IP Office 2.1
Product Description

11th May 2004 - Issue 11b

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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 2.1

For those already familiar with IP Office, this page lists the new features introduced in IP Office 2.1. This is not a exhaustive list, it covers just the major changes.

IP Office Core 2.1 Software

- Validated software upgrade option - allows remote upgrades.
- Full merge capability for all user and hunt group settings.
- IP403 can now support all trunk types (analog, BRI, PRI) in both slots.
- Integral CSU on T1 trunk modules.
- T1 Trunk module for the Small Office Edition.
- Wizard enhancements including "Moves, Adds and Changes" wizard.
- PIN (account code) restricted dialing based on number dialed.
- Paging to IP Phones.
- VPN support using IPSEC and L2TP.
- Support for 3810 wireless phone (North America only).
- Fax over IP interworking with ACM.

Voicemail Pro 2.1

- Voicemail Pro Networked Messaging (replaces VPIM).
- Fax tone detection at the mailbox with system and/or personal fax number.
- CTI Call Data Tagging

IP Office Conferencing Center

- Allows password controlled users to book conferences through a browser interface. Generated unique conference IDs and participant PIN numbers.
- Conference participants are notified by email or (using Voicemail Pro) called.
- Conferences can include a web session where conference host can display web server based documents and hold voting sessions.
- Listen only mode allows host to mute other participants.
- Caller name announced when joining conference.

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.

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**
  IP403 and IP406 support an eight port HUB that connects up to 8 PCs and/or supported IP devices. The Avaya IP Office – Small Office Edition offers 4 Switched Ethernet ports (Layer 2) and a dedicated Ethernet WAN port (Layer 3). The IP412 offers 2 switched Ethernet ports (Layer 3).
- **LDAP client support**
  For standards based directory synchronization.

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 PC Soft Phone, 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 and Avaya IP Office – Small Office Edition PCMCIA 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!6, 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, Voice Mail and other applications) are 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 Terminal Adaptor, Router or the optional Modem 2 package.

An IP Office Wizard can also be used to configure systems and to manage user adds, moves 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.

- **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:-
    - 2 Analog trunks, 4 analog extensions and 3 VoIP resources
    - 4 Analog trunks, 8 analog extensions and 3 VoIP resources
    - 4 Analog trunks, 4 analog extensions, 8 digital extensions***** and 3 VoIP resources
    - 4 Analog trunks, 4 analog extensions, 8 digital extensions***** and 16 VoIP resources
        - *Available as either Digital Station (64xx/44xx/24xx series handsets) or Digital Terminal (20xx series handsets). Digital Terminal versions are not available in all territories. Check with your Avaya representative for local availability.
- **IP401 Compact Office**
  The IP401 Compact Office is available in two versions (Not available in all territories). **IP401 Compact Office - Digital Terminal 2** supports two Digital Terminals, Two Analog Terminals, a single Basic Rate ISDN, 4 port dual speed LAN hub and 2 data channels. Data channels are used for Routing, RAS and Voicemail applications. **IP401 Compact Office - Digital Terminal 4** supports four digital Terminals, four Analog Terminals, two Basic Rate ISDN, 8 port dual speed LAN hub and 4 data channels.
- **IP403 Office**
  Supports 3 Expansion Modules providing a combination of up to 100 analog and digital extensions, with capacity for 8 analog trunks or 2 digital trunk (48 T1 channels or 60 E1 channels). Additional analog trunks can be provisioned using the IP400 Analog trunk 16 module. Features include 20 optional voice compression channels, 8 Digital Terminal/Station ports, 2 Analog Telephone ports, 8 dual speed LAN hub ports, and 18 data channels. Data channels are used for Routing, RAS and Voicemail applications.
- **IP406 Office**
  Supports 6 Expansion Modules providing a combination of up to 180 analog and digital extensions, with capacity for 8 analog trunks or 2 digital trunk (48 T1 channels or 60 E1 channels). Additional analog trunks can be provisioned using the IP400 Analog trunk 16 module. Features include 20 optional voice compression channels, 8 dual speed LAN hub ports, and 24 data channels. Data channels are used for Routing, RAS and Voicemail applications.
- **IP412 Office**
  Supports 12 Expansion Modules providing a combination of up to 360 analog and digital extensions, with capacity for 8 analog trunks or 4 digital trunk (96 T1 channels or 120 E1 channels). Additional analog trunks can be provisioned using the IP400 Analog trunk 16 module. 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.

All IP Office models support common software, telephones, and applications.

Endpoint Solution Options

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

- IP Office with the Integral H.323 Server supports selected Avaya 46xx series of H.323 IP telephones, 36xx Wireless VoIP sets and Phone Manager (which can operate in CTI or IP Soft phone modes)
- IP400 Digital Station 16 or 30 Module supports selected MERLIN MAGIX 44xx series and selected 64xx/24xx digital series sets. The IP Office Digital Station module will also support the TransTalk 9040 wireless handset. (The 4400 series telephones are not available in certain territories, check for local availability).
- The IP Office Digital Station module will support the new Avaya 3810 wireless handset and the existing TransTalk 9040 wireless handset
- IP400 Digital Terminal 16 or 30 Module supports the Avaya 20 Series telephones (not available in certain territories, check for local availability).
- IP400 Phone 8, 16 or 30 Modules support standard analog Phones, faxes and modems, with support for calling line identification.

2. IP Office - Small Office Edition Platform

IP Office - Small Office Edition Overview

IP Office - Small Office Edition supports all 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 available in six variants* which provide a different mix of Analog Trunks, Analog extensions, Digital Extensions and Voice Over IP capacity. Dependant on the model chosen, up to a maximum of 28 extensions can be supported (4 Analog, 8 Digital and 16 IP).

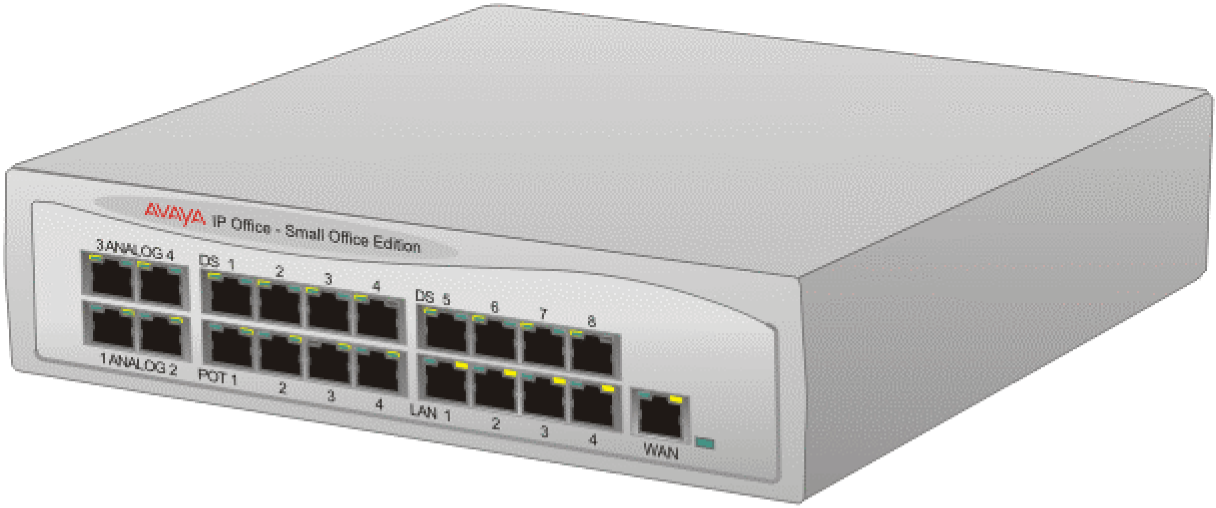
All IP Office - Small Office Edition's have a four port Ethernet Switch (layer 2) and a dedicated switched Ethernet WAN port (layer 3) making the system ideal for connection to broadband services such as ADSL and Cable. With Voice over IP as standard and the optional IPSec security, the system can be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

The IP Office - Small Office Edition supports an additional WAN slot (located at the back) to support other network connections such as V35, X21, T1 and BRI leased lines that may be encountered in frame relay applications.

The back of the unit provides a twin PCMCIA socket for a memory card when using embedded voicemail and a Wireless LAN card when using the system as an Access Point (see below for further detail)

As well as supporting the external license key server for licensed application 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.

- *Note: Not all variants are available in all territories, check for local availability.



The six pre-defined configurations are detailed in the following table.

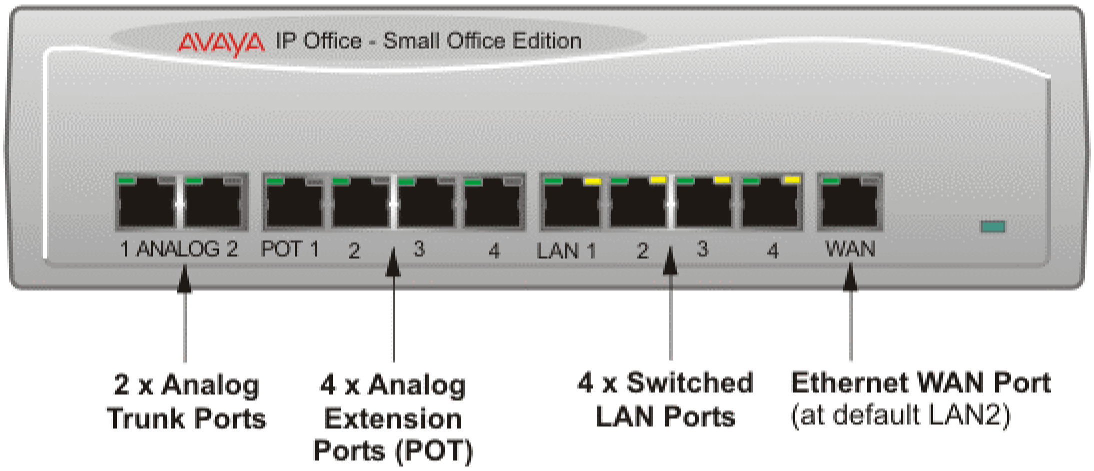
IP OFFICE - SMALL OFFICE EDITION	Analog Trunks	Analog Extensions	Digital Terminal (20 series)	Digital Station (64, 44 & 24 series, 3810 & 9040)	Voice Over IP Channels
2T+4A (3 VoIP)	2	4	0	0	3
4T+8A (3 VoIP)	4	8	0	0	3
4T+4A+8DT (3 VoIP)*	4	4	8	0	3
4T+4A+8DS (3 VoIP)	4	4	0	8	3
4T+4A+8DT (16 VoIP)*	4	4	8	0	16
4T+4A+8DS (16 VoIP)	4	4	0	8	16

- *Not available in all territories, check for availability.
- During power fail, Analog port 2 is connected to POT port 1.

IP Office - Small Office Edition 2T+4A (3 VoIP)

The IP Office - Small Office Edition 2T+4A (3 VoIP) provides:

- Two Analog Loop Start Trunks (Caller ID enabled).
- Four analog extension interfaces.
- 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, X.21, BRI, T1 PRI).
- DTE port.
- Audio port for external music on hold source.
- Two relay switch port for door entry systems (External O/P socket).



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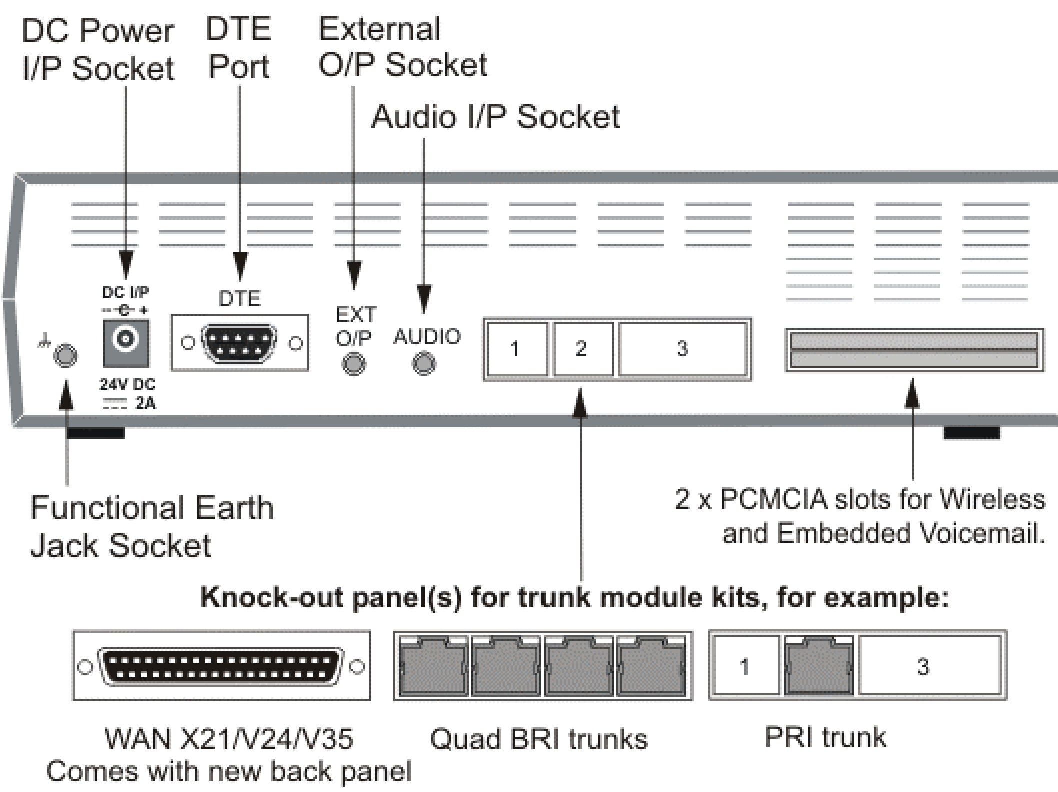

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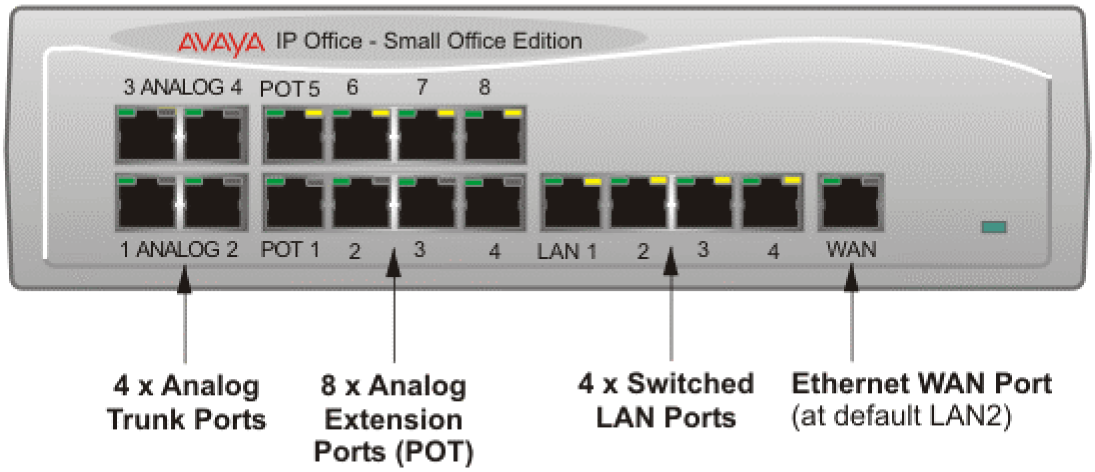
IP Office - Small Office Edition 4T+8A (3 VoIP)

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

- Four Analog Loop Start Trunks (Caller ID enabled).
- Eight analog Extension interfaces.
- 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, X.21, BRI, T1 PRI).
- DTE port.
- Audio port for external music on hold source.
- Two relay switch port for door entry systems (External O/P socket).



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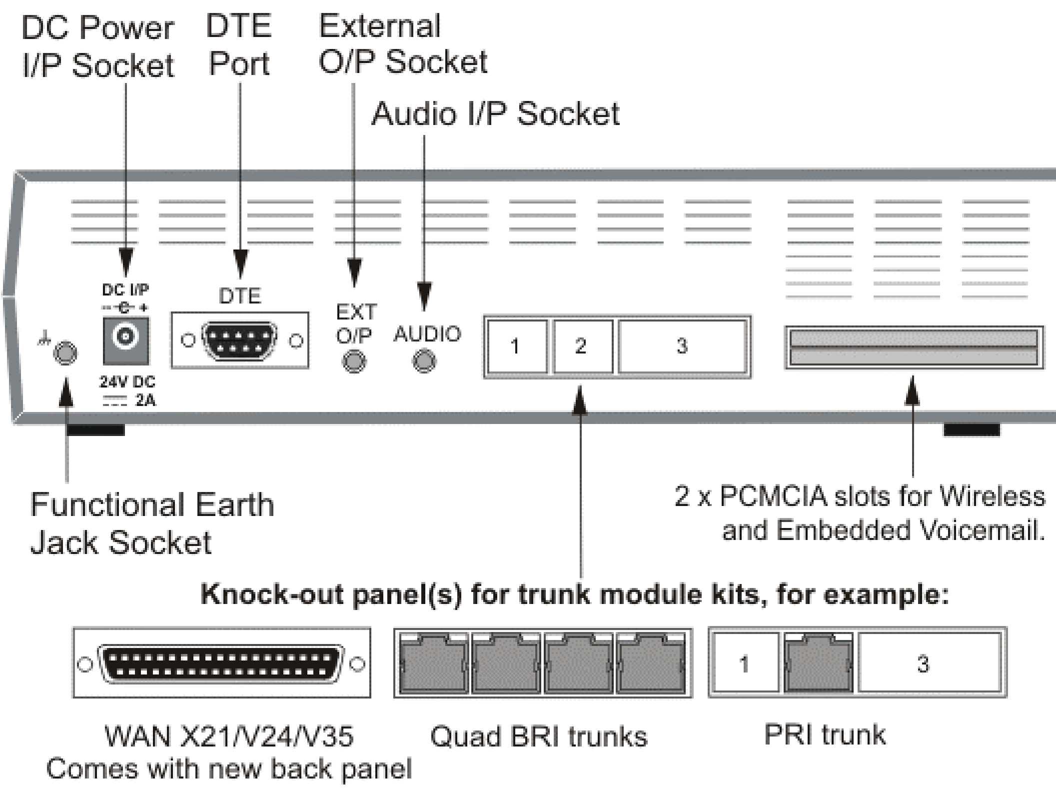

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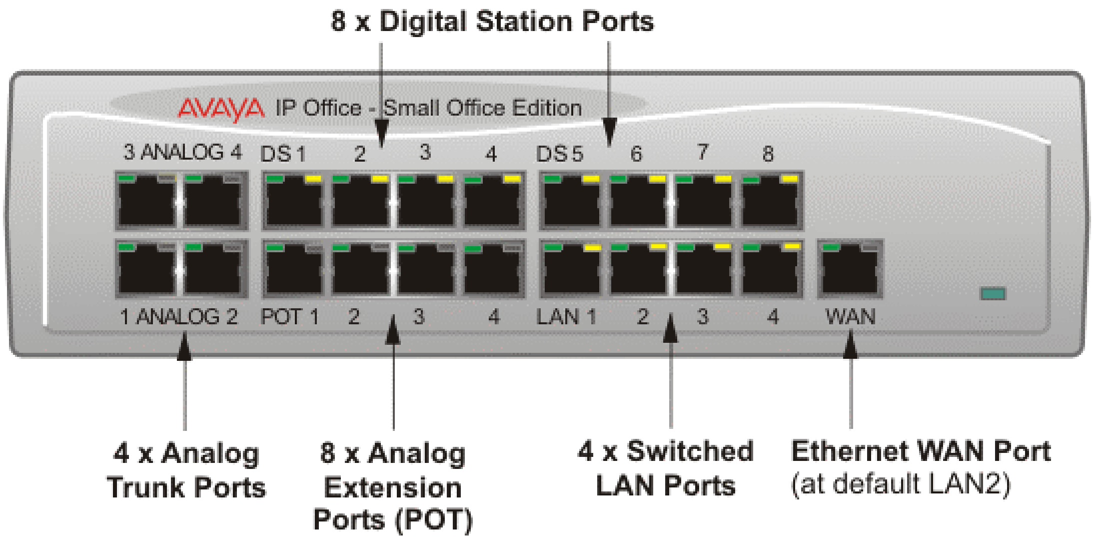
IP Office - Small Office Edition 4T+4A+8DS (3 VoIP)

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

- Four Analog Loop Start Trunks (Caller ID enabled).
- Four analog Extension interfaces.
- Eight Digital Terminals (24xx, 44xx, 64xx Series plus 3810 and 9040).
- 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, X.21, BRI, T1 PRI).
- DTE port.
- Audio port for external music on hold source.
- Two relay switch port for door entry systems (External O/P socket).



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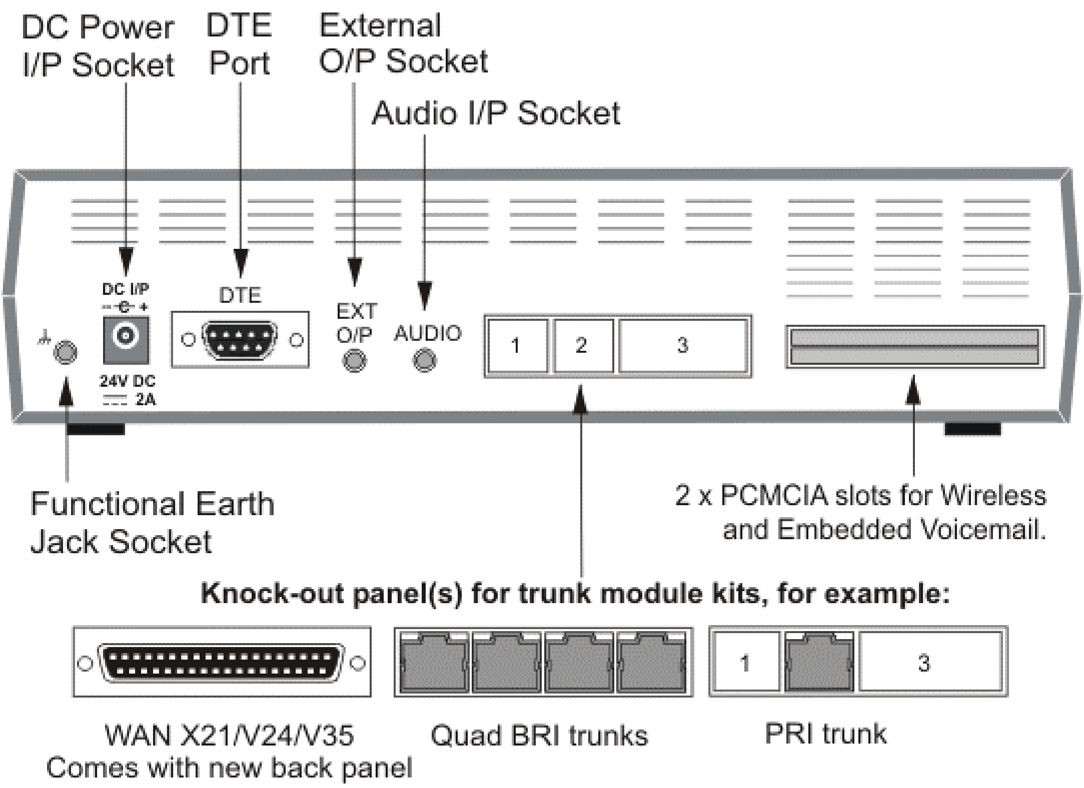


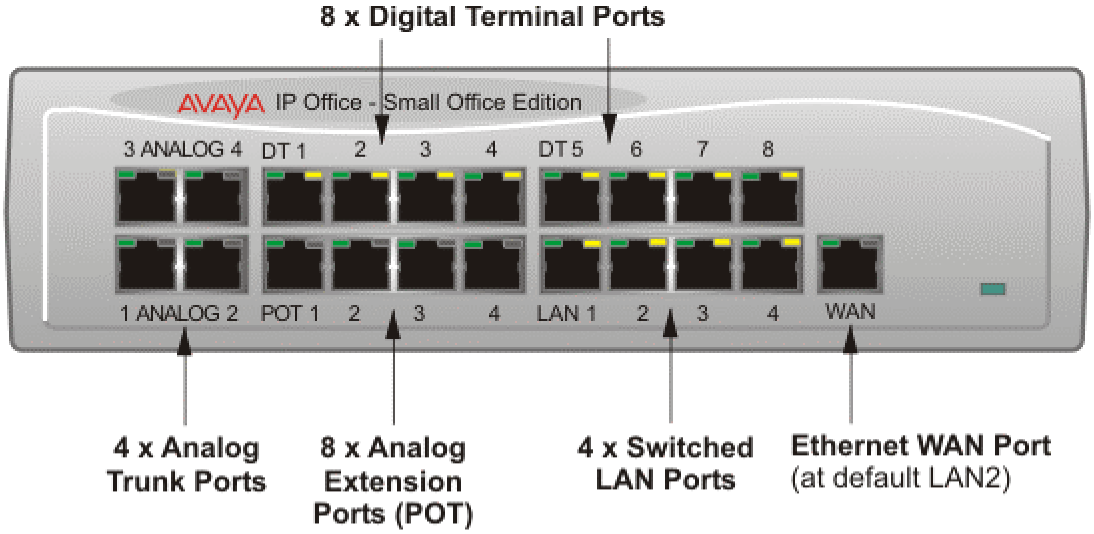
Rear Panel

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

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

- Four Analog Loop Start Trunks (Caller ID enabled).
- Four Analog Extension interfaces.
- Eight Digital Terminals (20 Series).
- 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, X.21, BRI, T1 PRI).
- DTE port.
- Audio port for external music on hold source.
- Two relay switch port for door entry systems (External O/P socket).

Not available in all territories, check for availability.



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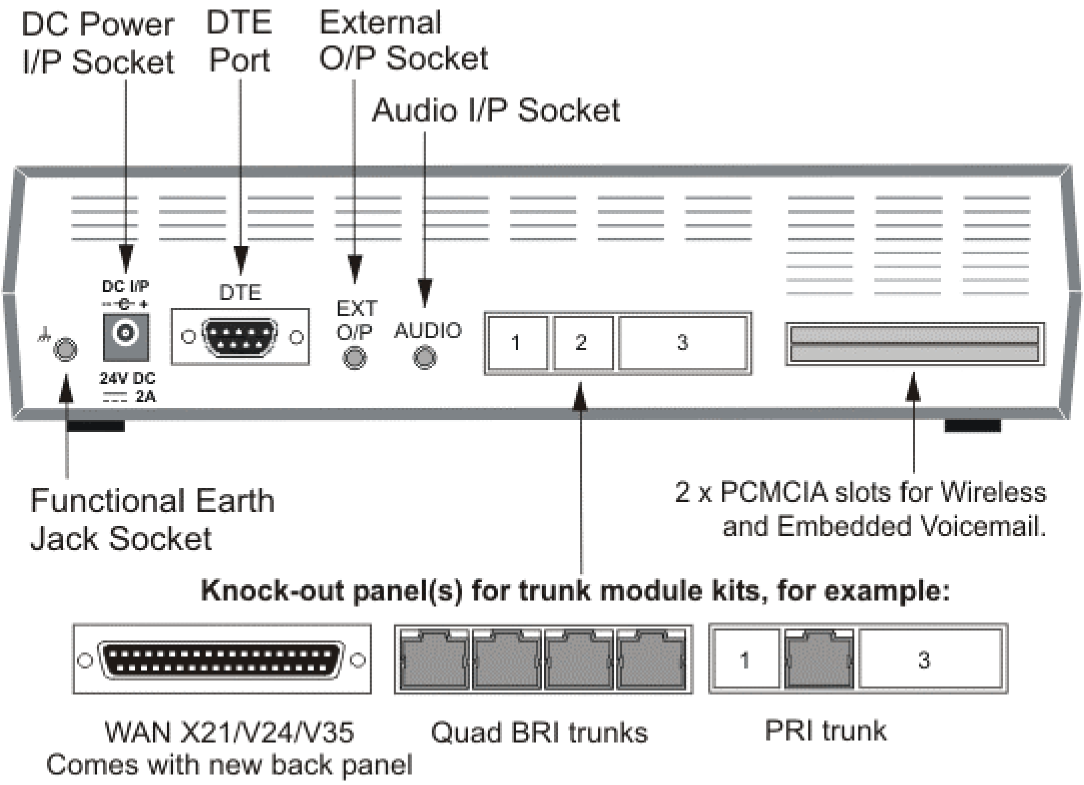

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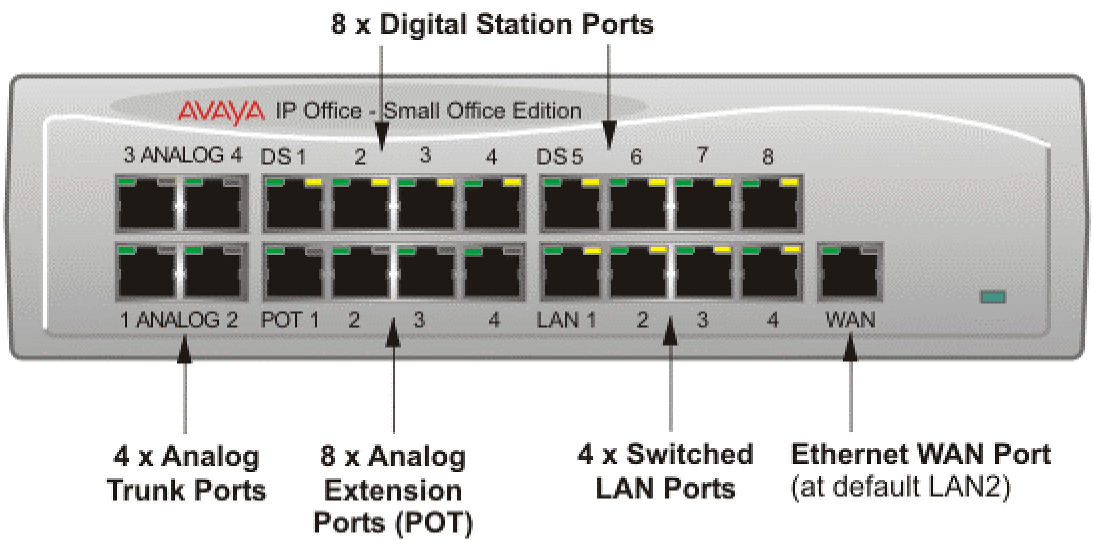
IP Office - Small Office Edition 4T+4A+8DS (16 VoIP)

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



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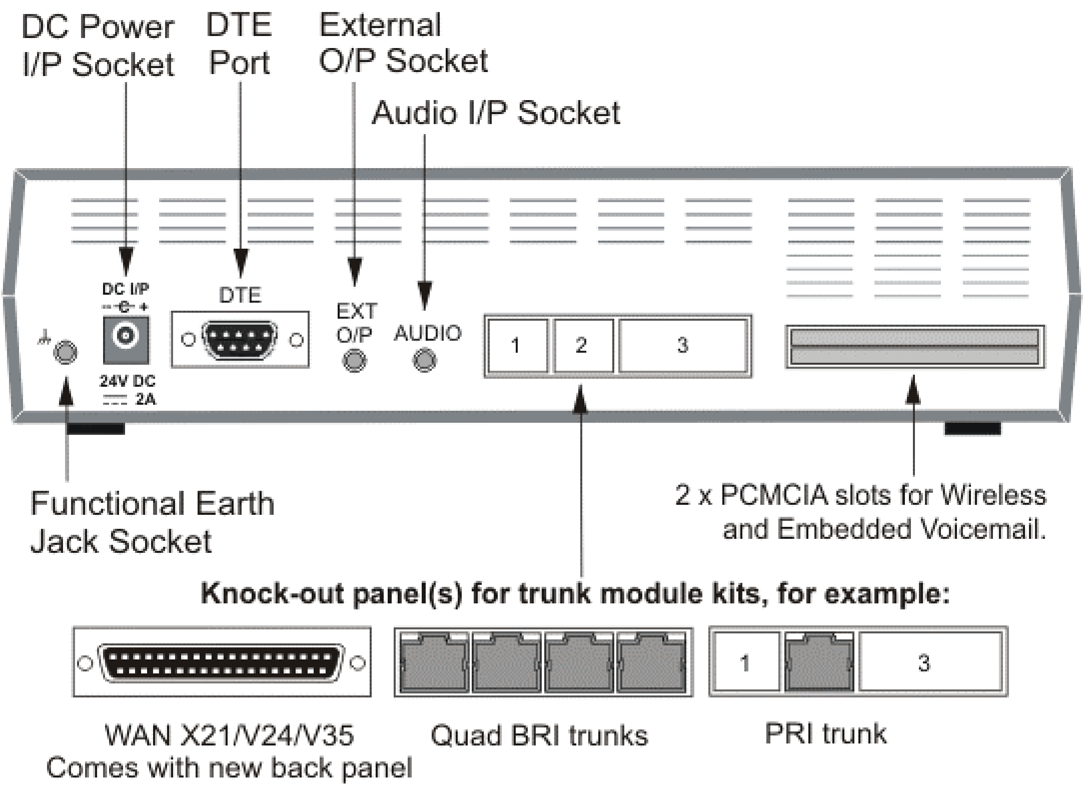

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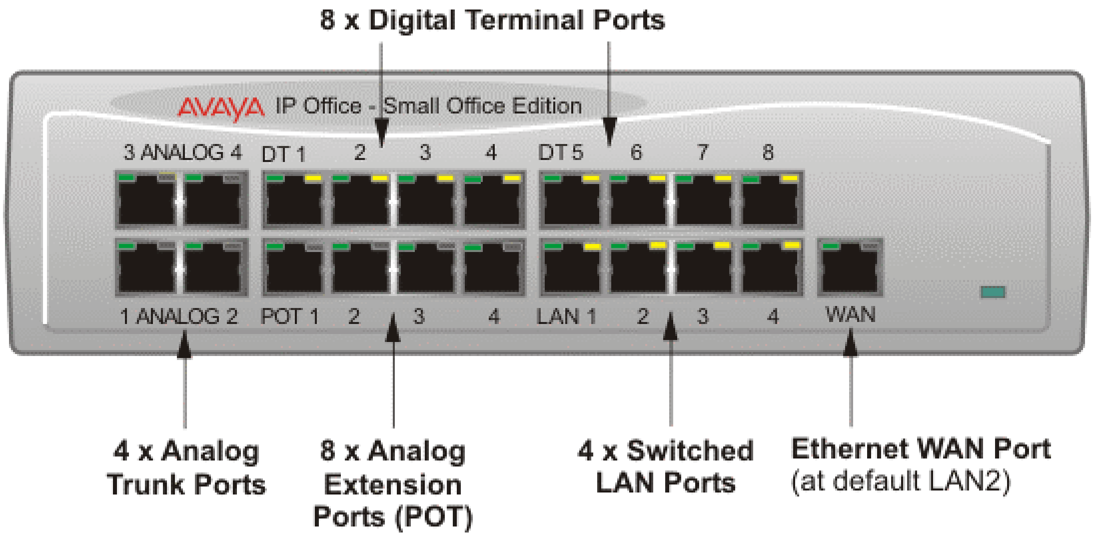
IP Office - Small Office Edition 4T+4A+8DT (16 VoIP)

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

Not available in all territories, check for availability.



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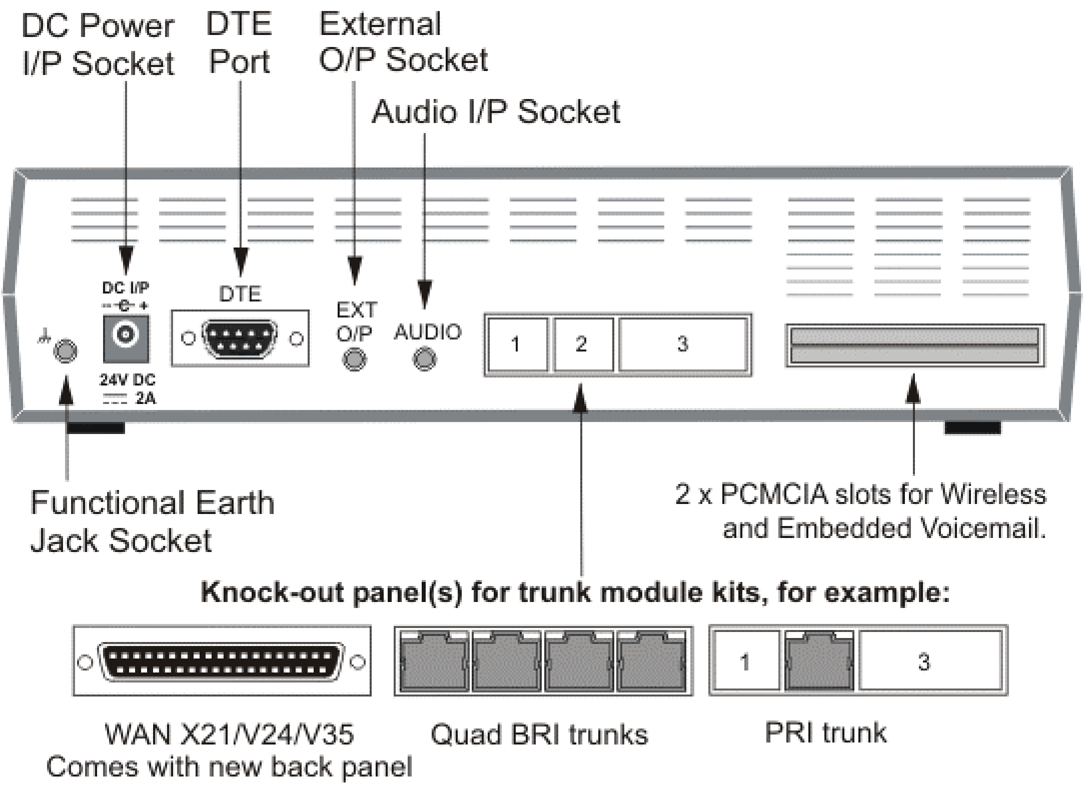

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IP Office - Small Office Edition WAN Expansion Interfaces

All IP Office - Small Office Edition 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.

IP401 WAN Expansion

The IP401 WAN Expansion card provides a single WAN connection (X21, V35 or V24 via a 37way 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.

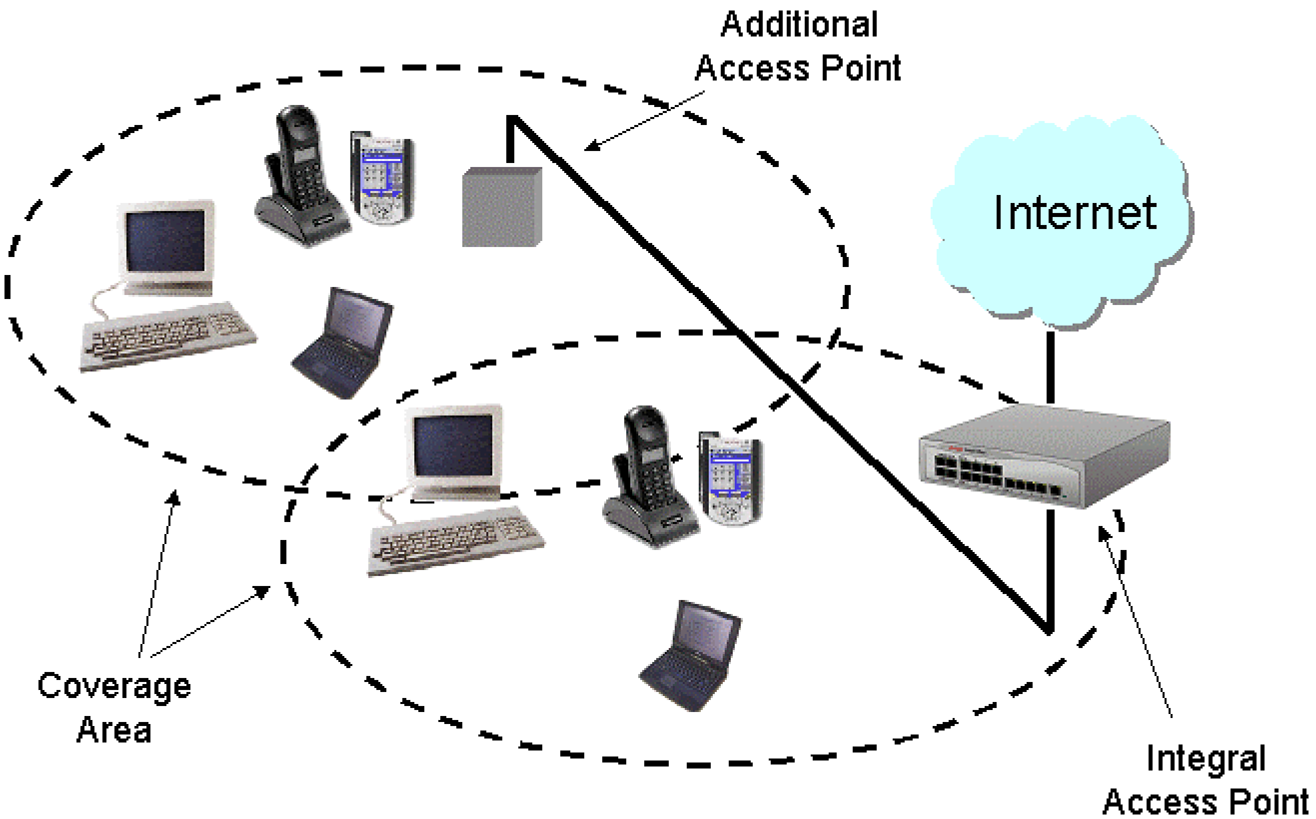
IP400 Office T1 PRI Card

The IP400 Office T1 PRI card provides a single primary rate trunk interface for supporting voice services and fractional leased lines, of 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.
- 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

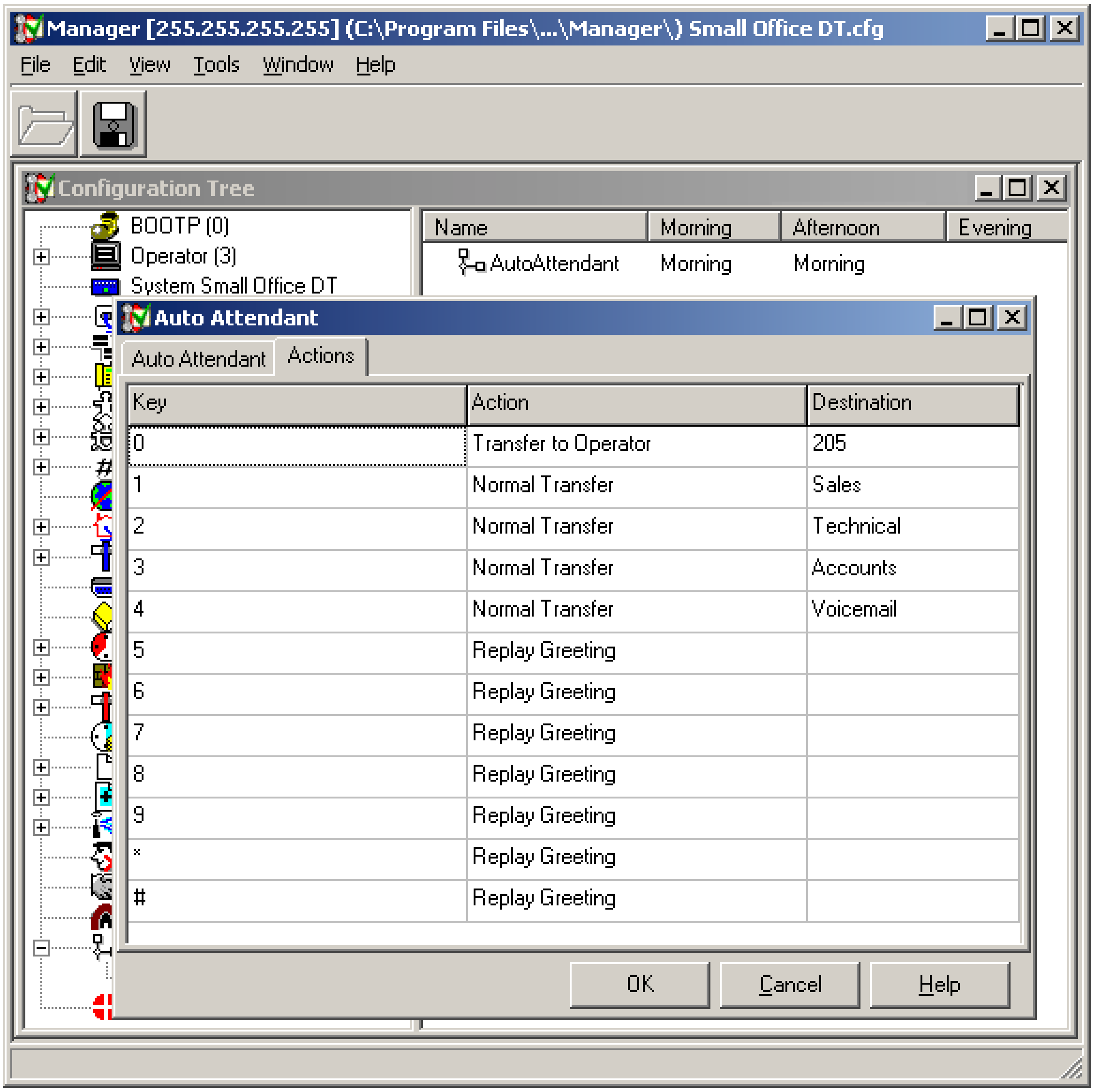
For wireless operation, IP Office - Small Office Edition must be fitted with a Wireless LAN card and the Wireless LAN Access Point license key.

Optional Embedded Voicemail and Auto-Attendant

Embedded voicemail and Auto-Attendant is provided on a pre-loaded 64M PCMCIA Memory card supplied by Avaya. 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 numbers 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.

The embedded voicemail supports up to 15 hours of storage. The number of simultaneous calls is limited by the number of Voice Compression channels that are available.



3. Platform Overview

IP Office Overview

This section provides an introduction to the main components of the IP Office platform that includes IP401 Compact Office, IP403 Office, IP406 Office and IP412 Office.

All IP Office platforms support identical applications therefore any platform can be used in any of the previously described IP Office solutions.

As with all IT and Communications equipment IP Office should be connected to a clean power supply or a UPS.

Additional Information on which components are available in which territories, along with configuration limits and examples, is provided in Appendix A.

IP401 Compact Office Units

IP401 Compact Office

The IP401 Compact Office provides a solution for the smallest of offices and those requiring sophisticated office facilities at home. IP401 Compact Office is a fully converged Voice and Data solution housed within a half sized rack mountable unit. The unit offers functionality including Voicemail, Internet Access, Homeworking and IP Telephony.

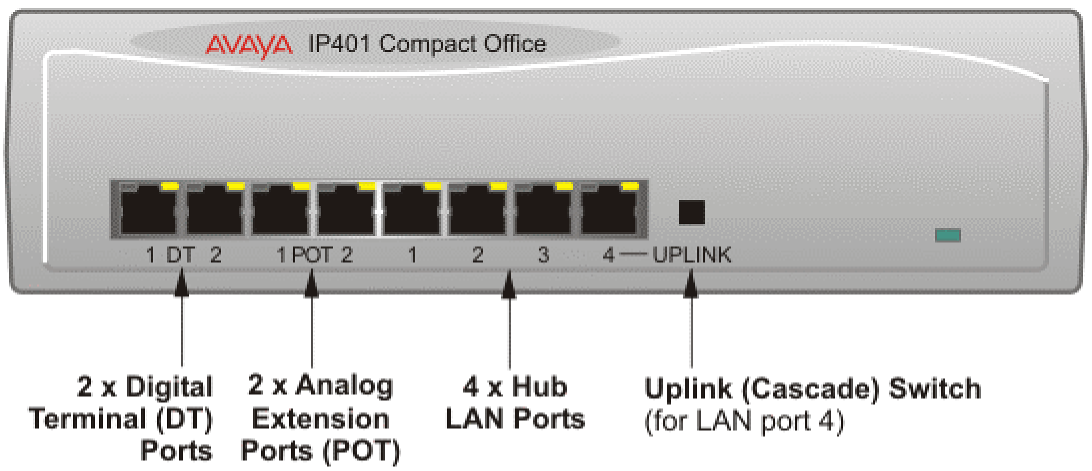
IP401 Compact Office is available in the following versions, not available in all territories.

- **IP401 Compact Office Digital Terminal 2.**
- **IP401 Compact Office Digital Terminal 4.**
- **IP401 Compact Office Upgrades.**

IP401 Compact Office Digital Terminal 2

Pre-configured with support of;

- 2 x Digital Terminal ports for Avaya 20 series handsets
- 2 x Analog telephone ports
- 1 x BRI (2 Lines)
- Four 10/100 Mbps LAN Hub ports (with Cascade Switch for connecting to external hubs)
- USB port
- DTE port
- Audio port for external music on hold source
- Two relay switch port for door entry systems (External O/P socket)
- 4 data channels:
  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.

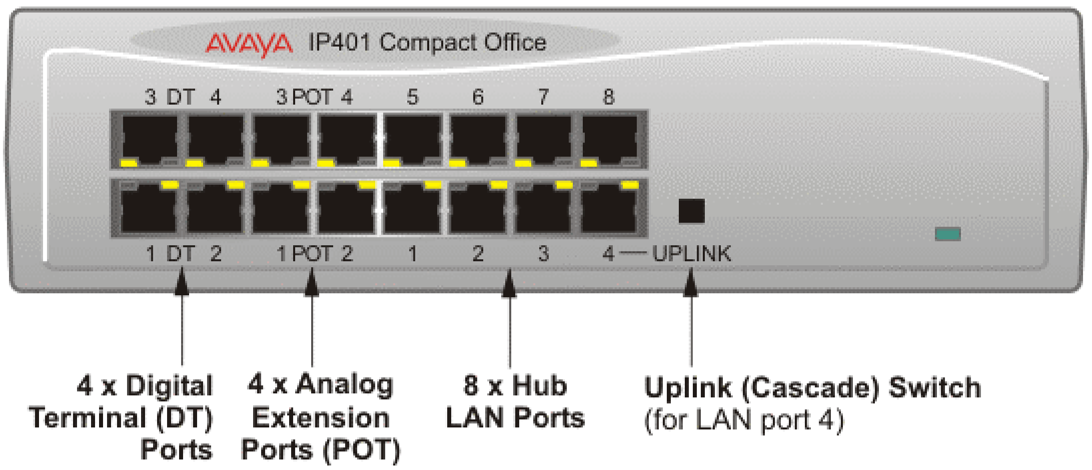


Front Panel

IP401 Compact Office Digital Terminal 4

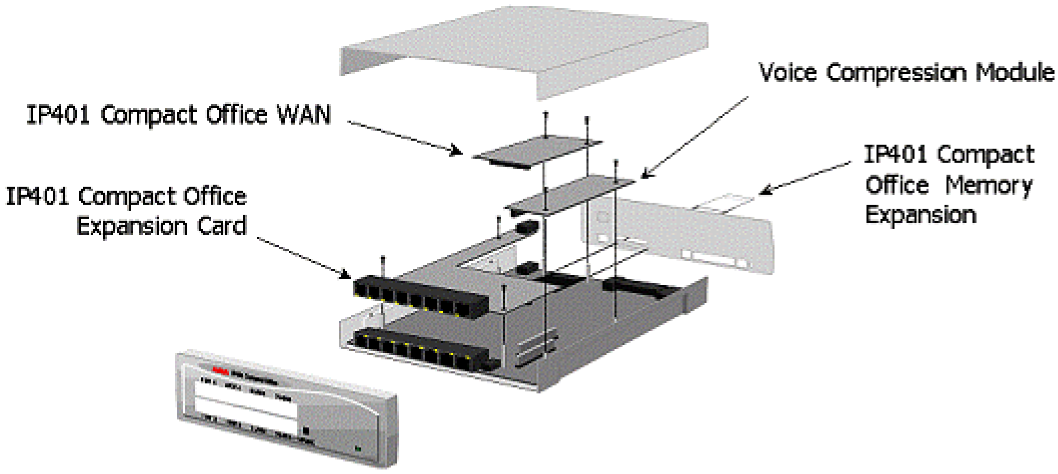
Pre-configured with support of

- 4 x Digital Extension ports for Avaya 20xx series handsets
- 4 x Analog telephone ports
- 2 x BRI (4 Lines)
- Eight 10/100 Mbps LAN Hub ports (with Cascade Switch for connecting to external hubs)
- USB port
- DTE port
- Audio port for external music on hold source
- Two relay switch port for door entry systems (External O/P socket)
- 4 data channels
  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.



Front Panel

IP401 Compact Office Upgrades



- **IP401 Compact Office WAN**
  The IP401 Compact Office WAN module is internally fitted to the IP401 Compact Office to provide a single WAN connection (X21, V35 or V24 via a 37way 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.
- **IP401 Compact Office Memory Expansion**
  A plug in SIM required for embedded Voicemail.
- **IP401 Expansion Kit**
  A kit comprising of Daughter Card, front panel and fittings for converting the IP401 Compact Office Digital Terminal 2 to a Digital Terminal 4.
- **IP401 VCM Upgrade**
  A plug in card required for both the embedded voice mail and support of voice over IP. The IP401 Compact Office supports 5 channels of Voice compression when fitted with the 5 channel Voice Compression module.

IP Office Servers - IP403, IP406 and IP412 Units

IP Office Servers – IP403, IP406 and IP412

The IP403, IP406 and IP412 Office are designed for the small medium enterprise. Scaling from 10 extensions to 360 extensions, each module is available pre-configured (not available in all territories) with a range of trunk configurations.

- **IP403 Office.**
- **IP406 Office.**
- **IP412 Office.**

IP403 Office

The IP403 Office base unit is a 19" rack mountable voice and data communication system and supports as standard -

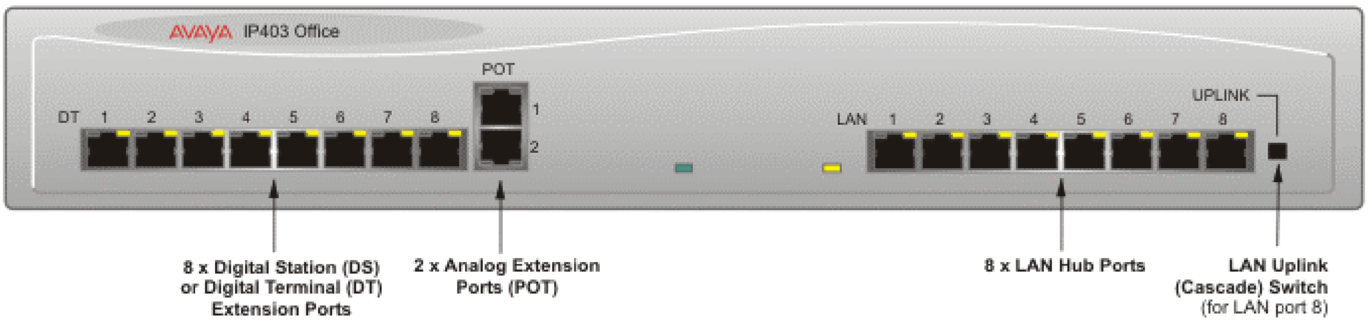
- Eight digital handset ports (Digital Terminal and Digital Station options of the IP403 Office base unit are available dependent on territory).
- Two Analog telephone ports.
- Eight 10/100 Mbps LAN Hub ports.
- DTE Port.
- X.21/V35 WAN interface.
- Support for 3 Expansion Modules.
- Two relay switch port for door entry systems (Ext O/P socket).
- Audio port for external music on hold source.
- 18 Data channels
  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.

The IP403 Office can be ordered in four trunk configurations Quad BRI, PRI (T1), PRI (E1) and Analog (loop start). A spare expansion slot is available to add an additional trunk card (Analog, BRI or PRI).

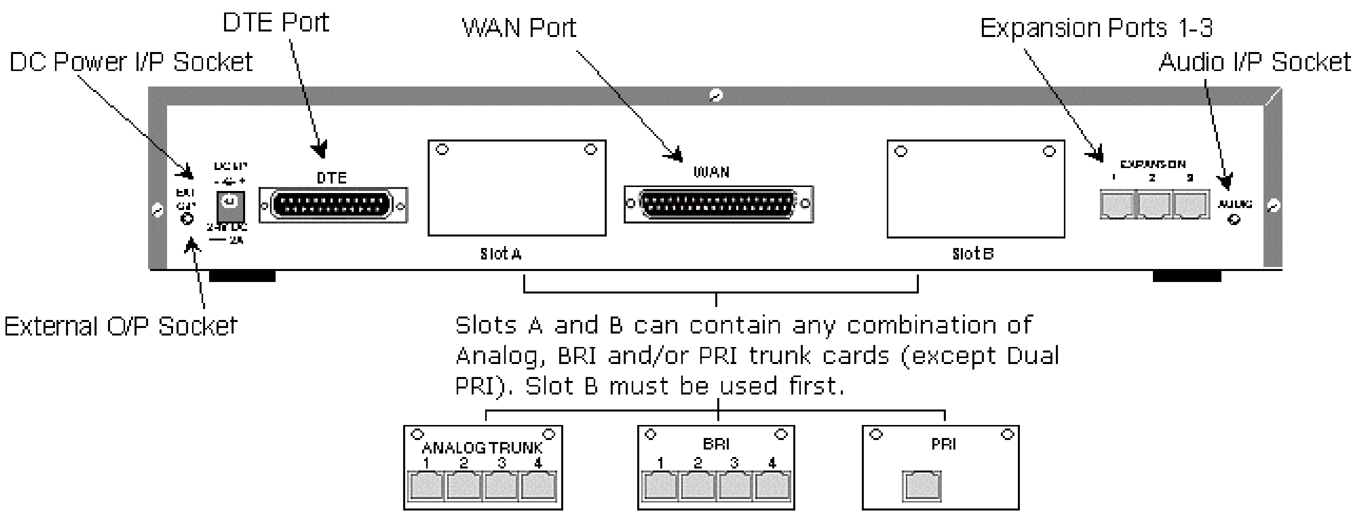
- **Note:** Some trunk configurations are only available in certain territories, please check for local availability.

Optional internal upgrades allow for support of 2 x V.90 modem calls and a 5, 10 or 20 channel Voice Compression Module (VCM). The VCM module supports 5, 10, or 20 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. An IP extension only uses the compression module whilst on a call to a non-IP extension/line. Hence, it is possible to support more extensions than the capacity of the VCM

Through the support of up to three external Expansion Modules, IP403 office can be enhanced to support a further 90 Analog or Digital Handsets.



Front Panel



Rear Panel

IP406 Office

The IP406 Office differs from the IP403 Office in that it supports six Expansion Modules but excludes the integral Digital extension and Analog extension ports. The IP406 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.
- Two relay switch port for door entry systems (Ext O/P socket).
- Audio port for external music on hold source.
- 24 Data channels.
  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.

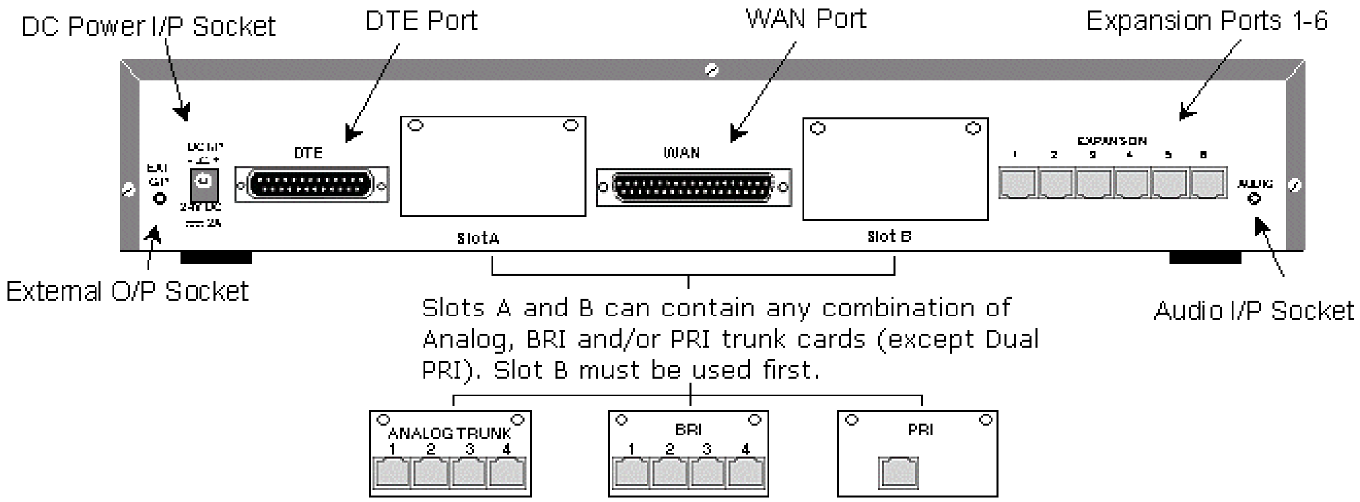
The IP406 Office can be ordered in four trunk configurations Quad BRI, PRI (T1), PRI (E1) and 4 x Analog (loop start) trunks. A spare expansion slot is available to add an additional trunk card (Analog, BRI or PRI).

- **Note:** Some configurations are only available in certain territories, please check for local availability.

Optional internal upgrades allow for the support of 2 x V.90 modem calls and a 5, 10 or 20 channel Voice Compression Module (VCM). The VCM module supports 5, 10 or 20 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. An IP extension only uses the compression module whilst on a call to a non-IP extension/line. Hence, it is possible to support more extensions than the capacity of the VCM.

Through support of up to six external Expansion Modules, IP406 office can be enhanced to support a mixture of Analog, Digital or IP Handsets to maximum of 180.

Front Panel



Rear Panel

IP412 Office

With a more powerful call processing engine and greater internal data transfer capability the IP412 Office 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 Office base unit is 19" rack mountable and supports as standard

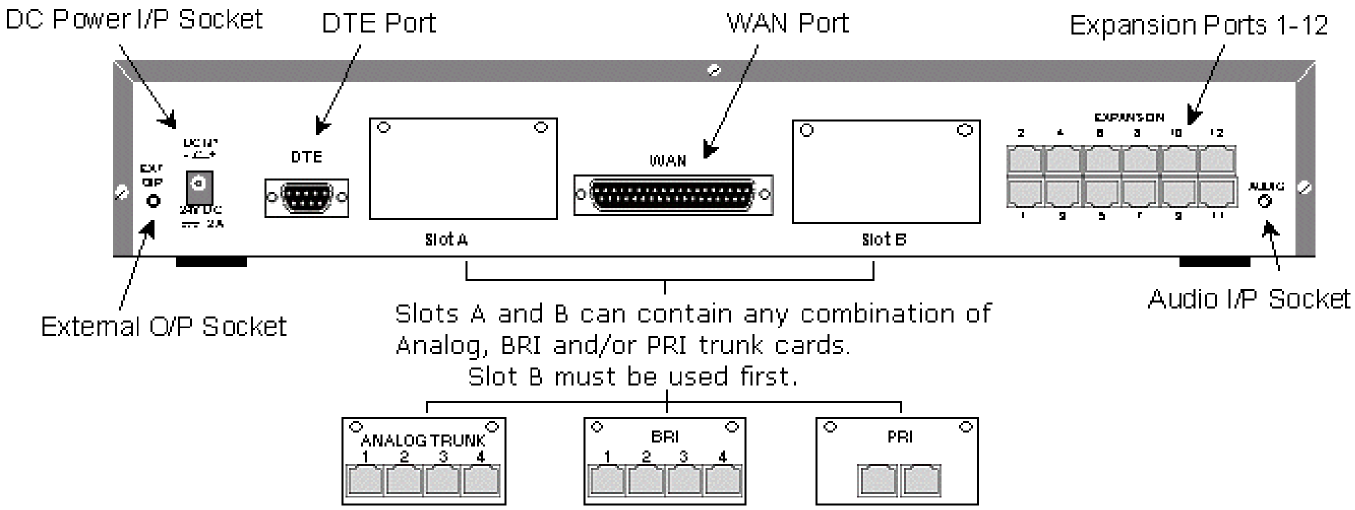
- Two 10/100 switched Ethernet ports.
- DTE Port.
- X.21/V35 WAN interface.
- Support for 12 Expansion Modules (360 extensions maximum).
- Two relay switch port for door entry systems (Ext O/P socket).
- Audio port for external music on hold source.
- 100 Data channels.
  Note: A data channel is used 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 2 x V.90 modem calls and two 5, 10, 20 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. An IP extension only uses the compression module whilst on a call to a non-IP extension/line. Hence, it is possible to support more extensions than the capacity of the VCM's.

IP412 Office can be expanded by 12 Expansion Modules however this is restricted to a maximum capacity of 360 Analog or Digital terminals.

The IP412 Office is available pre-configured with a single or dual PRI (E1/T1) and a spare slot for an optional Quad Analog Trunk, Quad Basic Rate, PRI or Dual PRI (E1/T1).

Front Panel



Rear Panel

External Expansion Module Units

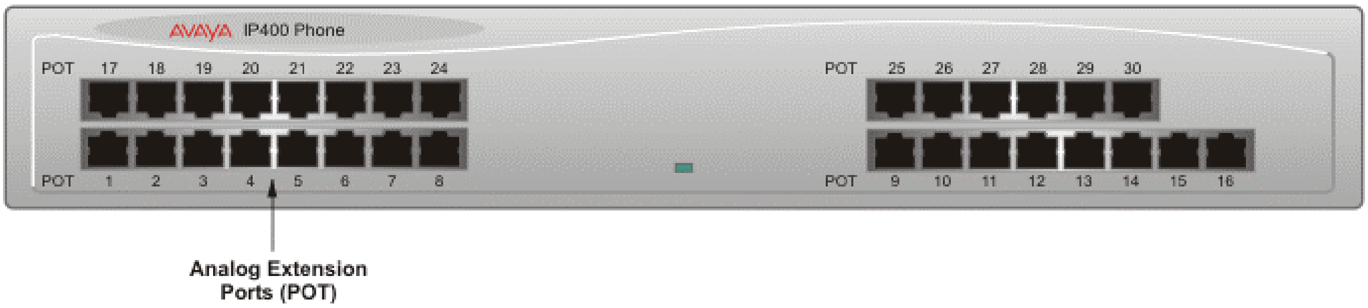
External Expansion Modules

There are ten 19" Expansion modules. The IP403 platform supports any three of these modules, IP406 supports any six, while IP412 supports any twelve (up to a maximum of 360 analog, digital or IP Extensions).

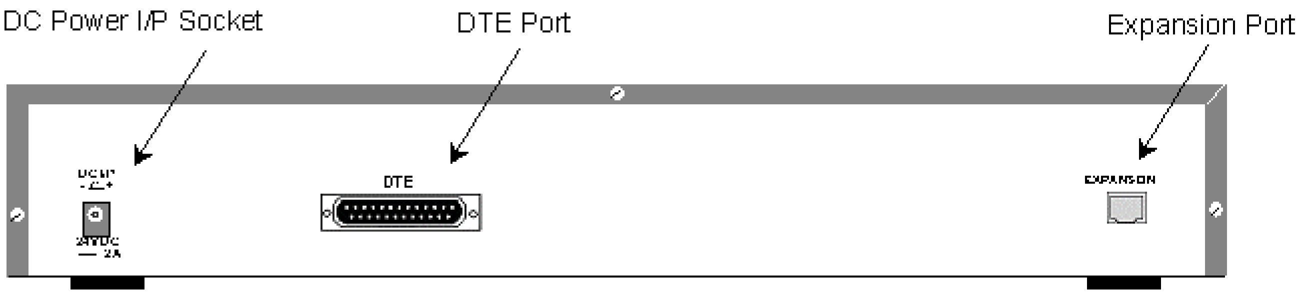
- IP400 Office Phone Module
  Available in three variants for 8, 16 or 30 extensions.
- IP400 Office Digital Terminal Module
  Available in two variants for 16 or 30 extensions.
- IP400 Office Digital Station Module
  Available in two variants for 16 or 30 extensions.
- IP400 Office So 8 Module
  Not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

IP400 Office Phone Module

Provides support for Analog telephones, the IP400 Office Phone module is available in 3 versions giving 8, 16 or 30 extensions. Telephones can be located up to 1km from the unit using CAT5 cabling (see Handset Cable Lengths).



Front Panel (30 port Version)

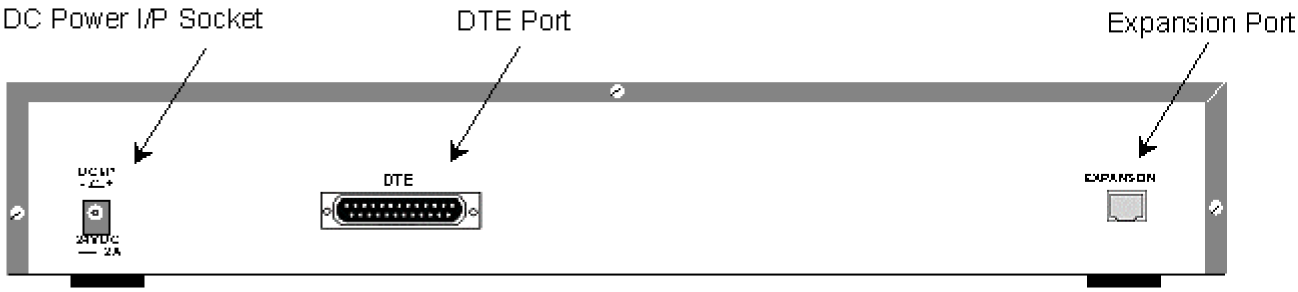


Rear Panel (all versions)

IP400 Office Digital Terminal Module

Provides support for the Avaya 20 series digital display extension terminals (2010, 2030, 2050, 20DS and 20CC terminals), the IP400 Office Digital Terminal module is available in 2 versions; 16 or 30 extensions. Terminals can be located up to 1km from the unit using CAT5 cabling (see Handset Cable Lengths).

Front Panel (30 port version)



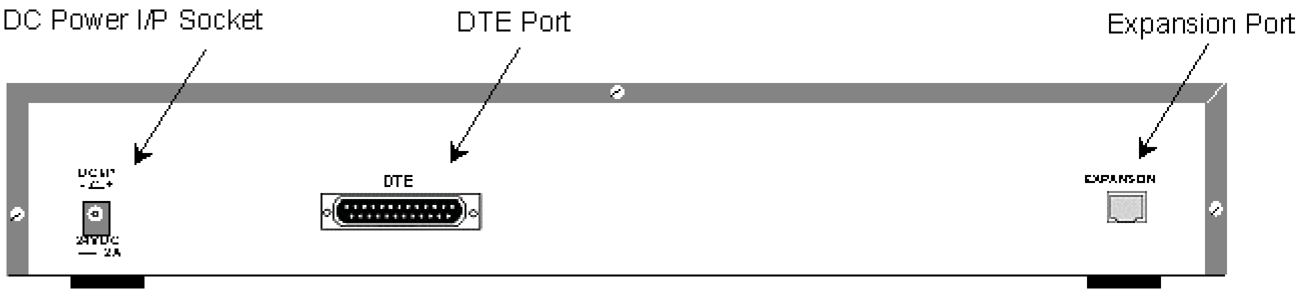
Rear Panel (all versions)

IP Office Digital Station Module

Provides support for the Avaya 4400 and 6400 series terminals. The IP400 Office Digital Station module is available in 2 versions; 16 or 30 extensions. Terminals can be located up to 1km from the unit using CAT5 cabling (see Handset Cable Lengths).

The handsets supported are the 4406D, 4412D, 4424D, 4450DSS, 9040 TransTalk wireless set, 6408D+, 6416D+M, 6424D+M and the XM24.

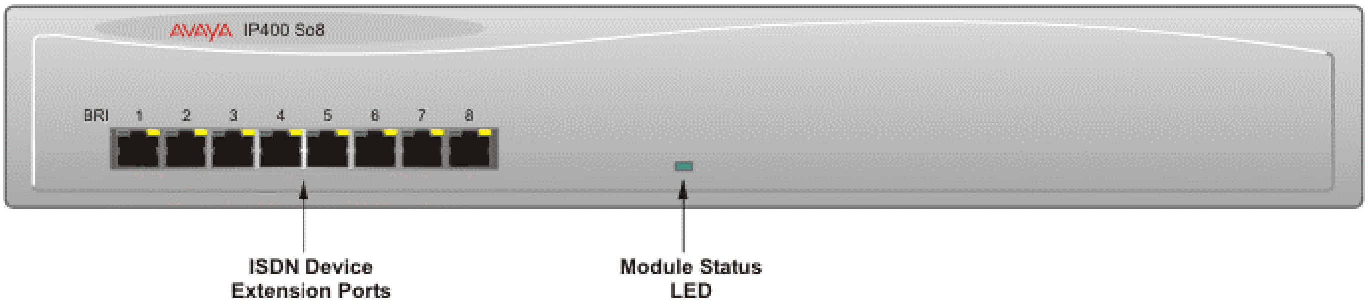
Front Panel



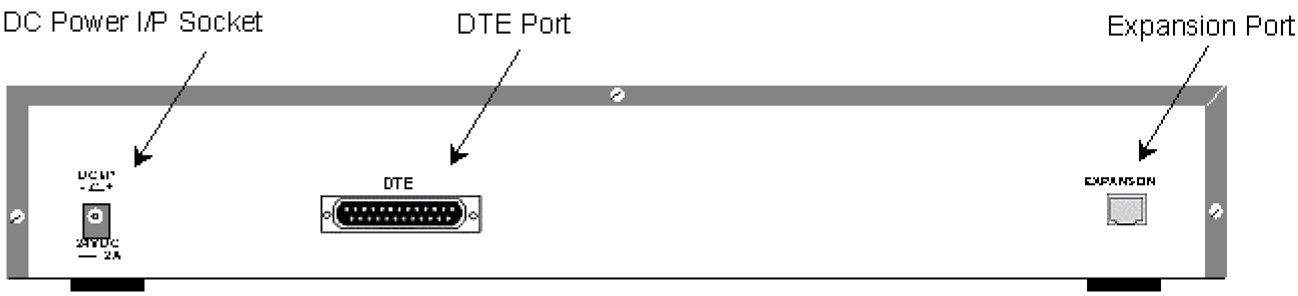
Rear Panel

IP400 Office So8 Module

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



Front Panel



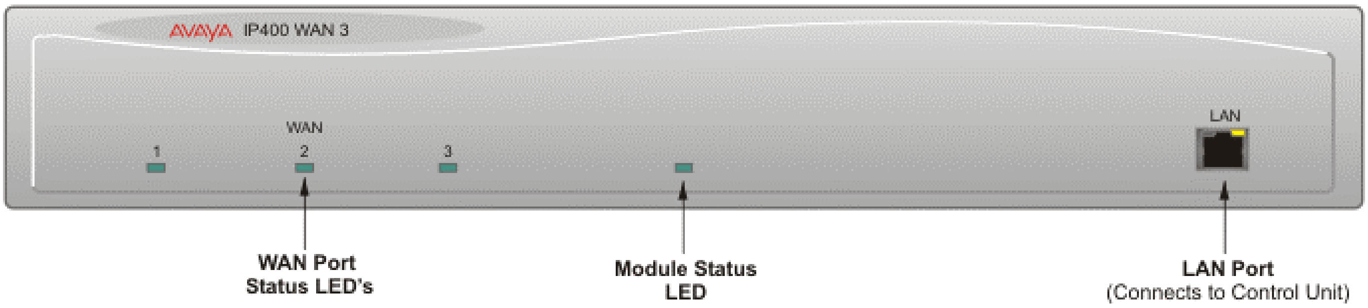
Rear Panel

IP400 Office WAN3

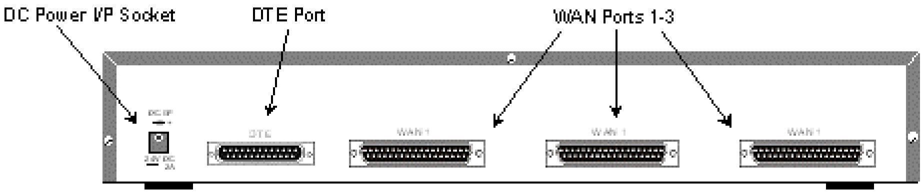
The IP400 Office WAN 3 module provides three WAN connections (X21, V35 or V24 via a 37way D Type socket). Line speeds up to and including 2Mbps are supporting on each interface, the carrier providing the line dictates the actual operating speed i.e. in some territories the maximum speed may be 1.544M. These interfaces are identical to the single connection provided as standard on the IP403, IP406 and IP412 platforms.

The IP400 Office WAN3 may be connected to the IP403, IP406 and IP412 platforms to provide additional WAN ports. Each platform can support two of these modules.

The IP400 Office WAN3 connects to the base unit through the Local Area Network and does not use an expansion port on the base module.



Front Panel

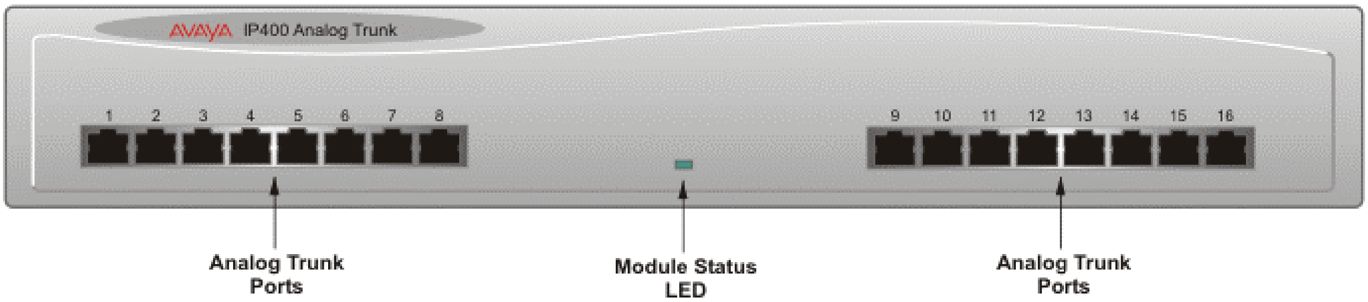


Rear Panel

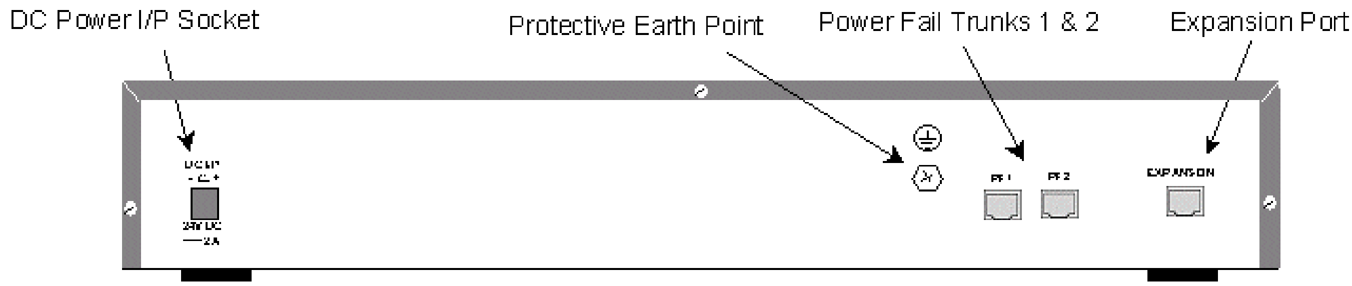
IP400 Office Analog Trunk 16

Each module supports up to sixteen Loop Start or Ground Start* 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.



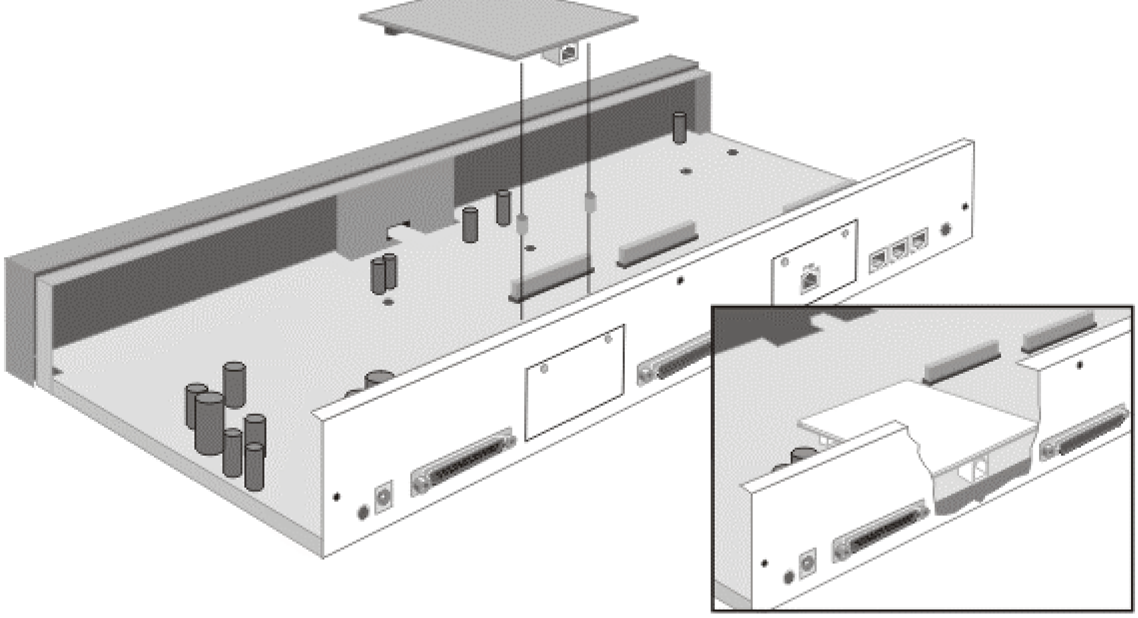
Front Panel



Rear Panel

Trunk Interface Cards

Trunk interface cards are rear mounted to provide flexible trunk connectivity for the IP403, IP406 and IP412 platforms. The IP403 supports a single trunk interface card with an optional plug-in analog trunk card, while the IP406 and IP412 support two trunk interface cards.



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

- IP400 Office Quad BRI.
- IP400 Office PRI E1.
- IP400 Office Dual PRI E1 (IP412 only).
- IP400 Office E1R2MFC.
- IP400 Office Dual E1R2MFC (IP412 only).
- IP400 Office PRI T1.
- IP400 Office Dual PRI T1 (IP412 only).
- IP400 Office 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.

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

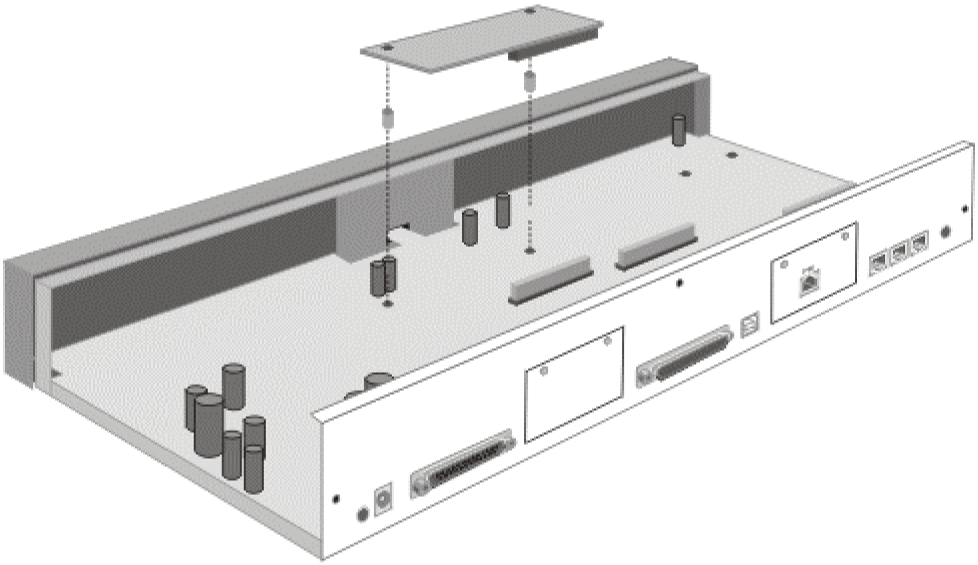
T1 trunk cards incorporate an integral CSU/DSU, eliminating the need for an external unit. The CSU function allows the trunk to be put in loop-back mode for testing purposes. This can be set manually, using the monitor application, or automatically from a Central Office sending a Line Loop Back (LLB)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

IP400 Office Quad Analog Trunk (LS) Card

Provides four Analog trunk 2 wire interfaces (loop start) including support for caller ID.

Internal Daughter Cards

Internal Daughter Cards are fitted inside the base module of the IP403, IP406 and IP412 platforms.



IP400 Office VC Module – 2/5/10/20/30

The Voice Compression Module (VCM) is used for Voice over IP (VoIP) applications in the IP401, IP403, IP406 and IP412 control units (on IP Office - Small Office Edition systems VCM/VoIP channels are pre-built). Four VCM variants are available supporting 5, 10, 20 and 30 channels of compression.

Up to two VCMs can be fitted to the IP412, the other control units support only a single VCM. The capacity of VCM module supported also varies.

- IP401: Supports only a single VCM5.
- IP403 & IP406: Support a single VCM5, VCM10 or VCM20.
- IP412: Supports any two from VCM5, VCM10, VCM20 and/or VCM30.

The VCM is also used for the embedded Voicemail system on the IP401 – see 'The Applications' section.

IP400 Office Modem 2 card

The integral dual V.90 (56kbps) digital Central Site modem card, allows termination of two simultaneous analog modem calls. These calls are presented over a digital BRI or PRI bearer. The IP400 Office Modem 2 Module is supported on the IP403, IP406 and IP412 platforms.

4. Terminals

Introduction to IP Office Terminals

Terminals are the natural focal point for the users of any telephone system. A communication platform may have very sophisticated functionality, but without user friendly telephone sets much of this is hidden and thus unused by the average user. All Avaya terminals 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.

The IP Office supports a wide range of dedicated digital terminals.

- 2010 Terminal
- 2030 Display Terminal
- 2050 Display Terminal
- 20CC Call Center Terminal
- 20DS Unit
- 20DT - DECT Cordless Handset
- 2420D Terminal
- 3810 Wireless Handset
- 4406D Terminal
- 4412D Terminal
- 4424D Terminal
- 4450 Unit
- 6408D Terminal
- 6416D Terminal
- 6424D Terminal
- XM24 Unit
- TransTalk 9040 Wireless Handset
- Analog Telephones/POTS

In addition a number of IP Hardphones are supported. For full details see IP Hardphones and Wireless VoIP.

- 4602 IP Hardphone
- 4606 IP Hardphone
- 4612 IP Hardphone
- 4620 IP Hardphone
- 4624 IP Hardphone
- 3616 Executive Wireless Phone
- 3626 Ruggedized Wireless Phone

- For maximum cabling distances please refer to Configurations and Factory Build Options.
- For details on IP hardphones, see the 4600 series.
- **Note:** Not all terminals are available in all territories. Please check for local availability.
- Note: IP Office does not support SIP hardphones as Release 2.1.

Display terminals employ an intuitive interface at the desktop for the user ensuring that the complex array of features are simple to access. The result is a 'context sensitive' display that offers the user features appropriate to the terminal's status and that visually prompts and assists the user in programming or using the terminal. The result is reduced complexity for the user ensuring all features are easily accessible. This benefits the user and the company by facilitating the use of features that improve communication, increasing efficiency and saving costs.

2010 Terminal

The 2010 supports the following features:

- Message Waiting Light.
- On Hook Dialing.
- Receive & Make Page.
- Hands Free Speech.
- Headset Capability.
- Wall Mountable.
- Hearing Aid Compatible.
- 6 Fixed Feature Keys.

2030 Display Terminal

The 2030 supports the following features:

- Message Waiting Light.
- On Hook Dialing.
- Receive & Make Page.
- Hands Free Speech.
- Headset Capability.
- LCD Display (Custom Large Call Information Widow and 2x16 Alphanumeric Display).
- 8 Key Direct Station Select.
- 4 Context Sensitive Soft Keys.
- 11 Fixed Feature Keys (Speaker, Scroll, Redial, Speed Dial, Hold, Answer/Release, Mute, Divert, No Calls, Group, Program).
- Wall Mountable.
- Hearing Aid Compatible.

2050 Display Terminal

The 2050 supports the following features:

- Message Waiting Light.
- On Hook Dialing.
- Receive & Make Page.
- Handsfree Speech.
- Headset Capability.
- LCD Display.
- Dual-Color BLF.
- 8 Key Direct Station Select.
- 4 Context Sensitive Soft Keys.
- 11 Fixed Feature Keys (Speaker, Scroll, Redial, Speed Dial, Hold, Answer/Release, Mute, Divert, No Calls, Group, Program).
- Wall Mountable.
- Hearing Aid Compatible.

20CC Call Center Terminal

The 20CC agent display terminal is a dedicated turret terminal for call center applications and supports the following features:

- Message Waiting Light.
- On Hook Dialing.
- Receive & Make Page.
- Headset Capability. (Note: Headsets are separately ordered)
- LCD Display.
- Dual-Color BLF.
- 8 Key Direct Station Select.
- Log On/Log Off To Register Each Agents Shift Duration.
- 11 Fixed Feature Keys (View, Scroll, Redial, Speed Dial, Hold, Answer/Release, Mute, Log On/Log Off, Busy Not Available, Busy Wrap Up, Program).
- 4 Context Sensitive Soft-Keys.

20DS Unit

The 20DS works in association with your chosen 20 Series display terminal.

It provides your phone with an additional 42 Direct Station Select keys and dual-color Busy Lamp Field (DSS/BLF). You can associate up to two 20DSs with your IP Office display terminal. Linking kits are provided to link the 20DS to your display terminal and any additional 20DS unit.

Each 20DS requires its own extension port and an AC (mains) power socket. Each IP Office DT module supports a maximum of two 20DS units only.

20DT - DECT Cordless Handset

IP Office DECT Handset 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).

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.

2420D Terminal

The 2420D supports the following features:

- High-end feature set with productivity local call log & speed dial directory.
- Advanced user interface.
- Reduced installation and move costs – no paper labels.
- Investment protection with downloadable firmware.
- Large screen 7 line x 29 character display.
- Twenty-four call appearance/feature buttons in 3 pages.
- Adjustable Desk Stand.
- Fully Global ready (Icons).

3810 Wireless Handset

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

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 10 Avaya 3810 wireless sets can be connected to the same PBX.

4406D Terminal

The 4406 supports the following features:

- 6 Programmable Feature buttons with LED.
- 5 Fixed Feature Keys: Speaker, Mute, Hold, Volume Up & Down.
- 3 Fixed Feature Keys below the Display: Conference, Transfer, Redial.
- 2 x 16 Character Display.
- Message waiting indicator.
- Two-way handsfree speaker phone.
- Hearing aid compatible.
- Optional wall mounting/desk stand.

4412D Terminal

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

- 12 Programmable keys with LED.
- 12 Programmable keys without LED.
- 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.

4424D Terminal

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

- 24 Programmable Feature Keys with LED.
- 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.
- 7x24 character display.

DSS4450 Unit

The DSS4450 works in association with the 4412D+ and 4424D+ telephone.

Auxiliary power is required. Each DSS4450 provides an additional 60 Direct Station Select keys and dual color Busy Lamp Field (DSS/BLF). Each telephone can support 2 DSS4450 adjuncts. No more than 2 DSS4450 adjuncts are supported per DS Module.

4602 IP Hardphone

The 4602 supports the following features:

- 9 Fixed Feature Keys: Conference, Transfer, Drop, Hold, Redial, Mute, Volume up & down, Speaker, Voice Mail.
- 2 X 24 Character based Eurofont Display.
- Message Waiting Indicator.
- Call Monitor Speaker.
- G.711, G.729a/B Voice CODECs.
- QoS Options of - UDP Port Selection, DiffServ and 802.1p/B (VLAN).
- Single 10/100 BaseT Ethernet port.
- Support for Simple Network management Protocol (SNMP).
- Hearing Aid Compatible.
- Microsoft NetMeeting Compatible.
- IP Address Assignment - DHCP Client or Statically Configured.
- Downloadable firmware for future upgrades.
- Wall Mountable with included desk/wall mount stand.
- Avaya Grey Color for all markets.

IP Office also supports the 4602SW, which includes all of the above features plus an integrated Ethernet switch for PC connection.

- Integrated Full Duplex 10/100 BaseT Ethernet Switched ports for connection of PC.
- Auto-negotiation provided separately for each port.
- 802.3 Flow Control.
- Phone has priority over PC port at all times.

4606 IP Hardphone

The 4606 supports the following features:

- 6 Programmable Feature Buttons With LED.
- 5 Fixed Feature Keys: Speaker, Mute, Hold, Volume Up & Down.
- 3 Fixed Feature Keys Below The Display: 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.
- Optional Integrated Ethernet Repeater Hub – for Connecting PC To Phone.
- Hearing Aid Compatible.
- Microsoft NetMeeting 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.

4612 IP Hardphone

The 4612 supports all of the features of the 4606 with the following differences:

- 12 Programmable keys with LED.
- 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.
- 2x24 Character Display.

4620/4620SW IP Hardphone

In addition to the features of the 4602/4602SW, the 4620/4620SW supports the following:

- 24 Programmable Feature Keys (presented in 2 pages of 12).
- Automatically labeled from the system (no paper labels).
- 6 Fixed Feature Keys: Speaker, Mute, Hold, Headset and Volume Up/Down.
- Large graphical gray-scale display (168 x 132 dots).
- 5 Fixed Feature Keys below the display: Conference, Transfer, Hold, redial and Drop.
- 4 Embedded applications: Speed Dial, Call Log, Web Browser (WAP/WML), Options.
- Full-duplex speaker phone with acoustic cavity for improved sound quality.
- Feature Key Module (FKM) interface jack for use with the EU24, 24 button expansion module (support for this module will be available in a later release of IP office).
- 7 Position adjustable desk stand/wall mount stand.
- Infrared (IrDA) port.
- Built-in headset jack.
- Multiple language support built-in: English, French, Italian, Spanish & KataKana.
- 8 Personalized ring patterns.

4624 IP Hardphone

The 4624 supports all of the features of the 4606 with the following differences;

- 24 Programmable Feature Keys With LED.
- 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.

6408D Terminal

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 Flexible dual LED feature keys.
- 8 Fixed feature keys: Speaker, Mute, Conference, Transfer, Redial, Hold, Volume Up & Down.

6416D Terminal

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 Flexible dual LED feature keys.
- 8 Fixed feature keys: Speaker, Mute, Conference, Transfer, Redial, Hold, Volume Up & Down.

6424D Terminal

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 Flexible dual LED feature keys.
- 8 Fixed feature keys: Speaker, Mute, Conference, Transfer, Redial, Hold, Volume Up & Down.

XM24 Unit

The XM24 is an expansion module that works in association with a 6416 or 6424 display phone and provides an additional 24 Direct Station Select keys and dual-color Busy Lamp Field (DSS/BLF). Only one XM24 per 6416/6424.

Each IP Office DS module supports a maximum of two XM24 units only.

TransTalk 9040 Wireless Handset

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.

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.

Analog Telephones / POTS

As well as providing a lower cost alternative to system specific terminals, analog terminals 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.

Uniquely, analog terminals 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 terminals 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.

5. Telephony Functions & Call Handling

Telephony Functions & Call Handling

For most businesses the telephone remains the prime means of contact with customers, prospects, suppliers and colleagues alike. IP Office provides a comprehensive telephony feature set to enable a fast, courteous and efficient response to a telephone call which can make the difference between winning and losing business.

Features such as CLI/ANI display and alpha 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 and following facilities 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 Dialing (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.

Feature / Handset Compatibility

	2010	2030	2050	20CC	4406D	4412D	4424D	4602	4606	4612	4620	4624	6408D	6416D	6424D	2420D	POT	CLI POT
Absent Text		●	●	●	●	●	●	●	●	●	●	●	●	●	●	●		
Account Codes		●	●	●	●	●	●	●	●	●	●	●	●	●	●	●		
Auto-Answer		●	●	●	●	●	●	●	●	●	●	●	●	●	●	●		
Automatic Call Distribution	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
BLF			●	●	●	●	●	●	●	●	●	●	●	●	●	●		
Call Barring	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Call Coverage	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●		
Call Forwarding	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Call History		●	●	●	●	●	●	●	●	●	●	●	●	●	●	●		
Call Hold	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Call Intrude	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Call Park	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Call Pickup	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Call Queue	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Call Steal	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Call Timer		●	●	●	●	●	●	●	●	●	●	●	●	●	●	●		
Call Transfer	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Call Waiting	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Callback	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Caller Display		●	●	●	●	●	●	●	●	●	●	●	●	●	●	●		●
Clear Call Waiting	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Conference Calls	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Dial Ahead		●	●	●	●	●	●	●	●	●		●	●	●	●			
Dial Emergency	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Dial On Pickup (Hotline)	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Directory Access		●	●	●	●	●			●		●	●	●	●	●	●		
Distinctive Ringing		●	●	●													●	●
Do Not Disturb	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
" Exceptions	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
E911	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Extension Password Change	●	●	●	●	●	●	●		●	●	●	●	●	●	●			
Follow Me Here	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Follow Me To	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Forward on Busy	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Forward on No Answer	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Forward to Specified Number	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Forward Unconditional	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●

	2010	2030	2050	20CC	4406D	4412D	4424D	4602	4606	4612	4620	4624	6408D	6416D	6424D	2420D	POT	CLI POT
Group In/Out	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Group Paging -Make	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
- Receive	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●		
Handsfree Speech	●	●	●		●	●	●		●	●	●	●	●	●	●	●		
Headset capability	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Hold	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Hot Desking	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Hot Transfer	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Least Cost Routes	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Lock	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●			
Login	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Manager-Secretary Working		●	●															
Message	●	●	●	●	●	●	●	●	●	●		●	●	●	●			
Message Waiting Light	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●		
Monitor Calls *	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Multi-language		●	●	●	●	●	●	●	●	●	●	●	●	●	●	●		
Mute	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●		
Night Service	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
On Hook Dialing	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●		
Park	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Personalized Ring		●	●	●	●	●	●		●	●		●	●	●	●	●		
Queuing a Transferred Call to a Busy extension	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Record a Call	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Redial	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Relay On/Off/Pulse	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Meet me Conference	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Ring Back When Free	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Self Administer																●		
Soft Key Labelling					●	●	●		●	●	●	●	●	●	●	●		
Speed Dialing	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Suspend Call Waiting	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Suspend/Resume	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Time/Date		●	●	●	●	●	●	●	●	●	●	●	●	●	●	●		
Toggle Calls	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Voicemail Collect	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Voicemail On/Off	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Voicemail Ringback On/Off	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●	●
Volume Adjustment		●	●	●	●	●	●	●	●	●	●	●	●	●	●	●		

*Notes: IP Phones can monitor calls but they cannot be monitored from another extension.

Extension Features

Absent Text

Allows a user to set an absent statement that will be displayed on the internal callers terminal as well as on Phone Manager and SoftConsole. The system has 10 pre-configured messages as well as the ability to customize text.

Call Coverage

Known as covering extensions, Call Coverage allows an extension to act as an answering point for another user's call. This is typically used in Personal Assistant / Manager environments.

Call 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 when there is no answer, when the extension is busy or for all calls. A user can enable/disable forwarding from their terminal, via the Phone Manager application or through Voicemail Pro.

If the user is also a member of a hunt group, they can separately control forwarding of their Hunt Group calls, thereby allowing them to choose whether to be presented with hunt group calls or not. 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.

Call 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 period.

See also **Call Park**.

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.

Call Park

As an alternative to placing a call on hold a call can be parked in 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. In default there are four system park areas, identified by Park IDs 1, 2, 3 or 4 which can be accessed from any extension. Additionally users can create their own personal or group Park IDs. Consequently, an unlimited number of parked slots can be set up for a specific user, for a department or for the entire organization.

After a period the parked call is re-presented to the extension that originally parked the call.

Call 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 Steal / Acquire Call

The Call Steal facility allows a user to take over (steal) the call, from another extension. This function is useful when you want to catch a call you have just missed, e.g. that has been diverted to voicemail. The RECLAIM function in the Phone Manager application also performs this function.

Call 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 a 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.

There is no differentiation between internal and external calls (other than ringing sequence) so it's just as easy to transfer a call to extension 201 as to a mobile/cellular telephone.

Call Waiting

If a user is busy on a call they can chose to have another call wait for them until they become free. The user will be made aware that a call that is waiting for them by a call waiting tone and, by using Phone Manager, they will receive additional information to assist them to decide which conversation they wish to continue with.

The user may:

- Ignore the Call Waiting.
- Disable Call Waiting for the duration of this call.
- Clear the current call and pickup the waiting call.
- Place the current call on hold and pick up the waiting call.
- Suspend the current call into the specified park slot and pick up the waiting call.
- Pass the call to voicemail.
- Send all calls to voicemail.

Clear Call Waiting

Similar to hold call waiting, Clear Call Waiting is a compound feature that clears an existing call and answers the waiting call.

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 IP403/IP406 systems can support one conference of 64 calls or multiple conferences of up to 64 parties, e.g. 21 conferences of 3 calls each. The IP412 has two 64-party conference circuits giving either 2 x 64-party or 42 x 3-party capacity.

- **Notes:** Only two analog trunks are permitted in any single conference. IP401 supports a single 3-way conference. Further conferencing options are available using Conferencing Center as detailed in the applications section.

Dial Ahead

This facility enables a user, when connected to a call, to establish the status of another extension without having to place the original caller on hold. The user can see whether the extension is free, busy, unobtainable or on divert (the divert destination is also shown). Dial ahead enables a user/operator to try several extensions very swiftly without interrupting the call and to then inform the caller in advance as to where they are being transferred to. This capability is available with the 20xx terminals and SoftConsole.

Dial On Pickup

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

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 dialing via Phone Manager or a display terminal.

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 terminal 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.

Distinctive Ringing

It is possible for an analog telephone to ring in 11 different styles. By default three of these styles are used for "External Calls", "Internal Calls", and "Ring Back Calls".

Do Not Disturb

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.

If specific numbers, internal and external, are required to override Do Not Disturb these can be added to the User's exception list.

Enhanced Intrusion (Whisper Page)

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.

Follow Me

The **Follow Me To** facility allows a user to take their calls from another location, whether this is an internal or external number. This feature can be set at the user's extension or via the Phone Manager application.

The **Follow Me Here** facility allows a user to take their calls from another extension. This feature can be set at the destination extension.

In both cases, if the redirected call receives busy tone or is not answered then the call behaves as though the user's extension had failed to answer and will follow the user's Forward settings.

Handset Dial By Name

Allows a user to look-up another user by spelling their name on the display terminals numeric keypad and then establish the call.

Hot Transfer

Hot transfer is the ability to transfer a call without personally answering the call. A user can perform a Hot Transfer via a PC application, for example, Phone Manager. This will display information regarding the caller, which may assist the user to decide who to pass the call on to. The extension receiving the transferred call will be informed, via Caller Display, where the call was transferred from and pass on any available information regarding the original caller.

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 feature.

Login

A contact center agent function, login is required before the agent is able to make or receive calls from that terminal. 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 unmanned.

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 IP Office Conferencing Solution.*

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

Monitor Calls

A user can monitor, i.e. listen in, on Hunt Group calls. The user must be a member of the Hunt Group to be monitored. If a User has Cannot Be Intrude feature programmed, their calls cannot be monitored. IP phones cannot be monitored.

Ring Back When Free

If an extension is busy and the user wants to be informed when the extension becomes free, the system will ring the user's telephone and give the appropriate Caller Display information to advise that the destination is free. When the telephone is picked up a call will be automatically made to the extension. This capability can be set via Short Code, Button Programming, or Soft keys. When accessed via Soft keys, the Display will provide options of Call Back when Free and Auto Call Back.

Relay On/Off/Pulse

IP Office is fitted with two independent relay switches 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.

Suspend/Resume

Suspend/Resume are only available on certain exchanges/Central Offices supporting this ISDN feature. When suspended the call is held at the local exchange freeing the ISDN channel for another call. Resume reconnects to the held call.

Suspend Call Waiting

Suspend Call Waiting is a compound feature that will hold a call and answer a waiting call. See 'Suspend/Resume' above.

Toggle Calls

Toggle Calls cycles round each call that is On Hold locally within the system. This does not include those Suspended at the Local Exchange or Central Office.

System Features

Account Codes

Account Codes allow a system to track calls. For example, a lawyers' office may wish to record the amount of time spent on calls to a client. Each client is given an Account Code and that code is used when making a call. This Account Code is then recorded with the call information in the call logger. Incoming calls from the client can be assigned Account Codes automatically using the CLI/ANI, via Phone Manager or a digital/IP display terminal. To ensure that every call to this client is recorded a user can be forced to use an account code when making an external call. Account codes can be either forced or voluntary but must be pre-registered within the IP Office system.

Automatic Call Distribution (Hunt Groups)

A Hunt Group is a collection of 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

- **Hunt mode / Linear mode:**
  One extension at a time sequentially.
- **Group mode:**
  All extensions simultaneously.
- **Rotary mode / Circular mode:**
  Start with extension next in list to extension that was used last time.
- **Idle mode / Most Idle mode:**
  Start with extension with the longest idle 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. Whilst 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.

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.

Caller Display

Caller Display uses the CLIP (Caller Line Identification Presentation) or ANI (Automatic Number Identifier) passed by the Telephone Company via the trunk line. For caller display on analog phones CLIP/ANI is converted to the analog version of the call display service. CLIP/ANI is also used by IP Office's PC programs such as Phone Manager and the PC TAPI interface. Thus users can see the telephone number of the person calling. Extensions can be configured to enable or disable Caller Display.

The **Directory** feature is used to assign names to recognized numbers.

Dial Emergency

Allows any user to dial a short code to override call barring and dial the emergency services.

External Control Port

The door release mechanism on the unit consists of two relay switches which can be either normally open, normally closed, pulsed open or pulsed closed.

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.

Group Paging

A group of users can be placed within a paging group for the purpose of receiving voice announcements via the speaker of their digital telephone when idle. IP Office also allows POT ports to be configured for connection to external tannoy or paging systems. With Release 2.1, IP phones are now able to be part of a paging group.

Hold Music

The system supports both internal and external music on hold. The internal source uses a WAV file of up to 30 seconds length. WAV files are industry standard making it simple to change the music to meet the customers needs. External music on hold devices connect to the 3.5mm Audio socket located on back of IP Office - Small Office Edition, IP401, IP403, IP406, and IP412 base unit.

- **Note**: The IP401 only supports external music on hold.

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 a telephony and Voicemail service without being permanently assigned a physical extension.

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, the calling parties telephone number or CLI/ANI (This could even be part of the number received such as an area code), routing based on presentation digits from the exchange such as DDI/DID or ISDN MSN, routing based on sub-address and routing based on the 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.

Intrusion Warning Tone

This is a system wide setting that enables or disables warning tones played to users that are being intruded upon.

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 also 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.

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.

Off Switch Call Inhibit

This is a system wide setting that prevents external calls being forwarded off switch as a precaution against toll fraud. This also prevents trunk to trunk transfers.

Outgoing Calls

When making an outbound call the system first checks if the dialed digits are an internal extension. If not, the system then checks to see if the dialed digits are a feature code. If the dialed digits are neither an internal extension number or a facility code, then the system concludes it is an external call. Hence, it is not necessary to prefix an external number with a line access code.

PIN Restricted Calling

IP Office can force a user to enter a PIN code or authorization code when they attempt to make a call. PINs can be applied to an individual telephone number or calls of a specific type, such as international calls. The Call Logging output includes the PIN code, allowing a system administrator to search for all calls made by a user regardless of the telephone used.

For added security, an option is provided to prevent PIN codes from being displayed on Phone Manager and digital/IP display terminals. Note: Caller-Display Analog Phones will display the PIN number dialed regardless of this option.

Personal Fax Numbers

Individuals and departments can have their own fax numbers. When an incoming fax is received on one of these numbers, DTMF tones (which identify the individual) are passed to the fax server. This feature allows many users/departments to share a fax server which could have as few as one or two lines. Alternatively fax machines can be connected to any extension port on the Phone module.

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.

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

Queuing a Transferred Call to a Busy Extension

When transferring a call, if the destination extension is busy and the caller wants to hold for that person, the call can be queued against the extension until the extension is free. The transferee is no longer involved in the call.

Short Codes

Short Codes are one of the most powerful facilities within IP Office. As well as being used by analog telephones to invoke terminal features, Short Codes allow for dialed number translation. For example, the creation of speed dial numbers, call baring and alternate carrier selection. Short Codes can be configured at user, System and Least Cost Route levels.

Speed Dialing

Short Codes can be used to create Speed Dials for system wide or individual use.

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.

6. IP Telephony, Hard Phones & Soft Phones

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 (iPhone Manager Pro) 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.

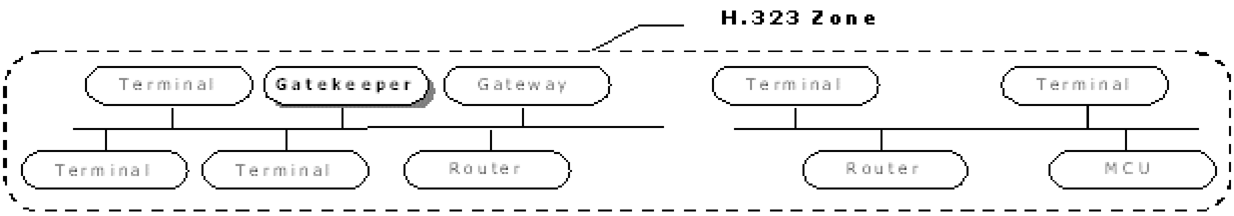
When making use of IP Extensions, quality of service should not be ignored. In situations where more than eight IP extensions are required or where LAN Bandwidth is limited, a quality of service capable LAN switch, such as the Avaya P130, should be used. For more information about implementing Voice over IP, refer to Appendix B.

Gateways, Gatekeepers and H.323 - Technology Overview

H.323 Architecture comprises of four logical components

- Terminals are H.323 devices that can support Audio, Video and Data calls in any combination
- Gateways 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, terminals, 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 terminal 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 terminal establish the call via a Gateway.



The design of IP Telephony systems has been driven with open standards in mind. IP Phones, Gateways & 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 Hardphones

4602 IP Hardphone

The 4602 supports the following features:

- 9 Fixed Feature Keys: Conference, Transfer, Drop, Hold, Redial, Mute, Volume up & down, Speaker, Voice Mail.
- 2 X 24 Character based Eurofont Display.
- Message Waiting Indicator.
- Call Monitor Speaker.
- G.711, G.729a/B Voice CODECs.
- QoS Options of - UDP Port Selection, DiffServ and 802.1p/B (VLAN).
- Single 10/100 BaseT Ethernet port.
- Support for Simple Network management Protocol (SNMP).
- Hearing Aid Compatible.
- Microsoft NetMeeting Compatible.
- IP Address Assignment - DHCP Client or Statically Configured.
- Downloadable firmware for future upgrades.
- Wall Mountable with included desk/wall mount stand.
- Avaya Grey Color for all markets.

IP Office also supports the 4602SW, which includes all of the above features plus an integrated Ethernet switch for PC connection.

- Integrated Full Duplex 10/100 BaseT Ethernet Switched ports for connection of PC.
- Auto-negotiation provided separately for each port.
- 802.3 Flow Control.
- Phone has priority over PC port at all times.

4606 IP Hardphone

The 4606 supports the following features:

- 6 Programmable Feature Buttons With LED.
- 5 Fixed Feature Keys: Speaker, Mute, Hold, Volume Up & Down.
- 3 Fixed Feature Keys Below The Display: 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.
- Optional Integrated Ethernet Repeater Hub – for Connecting PC To Phone.
- Hearing Aid Compatible.
- Microsoft NetMeeting 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.

4612 IP Hardphone

The 4612 supports all of the features of the 4606 with the following differences:

- 12 Programmable keys with LED.
- 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.
- 2x24 Character Display.

4620/4620SW IP Hardphone

In addition to the features of the 4602/4602SW, the 4620/4620SW supports the following:

- 24 Programmable Feature Keys (presented in 2 pages of 12).
- Automatically labeled from the system (no paper labels).
- 6 Fixed Feature Keys: Speaker, Mute, Hold, Headset and Volume Up/Down.
- Large graphical gray-scale display (168 x 132 dots).
- 5 Fixed Feature Keys below the display: Conference, Transfer, Hold, redial and Drop.
- 4 Embedded applications: Speed Dial, Call Log, Web Browser (WAP/WML), Options.
- Full-duplex speaker phone with acoustic cavity for improved sound quality.
- Feature Key Module (FKM) interface jack for use with the EU24, 24 button expansion module (support for this module will be available in a later release of IP office).
- 7 Position adjustable desk stand/wall mount stand.
- Infrared (IrDA) port.
- Built-in headset jack.
- Multiple language support built-in: English, French, Italian, Spanish & KataKana.
- 8 Personalized ring patterns.

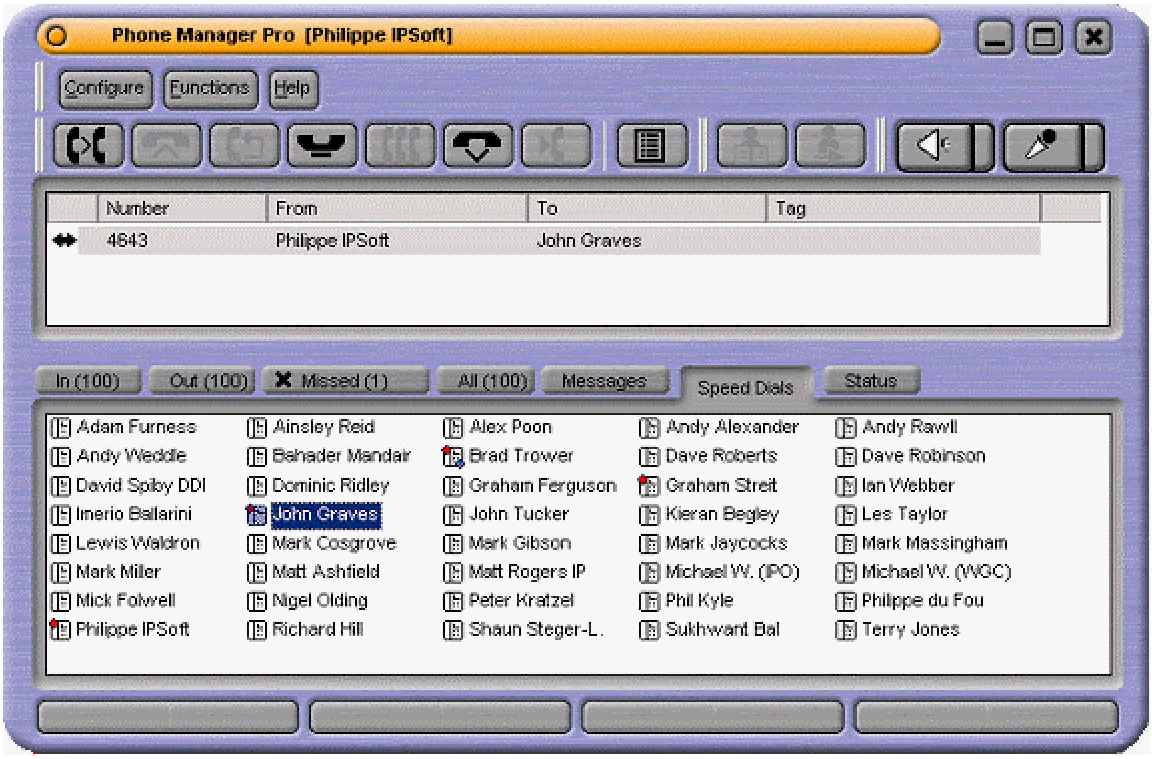
4624 IP Hardphone

The 4624 supports all of the features of the 4606 with the following differences;

- 24 Programmable Feature Keys With LED.
- 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.

IP Softphone (iPhone Manager Pro)

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



As with Phone Manager Pro the iPhone Manager Pro offers the same GUI interface for the user to take control of making and receiving telephone calls. Like Phone Manager Pro, iPhone Manager Pro communicates with the IP Office system unit via the LAN. The difference is that there is no physical terminal and conversation actually takes place via the PC's soundcard.

The physical set up must include a headset/microphone connected to the PC's soundcard or USB port.

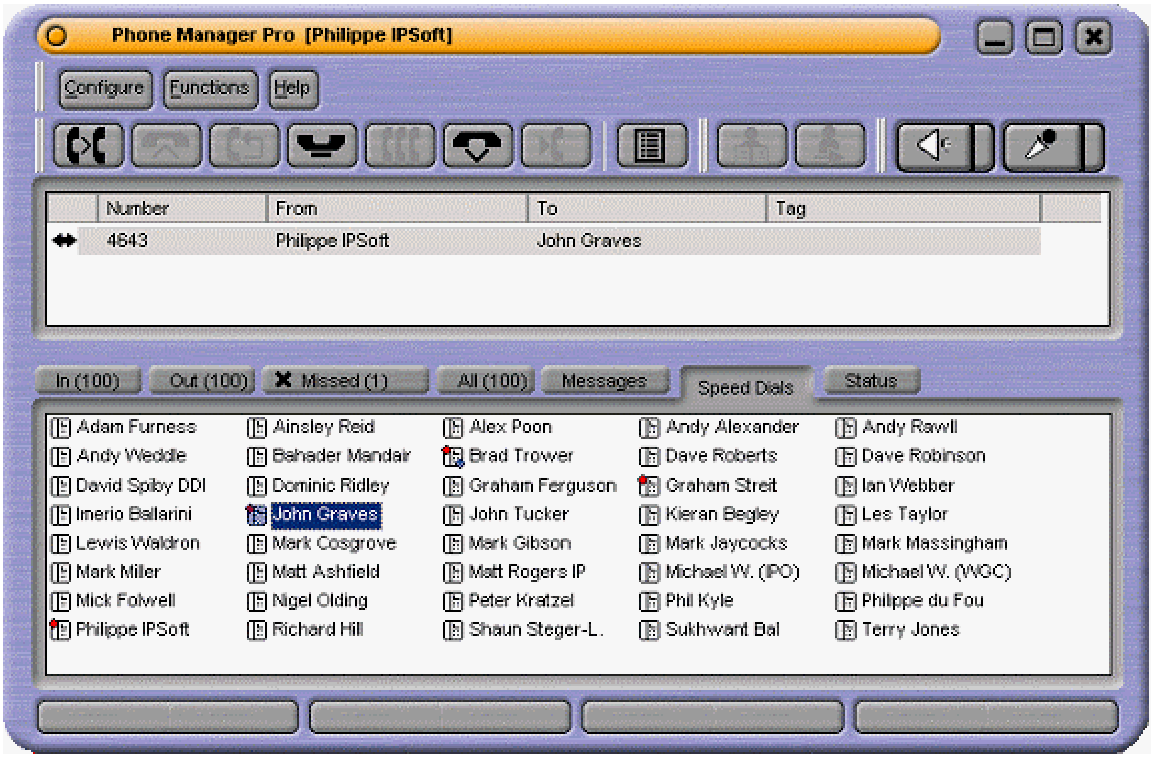
iPhone Manager Pro has the significant advantage for mobile users with remote access to the LAN, providing 'a phone within their laptop' with all the features available as in the office.

Notes:

- iPhone Manager Pro 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
- iPhone Manager Pro supports QoS in the form of DiffServ for both Windows XP and Windows 2000 when used in SoftPhone mode.

IP Softphone Used as a Wireless Deskset

IP Office's Phone Manager Pro can be configured to operate as an IP Softphone – 'iPhone Manager Pro' by use of a license key. IF this application is loaded onto a laptop and the laptop has a wireless LAN (Wi-Fi) card installed then this combination may be referred to as a Wireless Deskset.



As with Phone Manager Pro, the iPhone Manager Pro offers the same GUI interface for the user to take control of making and receiving telephone calls. iPhone Manager Pro communicates with the IP Office system unit via the wireless LAN. There is no physical terminal and conversation actually takes place via the laptop's soundcard.

The physical set up must include a headset/microphone connected to the laptop’s soundcard or USB port.

iPhone Manager Pro used as a wireless dekset provides 'a phone within their laptop' with all the features available as in the office.

Notes:

- iPhone Manager Pro requires on a multi-media Laptop 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 128 MB RAM minimum
- iPhone Manager Pro supports QoS in the form of DiffServ for both Windows XP and Windows 2000 when used in SoftPhone mode. However, when used with a wireless card Avaya does not have a built in QoS algorithm so the maximum number of wireless desksets that can be in use within a given access point may not exceed 3.

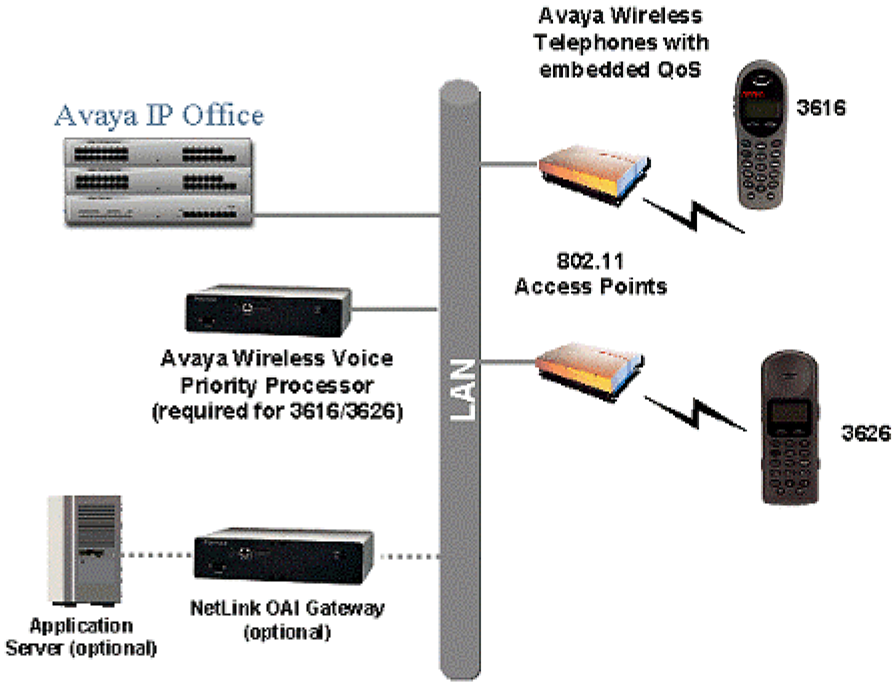
Wireless VoIP

Overview of Wireless VoIP

Leveraging the reliable technology from SpectraLink, a leader in wireless voice solutions for the workplace, 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 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 handsets 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	FH/DS	SVP Certified	Field Verification	Calls per Access Point
Alvarion	BreezeNET Pro 11 Series	FH	4.4.2 or 5.0.103		3
Cisco	Aironet 340	DS	10.13	11.03, 11.07, 1.10t	6
Cisco	Aironet 350	DS	11.03	11.07, 11.01t	7
Cisco	Aironet 4500 & 4800 Turbo DS	DS	8.12 & 8.24	8.55	5
Cisco	Aironet 3500	FH	8.12	8.24	3
Proxim	Orinoco AP1000	DS	D3.78S6 3.83	7.4a	5
Symbol	Spectrum 24 FH	FH	4.01-S2	4.02-12	3
Telxon	Air-I/O 802FH UAP	FH	8.12	8.24	3
Telxon	802 DS & 802 DS 11	DS	8.12	8.24	5
Avaya	AP-1, AP-2	DS		3.83, 3.92	6
Avaya	AP-3	DS		1.4 (v222)	7
Cisco	Aironet 1200	DS		11.40t	7
Enterasys	Roamabout AP2000	DS		V6.02	6
Intermec	Mobile LAN Access 2100, 2101, 2102	DS		1.51 or later	6
LXE	6250 Access Point	DS		3.83	6
Proxim	AP 2000	DS		7.4, 1.3	6
Symbol	Spectrum 24 DS	DS		2.21-23, 2.51-21, 3.50-18	6
Teklogix	9150 Wireless Gateway	DS		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. Frequency Hopping (FH) Avaya wireless telephones support 1Mbps data rate only. 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.

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.

3616 Executive Wireless Phone

The Avaya 3616 IP Wireless Telephone is designed for more general enterprise applications and uses a compact, cell phone-like form factor.

The 3616 supports the following features:

- Perfect for busy office environments.
- Lightweight innovative design .
- Simple to use.
- 802.11b standard-compatible.
- Radio Frequency 2.4000 – 2.835 GHz.
- Transmission type Direct Sequence Spread Spectrum (DSSS).
- FCC certification Part 15.247.
- Management of handsets via DCHP 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.

3626 Ruggedized Wireless Phone

The Avaya 3626 Wireless Telephone is designed specifically for use in commercial workplace applications. It is extremely durable and has no moving parts, no external antenna, and no complex configuration menus. The handset has a rugged, monolithic design that gives users a large earpiece to provide comfort and seal out background noise.

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.

IP Telephony Features

- **Gatekeeper**
  The Integral IP Office gatekeeper allows the registration of up to 360 IP extensions, less the number of traditional analog and digital telephones.
- **Gateway**
  IP Office fitted with a Voice Compression Module 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 quicker on answering a call.
- **Local Hold Music**
  This facility allows the choice of providing music from the IP Office, which obviously uses bandwidth on the LAN, or allowing the IP endpoint to generate its own.
- **Local Tones**
  This facility allows the choice of providing call supervision tones from the IP Office, which obviously uses bandwidth on the LAN, or allowing the IP endpoint to generate its own.
- **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 Over IP**
  Fax Over IP allows incoming and outgoing fax calls to be routed over an IP Network using a fax machine connected to a remote IP Office or Avaya Communication Manager. This feature is NOT T.38 faxing, it is a proprietary Avaya fax over IP protocol that only works between IP Offices or Avaya Communication Manager

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.

Connection to the Public Network

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 Basic Rate (ETSI CTR3) - IP400 Quad BRI
- North American T1 - IP400 Office PRI T1
- North American Primary Rate Interface - IP400 Office PRI T1
- Analog Trunks
- E1-R2 Trunks

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 service over Primary Rate Trunks.

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 base unit or as a stackable 16-port module.

The first two trunks on the stackable 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 module. See Section 2 for detail. 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 Co-Ax network connections. Each card provides 30 channels that can be configured for MFC, Pulse or DTMF Dialing dependent on the requirements of the network.

Traditional Private Voice Networking

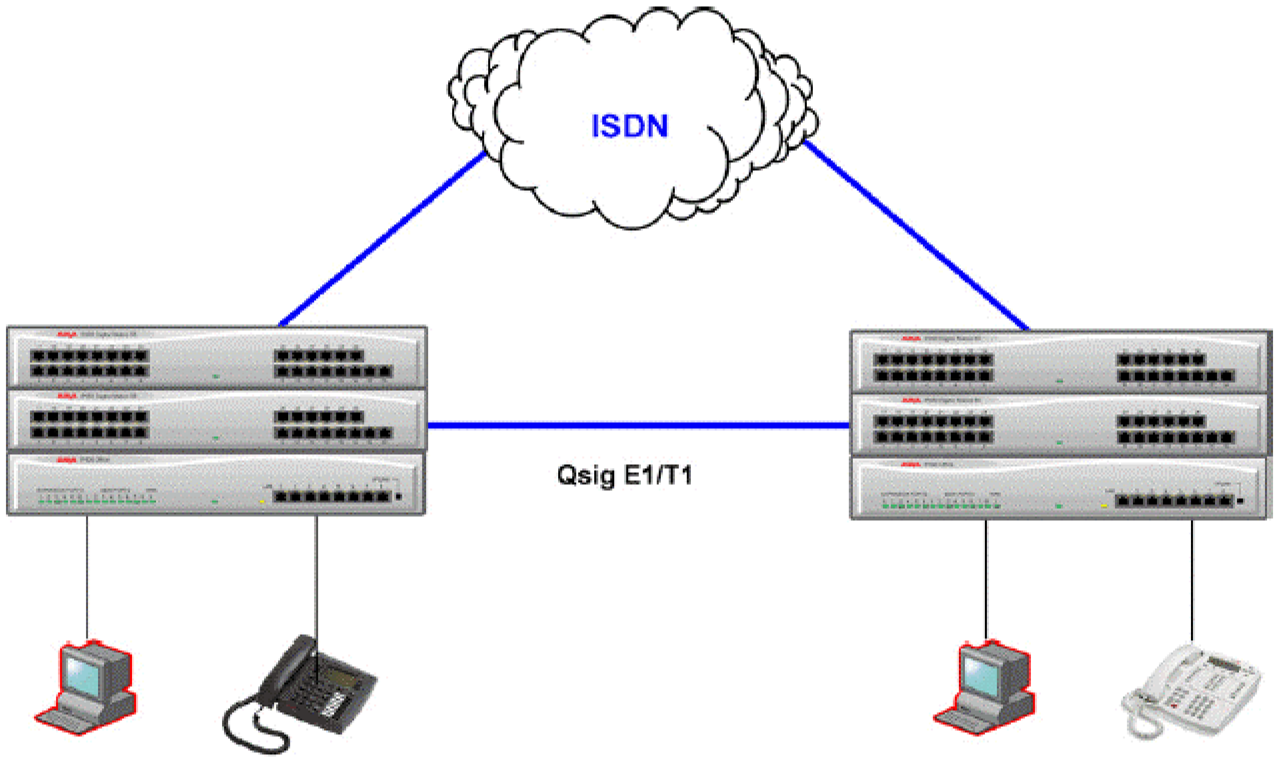
Private voice networks can be constructed utilizing structured leased line circuits (E1 or T1) or alternatively by establishing permanently connected 'B' channels between IP Office systems. Each channel within the Primary Rate interface can provide a single voice or 64K/56K data call.

If leased line circuits are used within a private networking scenario these PRI interfaces are typically configured in software to employ QSig signaling

QSig provides feature transparency between PBX's 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:** ETS300 237/238.
- **Message Waiting:** EN301 260/255.



Traditional voice networking

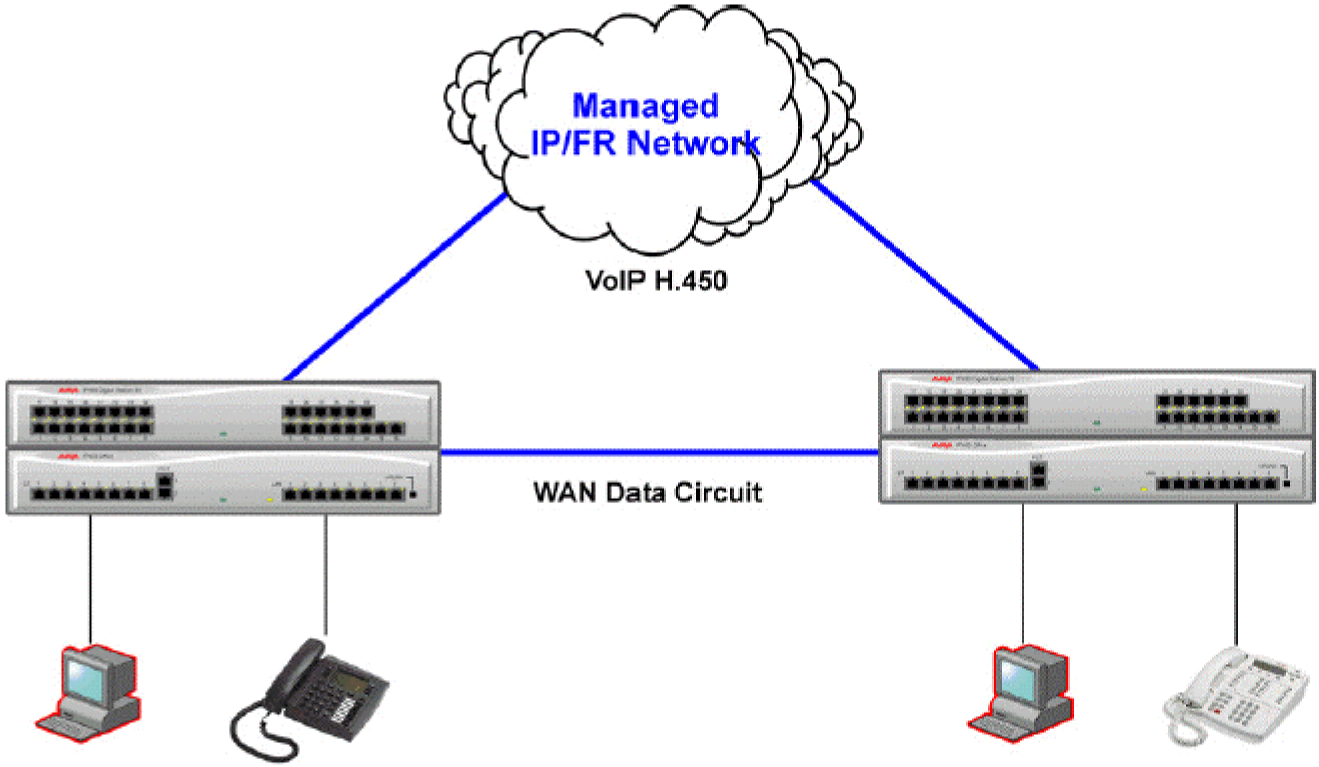
Packet Based Voice Networking

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 who 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 2 to 60 VoIP calls (see VCM). 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' 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.

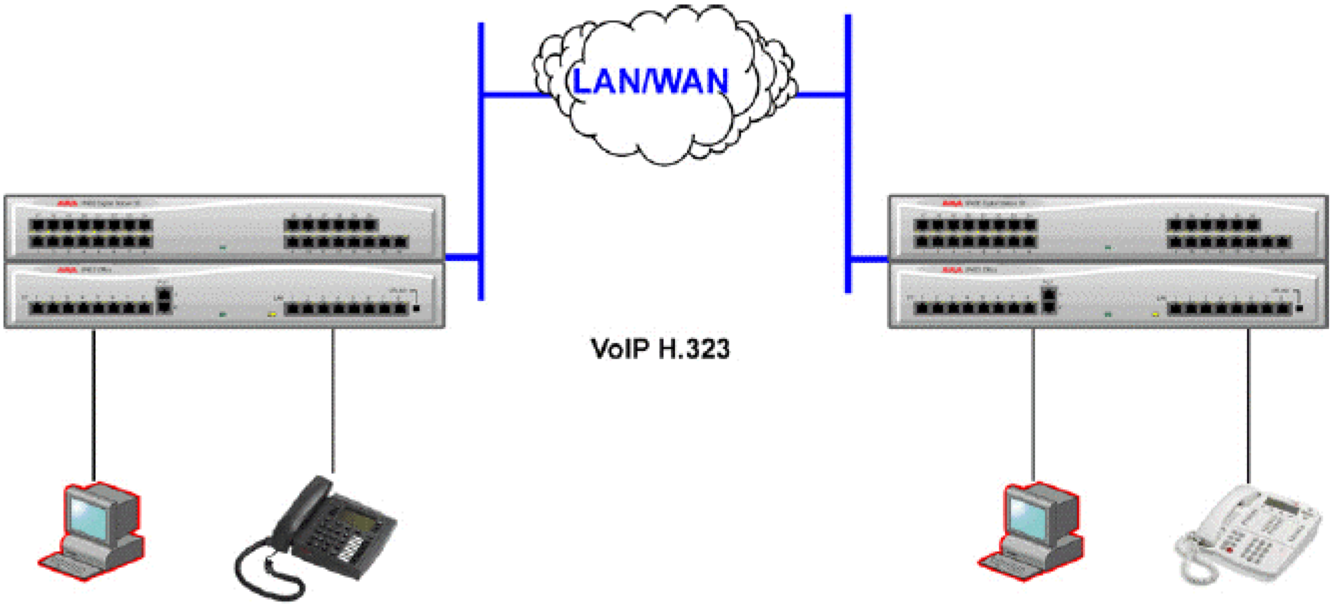
IP VPNs have some distinct advantages over Frame Relay networks: access bandwidth need not be pre-allocated between sites like Frame Relay's PVCs, they are generally cheaper and their global reach is normally greater. Access to the IP VPN is via one of IP Office's LAN ports.

- **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. Avaya's Cajun switches are the recommended option for this.



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 IP networks

Supplementary services within an IP environment are provided via Q.931 and H323. 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.323 are:

- 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).
- 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

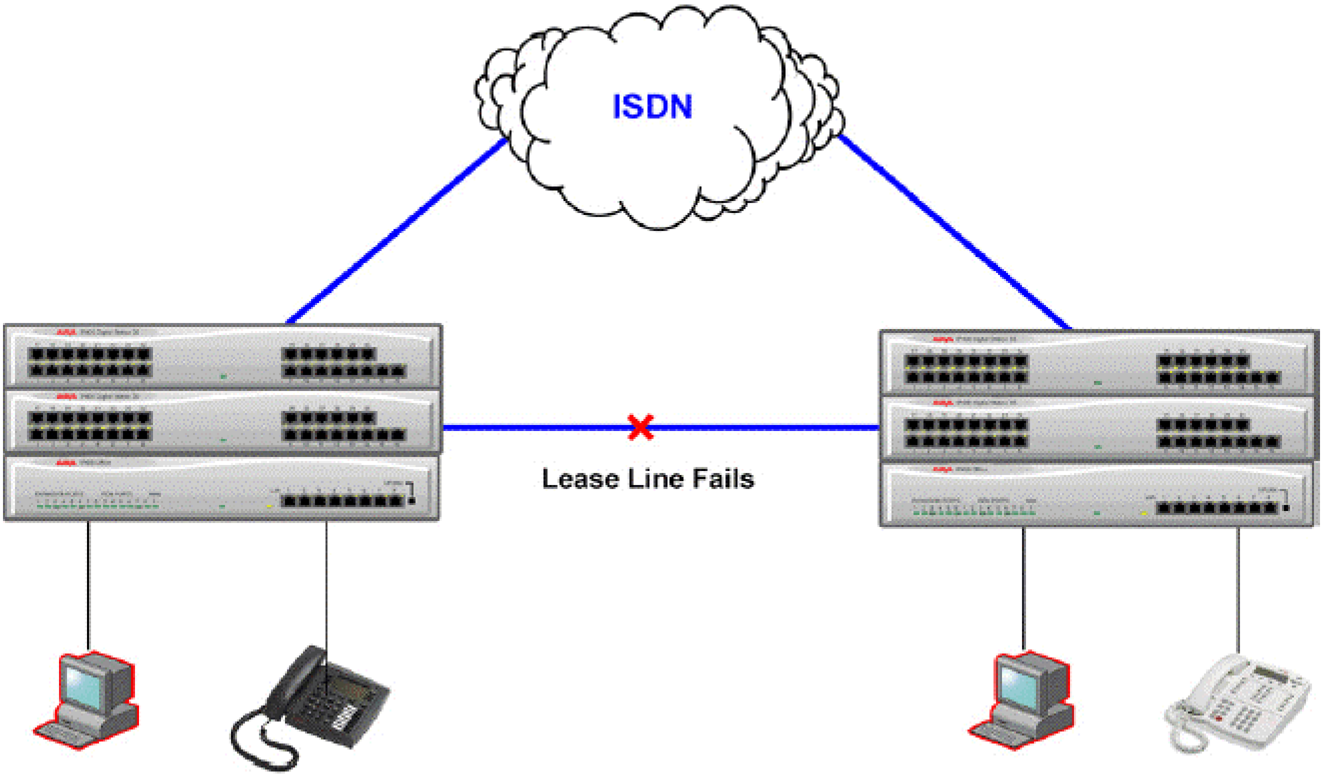
Least Cost Routing (LCR)

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 whilst preserving the data call, albeit with reduced bandwidth.

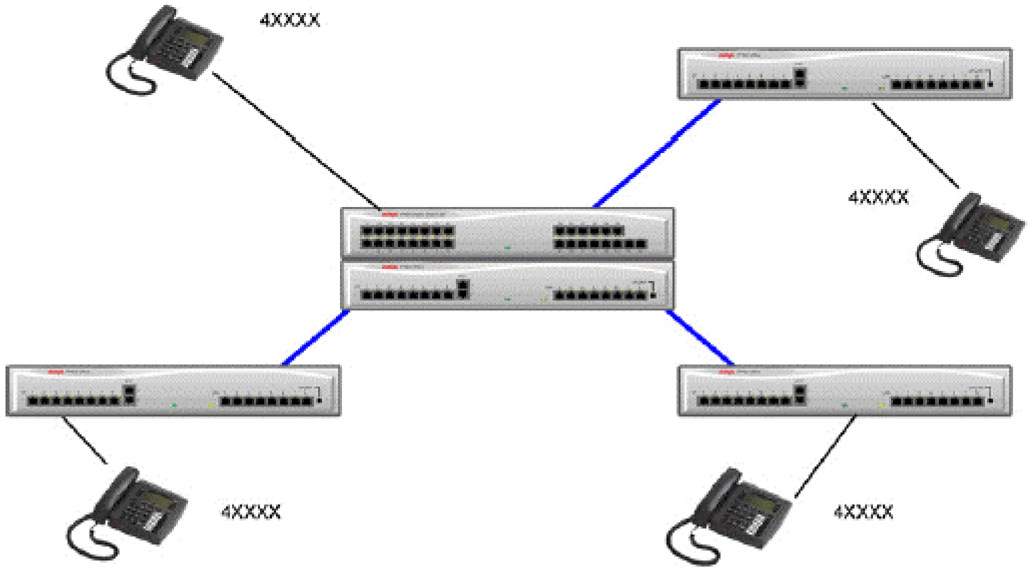


Alternate Call Routing

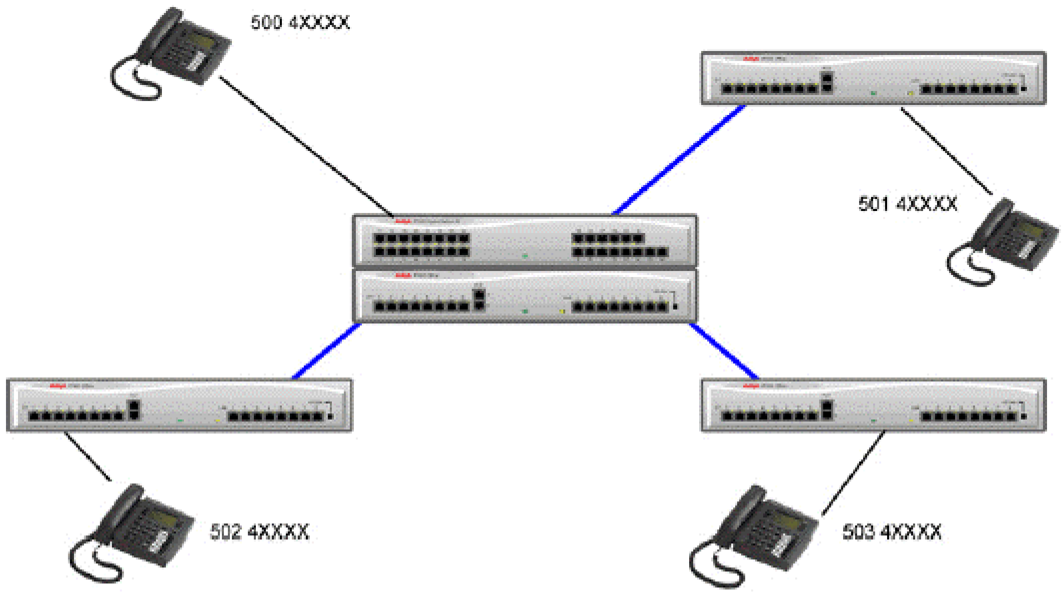
Network Numbering Schemes

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 whilst 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 IP401, IP403 and IP406 platforms incorporate an integral dual speed Ethernet hub (10/100 Base-T), while the IP Office - Small Office Edition and the IP412 support layer 2 and Layer 3 Ethernet Switching respectively, thereby allowing all users to easily utilize the data networking capabilities of IP Office.

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.

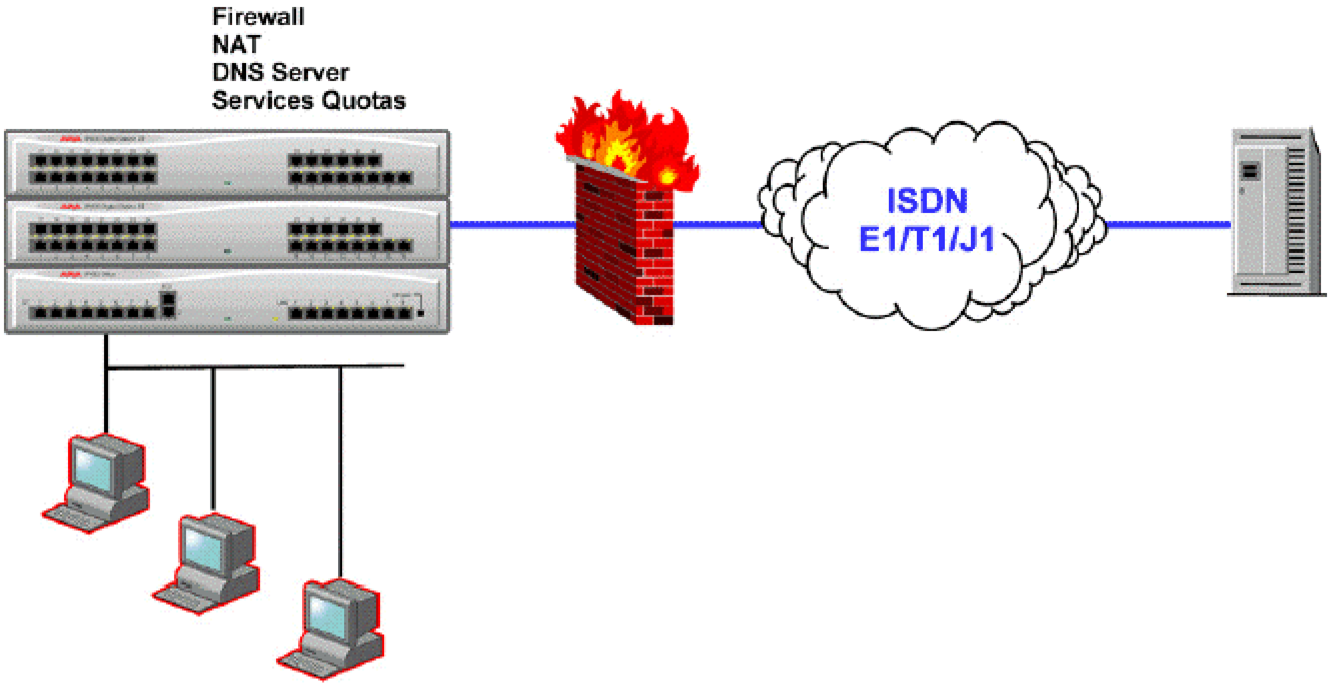
Internet Access

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, whilst 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 freephone services.

In addition to remote access from Terminal Adaptors, an optional dual 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 Quad Analog trunk modules, ATM16 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.

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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 Hub

- **401, 403 & 406 Only.**

Each IP Office, with the exception of the IP412 Office, is equipped with an integral 10/100 Auto-sensing hub. The IP403 Office and IP406 Office offer 8 hub ports, and the IP401 Compact Office offers either 4 or 8 dependent on the version. If more than eight computers and servers need to be supported the IP office can be connected to any commercially available external hub or switch. All hub ports on the IP Office are connected onto the same LAN segment regardless of their operational speed (10/100). This allows all devices, PCs and Servers to communicate with each other without the need for any configuration.

Integral 10/100 Mbit Layer 2 Ethernet Switch

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

All the IP Office - Small Office Edition platforms provide a four 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 intended for connecting to external xDSL or Cable Modems. This fifth port is Layer 3 switched to the other four ports as described below.

Integral 10/100 Mbit Layer 3 Ethernet Switch

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

In place of the integral HUB ports available on the other platforms, the IP412 Office supports a two-port Ethernet switch. Both of these switched ports have their own IP address. 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 Ethernet switch and its 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 Office 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 whilst 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 whilst 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.

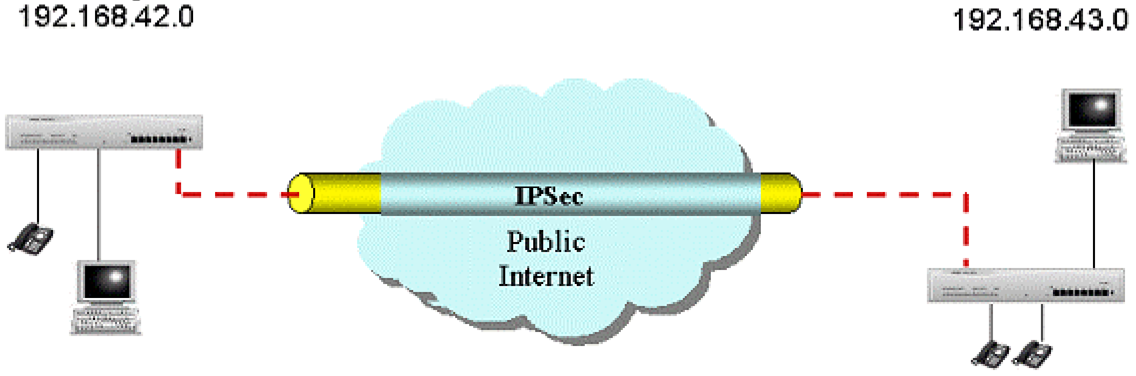
Remote Access Server (RAS)

IP Office provides RAS functionality allowing external users to dial in to the local area network from modems, terminal 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 whilst 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" terminal 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.

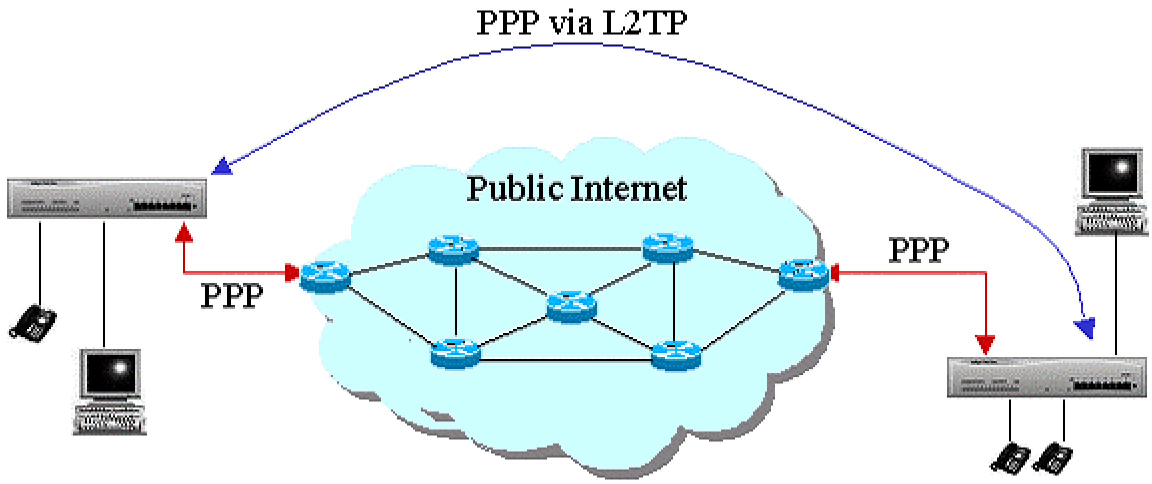
IPSec Tunneling



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.

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.

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.

9. The Applications

Introduction to IP Office Applications

One of the key strengths of the IP Office range is its level of application support. The applications on IP Office can be broadly divided into three categories: Personal Productivity (such as Voicemail, Integrated Messaging, Soft Phones and On-Site Mobility), Business efficiency (such as Operator Console and Automated Attendant) and Customer Relationship Management (CRM).

Covered in this section are:

- SoftConsole
- Voicemail
    - Voicemail Embedded
    - Voicemail Lite
    - Voicemail Pro
    - Centralized Intuity Audix
- Phone Manager
- Audio Conferencing
- Conferencing Center
- Cordless Solutions

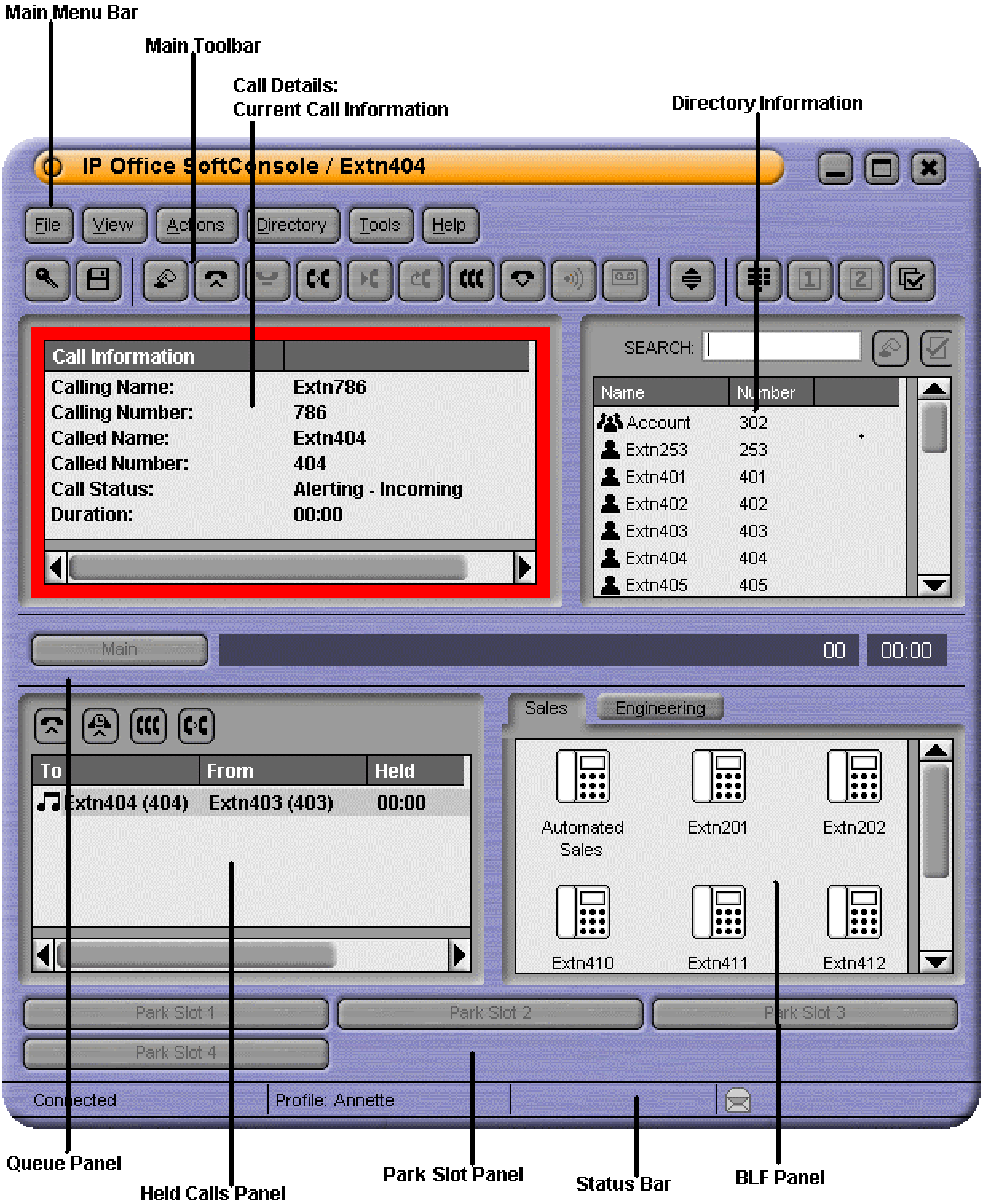
For Contact Center and CRM applications, please see IP Office Contact Center/CRM Solutions Product Overview.

For system management and reporting tools, please see Management Utilities.

SoftConsole

SoftConsole

The PC based Windows Operator Console 'SoftConsole' has been specifically designed to benefit businesses through improved operator service. Deployment of the SoftConsole provides 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.



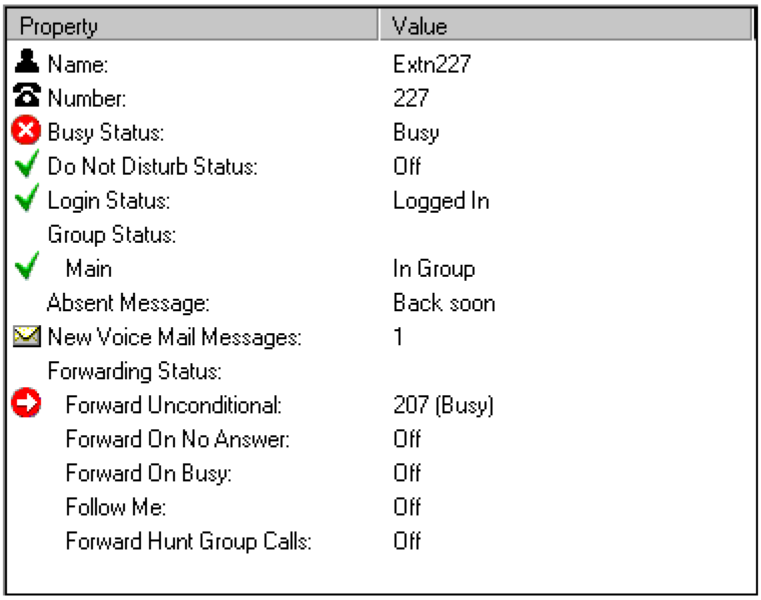
SoftConsole has been designed to be easy to use, whilst offering a look and feel, which will appeal to experienced and novice operators alike.

The console 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.

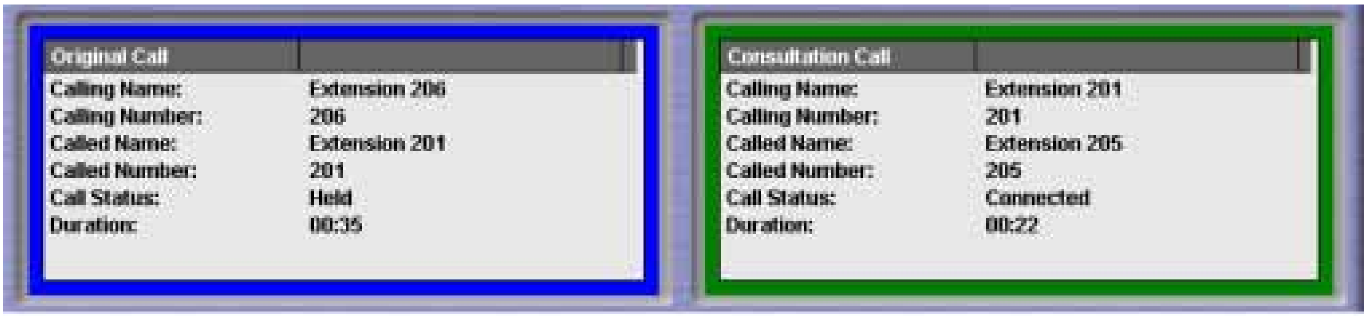
- **Call Details Panel**
  The call details panel on the left shows details of the current call which will include the following information:
  - **Calling Name**
    The system directory name associated with the calling number.
  - **Calling Number**
    The telephone number of the call originator.
  - **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.

- **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).



Property	Value
Name:	Extn227
Number:	227
Busy Status:	Busy
Do Not Disturb Status:	Off
Login Status:	Logged In
Group Status:	
Main	In Group
Absent Message:	Back soon
New Voice Mail Messages:	1
Forwarding Status:	
Forward Unconditional:	207 (Busy)
Forward On No Answer:	Off
Forward On Busy:	Off
Follow Me:	Off
Forward Hunt Group Calls:	Off

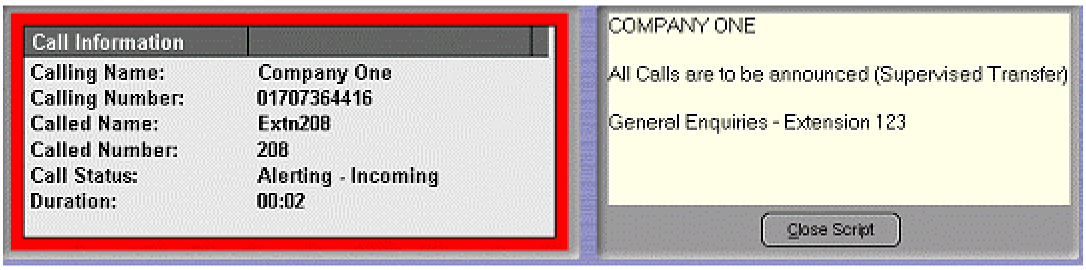
  - **Details about a consultation call**
    When operator wishes to carry out a supervised transfer.



Original Call	
Calling Name:	Extension 206
Calling Number:	206
Called Name:	Extension 201
Called Number:	201
Call Status:	Held
Duration:	00:35

Consultation Call	
Calling Name:	Extension 201
Calling Number:	201
Called Name:	Extension 205
Called Number:	205
Call Status:	Connected
Duration:	00:22

  - **Script file**
    When a script file has been configured for either the calling or called number. 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 file is displayed whenever a call is received for that company.

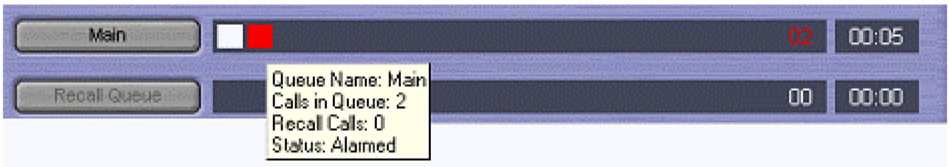


Call Information	
Calling Name:	Company One
Calling Number:	01707364416
Called Name:	Extn208
Called Number:	208
Call Status:	Alerting - Incoming
Duration:	00:02

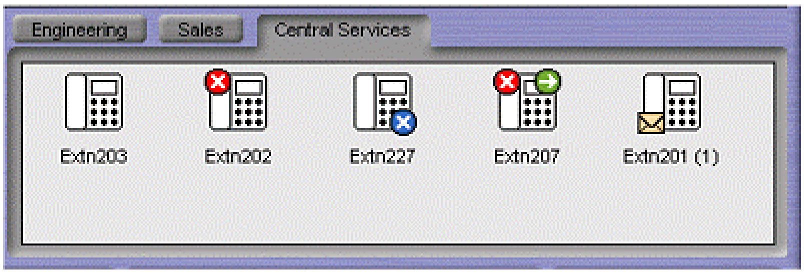
- **Conferencing**
  Within SoftConsole calls can be conferenced when held or a conference can be created through the two conference rooms:
  - **Conference Held Calls**
    An operator can conference calls that are in the Held Panel. All calls in the Held Panel will be conferenced.
  - **Conference 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.



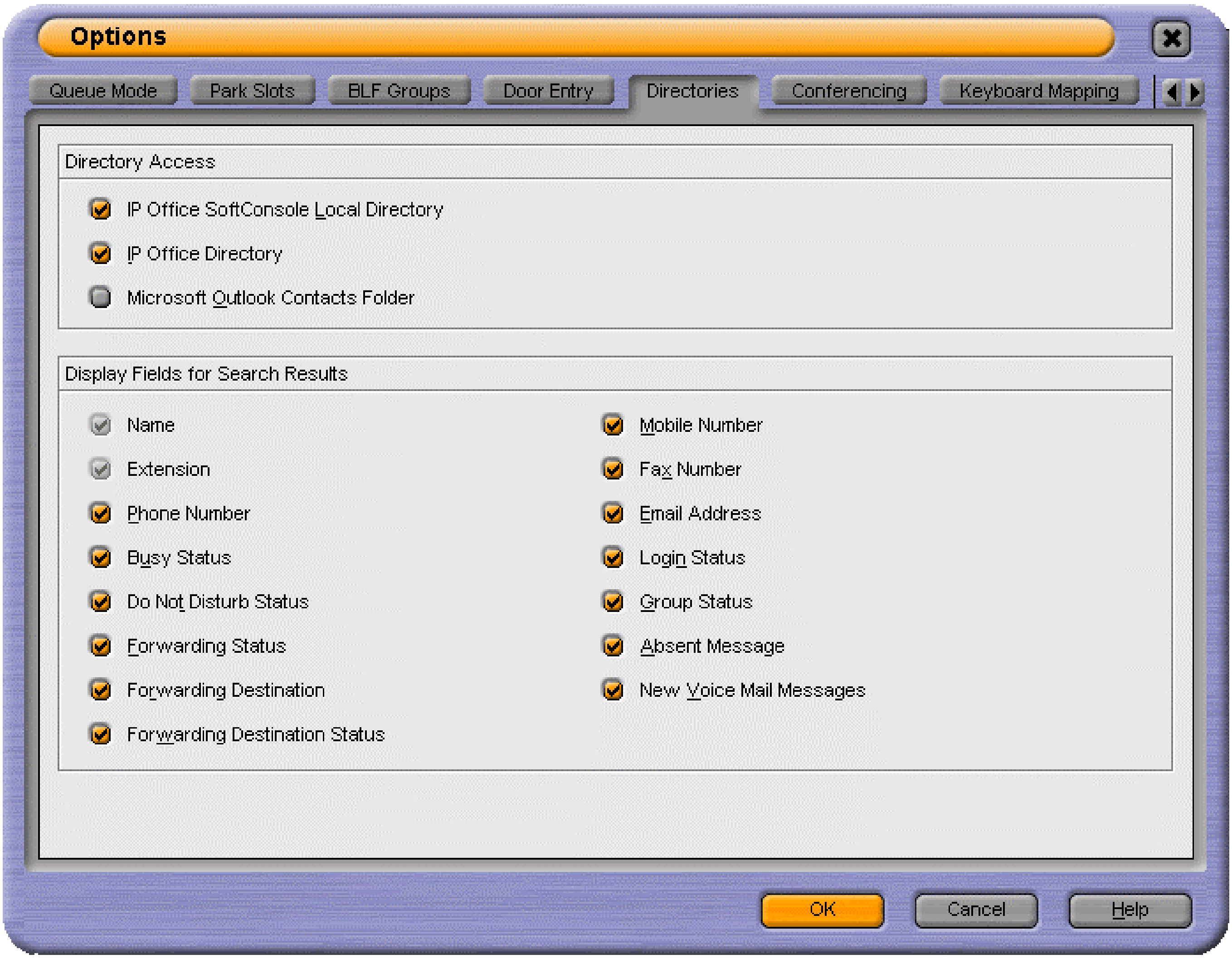
- **Held Calls Panel**
  The held call panel enables the operator to manage all calls held at the operator station. These will appear as a list in panel. The operator can perform the following the functions: Answer the highlighted held call, Answer the longest held call, Conference held calls (see conferencing 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 Configuration

SoftConsole has plethora of configurable options available to the operator to personalize the look and feel and tailor the usability specifically to each operator's 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 such as 20xx, 24xx, 64xx, 46xx and 44xx. These have differing display sizes, which means the administrator is able to tailor the call notes field according to the endpoints utilized.

SoftConsole PC Requirements

- IP Office switch software release 2.0 or higher
- Ethernet attached PC running Microsoft Windows 98/NT4/2000/XP operating systems, in conjunction with TCP/IP Networking.
- Minimum Pentium II processor 400Mhz or higher with 64MB RAM (or higher as specified by the Windows version) and 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).

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 whilst 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 IP Office's 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**
- **Voicemail Embedded** ***(IP401 and Small Office Edition Only)***
  - **with Auto Attendant** ***(IP Office - Small Office Edition only)***
- **Voicemail Pro networked with other voicemail systems**
- **Centralized Intuity Audix**

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 98, NT, 2000 or XP 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 Voicemail Embedded is the preferred option. Voicemail Embedded uses the Voice Compression Module (VCM) and can be used on the IP Office - Small Office Edition to provide an entry-level voicemail service and Auto Attendant – See IP Office - Small Office Edition section, or can be used on the IP401 Compact Office to provide an entry-level voicemail service.

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.

Centralized Intuity Audix

Where IP Office is deployed in Definity/Multi-Vantage/ACM Environments it may be desirable to utilize the Intuity Audix, connected to the Definity/Multi-Vantage/ACM, to provide voicemail services to IP Office users. Connectivity between IP Office and the Definity must be either a E1 or T1 circuit or an IP trunk running QSig services. In addition to the IP Office license Key (IP400 AUDIX RFA) that enables this service, further license keys may be required on the Definity/Multi-Vantage/ACM.

Voicemail Embedded

(IP401 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), Voicemail Embedded may be the preferred option to enable an entry-level voicemail service.

Voicemail Embedded uses Voice Compression Module and can be installed on the IP401 Compact Office to provide an entry-level voicemail service. See the Voicemail Feature Comparison table for functional details and the IP Office - Small Office Edition section for further details. A memory upgrade is required on IP401 Compact Office systems. IP Office – Small Office Edition uses its own Voice Compression resources and a PCMCIA card.

The maximum number of messages stored is only restricted by the memory capacity (currently about 3 hours on the IP401, on the IP Office - Small Office Edition the embedded Voicemail is provided on a 64MB memory card providing 10 hours recording time).

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 (31 days when running Intuity TUI). 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 whilst 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 Manager'. 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 whilst in the office or remotely when away from the office.

The Voicemail Pro Manager 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, whilst 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 Manager graphical user interface.
- Customizable voicemail services for individual business requirements.
- Personal Numbering.
- 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.
- Voice Forms/Questionnaire Mailboxes (Campaign Manager).
- Networked Messaging support for inter-working with other Avaya voicemail systems.
- 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.
- Comprehensive support of the Intuity Telephone User Interface.
- Recording of system prompts through the telephone handset or using multimedia facilities on a PC.
- Speaking Clock.
- 22 supported prompt languages: Chinese, 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.

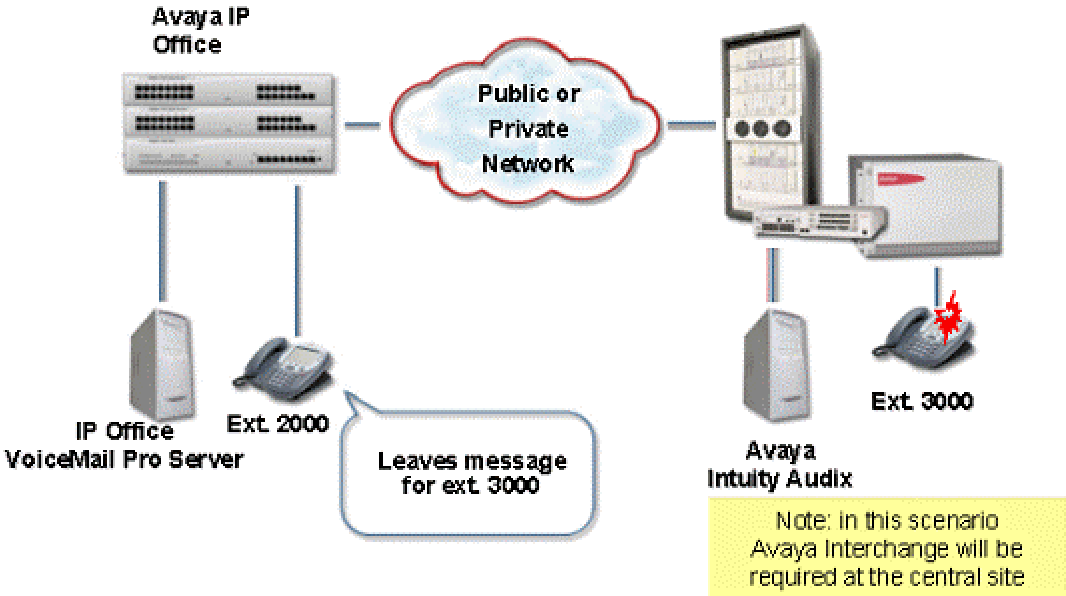
Further details on some of the Voicemail Pro Manager functionality listed above are described later in this section. Further information on Queue Announcements, Call Recording and Campaign Manager can be found in Compact Contact Center (CCC).

Networked Voicemail Environments – 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, whilst 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.



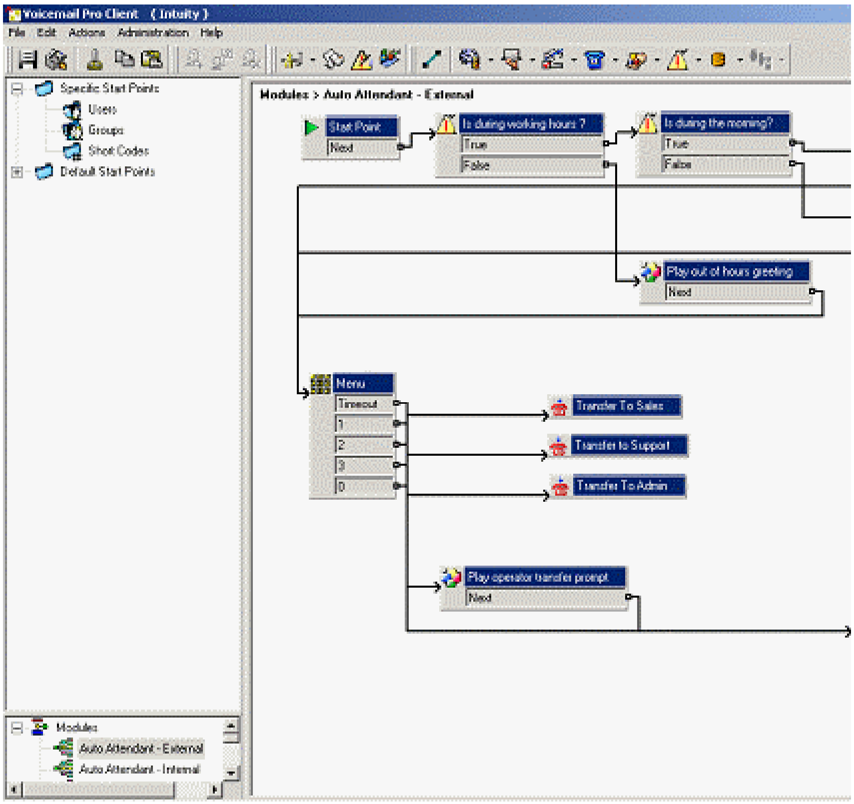
Auto Attendant

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, ...".



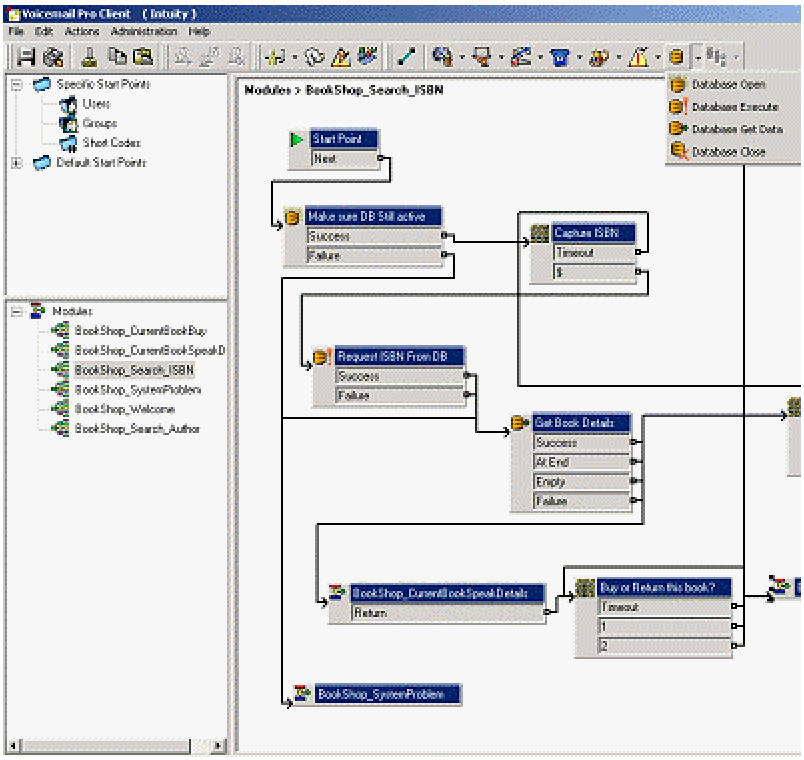
Auto-Attendant created using VoiceMail Pro Manager

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.

To 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 GUI are:

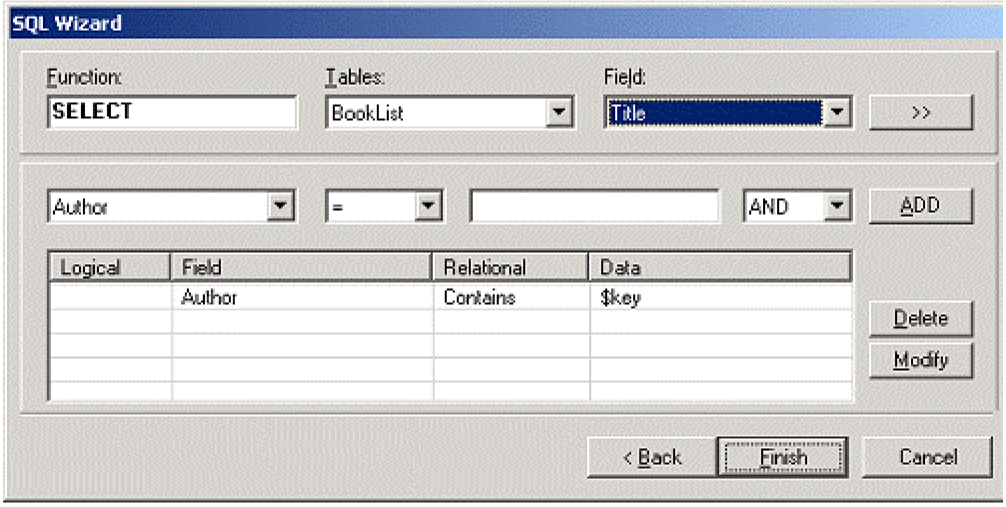
- **Database Open** – Opens a link to the required database. Multiple databases can be accessed during a call but only one database can be opened at one time.
- **Database Execute** – Provides the ability to enter a query on the opened database. The query can 'Select' data from the open database or can '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.

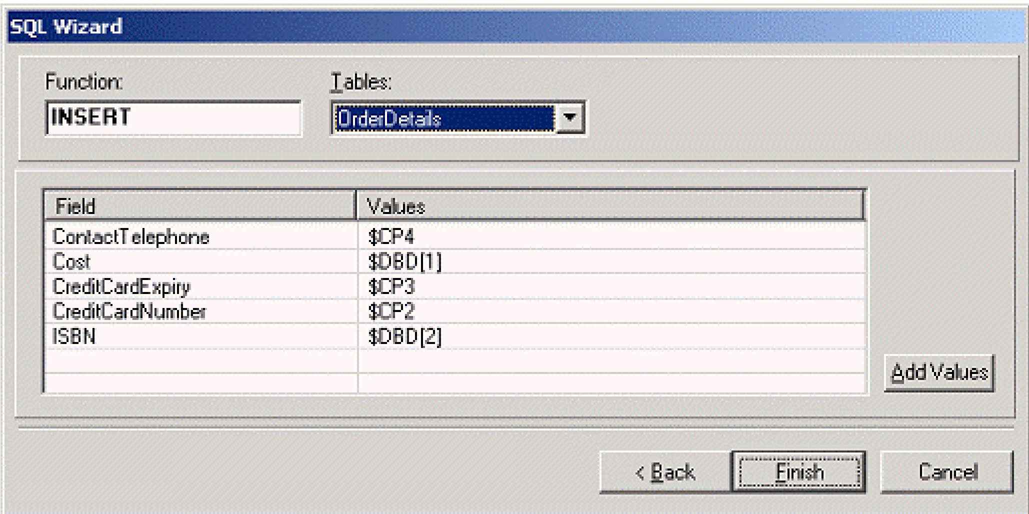
Interaction with the opened database is achieved through the use of Structured Query Language scripts (SQL). An administrator can enter SQL script directly into the 'specific' section of the Database Execute action. For administrators that are not familiar with SQL scripts a script can be automatically 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 GUI. 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.

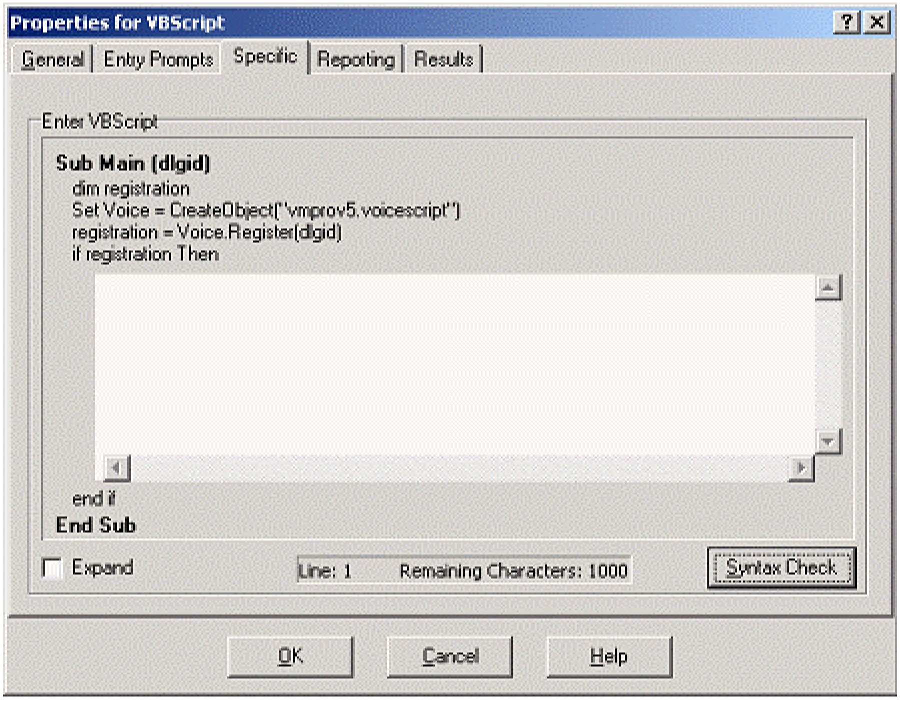
The languages currently supported by the Avaya TTS engine are:

- **Chinese**
- **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.

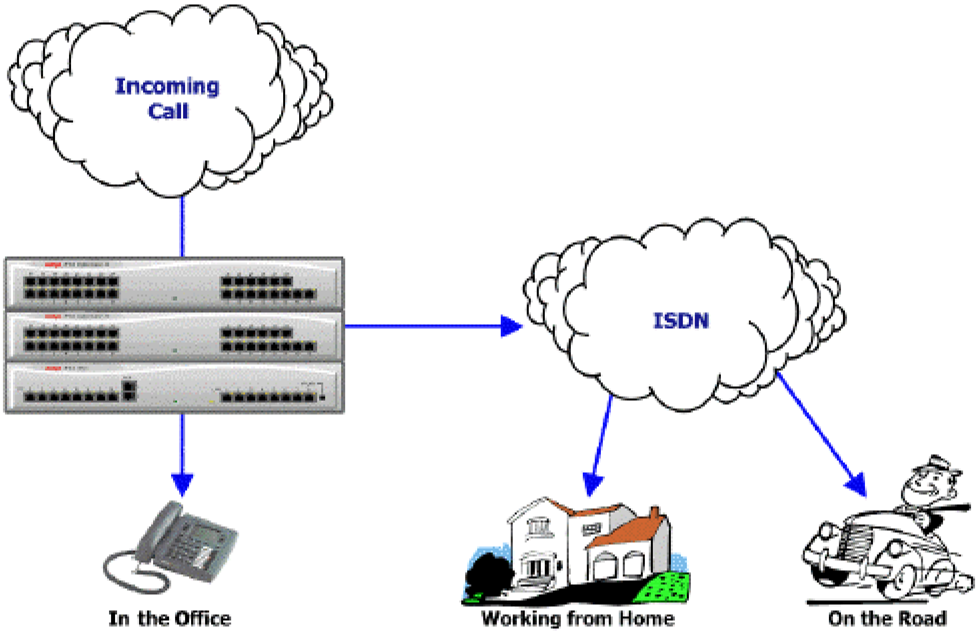


Diagram illustrating personal numbering

Extended Personal Greetings

In Intuity 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 the system 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:

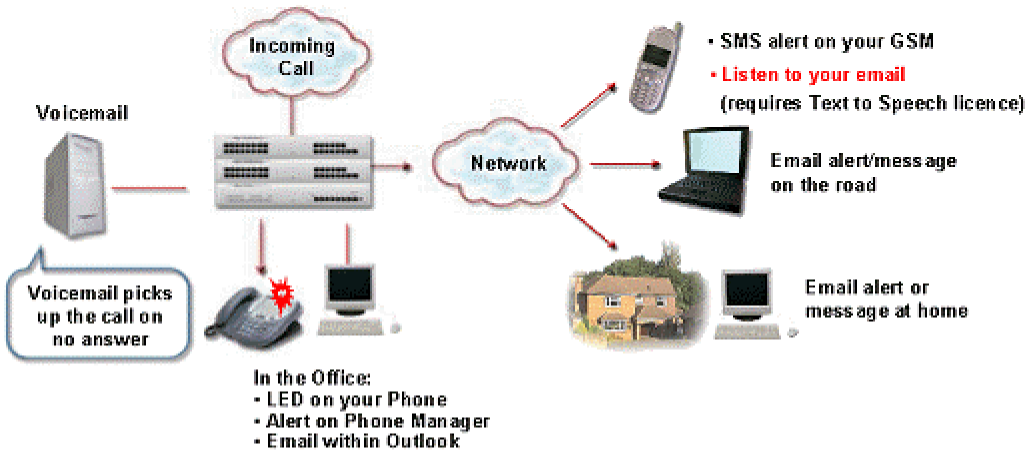
- 1st - Out of hours.
- 2nd - Busy/Engaged.
- 3rd - Internal/External greeting.
- 4th - No reply.

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

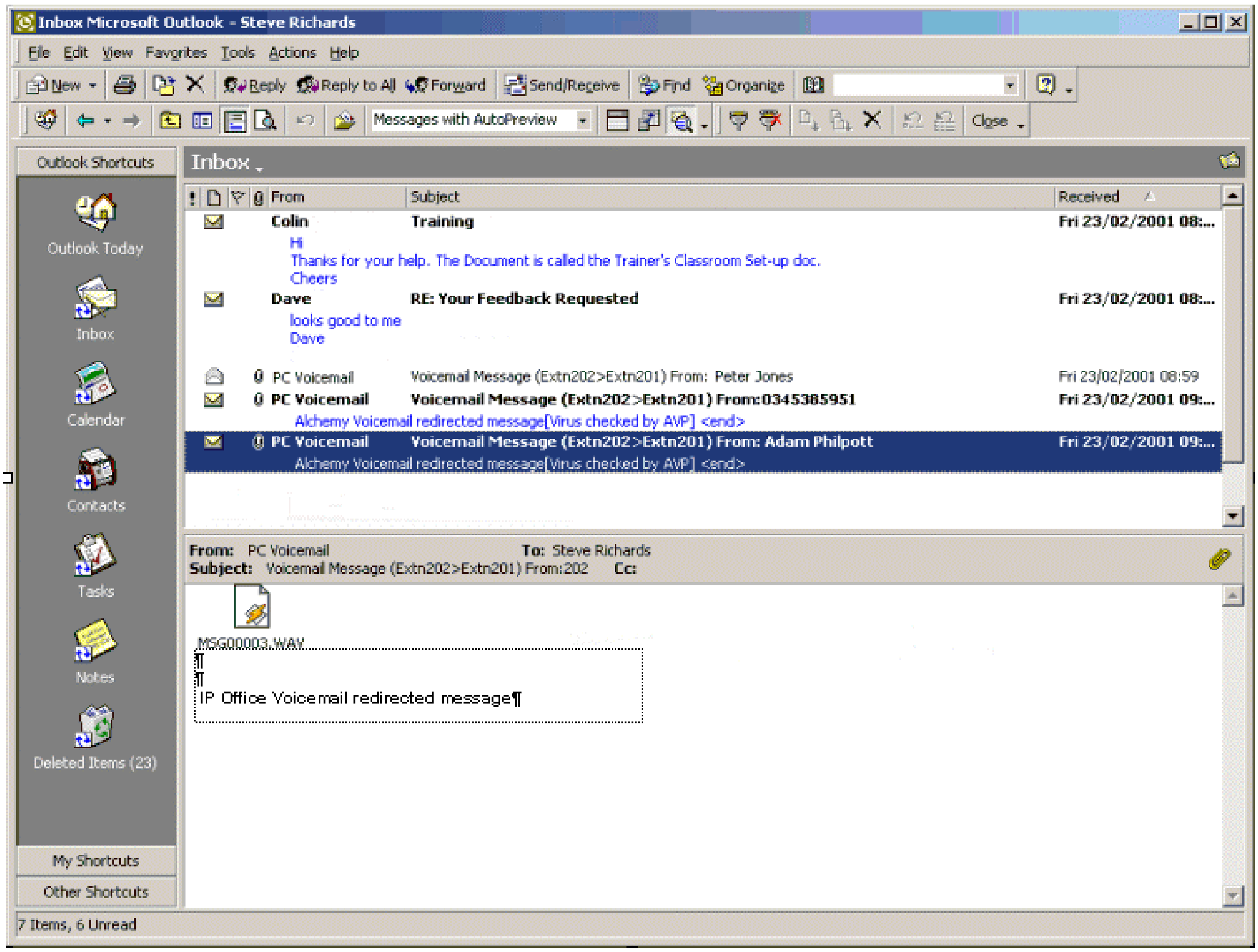
Interaction of Voicemail with Email Systems (Unified Mailbox)

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.

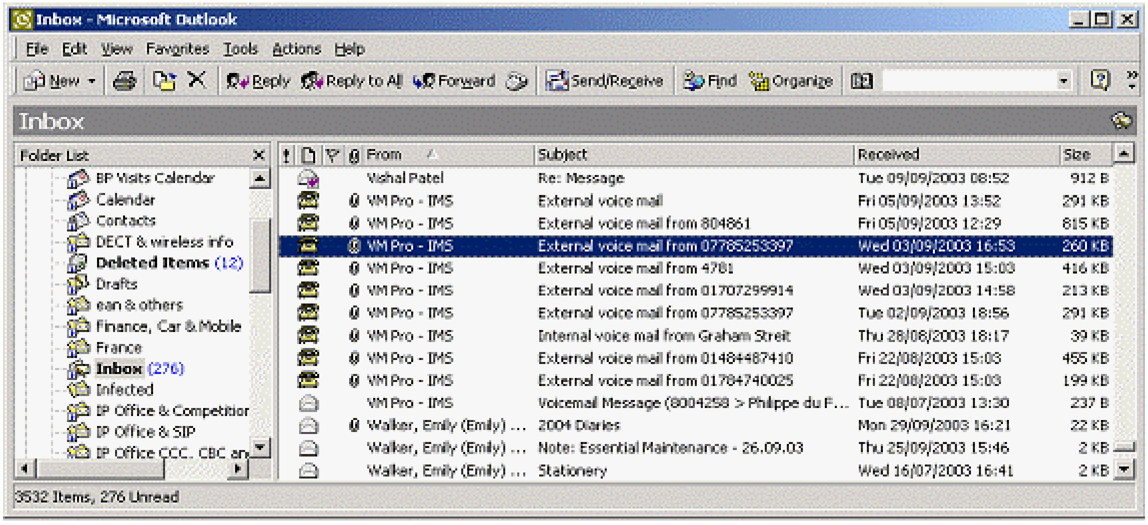
To enhance the support of Third Party Fax solutions, Voicemail Pro supports the automatic detection of incoming fax calls. Traditionally a dedicated telephone number will be provided for all incoming fax calls. In addition to, or as an alternative to, the Voicemail Pro 'Menu' action or a subscriber's 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 their own fax location number. If a voicemail box owner has set their own fax number then this number will be used instead of the default fax location. A voicemail box subscriber can set their own fax number through the telephone handset locally or remotely.

Integrated Messaging Pro (Microsoft Exchange only)

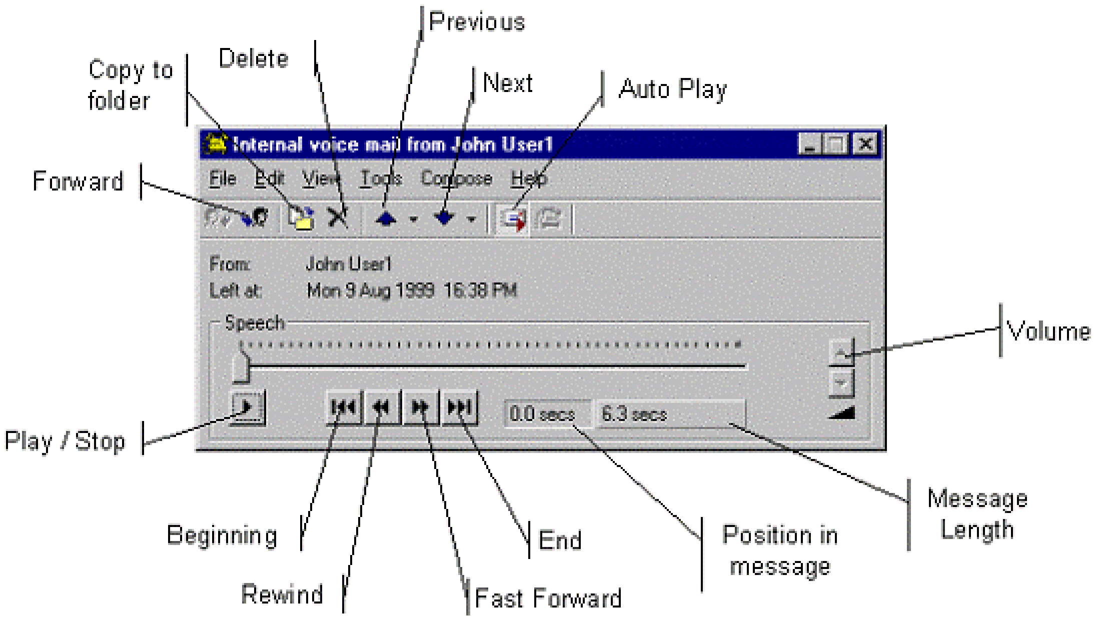
Integrated Messaging Pro allows increased productivity through easier management and prioritization of email and voicemail messages through one inbox. This optional application integrates IP Office Voicemail Pro and Microsoft Exchange email systems.

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 terminal, 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 terminal 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 terminal 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 whilst 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).

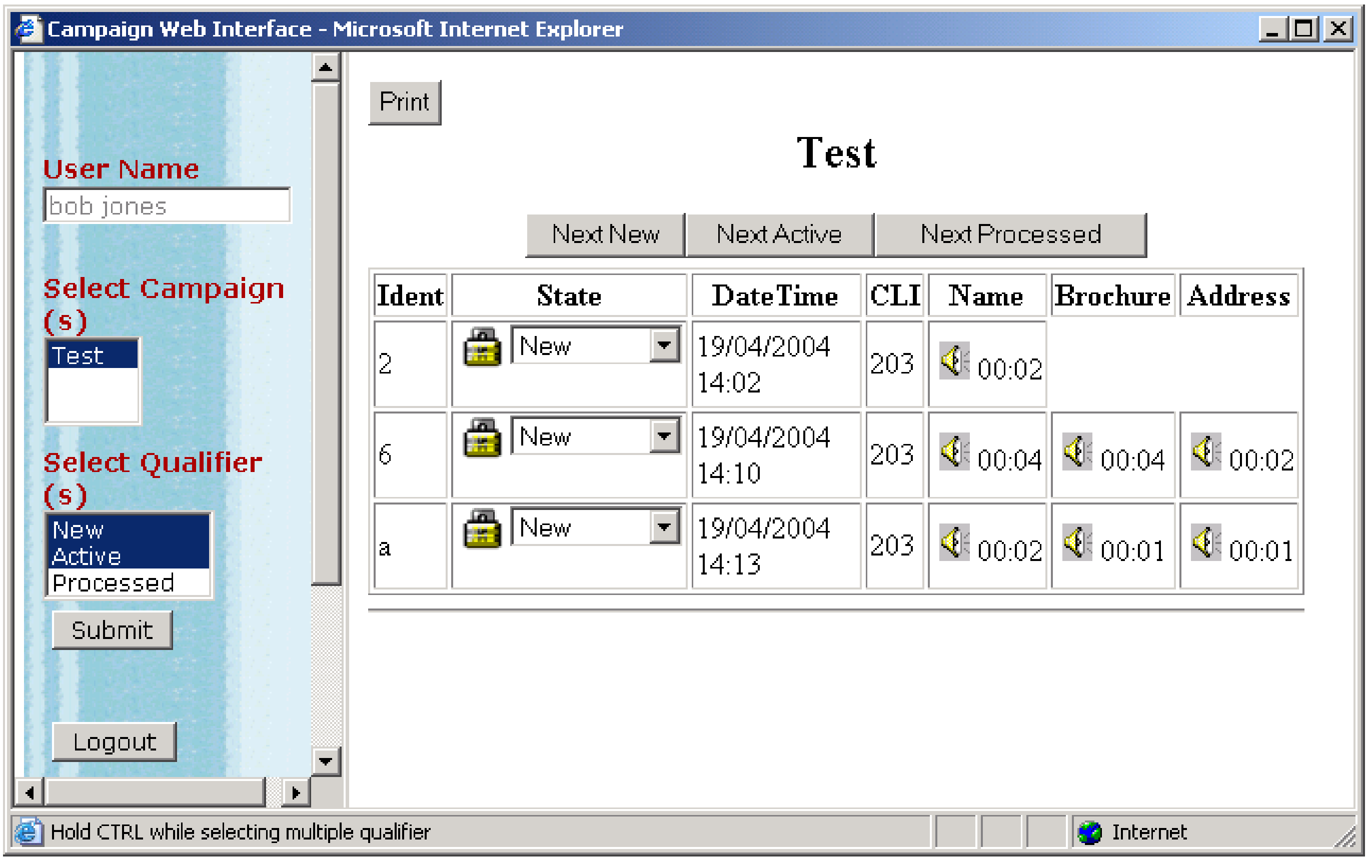
The differences between these two licenses are the language support and a difference in the Email reading capability. With the Avaya TTS RFA, a TTS engine is included with support for a minimum of 14 languages. All languages supported are included are made available as part of the license charge and 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. Recordings can be directed to the called extension's voicemail box or to any other mailbox for later retrieval. 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 Feature Comparison

Platform Support

	Voicemail Embedded	Voicemail Lite	Voicemail Pro
IP Office - Small Office Edition	Yes (Utilizes in built VCM resources)	Yes	Yes
IP401	Yes (Requires VCM and Memory Upgrade)	Yes	Yes
IP403	No	Yes	Yes
IP406	No	Yes	Yes
IP412	No	Yes	Yes

Capacities

Voicemail	Voicemail Embedded	Voicemail Lite	Voicemail Pro
Number of Mailboxes supported	Up to 100 on 401, no specific limit on IP Office - Small Office Edition.	No Limit	No Limit
Maximum Number of Concurrent Calls (ports)	2 simultaneous calls on IP 401 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 and IP412	Up to 30 dependent on license & platform (IP Office - Small Office Edition=10, IP401=2, IP403=10, IP406=20, IP412=30)
Recording Time	IP401: approx. 3 hours IP Office - Small Office Edition: 10 hours minimum	PC dependent (Requires 1MB per minute)	PC dependent (Requires 1MB per minute)

Features

	Voicemail Embedded	Voicemail Lite	Voicemail Pro
Runs as NT Service	No	No	Yes
Multi-lingual support	Yes	Yes	Yes
Voicemail services for Individual users	Yes	Yes	Yes
Voicemail services for Virtual users	Yes	Yes	Yes
Voicemail services 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
Visual Voice on 20XX digital handsets	No	Yes (limited support)	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

	Voicemail Embedded	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

	Voicemail Embedded	Voicemail Lite	Voicemail Pro
Multi-Level Tree Structure	Single tier option on IP Office - Small Office Edition, not available on IP401	No	Yes
Message Announcements	No	No	Yes
Whisper Announce	No	No	Yes
Alarm Calls	No	No	Yes
Assisted Transfers	No	No	Yes

Other Features

	Voicemail Embedded	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	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 whilst 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	Support planned for November 2004
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 4-7. Option 1, to manage distribution lists planned for November 2004.

6	Outcalling	Support planned for November 2004
7	Autoscan/Autoprint	Autoscan supported

Voicemail System Requirements

- Any IP Office system.
- Any desktop telephone.
- Ethernet attached PC running Microsoft Windows 98/NT/2000/XP/2003, with the following minimum recommended specification.
    - **Voicemail Lite:**
      Pentium 4, 2.4GHz or higher.
    - **Voicemail Pro:**
      Pentium 4, 2.8GHz or higher with 256 MB of RAM minimum.
    - **Voicemail Pro with Integrated Messaging Pro/Campaign Manager:**
      Pentium 4, 2.8GHz or higher with 512 MB of RAM
    - **Voicemail Pro TTS, IMS, Campaign Manager and Third Party Database:**
      Pentium 4, 2.8GHz or higher with 512 MB Ram (98,NT) or 1024Mb Ram (Win2K/XP/2003) minimum, HD minimum 20GB (preferably 7200rpm for better performance), 2GB for Operating System and VoiceMail Pro plus free space for voicemail storage.

Attempting to run the voicemail server on a lower specification PC may cause degradation of operation.

Phone Manager

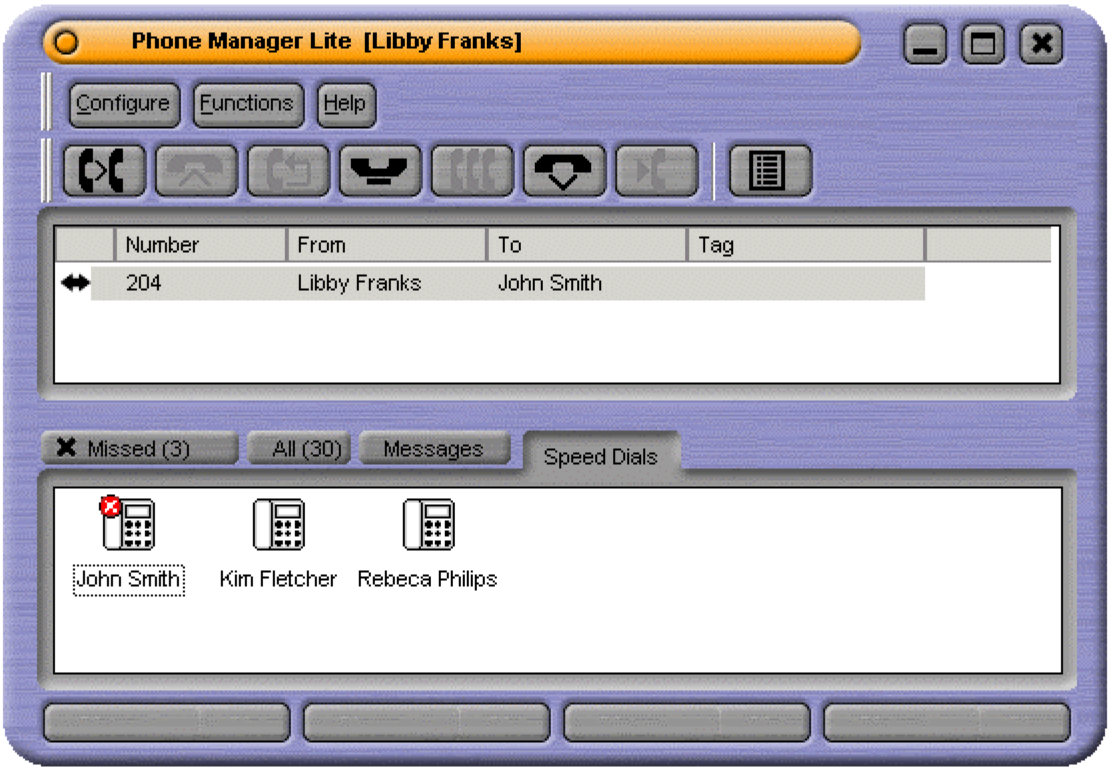
Phone Manager

The Phone Manager application offers control of the telephone terminal from the users' PC. Phone Manager is available in three versions; Phone Manager Lite, Phone Manager Pro and iPhone Manager Pro.

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 terminal rather than the PC's soundcard. The IP Soft-Phone variant adds PC-based telephony via a sound card to the product. See Section 6, IP Telephony for details.

Phone Manager Lite

Phone Manager Lite allows all employees to access the features and facilities only previously available to those working in call centers, or those companies deploying expensive, proprietary feature phones on every desktop. Using an analog telephone, a digital terminal or an IP hardphone, along with a networked PC on your desk, Phone Manager allows employees to take total control of their phone calls from their PC.

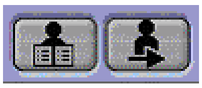


CLI/ANI is presented as standard so you can see who's calling you before you even answer. The caller's phone number and name (if known to IP Office) are clearly shown on your PC, allowing you to have a good idea what the call's about before you take it. Also shown is information on the actual number 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.

Phone Manager's call history keeps a record of all your received, outgoing and missed calls. Double-clicking on any item calls that number back to return a missed call, or to redial a previously called or received 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.

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 rather than cumbersome features codes.

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. 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. 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.

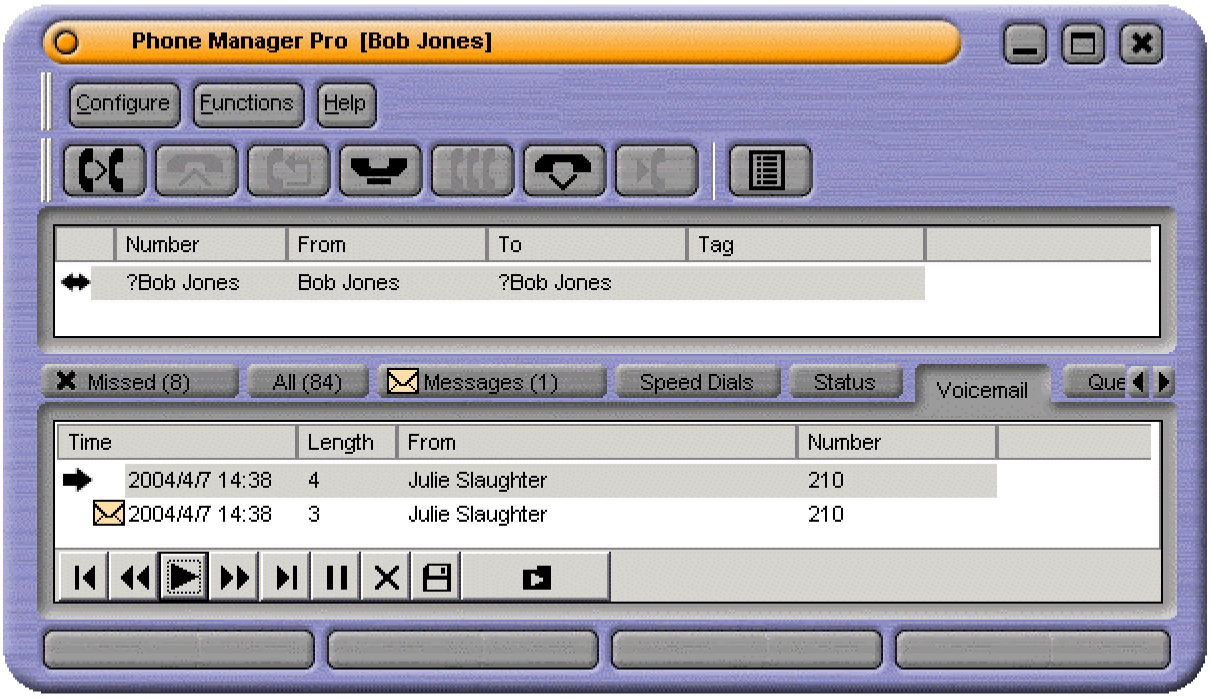


Phone Manager also offers Conferencing Center toolbar buttons that allow users to book a conference or join a web conference. Note: The booking feature is only available if permission is specified by your system administrator and Conferencing Center has been installed (see the Conferencing Center section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

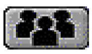
Phone Manager Pro

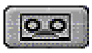
Phone Manager Pro builds upon Phone Manager Lite by offering the following additional features:

- **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 whilst 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.



- **Private 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
    - **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.
- **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 and  Stop Call Recording

  If your Phone Manager Pro is also VoIP enabled, then you can act as a contact center agent entirely through your PC

- **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**.

Feature Comparison

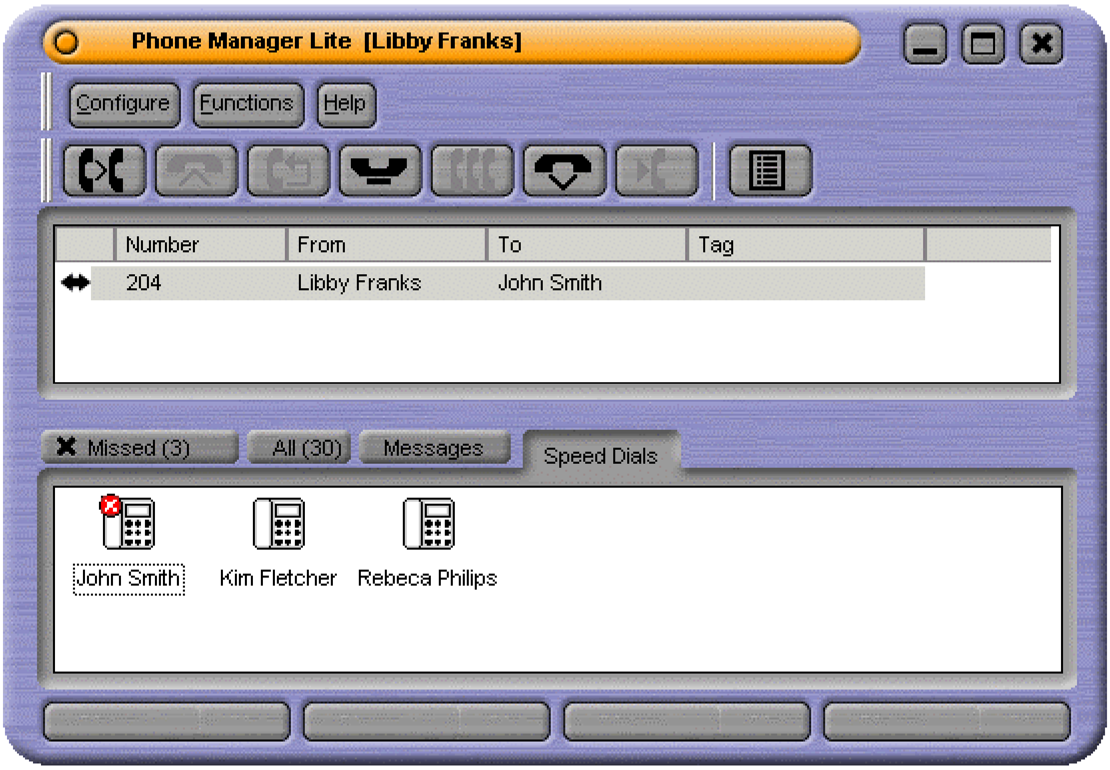
Feature	Phone Manager Lite	Phone Manager Pro
Inbound/outbound call handling.	Yes	Yes
Phone call control.	Yes	Yes
Configure phone preferences.	Yes	Yes
Configure keyboard short cuts.	Yes	Yes
CLI (ANI) /Name display.	Yes	Yes
Speed dial management.	Yes - 15 max	Yes
Busy lamp field (via speed dial).	Yes - 15 max	Yes
Local Phone Directory.		Yes
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
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 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
Conferencing Center.	Yes	Yes
Post Connect dial (sending DTMF whilst connected to another party).	Yes	Yes
VoIP mode (to run as an IP softphone)		Option

System Requirements

- Any IP Office system and supported telephone.
- Ethernet attached PC running Microsoft Windows 98/NT4/2000/XP, in conjunction with TCP/IP Networking.
- Phone Manager Lite/Pro: Minimum Pentium 266Mhz or above with 64MB RAM and 50Mb of free disk space (sound card if audio features required)
- iPhone Manager Pro (IP softphone version): requires a VoIP license in addition to the Phone Manager Pro user license. Please see section 6 (IP Telephony) for the minimum PC requirements.
- Optional Microsoft Outlook 98/2000/2003/XP, Act! 6.0, Maximizer 7.5 and Goldmine 6.0 for contact management integration.
- Optional Internet Explorer 6.0 or above for Conferencing Center integration.

Phone Manager Lite

Phone Manager Lite allows all employees to access the features and facilities only previously available to those working in call centers, or those companies deploying expensive, proprietary feature phones on every desktop. Using an analog telephone, a digital terminal or an IP hardphone, along with a networked PC on your desk, Phone Manager allows employees to take total control of their phone calls from their PC.

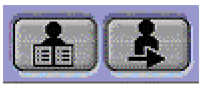


CLI/ANI is presented as standard so you can see who's calling you before you even answer. The caller's phone number and name (if known to IP Office) are clearly shown on your PC, allowing you to have a good idea what the call's about before you take it. Also shown is information on the actual number 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.

Phone Manager's call history keeps a record of all your received, outgoing and missed calls. Double-clicking on any item calls that number back to return a missed call, or to redial a previously called or received 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.

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 rather than cumbersome features codes.

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. 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. 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.

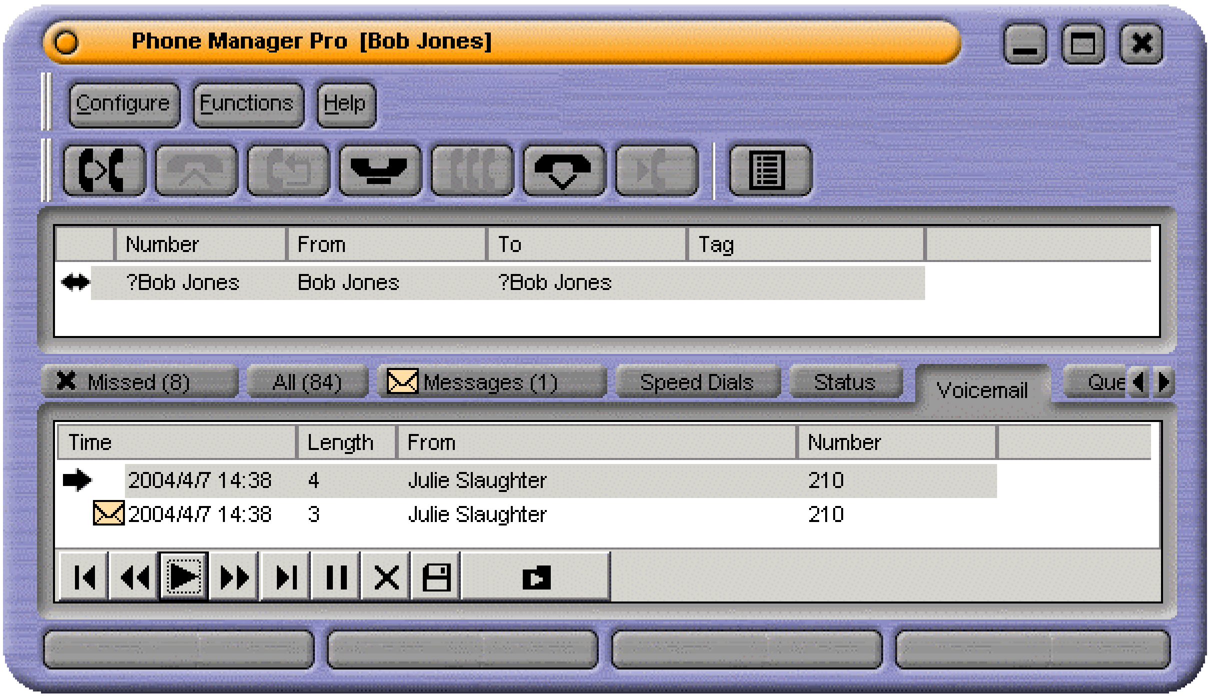


Phone Manager also offers Conferencing Center toolbar buttons that allow users to book a conference or join a web conference. Note: The booking feature is only available if permission is specified by your system administrator and Conferencing Center has been installed (see the Conferencing Center section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

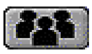
Phone Manager Pro

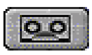
Phone Manager Pro builds upon Phone Manager Lite by offering the following additional features:

- **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 whilst 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.



- **Private 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
  - **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.
- **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 and  Stop Call Recording

  If your Phone Manager Pro is also VoIP enabled, then you can act as a contact center agent entirely through your PC
- **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**.

Phone Manager Feature Comparison

Feature	Phone Manager Lite	Phone Manager Pro
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 max	Yes
Busy lamp field (via speed dial).	Yes - 15 max	Yes
Local Phone Directory.		Yes
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
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 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
Conferencing Center.	Yes	Yes
Post Connect dial (sending DTMF whilst 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 telephone.
- Ethernet attached PC running Microsoft Windows 98/NT4/2000/XP, in conjunction with TCP/IP Networking.
- Phone Manager Lite/Pro: Minimum Pentium 266Mhz or above with 64MB RAM and 50Mb of free disk space (sound card if audio features required)
- iPhone Manager Pro (IP softphone version): requires a VoIP license in addition to the Phone Manager Pro user license. Please see section 6 (IP Telephony) for the minimum PC requirements.
- Optional Microsoft Outlook 98/2000/2003/XP, Act! 6.0, Maximizer 7.5 and Goldmine 6.0 for contact management integration.
- Optional Internet Explorer 6.0 or above for Conferencing Center integration.

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).

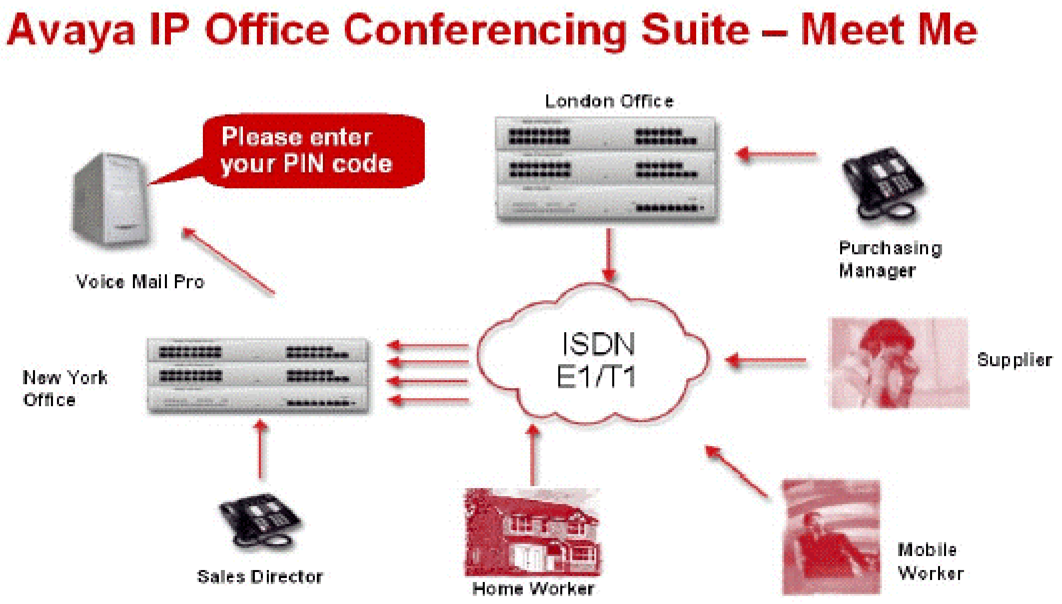
As a result of using conferencing, the benefits gained are:

- 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 IP403, IP406 and IP412 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.

IP Office Conferencing Capacity

The IP Office platforms provide maximum flexibility for conferencing. Today IP403 and IP406 Office 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 Office 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	IP Office - Small Office Edition	IP403	IP406	IP412
T1/PRI-T1	3	48/46	48/46	96/92
IP	3	20	20	60
Internal users	3	64	64	2x64
Total max.	**3**	**64**	**64**	**2x64**

Rest of the World

Maximum Participants	IP Office - Small Office Edition	IP401	IP403	IP406	IP412
ISDN	3	3	60	60	120
IP	3	3	20	20	60
Internal users	3	3	64	64	2x64
Total max.	**3**	**3**	**64**	**64**	**2X64**

Important 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).

Voicemail Pro Requirements (if PIN codes or guidance are required)

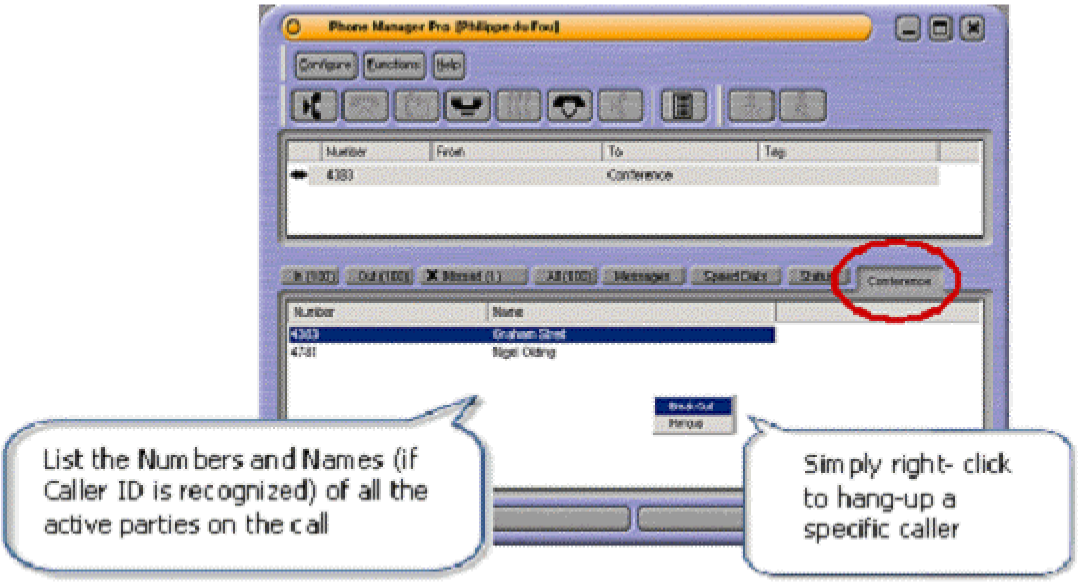
Dial in prompts and PIN codes for security were added with Voicemail Pro Release 1.3. This facility simply requires the Voicemail Pro system license offering simultaneous access to 4 people i.e. 4 participants can type in the PIN or access a menu at the same time. Once they are on the conference bridge, 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 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 Built-In 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 (see below).



- **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

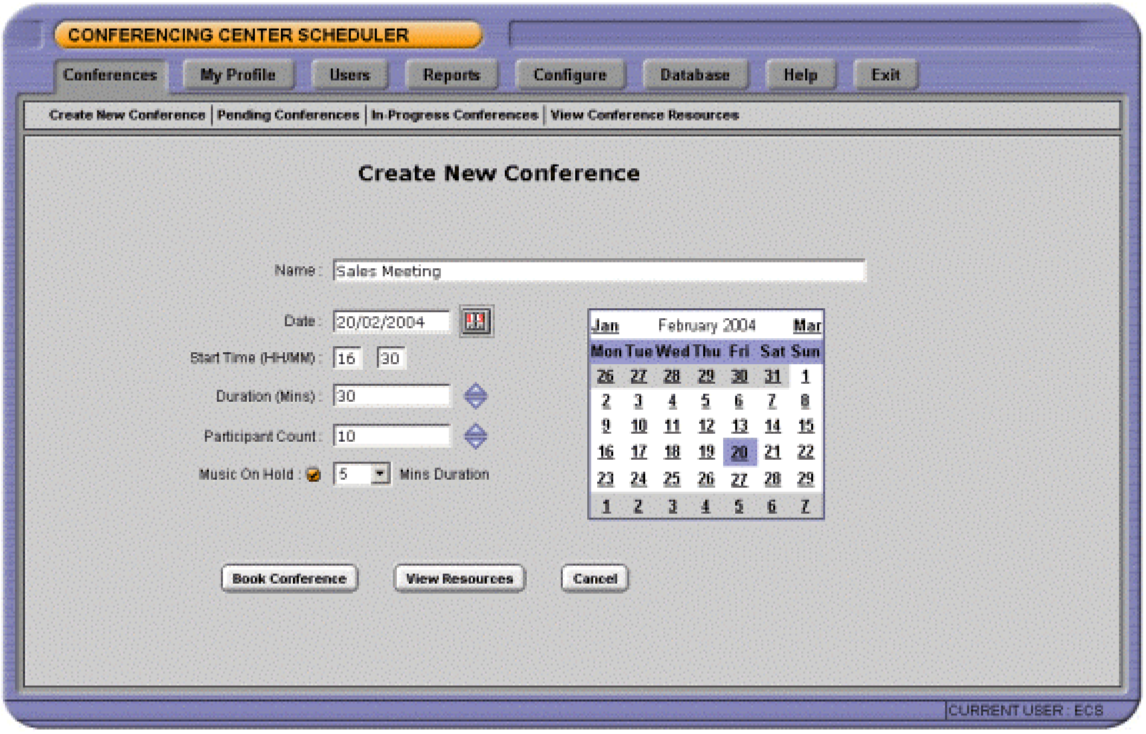
The integrated conferencing functionality on IP Office can be greatly enhanced by adding Conferencing Center. This optional application is a web-based software package that consists in two parts:

- 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.



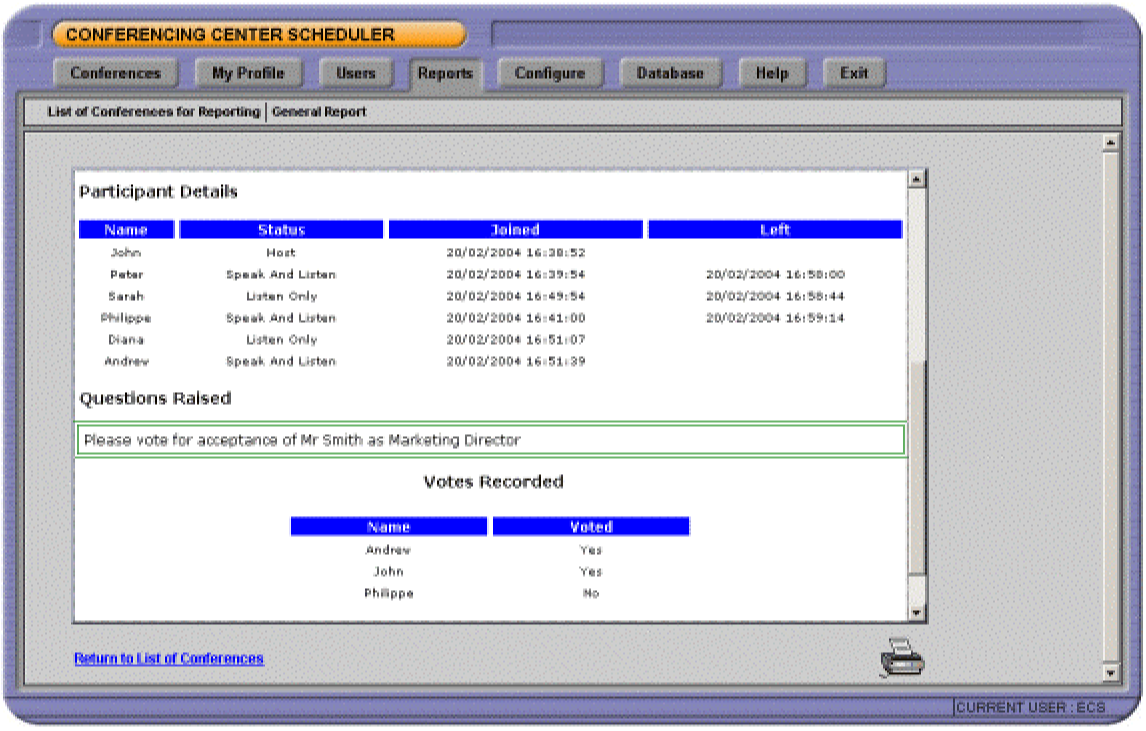
Access to the Web Scheduler requires a user to be set up by the administrator and have Internet Explorer (6.0 or above) installed on their PC. No other software is required. The System Administrator can set up an unlimited number of registered users on the Conferencing Center application. Once registered, 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.

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

- IP Office version 2.1 or higher.
- Voicemail Pro version 2.1 or higher.
- Conferencing Center Server:
    - Pentium 450Mhz (1.4Mhz recommended) or above with 256MB RAM (512MB + recommended) running Windows 2000 Server or Windows 2003 Server (Windows XP Professional, Windows 2000 Professional could be used but would typically support 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.
- Phone Manager version 2.1 or higher (optional).
- SoftConsole version 2.1 or higher (optional).

Digit Cordless Solutions (non VoIP)

Digit Cordless Solutions (non VoIP)

IP Office supports three wireless solutions:

- TransTalk which is primarily for the North American market.
- DECT which is primarily for the European market.
- Avaya 3810 which is primarily for the North American market.

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 Onsite Mobility Solution is to improve communication with staff who, because of the function they perform, are mobile within the workplace. Using cordless technology such individuals may be instantly contactable, with many obvious benefits.

The cordless 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 cordless 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.

IP Office DECT

IP Office DECT

Based on the Digital Enhanced Cordless Telecommunications (DECT) standard, IP Office's cordless system supports 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, IP Office's 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 two DECT options on IP Office: the Compact DECT Control Unit and the larger external DECT Control Unit.

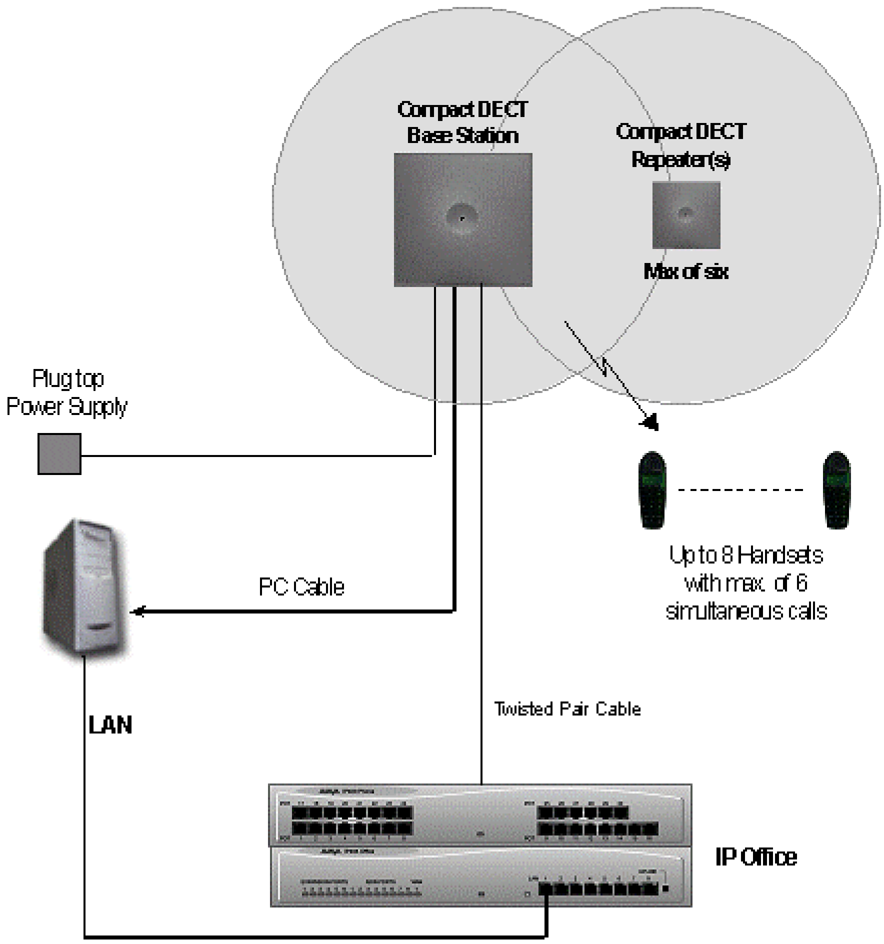
Compact DECT Control Unit

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 cordless handsets and 7 DECT Base Stations (DBS). The Compact DECT CU is connected to the IP Office control cabinet by 2 wire analogue 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 terminals 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 cordless 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

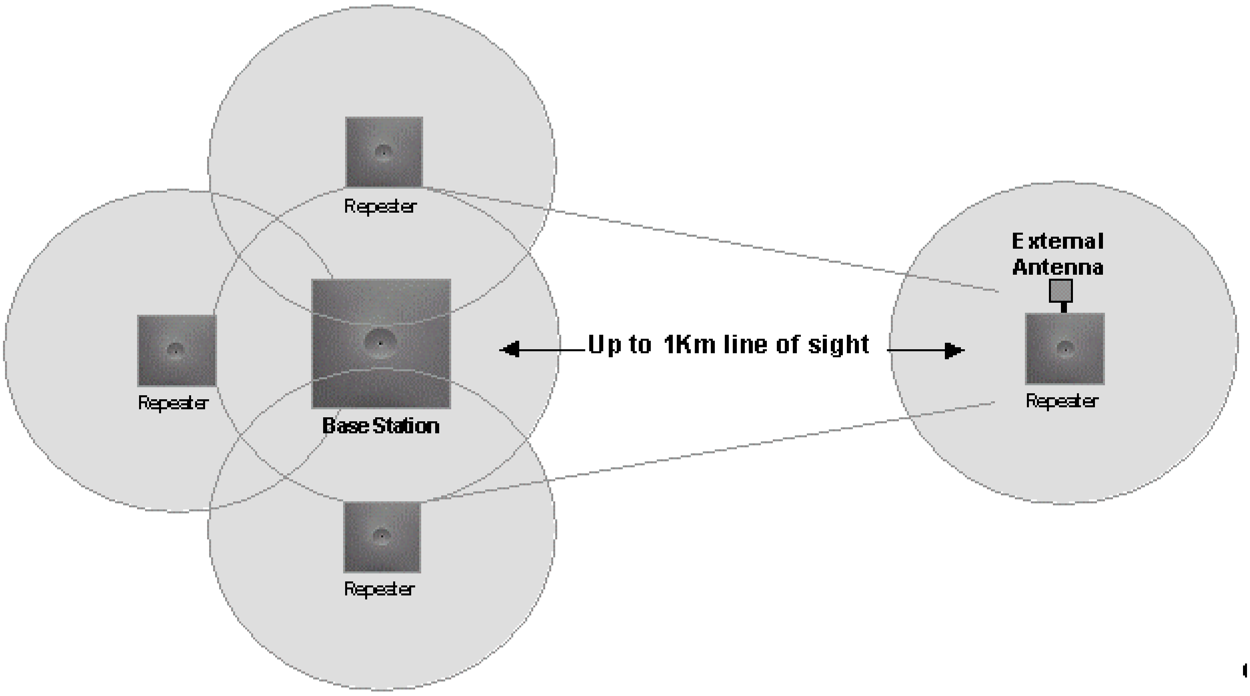
The coverage offered by the Compact DECT CU can be supplemented by up to 6 DECT Repeater Base Stations.

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

The compact DECT solution provides smaller businesses with a highly functional entry-level cordless 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. 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 cordless handsets and 32 base stations, two DCU's are linked using two Link Cards.

Each cordless 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.

DECT Comparison

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

DECT Feature Integration

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

- **DDI/DID**
  Since each cordless 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.
- **Group working**
  Cordless 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 20 Series terminal to a cordless handset. This is particularly useful for extension users who are desk-based most of the time and want access to the full range of features available to 20 Series extension users, but who need to be accessed quickly and efficiently when away from their desk.

Features available through the CTI DECT license

- 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 - Cordless users require many of the standard, as well as advanced, functions available to users of Desktop handsets. All terminal 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 cordless 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)
  - **20DT - DECT Cordless Handset** - IP Office's DECT Handset, the 20DT, forms an integral part in the IP Office terminal range in terms of design and functionality. The 20DT incorporates design ideals from GSM technology phones into the work place through its modern and robust shape. The 20DT is described in section 3 referring to Terminals.

TransTalk

Introduction to TransTalk

Avaya's TransTalk 9040 system delivers the benefits and accessibility of a wireless phone with all the power and functionality of a wired desk telephone.

The TransTalk 9040 is an in-building wireless system that provides a mobility solution up to a 900' range from the Digital Radio Module (DRM) depending on environment.

An outdoor enclosure is also available, allowing the mounting of up to 2- DRMs per Outdoor Box outside the building for extended coverage. All DRMs must be connected together with the provided Sync. Cable. Longer cables for Outdoor Box to Outdoor Box DRM cabling are available.

Avaya's TransTalk 9040 solution integrates fully with IP Office. The DRM 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 TransTalk handsets

With the TransTalk 9040 system's 4-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.

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

The TransTalk family (Two 9040 Voice Terminals with DRM)

9040 TransTalk Phone

The TransTalk 9040 Wireless Telephone

- **Voice Quality -** The TransTalk 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")
- **Noise Cancellation/Sound Enhancement -** Noise Cancellation/Sound Enhancement helps workers in noisy environments, such as a manufacturing line.
- **Test Mode -** The TransTalk 9040 test mode functionality can be used to actually measure radio reception eliminating, in many cases, the need for RF meters and range estimates. This special capability is patented and is used to:
  - Determine proper module station placement
  - Ensure optimal system performance
  - Ensure optimal customer satisfaction.
- **Vibrator Alert -** The 9040 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.

TransTalk 9040 Voice Terminal Attributes

Each TransTalk 9040 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. See also TransTalk 9040 Wireless Handset.

Battery Charging

Each TransTalk 9040 handset comes with its own charging cradle. The TransTalk's 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 TransTalk 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 1.5 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 3.5 hours of talk time, and 22 hours 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 8 hours of talk time and 72 hours of standby time.

When batteries are recharged repeatedly without first being fully discharged, they can lose the ability to provide the output voltage for the full time for which they were designed. This phenomenon is called the "memory effect." The TransTalk's charging cradle includes the capability to provide a deep discharge phase prior to charging in order to eliminate this failing. Although the NiMH batteries used for the TransTalk are described as being highly resistant to the memory effect, the charging cradles provide the deep discharge as a precaution.

The charging cycle for the spare battery always includes a deep discharge of the battery (a ½ hour process) and then a one-hour full re-charge. The charging cycle used for the battery in the handset is user selectable, and can optionally bypass the deep discharge phase, thereby allowing a full charge in one hour.

Capacity

The TransTalk 9040 handset has a backlit four-line display, incorporating icons for line and feature access and status. The display includes:

- A 2 x 16 character alphanumeric display capability for internal called and calling party information, and external called number display
- The capability for external Caller ID, if the network supports it.
- Access to either 10 lines/intercoms/feature buttons with physical appearances on the handset display

TransTalk 9040 Accessories

- **Carrying Clip and Lanyard**
  A belt/pocket clip has been designed into the 9040 handset for ease of carrying and access. The clip is built into the back of the handset and is removable. A lanyard (wrist strap) is also included with the TransTalk handset for easy carrying.
- **Replaceable Antenna**
  The antenna on the handset is user replaceable. A customer-replaceable antenna is available in the event of breakage.
- **Holster Option**
  Leather holsters are optionally available for the 9040 TransTalk. These holsters are available in black.
- **Headset Option**
  There is a 2.5mm headset jack that allows headset support and integration. Two types of headsets are now available for the 9040 TransTalk, making hands-free operation possible. The Supra headset is an over-the-head model, while the Radium model merely hangs over the user's ear. A special adapter cord allows the use of the headsets' "quick-disconnect" cords.
- **Security**
  The TransTalk telephone system uses digital rather than analog radios. Digital radio transmissions are very difficult, if not impossible, to monitor.
    - The TransTalk system also uses "spread spectrum frequency hopping," a design that has each radio/handset combination constantly changing transmit-receive frequencies within the 150 available channels. Since each conversation is constantly switched throughout the range of channels, conversations cannot be monitored.

Avaya 3810

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) and Magix (minimum release 3.0). 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 10 Avaya 3810 wireless sets 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.

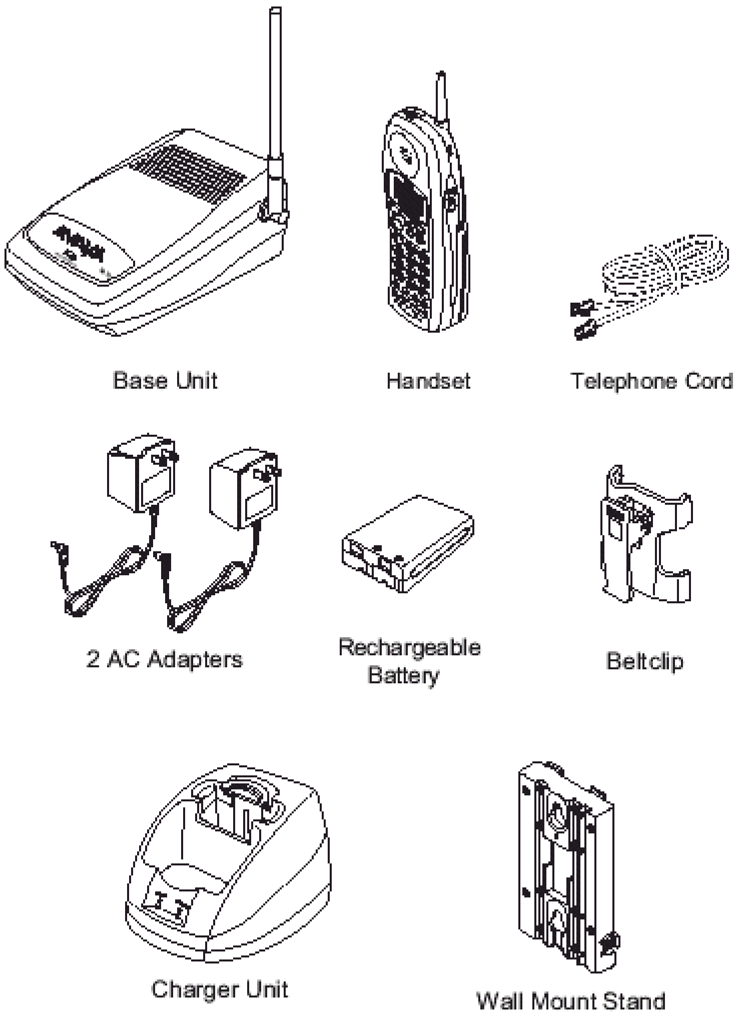
The 3810 provides the following features:

- 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 Terminal 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.

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 PC (running Microsoft Windows 98, Windows NT, Windows 2000 or XP) must also be provided, and must be connected to the same LAN as the IP Office switch. This PC must run the IP Office License Service, and should be permanently switched on and connected to the LAN.

Licensing is achieved by the use of a feature key (dongle), which plugs into the PC's parallel port, USB port or to the serial port of the Small Office Edition control unit. 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, significantly increasing the resilience of the system.

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.

10. The Contact Center

IP Office Contact Center/CRM Solutions Product Overview

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 small 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
- MultiMedia Module for CCC (New for Version 4)

Compact Business Center

IP Office Compact Business Center is an entry-level management tool for small customer facing departments, typically handling from 2 to 15 agents. It provides graphs on real-time and historical information (up to 31 days) for up to three groups, as well as providing system-wide information on the operation of the system as a whole. It provides information on key performance indicators of the business - lost calls, trunks free, agents free, queuing time and much much more.

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:

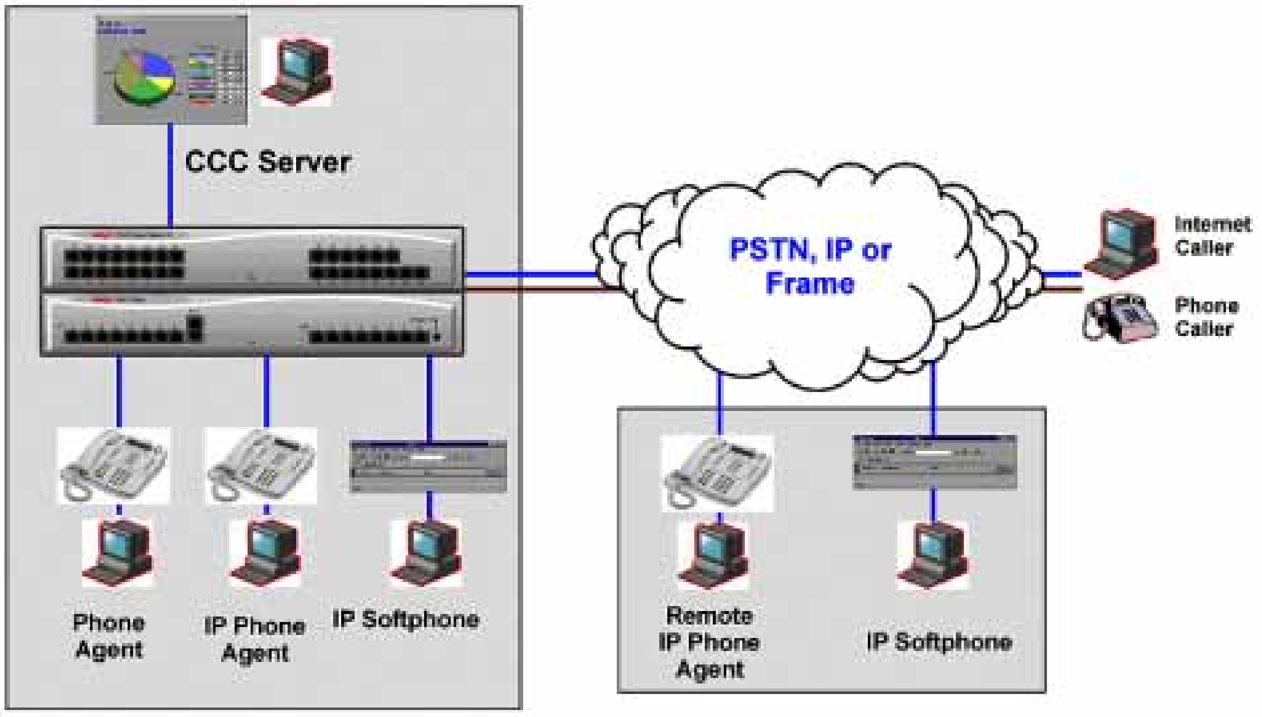
- **CCC Server**
  Provides one supervisor position, with real-time information view, management by exception, plus historical reports for any aspect of the contact center. Also included are 5 agents, and one full-functionality PC-based wallboard. Up to 70 standard reports can be produced. All users can immediately use a PC-based wallboard to view basic statistics for the whole contact center.
- **Report Manager**
  A easy to use application that allows the supervisor to create and schedule reports on all the activity of the contact center. Scheduling, saving to HTML and PDF, and easy to read descriptions are built in to the Report Manager product.
- **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.
- **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, iPhone Manager Pro 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.
- **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.
- **Additional Supervisor Positions**
  As many as 5 supervisor positions can be purchased in total for CCC. This provides a supervisor with the ability to monitor in real time the service being provided to callers.
- **Report Designer**
  Provides supervisors with the ability to create their own customized reports on the contact center's activity
- **Agent Rostering interface**
  CCC works with a number of workforce rostering packages, including Blue Pumpkin and Qmax. This module enables the interoperability of these packages with the Compact Contact Center.

MultiMedia Module for CCC

The MultiMedia Module provides CCC with new routing schemes. It will also provide combined reporting for all interactions within the contact center. There are several queue types supported in the MMM; they are:

- Email Queues.
- Chat Queues.
- Web Callback.
- Proactive List Dialing.

IP Office MultiMedia Module extends the routing and reporting of voice calls into the realm of emails and chat sessions initiated via a web browser. In this way, 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.



Overview of the Compact Contact Center

Compact Business Center

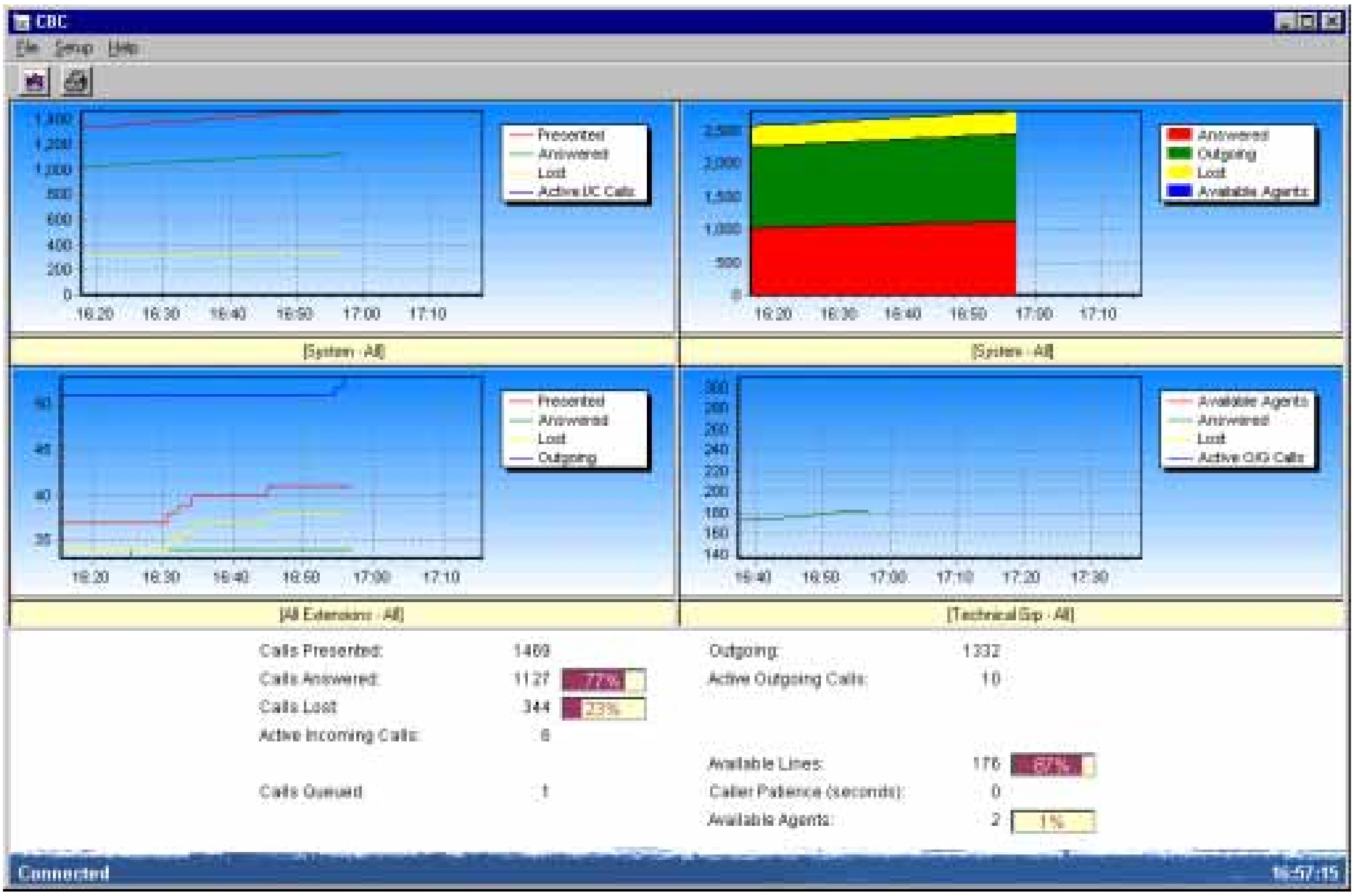
Compact Business Center

Compact Business Center is enabled by a license key 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), powers the server piece. 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.

- Department/Hunt Groups.
- Period start time (24hr period).
- Period end time (24hr period).



Compact Business Center Example

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.

Compact Contact Center (CCC)

Compact Contact Center (CCC)

Compact Contact Center is a pre-packaged suite of 4 modules that runs as a client/server application. The suite of pre-packaged modules consists of:

- **Call Center View.**
- **Wallboard Manager.**
- **Report Manager.**
- **MultiMedia Module.**

It is designed to provide a tightly integrated real time and historic reporting package and wallboard support for IP Office. The Compact Contact Center has been designed to allow customers to manage their customer facing department or contact center effectively and improve the service they provide to their customers.

The product consists of a set of fully 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 whilst accuracy is assured through the single configuration database held on IP Office.

Call Center View

Whilst Wallboard Services are useful for monitoring the service provided by the Customer facing department in real time, human resource management is required to manage customer service effectively. 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.

Many traditional management information systems (MIS) rely on a busy supervisor constantly monitoring queues and agents. Avaya have taken the approach that allows Call Center View to do the work and only informs the supervisor when a problem has or is about to occur i.e. exception management and reporting. Alarms may be set on up to 16 parameters per device, 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

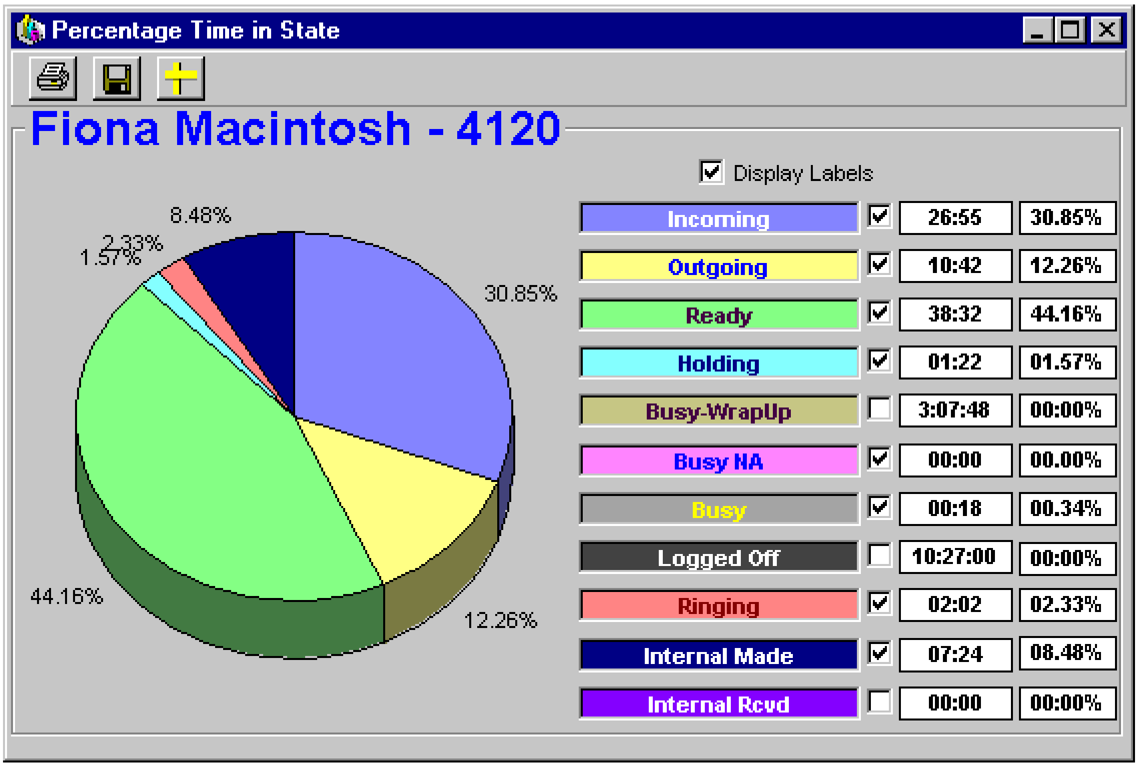
- Alarm Handling.
- BLF Details.
- Extension Activity.
- Callback Request.

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

Wallboard Manager

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 Manager is a PC based application utilizing a standard PC running Windows NT4/2000/XP.

The Wallboard Manager architecture has been designed to be upgradeable as users requirements expand, allowing the same PC hardware platform to be used to support additional CCC modules. 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 local 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	407689827
Spectrum Wallboard Master Kit US	Not Applicable	407679174
Spectrum Wallboard Master Kit UK	5340-UKK	408097350
Spectrum Wallboard Master Kit Europe	5340-IKT	407689892
Wireless Keyboard (Remote Control)	5340-905	161336
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.

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.

CCC version 4 also provides a supervisor template that will prevent users from continually changing the parameters without permission.

Report Manager and Report Designer

The Report Manager provides in depth historical reporting on the customer facing department's activity. Report Manager provides 48 standard reports for measuring both overall contact center call handling and individual/team performance to enable improved human resource management. These standard report templates may be parameterized by the user (in terms of date range, group, shift etc) to create their own 'management ready' reports. These reports can be scheduled to run at a specified date and time, or repeated at defined intervals.

Optionally Report Designer may be added to allow the user to create custom reports, or modify and change standard reports, providing total flexibility in the presentation of traffic and agent information. Report Designer is aimed at the contact center manager who requires a greater degree of flexibility via completely tailorable reports and ad hoc querying to allow better informed decisions. Report Designer adds the flexibility of only generating reports when exceptions occur and allows reports to be exported in a variety of formats.

Reporting Enhancements in CCC Version 4

- Supervisors can now schedule reports to be delivered to various places within the contact center.
- Reports can now be delivered to multiple recipients via email in the following formats; PDF, HTML, and REP.
- Reports can also be scheduled for delivery to multiple printers within the network at the same time.

CCC Reports

1. Account Code Log by Agent Group (All Media).
2. Account Code Log by Agent Group (Graphical - All Media).
3. Account Code Log by Agent Group (Graphical).
4. Account Code Log by Agent Group.
5. Account Code Log by DDI (Graphical).
6. Account Code Log by DDI.
7. Account Code Log by Pilot (Graphical).
8. Account Code Log by Pilot.
9. Account Code Log by Target (Graphical).
10. Account Code Log by Target.
11. Agent Activity Trace.
12. Agent Activity.
13. Agent Callback Request.
14. Agent Group Busy Status.
15. Agent Group Graphical Summary (All Calls).
16. Agent Group Graphical Summary (All Media).
17. Agent Group Graphical Summary.
18. Agent Group Member Call Duration Report (All Calls).
19. Agent Group Member Duration (All Media).
20. Agent Group Member Duration.
21. Agent Group Tabular Summary (All Calls).
22. Agent Group Tabular Summary.
23. Agent Group Tabular.
24. Agent Individual (All Media).
25. Agent Individual.
26. Agent Tabular (All Media).
27. Agent Tabular.
28. Customer Tracking by Call Identifier.
29. Customer Tracking by CLI.
30. DDI Call Duration.
31. DDI Distribution by Target.
32. DDI Distribution.
33. DDI Response.
34. DDI Routing.
35. DDI Summary (All Calls).
36. DDI Summary.
37. External Transferred Account Code.
38. Incoming DDI Summary.
39. Incoming Duration Summary (All Media).
40. Incoming Duration Summary.
41. Incoming Pilot Summary.
42. Lost Call CLI.
43. Multi-Media Summary.
44. Outgoing Account Code Costing Log (All Media).
45. Outgoing Account Code Log (All Media).
46. Outgoing Account Code Log (Graphical).
47. Outgoing Account Code Log.
48. Outgoing Most Common Destination by Agent Group.
49. Pilot Call Duration.
50. Pilot Distribution by Target.
51. Pilot Distribution.
52. Pilot Response.
53. Pilot Routing.
54. Pilot Summary (All Calls).
55. Pilot Summary
56. System Summary.
57. Target Graphical Summary (All Media).
58. Target Graphical Summary.
59. Target Member Duration (All Media).
60. Target Member Duration.
61. Transfer Call Tracking Detail by Agent.
62. Trunk Group Activity.
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.

Multi Media Report Integration

When using the MultiMedia Module (MMM) within the contact center, all agent interactions when logged in, be they email, chat, and of course, over the phone, will be captured and reported on. This is represented by several new reports that tabulate the media interactions within the center.

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.

MMM 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 E-mail 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, E-mail, 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.

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, E-mail messages, Web calls, and Web Chats and can communicate with group members from one centralized view.
- **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.

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.

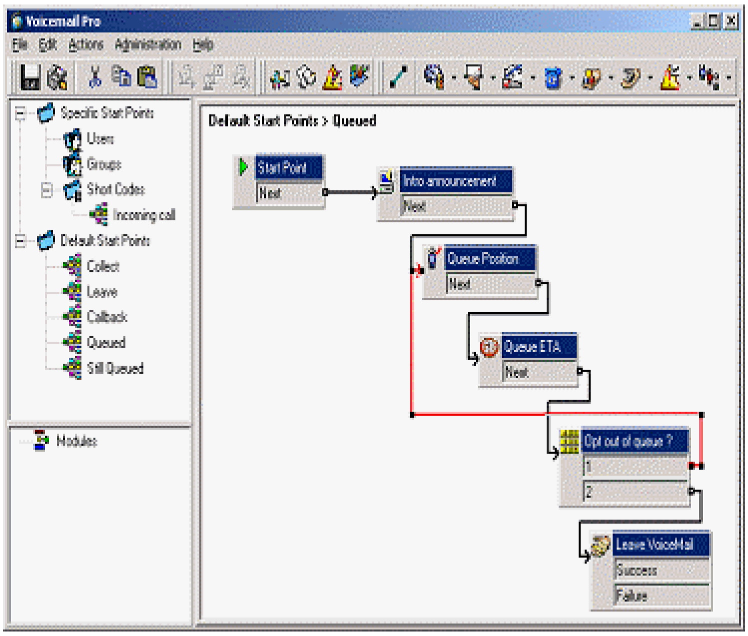
Thanks to a powerful graphical user interface, it can easily and rapidly create and modify call flows from simple announcements to multiple single digit tone menus, to provide Audiotex services and Automated Attendant functionality.

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), whilst 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

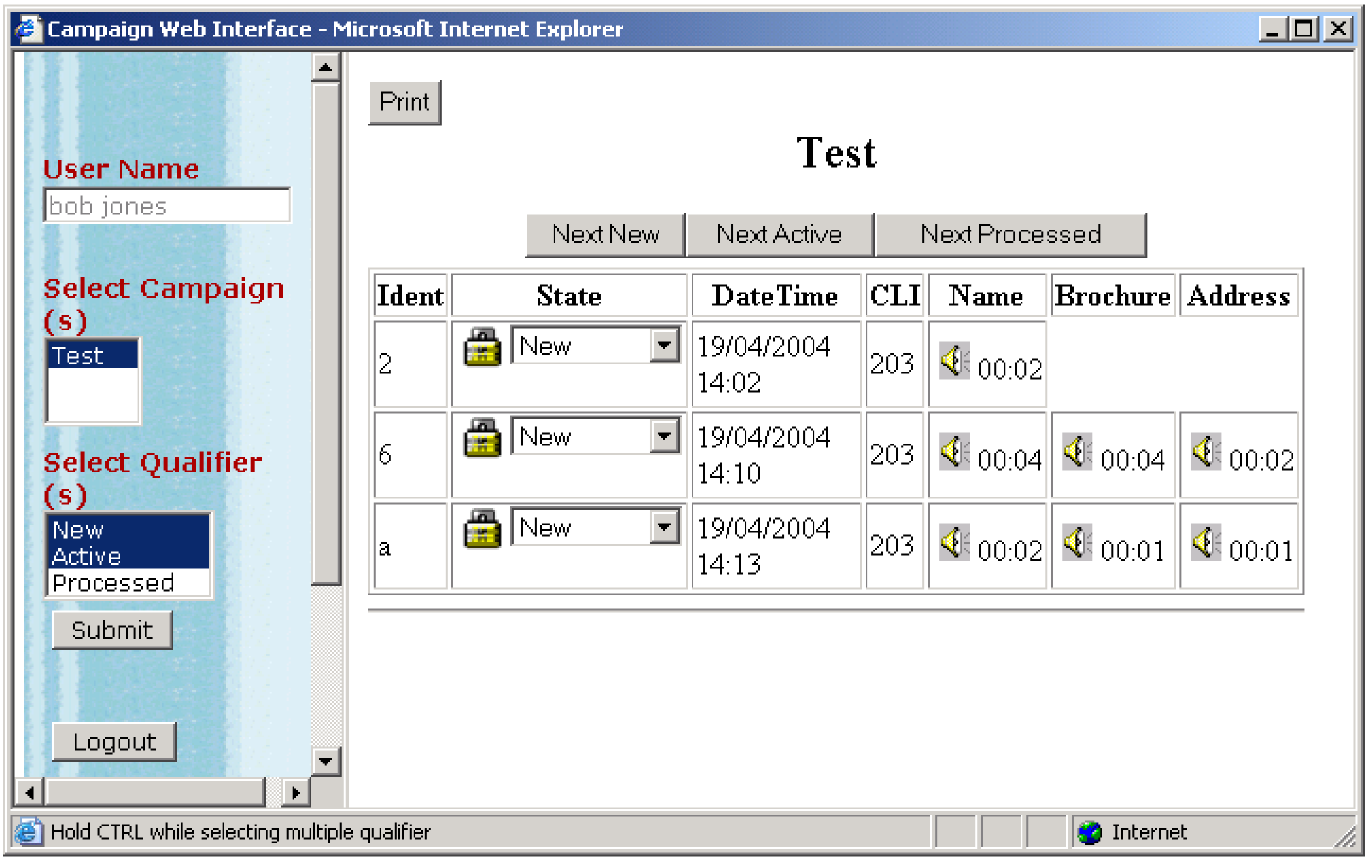
An Integral part of Voicemail Pro, this application enables repetitive information gathering (such as brochure requests) to be fully automated, leaving agents free to deal with other, more complex calls which require human interaction. Campaign Manager enables round-the-clock immediate answering, offering 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 terminals 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	12 months +
Pre-defined reports	None	48
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	Win 2000	Win 2000
PC Wallboard	Not available	Optional
Report Designer	Not available	Optional
WFM Interface	Not available	Optional
Agents	N/A	75
Supervisor	3	5

Technical Description/Configuration

All CCC & CBC applications are based on industry standards and exploit the resilient Windows NT4/2000/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 requirements for both the server and client platforms as follows:

Compact Business Center

Server PC (Delta Server):

- **Operating Systems Supported:**
    - Windows 2000 Server SP2 and later.
    - Windows XP Professional.
    - Windows 2000 Professional SP2 and later.
- **PC Specification:**
    - Pentium III 500MHz or higher.
    - 10GByte hard disk.
    - Minimum 256 Mbytes of RAM.

Client PC:

- **Operating Systems Supported:**
    - Windows 2000 Server SP2 and later.
    - Windows XP Professional.
    - Windows XP Home.
    - Windows 2000 Professional SP2 and later.
    - Windows NT Workstation SP6.
    - Windows 98.
- **PC Specification:**
    - Pentium III 500MHz or higher.
    - Minimum 128 Mbytes of RAM.

Customer Contact Center

Server PC (Delta Server):

Supporting a maximum of 5 Supervisor Positions:

- **Operating Systems Supported:**
    - Windows 2000 Server SP2 and later.
    - Windows NT Server SP6.
    - Windows 2000 Professional SP2 and later.
      Use of Windows 2000 Professional is recommended only in small contact centers (10 agents) that have low call volumes (300-500 calls per day), if you are unsure whether these parameters are being met in the customer's business, use the server configurations.
- **PC Specification:**
    - Pentium III 800MHz or higher with 1 x 10GByte hard disk.
    - Minimum 256 Mbytes of RAM.

Client PC:

Supervisor running CCV, Wallboard Manager, Report Manager

- **Operating Systems Supported:**
    - Windows XP Professional.
    - Windows 2000 Professional SP2 or higher.
      Note: Use of Windows 2000 Professional is recommended only in small contact centers (10 agents) that have low call volumes (300-500 calls per day), if you are unsure whether these parameters are being met in the customer's business, use the server configurations.
    - Windows NT Workstation SP6.
    - Windows 98.
- **PC Specification:**
    - Pentium III 500MHz or higher.
    - 1 GByte hard disk.
    - Minimum 128 Mbytes 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

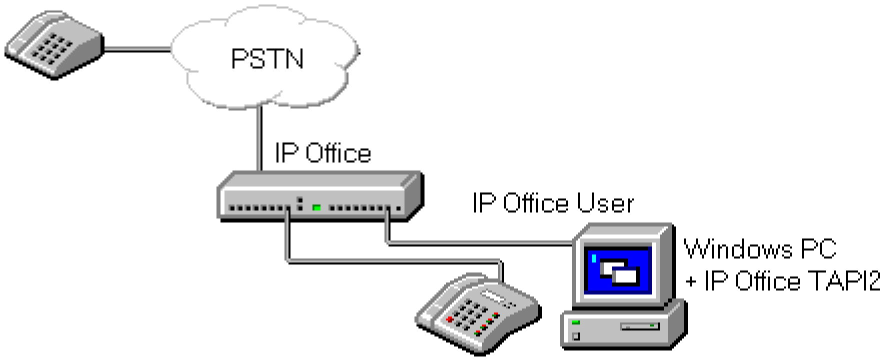
IP Office offers a significant CTI capability. Several interfaces are supported:

- **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 programme, called the Developer Connection Programme.

The Developer Connection Programme ("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 programme 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.

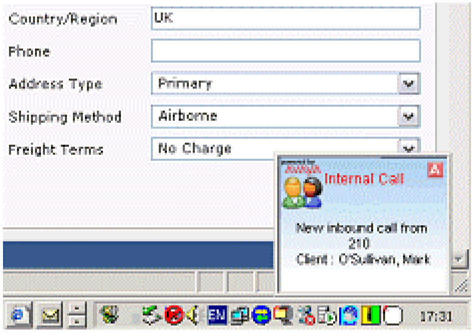
More information on the DeveloperConnect programme can be found at: www.devconnectprogram.com.

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, an example of the screen pop associated with the application is shown below.



11. Common Management Utilities

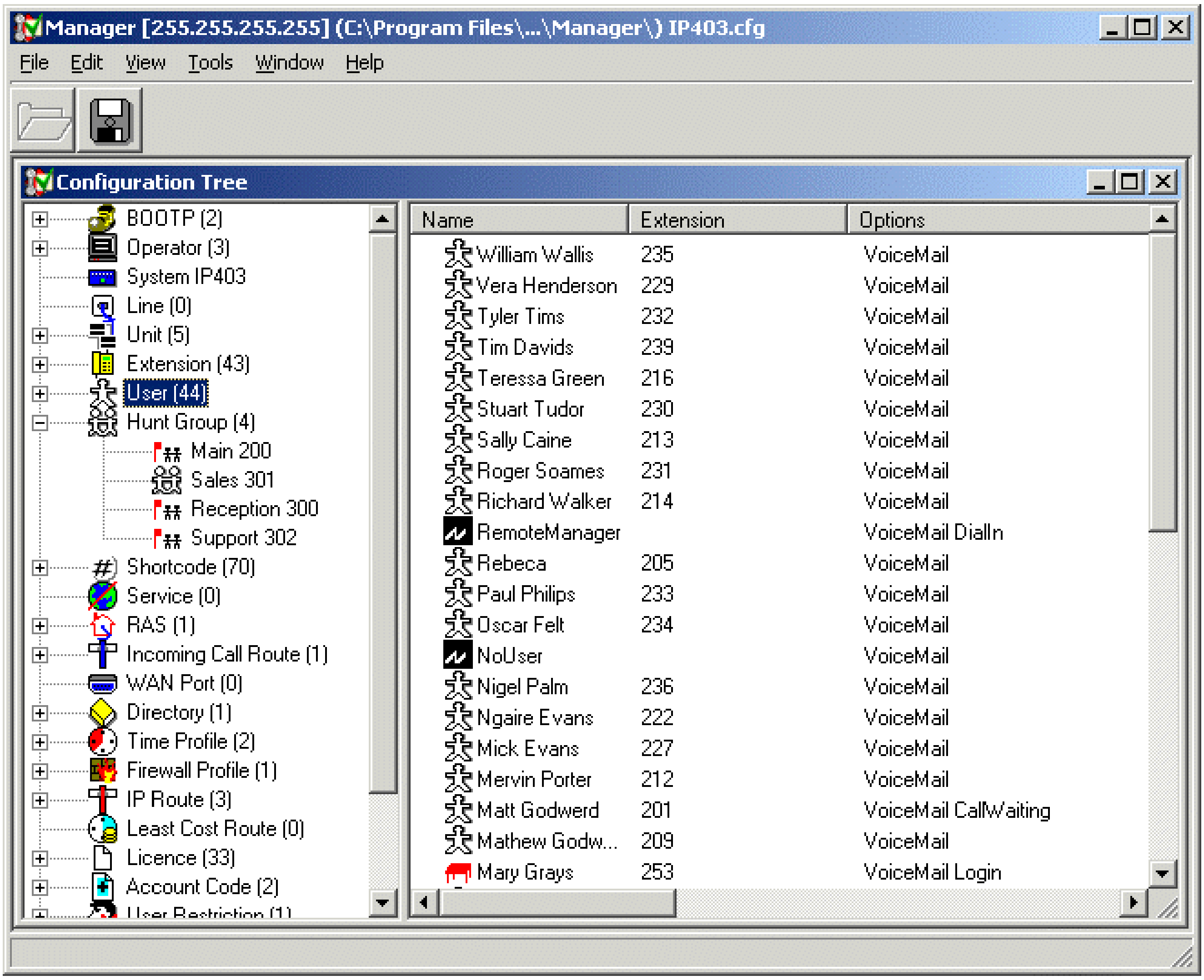
Introduction to IP Office Management Utilities

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.

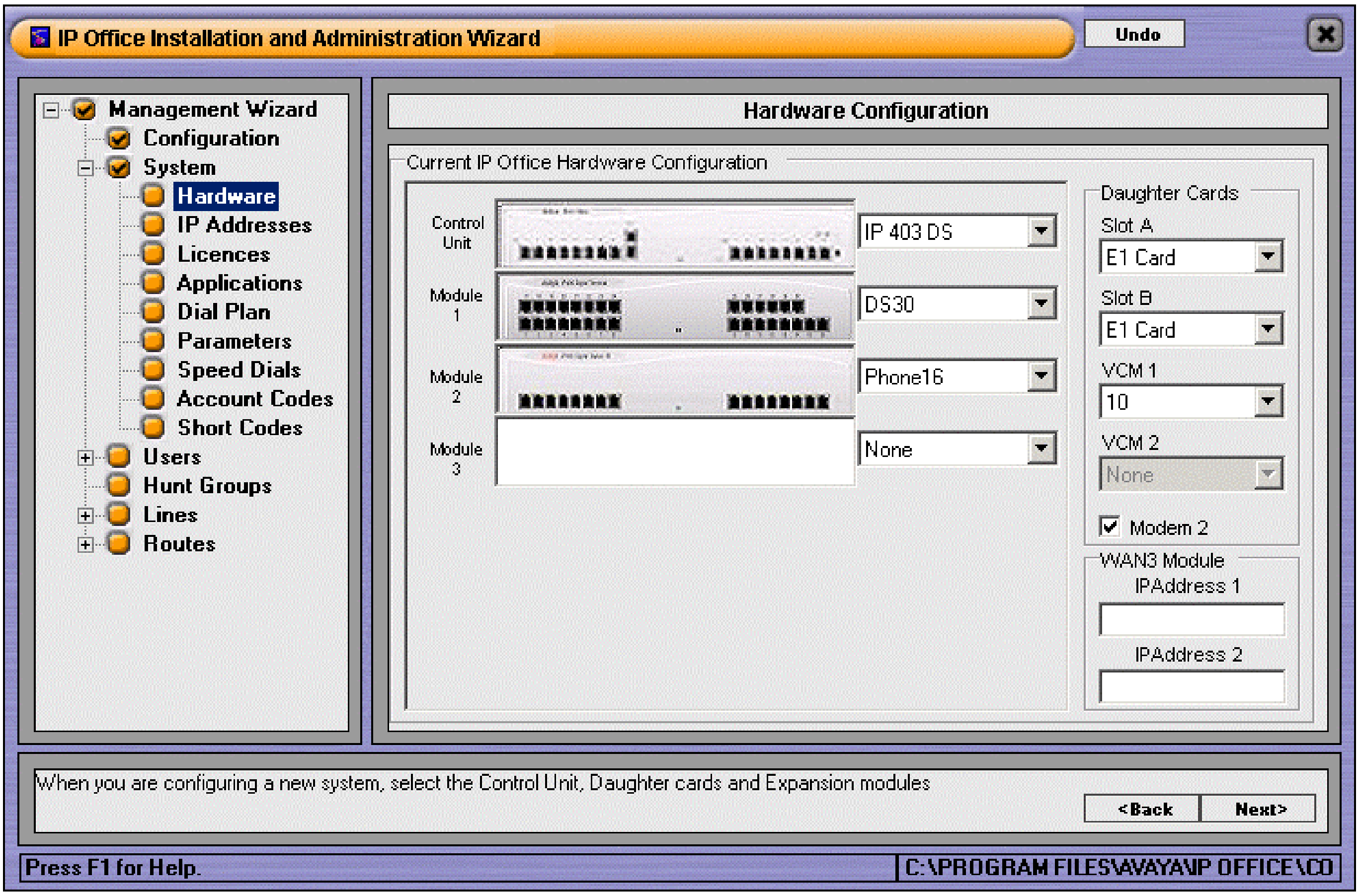
A number of other convenient features are provided by the manager including:

- 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 Installation and Administration 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.

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 Wizard has three versions, IP Office - Small Office Edition Wizard, IP Office Installation and Maintenance Wizard and Modifications Wizard for Users and Hunt Groups.

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 Embedded Voice Mail and 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 Maintenance 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 within the Voice Mail Auto Attendant.

The Maintenance and Modifications 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 prevents unintentional system changes while providing a simple and intuitive interface for the most commonly accessed system management functions – users and hunt groups.

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)

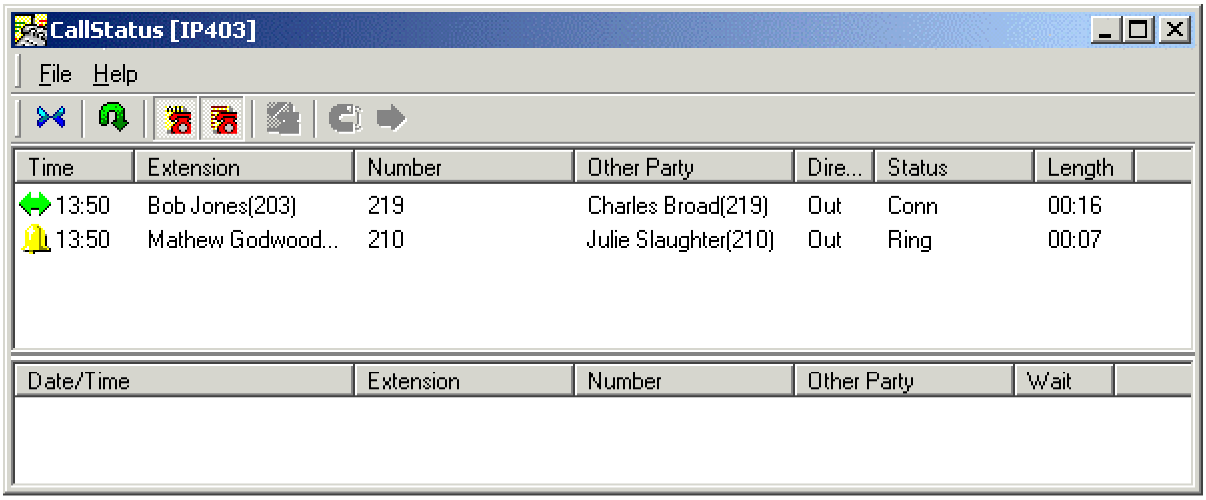
CSV File Format

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

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.

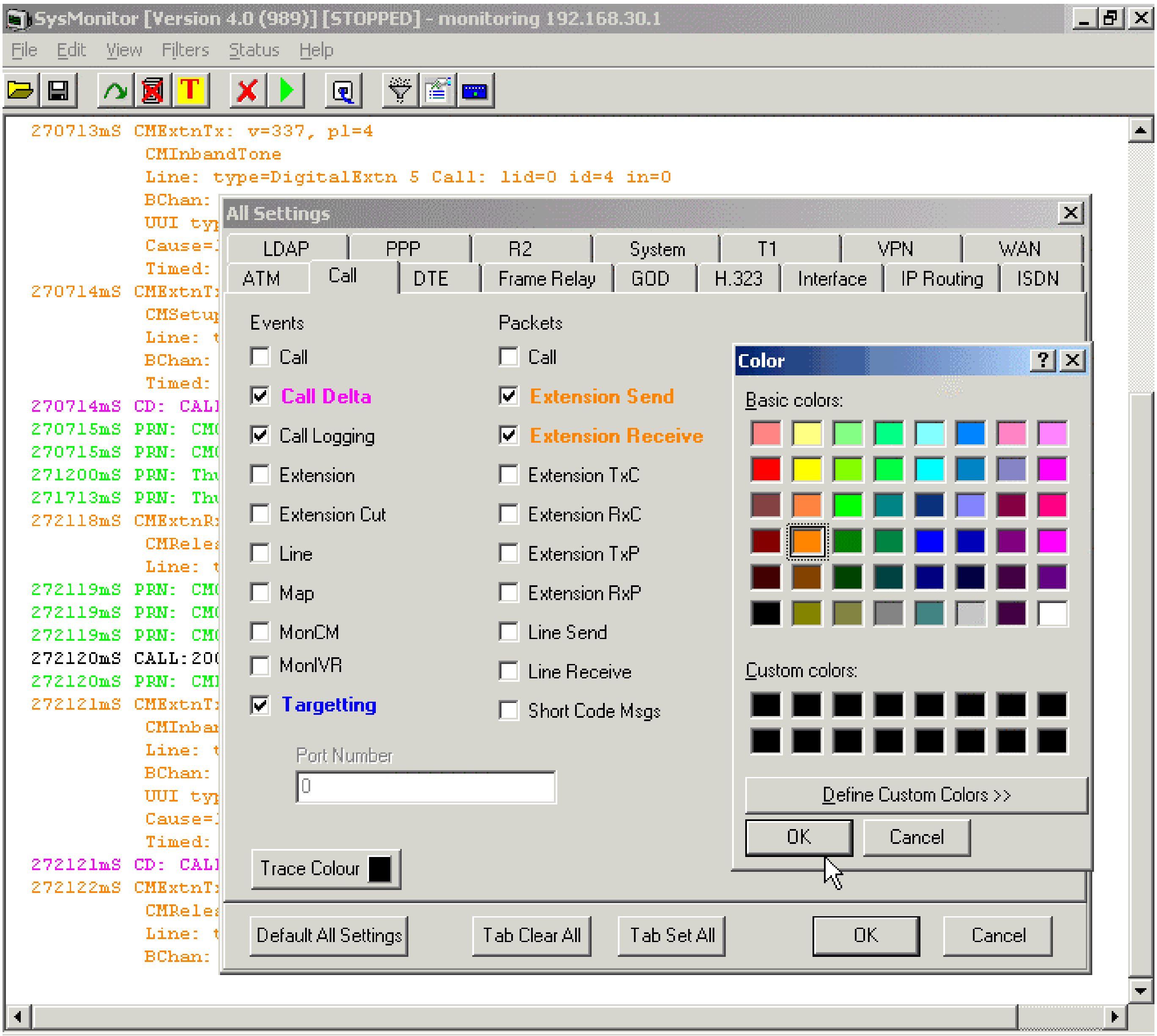
Call Status is extremely useful in establishing if there are any data calls in progress.

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).

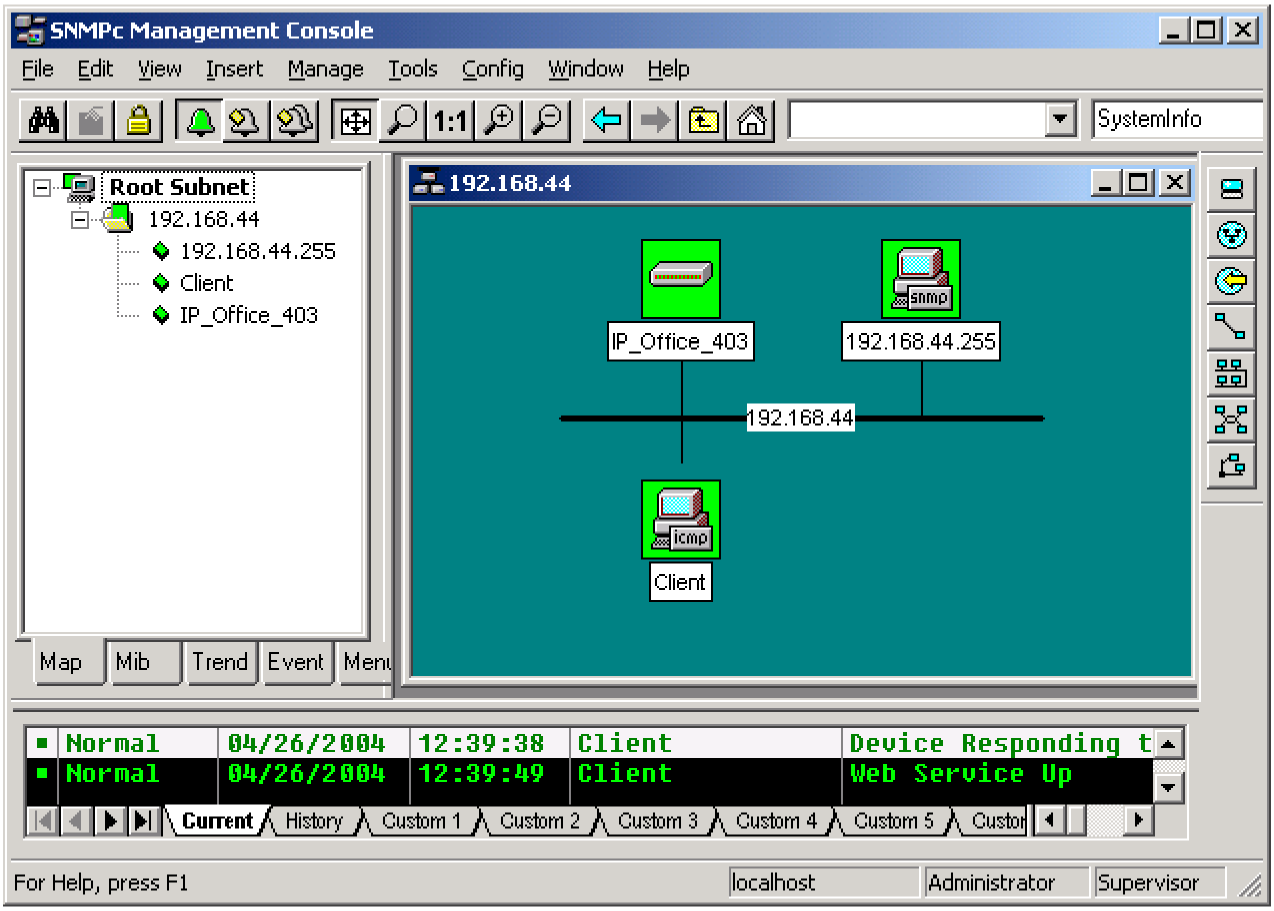
A simple interface allows an engineer to select which protocols and interfaces are to be monitored and decoded. The trace can either be captured directly to screen or as a log file for later analysis.

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 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.

A: Configurations and Factory Build Options

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.

- IP401 Compact Office Digital Terminal 4
- IP403 Compact Office DT
- IP406 Office
- IP412 Office
- Factory Configurations
- Country Availability

IP401 Compact Office Digital Terminal 4

Scenario:

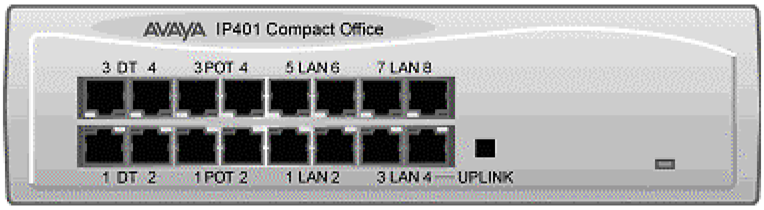
A customer requiring a voice and data solution for a warehousing facility remote from a regional office. Three administrators and two pickers staff the warehouse.

This configuration provides support for four Avaya 20 series digital terminals, one for each of the administrators, leaving a spare port for future growth. Up to four analog telephones (POTS), two of which support a DECT wireless solution to allow the pickers freedom of movement, with one of the remaining ports being used for a fax machine.

The eight port 10/100M Hub allows the local PCs and Printers to be networked. Connectivity for all voice and data traffic between the Warehouse and the regional office is carried over the optional IP401 WAN interface using Voice over IP and standards based compression (through the optional IP400 VCM 5 media card). Two ISDN ports allow up to four simultaneous calls to the public network.

Kit List

- IP401 Compact Office DT4.
- IP401 Compact Office WAN Expansion.
- IP400 Office Voice Compression Module 5.
- 2 x 2030 Display Terminals.



IP403 Compact Office DT

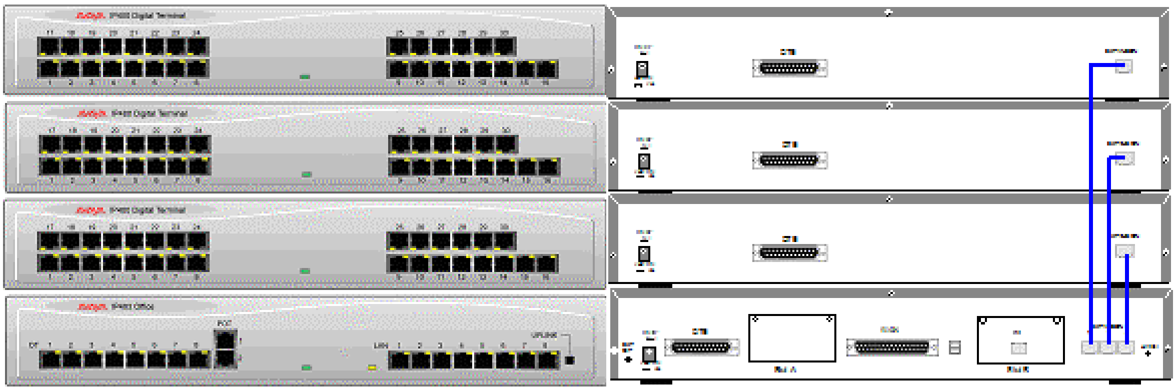
Scenario:

A customer with sophisticated telephony requirements, needing 30 exchange lines and 80 Display Terminals.

This configuration provides support for 98 Avaya 20 series digital terminals (18 spare for growth) and a single Primary Rate ISDN connection. If growth beyond 18 users or additional line capacity were anticipated, the IP406 Office would be considered more appropriate. Typically, a business of this size would have a data network built using LAN switches such as the Avaya Cajun range. The IP403 Compact Office would be connected to the data network through its integral 8 port Hub, providing all users access to the Internet and IP Office productivity applications.

Kit List

- IP403 Office DT PRI 30 E1.
- 3 x IP400 Digital Terminal Module 30.
- 80 x 2030 Display Terminals.



IP406 Office

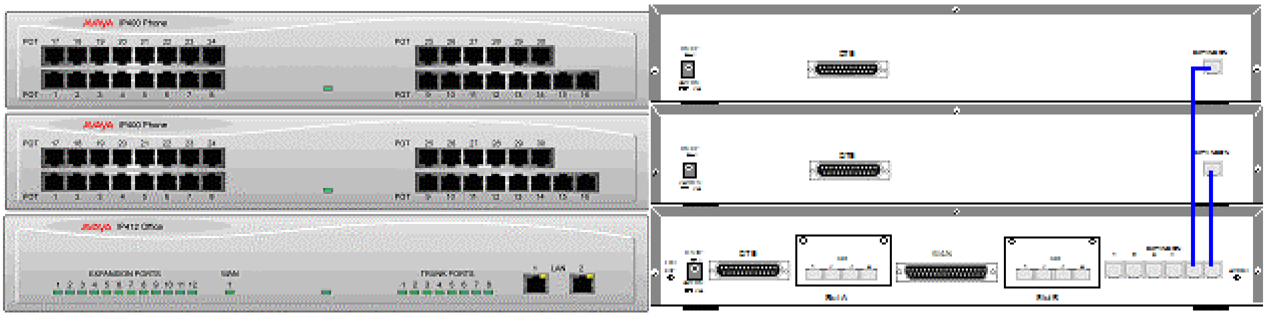
Scenario 1:

A business requiring 60 analog Telephones and 8 Basic Rate ISDN lines (16 channels).

The IP406 Office BRI 16 with two IP400 Office Phone 30 modules provides the required line and extension capacity. Through the use of Phone Manager Lite the functionality provided by the Analog Telephones is greatly enhanced. Expansion capability for an additional 4 Modules allows the system to be expanded to a full 180 extensions. Additional lines can be added by replacing one of the BRI interfaces for a Primary rate.

Kit List

- IP406 Office BRI 16.
- 2 x IP400 Office Phone Module 30.



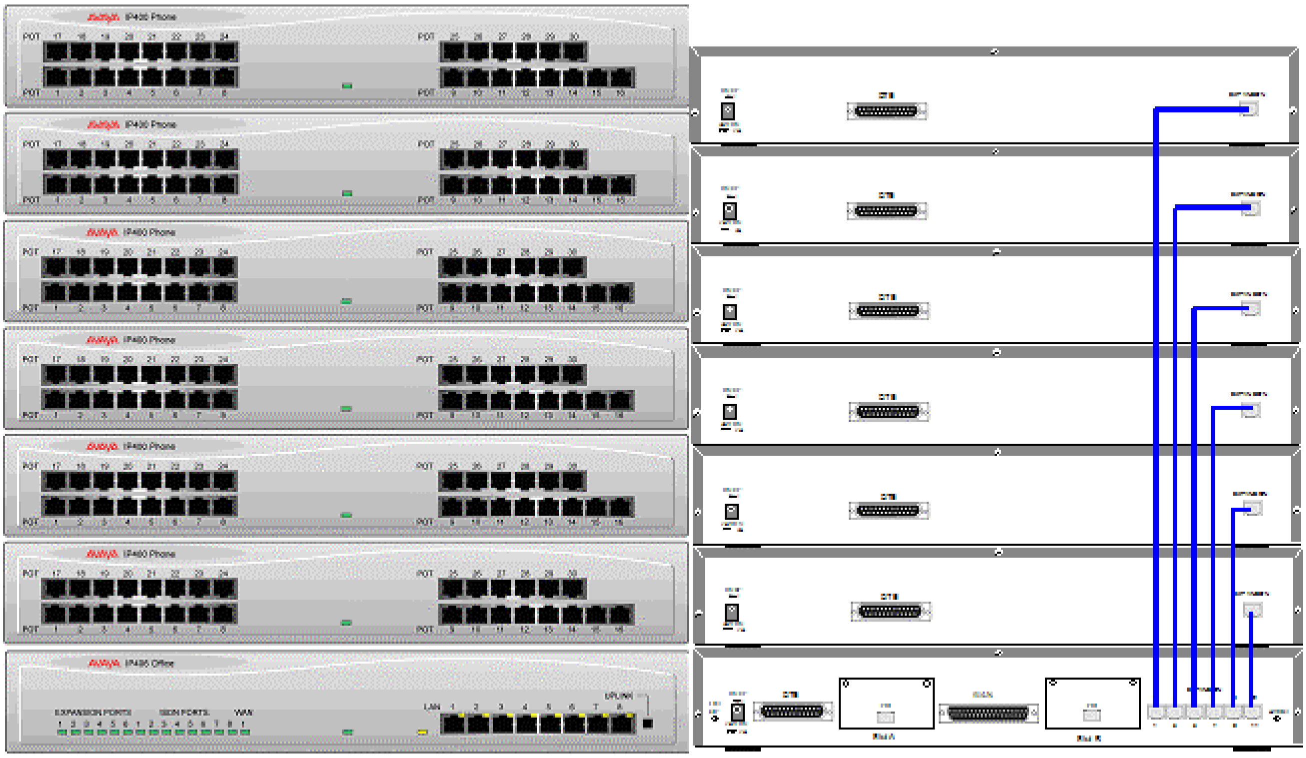
Scenario 2:

A business requiring 180 analog Telephones and 60 lines.

The configuration illustrates a fully configured IP406 Office providing 180 extensions and 60 trunks. Factory shipped with a single PRI the system is fitted with an extra trunk card in its spare slot to provide the additional 30 lines.

Kit List

- IP406 Office PRI 30 E1.
- IP400 PRI E1 Media Card.
- 6 x IP400 Office Phone Module 30.



IP412 Office

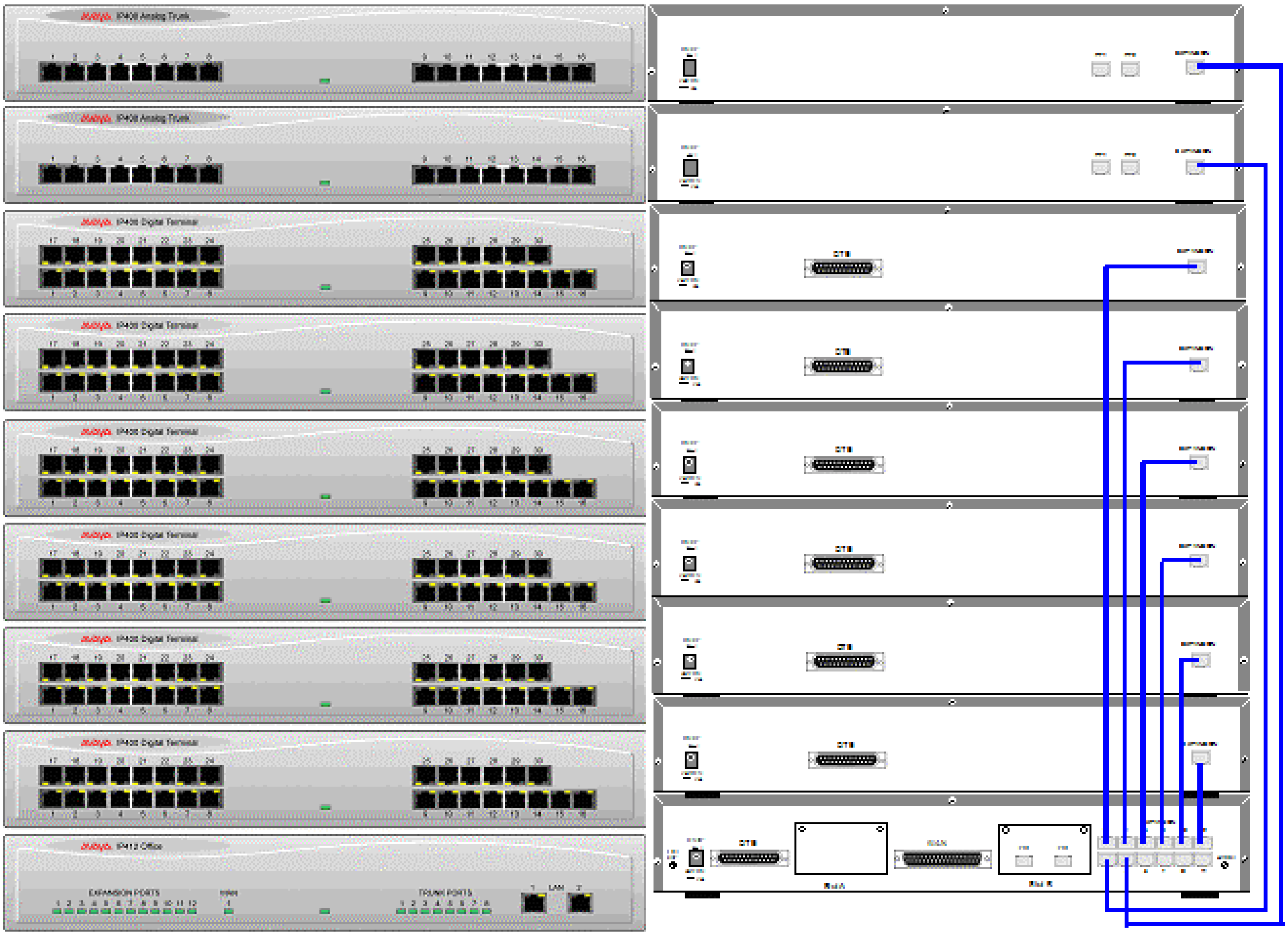
Scenario 1:

A North American business requiring 180 Display Telephones and 96 Digital lines with 20 Analog lines for fall back purposes in the event of the T1 service failing.

The configuration illustrates a IP412 Office 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

- IP412 Office PRI 48 T1.
- IP400 Office Dual PRI T1.
- 6 X IP400 Office Digital Station 30 Module.
- 2 x IP400 Office Analog Trunk 16.
- 180 x Avaya 6412 Digital Terminals.



Scenario 2:

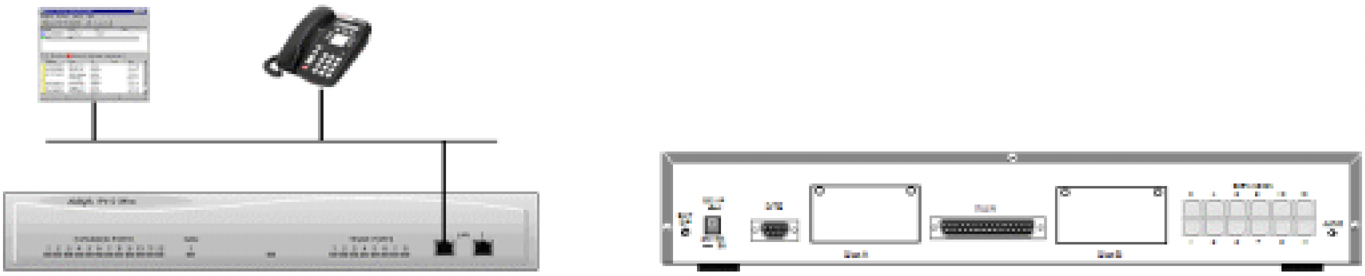
A Business requiring 90 IP hardphones, 90 IP softphones and 60 lines.

This configuration illustrates an IP412 Office PRI 60 E1 fitted with two optional IP400 Office Voice Compression Module 30s. These two internally fitted cards allow up 60 simultaneous calls to external parties, as they are only used when an IP extension is calling a non-IP telephone or line. If less 'Gatewayed' calls were required, one of the 30 channel cards could be substituted for a smaller variant.

The IP Office softphone is 'iPhone Manager Pro' which requires two types of License Keys which allow Phone Manager Lite, supplied as standard, to run as IP Extensions.

Kit List

- IP412 Office PRI 60 E1.
- 2 x IP400 Voice Compression Module 30.
- 90 x 4612 IP Hardphones.
- IP400 Phone Manager Pro RFA.
- IP400 iPhone Manager Pro RFA 50.
- IP400 iPhone Manager Pro RFA 40 (50+40 = 90).



Factory Configurations

Avaya IP Office - Small Office Edition Control Units

- **Avaya IP Office Small Office Edition - 2T+4A(3 VC) US *(700280167)***
  Providing two US specification analog trunks and four analog extensions. Comes with three voice compression resources as standard for VoIP applications. Also includes twin PCMCIA slot for voicemail and Wireless access point, four port Ethernet switch, single Ethernet WAN port and a slot for optional V35, X21, T1 WAN.
- **Avaya IP Office Small Office Edition - 2T+4A(3 VC) INT *(700280175)***
  Providing two analog trunks (not US) and four analog extensions. Comes with three voice compression resources as standard for VoIP applications. Also includes twin PCMCIA slot for voicemail and Wireless access point, four port Ethernet switch, single Ethernet WAN port and a slot for optional V35, X21, BRI, T1 WAN.
- **Avaya IP Office Small Office Edition - 4T+8A(3 VC) US *(700293053)***
  Providing two US specification analog trunks and eight analog extensions. Comes with three voice compression resources as standard for VoIP applications. Also includes twin PCMCIA slot for voicemail and Wireless access point, four port Ethernet switch, single Ethernet WAN port and a slot for optional V35, X21, T1 WAN.
- **Avaya IP Office Small Office Edition - 4T+8A 3 VC) INT *(700293061)***
  Providing two analog trunks (not US) and eight analog extensions. Comes with three voice compression resources as standard for VoIP applications. Also includes twin PCMCIA slot for voicemail and Wireless access point, four port Ethernet switch, single Ethernet WAN port and a slot for optional V35, X21, BRI, T1 WAN.
- **Avaya IP Office Small Office Edition - 4T+4A+ 8DS(3 VC) US *(700280183)***
  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. Also includes twin PCMCIA slot for voicemail and Wireless access point, four port Ethernet switch, single Ethernet WAN port and a slot for optional V35,X21, T1 WAN.
- **Avaya IP Office Small Office Edition - 4T+4A+8DS(16 VC) US *(700280191)***
  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. Also includes twin PCMCIA slot for voicemail and Wireless access point, four port Ethernet switch, single Ethernet WAN port and a slot for optional V35, X21, T1 WAN.
- **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. Also includes twin PCMCIA slot for voicemail and Wireless access point, four port Ethernet switch, single Ethernet WAN port and a slot for optional V35, X21, BRI, T1 WAN.
- **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. Also includes twin PCMCIA slot for voicemail and Wireless access point, four port Ethernet switch, single Ethernet WAN port and a slot for optional V35, X21, BRI, T1 WAN.
- **Avaya IP Office Small Office Edition - 4T+4A+8DT(3 VC) INT *(700280225)***
  Providing four analog trunks (not US), four analog extensions and eight Digital Terminals. Comes with three voice compression resources as standard for VoIP applications. Also includes twin PCMCIA slot for voicemail and Wireless access point, four port Ethernet switch, single Ethernet WAN port and a slot for optional V35, X21, BRI, T1 WAN.
- **Avaya IP Office Small Office Edition - 4T+4A+8DT(16 VC) INT *(700280233)***
  Providing four analog trunks (not US), four analog extensions and eight Digital Terminals. Comes with sixteen voice compression resources as standard for VoIP applications. Also includes twin PCMCIA slot for voicemail and Wireless access point, four port Ethernet switch, single Ethernet WAN port and a slot for optional V35,X21,BRI, T1 WAN.

Avaya IP Office - Small Office Edition Expansion Cards

- **Avaya IP Office Small Office Edition - WAN Expansion Kit *(700289713)***
  Optional card supporting V.35 and X.21 private circuits.
- **Avaya IP Office Small Office Edition - 64MB Memory Card *(700289721)***
  64M PCMCIA memory card for embedded auto-attendant and voicemail.
- **Avaya IP Office Small Office Edition - Wireless LAN Card *(700289739)***
  PCMCIA Wireless card for Access Point functionality.

Factory Configured Base Modules

- **IP401 Compact Office DT2 *(700184617)***
  IP401 Compact Office supporting - 2 x 20 Series Digital Terminals, 2 Plain Ordinary Telephones (CLI), 1 x ISDN Basic Rate (2 lines), 4 Port 10/100M Ethernet Hub, USB port and DTE port for Terminal Adaptor or TPAD applications, Music on Hold input and 2-switch external Door relay control port.
- **IP401 compact Office DT4 *(700184633)***
  IP401 Compact Office supporting - 4 x 20 Series Digital Terminals, 4 Plain Ordinary Telephones (CLI), 2 x ISDN Basic Rate (4 lines), 8 Port 10/100M Ethernet Hub, USB port and DTE port for Terminal Adaptor or TPAD applications, Music on Hold input and 2-switch external Door relay control port.
- **IP403 Office DT-BRI 8 *(700184641)***
  IP403 Office Digital Terminal supporting - 8 x 20 Series Digital Terminals, 2 Plain Ordinary Telephones (CLI), 8 port Ethernet Hub, USB port and DTE port for Terminal Adapter or TPAD applications, 1 x X.21/V35 WAN port, Music on Hold input and 2-switch external Door relay control port. Pre-configured in the factory with a quad ISDN Basic Rate card giving 8 lines (IP400 Office Quad BRI). Expandable by 3 Expansion Modules.
- **IP403 Office DT-PRI 30 E1 *(700184658)***
  IP403 Office Digital Terminal supporting - 8 x 20 Series Digital Terminals, 2 Plain Ordinary Telephones (CLI), 8 port Ethernet Hub, USB port and DTE port for Terminal Adapter or TPAD applications, 1 x X.21/V35 WAN port, Music on Hold input and 2-switch external Door relay control port. Pre-configured in the factory with a ISDN Primary Rate/E1 card giving 30 lines (IP400 Office PRI E1). Expandable by 3 Expansion Modules.
- **IP403 Office DS-PRI 24 T1 *(700184666)***
  IP403 Office Digital Station supporting - 8 x 44 or 64 Series Digital Terminals, 2 Plain Ordinary Telephones (CLI), 8 port Ethernet Hub, USB port and DTE port for Terminal Adapter or TPAD applications, 1 x X.21/V35 WAN port, Music on Hold input and 2-switch external Door relay control port. Pre-configured in the factory with a T1 card giving 24 lines (IP400 Office PRI T1). Expandable by 3 Expansion Modules.
- **IP403 Office DS-Analog 4 *(700184674)***
  IP403 Office Digital Station supporting - 8 x 44 or 64 Series Digital Terminals, 2 Plain Ordinary Telephones (CLI), 8 port Ethernet Hub, USB port and DTE port for Terminal Adapter or TPAD applications, 1 x X.21/V35 WAN port, Music on Hold input and 2-switch external Door relay control port. Pre-configured in the factory with a quad analog loop start trunk card (IP400 Office Quad Analog Trunk (Loop Start)). Expandable by 3 Expansion Modules.
- **IP406 Office BRI 16 *(700184682)***
  IP406 Office supporting - 8 port Ethernet Hub, DTE port, 1 x X.21/V35 WAN port, Music on Hold input and 2-switch external Door relay control port. Pre-configured in the factory with two quad ISDN Basic Rate card giving 16 lines (IP400 Office Quad BRI). Expandable by 6 Expansion Modules.
- **IP406 Office PRI 30 E1 *(700184690)***
  IP406 Office supporting - 8 port Ethernet Hub, DTE, 1 x X.21/V35 WAN port, Music on Hold input and 2-switch external Door relay control port. Pre-configured in the factory with an ISDN Primary Rate/E1 card giving 30 lines (IP400 Office PRI E1). Expandable by 6 Expansion Modules.
- **IP406 Office PRI 24 T1 *(700184708)***
  IP406 Office supporting - 8 port Ethernet Hub, DTE port, 1 x X.21/V35 WAN port, Music on Hold input and 2-switch external Door relay control port. Pre-configured in the factory with a T1 card giving 24 lines (IP400 Office PRI T1). Expandable by 6 Expansion Modules.
- **IP406 Office Analog 4 *(700184716)***
  IP406 Office supporting - 8 port Ethernet Hub, DTE port, 1 x X.21/V35 WAN port, Music on Hold input and 2-switch external Door relay control port. Pre-configured in the factory with a quad analog loop start trunk card (IP400 Office Quad Analog Trunk (Loop Start)). Expandable by 6 Expansion Modules.
- **IP412 Office PRI 30 E1 *(700184724)***
  IP412 Office supporting - 2 port Ethernet switch, DTE Port, 1 x X.21/V35 WAN port, Music on Hold input

and 2-switch external Door relay control port. Pre-configured in the factory with an ISDN Primary Rate/E1 card giving 30 lines (IP400 Office PRI E1). Expandable by 12 Expansion Modules.

- **IP412 Office PRI 60 E1** ***(700184732)***
  IP412 Office supporting - 2 port Ethernet switch, DTE Port, 1 x X.21/V35 WAN port, Music on Hold input and 2-switch external Door relay control port. Pre-configured in the factory with a dual ISDN Primary Rate card giving 60 lines (IP400 Office Dual PRI E1). Expandable by 12 Expansion Modules.
- **IP412 Office PRI 24 T1** ***(700184740)***
  IP412 Office supporting - 2 port Ethernet switch, DTE port, 1 x X.21/V35 WAN port, Music on Hold input and 2-switch external Door relay control port. Pre-configured in the factory with a T1 card giving 24 lines (IP400 Office PRI T1). Expandable by 12 Expansion Modules.
- **IP412 Office PRI 48 T1** ***(700184757)***
  IP412 Office supporting - 2 port Ethernet switch, DTE port, 1 x X.21/V35 WAN port, Music on Hold input and 2-switch external Door relay control port. Pre-configured in the factory with a dual T1 card giving 48 lines (IP400 Office Dual PRI T1). Expandable by 12 Expansion Modules.

IP401 Compact Office Upgrades

- **IP401 Compact Office WAN Expansion** ***(700185093)***
  IP401 Compact Office WAN Expansion is an internally fitted card for the IP401 Compact Office providing a single X.21/V.35 WAN port.
- **IP401 Compact Office Memory Expansion** ***(700198351)***
  IP401 Compact Office Memory Expansion is a plug in SIM required for the embedded voicemail option. Not required for the external Voicemail Lite.
- **IP401 Compact Office Expansion Digital Terminal 2** ***(700185085)***
  IP401 Compact Office Expansion Digital Terminal 2 is a kit for converting an IP401 Compact Office DT2 into an IP401 compact Office DT4.

IP Office Expansion Modules

- **IP400 Office Phone Module 8** ***(700184773)***
  Adds an additional 8 Plain Ordinary Telephone ports to the IP403, IP406 and IP412.
- **IP400 Office Phone Module 16** ***(700184781)***
  Adds an additional 16 Plain Ordinary Telephone ports to the IP403, IP406 and IP412.
- **IP400 Office Phone Module 30** ***(700184799)***
  Adds an additional 30 Plain Ordinary Telephone ports to the IP403, IP406 and IP412.
- **IP400 Office Digital Terminal Module 16** ***(700185606)***
  Adds an additional 16 Digital Terminal ports to the IP403, IP406 and IP412, to support 20 Series Digital Terminals.
- **IP400 Office Digital Terminal Module 30** ***(700185069)***
  Adds an additional 30 Digital Terminal ports to the IP403, IP406 and IP412, to support 20 Series Digital Terminals.
- **IP400 Office Digital Station Module 16** ***(700184807)***
  Adds an additional 16 Digital Station ports to the IP403, IP406 and IP412, to support 44 and 64 Series Digital Terminals.
- **IP400 Office Digital Station Module 30** ***(700184880)***
  Adds an additional 30 Digital Station ports to the IP403, IP406 and IP412, to support 44 and 64 Series Digital Terminals.
- **IP400 Office So8 Module** ***(700185077)***
  Provides 8 ISDN device lines to the desktop.
- **IP400 Office Analog Trunk 16 - North America only** ***(700211360)***
  Provides an additional 16 Analog trunks (loop start or ground start) and two power fail sockets.
- **IP400 Office Analog Trunk 16 EU** ***(700241680)***
  Provides an additional 16 Analog trunks (loop start) and two power fail sockets. European CTR21 specification.
- **IP400 Office Analog Trunk 16 NZ** ***(700241698)***
  Provides an additional 16 Analog trunks (loop start) and two power fail sockets. New Zealand specification.

- **IP400 Office 10/100 WAN 3 Module *(700262009)***
  Provides an additional three X.21/V.35 ports. This expansion module is connected back to the IP403, IP406 and IP412 using the LAN and does not impact on the maximum number of Expansion Modules supported.

Internal Daughter Cards

Voice Compression Modules

- **IP400 Office Voice Compression Module 5 *(700185119)***
  5 Channel Voice Compression module required for IP Trunking and IP extensions.
- **IP400 Office Voice Compression Module 10 *(700185127)***
  10 Channel Voice Compression module required for IP Trunking and IP extensions.
- **IP400 Office Voice Compression Module 20 *(700185135)***
  20 Channel Voice Compression module required for IP Trunking and IP extensions.
- **IP400 Office Voice Compression Module 30 *(700293939)***
  30 Channel Voice Compression module required for IP Trunking and IP extensions.

Modems

- **IP400 Office Modem 2 *(700185226)***
  Internally fitted card allowing two simultaneous V.90 modem call.

Trunk Interface Cards

- **IP400 Office BRI-8 (UNI) *(700262017)***
  Interface card for the IP403, IP406 and IP412 providing 4 x ISDN basic rate ports (8 lines).
- **IP400 Office PRI 30 E1 (1.4) *(700272461)***
  Interface card for the IP403, IP406 and IP412 providing 1 x ISDN Primary rate port (30 lines).
- **IP400 PRI 30 E1R2 RJ45 EXP - CALA *(700241631)***
  Interface card for the IP403, IP406 and IP412 providing 1 x E1R2 Primary rate port (30 lines). RJ45 termination.
- **IP400 PRI 30 E1R2 COAX EXP - CALA *(700241656)***
  Interface card for the IP403, 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 IP403, IP406 and IP412 providing 2 x ISDN Primary rate ports (60 lines).
- **IP400 Office PRI T1 *(700185200)***
  Interface card for the IP403, IP406 and IP412 providing 1 x T1/PRI port (24 lines).
- **IP400 Office Dual PRI T1 *(700185218)***
  Interface card for the IP403, IP406 and IP412 providing 2 x T1/PRI (48 lines).
- **IP400 Office Quad Analog Trunk (Loop Start) *(700185192)***
  Interface card for the IP403, IP406 and IP412 providing 4 x Loop start analog trunks (North American specification).
- **IP400 ANLG 4 EU (LS) EXP *(700241672)***
  Interface card for the IP403, IP406 and IP412 providing 4 x Loop start analog trunks (European CTR21 specification).
- **IP400 ANLG 4 NZ (LS) EXP *(700241706)***
  Interface card for the IP403, IP406 and IP412 providing 4 x Loop start analog trunks (New Zealand specification).

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**
- **Hungary**
- **Iceland**
- **Ireland**
- **Italy**
- **Korea**
- **Luxembourg**
- **Mexico**
- **Netherlands**
- **New Zealand**
- **Norway**
- **Peru**
- **Poland**
- **Portugal**
- **Russia**
- **Spain**
- **Sweden**
- **Switzerland**
- **United Kingdom**
- **USA**

B: Implementing Voice over IP FAQ

What is Quality of Service?

First and foremost, Quality of Service is a goal not a standard. There are a number of measures that can be taken on the Local Area Network and Wide Area Network to make them 'good enough' for voice traffic. Some of these are standard based, others simply a matter of network architecture.

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 practice. To others the additional expense may be viewed as unnecessary.

What are the Symptoms of Quality Problems?

Poor speech quality manifests itself in three distinct ways.

- Echo induced by delay.
- Warble to severe clipping induced by lost packets and variable delay (jitter).
- Distortion as a result of errors introduced by the conversion of speech to data and back again.

How Do I Minimize Delay Induced Echo In My Network?

Delay in a network comes from a number of different sources and phenomena. The first source of delay comes from the process of converting speech to VoIP packets. The IP Office supports a number of standards based encoding methods to allow the optimum trade off between delay and bandwidth to be made (see What Bandwidth Do I Require For Each Voice Call?). IP Office also incorporates integral echo cancellation to maximize speech quality.

The next source of delay comes from data and voice traffic queuing at the ports of the switches, routers and bridges making 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 in the network the IP Office prioritizes voice traffic using a standard known as DiffServ. This marks each IP packet carrying voice with a flag so that switches and routers can force packets containing voice to the front of the 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 (UDP Port) is being used. All voice traffic is carried using two identifying protocols RTP and RTCP. Both methods are equally good, leaving the choice of which to use as being between the most cost effective and the easiest to implement and manage.

Another source of delay can come from collisions of a particular segment of the Local Area Network. 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 the way of life on most 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 very little.

How Do I Minimize Warble and Clipping In My Network?

As mentioned earlier warble and clipping are symptoms of variable delay (Jitter) and packet loss. 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 that will 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 it can be ignored. Where packet loss is excessive (that is greater than 2%) the cause should be established and fixed. This could be due to a fault or simply an over worked device discarding packets.

Avaya's Cajun Ethernet switches are an ideal complement to IP Office as they have been engineered to minimize delay, jitter and packet loss.

How Do I Minimize Distortion In My Network?

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 will be.

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

Different coding methods will have different effects. IP Office supports a range of methods to allow you to choose the one with the right quality vs bandwidth for your network. Generally speaking double conversions should be avoided wherever possible.

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

IP office will allow 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. When using IP Office with Avaya Cajun LAN switches, it is even possible to divide the data bandwidth to provide guarantees for different types of data traffic such as SAP or e-business applications.

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.

Compression 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?

Effort should be made to keep the overall end-to-end delay below 150 milli-seconds.

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 data network, the ultimate scenario for Voice would be for every device on the Local Area Network to have its own dedicated port on a DiffServ capable layer 3 switch such as the Avaya Cajun. Connections to the Wide Area Network should, once again, be via devices supporting DiffServ such as the 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 Wide Area Network 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 That IP Office Supports

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

- The IP401 Compact Office 2/4, IP403 Office and IP406 Office can each be fitted with a single module offering 5, 10 or 20 simultaneous calls.
- The IP412 Office is capable of supporting two modules, including a 30 channel module that is exclusive to the IP412 Office, allowing between 5 and 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, it is supported between IP Office systems and Avaya Communication Manager. IP Office supports fax speeds up to 14.4 Kbps, it does not confirm to the T.38 standard. The bandwidth requirements for the call will initially be as per the specified or negotiated compression method then the bandwidth requirement will change to accommodate the Fax data. The Fax bandwidth will vary depending on the speed that the Fax devices are communicating at 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.

Network Assessment

With IP Office, optimum network configurations can support VoIP with perceived voice quality equivalent to that of the Public Switched Telephone Network (PSTN). However, not every network is able to take advantage of packet voice 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 with IP phones 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 Voice over IP 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 involved 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.
- A measurement of packet loss, jitter, and delay over the course of multiple days while measured on a per minute basis. A graphical representation of the data is the preferred output method.
- Examination of QoS/CoS parameters in place in the network.
- Summarization 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 Avaya Support.

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.

Voice over IP Relevant Standards Supported

The 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.

C: 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
- LineHandoff
- LineHold
- LineInitialiseEx
- LineMakeCall
- LineNegotiateTAPIVersion
- LineOpen
- LinePark
- LineRedirect
- LineRemoveFromConference
- LineSetAppPriority
- LineSetAppSpecific
- LineSetCallPrivilege
- LineSetStatusMessages
- LineSetupTransfer
- LineShutdown
- LineSwapHold
- LineUnhold
- LineUnpark

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 is 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 2.0 developers SDK CD. The following table shows the device specific data available via TAPI. A "Y" in the column indicates that the field is already described in the TAPI manual.

- 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

D: Technical Specifications

General

Dimensions

Unit Dimensions (mm/inches)	Width	Height	Depth
IP401 Compact Office	255mm/10"	71mm/2.8"	235mm/9.3"
IP403, IP406, 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"	235mm/9.3"

- The recommended minimum clearance, front and rear, for the connection of cables and other devices is 62mm/3".

Environmental

- 0 to +40C. 95% relative humidity, non-condensing.

Terminal/Extension Cable Lengths

The following table details the maximum cable lengths supported for the telephone range using AWG22, 24 and 26 cabling.

Phone	AWG22	AWG24 (~ 0.5mm Ø)	AWG26
20 Series	1km - 3280 feet	1km - 3280 feet	0.5km - 1640 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
64 Series	1km - 3280 feet	1km - 3280 feet	0.4km - 1310 feet
POT's	1km - 3280 feet	1km - 3280 feet	0.5km - 1640 feet

Weight & Power Consumption

Unit	Weight	Power Consumption (Nominal Watts)
IP401 Control Unit	1.2Kg/2.6lbs	24
IP403 Control Unit	2.6Kg/5.8lbs	26
IP406 Control Unit	3.0Kg/6.7lbs	16
IP412 Control Unit	3.0Kg/6.7lbs	17.5
IP Office - Small Office Edition	1.2Kg/2.6lbs	–
Analog 16 Module	2.9Kg/6.5lbs	5
DT/DS 16 Module	3.0Kg/6.7lbs	24
DT/DS 30 Module	3.5Kg/7.8lbs	30
WAN3 Module	2.8Kg/6.3lbs	12
So8 Module	2.8Kg/6.3lbs	24
Phone 8 Module	2.8Kg/6.3lbs	12
Phone 16 Module	2.9Kg/6.5lbs	16
Phone 30 Module	3.1Kg/6.94lbs	30

Power Supply

- **Input**
  - **Small Office Edition:** 2.5mm DC inlet socket. 24Vdc power input. Rating 24Vdc, 1.8A maximum.
  - **All Other Units:** 2.5mm DC inlet socket. 24Vdc power input. Rating 24Vdc, 2A maximum.
- **Power Supply Units:** All CE/UL/Dentori Safety Approved.
  - **Small Office Power Supply Unit**
    - Input: 100-240Vac, 50/60Hz, 81-115VA, 1.5A maximum.
    - Output: 24Vdc, 1.875A, output power 45W maximum.
  - **Standard Power Supply Unit**
    - Input: 100-240Vac, 50/60Hz, 81-115VA, 2A maximum.
    - Output: 24Vdc, 1.875A, output power 45W maximum.
  - **DS30 80W Power Supply Unit**
    - Input: 100-240Vac, 50/60Hz, 81-115VA, 2A maximum.
    - Output: 16Vdc, 5A, output power 80W maximum.

Interfaces

Interface	Information
DTE Port	• 25 way D-Type female connector, V.24/V.28. • 9 way D-type on IP412 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 ports	• RJ45 sockets: Loop start/Ground start (regional dependant)
Power Fail ports	• RJ45 sockets: Telephone ports acts as master socket
ISDN Data Rates	• BRI: B-channel 64kbps or 56kbps, D-channel 16kbps. • PRI: B-channel 64kbps or 56kbps, D-channel 64kbps.
Telephone	• RJ45 sockets: EU - Telephone ports act as Master sockets. • CLI Schemes: DTMFA, DTMFC, DTMFD, FSK and UK20. • REN = 2. • Off Hook current = 25mA. • Ring Voltage = 40V (nominal) RMS. • External Bell (via POT port); REN = 1
LAN	• RJ45 sockets. Auto-negotiating 10/100BaseT Ethernet (10Mbps). Port 8 is MDI/MDIX switchable via the adjacent Cascade push button switch.
WAN	• Small Office Edition: RJ45 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 a.c. signal – 200mV rms.
External Output Port	• 3.5mm Stereo Jack socket. Switching Capacity - 0.7A. • Maximum Voltage - 55V d.c. On state resistance - 0.7. • Short circuit current - 1A. Reverse circuit current capacity - 1.4A.
Wireless Module	• 16bit Type II PCMCIA format PC card. • IEEE 802.11b WiFi.
Embedded Voice Memory	• 16bit Type II PCMCIA format PC card. • 64MB Flash Memory.

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.
PAP	RFC1334	Password Authentication 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.
IPCP	RFC1332	Internet protocol Control Protocol.
TCP/IP	RFC793	Transmission Control Protocol/Internet 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.
Entity MIB	RFC2737	Entity MIB (Version 2).

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 from voicemail which, if it returns again to voicemail, will be treated as a new call.

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 terminals in the 20xx series. Supported by DS sockets on IP Office control units and Digit Station modules. Note: Not all terminals in the above range are supported on IP Office.

Digital Terminals: Refers to Avaya terminals in the 24xx, 44xx and 64xx series. Supported by DT sockets on IP Office control units and Digit Terminal 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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D

E

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

Avaya
Sterling Court
15 - 21 Mundells
Welwyn Garden City
Hertfordshire
AL7 1LZ
England

Tel: +44 (0) 1707 392200
Fax: +44 (0) 1707 376933

Email: contact@avaya.com
Web: http://www.avaya.com.