 Digital Phones
- Support for 4601 and 4610 IP Sets
- EU24 Expansion Module
- Modem 12 Card
- Embedded Voicemail for IP406 V2

System software enhancements:

- Key System Features
    - Call Appearance
    - Bridged Appearance
    - Line Appearance
    - Call Coverage
    - Idle Line Preference
    - Ringing Line Preference
    - Hold Functionality redesign
    - LED Feedback redesign
    - Distinctive Ringing
- SNMP Enhancement
    - Disk space alarms for Voicemail Pro and Embedded Voicemail.
- Enhancements to Embedded Voicemail
    - Default message length - increased to 2 minutes, configurable to a maximum of 3 minutes.
    - Auto-attendant time-out - in the absence of DTMF input the caller will time-out to a pre-defined position

Changes in Manager

- Call Coverage Tab removed from "User Form" and replace by "Coverage Appearance" button option.
- Gain Control for IP Phones added to "VoIP" tab in Extension form

IP Office Voicemail Pro

- Intuity Mode Personal Distribution Lists
- Group Message Broadcast
- Introduction of ContactStore for IP Office
- Fax Server Support tested and verifed
    - Equisys - Zetafax
    - Captaris - RightFax
    - Fenestrae – Faxination
    - GFI - GFI FAXMaker
- SNMP Alarms: Disk Full Warning

IP Office Conferencing Center

- Local Address Book
- Conference Templates

Phone Manager

- Profiles.
- Compact Mode (Pro Only).
- Speed Dial Enhancements - 10 Tabs 100 per Tab.
- Personal Distribution List Support (Pro & Intuity Mode Voicemail Pro).
- Microsoft LIVE Communication Server Support - for Instant Messaging.
- Import/Export of Local Directories (Pro Only).
- Call History Enhancements (Pro Only).
- Programmable Date and Time Format (Pro Only).
- Phone Manager PC Softphone USB Settings.

SoftConsole

- • The Call Information Panel now has the ability to show multiple calls waiting. This allows the SoftConsole user to either answer calls from the Call Information panel based on the Caller ID or from the Queuing Panel based on the dialled number (target hunt group).

Wizards

- Password Protection
- Wizard Support for Embedded Voicemail and Voicemail Pro

CCC Compatibility

- IP Office R3.0 is compatible with both CCCv4 and CCCv5, however the new Key and Lamp features are not supported.

IP Office 2.1

New hardware capabilities:

- Change to IP403 digital trunk card restrictions. Single PRI cards now allowed in either slot.
- Integral CSU support (applicable for North America only).
- T1 Support for Small Office Edition (applicable for North America only).
- 3810 Wireless Handset (applicable for North America only).
- IP406 V2 with release 2.1.27 core software.

Core Software:

- Reliable disconnect on analog lines with and without ICLID.
- PIN restricted terminals.
- Paging over IP Phones.
- Integrated VPN capabilities.
- All Hunt Group and User parameters are now mergeable.
- Personal ringing on 4400 Series phones, replaced by distinctive ringing in IP Office 3.0.

Management Tools:

- Streamlined Installation and Administration Wizard.
- IP Office - Small Office Edition Wizard.
- Moves, Adds and Changes Wizard.

Compact Business Center (CBC)

- Added new alarms: Lost Calls, Trunk Utilization, Calls Queued and Available Agents.
- New Key Performance Screen: Current Alarm, Last Alarm and Longest Call Waiting Alarm.
- Trunk Utilization Graphs.
- Email Notification.
- New language support - Italian & Russian.

Voicemail Pro:

- Email reading enabled via TTS.
- Call Data Tagging.
- VM Pro Fax Detection without call flows.
- VB Scripting "GetDTMF" & RecFile" function present a beep on calls to indicate start of recording of DTMF entries.
- Voicemail Pro Networked Messaging.

IP Office Conferencing Center:

- Web-based Conference booking and scheduling system (requires Windows 2000/2003 server running IIS):
    - Web-based conference booking tool to reserve conference resources (immediate or future).
    - Ability to select “Listen-only” or “Speak & Listen” mode for each participant.
    - Email notification to all participants.
    - Voice Conference Notification (VCN) to dial out participants.
    - Participants name announcements as they enter/leave the conference bridge.
    - Unique computer-generated Conference ID for security.
    - Unique PIN code for each participant for security and authentication.
    - Web-based reports on conference usage and voting results.
- Value Proposition Web client interface (requires Internet Explorer 6.0 or higher) enabling:
    - Real-time view of participants status (Dialed in, Logged on to Web client, Speak & Listen, Listen Only).
    - Ability for the host to change participant status in real-time.
    - Ability for participants in listen-only mode to request the right to speak (raise hand function).
    - Mute All / Un-Mute All facility for the host.
    - Whisper facility for the host to have a private conversation with one of the participants.
    - Viewing area for reviewing PowerPoint™ presentations (or any other document saved in html format).

IP Office 2.0

New hardware capabilities:

- VCM 30 module (IP412 Only).
- Avaya 4620 IP Telephone.
- Avaya 3616 Wireless Telephone.
- Avaya 3626 Wireless Telephone.
- Avaya IP Office - Small Office Edition Platform.
    - Embedded Voicemail Card for Small Office Edition.
    - WiFi (802.11b) card for Small Office Edition.
- Serial Dongle.
- Surge protection module (IROB) – Protects analog extension ports
- Surge protection module (IROB) rack mount kit

System software enhancements:

- Voice features:
    - Increased extension support on the IP412 from 256 to 360.
    - Call Priority for incoming callers.
    - Transfer Recall/Return.
    - Auto Attendant Fall Back Extension.
    - Enhanced Call Recording and Intrusion configuration options.
- Data features:
    - RIP support.
    - V.32 modem with V.42 error detection and correction on first Analog port of ATM4 trunk card and Small Office Edition integral analog trunks.
- Management:
    - New installation and administration wizard.
    - SNMP Notifications for centralized monitoring.

PhoneManager:

- New look and feel.
- Support for Voicemail Pro Intuity TUI mode.
- Screen-pop support for Symantec ACT!, Goldmine, Maximizer as well as MS Outlook.
- DiffServ QoS support.
- Call record start and stop in conjunction with VM Pro.
- Post connect dial and DTMF.

Soft Console:

- New Soft Console replaces eConsole and eBLF applications.
- BLF integrated into Console screen.
- Simplified and efficient new user interface.
- Enhanced directory access.
- Record start and stop in conjunction with VM Pro.
- Multi-profile support.

Voicemail Pro:

- 3rd party IVR database support.
- Text to Speech within call flows.
- Text to Speech for listening to emails.
- Fax detection within auto attendant call flows.
- Enhanced Audio compression for Integrated Messaging Pro users.
- Forwarding voicemail as email to non MS Exchange email servers via SMTP.
- Improved housekeeping for message storage.
- WAV editor with the capability to utilize a telephone.
- Additional Personal Greetings (Internal, External, Busy, Out of Hours, No Reply).
- Additional Message Capture options (Private and Priority).

IP Office 1.4

New hardware capabilities:

- New Digital Phone 2420
- Enhanced WAN 3 Module

System software enhancements:

- Increased networking capabilities through support of quality of service over Frame Relay networks
- Additional phone devices supported on the core platform.
- Increased system performance and reliability through improvements to the core system software.
- The use of remote access software for system support via DameWare.
- Localization for China and Russia country connection.

Applications enhancements:

- IP Office Compact Contact Center Version 4.

IP Office 1.3.2

System software enhancements:

- Increased networking capabilities through increased support of the QSIG protocol, including Intuity AUDIX (via DEFINITY and MultiVantage platforms).
- Increased system performance and reliability through improvements to the core system software.

IP Office 1.3

New hardware capabilities:

- IP412 control unit.
- Dual T1/PRI trunk cards (supported only on IP412 only).
- Support for Avaya 4602 IP Hard Phone (Release 1.6.69).
- Support for VCM 20 (Voice Compression Module 20 channels) on the IP403 Office.

System software enhancements:

- Larger capacity: support for 256 endpoints on IP412.
- Enhanced Boss/Secretary operation including call coverage.
- Improved IP Telephone operation, including support for the 4600 Series Terminal 1.6.17 software single-connect and support for the 4602 IP terminal.
- Better networking and interoperability, including QSIG enhancements and Name on PRI.
- More efficient use of feature keys, including support for single button on/off control for popular features.

Applications enhancements:

- Voicemail Pro Release 1.2. (12), including Dial by Name and Pin Code Check for Meet Me Conferencing
- Phone Manager Pro Release 1.3, including Per Seat Licensing and Agent Enabled.
- CTI Link 1.5
- Improved interface to 3rd party call accounting packages through IP Office SMDR.
- Support for Windows XP on client applications.

E: Miscellaneous

Discontinued Units

The following items are still supported by IP Office 3.1 but are no longer available from Avaya. This page and any other references to these units within the Product Description are for reference only.

IP Office Control Units

- **Small Office 2T+4A (3 VoIP)**
- **Small Office 4T+8A (3 VoIP)**
- **IP403 Office.**
- **IP406 Office V1.**

IP Office Expansion Modules

- **Phone Expansion Module (8, 16 and 30 port variants)**
  These expansion modules have been superseded by the equivalent Phone V2 expansion modules.
- **Digital Station Expansion Module (16 and 30 port variants)**
  These expansion modules have been superseded by the equivalent Digital Station V2 expansion modules
- **WAN3 Expansion Module**
  This module has been superseded by the WAN3 10/100 expansion module.

IP Office Trunk Interface Cards

- **ATM4, ATM4 EU and ATM4 NZ**
  These quad analog trunk cards have been superseded by the ATM4U (Universal).

IP Office Internal Daughter Cards

- **VCM5, VCM10, VCM20**
  These VCM cards have been superseded by the VCM4, VCM8 and VCM16 versions. Though the new cards have a lower channel capacity, they support echo-cancellation of 64ms rather than the formers 25ms.
- **Modem2**
  The Modem2 card has been superseded by the Internal Modem Card.

Avaya Phones

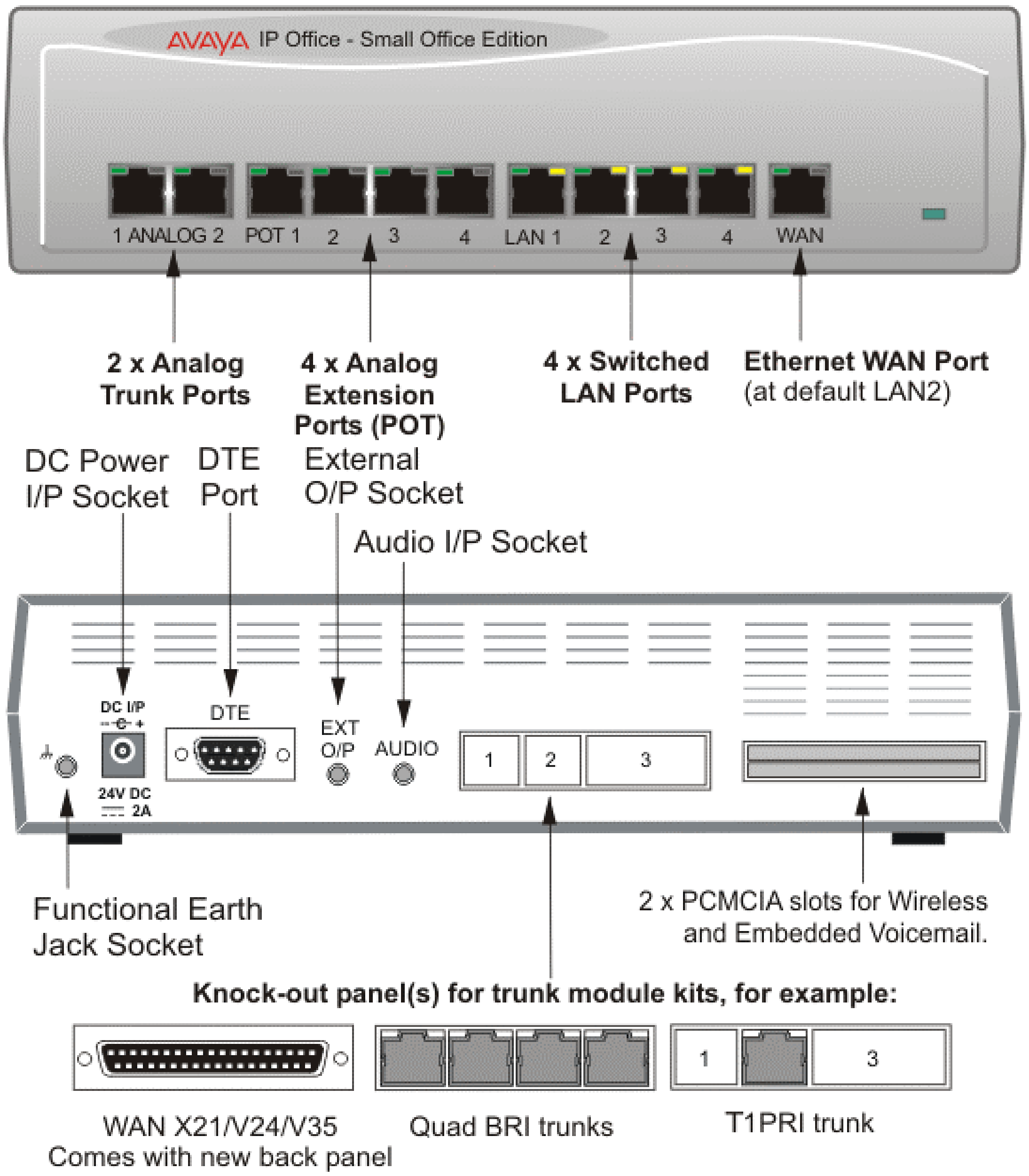
- 4602IP and 5602IP
  These phones are no longer available. The recommended IP Office replacement is the 5602SW.
- 4606, 4612 and 4624
  These phones are no longer available. The recommended IP Office replacements are the 5610SW and 5620SW.

IP Office - Small Office Edition 2T+4A (3 VoIP)

This unit is no longer available from Avaya but is still supported by IP Office 3.1 software. This section is included for reference for existing units.

The IP Office - Small Office Edition 2T+4A (3 VoIP) provides:

- Two Analog Loop Start Trunks (Caller ID enabled).
- Four analog extension (POT) ports. During power fail, analog trunk port 2 is connected to analog extension port 1.
- Three VoIP Codecs (G.723.1, G.711 and G.729a).
- 4 Switched Ethernet ports (Layer 2).
- Dedicated Switched Ethernet WAN port (Layer 3).
- Two PCMCIA slots for wireless and memory card support.
- WAN slot for optional WAN card (V35/V24/X.21, BRI or T1 PRI).
- DTE port.
- Audio port for external music on hold source.
- Two relay switch port for door entry systems (External O/P socket).

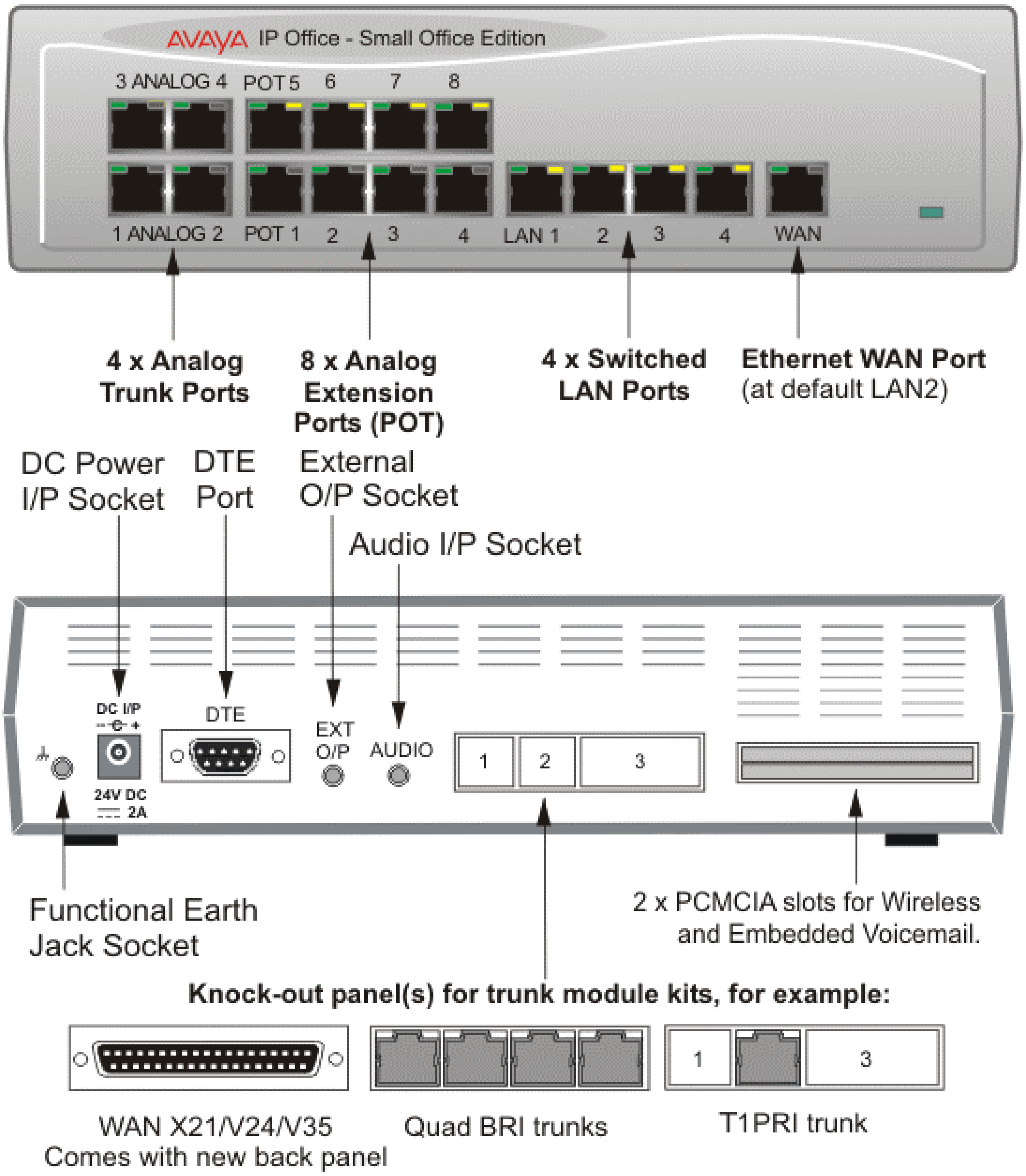


IP Office - Small Office Edition 4T+8A (3 VoIP)

This unit is no longer available from Avaya but is still supported by IP Office 3.1 software. This section is included for reference for existing units.

The IP Office - Small Office Edition 4T+8A (3 VoIP) provides:

- Four Analog Loop Start Trunks (Caller ID enabled).
- Eight analog extension ports (POT). During power fail, analog trunk port 2 is connected to analog extension port 1.
- Three VoIP Codecs (G.723.1, G.711 and G.729a).
- 4 Switched Ethernet ports (Layer 2).
- Dedicated Switched Ethernet WAN port (Layer 3).
- 2 x PCMCIA Slots for Wireless and Memory card support.
- WAN Slot for Optional WAN card (V35/V24/X.21, BRI or T1 PRI).
- DTE port.
- Audio port for external music on hold source.
- Two relay switch port for door entry systems (External O/P socket).



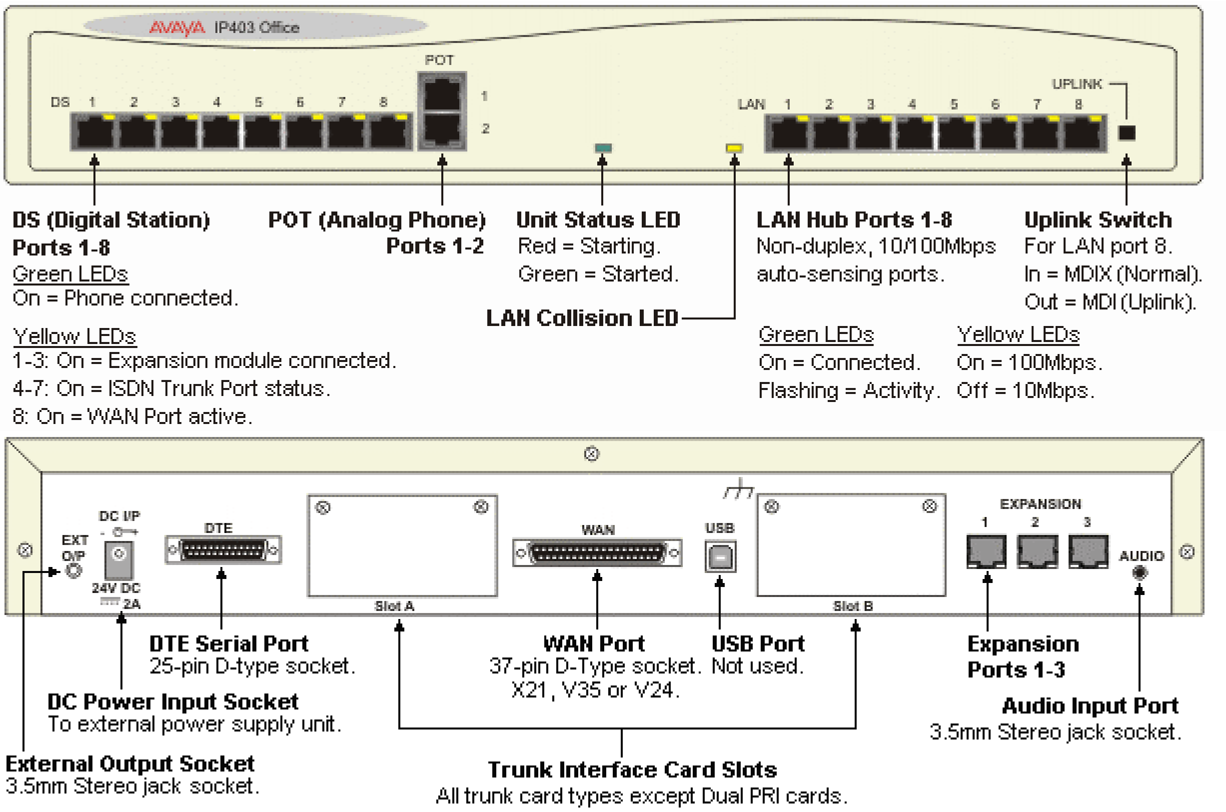
IP403 Office

This unit is no longer available from Avaya but is still supported by IP Office 3.1 software. This section is included for reference for existing units.

The IP403 Office control unit is a 19" rack mountable voice and data communication system and supports as standard -

- Eight Digital Station (DS) ports for selected 24xx, 44xx, 54xx and 64xx Series phones (plus 3810 and 9040 wireless (US) phones).
- Two Analog telephone ports.
- Eight 10/100 Mbps LAN Hub ports.
- DTE Port.
- X.21/V35 WAN interface.
- Support for 3 Expansion Modules.
- External output port containing two switches for door entry systems.
- Audio port for external music on hold source.
- 18 Data channels *(maximum of 10 useable for Voicemail Pro)*
  Note: A data channel is used whenever a call is made from the IP network to an exchange line (Central Office). For example, four people surfing the Internet will use a single data channel since they all share the same line to the ISP. Two people remotely accessing the Office LAN from home will use two data channels since they have dialed in on separate lines. IP extensions do not use data channels.

Optional internal upgrades allow for support of up to 4 x V.90 modem calls and a Voice Compression Module (VCM) of up to 20 channels. Through the support of up to three external Expansion Modules, IP403 office can be enhanced to support a further 90 Analog, Digital or IP phones.



IP406 Office V1

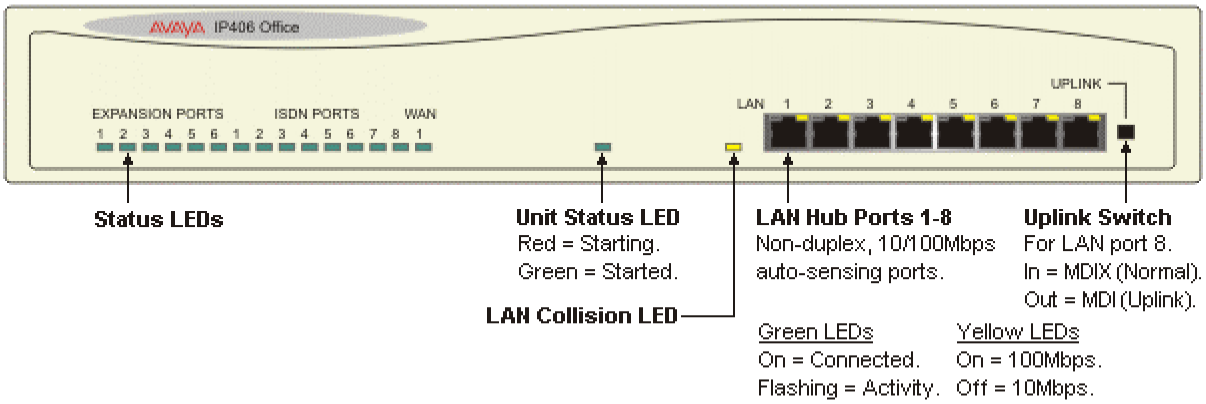
This unit is no longer available from Avaya but is still supported by IP Office 3.1 software. This section is included for reference for existing units.

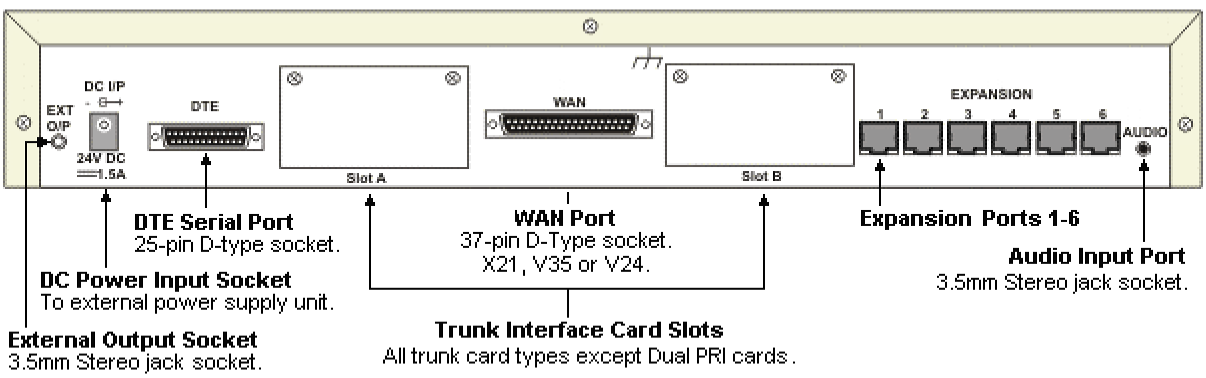
The IP406 V1 (formerly just called the IP406) differs from the IP403 Office in that it supports six expansion modules but excludes the integral Digital extension and Analog extension ports.

The IP406 V1 Office base unit is 19" rack mountable and supports as standard-

- Eight 10/100 Mbps LAN Hub ports.
- DTE Port.
- X.21/V35 WAN interface.
- Support for 6 Expansion Modules.
- External output port containing two switches for door entry systems.
- Audio port for external music on hold source.
- 24 Data channels *(Maximum 20 useable for Voicemail Pro)*.
  Note: A data channel is used whenever a call is made from the IP network to an exchange line (Central Office). For example, four people surfing the Internet will use a single data channel since they all share the same line to the ISP. Two people remotely accessing the Office LAN from home will use two data channels since they have dialed in on separate lines. IP extensions do not use data channels.

Optional internal upgrades allow for the support of up to 2 x V.90 modem calls and a single Voice Compression Module (VCM) of up to 20 channels. Through support of up to six external Expansion Modules, IP406 V1 office can be enhanced to support a mixture of Analog, Digital or IP phones to maximum of 180.





EU24

This unit is no longer available from Avaya but is still supported by IP Office 3.1 software. This section is included for reference for existing units.

The EU24 is an add-on unit that work in association with a 2420, 4620SW, 4621, 5420, and 5620SW. It provides an additional 24 programmable buttons with associated display label and status icons. Only one EU24 per phone.

Although the EU24 supports an additional 24 Call Appearance/Feature buttons, it displays only the button labels for one column of 12 buttons at a time. A dotted line separates the left column from the right column. When you view the labels and icons for the left column, the icons for any active or selected right column features display to the right of the dotted line. To view the column not currently displayed, press the Alternate Display button. You can alternately press any Call Appearance/Feature button on the column not currently displayed. Doing so displays that column and selects the line/feature associated with the button you pressed.

Each IP Office DS module, including control units with integrated DS ports, supports a maximum of eight EU24 or eight EU24BL or two XM24 or two 4450 units only. If using an EU24 with only IP telephones the IP Office will support up to 8 EU24 modules per IP Office switch.

- 24 Programmable call appearance/feature keys.
- Automatically labeled from the system (no paper labels).
- Connects directly to the associated phone.
- Use the EU24 with these Avaya telephones:
    - 2420 and 5420 digital phones.
    - 4620SW, 4621BL and 5620 IP phones.
- Requires a 1151B1 or 1151B2 power supply unit for the phone including for IP phones using Power over Ethernet (PoE).

4606 IP Hardphone

This unit is no longer available from Avaya but is still supported by IP Office 3.1 software. This section is included for reference for existing units.

The 4606 supports the following features:

- 6 Programmable call appearance/feature keys with twin lamps.
- 8 Fixed Feature Keys: Speaker, Mute, Hold, Volume Up & Down, Conference, Transfer, Redial.
- 2 x 16 Character Display.
- Message Waiting Indicator.
- Full Duplex Speakerphone with Echo Cancellation.
- G.711, G.722, G.723.1a, G.729a/B Voice CODECs.
- QoS Options Of - UDP Port Selection, DiffServ And 802.1p/B.
- 10/100 BaseT Ethernet Connection to the IP Office.
- Optional Integrated Ethernet Repeater Hub – for pass through connection of a PC via the phone.
- Hearing Aid compatible.
- IP Address Assignment - DHCP client or statically configured.
- Infrared Port To Support Future Applications.
- Downloadable Firmware for future upgrades.
- Wall Mountable with a separate orderable stand.
- Connects to IP Office via the LAN.

4620 IP Telephone

This unit is no longer available from Avaya but is still supported by IP Office 3.1 software. This section is included for reference for existing units.

The 4620 supports the following:

- 24 Programmable call appearance/feature keys (arranged in 2 switchable display pages of 12 matching the 12 physical display buttons).
- Automatically labeled from the system (no paper labels).
- 11 Fixed Feature Keys: Speaker, Mute, Hold, Headset and Volume Up/Down, Conference, Transfer, Hold, Redial and Drop.
- Large graphical gray-scale display (168 x 132 pixels).
- 4 Embedded applications: Speed Dial, Call Log, Web Browser (WAP/WML), Options.
- Two-way handsfree speaker and microphone.
- Socket for use with the EU24 expansion module.
- 7 Position adjustable desk stand/wall mount stand.
- Infrared (IrDA) port.
- Built-in headset jack.
- Multiple language support: English, French, Italian, Japanese (Katakana), Spanish, German, Dutch and Portuguese.
- 8 Personalized ring patterns.
- Connects to IP Office via the LAN.
- Second full duplex 10/100 BaseT Ethernet Switched ports for PC pass through connection
    - Auto-negotiation provided separately for each port.
    - 802.3 Flow Control.
    - Phone has priority over PC port at all times.

4612 IP Hardphone

This unit is no longer available from Avaya but is still supported by IP Office 3.1 software. This section is included for reference for existing units.

The 4612 supports all of the features of the 4606 with the following differences:

- 12 Programmable call appearance/feature keys with twin lamps.
- 4 Display Navigation Keys, right of the display: Menu, Previous, Next & Exit.
- 4 Display Soft Keys below the Display.
- 8 Fixed Feature Keys: Speaker, Mute, Conference, Transfer, Redial, Hold, Volume Up & Down.
- 2 x 24 Character Display.
- Connects to IP Office via the LAN.

4624 IP Hardphone

This unit is no longer available from Avaya but is still supported by IP Office 3.1 software. This section is included for reference for existing units.

The 4624 supports all of the features of the 4606 with the following differences;

- 24 Programmable call appearance/feature keys with twin lamps.
- 8 Fixed Feature Keys: Speaker, Mute, Conference, Transfer, Redial,Hold, Volume Up & Down.
- 4 Display Soft Keys below the display.
- 4 Display Navigation Keys, right of the display: Menu, Previous, Next & Exit.
- Connects to IP Office via the LAN.

TransTalk 9040 Wireless Telephone

This unit is no longer available from Avaya but is still supported by IP Office 3.1 software. This section is included for reference for existing units.

The 9040 supports the following features:

- 2x16 character LCD display.
- Intuitive Keys for driving the display.
- 10 feature Keys.
- Vibrating ringer option.
- Fixed Redial button.
- Headset connection.
- Belt clip.
- 3.5 hours talk time and 22 hours stand by time.
- Lightweight, weighing less than 8oz.
- Pocket size (dimension 6" x 2" x 1").
- A desktop charger.
- Headset option.
- Base station connects to an IP Office DS (Digital Station) port.

Note: The 9040 requires the DRM-D (Dual Radio Module for TDL/DCP) for connectivity to DS ports. One radio module can support two handsets in a common area. Site survey highly recommended.

Glossary

A

ANI: Automatic Number Identification (ANI). See CLIP

Assisted Transfer: A call transferred from voicemail, which if it returns again to voicemail, will return to the previous position.

B

BACP: Bandwidth Allocation Control Protocol (BACP) is a protocol specification for PPP that allows Multilink PPP routers to negotiate extra bandwidth dynamically over time. Using BACP, two routers can dynamically connect extra "B" channels at times of higher load, then can drop the channels when they are no longer needed. BACP is described in RFC2125.

BDC: Backup Domain Controller is a server in a network domain that keeps and uses a copy by a computer without interrupting its current or primary task. For Windows NT Server domains, BDC refers to a computer that receives a copy of the domain's security policy and domain database and authenticates logons.

Blind Transfer: A call transferred without waiting for the transfer destination to answer first.

BOOTP: This protocol was invented when it was expensive to store software or configurations in small hosts (and even more expensive to upgrade them) so when the host was switched on it would ask (broadcast) on the LAN for its software. A machine with a disk would reply and send the software. Typically the BOOTP Server would send a file to the host using Trivial File Transfer Protocol (TFTP). The main unit uses BOOTP to obtain new versions of its operational software (which it stores in its flash memory). The Manager program acts as the BOOTP server. The BOOTP server recognizes the main unit by its MAC address, this is a hardware address built into the unit at manufacture. This information is obtained from a BOOTP entry which must also include the unit's IP Address and name of the software file to be sent. BOOTP entries are created automatically and stored in the PC's registry.

C

Callflow: A general term for a sequence of actions used to determine what facilities are offered to a caller.

CAPI: Common Application Programming Interface.

CHAP: Challenge Handshake Authentication Protocol (CHAP). An authentication scheme used by PPP servers to validate the identity of the originator of a connection, upon connection or any time later.

CLI: Calling Line ID. Information passed from the telephone network exchange to the IP Office. Also called ICLID and CLID.

CLID: Calling Line ID. See CLI.

CLIP: Calling Line Identity Presentation. Displays the calling party's number to the called party. Variations include withholding CLI and displaying alternative presentation numbers. ANI (automatic Number Identification) is the USA equivalent.

CLIR: Calling Line Identification Restriction (CLIR) Inhibits the telephone number of the IP Office being presented on an outbound call.

COLP: Connected Line Identity Presentation (COLP). Displays the connected party's number to the calling party. Useful where the call has been diverted away from the originally dialed party.

COLR: Connected Line Identification Restriction (COLR) Inhibits the COLP service.

CSU: Channel Service Unit: Used to terminate an incoming digital trunk at the customer premises. Incorporates features to allow trunk testing and checking, including loop-back functions.

CTI: Computer Telephony Integration, a technology that acts as an electronic bridge connecting telephones or switches with computers. CTI controls or coordinates business processes and related applications through the exchange of commands and messages between computers and telephone systems.

D

DDI(DID)/MSN: Direct Dial In (DDI/DID) and Multiple Subscriber Numbering (MSN) are telephone company services that can be subscribed to. Call destinations can therefore be passed down the ISDN line and the system can use this information to deliver the calls to their final destination, perhaps individuals or departments.

DHCP: Dynamic Host Configuration Protocol, a standards-based protocol for dynamically allocating and managing IP addresses. DHCP runs between individual computers and a DHCP server to allocate and assign IP addresses to the computers and also limits the time computers can use the address. When time expires on the use of the IP address, the computers contact the DHCP server again to obtain an address.

DiffServ: DiffServ (RFC 2474) is a TCP/IP quality of Service mechanism used to ensure that IP packets are prioritized according to their importance, for example prioritization of voice packets over data packets. Prioritization is based upon the Type of Service (ToS) field in the IP header.

Digital Stations: Refers to Avaya telephones in the 2400, 4400, 5400 and 6400 series. Supported by DS sockets on IP Office control units and Digit Station modules. Note: Not all terminals in the above ranges are supported on IP Office.

Dn: Directory number.

DNIS: Dialed Number Identification Service (DNIS). Available in US markets. DNIS identifies to the called party the dialed number. Can be used to identify the purpose of inbound calls.

Domain: The part of the computer network in which the data processing resources are under common control.

DSS: Direct Station Select - A DSS key can be programmed with a number or feature code.

DSU: Data Service Unit: Normally incorporated within the CSU of digital trunk connections. The DSU allows the trunk to be shared between data and voice services.

E

Embedded Voicemail: A voicemail system stored on a memory card inserted into the IP Office telephone system's control unit.

ESP: Encapsulation Security Payload: A standard (RFC2406) that forms part of IPSec.

F

Frame Relay: Connections to private or public Frame Relay services, such as BT FrameStream, can be made via the WAN port on the rear of main unit, or the WAN port of an associated WAN 3 module. Both data and Voice over IP (requires the use of the Voice Compression Module) are supported across Frame Relay.

G

G.711 A-Law 64K: A VoIP compression mode. Each voice call is converted from analog to digital (refer to G.723) and uncompressed.

G.723.1 6K3 MP-MLQ: A VoIP compression mode. A real-time implementation of the ITU-T Multi-Pulse Maximum Likelihood Quantization (MP-MLQ) 6.4 Kbps and Algebraic Codebook Excited Linear Prediction (ACELP) 5.3 Kbps speech coding algorithms. The G.723.1 speech coder operates upon 30 ms frame of digitized, telephone bandwidth speech signals sampled at 8 kHz. The frames are divided into four 7.5 milli-second sub frames of 60 samples each. Each frame of 240 input samples is converted into 12 16-bit word of compressed data at the high rate or 10 16-bit words of compressed data at the low rate. The Voice Activity Detection/Comfort Noise Generation (VAD/CNG) specified in Annex A to ITU-T G.723.1 is fully implemented,and may be used to further reduce the average bit rate.

G.726 ADPCM 16K/32K: A VoIP compression mode. Each voice call is compressed using the standard ADPCM compression technique (refer to G.732). This algorithm uses 16,000 or 32,000 bits per second.

G.729(a) 8K CS-ACELP: A VoIP compression mode. A fully compliant, real-time implementation of the ITU-T fixed-point conjugate-structure, algebraic code-excited linear prediction (CS-ACELP) speech coding algorithm. The CS-ACELP operates at 8Kkbps. The coder processes 10 millisecond frames of speech sampled at an 8 kHz rate, which together with a 5 millisecond look-ahead results in a total algorithmic delay of 15 milliseconds. For each frame of 80 samples of 16-bit linear PCM data, the coder outputs five 16-bit words. Applications using the G.729 vocoder include digital telephony, satellite and wireless communications.

Gatekeeper: An H.323 entity that provides address translation, controls access, and sometimes bandwidth management to the LAN for H.323 terminals, Gateways, and Multipoint Control Units. IP Office units can register themselves with multiple external H.323 gatekeepers.

GUI: Graphical User Interface.

H

H.323 VoIP: Allows voice and data traffic to be networked between systems. Connections between platforms across the WAN, at speeds up to 2.048Mbps (in conjunction with the Voice Compression Module), or across the LAN at 10 or 100 Mbps. Multiple WAN links maybe supported utilizing the optional WAN3 modules. Also allows telephone calls to be made from PCs running Microsoft's NetMeeting when fitted with a sound card, speakers and microphone. Calls can be made between PCs or to standard analog or digital telephones. Please note that at this point in time, we do not consider NetMeeting to offer a Toll Quality voice service. The addition of the IP Telephony Extensions to the H.323 Gateway protocol allows physical H.323compliant IP "Hardphones" and PC based, IP "Softphone" applications to make and receive phone calls.

H.450: VoIP Supplementary Services H.450 provides extended features within H.323 based VoIP networks similar in concept to QSig within ISDN.

HTML: Hyper Text Markup Language, the authoring language used to create hypertext documents for the World Wide Web.

HTTP: Hyper Text Transfer Protocol, the application protocol for moving hypertext files across the Internet. The protocol requires an HTTP client program on one end of a connection and an HTTP server program on the other.

I

iChat: iChat is a service that forwards chat requests to the chat queues. iContact converts the PC to an all-in-one communications and data tool by allowing enterprise knowledge workers to prioritize and manage all interactions from one interface. An agent can see queued Telephone calls, Emails, Web calls, chats, and can communicate with group members from one centralized view.

ICLID: Incoming Caller ID. See CLI.

iEmail: iEmail is a service that is responsible for forwarding incoming E-mail messages to the E-mail queue or to the agent. iEmail also forwards web callback requests to the web callback queue.

IKE: Internet Key Exchange: A standard (RFC2409) that forms part of IPSec operation.

IMAP: Internet Mail Access Protocol: An essential Internet protocol for E-mail communication. IMAP4, which is both a client and server protocol, can enable voice and fax message access and storage through a PC interface. IMAP4 also complements SMTP for retrieval/access of messages.

IP: The Internet Protocol (IP) is the method or protocol by which data is sent from one computer to another on the Internet. Each computer (known as a host) on the Internet has at least one IP address that uniquely identifies it from all other computers on the Internet. When you send or receive data (for example, an email note or a Webpage), the message gets divided into little chunks called packets. Each of these packets contains both the sender's Internet address and the receiver's address. Any packet is sent first to a gateway computer that understands a small part of the Internet. The gateway computer (or router) reads the destination address and forwards the packet to an adjacent gateway that in turn reads the destination address and so forth across the Internet until one gateway recognizes the packet as belonging to a computer within its immediate neighborhood or domain. That gateway then forwards the packet directly to the computer whose address is specified. Because a message is divided into a number of packets, each packet can, if necessary, be sent by a different route across the Internet. Packets can arrive in a different order than the order they were sent in. The Internet Protocol just delivers them. It's up to another protocol, typically TCP, to put them back in the right order. IP is a connectionless protocol, which means that there is no established connection between the end points that are communicating. Each packet that travels through the Internet is treated as an independent unit of data without any relation to any other unit of data. (The reason the packets do get put in the right order is because of TCP, the connection-oriented protocol that keeps track of the packet sequence in a message.) In the Open Systems Interconnection (OSI) communication model, IP is in layer 3, the Networking Layer.

iPhone: iPhone is a service that applies telephony rules.

IPSec: IP Security: A set of methods and standards (starting with RFC2401) for the secure (authenticated and/or encrypted) routing of private network traffic across the Internet.

ISAKMP: Internet Security Association and Key Management Protocol: A standard (RFC2408) for the bodies and processes that keys used by IPSec.

iServer: iServer consists of two parts. One is WT service, and the other is a combination of different server components, that run on the Microsoft transaction server.

ISP: Internet Service Provider. A business that supplies Internet connectivity services to individuals, businesses and other organizations.

L

L2TP: Layer Two Tunneling Protocol: A standard (RFC2661 and RFC3193) for the connections of private network connections across the Internet.

LAN: Local Area Network.

LCP: In the Point-to-Point Protocol, the Link Control Protocol (LCP) establishes, configures and tests data-link Internet connections. Before establishing communications over a point-to-point link, each end of the PPP link must send out LCP packets. The LCP packet either accepts or rejects the identity of its linked peer, agrees upon packet size limits, and looks for common mis-configuration errors. Basically, the LCP packet checks the telephone line connection to see whether the connection is good enough to sustain data transmission at the intended rate. Once the LCP packet accepts the link, traffic can be transported on the network; if the LCP packet determines the link is not functioning properly, it terminates the link. LCP packets are divided into three classes: 1. Link configuration packets used to establish and configure a link. 2. Link termination packets used to terminate a link. 3. Link maintenance packets used to manage and debug a link.

LDAP: Lightweight Directory Access Protocol, a protocol used to access a directory listing. LDAP support is being implemented in Web-enabled and Email programs, which can query an LDAP-compliant directory. LDAP has become the Internet standard for directory infrastructure and is expected to provide a common method for searching Email addresses on the Internet.

M

MAC address: The address of a device identified at the media access control (MAC) layer of the network architecture.

MAPI: Messaging Application Programming Interface - Part of Microsoft's Window's Open Service Architecture (WOSA). Allows programs and devices to send emails via email clients if those clients support MAPI.

ML-PPP: Multilink PPP (ML-PPP) is a standard, based on the original PPP standard, that allows a router to open a number of different connections to a remote router. ML-PPP defines a way to divide up the data and send it down multiple paths in such a way that the remote router can put the pieces back in the original order on reception. The main justification for ML-PPP is bandwidth allocation (sometimes known as Bundling or Bonding). The application only sees one "logical link" giving a bandwidth of (say)256Kbps, even though there are actually four "B" channels connected between the two sites. This is achieved by adding an additional data header on each packet sent. For example, if a router has an ISDN BRI interface, it could transfer data at 64Kbps on one "B" channel, but then in times of higher load could connect extra "B"channels and so have an aggregate rate of 128 Kbps and above. There is a new standard for the PPP protocol called BAP (Bandwidth Allocation Protocol), which enhances the ML-PPP specification by making sure that all vendors implement the same rules for when extra channels are connected, and when they are disconnected.

N

NAT: Network Address Translation is a mechanism that allows you to hide internal IP addresses from external networks. You may have an established network using your own numbering scheme, and would like to access the Internet. There are many cost effective Internet Service Providers (ISP) but they want you to use a different IP address. By using NAT between your machine and their network everyone is satisfied, without any need to renumber your network. An additional benefit is that all your machines can use the NAT facility and access the Internet via the one address. NAT is the translation of an IP address within one network to a different IP address known within another network. One network is designated the inside network and the other is the outside. Typically, a company maps its local inside network addresses to one (or more) global outside IP address and unmaps the global IP address on incoming packets back into local IP addresses. This helps ensure security since each outgoing or incoming request must go through a translation process that also offers the opportunity to qualify or authenticate the request or match it to a previous request. NAT also conserves on the number of global IP addresses that a company needs and it lets the company use a single IP address in its communication with the world.

NU: Number Unobtainable.

P

PAP: Password Authentication Password is a method for verifying the identity of a user attempting to log on to a PPP server. PAP is used if the password is to be sent without encryption.

PDC: Primary Domain Controller. For a Windows NT Server domain, the computer that authenticates domain logons and maintains the security policy and the master database for a domain.

PDF: Portable Document Format. The file format used for Adobe Acrobat files.

PPP: Point-to-Point Protocol. This is a Protocol for communication between two computers using a Serial interface, typically a personal computer connected by phone line to a server. For example, your Internet service provider may provide you with a PPP connection so that the provider's server can respond to your requests, pass them on to the Internet, and forward your requested Internet responses back to you. PPP uses the Internet protocol (IP), and is designed to handle others). It is sometimes considered a member of the TCP/IP suite of protocols. Relative to the Open Systems Interconnection (OSI) reference model, PPP provides layer 2 (data-link layer) service. Essentially, it packages your computer's TCP/IP packets and forwards them to the server where they can actually be put on the Internet. PPP is a Full Duplex protocol that can be used on various physical media, including twisted pair or fiber optic lines or satellite transmission. It uses a variation of High Speed Data Link Control (HDLC) for packet encapsulation. PPP is usually preferred over the earlier de facto standard Serial Line Internet Protocol (SLIP) because it can handle Synchronous as well as Asynchronous communication. PPP can share a line with other users and it has error detection that SLIP lacks. Where a choice is possible, PPP is preferred.

PPTP: Point-to-Point Tunneling Protocol. This is a Protocol (set of communication rules) that allows corporations to extend their own corporate network through private "tunnels" over the public Internet. Effectively, a corporation uses a wide-area network as a single large local area network. A company no longer needs to lease its own lines for wide-area communication but can securely use the public networks. This kind of interconnection is known as a virtual private network (VPN).

Presumed User: Some actions presume who the user associated with a call is from factors such as the original target extension or mailbox of the call. This allows those action to be used in modules without having to specify the mailbox on which they should act.

Proactive List Manager: The Proactive List Manager module facilitates the importing and assignment of outbound calling lists to Proactive Campaigns. It provides the administrator with the ability to manage outbound Proactive Campaign Lists. It furnishes the tools to create draft calling lists, attach them to campaigns and run the campaigns.

R

Reporting: The browser-based Reporting module provides complete enterprise management reporting through textual and graphical reports. These reports provide enterprise managers with a record of every step in the customer interaction process, and allow them to view and analyze how effectively interactions are being handled and how resources are being deployed. The reports can also provide a better understanding of how their operation and performance affects your networks, resources and people.

Resource Manager: The Resource Manager administration module consists of components that enable you to add queues, define interaction results, and assign human resources to all from a single, unified console. Resource Manager has a user-friendly Microsoft Explorer look and feel interface.

RSVP: RSVP (Resource Reservation Protocol) is a protocol that allows channels or paths on the Internet to be reserved for the multicast (one source to many receivers) transmission of video and other high-bandwidth messages. RSVP is part of the Internet Integrated Service (IIS) model, which ensures: best-effort service, real-time service, and controlled link-sharing. The basic routing philosophy on the Internet is "best-effort," which serves most users well enough but isn't adequate for the continuous stream transmission required for video and audio programs over the Internet. With RSVP, people who want to receive a particular Internet "program" (think of a television program broadcast over the Internet) can reserve bandwidth through the Internet in advance of the program and be able to receive it at a higher data rate and in a more dependable data flow than usual. When the program starts, it will be multicast to those specific users who have reserved routing priority in advance. RSVP also supports unicast (one source to one destination) and multi-source to one destination transmissions.

S

SNMP: Simple Network Management Protocol: A method of communication between a network monitoring agent and a network management application to provide information regarding its operational status.

SQL: Structured Query Language is a database language used for creating, maintaining and viewing database data.

Standard Voicemail: Also called Voicemail Lite. Provides basic voicemail operation for the telephone system. The Voicemail Pro Server contains all the same functions as Voicemail Lite.

T

TAPI: Telephony Application Program Interface.

TCP: Transmission Control Protocol (TCP) is a method protocol used along with the Internet Protocol (IP) to send data in the form of message units between computers over the Internet. While IP takes care of handling the actual delivery of the data, TCP takes care of keeping track of the individual units of data (called packets) that a message is divided into for efficient routing through the Internet. For example, when an HTML file is sent to you from a Web server, the Transmission Control Protocol (TCP) program layer in that server divides the file into one or more packets, numbers the packets, and then forwards them individually to the IP program layer. Although each packet has the same destination IP address, it may get routed differently through the network. At the other end (the client program in your computer), TCP reassembles the individual packets and waits until they have arrived to forward them to you as a single file. TCP is known as a connection-oriented protocol, which means that a connection is established and maintained until such time as the message or messages to be exchanged by the application programs at each end have been exchanged. TCP is responsible for ensuring that a message is divided into the packets that IP manages and for reassembling the packets back into the complete message at the other end. In the Open Systems Interconnection (OSI) communication model, TCP is in layer 4, the Transport Layer.

TCP/IP: Transmission Control Protocol/Internet Protocol is a networking protocol that provides communication across interconnected networks, between computers with diverse hardware architecture and various operating systems.

TFTP: Trivial File Transfer Protocol: A standard protocol (RFC1350) used to send and receive files. Used by IP Office applications and devices to exchange information.

Trusted Location: This is a location from which the System will allow data access, e.g. a user dialing in from home, or access to Voicemail without a Voicemail Code e.g. a user collecting his Voicemail messages from a mobile, or the location the Voicemail Server will call to inform the user of a new message.

U

UDP: User Datagram Protocol is a protocol that can be used as an alternative to TCP for IP packet transfer. UDP differs from TCP in that it does not open connections before it sends data and does not number or sequence its datagrams (packets) in any way. Packets can therefore arrive out of sequence, get lost, get duplicated and succesful packets are not acknowledged. UDP is used for those applications where the rapid real-time send of packets is required without the administrative burden of TCP, for example VoIP.

URL: Universal Resource Locator is an address that can lead you to a file on any computer connected to the Internet.

V

V.110/V.120: V.110 and V.120 are ITU Protocol standards which support the transport of an RS232(V.24/V.28) interface and asynchronous characters across a link. Thus simple terminals of between 50bps to 19.2Kbps can be connected to the TA RS232/V.24 port and communicate over a 'B' channel. V.120 offers enhancements over V.110 in that it uses a LAPD-like protocol on the "B" channel so it is possible to support a number of multiplexed low-speed devices over one channel i.e. V.120 makes better use of the bandwidth.

Voice Compression Module: Support for the optional Voice Compression Module allows voice calls to be networked between Systems when WAN links are used. Five compression algorithms are supported from 64kbp to 6.3kbps, while the Voice Compression Module also provides echo cancellation where voice calls between systems are then broken out on to the public network. Support is provided for the 5, 10 and 20 channel variants of the Voice Compression Module.

VoIP: Voice over Internet Protocol (VoIP). The technology used to transmit voice conversations over a data network using the Internet Protocol.

VPIM: Voice Profile for Internet Messaging. Allows different voice messaging systems to exchange voicemail over the internet.

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