IP Office 4.1

Product Description

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Contents

1. Introduction

Notice

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 your Avaya Representative for further details. This document should be read in conjunction with any issued technical bulletins and/or product offer announcements.

Avaya IP Office Family

The Intelligent Communications solution for small and midsize businesses.

What is IP Office?
A solution for voice and data communications, messaging and customer management. It uses IP technology to deliver more functionality at a lower cost.

How can I use it in my business?
To connect with colleagues and customers... simplify access to information... keep remote workers in touch. To save money through conferencing, networking, time/ call management, Voice over IP and more.

What are my choices?
Does your business have one location? Multiple locations? Are you a branch office of a larger organization? A home office? With IP Office you can choose from a range of models and add capacity, applications and phones, as you need them. Whether you have 2 employees, 200 or more, IP Office is the right choice.

IP Office: Three key things to know
Every small- and medium-size business needs ways to reduce costs and improve the way it operates. Like every business, you're looking to keep all your customers, add new ones and grow at the pace that's right for you. Avaya understands this. With over one hundred years of experience as a leader in communications, we know that the right solution for your business is one that helps you increase profitability, improve productivity and gain competitive advantages.

Get big business communications —at small business prices
Over one million businesses rely on Avaya solutions like Avaya IP Office —the award-winning business communications system that gives growing companies an "all-in-one" solution for telephony, messaging, networking, conferencing, customer management and much more. Growing businesses know they can rely on Avaya for big business capabilities at small business prices—Avaya has an entire division focused on the needs of small- and medium-size businesses. We support extensive research into new technologies and standards, and we make it easy for businesses like yours to acquire our solutions by offering an array of financing options.

See what Avaya can do for you
You need a communications system—every business does. To find one that's right for your business, start with Avaya. With solutions like IP Office, we're revolutionizing how small and medium businesses communicate. Now is the time to see what an Avaya solution can do for your business.

Reduce monthly costs. Now.
IP Office will help you lower the cost of communications, with capabilities like conferencing, making calls over a managed Internet service (Voice over IP) and the "all-in-one" benefits of a converged communications system.

Leave the office. Be accessible.
With easy, flexible options for call/message forwarding and one-number reachability, IP Office keeps everyone in touch. Get the freedom to go where you want and never miss important business calls.

Serve better. Sell more.
IP Office can give you a customer sales and service center designed for your needs and your budget—with all the routing and reporting capabilities you need. Deliver the personal service that builds sales and loyalty.

Get connected.
Talk to your Avaya BusinessPartner. Discuss where you want communications to add value to your business. Learn about the different service and support options that are available. See why thousands of growing businesses rely on the innovative Avaya IP Office solution.

The right choice for you... and your business.
How we communicate is a personal choice—it has to match the needs of your business. And your needs change depending on whether your employees are working in the office, at home, or on the road. That's why when you choose IP Office you can also choose from a whole range of communication tools and applications designed to boost productivity. Choose a basic phone or one with all the bells and whistles. Connect our IP phones directly to your office LAN—also use them at home and get all the features you have at the office. Avaya Phone Manager software can turn the screen of your PC into a phone. And our wireless solutions make it easier to roam the office. With all of our IP Office capabilities, our goal is to make your communications simple and cost-effective. Let your Avaya BusinessPartner put together a selection of tools and applications that's right for you.

Fine-tuning performance.
How many calls are you handling an hour, a day? What are your peak calling periods? How many calls typically turn into sales? Avaya IP Office reporting capabilities can help you measure and manage your availability and response to customers.

Day-to-day administration.
Once your system is up and running you will benefit from the menu-driven administration tools that simplify day-to-day tasks, such as updating directories and moving phone extensions.

Getting started.
Is your communications network ready for IP Office? We'll make sure. Avaya has created a whole set of assessment and automated configuration tools to make sure that when your system is installed it's ready to meet your needs starting Day One.

Keeping ongoing management simple.
Concerned about needing extra resources to administer a system as powerful as IP Office? There's no need for worry. IP Office comes with a whole set of menu-driven tools to keep ongoing management simple.

Does my current phone system give my business what it needs?
If it is based on old technology, probably not. Your competitors will react faster and appear more professional with the latest in communications software. IP Office delivers the capabilities that allow you to keep up with or overtake the competition.

Do I need to understand the technology to implement it?
No. IP Office is designed specifically to give you more functionality without making more demands on your resources. Rely on your certified Avaya BusinessPartner for support before, during and after your purchase. We'll take care of you so you don't have to worry.

Do I need to spend a lot?
Not at all. You have choices based on your budget needs. Easy leasing or financing plans not only make this affordable; they help you quickly cut monthly expenses immediately. And you only have to buy/lease what you need, when you need it.

Is IP technology so new that it's not reliable?
With approximately over 100,000 systems deployed worldwide (Avaya is #1 in IP Telephony shipments*), Avaya IP Office has the track record businesses like yours can rely on. Aside from receiving the Product of the Year award by Internet Telephony magazine and being named Best in Test by Miercom in 2004, customers like you are saving money and boosting productivity. Many are managing the system themselves via menu-driven tools.

I have old systems but am adding an office. Should I consider the new technology?
Not only would this be a way for you to experience the rich functionality of the latest communications applications, but we may be able to network with your existing equipment, as well as provide a gradual migration plan for your other locations.

How quickly can I get up and running?
Just say "when"—an authorized Avaya BusinessPartner can tailor a solution to your needs and your budget. By saving you money and helping you grow, IP Office repays your investment and lets you reallocate resources to other business priorities.

Lowering long distance costs.
Routing phone calls over IP lines—Voice over IP—is growing in popularity. Particularly in the case of international calls, VoIP generates significant savings. If your company is already linking multiple offices using high-speed lines, the VoIP capabilities in IP Office make it possible to route voice calls over the existing infrastructure, providing another way to lower costs and leverage your investment. However you do it, the VoIP capabilities of IP Office are a way to put money back in your pocket.

Eliminating conferencing fees.
For connecting with partners, suppliers and dispersed employees, conference calls keep people working together and keep travel costs down. Many companies rely on third party teleconferencing services and pay a price for the convenience. This is particularly true—and irritating—if a call that's scheduled doesn't happen: you still pay the fee.

Now there's an alternative that will save you money. With Avaya IP Office, your organization can have its own private, secure conference bridge and entirely eliminate fees to third party providers.

Supporting multiple offices/remote workers
When employees can't get to the office (because of storms, medical issues or other reasons) but can still work productively at home, your business benefits. IP Office Phone Manager lets you turn any PC into a phone, making it easy and productive to work anywhere. And the ability to network phone systems and share messaging systems between offices reduces up front investment and drives long-term productivity.

And keep in mind…
IP Office delivers a whole range of capabilities. Only you can put a number on the value that many of these capabilities will have for your business.

Examples:

- Having calls automatically routed to a cell phone or other location, so important customers can get through to the right person in real-time
- Being able to operate as a 24/7 business, without a 24/7 staff
- Using your communications to quickly identify when your top customers call.

How IP Office is benefiting businesses today.

- More room for sales
  With IP Office, a leading provider of commercial food service equipment now handles 50% more calls per day, without extra staff and without sacrificing the personal service it knows is the key to sales.
- At the head of the class
  By relying on IP Office to connect nearly 50 buildings, a public school system saved thousands of dollars on inter-office calls and simplified communications.
- Lowering global costs
  By using IP Office to hold teleconferences and make phone calls across the IP network, a strategic consulting firm is saving up to $30,000 per year.

The right model for your business with several models to choose from, there's an IP Office to meet your needs. Ready to grow Capacities: 2-360 extensions; up to 192 analog lines; 96/120 T1/E1 lines; 128 VoIP trunk lines.

Call handling and messaging.
Get 24-hour support for callers/customers without a 24-hour staff. IP Office has a range of messaging, auto attendant and Interactive Voice Response (IVR) capabilities. Integrate messaging and advanced call handling into your customer service operations. Handle voice mail and email in a single mailbox.

Communication with customers.
Set up a formal or informal customer service center. Integrate your customer data base into your call handling. Manage the quality of your customer interactions.

Work anywhere.
Give your employees all the communications capabilities they have at the office whether they are working from home, a hotel or a remote office.

A complete conferencing solution.
Don’t pay any more fees to outside conferencing service providers. Get Web and audio-based conferencing that are easy to set up and use.

Secure converged communications.
Use IP Office as a secure router with a built-in firewall/VPN. Route voice calls over a managed Internet service (VoIP) and pocket the savings. Simple administration Windows-based, menu-driven tools cut the time and expense of administration.

What's New in IP Office 4.1

For those already familiar with IP Office, this page lists the new features introduced in IP Office 4.1. This is not an exhaustive list, however, it covers the major changes that are aimed at improving product flexibility and end user mobility.

Hardware Support

IP Office Control Unit Support

- IP Office 4.1 is supported on the IP500 as well as the Small Office Edition, IP406 V2 and IP412.
- IP Office 4.1 is not supported on the IP403 and IP406 V1.

Note: Some early IP406 V2 systems (typically pre PCS08) do not have enough memory to run IP Office 4.1 software. If your IP406 V2 system does not have enough memory there is a process in place to enable you to upgrade the system free of charge.

IP500 Universal PRI daughter card

This new card provides digital trunk interfaces for the IP500.

- Each card is configurable to connect to T1, PRI, E1 or E1R2 lines.
- The card is available in either a single or dual PRI variant. The single variant can support up to 24 T1 channels or up to 30 E1 channels. The dual variant can support up to 48 T1 channels or 60 E1 channels.
- On each card, 8 channels are enabled by default. Further channels may be enabled by the purchase of additional licenses in 2-channel or 8-channel increments.
- The IP500 PRI daughter card can be fitted to any IP500 VCM or extension base card (not the Legacy Card Carrier).
- Up to four Universal PRI cards can be installed in any combination in the IP500 chassis (max. 192 T1 or 240 E1 channels).
- Diagnostics capabilities:
  - Visual indicators to show service state
  - Physical test points to monitor traffic

IP500 Expansion Modules

Avaya has introduced new versions of the following IP Office expansion modules. They are functionally identical to the existing IP400 versions but have been refreshed in the look and feel of the IP500 (dark gray color) and require the IP500 rack mounting kit. New modules:

- IP500 DS 16
- IP500 Phone 16
- IP500 Analog Trunk 16 (North American version)
- IP500 BRI So8

Terminal Support

The following terminals are not supported by IP Office 4.1. They may function but have not been tested with 4.1 and any faults reported with 4.1 will not be fixed.

- 20DT Analog DECT used with IP Office Analog DECT and Compact DECT
- 4606, 4612 and 4624 IP phones
- TransTalk 9040

Small Community Networking

- **Support on IP500 running Standard Edition software**
  Since the Q3 2007 Maintenance Release of IP Office Release 4.0 software, the Voice Networking Licenses and Advanced Networking Licenses work on Standard Edition and do not require an upgrade to Professional Edition. Only the main site of the Small Community Network (SCN) requires Professional Edition (since centralized Voicemail Pro would be running).
  - If the customer requires voice messaging, the only messaging supported in the SCN is Voicemail Pro. Therefore, the site running Voicemail Pro would require Professional Edition.
  - If the customer does not require voice messaging at all, all sites in the SCN could operate running Standard Edition while still benefiting from the other SCN features:

Desk to desk dialing.	Absent Text Messaging.	Hot desking across network.*	Camp–On.
Hold / Transfer.	Busy Lamp Field.	Paging.	Remote Hot Desking.*
Call Forward.	Dynamic User Directory.	Call Pick-Up.	Distributed Hunt-Groups.*
Conference.	Centralized Receptionist.	Call Back When Free.	Anti-Tromboning.

*Requires Advanced Small Community Networking licenses.

System Status

- **System Status Application**
  The ability to play back previously recorded logs has been added to SSA in Release 4.1.

Embedded Voicemail

The following enhancements have been added to make the embedded voicemail solution more complete.

- **Up to 40 Auto Attendants on Embedded Voicemail**
  Embedded Voicemail for IP500, IP406 V2 and Small Office Edition allows up to 40 independent Auto Attendants to be configured, which may be linked together to enable multiple tiers (e.g. 8 attendants with 5 levels each, or other similar permutations).
- **Ability to give labels to recordings**
  The Auto Attendant form has a new recording label field added against each 'greeting' option and the 'menu' option. A recording can be referenced by name and re-used in multiple Auto Attendants without the need to create a recording for each occurrence/instance.
- **Embedded Voicemail shutdown feature**
  Since the Q2 2007 Maintenance Release of IP Office Release 4.0 software, IP Office includes a new short code feature – "ShutdownEmbeddedVoicemail". This allows a polite shutdown of the embedded Voicemail card so that it can be removed from the IP500 or IP406 V2 without having to power off the unit. The card LED will extinguish when this short code feature is run to indicate that it is safe to remove the card.

Voicemail Pro

- **Call Transfer Announcements**
  When calls are transferred to a number, Voicemail Pro is able to announce the destination name or number to the caller. This feature is configurable through the Auto Attendant Properties for Transfer.
- **Call Transfer Data Tagging**
  Call Data tagging has been enhanced to support unsupervised transfers so that all call transfers support the ability for Call Data tagging information to be passed with the transferred call.
- **Queue Announcements**
  Voicemail Pro can announce to callers in queue the length of time the call has been in the system, as well as the length of time the call has been in queue. This will help alleviate frustration since callers will know they are not stuck in a non-progressing queue.
- **Variable routing via Call Flow**
  The existing CLI routing call flow action has been enhanced to offer routing by additional variables, including a new DDI/DID variable ($DDI). It also allows the use of wildcards for matching. The action is now known as 'Variable Routing' where the user can select the variable to test against a configured routing condition. The administrator can program conditional re-routing of calls based on queue position or time spent in the queue.
- **Call Recording Enhancements**
  The IP-Office Manager now has the ability to select the mailbox that Hunt Group and Account code recordings should be targeted to.
- **LIFO/FIFO playback**
  Voicemail Pro administrators now have the ability to select the playback order of new or saved messages, either on a last-in/first-out (LIFO) or first-in/first-out (FIFO) basis. This is a system-wide setting.
- **Castelle Fax Server**
  In addition to a number of other supported fax servers, Voicemail Pro now supports Castelle Fax Servers.

Phone Manager Pro

- **Telecommuter Mode**
  Phone Manager Pro allows the making and receiving of calls and the retrieving of voicemails from an external phone number as if they were in the office, with Phone Manager providing the call control. This also provides the convenience of centralized billing as well as potential cost savings for remote workers and mobile work force. To use Telecommuter mode, a Phone Manager Pro license is required for each remote worker.

Twinning of Appearance Keys

Users with internal twinning enabled can now receive line appearance, bridged appearance and coverage calls on their twinned phone.

VPN Phones

- **VPN capability in 4600 and 5600 series phones**
  Release 4.1 supports licenses to allow remote IP handsets to access the IP Office over a secure IPSec Virtual Private Network (VPN) without the need for a separate VPN gateway at the remote location. Note that this requires a compliant VoIP-ready VPN device at the central site; it is not possible to terminate the VPN tunnel directly on the IP Office.

Improved IP DECT Licensing

Pre-licensed IP-DECT base stations are now available making both first time installation and upgrades easier and more flexible.

Other Features

- **Time of Day and Date Routing of calls**
  A new calendar facility has been added to the IP Office Time Profiles to define dates and times for specific operations. This provides the flexibility to use date and time on any feature that makes use of Time Profiles, e.g. for public holidays. This is also supported on Incoming Call Routes.
- **Queue Threshold Alert**
  A new feature has been added that alerts at a selected analog extension port when the number of calls queued against a Hunt Group exceed a threshold. Typically the User to Alert will be a loud ringer or other alerting device.
- **Transport Layer Security (TLS) and enhanced Password policy**
  The security administrator can select whether the session between Manager and IP Office can utilize TLS v1.0. Enabling this option will disable access to versions of Manager that do not have TLS capability. The selection of whether TLS is to be enforced will reside within the security settings, not within general configuration settings.
    - Manager security can be set to none, low, medium or high. For medium or high settings, an idle timeout causes the user name and password to be re-requested before any Manager operation can be continued. Any Manager data is grayed out until the password is successfully entered. Different levels of password expiry and complexity are also supported.
- **Syslog support**
  A Syslog client has been added to IP Office which is capable of reporting administrative changes and security events to up to two external Syslog servers connected via IP (LAN or WAN).
- **Disable Speakerphone**
  This feature gives the administrator the ability to turn off the hands free speakers on both IP Office digital handsets and IP handsets. The default operation is for the speaker to be enabled. When Speakerphone is disabled pressing the Speaker button will have no action and an error bleep will sound (through the speaker). Disable Speakerphone is supported on the 2400, 5400, 4600, 5600 and 6400 series phones. It is not supported on 4400 series phones or T3 series phones (digital, IP or analog).
- **Group Listen**
  IP Office digital handsets now support the "Group Listen" feature, which allows the audio path to be two-way on the handset (or headset) while the speaker is one-way listen only. This allows the person with the handset to speak to the far-end, while everyone else in the room can hear the responses (with side discussions, etc. that would not be picked up by the handset). Group Listen is only supported on the 2402, 2410, 2420, 5402, 5410, 5420, 6408, 6416, 6424 phones. It is not supported on IP phones.
- **Manager Start-up Banner**
  On starting Manager, an optional banner can display textual information stored within a text file. The user will be required to select to continue using Manager. If the text file cannot be found then no information will be presented.
- **Manager Warning Dialog**
  Manager now includes a small text field that might be used to record general notes about a site or contact details for a centrally managed system. When used with the appropriate flags, this feature can be used to avoid conflicts when more than one simultaneous attempt is made to configure the IP Office.
- **LAN 2 interface on port 8 of the IP406V2**
  The IP406V2 provides the facility to configure a second logical LAN interface on port 8 of the built-in Ethernet switch. Once enabled, the LAN 2 interface is available as an IP route destination.

Voice Communication Solution Features

IP Office offers 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 PABX's.
- **Integrated H.323 Gatekeeper and Gateway for converged communications**
  The IP Office acts as an IP telephony server with Quality of Service (QoS) support through DiffServ for routing and up to 128ms of Echo cancellation depending on VCM card fitted.
- **SIP Trunking**
  IP Office 4.0 and above supports SIP trunking to Internet Telephony Service Providers. This approach allows users with non-SIP phones to make and receive SIP calls.

Data Communication Solution Features

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 and IPSec support allows secure VPN 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 Switching**
  The Avaya IP Office – Small Office Edition has a 4 port Ethernet switch (Layer 2) plus a fifth Ethernet WAN port (Layer 3). The IP406 V2 offers an 8 port Ethernet switch (Layer 2), with port 8 able to act as a second LAN interface. The IP412 and IP Office 500 offer 2 switched Ethernet ports (Layer 3).
- **LDAP client support**
  For standards based directory synchronization for Phone Manager.

Applications Platform Features

IP Office provides big business benefits and enhanced productivity for small and mid-size businesses with a full compliment of sophisticated applications. IP Office provides free-of-charge applications, including Phone Manager Lite, Voicemail Lite and CTI interfaces. These free-of-charge applications can be upgraded to provide enhanced functionality by chargeable license keys.

- **Operator SoftConsole**
  A graphical User Interface (GUI) for attendants on their PC desktop for call handling. Works with a telephone and is 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, Pro, and PC Softphone versions to allow you to control and manage phone calls from your Windows desktop.
- **Open CTI interfaces**
  IP Office has a built in TAPI server that integrates easily with popular contact management applications such as Outlook, ACT!, GoldMine and Maximizer. Sophisticated custom applications can be rapidly developed and deployed with our full software development kit.
- **Voicemail**
  Callers can always be answered with a personal voicemail greeting before a message is taken and message notification set. Messages can be shared (forwarded) with colleagues and retrieved by any phone capable of tone dialing. When used with Phone Manager Pro, the PC can be used to control message playback.
- **Integrated Messaging**
  Voice messages can be can be copied into email messages and delivered into the email system. IP Office uses SMTP or MAPI to deliver a copy of the voice message. Integrated Messaging Pro provides a higher level of integration with Microsoft Exchange Server to synchronize both voicemail and email inboxes
- **Auto-Attendant**
  Simplify service for administrators with this easy-to-use feature with the ability to construct customized automated services allowing callers to efficiently navigate the system, and reach the right person, without the assistance of an operator. Available with Voicemail Pro and with Embedded Voicemail for IP500, IP406 V2 and Small Office Edition.
- **Interactive Voice Response (IVR) and Text to Speech**
  Create automated customized systems allowing callers to interact with business information, for example, reading email, 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.
- **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.
- **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 installation, and more.
- **Compact Contact Center**
  This is the IP Office Contact Center option, with a full customer management toolset including real time agent, system, group management, standard and custom reporting. It provides real time tracking and analysis, options for agent connection, and remote agent support and wallboards for installations of up to 75 agents.

Management Tools

The IP Office solution (phone system, router/firewall/DHCP server) is easily managed through the IP Office Manager. 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 a Telephone Adaptor, Router or the optional Internal Modem Card.

The System Status application is a useful diagnostic tool that provides enhanced details about equipment and resources in the IP Office system. This includes indication of alarms and details of current calls in progress for local or remote diagnostics.

Scalable Platform

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

- **Avaya IP Office 500**
  Modular, flexible chassis which supports up to 32 extensions (up to 272 with expansion modules), with capacity for up to 16 analog trunks or 240 digital trunks (up to 192 T1 channels or 240 E1 channels) using internal daughter cards. Up to 8 Expansion Modules may be added to provide a combination of up to 272 analog, digital or IP extensions, with additional analog trunks through external Analog 16 modules. Features include 128 optional voice compression channels, 2 independently switched LAN ports and an optional Embedded Messaging card.
- **Avaya IP Office - Small Office Edition**
  The IP Office - Small Office Edition is a compact platform specifically designed to meet the needs of very small businesses and home offices. In a single unit, it can provide a PABX with Auto Attendant and Voicemail, Broadband Access, Wireless Access Point (WiFi) and VPN tunneling. Voice Compression is included as standard to support IP Extensions or provide IP Trunks back to a head office. The IP Office - Small Office Edition is available in two configurations
  - 4 Analog trunks, 4 analog extensions, 8 digital stations and 3 or 16 VoIP resources.
- **Avaya IP Office IP406**
  Supports 6 Expansion Modules providing a combination of up to 190 analog, digital or IP extensions, with capacity for 8 analog trunks or 2 digital trunks (up to 72 T1 channels or 90 E1 channels). 8 Digital Station ports (DS), 2 analog phone ports, a socket for optional embedded voicemail. Additional analog trunks can be added using IP400 Analog Trunk 16 modules. Features include up to 30 optional voice compression channels and an 8 Ethernet port switch (Layer 2). An Internal Modem Card can be added to answer up to 12 V.90 analog modem calls.
- **Avaya IP Office IP412**
  Supports 12 Expansion Modules providing a combination of up to 360 analog, digital or IP extensions, with capacity for 8 analog trunks or 4 digital trunks (up to 96 T1 channels or 120 E1 channels). Additional analog trunks can be added using IP400 Analog 16 modules. Features include 60 optional voice compression channels, 2 independently Switched LAN ports, and 108 data channels. An Internal Modem Card can be added to answer up to 12 V.90 analog modem calls.

Telephone Options

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

- **IP Telephones**
  IP Office's integral H.323 Server supports Avaya 5600 series IP telephones, selected Avaya 4600 series IP telephones, Avaya T3 series IP telephones, Avaya 3600 series Wireless VoIP telephones and Phone Manager PC Softphone.
- **Digital Telephones**
  IP Office Digital Station 16 or 30 Modules support the Avaya 5400 Series of digital phones and Avaya T3 Series telephones. The IP Office Digital Station modules also support existing selected 2400, 4400, 6400 Series phones
- **Analog Phones**
  IP Office Phone 8, 16 or 30 Modules support standard analog phones, faxes and modems, with support for calling line identification and message waiting indication (where service is provided).
- **Wireless Telephones**
  Avaya IP DECT base stations can be added to support the Avaya IP DECT 3701 and 3711 telephones. The IP Office Digital Station modules support the Avaya 3810 telephone and the Avaya 3600 series wireless VoIP telephones

Application and Feature Licensing

IP Office is an applications platform and includes a number of applications as part of the solution. These Lite versions of applications do not require any additional licensing, but upgrades to Pro versions or optional applications will need additional IP Office licenses to operate. The licensed applications require both a license key, a unique number that enables the application to run, and a feature key. The feature key is an electronic key installed on the IP Office system that determines which licensed applications can run.

Licensed applications are supplied in two forms; time limited trial licenses and full indefinite licenses. Trial licenses allow applications to run in fully functional form for 45 days (from the date of license generation), after which time they cannot be used until upgraded at cost to the full license but can be ordered at any time during the product ownership. Trial licenses are available for:

- Avaya Text To Speech (1 port).
- Centralized Voicemail with ACM.
- Compact Business Center (CBC).
- Conferencing Center.
- Integrated Messaging Pro.
- Mobile Twinning (5 users).
- Phone Manager PC Softphone (10 User).
- Phone Manager Pro (10 user).
- SoftConsole (1 user).
- Third Party Text To Speech (1 port).
- VB Scripting.
- Voicemail Pro (4 ports).
- Voicemail Pro Networked Messaging.
- VPN IPSec/L2TP.
- 3rd Party Database/IVR.
- SIP Trunking (1 trunk).
- Standard Edition upgrade to Professional Edition (for IP500 only).
- Voice Networking (4 channels, for IP500 only).
- Advanced Networking.
- VPN Phone (1 user).

ContactStore has a 45 day trial built into the software and therefore does not require a separate license key, but this 45 day trial runs from when the software is installed.

2. IP Office Platform

IP Office Overview

IP Office is a modular communications solution that scales from 2 to 360 users. It provides a hybrid PBX with both Time Division Multiplexing (TDM) and IP phone support that can be used in either mode or both concurrently. IP Office has data capabilities built in, providing IP routing, switching and Firewall protection between LAN and WAN. IP Office has an integrated software applications suite that delivers a contact center, voice and email messaging, Interactive Voice Response, conferencing and computer telephony integration.

IP Office solutions are built from hardware units and application software. Hardware provides the connectivity for voice and data circuits and processor units for the solution software. Each IP Office solution will require a system control unit (Small Office Edition, IP406 V2, IP412 and IP500), trunk connections to service provider, and expansion modules for TDM phone cabling. IP Phones connect over LAN connections to the IP Office solution.

IP Office - Small Office Edition

The IP Office - Small Office Edition is the entry level control unit of the IP Office solution and is delivered in a compact configuration that provides a mix of Analog trunks, Analog and Digital extensions and Voice over IP (VoIP) capacity. Dependant on the model chosen, up to a maximum of 28 extensions can be supported (4 Analog, 8 Digital and 16 IP).

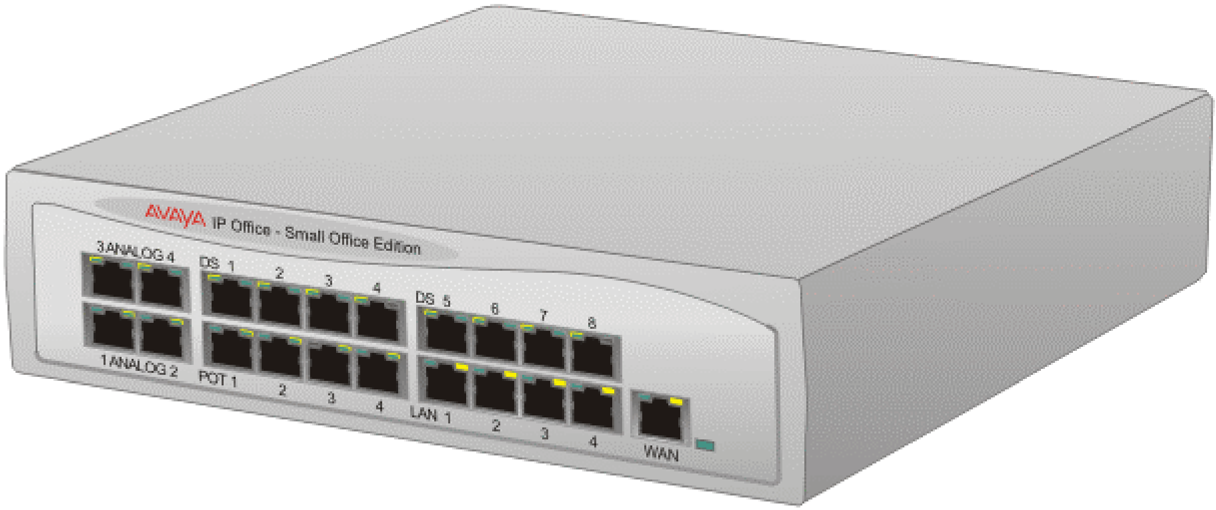
All IP Office - Small Office Edition's variants have a four-port Layer 2 Ethernet Switch and a dedicated switched Ethernet WAN port (Layer 3), making the system ideal for connection to local area networks and broadband wide area network services such as ADSL and Cable. With Voice over IP as standard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

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

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

To enable licensed IP Office applications, a serial Feature Key can be attached directly to the IP Office - Small Office Edition removing the need for an external PC for license verification.

For resilience, under power fail conditions Analog trunk port 2 is connected to POT extension port 1.



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

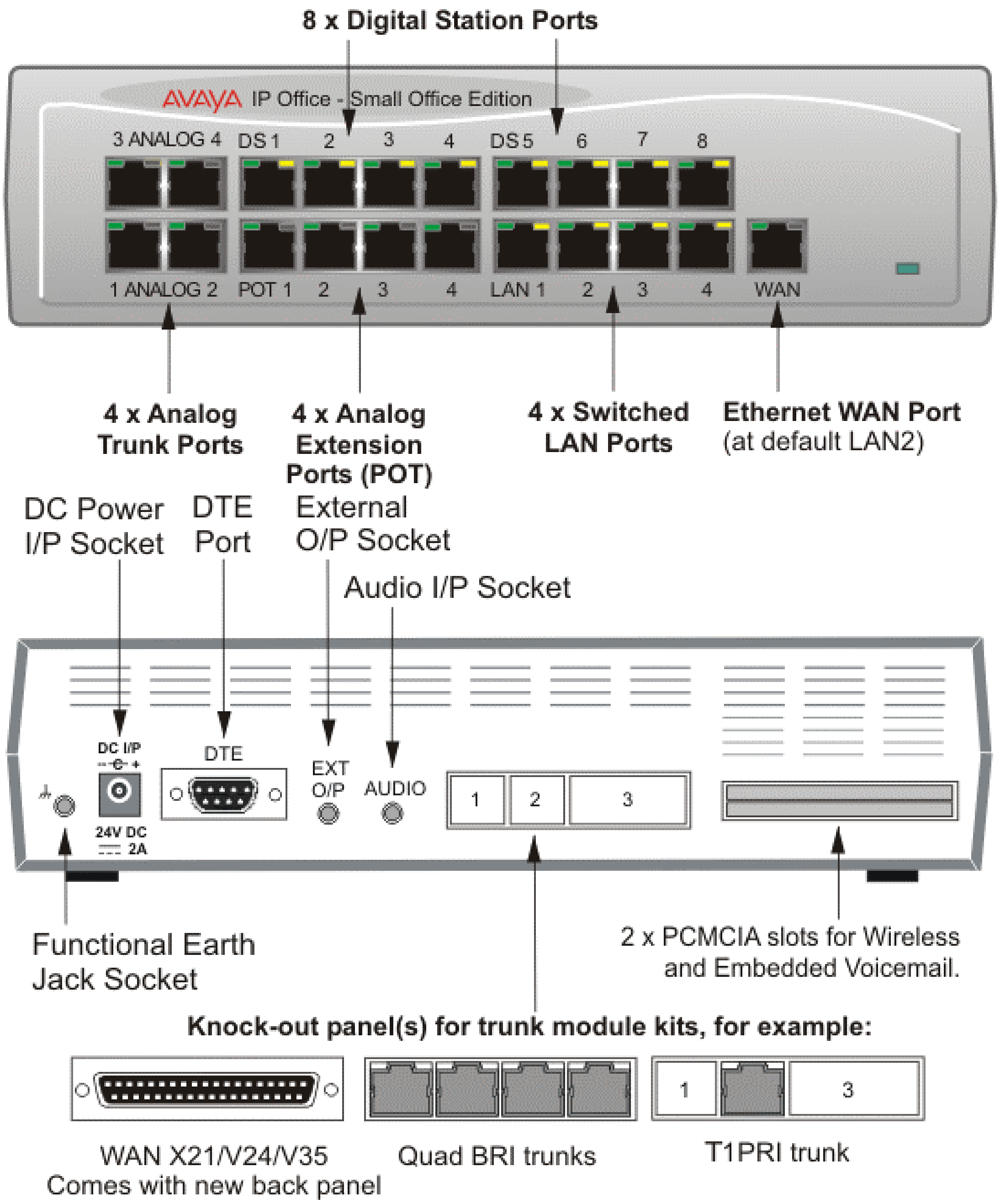
IP Office - Small Office Edition	Analog Trunks	Analog Extensions	Digital Stations	IP Extensions	VoIP Channels
4T+4A+8DS (3 VoIP)	4	4	8	16	3
4T+4A+8DS (16 VoIP)	4	4	8	16	16

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

IP Office - Small Office Edition 4T+4A+8DS (3 or 16 VoIP)

The IP Office - Small Office Edition 4T+4A+8DS provides:

- Four Analog Loop Start Trunks (Caller ID enabled).
- Four Analog extension (POT) ports with power fail switchover such that analog trunk port 2 is connected to analog extension port 1.
- Eight Digital Station (DS) ports for selected 2400, 5400 and 6400 phones plus 3810 wireless (US) phones. T3 Series phones are not supported from Release 4.0 onwards on Small Office Edition.
- 3 or 16 VoIP Codecs (G.723.1, G.711 and G.729a) and 48ms echo cancellation.
- 4 Switched Ethernet ports (Layer 2).
- Dedicated Switched Ethernet WAN port (Layer 3).
- 2 x PCMCIA Slots for optional Wireless and Embedded Voicemail card support.
- Expansion Slot for optional WAN card (V35/V24/X.21), BRI or T1 PRI.
- Serial DTE port.
- Audio input port for external music on hold source.
- External O/P socket supporting two relay on/off switch ports, e.g. for door entry systems.



IP Office - Small Office Edition WAN Expansion Interfaces

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

IP400 WAN Expansion

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

IP400 BRI Card

The BRI trunk card provides 4 European Basic Rate ISDN T 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.

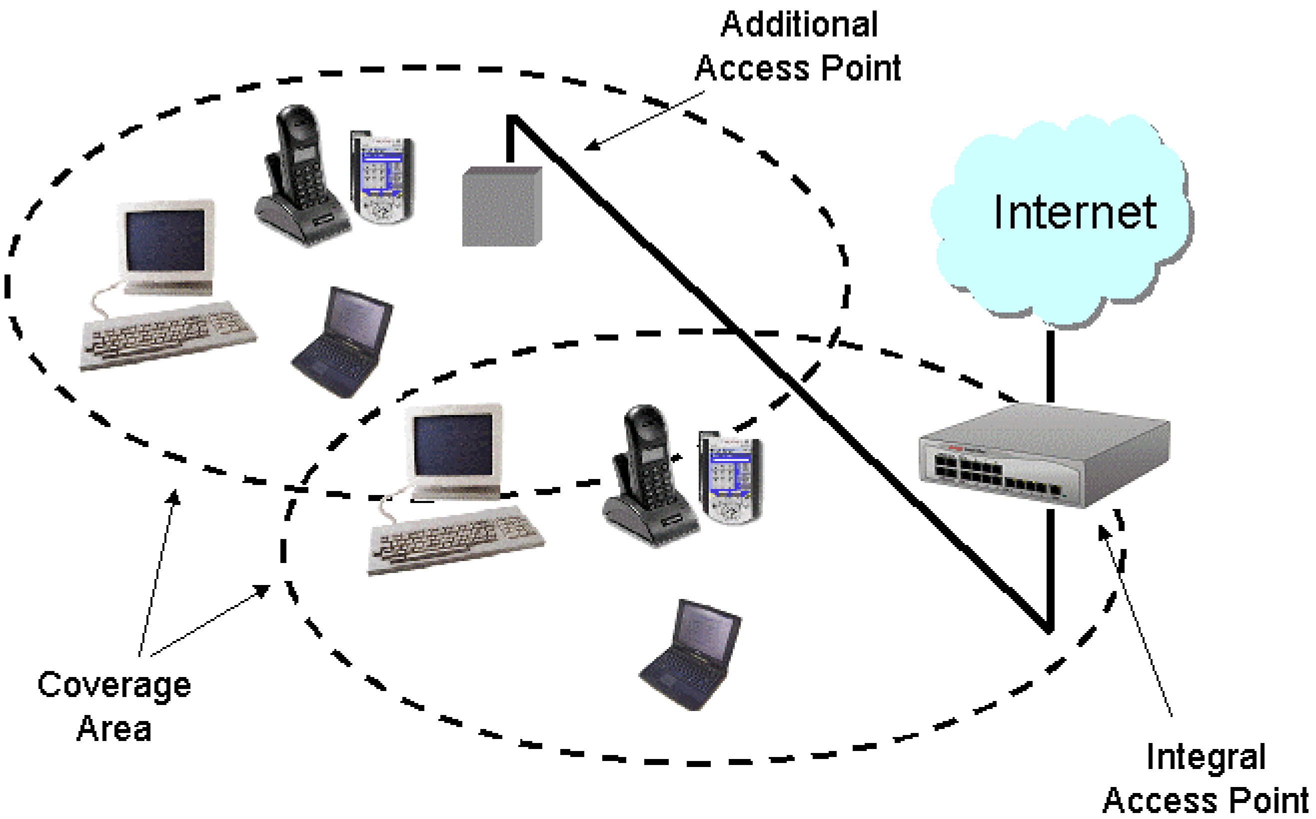
IP400 T1 PRI Card

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

- Not available in all territories, check for availability.

Optional Wireless Access Point

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



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

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

Summary

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

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

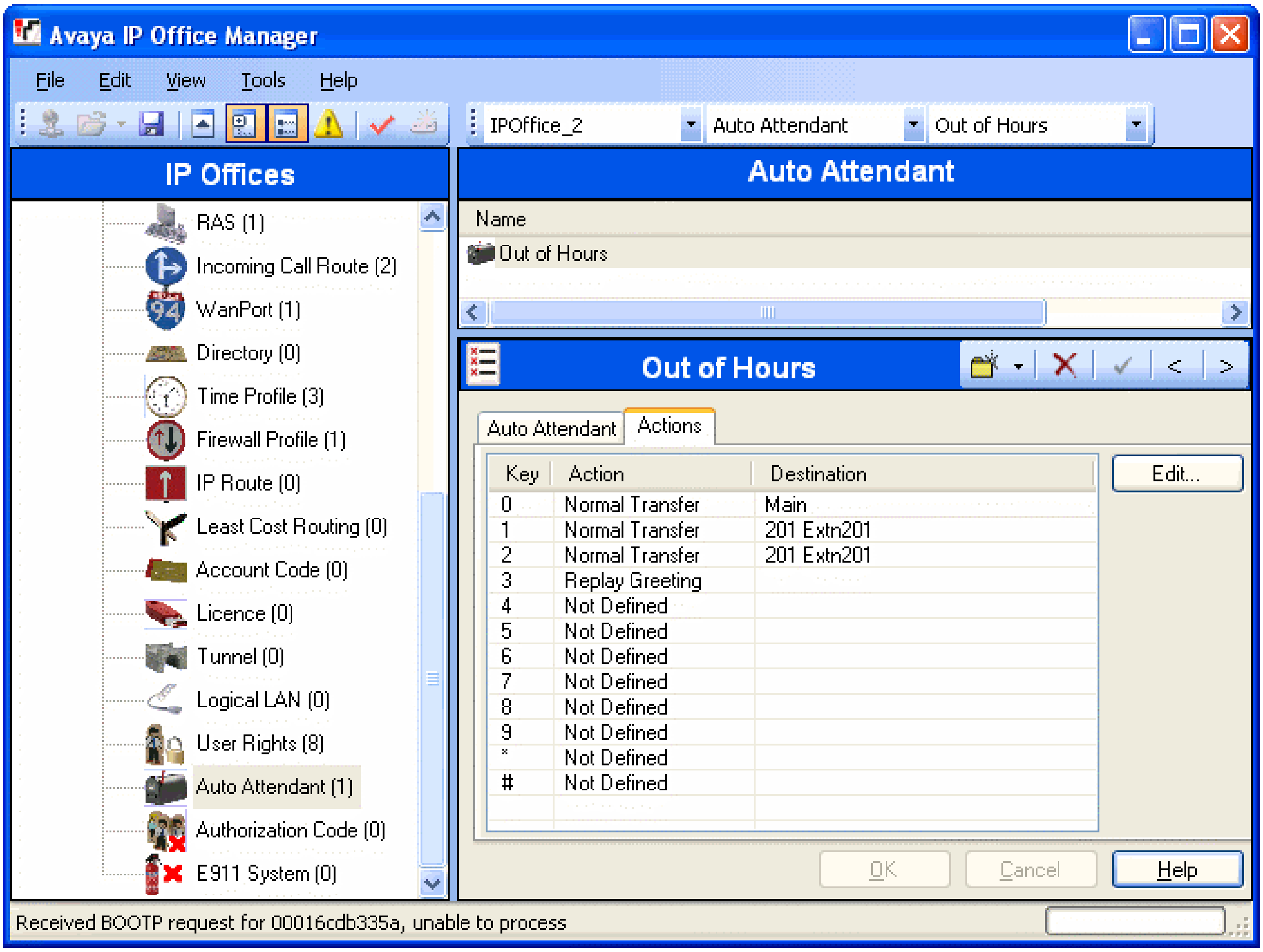
For wireless operation, IP Office - Small Office Edition must be fitted with a Wireless LAN card and the Wireless LAN Access Point license key. Alternatively, a 3rd party wireless access point can be connected directly to one of the LAN ports.

Optional Embedded Voicemail with Auto-Attendant

Entry-level voicemail and auto attendant applications are available using the Avaya memory expansion kit in one of the PCMCIA slots on the rear of the Small Office Edition. This provides small locations with an effective embedded messaging solution with auto-attendant without the additional costs of an external PC. The embedded voicemail supports up to 10 hours of message storage. The number of available voicemail ports (to support simultaneous calls to voicemail) is 3 ports on the 3 VoIP model or 10 ports on the 16 VoIP model

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

Up to 40 independent auto-attendants can be configured on the platform. These may be linked together to form multiple tiers of attendants. The choice of which auto-attendant is to answer a call can be made on any of the criteria on the Incoming Call Routing form such as called number, calling number and time of day.



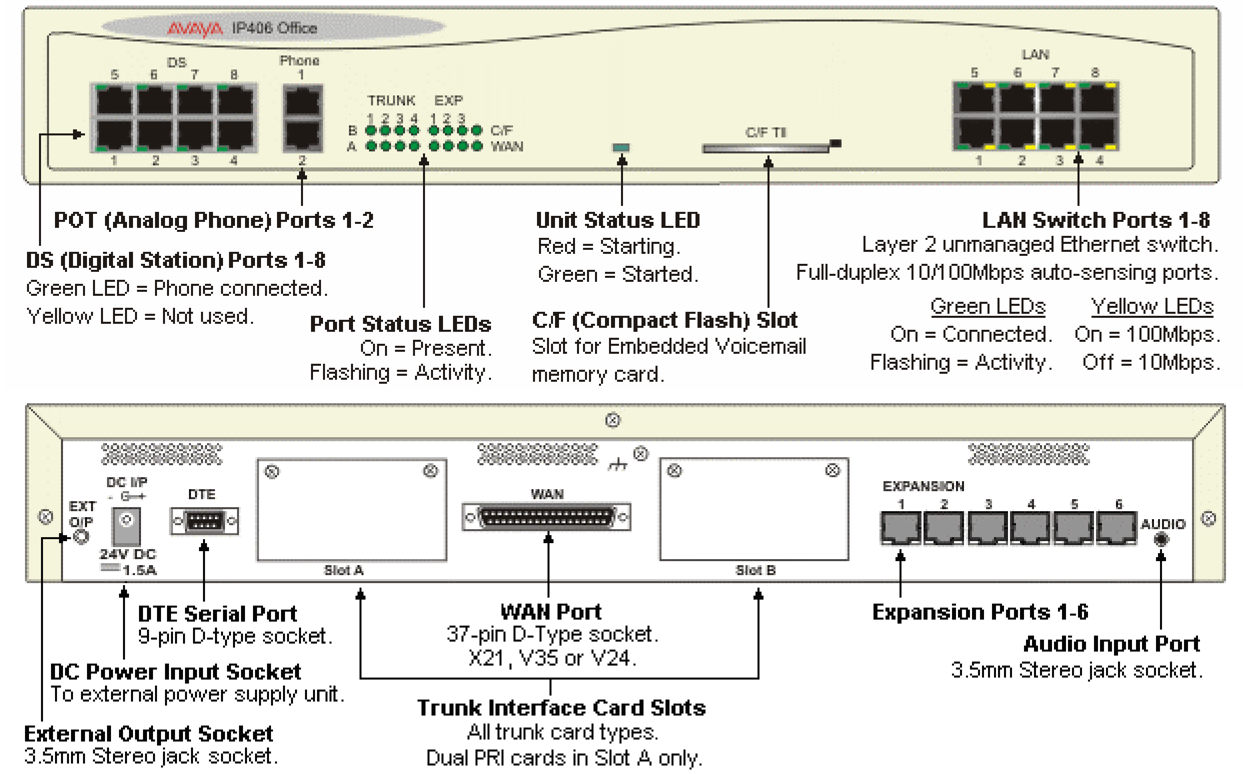
Each auto-attendant has a single menu of 12 items (0...9, *, #) that a caller can select from to either be transferred to a predefined number, to another auto attendant or to replay the greeting. The greeting for the menu is controlled by time profiles to allow three alternative messages to be played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

Please note that the IP406 V2 and IP500 embedded voicemail memory cards are identical but these are not interchangeable with the Small Office Edition card. Only Avaya supplied memory cards with the voicemail and auto attendant applications pre-installed can be used.

Avaya IP Office IP406 V2 Control Unit

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

- Eight Digital Station (DS) ports for supported 2400, 4400, 5400, 6400 and T3 Series phones plus 3810 wireless (US) phones.
- Two Analog telephone ports.
    - Two Wire
    - DTMF signaling (No rotary or Loop Disconnect)
    - Timed Break Recall (No Earth Loop Recall)
    - Caller ID capable – a variety of standards, see later
    - MWI capable – 82.5V and Line Reversal
- Eight 10/100 Mbps LAN Switched ports (Layer-2, unmanaged).
- Support for optional embedded voicemail/auto-attendant (Compact Flash card)
- 9-pin DTE Port (for maintenance or Feature Key connection for application licensing).
- X.21/V35 WAN interface.
- Support for up to 6 IP Office Expansion Modules:
    - Phone modules (8, 16, 30)
    - Digital Station modules (16, 30)
    - Analog Trunk Module 16
    - So8 module
- External O/P socket supporting two relay on/off switch ports, e.g. for door entry systems.
- Audio input port for external music on hold source.
- Two trunk interface card slots for analog, BRI, PRI (T1, E1) or CAS (E1R2)
- Internal socket for IP Telephony expansion – voice compression modules (from 4 to 30 channels)
- Internal socket for internal modem (2 or 12) for Remote Access Services
- 50 Data channels
- Up to 20 Voicemail Pro ports



Expansion Modules

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

If additional analog trunks are required, these can be aggregated in groups of 16 on each analog expansion module

Data Channels

A Data Channel is used for Remote Access (RAS), Internet Access, and Voicemail sessions. A data channel is an internal signaling resource 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. Data channels are used for voicemail connections with a maximum of 20 available for Voicemail Pro on an IP406 V2.

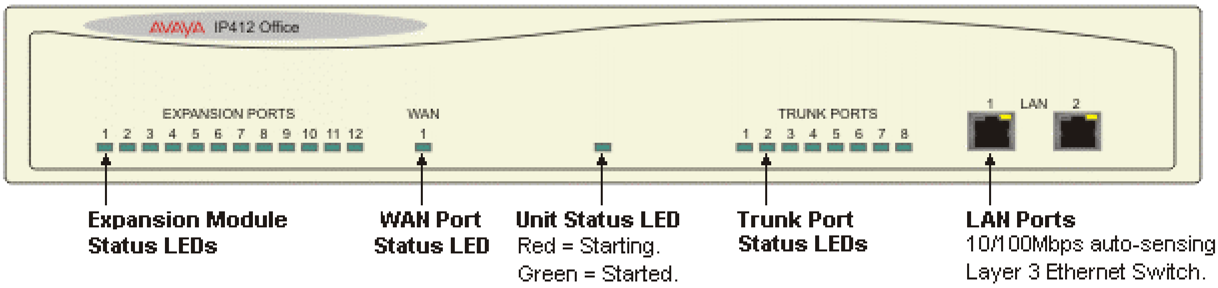
Modems and Voice Compression modules

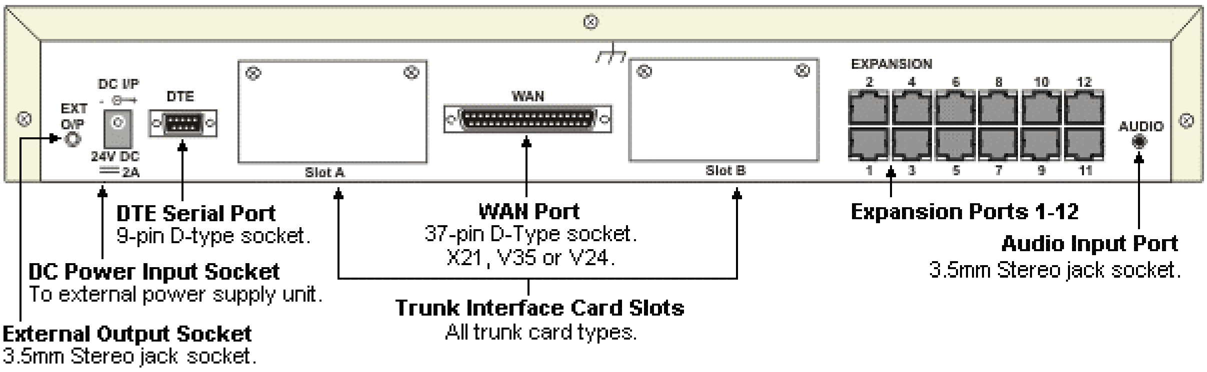
You can add additional hardware to the IP406 system to add one modem card (2 or 12 V.90 modems) and 1 Voice Compression Module (VCM). The VCM supports from 4 to 30 simultaneous Voice over IP sessions and is used for either providing networking between sites over a Wide Area Network or supporting IP Telephones and Soft phones.

Avaya IP Office IP412 Control Unit

With a greater internal data transfer capability than the IP406 V2, the IP412 is more suitable for meeting the needs of the small contact center or businesses with a CRM focus. The IP412 differs from the IP4060 V2 by providing a greater trunk expansion capability of up to four PRI trunks. The IP412 is a stackable unit with an optional 19" rack mounting kit. The IP412 includes:

- 9-pin DTE Port (for maintenance or Feature Key connection for application licensing).
- X.21/V35 WAN interface.
- Support for up to 12 IP Office Expansion Modules:
    - Phone modules (8, 16, 30)
    - Digital Station modules (16, 30)
    - Analog Trunk Module 16
    - So8 module
- External O/P socket supporting two relay on/off switch ports, e.g. for door entry systems.
- Audio input port for external music on hold source.
- Two trunk interface card slots for analog, BRI, PRI (T1, E1) or CAS (E1R2)
- 2 internal sockets for IP Telephony expansion – voice compression modules (from 4 to 30 channels)
- Internal socket for internal modem (2 or 12) for Remote Access Services
- 108 Data channels
- Up to 30 Voicemail Pro ports
- Two 10/100 switched Ethernet ports (Layer 3).





Expansion Modules

Through support of up to twelve external Expansion Modules, IP412 can be enhanced to support a mixture of analog, digital or IP phones, to maximum of 360 phones in any combination.

If additional analog trunks are required, these can be aggregated in groups of 16 on each analog expansion module.

Data Channels

A Data Channel is used for Remote Access (RAS), Internet Access, and Voicemail sessions. A data channel is an internal signaling resource 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. Data channels are used for voicemail connections with a maximum of 30 available for Voicemail Pro on a IP412.

Modems and Voice Compression modules

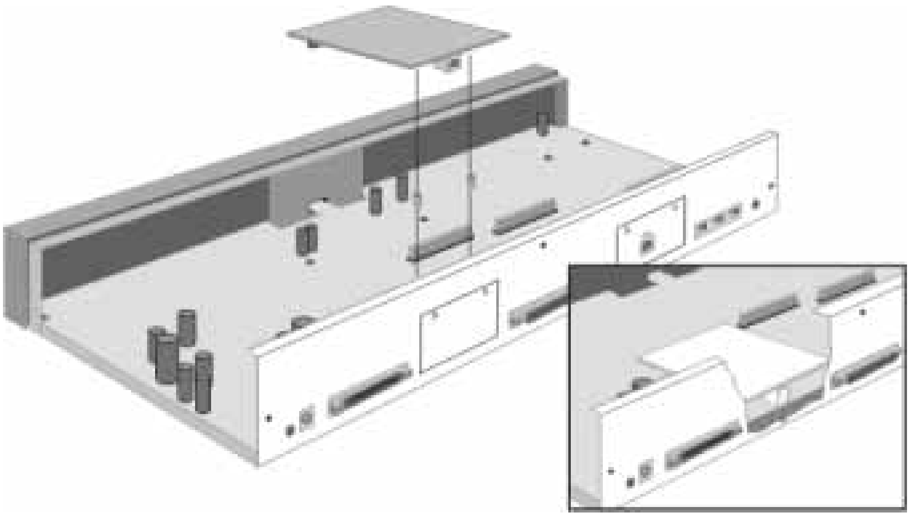
You can add additional hardware to the IP412 system to add one modem card (2 or 12 V.90 modems) and 2 Voice Compression Modules (VCM). Each VCM supports from 4 to 30 simultaneous Voice over IP sessions and is used for either providing networking between sites over a Wide Area Network or supporting IP Telephones and Soft phones.

IP400 Trunk Interface Cards

IP400 trunk interface cards fit into the card slots on the Small Office Edition, IP406 V2 and IP412 control units and in any slot of the IP500 when combined with the IP500 Legacy Card Carrier. They provide analog, ISDN or CAS trunk connectivity. Not all interfaces are available in all territories. The following table shows how many of each card type are supported by each of control unit.

	Small Office Edition	IP406 V2	IP412	IP500*
IP400 Universal Analog Trunk 4	✗	2	2	2
IP400 Quad BRI	1	2	2	2
IP400 PRI E1	✗	2	2	2
IP400 Dual PRI E1	✗	1 (Slot A)	2	2
IP400 E1R2	✗	2	2	2
IP400 Dual E1R2	✗	1 (Slot A)	2	2
IP400 PRI T1	1	2	2	2
IP400 Dual PRI T1	✗	1 (Slot A)	2	2

* Each card requires the use of an IP500 Legacy Card Carrier.



IP400 BRI Card

The BRI trunk card provides 4 Basic Rate ISDN T interfaces (8 trunks). Details of the supported ISDN supplementary services on BRI interfaces are given in the 'Public and Private Voice Networks' section.

IP400 PRI Cards (T1/E1/E1R2)

Available in single and dual versions the IP400 PRI card provides single and dual primary rate trunk interfaces respectively. The PRI is available in T1, E1 and E1R2 MFC variants depending on the territory. The dual version is only supported on the IP412, in slot A of the IP406 V2 and in the IP500 using an IP500 Legacy Card Carrier. The Small Office Edition only supports the single T1 PRI card (not E1 or E1R2).

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

T1 trunk cards incorporate an integrated CSU/DSU. 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.

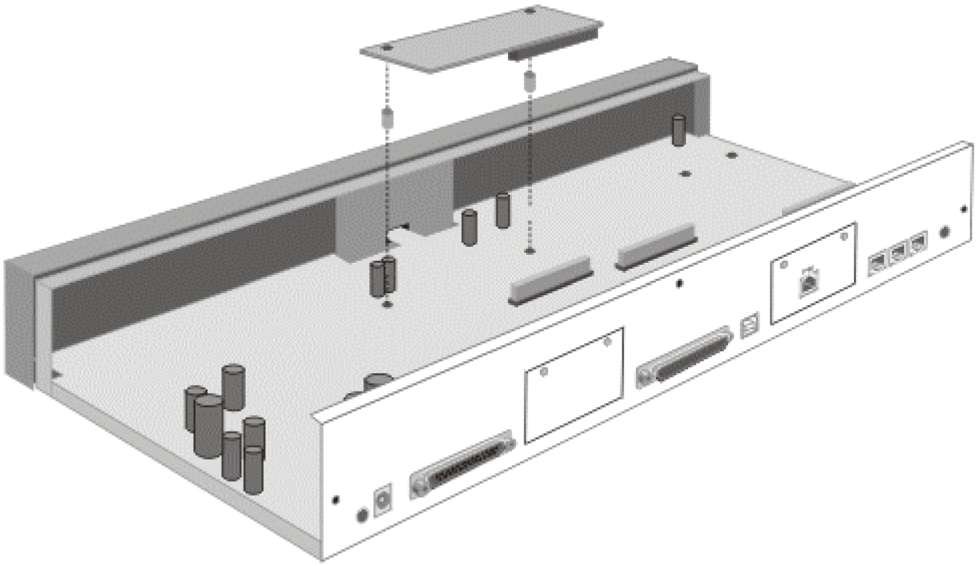
IP400 Universal Quad Analog Trunk (LS) Card

This card provides four analog trunk ports. These are 2-wire loop start interfaces and are available in all territories. This card supports Caller ID where provided. With IP Office R3.1 and later, this module supports optional 16ms echo cancellation.

Please note that ground start analog trunks are supported via the IP Office Analog Trunk 16 Expansion Module.

Internal Daughter Cards

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



IP400 Voice Compression Module – 4/8/16/24/30 ports

The Voice Compression Module (VCM) is used for Voice over IP (VoIP) applications in the IP406 and IP412 control units. Five VCM variants are available supporting 4, 8, 16, 24 and 30 channels of compression. The echo cancellation capabilities of the VCM cards vary. The VCM 4, 8, 16 and 24 cards offer 64ms of echo cancellation while the VCM 30 card offers 25ms.

On IP Office - Small Office Edition systems, either 3 or 16 VCM channels are built-in and cannot be changed. The IP406 V2 supports a single VCM while the IP412 can have any two VCMs installed.

The following table shows how many of each card are supported by each platform.

	IP406 V2	IP412	IP500
IP400 VCM 4	1	2	2*
IP400 VCM 8	1	2	2*
IP400 VCM 16	1	2	2*
IP400 VCM 24	1	2	2*
IP400 VCM 30	1	2	2*
IP400 VCM 5/10/20 (no longer sold)	1	2	✕
IP500 VCM 32	✕	✕	2
IP500 VCM 64	✕	✕	2

*Each card requires the use of an IP500 Legacy Card Carrier.

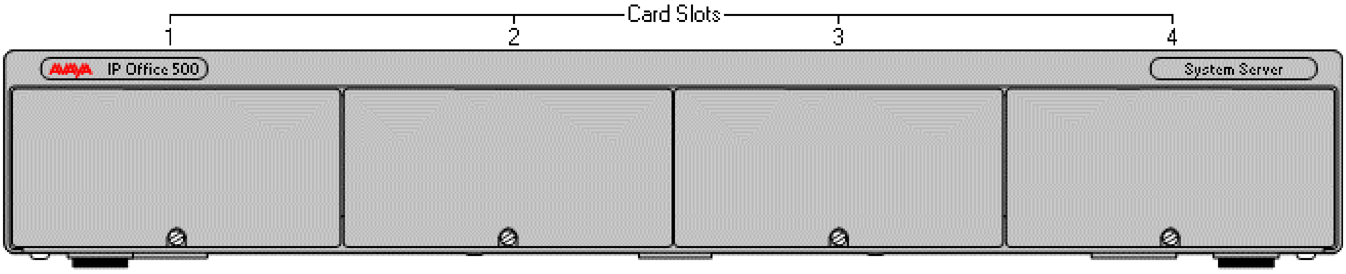
IP400 Internal Modem Card

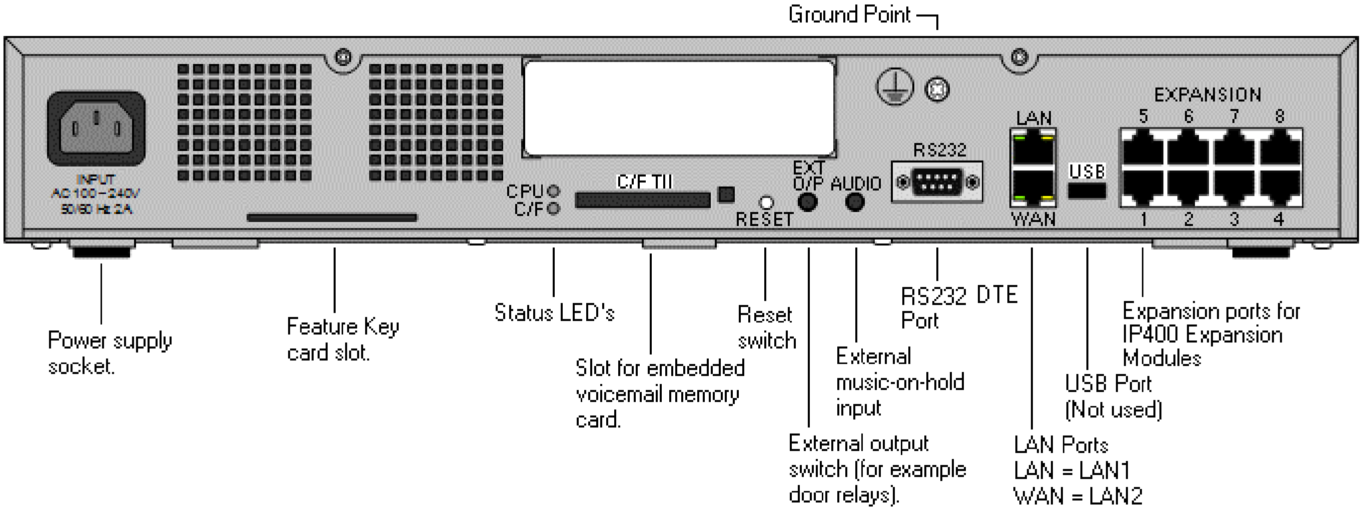
An internal modem card with 12 modems can be installed in both the IP406 V2 and IP412 to provide dial-up capacity that is better matched to remote access requirements of customers. The Internal Modem card allows up to 12 simultaneous V.90 (56kbps) analog modem calls into the IP Office.

IP Office 500 Control Unit

With a greater VCM channel capacity and performance, the IP Office 500 (IP500) is the most suitable of the IP Office range for IP Telephony applications. It also provides an entry level offer into the IP Office family through IP Office Standard Edition software. The IP500 also differs from the IP406 V2 and IP412 by providing a greater trunk expansion capability of up to eight four PRI interfaces (maximum 192/240 trunks). The IP500 is a stackable unit with an optional 19" rack mounting kit and an optional wall mounting kit for smaller configurations. The IP500 includes:

- 4 slots to house a mixture of extension cards and VCM cards
    - Digital Station 8 card
    - Phone 2 and Phone 8 cards
    - VCM-32 and VCM-64 cards
- Optional trunk daughter card support:
    - Analog Trunk Module 4 card
    - BRI-4 and BRI-8 cards (2 x 2B+D and 4 x 2B+D channels respectively)
    - Single and Dual Universal PRI cards
- Support for IP400 trunk and VCM cards using a Legacy Card Carrier
- Slot for smart card Feature Key – required for system operation as well as licensing of optional features.
- 9-pin DTE Port for maintenance.
- Support for up to 8 IP500 Expansion Modules (requires upgrade to Professional Edition):
    - Phone modules (8, 16, 30)
    - Digital Station modules (16, 30)
    - Analog Trunk Module 16
    - BRI So8 module
    - IP400 expansion modules (not WAN3 10/100 or Network Alchemy modules)
- External O/P socket supporting two relay on/off switch ports, e.g. for door entry systems.
- Audio input port for external music on hold source.
- 48 Data channels
- Up to 30 Voicemail Pro ports
- Two 10/100 switched Ethernet ports (Layer 3).





IP500 Voice Networking License

QSIG, H.323 and SCN capabilities are not enabled by default in the IP500. An additional license is required to enable this functionality with 4 simultaneous networking channels (no channel limit for QSIG). Additional channels can then be licensed in increments of 4.

IP Office Standard Edition

By default the IP500 control unit runs a subset of full IP Office functionality called IP Office Standard Edition. In this mode the IP500 is restricted to a maximum of 32 users in the base control unit with no expansion. Supported options include Embedded Voicemail, Phone Manager Lite/Pro/PC Softphone, SoftConsole, TAPI, SMDR, CBC, SIP trunking, mobile twinning, VPN Phone and IP DECT, as well as licenses for voice networking (H.323 or SCN). IP Office Standard Edition does not support advanced applications (Voicemail Pro, CCC, Conference Center, etc). This restriction can be removed by adding an IP Office Professional Edition Upgrade license to the configuration.

IP Office Professional Edition

By purchasing the upgrade license from Standard Edition to Professional Edition, additional functionality is enabled. This includes the ability to expand the system using up to eight external Expansion Modules, allowing the IP500 to support a maximum of 272 phones through a mixture of analog, digital or IP handsets. If additional analog trunks are required, these can be aggregated in groups of 16 on each analog expansion module. Note that the Professional Edition also enables the licensing of advanced applications such as Voicemail Pro.

The following table shows which features are supported by Standard Edition and which require the upgrade to Professional Edition.

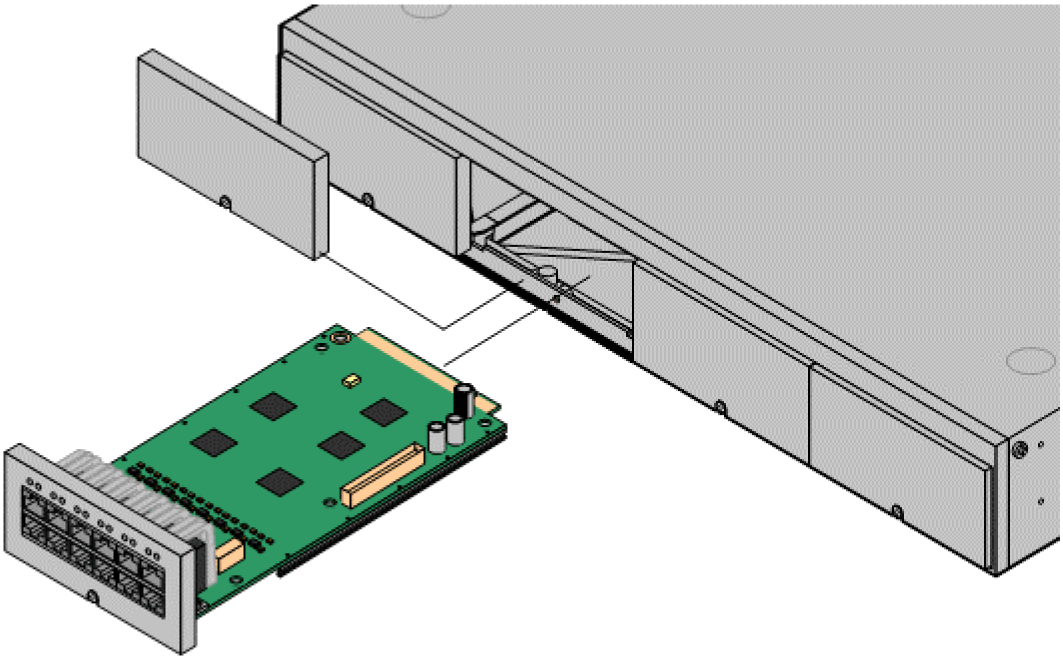
	Standard Edition		Professional Edition	
	Included	Optional	Included	Optional
Up to 32 users.	✓		✓	
IP phone support	✓		✓	
Phone Manager Lite	✓		✓	
TAPI	✓		✓	
SMDR	✓		✓	
64-way basic conferencing	✓		✓	
IP Office Manager	✓		✓	
System Status Application	✓		✓	
TAPI	✓		✓	
SMDR	✓		✓	
Voice Networking		✓		✓
Advanced Networking		✓		✓
Embedded Voicemail		✓		✓
Phone Manager Pro / PC Softphone		✓		✓
SoftConsole		✓		✓
CBC		✓		✓
SIP Trunking		✓		✓
IP DECT		✓		✓
Mobile Twinning		✓		✓
VPN phone support		✓		✓
CTI Link Pro		✓		✓
32 - 272 Users			✓	
Expansion Module support.			✓	
64-way Meet-Me conferencing			✓	
Voicemail Lite			✓	
Voicemail Pro				✓
ContactStore				✓
Conferencing Center				✓
CCC				✓
TAPI WAV				✓

IP500 Cards

The IP500 control unit has 4 slots for the insertion of cards. These cards can be divided into two types; base cards and daughter cards. Base cards include a front panel and ports for cable connections. Daughter cards can be added to a base card in order to provide additional facilities (typically trunk connections).

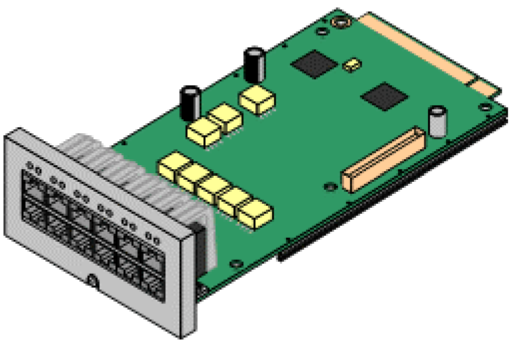
The following base cards are available:

- **IP500 Digital Station 8 Card *(Maximum 3)*.**
- **IP500 Analog Phone 2 Card and Phone 8 Card *(Maximum 4)*.**
- **IP500 VCM Card *(Maximum 2)*.**
- **IP500 Legacy Card Carrier *(Maximum 2)*.**



IP500 Digital Station 8 Card

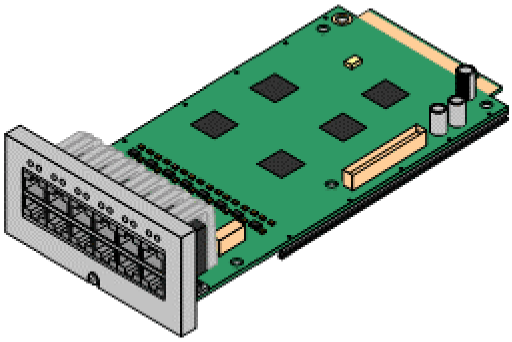
This card provides 12 RJ45 ports. The first 8 ports are DS ports for the connection of Avaya digital phones other than IP phones. The card can be fitted with an IP500 daughter trunk card, which then uses the additional 4 RJ45 ports for trunk connections.



- This card accepts one IP500 trunk daughter card of any type.
- 4400 Series phones (4406D, 4412D and 4424D) are not supported on this card, only on Digital Station expansion modules. Therefore a maximum of 240 4400 Series phones are supported in the system.

IP500 Analog Phone 2 Card

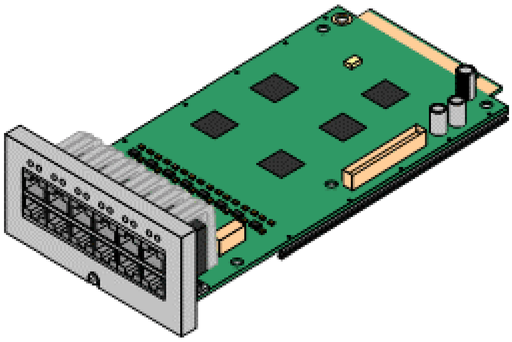
This card provides 2 analog extension ports (1-2) for the connection of analog phones. The card can be fitted with an IP500 daughter trunk card, which then uses the last 4 RJ45 ports (9-12) for trunk connections.



- This card accepts one IP500 trunk daughter card of any type.

IP500 Analog Phone 8 Card

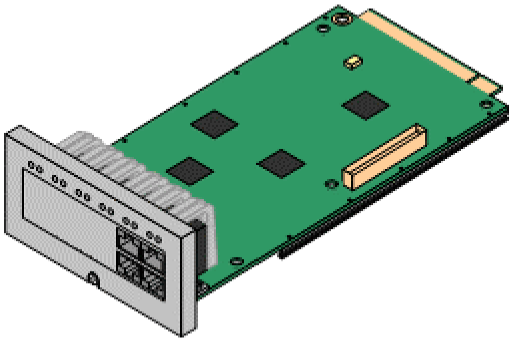
This card provides 8 analog extension ports for the connection of analog phones. The card can be fitted with an IP500 daughter trunk card, which then uses the additional 4 RJ45 ports for trunk connections.



- This card accepts one IP500 trunk daughter card of any type.

IP500 VCM Card

This card provides voice compression channels for use with VoIP calls, SIP trunks and IP-based voice networking. The module is available in variants supporting up to 32 or 64 channels. The actual number of channels provided is controlled by the VCM Channels licenses entered into the IP500 system configuration.



Each VCM card has 4 VCM channels enabled by default. Further channels can be enabled up to the maximum (32 or 64) through adding one or more licenses (available in 4, 8, 16, 28 and 60 channel increments).

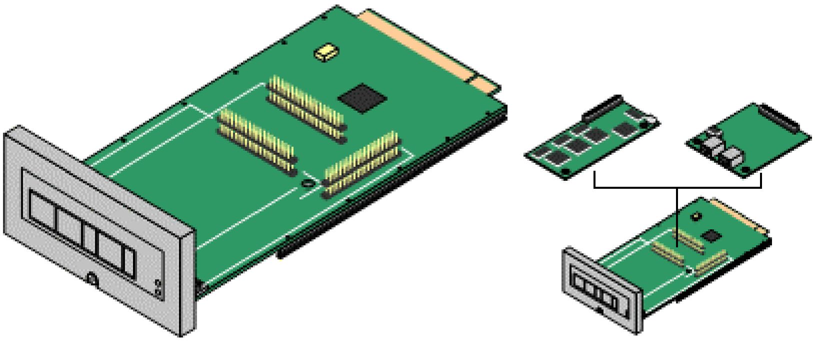
The maximum number of voice compression channels supported, using IP500 VCM base cards and / or IP400 VCM cards on IP500 Legacy Card Carriers, is 128.

The card can be fitted with an IP500 daughter trunk card, which uses the 4 RJ45 ports for trunk connections.

- This card accepts one IP500 trunk daughter card of any type.

IP500 Legacy Card Carrier

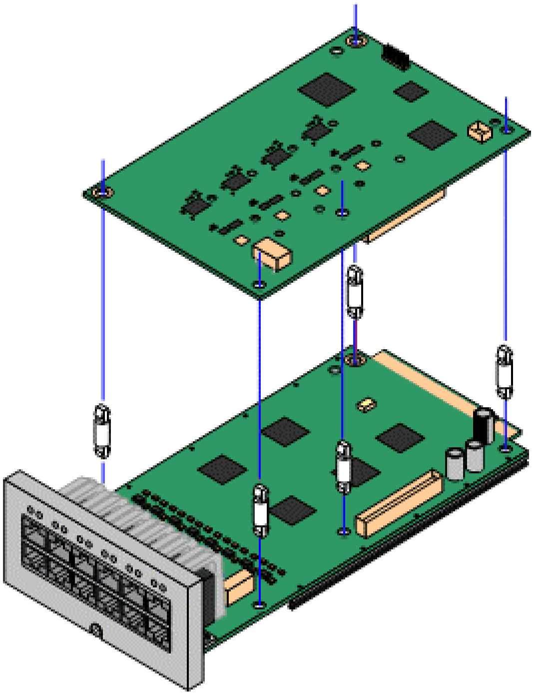
This card allows a variety of IP400 trunk and VCM cards to be used with the IP500 control unit. The front of the card includes a number of panels that can be snapped off to match the ports available when an IP400 trunk card is fitted.



- This card does not accept any IP500 daughter trunk cards.
- The IP500 control unit can accept up to 2 IP400 trunk or VCM cards by mounting each card on an IP500 Legacy Card Carrier
- This card supports the following IP400 cards:
  - ✓ PRI T1
  - ✓ Dual PRI T1
  - ✓ PRI 30 E1 (1.4)
  - ✓ Dual PRI E1
  - ✓ PRI 30 E1R2 RJ45
  - ✓ Dual PRI E1R2 RJ45
  - ✓ ANLG 4 Uni
  - ✓ BRI-8 (UNI)
  - ✓ VCM 4
  - ✓ VCM 8
  - ✓ VCM 16
  - ✓ VCM 24
  - ✓ VCM 30

IP500 Trunk Cards

IP500 daughter trunk cards can be fitted to existing IP500 base cards to provide support for trunk ports. The daughter card uses the ports provided on the base card for cable connection. The addition of an IP500 daughter trunk card is supported on IP500 Digital Station, IP500 Analog Phone and IP500 VCM base cards. They are not supported on the IP500 Legacy Card Carrier base card.



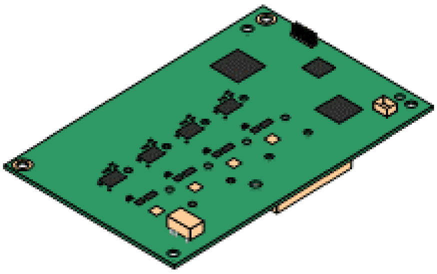
For those base cards that support daughter cards, there are no restrictions on the combination of card types. However in systems with both Analog Phone 8 base cards and analog trunk daughter cards, combining the two types are recommended as it then provides analog power failure support for one trunk/extension (Not applicable to the Analog Phone 2 base card).

Each daughter card is supplied with the spacer pegs required for installation and a label to identify the cards presence on the physical unit once installed.

- **IP500 Analog Trunk Card (Maximum 4).**
- **IP500 BRI Trunk Card (Maximum 4).**
- **IP500 Universal PRI Trunk Card (Maximum 4).**

IP500 Analog Trunk Card

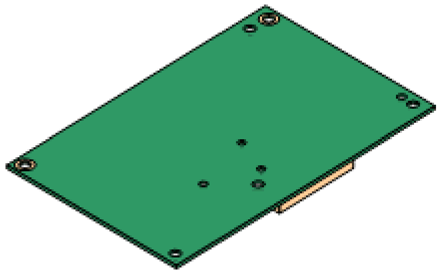
This card can be added to an IP500 Digital Station card, IP500 Analog Phone base card, or IP500 VCM card. It allows that card to then also support 4 analog loop-start trunks. It also provides one analog V.32 modem.



- When fitted to an IP500 Analog Phone 8 base card, the combination supports 1 power failure extension to trunk connection.

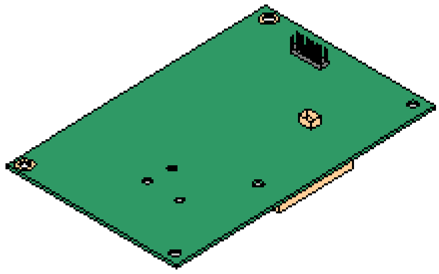
IP500 BRI Trunk Card (Euro ISDN)

This type of card can be added to an IP500 Digital Station card, IP500 Analog Phone card, or IP500 VCM card. It allows that card to then also support up to 4 BRI trunk connections, each trunk providing 2B+D digital channels. The card is available in 2 port (4 channels) and 4 port (8 channels) variants.



IP500 Universal PRI Trunk Card

This type of card can be added to an IP500 Digital Station card, IP500 Analog Phone card, or IP500 VCM card. It allows that card to then also support primary rate digital trunk connections. Available in single and dual versions the IP400 PRI card provides single and dual primary rate trunk interfaces respectively. The PRI is configurable for T1, E1 or E1R2 MFC use depending on the territory.



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

The IP500 Universal PRI trunk cards incorporate an integrated CSU/DSU. 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.

Here is a summary of the capabilities of the card:

- Each card is configurable to connect to T1, E1 or E1R2 lines.
- The card is available in either a single or dual PRI variant. The single variant can support up to 24 T1 channels or up to 30 E1 channels. The dual variant can support up to 48 T1 channels or 60 E1 channels.
- On each card, 8 channels are enabled by default. Further channels may be enabled by the purchase of additional licenses in 2-channel or 8-channel increments.
- The IP500 PRI daughter card works on any IP500 VCM or extension base card (not the Legacy Card Carrier).
- Up to four Universal PRI cards can be installed in any combination in the IP500 chassis.
- Diagnostics capabilities:
    - Visual indicators to show service state
    - Physical test points to monitor traffic

External Expansion Modules

External Expansion Modules

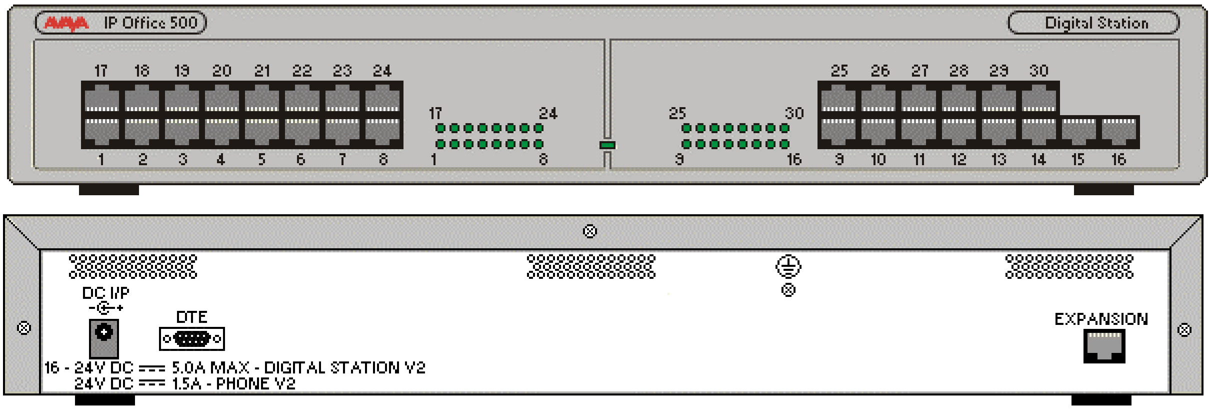
Unless otherwise stated, each of these modules may be used with the IP500, IP406 V2 and IP412.

- **IP500 Phone Expansion Module**
  Available in two variants for 16 or 30 analog extensions with calling line presentation.
- **IP500 Digital Station Expansion Module**
  Available in two variants for 16 or 30 digital extensions for Avaya series digital telephones.
- **IP500 BRI So8 Expansion Module**
  Available regionally, offering 8 BRI S-interfaces for ISDN connection.
- **IP500 Analog Trunk 16 Expansion Module** ***(US version only)***
  Provides 16 analog loop start or ground start trunks, with power failover of two trunks.
- **IP400 Phone Expansion Module**
  Available in three variants for 8, 16 or 30 analog extensions with calling line presentation.
- **IP400 Digital Station Expansion Module**
  Available in two variants for 16 or 30 digital extensions for Avaya series digital telephones.
- **IP400 So8 Expansion Module**
  Available regionally, offering 8 BRI S-interfaces for ISDN connection.
- **IP400 Analog Trunk 16 Expansion Module**
  Available in one variant for 16 analog loop start or ground start trunks, with power failover of two trunks.
- **IP400 WAN 3 Expansion Module**
  Provides 3 wide area interfaces and connects to IP Office via Ethernet. A maximum of 2 WAN3 10/100 modules are supported on the IP406 V2 and IP412. It is not supported on the IP500 or Small Office Edition.

IP500 Digital Station Module

This expansion module provides additional Digital Station (DS) ports for selected Avaya 2400, 4400, 5400, 6400, T3 (EMEA only) series phones and 3810 wireless phones (North America only). The IP500 Digital Station module is available in 2 variants; 16 or 30 extensions.

For installations in a rack, this module requires the IP500 Rack Mounting Kit. The IP500 Digital Station Module is functionally identical to the IP400 Digital Station V2 Module.



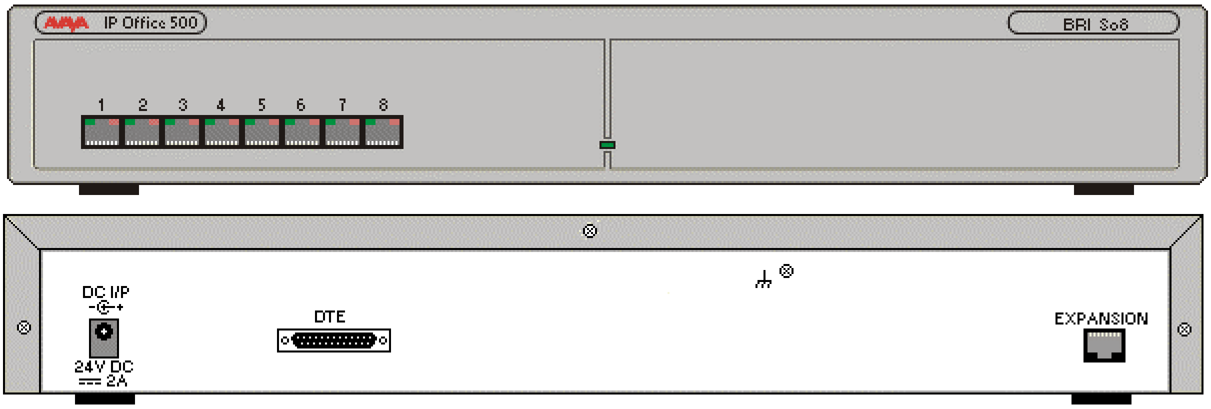
- Telephones can be located up to 1km from the control unit. For extensions located "out-of-the-building" additional line protection will be needed. For more information on cabling and out of building guidelines, see the IP Office Installation Manual.
- For systems where Direct Station Select (DSS) Units are being used, IP Office supports a maximum of:
  - Eight EU24 and or EU24BL per system.
  - Two XM24 units on each Digital Station expansion module, including the IP406 V2 control unit, to a maximum of 10 XM24 units per system.
  - Two 4450 units on each Digital Station expansion module, including the IP406 V2 control unit, to a maximum of 10 4450 units per system.
  - T3 DSS units.

See the Telephones Section for specific limits on the number of each type of telephone supported on DS modules.

IP500 BRI So8 Module

The IP500 BRI So8 module provides 8 S-Bus interfaces for Basic Rate ISDN devices, such as video conferencing, fax servers or ISDN telephones.

For installations in a rack, this module requires the IP500 Rack Mounting Kit. The IP500 BRI So8 Module is functionally identical to the IP400 So8 Module.

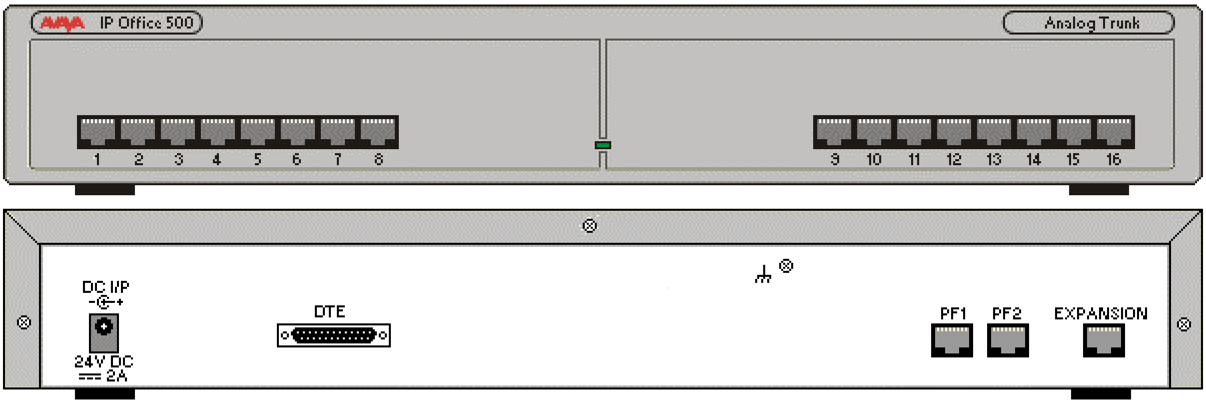


The IP500 BRI So8 expansion module supports both point-to-point and point-to-multipoint connections. A maximum of 10 terminal endpoints identifiers (TEIs) are supported on each bus.

IP500 Analog Trunk 16 Module

This expansion module provides an additional sixteen Loop Start or Ground Start two-wire analog trunks. (Ground start trunks are not available in all territories) The first two trunks on the module which are automatically switched to power fail sockets on the rear of the unit in the event of power being interrupted must be loop start for correct power fail operation.

For installations in a rack, this module requires the IP500 Rack Mounting Kit. The IP500 Analog Trunk 16 Module is functionally identical to the IP400 Analog Trunk 16 Module.

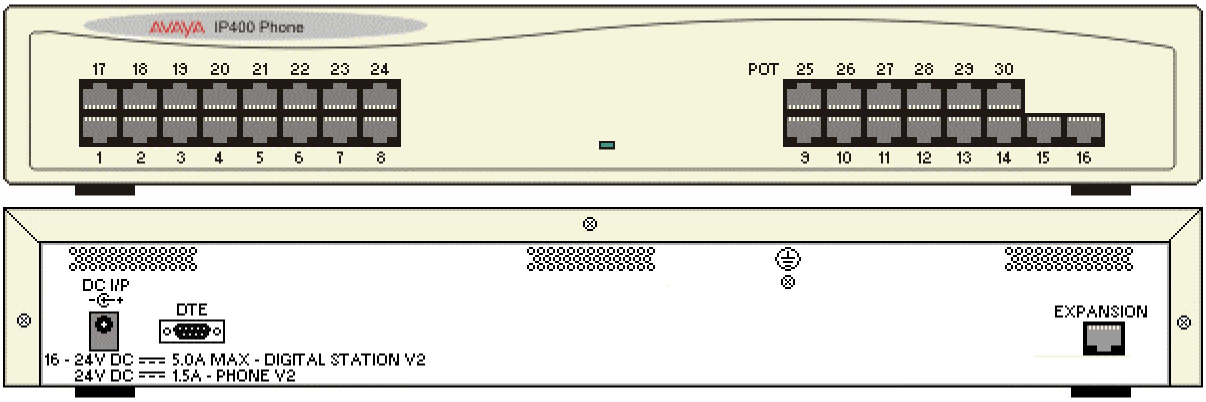


IP400 Phone Module

This module provides additional analog telephone interfaces:

- Two Wire
- DTMF signaling (No rotary or Loop Disconnect)
- Time Break Recall (No Earth Loop Recall)
- Caller ID capable
- Message Waiting Indication (MWI) capable - High Voltage, Pulsed High Voltage, Line Reversal

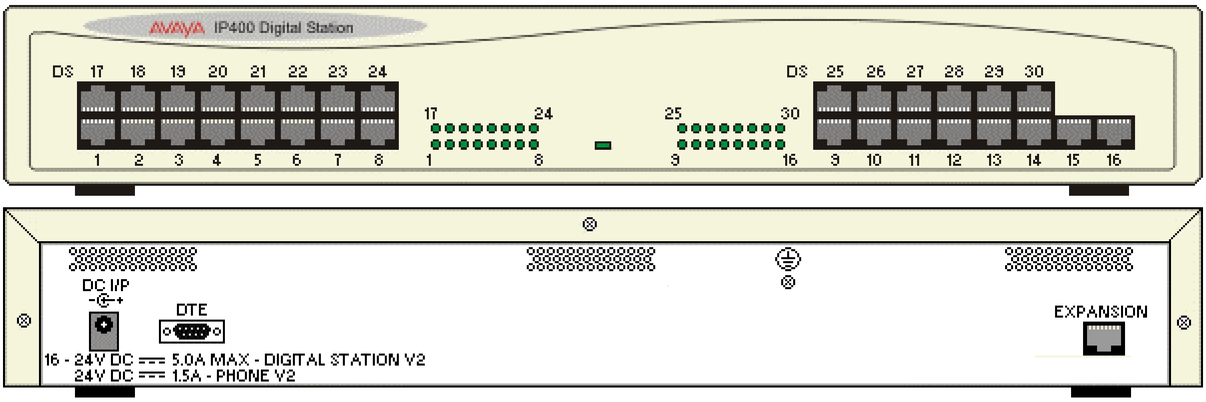
The IP400 Phone module is available in 3 versions, giving 8, 16 or 30 extensions. Telephones can be located up to 1km from the control unit. For extensions located "out-of-the-building" additional line protection will be needed. For more information on cabling and out of building guidelines, see the IP Office Installation Manual.



- IP Office Phone Modules provide support for a variety of analog MWI methods. These methods are 51V Stepped, 81V, 101V and Line Reversal. The 101V method is only supported when using a Phone V2 expansion module.
- Each analog port can support a device of maximum 1 REN.
- On analog ports, call information is sent while the phone is ringing, and cannot be updated during a call or set on an outbound call (the phone may do a local match but this is not controlled by the IP Office). The primary purpose of displays is to give information about incoming calls. Where the Caller Display standard chosen supports the delivery of text (extension name) as well as the number, both are delivered.
- An analog extension port can be set for external Paging operation. It does not operate like a normal extension and is connected to external equipment through an isolation device. The Port will always be busy so it cannot be called directly and can only be accessed by using a shortcode. When not receiving a Page the port will remain silent, when being paged the page tone is sent before the speech path is opened.

IP400 Digital Station V2 Module

This expansion module provides additional Digital Station (DS) ports for selected Avaya 2400, 4400, 5400, 6400, T3 (EMEA only) series phones and 3810 wireless phones (NA only). The IP400 Digital Station module is available in 2 variants; 16 or 30 extensions.

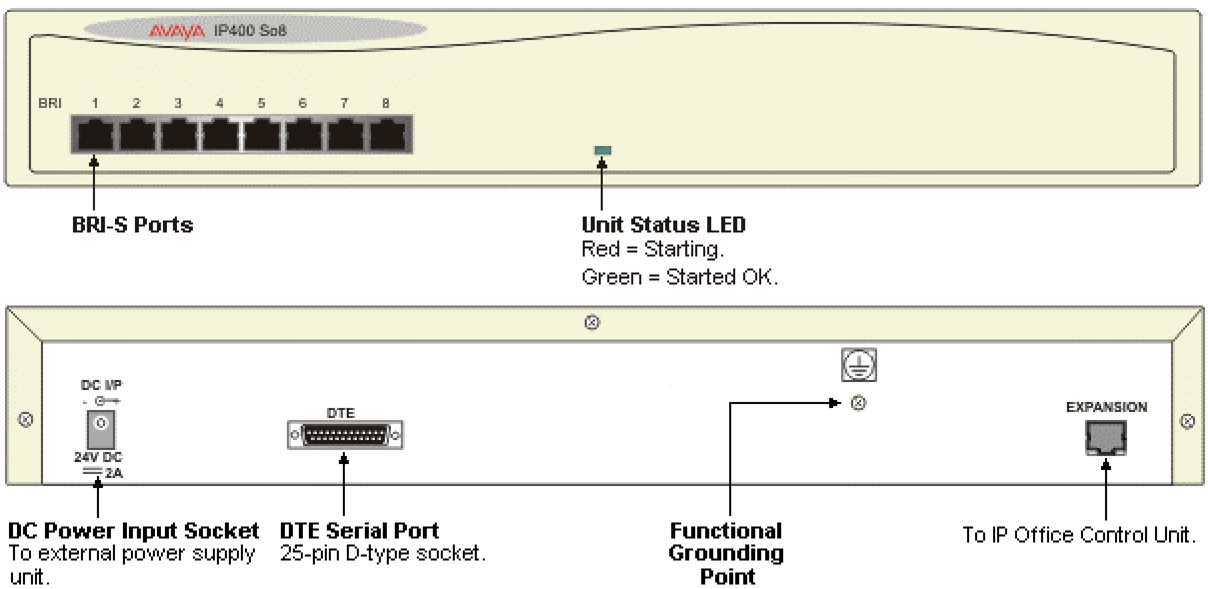


- Telephones can be located up to 1km from the control unit. For extensions located "out-of-the-building" additional line protection will be needed. For more information on cabling and out of building guidelines, see the IP Office Installation Manual.
- For systems where Direct Station Select (DSS) Units are being used, IP Office supports a maximum of:
  - Eight EU24 and or EU24BL per system.
  - Two XM24 units on each Digital Station expansion module, including the IP406 control unit, to a maximum of 10 XM24 units per system.
  - Two 4450 units on each Digital Station expansion module, including the IP406 control unit, to a maximum of 10 4450 units per system.
  - T3 DSS units.

See the Telephones Section for specific limits on the number of each type of telephone supported on DS modules.

IP400 So8 Module

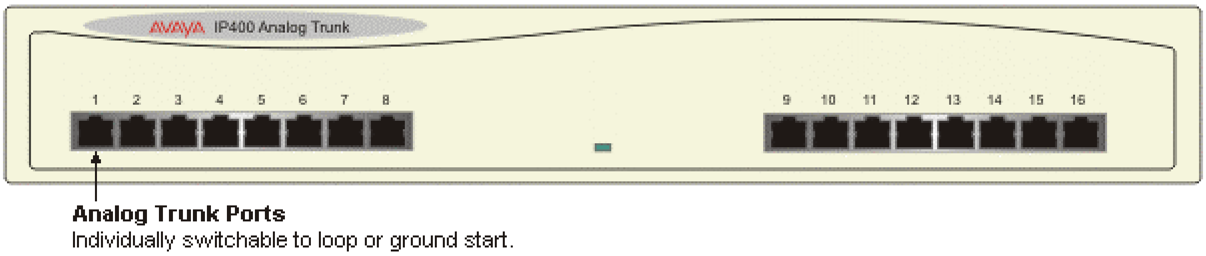
The IP400 So8 module provides 8 S-Bus interfaces for Basic Rate ISDN devices, such as video conferencing, fax servers or ISDN telephones.

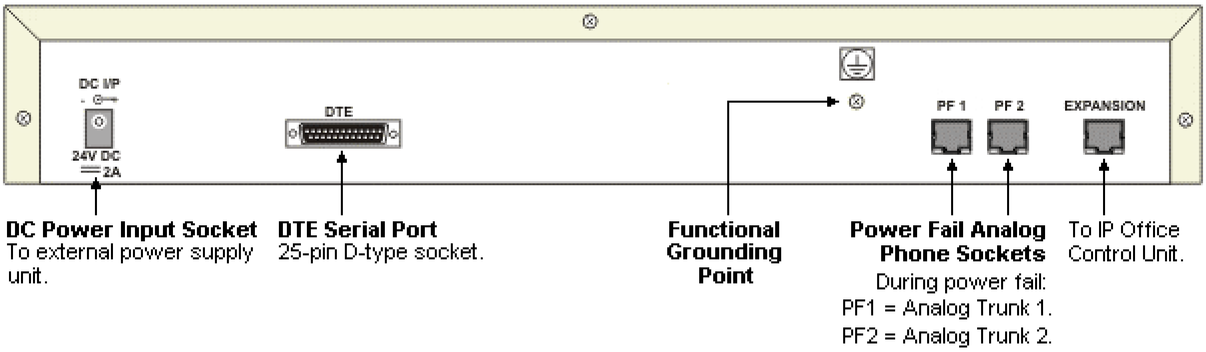


The IP Office So8 expansion module supports both point-to-point and point-to-multipoint connections. A maximum of 10 terminal endpoints identifiers (TEIs) are supported on each bus.

IP400 Analog Trunk 16 Module

This expansion module provides an additional sixteen Loop Start or Ground Start two-wire analog trunks. (Ground start trunks are not available in all territories) The first two trunks on the module which are automatically switched to power fail sockets on the rear of the unit in the event of power being interrupted must be loop start for correct power fail operation.

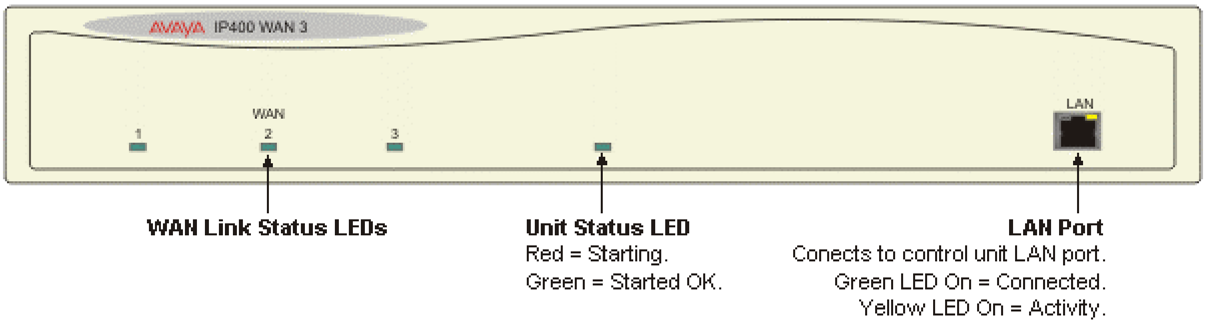


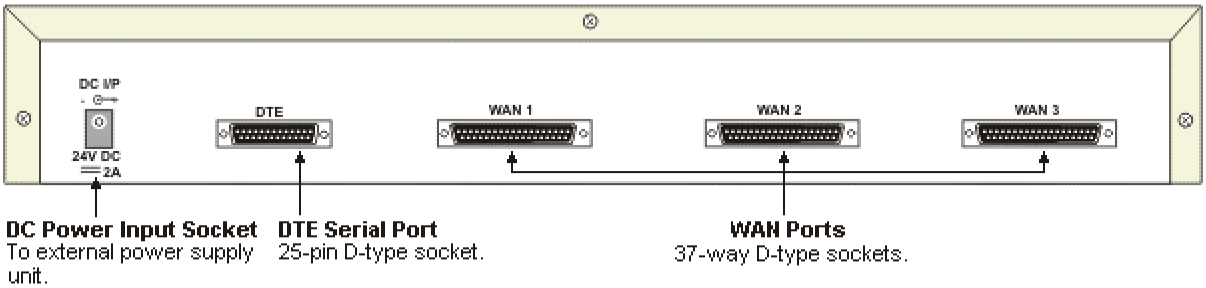


IP400 WAN3 10/100

The IP400 WAN3 10/100 module provides three WAN connections (X21, V35 or V24 via a 37way D Type socket and using an appropriate connector cable). Data rates of up to 2 Mbps are supported on each interface, the carrier providing the line dictates the actual operating speed i.e. in some territories the maximum speed may be limited to 1.544 Mbps. These WAN interfaces are identical to the single WAN connection provided as standard on the IP406 and IP412 platforms.

The IP400 WAN3 10/100 connects to the control unit through the Local Area Network via a 10/100Mbps connection and does not use an expansion port on the control unit. Small Office Edition and IP500 do not support the WAN3 10/100. All other platforms support up to two WAN3 10/100 modules.





3. Telephones

Introduction to IP Office Telephones

Avaya's range of digital and IP telephones deliver advanced productivity-boosting features, including a large display and up to a 100-entry call log. They are designed to be a cost-effective choice for any business or contact center using IP Office and bring Avaya state-of-the-art technology directly to your desktop. These telephones deliver efficient service, superior voice quality, along with cutting-edge communications features such as screen labels for call appearance/feature keys to simplify user administration.

IP Office is compatible with a wide range of wired Avaya telephones that were designed as part of other Avaya product ranges as well as the IP Office exclusive 5000 series phones. Compatible phones are as follows:

Digital Stations (DS) – connecting via DS extension ports:

- IP Office 5400 series (5402, 5410, 5420)
- MERLIN MAGIX Integrated System 4400 series (4406D, 4412D+, 4424D+) in North America
- Avaya Communication Manager 2400 series (2402, 2410, 2420)
- Integral T3 digital series (Compact, Classic, Comfort) in selected European countries
- DEFINITY 6400 series (6408D, 6416D+M, 6424D+M) (supported but not sold anymore)

IP Telephones (LAN) – connecting via Powered LAN (local or PoE)

- IP Office 5600 series (5601, 5602, 5610, 5621)
- Integral T3 IP series (Compact, Classic, Comfort) in selected European countries
- Avaya Communication Manager 4600 IP series (4601, 4602, 4610, 4620, 4621, 4625)

Wireless Telephones – connecting via a base station/access point

- Avaya 3701 and 3711 IP DECT telephones
- Avaya 3810 wireless 900 MHz telephone
- Avaya 3616, 3620 and 3626 WiFi telephones
- Avaya 3641 and 3645 WiFi telephones
- Avaya TransTalk 9040 wireless 900 MHz telephone

Avaya IP Office telephones fall into three categories;

- **Basic:** 5402, 5601, 5602, 2402, 4601, 4602, T3 Compact
- **Regular:** 5410, 5610, 2410, 4610, T3 Classic
- **Executive:** 5420, 5621, 2420, 4621, 4625, T3 Comfort

The following descriptions highlight both the common features and differences between models.

5601, 4601 Telephones

Telephone	Works on IP Office	Works on IP Office and Communication Manager
5601	✓	✗
4601	✓	✓

Common Features:

- **Display:** None.
- **Fixed Feature Buttons:** 8 - Conference, Transfer, Drop, Redial, Messages, Hold, Volume Up, Volume Down.
- **Programmable Feature Buttons:** 2 with single color indicator lamps.
- **Key Labels:** Icons used on fixed feature keys. None on programmable feature keys.
- **Speakerphone:** No.
- **Hearing Aid Compatible:** Yes.
- **Message Waiting Indicator:** Yes.
- **Personalized Ring Patterns:** No.
- **Headset Socket:** No, this phone does not support headset operation.
- **Embedded Applications:** None.
- **Upgradeable Firmware:** Yes.
- **Expansion:** None.
- **Color:** Multi-gray.
- **Power Supply:** IEEE 802.11af Power over Ethernet (PoE) or individual Avaya power supply unit (1151).
- **Connect to:** LAN using H.323 VoIP.
- **Mounting:** Desk or wall mountable.
- **Adjustable Desk Stand:** No.
- **Codecs:** G.711, G.729a/b.
- **QoS Options:** UDP Port Selection, DiffServ and 802.1p/B (VLAN)
- **SNMP Support:** Yes.
- **IP Address Assignment:** Dynamic IP address assignment only
- **Ethernet Ports:** Single 10/100 BaseT Ethernet port.

5402, 5602 SW, 2402, 4602 SW Telephones

Telephone	Works on IP Office	Works on IP Office and Communication Manager
5402	✓	✗
5602 SW	✓	✗
2402	✓*	✓
4602 SW	✓	✓

*Early 2402 telephones can make and receive call but the display will not function.

Common Features:

- **Display:** 2 lines x 24 characters.
- **Fixed Feature Buttons:** 10 - Conference, Transfer, Drop, Redial, Speaker, Messages, Hold, Mute, Volume Up, Volume Down.
- **Programmable Feature Buttons:**
    - **DS Phones:** 2 plus an additional 12 programmable feature keys can be accessed via the **FEATURE** key.
    - **IP Phones:** 2.
- **Key Labels:** Icons used on fixed feature keys. Display labels and icons used on 2 programmable feature keys.
- **Speakerphone:** Listen-only hands free speaker (no microphone).
- **Hearing Aid Compatible:** Yes.
- **Message Waiting Indicator:** Yes. On the 2402 and 5402 this is also used as a ringing call alert indicator.
- **Personalized Ring Patterns:** Yes - 8.
- **Headset Socket:** No, this telephone does not support headset operation.
- **Embedded Applications:** None.
- **Upgradeable Firmware:** DS Phones - No. IP Phones - Yes.
- **Expansion:** None.
- **Color:** Multi-gray.
- **Mounting:** Desk or wall mountable.
- **Adjustable Desk Stand:** No.

Requirements for 5402 and 2402:

- Connect to: Digital Station (DS) port.
- Power Supply: From phone system.

Requirements for 5602 SW and 4602 SW:

- **Power Supply:** IEEE 802.3af Power over Ethernet (PoE) or individual power supply unit (Avaya 1151 series).
- **Codecs:** G.711, G.729a/b.
- **QoS Options:** UDP Port Selection, DiffServ and 802.1p/q (VLAN)
- **SNMP Support:** Yes.
- **IP Address Assignment:** Static or dynamic IP address assignment.
- **Ethernet Ports:** Two port full-duplex 10/100 BaseT Ethernet switch for PC pass-through connection.
    - Auto-negotiation provided separately for each port.
    - 802.3 Flow Control.
    - Phone has priority over PC port at all times.

5410, 5610 SW, 2410, 4610 SW Telephones

Telephone	Works on IP Office	Works on IP Office and Communication Manager
5410	✓	✗
5610 SW	✓	✗
2410	✓	✓
4610 SW	✓	✓

Common Features:

- **Display:** 5 lines x 29 characters (168 x 80 pixel 4-grayscale).
- **Fixed Feature Buttons:** 10 - Conference, Headset, Transfer, Drop, Redial, Speaker, Hold, Mute, Volume Up, Volume Down.
- **Programmable Feature Buttons:**
    - **DS Phones:** 12 - in 2 switchable display pages of 6 matching the 6 physical display buttons.
    - **IP Phones:** 24 - in 4 switchable display pages of 6 matching the 6 physical display buttons.
- **Key Labels:** Icons used on fixed feature keys.
- **Speakerphone:** Two-way hands-free speaker and microphone.
- **Hearing Aid Compatible:** Yes.
- **Message Waiting Indicator:** Yes - also used as ringing call alert indicator.
- **Personalized Ring Patterns:** Yes - 8.
- **Headset Socket:** Yes.
- **Embedded Applications:** Speed Dial List (48) and Call Log (Missed, Incoming, Outgoing). Also WAP WML browser supported on IP phone models.
- **Upgradeable Firmware:** Yes.
- **Expansion:** None.
- **Color:** Multi-gray.
- **Mounting:** Desk or wall mountable.
- **Adjustable Desk Stand:** Yes - Supplied with phone.

Special Features for the 5410 and 2410:

- **Messages Button:** Dedicated button to collect voicemail.

Requirements for 5410 and 2410:

- **Connect to:** Digital Station (DS) port.
- **Power Supply:** From phone system.

Requirements for 5610 and 4610:

- **Power Supply:** IEEE 802.3af Power over Ethernet (PoE) or individual power supply unit (Avaya 1151 series).
- **Codecs:** G.711, G.729a/b.
- **QoS Options:** UDP Port Selection, DiffServ and 802.1p/q (VLAN)
- **SNMP Support:** Yes.
- **IP Address Assignment:** Static or dynamic IP address assignment.
- **Ethernet Ports:** Two port full-duplex 10/100 BaseT Ethernet switch for PC pass-through connection.
    - Auto-negotiation provided separately for each port.
    - 802.3 Flow Control.
    - Phone has priority over PC port at all times.

5420, 5621, 2420, 4621, 4625 Telephones

Telephone	Works on IP Office	Works on IP Office and Communication Manager
5420	✓	✗
5621 SW	✓	✗
2420	✓	✓
4621 SW	✓	✓
4625 SW	✓	✓

Common Features:

- **Display:** 7 lines x 29 characters.
- **Fixed Feature Buttons:** 10 - Conference, Headset, Transfer, Drop, Redial, Speaker, Hold, Mute, Volume Up, Volume Down.
- **Programmable Feature Buttons:**
    - **DS Phones:** 24 - arranged in 3 switchable display pages of 8 matching the 8 physical display buttons.
    - **IP Phones:** 24 - arranged in 2 switchable display pages of 12 matching the 12 physical display buttons.
- **Key Labels:** Icons used on fixed feature keys.
- **Speakerphone:** Two-way hands free speaker and microphone.
- **Hearing Aid Compatible:** Yes.
- **Message Waiting Indicator:** Yes - also used as ringing call alert indicator.
- **Personalized Ring Patterns:** Yes - 8.
- **Headset Socket:** Yes.
- **Embedded Applications:** Speed Dial List (104) and Call Log (Missed, Incoming, Outgoing). Also WAP WML browser supported on IP phone models.
- **Upgradeable Firmware:** Yes.
- **Expansion:** Supports the EU24 DSS expansion module (with additional Avaya 1151 power supply).
- **Color:** Multi-gray.
- **Mounting:** Desk or wall mountable.
- **Adjustable Desk Stand:** Yes - Supplied with phone.

Special Features for the 5420 and 2420:

- **Messages Button:** Dedicated button to collect voicemail.

Requirements for 5420 and 2420:

- **Connect to:** Digital Station (DS) port.
- **Power Supply:** From phone system.

Requirements for 5621 SW, 4621 SW, 4625 SW:

- **Power Supply:** IEEE 802.3af Power over Ethernet (PoE) or individual power supply unit (Avaya 1151 series).
- **Codecs:** G.711, G.729a/b.
- **QoS Options:** UDP Port Selection, DiffServ and 802.1p/q (VLAN)
- **SNMP Support:** Yes.
- **IP Address Assignment:** Static or dynamic IP address assignment.
- **Ethernet Ports:** Two port full-duplex 10/100 BaseT Ethernet switch for PC pass-through connection.
    - Auto-negotiation provided separately for each port.
    - 802.3 Flow Control.
    - Phone has priority over PC port at all times.

Special Features for 5621 SW and 4621 SW:

- **Display Backlight:** The display has a backlight for improved contrast. Standby mode turns off backlight after time-out.

Special Features for 4625 SW:

- **Color Backlight Display:** The display is full color and has a backlight for improved contrast.

Note: While still supported, the 5620SW and 4620SW phones are no longer available for purchase.

EU24 and EU24 BL Expansion Modules

The EU24/EU24BL are phone expansion modules that work in association with a 5420, 5620/1, 2420, 4620/1, 4625 phones. They provide an additional 24 programmable buttons with associated display label and status icons. Only one EU24 can be used per phone. The EU24BL has a backlight and is for use with the 4621 and 5621 only.

The EU24/EU24BL supports an additional 24 Call Appearance/Feature buttons, by displaying a column of 12 buttons at a time, with a dotted line separating the two columns.

Common Features

- 24 Programmable call appearance/feature keys.
- Automatically labeled from the system (no paper labels).
- Connects directly to the associated phone.
- Requires an Avaya 1151 series power supply, even for IP phones already using Power over Ethernet (PoE).
- IP Office supports a maximum of eight EU24/EU24 BL's on each IP Office system.

Telephone	EU24	EU24BL
2402/5402	✗	✗
2410/5410	✗	✗
2420/5420	✓	✗
4601/5601	✗	✗
4602/5602	✗	✗
4610/5610	✗	✗
4620/5620	✓	✗
4621/5621	✓	✓

T3 Series Phones

T3 Telephone Range

The T3 range of digital Upn and IP telephones provide European style with context sensitive displays and are available in select European countries only.

T3 IP phones do not support direct media and require the use of a VCM channel for the duration of a call except when calling another T3 IP phone, see **T3 Interworking**. The number of simultaneous T3 IP phone calls is limited to the number of VCM channels available up to a maximum of 50.

T3 Compact

Common Features:

- **Display:** 1 line with 24 characters alphanumerical plus one line icons
- **Fixed Feature Buttons:** 3 keys with printed text labels and 2 for Audio Volume control
- **Programmable Feature Buttons:** 3 keys with indicators and printed text labels, 2 keys with printed text labels
- **Speakerphone:** full duplex hands free speaker and microphone.
- **Hearing Aid Compatible:** Through optional handset
- **Message Waiting and call log Indicator:** Yes
- **Personalized Ring Patterns:** Yes, 8 ring patterns
- **Embedded Applications:** Navigation Cursor Control, Call signaling via LED and/or ringer; Alphanumeric entry via dialing keypad.
- **Color:** Graphite gray or polar white.
- **Mounting:** Desk or optional wall mountable.
- **Adjustable Desk Stand:** No

Features for T3 Upn only:

- **Upgradeable Firmware:** No.
- **Optional Add-Ons:** up to 3 DSS Modules, T3 Headset link for wired headsets only
- **Headset Socket:** No
- **Connect to:** Digital Station (DS) port.
- **Power Supply:** From phone system.

Features for T3 IP only:

- **Upgradeable Firmware:** Yes
- **Headset Socket:** Yes
- **Optional Add-Ons:** No
- **Power Supply:** IEEE 802.3af Power over Ethernet (PoE) or individual power supply unit.
- **Codecs:** G.711, G.729a/b.
- **QoS Options:** UDP Port Selection, DiffServ and 802.1p/q (VLAN)
- **SNMP Support:** No.
- **IP Address Assignment:** Static or dynamic IP address assignment.
- **Ethernet Ports:** Two port full-duplex 10/100 BaseT Ethernet switch.
    - Auto-negotiation provided separately for each port.
    - 802.3 Flow Control.

T3 Classic

Common Features:

- **Display:** graphical, 4 lines x 26 characters
- **Fixed Feature Buttons:** 5 preprogrammed keys with printed text labels and 2 for Audio Volume control
- **Programmable Feature Buttons:** 6 preprogrammed keys with indicators and printed text labels, 4 programmable keys with printed text labels
- **Speakerphone:** Two-way hands free speaker and microphone.
- **Hearing Aid Compatible:** Through optional handset
- **Message Waiting and call log Indicator:** Yes
- **Personalized Ring Patterns:** Yes, 8 ring patterns.
- **Headset Socket:** no
- **Embedded Applications:** Navigation Cursor Control, Call signaling via LED and/or ringer; Alpha entry via dialing keypad.
- **Optional Add-Ons:** T3 Headset link for wired headsets only
- **Color:** Graphite gray or polar white.
- **Mounting:** Desk
- **Adjustable Desk Stand:** Display adjustable

Features for T3 Upn:

- **Upgradeable Firmware:** No.
- **Optional Add-Ons:** up to 3 DSS Modules
- **Connect to:** Digital Station (DS) port.
- **Power Supply:** From phone system.

Features for T3 IP:

- **Upgradeable Firmware:** Yes.
- **Optional Add-Ons:** up to 3 DSS Modules with AEI/Headsetlink,
- **Power Supply:** IEEE 802.3af Power over Ethernet (PoE) or individual power supply unit.
- **Codecs:** G.711, G.729a/b.
- **QoS Options:** UDP Port Selection, DiffServ and 802.1p/q (VLAN)
- **SNMP Support:** no.
- **IP Address Assignment:** Static or dynamic IP address assignment.
- **Ethernet Ports:** Two port full-duplex 10/100 BaseT Ethernet switch.
    - Auto-negotiation provided separately for each port.
    - 802.3 Flow Control.

T3 Comfort

Common Features:

- **Display:** graphical 17 lines x 40 characters, Integrated keyboard
- **Fixed Feature Buttons:** 5 preprogrammed keys with printed text labels and 2 for Audio Volume control
- **Programmable Feature Buttons:** 6 preprogrammed keys with indicators and printed text labels, 6 preprogrammed keys with printed text labels, 10 user programmable keys with associated display labels
- **Speakerphone:** Two-way hands free speaker and microphone.
- **Hearing Aid Compatible:** Through optional handset
- **Message Waiting and call log Indicator:** Yes
- **Personalized Ring Patterns:** Yes, 8 ring patterns.
- **Headset Socket:** No
- **Embedded Applications:** Navigation Cursor Control, Call signaling via LED and/or ringer
- **Optional Add-Ons:** T3 Headset link for wired headsets only
- **Color:** Graphite gray or polar white.
- **Mounting:** Desk
- **Adjustable Desk Stand:** Display adjustable

Features for T3 Upn:

- **Upgradeable Firmware:** No.
- **Optional Add-Ons:** up to 3 DSS Modules
- **Connect to:** Digital Station (DS) port.
- **Power Supply:** From phone system.

Features for T3 IP:

- **Upgradeable Firmware:** Yes.
- **Optional Add-Ons:** up to 3 DSS Modules with AEI/Headsetlink,
- **Power Supply:** IEEE 802.3af Power over Ethernet (PoE) or individual power supply unit.
- **Codecs:** G.711, G.729a/b.
- **QoS Options:** UDP Port Selection, DiffServ and 802.1p/q (VLAN)
- **SNMP Support:** No.
- **IP Address Assignment:** Static or dynamic IP address assignment.
- **Ethernet Ports:** Two port full-duplex 10/100 BaseT Ethernet switch.
    - Auto-negotiation provided separately for each port.
    - 802.3 Flow Control.

T3 DSS Expansion Modules

The T3 DSS Module is a phone expansion module that is compatible with all T3 Upn and T3 IP Telephones except the T3 IP Compact. Each module provides an additional 36 programmable buttons with associated printed text labels and indicators, and can be programmed for lines, groups or speed dial numbers. 3 DSS Modules can be added to each T3 phone. Power is provided by T3 Upn telephones, but an external power adapter is needed for each DSS module when used on T3 IP telephones.

IP Office 406, IP412 and IP500 support a maximum of 30 T3 DSS modules per control unit.

T3 IP telephone interworking with other Avaya telephones and endpoints

The Avaya T3 IP Telephones are compatible with different Avaya telephones and endpoints and use Voice Compression Channels (VCMs) according to the following table.

From	To	Method
T3 IP telephone	T3 IP Telephone	RTP relay, no VCM
	IP DECT 3700 series telephone	RTP relay, no VCM
	PhoneManager PC Softphone	RTP relay, no VCM
	Analog or ISDN or digital telephone	1 VCM channel
	Connection across the Small Community Network	RTP relay, no VCM

Mobility Solutions

Avaya Mobility Solutions

Avaya IP Office Mobility Solutions include analog, digital and IP-based WiFi wireless phones. These are solutions employees can use every day to work more effectively and be more responsive to customers — all while increasing revenues and keeping communication costs firmly under control. Also, Avaya IP Office Mobility Solutions integrate seamlessly with IP Office, enhancing each customer's investment. IP Office's in-building Mobility Solutions improve communication with staff that, because of the function they perform, are mobile within the workplace. Using wireless technology, such individuals may be instantly contactable, with many obvious benefits;

- The wireless telephone is carried in the pocket, so users are not tied to the desk in order to remain in contact.
- Users may be contacted instantly to ensure fast, accurate decision making and immediate response to problems through planned radio coverage with no blind spots

Avaya Mobility Solutions

IP Office supports the following wireless solutions:

- DECT in the EMEA and NA regions and in selected APAC countries.
- Digital Wireless North American market.
- Avaya VoIP Wi-Fi Solution offered worldwide.

IP Office supports the following VPN mobility solutions:

- VPN phone client on selected 46XX and 56XX Series IP phones offered worldwide.

Mobility - Avaya IP DECT

The IP DECT solution delivers the productivity-boosting benefits of IP and wireless communications across multiple offices in a convenient, lightweight handset. It provides businesses with a highly functional wireless solution with the ability to scale to support large numbers of users. This system also supports users in different offices connected via a WAN. The Avaya IP DECT solution radio fixed part (RFP) or base station connects to the IP Office using an IP protocol based on H.323.

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

In EMEA and APAC this system supports the 3701 and 3711 handsets.

In North America, only the 3711 handset is supported.

Avaya recommends that for new deployments, for full feature functionality the 3711 handset be used with the IP DECT solution.

Note: The regulatory requirements for the radio part (base station and Handset) are slightly different in the US and Canada compared to EMEA and APAC. Therefore, while providing the same functionality, the hardware is different in these two regions.

Each Base station has the following features:

- 8 simultaneous Voice and up to 12 Signaling Channels.
- Codec G.711, G.723, G.729 for base station IP trunk connection.
- Handover
  While in motion, the handset performs continuous measurements to determine which IP DECT base station has the strongest signal. The one that can be best received is defined as the active Base station. To prevent the handset from rapidly switching back and forth between two base stations that are equally well received, threshold values are used. Handover between base stations occurs seamlessly whether a call is active or not.
- DECT Networking
  An IP DECT telephone can travel from one office to another which is connected over a wide area network (WAN) link and make and take calls. In this scenario the main IP DECT controller remains at one “headquarters” location.

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

- **DDI/DID**
  Since each wireless handset is an extension on the IP Office system calls may be routed directly using a DDI/DID number.
- **Transfer**
  Calls may be transferred to DECT extensions by operators or other extension users and DECT extension users may transfer callers to any other extension user.
- **Hunt Group compatibility**
  Wireless handsets may be programmed as members of groups and answer calls in the same manner as any other extension within that group.
- **Group working**
  Wireless handsets may be programmed as members of groups and attract calls in the same manner as any other extension within that group. DECT handsets must NOT be configured into collective groups.
- **Divert destination**
  Users may initiate any or all diverts from an Avaya desk phone to a wireless handset.
- **Twinning**
  Twinning allows calls to a user main extension number to alert at both that extension and a secondary extension. Though not restricted to DECT, this feature is aimed primarily at users who have both a desk phone and a wireless extension. Calls from the secondary twinned extension are presented as if from the users main extension. Presentation of call waiting and busy is based on whether either of the twinned extensions is in use. In North America this functionality became available in Release 4.0.7.

Avaya IP DECT System licensing

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

No separate PC or software is required with this system.

In all regions, a “plug and play” licensing mechanism is available: It consists of a pre-licensed and ready to go two-base-station bundle (“IP DECT IPO STARTER KIT ”) and two pre-licensed base stations (“IP DECT RFP32/34 UPG KIT”) that can be added to the system independent of the number of licenses in ADMM. This allows easier deployment and upgrades of systems without the need to buy a separate upgrade-license. For IP Office we recommend to use the “Starter Kits” and the “Upgrade Kits” for new installations for added flexibility and to minimize the installation effort.

The bundles that have previously been available in EMEA will continue to be available and are compatible with the pre-licensed base stations described above, if the latest software is installed on the IP-DECT system, e.g. upgrade an existing 5-base-station system with an “Upgrade-Kit” when adding an extra base-station instead of upgrading the system-license to a 6+ license.

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

IP DECT Capacities

Feature	IP DECT
Maximum handsets	120
Maximum base-stations	32
Total base-stations/repeaters	32
Maximum simultaneous calls	100*

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

Mobility - 900MHz Digital Wireless

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

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

A maximum of 5 Avaya 3810 wireless handsets can be used in the same zone of radio coverage, 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.

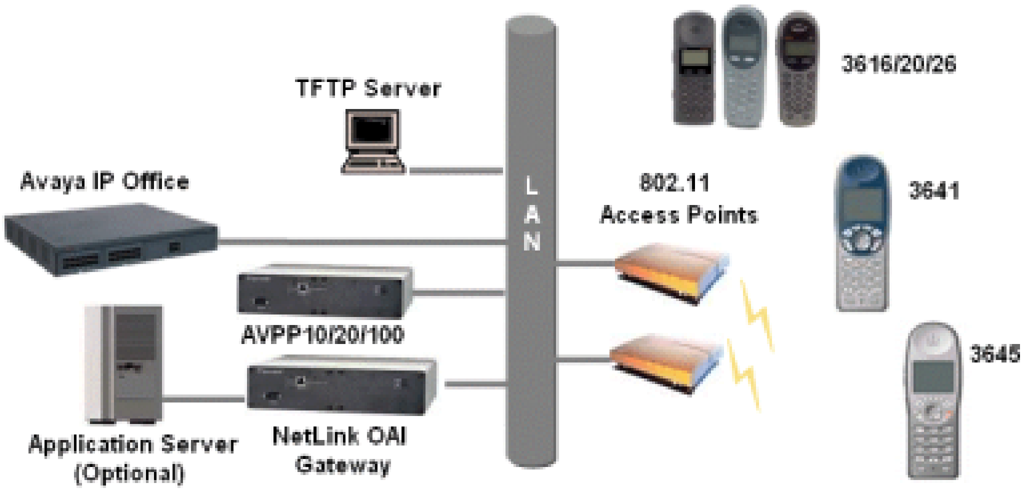
Mobility - WiFi (802.11)

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.

Avaya 3616, 3620 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 Processors (10, 20 and 100) to ensure voice quality over Wireless LAN's.

The newly available 3641 and 3645 phones provide an improved user-interface, a new lightweight design and a radio that supports several WIFI standards (802.11a/b/g). With these handsets customers have an increased choice to fit their needs and infrastructure.

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. 3616, 3620 and 3626 telephones are available for direct sequence 802.11b Wi-Fi networks; the 3641 and 3645 will also work in 802.11a and 802.11g networks. These phones are also field upgradeable through external TFTP clients (not included), so telephones can be updated with new protocols, features, and capabilities as they become available.



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.

Wireless IP Terminals

Users can have a choice of 5 WiFi phones to meet their in building mobility needs:

- 3 Phones supporting the 802.11b standard:
    - Avaya 3616 supports a broad range of enterprise applications and is ideally suited for general office, financial or hospitality industries. This compact handset offers a high-resolution graphic display and menu driven functions.
    - Avaya 3620 is specifically designed to meet the needs of the healthcare vertical. The 3620 comes standard with a backlit display.
    - Avaya 3626 is an extremely durable handset for workplace applications in industrial environments. This phone is easy to use and requires minimal training. Push-to-talk functionality is also available for broadcast communication between employees, eliminating the need for two-way radios or walkie talkies. The large ear piece seals out background noise and provides comfort for frequent or lengthy calls.
- 2 Phones supporting the 802.11 a/b/g standard. Both of these handsets are resistant to dust and spraying water and therefore also suitable for harsh environments. They also offer office-quality speaker-phone functionality.
    - Avaya 3641 supports a broad range of enterprise applications and is ideally suited for general office, financial or hospitality industries. This compact handset offers a high-resolution backlight graphic display a new, improved user-interface and design and a lightweight form factor.
    - Avaya 3645 is a slightly larger version that in addition supports “push-to-talk” functionality for broadcast communication between employees. Due to its rubberized sized grips and the larger ear cup it is especially well suited in noisy and industrial environments.

Avaya Voice Priority Processors

The Avaya Voice Priority Processor (AVPP) is an Ethernet LAN appliance that works with access points to provide Quality of Service (QoS) on the wireless LAN. All packets to and from the wireless phones pass through the AVPP and are encapsulated for prioritization as they are routed to and from IP Office. AVPP is fully compliant with the IEEE 802.11a/b/g standards.

AVPP is required for QoS because the current IEEE 802.11a/b/g wireless LAN standards provide only limited mechanism for differentiating audio packets from data packets. It also delivers quality of service by limiting the number of phones that are connected to one access point in order to avoid quality problems. In addition AVPP ensures that the phone can run in energy-efficient mode when not in use. The following AVPPs are available to meet customer needs:

- AVPP100: Serves 80 calls simultaneously.
- AVPP020: Serves 20 powered-on handsets.
- AVPP010: Serves 10 powered-on handsets.

Wireless Access Points

When using the Avaya Wireless IP solution, customers can utilize wireless access points from various vendors. The list of compatible wireless access points is large and constantly growing. Please visit http://www.spectralink.com/consumer/support/index.jsp and select “WLAN Compatibility List” for the latest information.

Benefits

- Supports the 802.11b or 802.11 a/b/g standard for Wi-Fi networks converging voice and data over a single network.
- Seamless integration with IP Office.
- Excellent voice quality on converged wireless networks.
- Lightweight, durable handsets specifically designed for workplace use.
- Improved display, battery life, processor power all with lower costs.
- Increased range of AVPP's to address the needs of diverse construct sizes.
- 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).
- Single, Dual, & Quad Chargers for the 3641 and 3645 phones.
- Belt Clip.
- Nylon Pouch.
- Carrying case with Lanyard.
- Hands Free Pouch.
- Noise canceling headset.
- Over the ear headset.

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

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

3616 Wireless Telephone

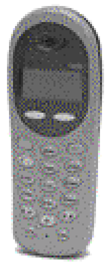
The Avaya 3616 IP Wireless Telephone is a WiFi (802.11b) telephone that runs using H.323.

The 3616 supports the following features:

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

3620 Healthcare Wireless Telephone

The Avaya 3620 IP Wireless Telephone is a WiFi (802.11b) telephone that runs using H.323.



The 3620 supports all of the features of 3616 with the following differences:

- Designed for health care environments
- Waterproof durable design.
- Display Backlight:
- Manufacturer's Liquid damage warranty

3626 Ruggedized Wireless Telephone

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

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

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

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

3641 Ruggedized Wireless Telephone

The Avaya 3641 Wireless Telephone is a WiFi standard (802.11a/b/g) telephone that runs using H.323.

The 3641 supports the following features:

- Slim lightweight design with large display.
- Backlight display with Icons.
- Simple to use with improved user interface.
- Navigation and soft keys for simple access to frequently used operations.
- Office-quality speakerphone for hands-free operation.
- 802.11a/b/g standard-compatible.
- Radio Frequency 2.4000 GHz (b/g) or 5.8 GHz (a).
- FCC certification Part 15.247.
- Management of telephones via DHCP and TFTP.
- Voice encoding G711, G.729a.
- Wired Equivalent Privacy (WEP), 40bit and 128 bit and 802.11i (PSK) for secure communication.
- Lithium Ion Battery pack with up to 8 hours talk time and 160 hours standby.
- IP-53 Design (Liquid/dust protection).
- MIL 810F Design (Shock protection).
- Clips, cases, lanyard.

3645 Ruggedized Wireless Telephone

The Avaya 3645 Wireless Telephone is a WiFi standard (802.11a/b/g) telephone that runs using H.323.

The 3645 supports all of the features of 3641 with the following additions:

- Push-to-talk (PTT) functionality for workgroup communication
- Enlarged earpiece for operation in noisy environments
- Rubberized grips for improved ergonomics and durability

3701 IP DECT Telephone

This handset is supported on the Avaya IP DECT system only. This handset is not available in North America.

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

Optional telephone accessories include:

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

3711 IP DECT Telephone

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

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

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

Optional handset accessories include:

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

Digital Wireless 3810 Telephone

Features

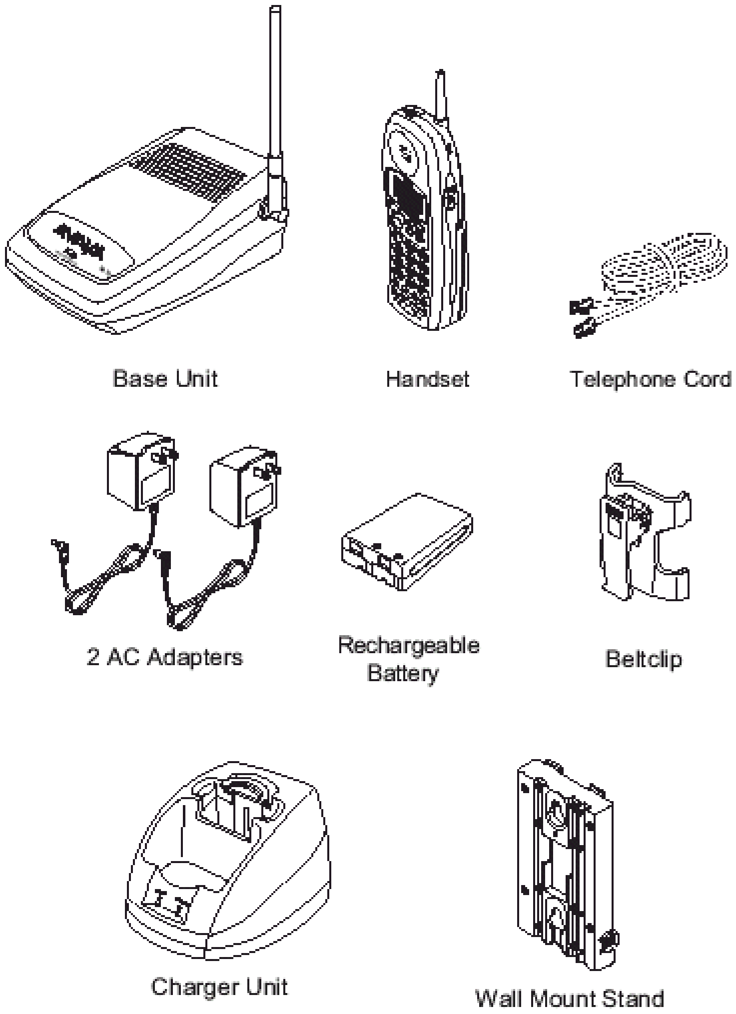
- 2-line, 32 character Handset Liquid Crystal Display (LCD).
- 10 hours of talk time, and 4 days of standby time.
- 4 displayed operation modes indicating Talk, Ringer On/Off, Battery Low, and Message Waiting.
- Single button access to fixed features – Hold, Transfer, Conference, and Redial.
- 4 programmable buttons to access features on the PBX.
- 20 Number Memory for quick and easy speed dialing
- 10 channels, supporting up to 10 simultaneous conversations in overlapping radio coverage areas.
- Headset jack.
- Ringer and Handset volume control.
- User selectable ring type.
- Vibrate alert.
- Redial Button
- Base Unit and Charger Unit.

The Avaya 3810 Wireless Telephone is a digital telephone designed to work with IP Office from release 2.0 and above by connecting to a Digital Station (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.

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

The Avaya 3810 is delivered 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.



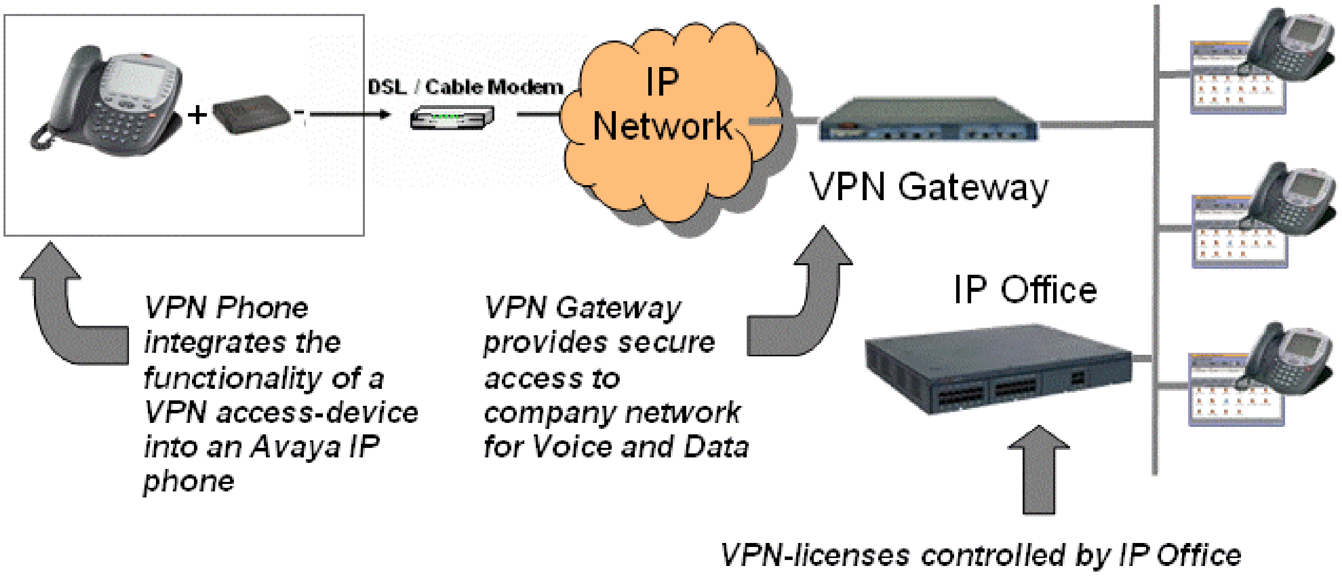
VPN Phone Software

VPN Phone is a full-featured IP Telephony solution that provides secure communication over public ISP networks to an IP Office system at the company headquarters.

It is a software-only product that runs on the standard 5610/5620/5621 or 4610/21 IP telephones. In combination with one of these phones and the most popular VPN gateway products, the software extends enterprise telephony to remote locations.

VPN Phones offer the full IP Office telephony features that are available on IP Office IP phones at the users desktop in a remote location like a home-office:

Licenses for VPN Phone are controlled by IP Office.



VPN Phone is ideal for IP Office customers supporting "work-at-home" users:

- Virtual Office workers
- Remote workers
- Remote call center
- Business continuity support
- Very small locations that require a single phone only
- Temporary installations such as conferences, off-site meetings, and trade shows

VPN Phone has been tested with a number of VPN-gateways from major vendors like Cisco or Juniper as well as with smaller VPN-access devices from companies like Netgear, Kentrox and Adtran. Refer to the support pages (support.avaya.com) for a list of available application notes on VPN-gateways.

Other Ranges

Other Ranges of Telephones Compatible with IP Office

Avaya has a wide range of communication products so we do our best to support as many telephones from other Avaya product families established in the global market such as MERLIN MAGIX and DEFINITY.

4400 Series

4406D Telephone

This range of telephones is only available in North America.

The 4406 supports the following features:

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

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

This phone is not supported on the IP500 DS8 Extension Card. For the IP500 it will work on an external Digital Station Expansion Module.

4412D Telephone

This range of telephones is only available in North America.

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

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

Note: A maximum of twenty-seven 4412D telephones are supported on the DS30 (version 2) expansion module at PCS level 5. Earlier DS30 expansion modules will only support sixteen of these telephones.

This phone does not support personalized ringing.

This phone is not supported on the IP500 DS8 Extension Card. For the IP500 it will work on an external Digital Station Expansion Module.

4424D Telephone

This range of telephones is only available in North America.

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

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

Note: A maximum of twenty-four 4424D telephones are supported on the DS30 (version 2) expansion module at PCS level 5. Earlier DS30 expansion modules will only support sixteen of these telephones.

This phone does not support personalized ringing.

This phone is not supported on the IP500 DS8 Extension Card. For the IP500 it will work on an external Digital Station Expansion Module.

DSS4450 Unit

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

Each DSS4450 provides an additional 60 programmable keys with single red lamps except for the bottom two rows that have green lamps. The DSS4450 requires an auxiliary Avaya power supply unit and must be used with the cables supplied.

IP Office supports a maximum two 4450 units on each Digital Station expansion module, including the 406 V2 control unit.

This phone is not supported on the IP500 DS8 Extension Card. For the IP500 it will work on an external Digital Station Expansion Module.

Analog Telephones

Analog Telephones / POTS

As well as providing a lower cost alternative to system specific telephones, analog telephones can still deliver a high degree of functionality on IP Office. They are particularly appropriate in applications where users require lower entry costs and can be used with Phone Manager for a high proportion of call control.

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

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

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

- Analog phones connect to IP Office via ports marked PHONE ports.

Avaya 6200 Analog Telephone (North America)

The 6200 range of telephones are single-line analog phones that require one tip and ring pair for operation. This series of telephones have a Ringer volume control on the side of the telephone and a Handset volume control on the front of the phone. They use DTMF dialing only and support the Positive Disconnect function. In addition, these phones have a Message light, a recall button that allows access to system features, a redial button that allows automatic redial, a hold button with a single associated light, and a data jack on the rear of the telephone. The 6219 phone adds 10 programmable dialing buttons and the 6221 phone adds a built-in speakerphone with mute capability.

Feature Table

Analog Telephone Features	6211	6219	6221
Programmable buttons (10 buttons)	✗	✓	✓
Program Keylock	✗	✓	✓
Pause	✗	✓	✓
Redial	✓	✓	✓
Speaker	✗	✗	✓
Flash	✓	✓	✓
Hold (with indicator light)	✓	✓	✓
System Hold	✗	✓	✓
Mute	✗	✗	✓
Handset Volume Control	✓	✓	✓
Ringing Volume Control (3 position)	✓	✓	✓
Ringing Patterns (2)	✓	✗	✗
Personalized Ringing	✗	✓	✓
Message Waiting Light	✓	✓	✓
Desk/Wall Mount	✓	✓	✓
Data Jack	✓	✓	✓
Colors	White/Grey	White/Grey	White/Grey
Ringer Equivalency	0.7A, 1.6B	0.5A, 1.5B	0.5A, 1.5B
Hearing Aid Compatible	✓	✓	✓
Positive Disconnect	✓	✓	✓
DTMF Dialing	✓	✓	✓
Specialty Handset Support	✓	✓	✓

Interquartz Gemini Phones (EMEA and APAC)

Avaya have tested the new generation Interquartz Gemini analog telephones with IP Office to ensure that telephone and system are compatible. The Gemini phones offer good value for money without compromising on quality. Their stylish design and rugged build quality make them a popular choice for buyers on a limited budget.

For sales enquiries and product information contact Interquartz at avaya-enquiries@interquartz.co.uk.

Basic telephone 9330-AV

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

CLI Feature phone 9335-AV

All features of 9330-AV plus:

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

Hotel Phone 9281-AV

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

Doorphone Entry Systems for IP Office

Doorphones offer convenience and security. Depending on the needs of the environment, door phones may allow internal users to not only speak with someone who is outside, but also to easily allow the visitor entrance to the facility or residence. Doorphones can be connected to the Avaya IP Office base unit in a variety of ways, providing design flexibility based upon needs.

All of the IP Office base units include an external output port. Connections of doorphones to these ports enable the user to gain access to the premises through default system short codes, through the optional Phone Manager Pro application, and through the optional VoiceMail Pro application. The flexibility of the IP Office provides the ability for short codes to be customized to a code more desirable for users. By using the Phone Manager Pro application, users can label the icons within the application a descriptive name such as Receiving Door or Front Door. The flexibility of VoiceMail Pro allows the visitor to enter a predetermined code from the phone granting access. This scenario is particularly useful in areas when co-workers are working at another site. Additionally, many doorphones can be connected to station or trunk ports available on IP Office.

The Avaya IP Office system offers three doorphone solutions to choose from:

- **Avaya Universal Doorphone System (North America)**
- **Kalika Communications Doorphone Entry System (EMEA)**
- **Interquartz Doorphone (EMEA)**

Avaya Universal Doorphone System:

- System consists of a controller and a speaker.
- The speaker is mounted securely on the wall and is connected to the controller, which normally resides in the equipment room. The controller is connected to a trunk port.
- Users with the trunk appearance will be notified when a visitor has pressed the Push button located on the weatherproof speaker.
- Each controller supports two speakers, for example Front Door and Back Door.
- Custom ringing mode distinguishes doorphone calls from external calls.
- Call waiting tones indicate which doorphone is calling and distinguish a doorphone call from an external line call.
- Calls can be placed on hold when visitors call from the doorphone.
- Commercial or residential security is provided via two-way hands-free communication from a door or gate.

Kalika Communications Doorphone Entry System:

- Supports a doorway intercom system with up to 46 buttons.
- The system can be programmed to enable multiple extensions to answer and control the operation of the door and can be used with both single and multiple door entry systems.
- It is ideal for apartment complexes or where different companies occupy different floors and require their own unique door entry solution.
- The Kalika Communications Control Unit is available in several versions and is equipped to provide two-way voice communications, electrical lock control and label lamps.
- It is weatherproof and remotely programmable.

Interquartz Doorphone:

- Choice of models (1, 2, and 4 button)
- Slim design (16mm thick)
- Strong aluminium casing
- Optional PC configuration
- Remotely programmable via DTMF
- Connection via analog extension port or trunk port
- Relay lock control
- Backlit inlay cards
- Internal heating system
- Day/Night service
- Combination lock control

Headsets

Headsets

Avaya offers ergonomically designed communication headsets and amplifiers for the Avaya IP Office telephones. This full line of professional and contact center solutions set the standard in sound quality and durability. Avaya headsets are designed for maximum, all-day comfort and are available in styles that suit nearly any wearer and any usage pattern.

Whether you want the freedom to communicate hands-free while working at your desk, or the ability to roam while talking, you will find a solution that suits your individual needs.

To view the full range of Avaya headsets:

1. Go to **http://www.avayaheadsets.com/**.
2. Identify the IP Office telephone you are using.
3. Choose an amplifier based on compatibility and features.
4. Choose the style of headset that best suits your needs. For instance, noise-canceling headsets are great in a busy office or when using VoIP telephones.

Summary

Summary

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

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

IP Office worldwide digital phones:

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

IP Office worldwide H.323 IP phones:

- 5601 IP Telephone.
- 5602SW IP Telephone.
- 5610SW IP Telephone.
- 5621 IP Telephone.

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

North America and CALA

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

EMEA and APAC

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

Phones supported worldwide in addition to 5400 Series.

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

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

- 4601 IP Telephone.
- 4602 IP Telephone.*
- 4602SW IP Telephone.
- 4610 IP Telephone.
- 4621 IP Telephone.
- 4625 IP Telephone.*
- 3616 Executive Wireless (WiFi) Phone.
- 3620 Healthcare Wireless (WiFi) Phone.
- 3626 Ruggedized Wireless (WiFi) Phone.
- 3641 Ruggedized Wireless (WiFi) Phone.
- 3645 Ruggedized Wireless (WiFi) Phone.

- For maximum cabling distances please refer to the IP Office Installation Manual.
- Those phones that support hands free operation are intended for individual use only, not for group and conference room operation.

*These phones are no longer available as new from Avaya but are still supported by Avaya IP Office R4.0.

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

4. Features

Telephony Functions & Call Handling

IP Office provides a comprehensive telephony feature set to enable a fast and efficient response to a telephone call. Features such as Caller ID display and call tagging allow employees to see who is calling and who they are calling before they pick the call up. Client information can even be 'popped-up' on the user's PC.

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

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

Basic Call Handling

Tones

IP Office generates the correct user tones for the geography. These tones are generated for all IP Office extension types, analog, digital and IP.

Supported tones are:

- Dial, both primary and secondary depending on geography
- Busy
- Unobtainable
- Re-order
- Conferencing tone depending on geography

Caller ID

Feature

- Display of the caller's number on incoming calls, where supplied by the service provider.
- Sending of calling number on outgoing external calls.

Benefit

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

Description

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

Where IP Office Phone Manager, or the TAPI service is used to link to database software on the users PC, it is possible to have an automatic query performed on the supplied Caller ID and have the caller's record in front of the user before the call is answered.

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

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

Hold

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

See also **Park**.

Toggle Calls

Toggle Calls cycles round each call that the user has On Hold to their extension locally within the system, presenting them one at a time to the user

Hold Call Waiting

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

Hold Music (Music on Hold)

The IP Office system supports a single source of music on hold, either internal or external. The internal source uses a .WAV file saved either in volatile memory, or on the optional memory card in a Small Office Edition or IP 406. The .WAV file must be 16bit PCM mono and sampled at 8Khz with a maximum duration of 30 seconds.

External music on hold sources connect to the 3.5mm Audio socket on all IP Office control units.

Park

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

The call park facility is available through the user's telephone, Phone Manager or SoftConsole. Calls are Parked against a ‘park slot number’ which can be announced over a paging system so the person the call is for can go to any phone and collect the call by dialling the park slot number.

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

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

Automatic Callback

Feature

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

Benefit

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

Description

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

You can also set a callback when free or a callback when next used using a short code without attempting a call.

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

This feature is supported across the IP Office Small Community Network.

Direct Inward Dialing (DID /DDI)

This relies on the local telephone exchange passing all or part of the dialed number to the IP Office. This number can then be used by IP Office call routing software to route the call to an individual phone, or groups of phones. This service is typically used to reduce the workload on a reception position by giving members of staff or departments individual numbers so they can be called directly. For convenience it is common to have the extension or group number the same as the digits supplied from the network, but IP Office can convert the number to what ever number is needed by the business, within limits

In North America, T1 circuits are required for DID.

Transfer

Call Transfer allows users to transfer a call in progress to another phone number – either internal extension or external public number. The caller is placed on hold while the transfer is performed.

If the phone is put down before the destination has answered, the original caller will be automatically transferred. This is called an Unsupervised or Blind Transfer. Alternatively, a user can wait for the destination to be answered and announce the transfer before hanging up to complete the transfer. This is called a Supervised Transfer.

Unless restricted by the system administrator, the IP Office makes no differentiation between internal or external call transfers.

Distinctive and Personalized Ringing

The IP Office uses different ringing sequences to indicate the type of call, for example whether internal or external. This feature is called 'distinctive ringing'. For analog phones the distinctive ringing sequences used are adjustable. For digital and IP phones the distinctive ringing sequences are fixed as follows;

- Internal Call: Repeated single-ring.
- External Call: Repeated double-ring.
- Ringback Call: Single ring followed by two short rings.

This ring is used for calls returning from park, hold or transfer. It is also used for call back when free and voicemail ringback calls.

This feature is supported across the IP Office Small Community Network

Personalized Ringing

In IP Office the term personalized ringing is used to refer to changing the sound or tone of a phone's ring. On many Avaya digital phones, the ringer sound can be personalized. Changing the ringer sound does not alter the ring sequence used for distinctive ringing. This feature is local to the telephone and not supported on all types of telephones.

Message Waiting Indication

Message waiting indication (MWI) is a method IP Office uses to set a lamp or other indication on compatible telephones when a new message has been left for the user, either in a personal voice mailbox or in a group mailbox or call back message. When the message has been played or acknowledged, the lamp is turned off.

All Avaya digital and IP phones all have in-built message waiting lamps, and the IP Office Phone Manager application provides message waiting indication on screen

For analog phones, from IP Office 3.1 a variety of analog message waiting indication (MWI) methods are provided. Those methods are 51V Stepped, 81V, 101V and Line Reversal. The MWI method must be selected from the IP Office Manager application when configuring a system to match the properties of the analog phones. Note that the 101V signaling is only available on version 2 IP400 Phone 8, 16 and 30 modules, not on the IP406 system unit.

Visual Voice

Feature

- Provides interface to voicemail through handset display and buttons e.g. Listen, Save, Delete, Fast Forward….

Benefit

- Quick access to voicemails and commonly used messaging features.

Description

With IP Office R4.0, you can now access and control voice messages via the display on Digital or IP phones. Visual Voice requires Voicemail Pro or Embedded Messaging, and can be used with large display LCD sets only (2410, 2420, 5410, 5420, 4610, 4620, 4621, 4625, 5610, 5620, and 5621 sets are supported)

Features supported are:

- access new/old/saved messages for personal and hunt group mailboxes.
- next and previous message.
- fast forward and rewind.
- pause message.
- save, delete and copy message to other users of the system.
- change default greeting.
- change password.
- change email settings (Voicemail Pro only).

Note: Visual Voice NOT available on Voicemail Lite and not supported on T3 sets.

Advanced Call Handling

Advanced Call Handling

Description

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

Absence Text

Feature

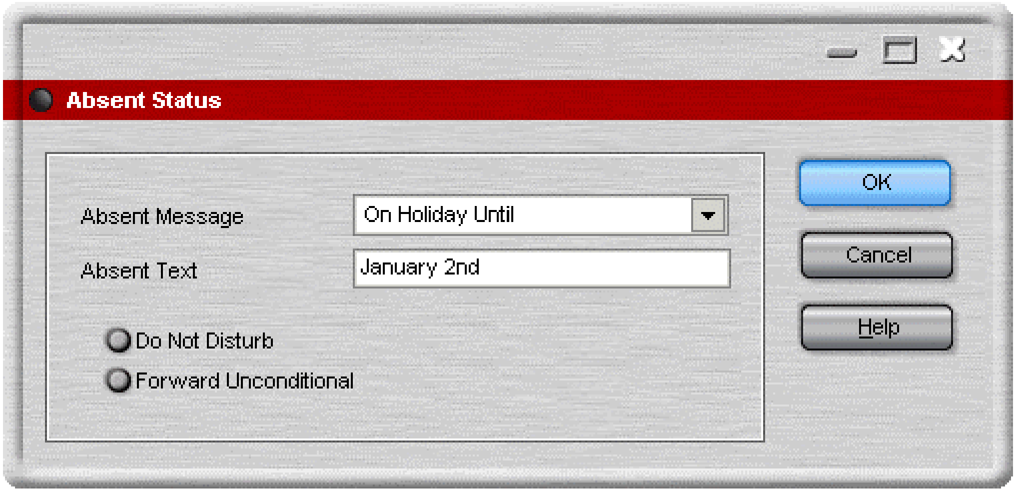
- Display a text message on the user's phone and IP Office Phone Manager application.
- Display the same message on other internal phones and IP Office applications when calling the user.

Benefit

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

Description

Any user can set Absence Text on their phone, even users of standard analog phones, but it can only be displayed on selected display phones, Phone Manager and SoftConsole that call the user. Most supported feature phones give the option of adding some text, for example, “At lunch until 16:00”.



When a user has an absence text message set, call processing is not affected to the user and they still have the choice of using features like Do Not Disturb or Forward on No Answer as appropriate. Phones that support the interactive setting of Absence Text will also display it on the users own phone for the benefit of people who come to their desk. There are 10 predefined strings for Absence Text:

- None (no text message)
- "On vacation until"
- "Will be back"
- "At lunch until"
- "Meeting until"
- "Please call"
- "Don't disturb until"
- "With visitors until"
- "With cust. til"
- "Back soon"
- "Back tomorrow"
- Custom

All may have additional text entered, eg message 4 plus 10:00 will show "Meeting until 10:00" and the text strings are localized to the system language

This feature is supported across the IP Office Small Community Network

Call Tagging

Feature

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

Benefit

- Provide additional information about the call.

Description

This feature is used to provide additional information about the call to the targeted user before they answer it. Call Tagging may be used when transferring a call from Phone Manager or Soft Console to give caller info if the user doing the transfer is not able to announce the call.

It is possible to add a tag to a call automatically using CTI and IP Office Voicemail Pro. This is also possible based on an Incoming Call Route. On some telephones, displaying the Tag may mean that it is not possible to display the usual call source and target information.

Reclaim Call

Feature

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

Benefit

- If you just miss a call and it goes to voicemail or call coverage, you can get the call back while it is still being presented or connected through IP Office

Description

This is a special version of the Acquire Call feature that only applies to the last call at your extension.

Hunt Group Enable/Disable

Feature

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

Benefit

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

Description

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

Call Waiting

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

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

Do Not Disturb (DND)

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

It is possible to have some calls bypass the DND setting and ring the phone. For example a manager might have their secretary's extension number on the DND exceptions list. The exceptions list can be easily managed by the Phone Manager application. Both internal and external numbers can be on the exception list.

Dial Plan

IP Office has a very flexible numbering scheme for extensions, hunt groups and feature commands. While the system has default numbering for feature codes and extensions, they can all be re-defined. Default extensions and hunt groups have 3 digit numbers starting at 200 but these can be changed from 2 to 9 digits through the IP Office Manager application. There is a default set of feature access “short codes,” but these can be changed to what ever the end user requires, within limits. This is useful for example, if IP Office is replacing a system where DND was accessed by dialling *21, it is possible to change the IP Office Short Code to mimic the code of the replaced system.

In certain countries IP Office can support a Secondary Dial Tone when an access digit is dialled, though this limits some functionality like Alternate Route Selection (ARS). IP Office can also be configured to work without line access digits, by analyzing digits as they are dialled and determining if they are for an internal number or should be sent out on a line – this is valuable in SOHO installations where users will not necessarily be used to dialling an access digit for an outside line.

Paging

All Avaya digital and IP phones supported on the IP Office that have loudspeakers can be used to receive broadcast audio messages without having to install a separate paging system. Paging can be to individual phones or groups of phones.

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

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

This feature is supported across the IP Office Small Community Network

Intrude

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

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

Inclusion

This feature enables selected users 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 user, 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.

Private Call

Users can set a status of private call using short codes or a programmed button. Private calls cannot be recorded, intruded on, bridged into or monitored.

Hot Desking

Hot Desking allows a number of users non-exclusive use the same extension. Each user logs in with their own identity so they can receive calls and can access their own Voicemail and other facilities. For example, sales personnel who visit the office infrequently can be provided with telephony and Voicemail services without being permanently assigned a physical extension. When finished, they simply log off to make the extension available to others or if users log on at another phone, they are automatically logged off the original extension.

Remote Hot Desking

Feature

- The ability for a user to Hot Desk to other locations within the Small Community Network.
- Available on Digital, Analog and IP phones.

Benefit

- A user can make and receive calls from any office as if using the phone on their own desk.
- Single number, improved mobility and easy access to familiar features.
- Great for consultants, managers, lawyers working on different offices on different days.

Description

IP Office 4.0 supports remote hot desking between IP Office systems within a Small Community Network. The system on which the user configured is termed their 'home' IP Office, all other systems are 'remote' IP Offices. To log on at a remote IP Office requires that IP Office to have a Small Community Advanced Networking license. A license is not necessary on the user's home IP Office.

- **User Settings**
  When a user logs on to a remote IP Office system, all their user settings are transferred to that system.
  - The user's incoming calls are rerouted across the SCN.
  - The user's outgoing calls use the settings of the remote IP Office.
  - However some settings may become unusable or may operate differently. For example if the user uses a time profile for some features, those feature will only work if a time profile of the same name also exists on the remote IP Office.
- **Break Out Dialing**
  In some scenarios a hot desking user logged on at a remote system will want to dial a number using the system short codes of another system. This can be done using either short codes with the Break Out feature or a programmable button set to Break Out. This feature can be used by any user within the Small Community Advanced Network but is of significant use to remote hot deskers.

Note: Remote Hot Desking is not supported for use with CBC and CCC. Features handled by the telephone itself are not affected by Hot Desking (e.g. call log and phone speed dials).

Relay On/Off/Pulse

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

Pickup

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

- Pick up any call ringing on another extension.
- Pick up a Hunt Group call ringing on another extension, where 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.

This feature is supported across the IP Office Small Community Network

Call Recording

Where IP Office has Voicemail Pro installed it is possible to record a call and save the recording to the user's mailbox, a group mailbox or the voice recording library. For example, this is useful when a caller is going to give detailed information like an address or phone number and the caller will hear a warning message or tone that the call is being recorded in some countries. Where call recording is required for Quality Assurance, it is possible to set the IP Office system to automatically record a percentage of calls for later review.

Beginning with IP Office R4.0, any call (normal, conference, or intrusion) and any phone type (including IP) can be recorded. Where "advice of recording" needs to be played, IP Office will ignore Voicemail port licensing if an insufficient number of voicemail channels have been licensed.

Note: for IP phones, a VCM channel will be required for the duration of the recording.

Telecommuter Mode

Phone Manager Pro allows the making and receiving of calls and the retrieving of voicemails from an external phone number as if they were in the office, with Phone Manager providing the call control. The typical scenario is the remote worker that occasionally works from home or from a hotel room.

This feature also provides billing convenience and potential cost savings for remote workers and mobile work force as all the calls are established by IP Office: there is no need to check bills, nor to pay for expensive hotel calls.

Twinning

Twinning allows a primary extension and a secondary number (extension or external) to operate together as a single telephone, typically used in scenarios like workshops or warehouses where team supervisors may have a desk with a fixed phone but also have a Mobile/Cell phone. When a call is presented to the primary phone the secondary will ring. If the primary telephone does not ring, for example in Do Not Disturb, the secondary phone will not ring. When a call is made from either twinned phone, the call will appear to have come from the primary phone (when the secondary is an extension on the IP Office system). Other users of the system need not know that the supervisor has two different phones. The supervisor's Coverage Timer and No Answer Time are started for the call and if the call is not answered within that time, the call will be delivered to available coverage buttons (if applicable) and then Voicemail (if applicable).

Users may be allowed to enter a twinned number, or may just be able to activate/deactivate the twinning function depending on administrative settings.

The following types of calls are eligible for twinning:

Call Type	Internal Twinning	External (Mobile) Twinning
Any internal call on a Call Appearance button	✓	✓
Internal or external calls transferred to the extension	✓	✓
Direct Dial calls to that extension	✓	✓
Hunt Group Calls	✓	✓
Calls forwarded from another extension	✓	✓
Line Appearance calls (configurable)	✓	✗
Bridged Appearance calls (configurable)	✓	✗
Coverage calls (configurable)	✓	✗
Automatic Intercom calls	✓	✗
Returning transferred, held or park calls	✓	✗
Callback calls from the system (Transfer and Park Return)	✓	✗
Paging Calls	✓	✗
Follow Me calls	✓	✗

Key and Lamp Operation

Key and Lamp Operation

IP Office offers a full range of Key and Lamp features on Avaya feature phones. These features include; Line Appearance, Call Appearance, Bridged Appearance and Call Coverage. As these features require a phone with buttons and indicators, the features are only supported on certain Avaya digital and IP phones. Key and Lamp operation is not supported on analog phones.

IP Office can have a ring delay set on each appearance button to allow time for the target number to answer before other extensions ring, or visual alert only without ring.

In Key and Lamp operation, IP Office supports up to 10 buttons on each telephone and 10 telephones with the same line appearance.

Appearance Buttons

Feature

- Use the programmable buttons available on Avaya digital and IP telephones to represent individual calls.
- Answer, originate and join calls by pressing the appropriate appearance buttons.

Benefits

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

Description

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

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

Line Appearance

A Line Appearance is a representation of a trunk line on the IP Office system where the indicator tracks the activity on the Line. Only external calls can be answered or made on Line Appearances. Line appearances can be used with Analog, E1 PRI, T1 PRI and BRI trunks PSTN trunks. They cannot be used with E1R2, QSIG and IP trunks.

Call Appearance Buttons

Feature

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

Benefit

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

Description

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

Note that the use of call appearance buttons overrides IP Office call waiting features. It is only when all call appearances are in use that subsequent callers receive either busy tone, voicemail or follow a forward on busy action

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

Bridged Appearance Buttons

Feature

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

Benefit

- Answer and make calls on behalf of the other user.
- Audible indication of calls presented to the bridged user, where programmed
- Visual indication of when the other user has calls presented, held or connected.
- Join and exchange calls using the paired call appearance and bridged appearance buttons.

Description

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

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

Bridged appearance buttons allow paired 'manager/secretary' style operation between two users, and are only supported for users who have call appearance buttons.

Call Coverage

Feature

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

Benefit

- Provide users the opportunity to answer colleague’s unanswered calls before they go to voicemail.

Description

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

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

Outbound Call Handling

Outbound Call Handling Features

Every business needs to make calls, but depending on the type of business these calls may need to be treated in a special way, such as recorded against a project or client through the use of Account Codes. A business may have several sites linked via a private network but certain users, like customer services agents, may need to be able to call colleagues in other offices even when the network is busy, while other users can wait for a line to come free, Least Cost Routes can automatically translate the internal number to a direct dial call over the public network while other users wait.

Account Codes

Feature

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

Benefit

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

Description

IP Office stores a list of valid account code numbers. When making a call or during the call, the user can enter the account code they want associated with that call. IP Office will check the account code against its list of valid codes and request the user to re-enter the code if it is not valid. For incoming calls, the Caller ID can be used to match it with an account code from the IP Office's list of valid codes and report the account code with the call for billing.

Individual users can be set to Forced Account Code operation where they are required to enter a valid account code before making external calls. By using IP Office Short Codes it is possible to identify certain numbers or call types as requiring a valid account code before permitting the call to proceed, for example long distance or international numbers. Analog phone users can only enter account codes before making a call or in response to an audible system prompt to enter a code when making the call.

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

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

Authorization Codes

Authorization codes allow an IP Office user to go to another extension on the system and make calls using their personal toll restrictions; this may grant the user greater or fewer privileges than the normal owner of the extension they use. Since Authorization Codes are independent of Account Codes, the user has to enter both if the required by the system configuration. All entered codes are logged in CDRs.

Dial Emergency

Dial emergency is an IP Office Short Code and, permits certain numbers to be dialed regardless of call barring or a phone being logged off.

Call Barring

Feature

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

Benefit

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

Description

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

For users, the short codes can be allocated to a User Rights template. This template is then applied to the Users whose calls need restriction. In addition to barring the dialling of certain numbers, IP Office can be set to bar the forwarding of calls to external numbers on a per user basis.

Alternate Route Selection (ARS)

IP Office supports Alternate Route Selection, which is more flexible and easier to configure than Least Cost Routing (LCR). If a primary trunk is unavailable, then ARS provides automatic fallback to an available trunk (e.g., analog trunk fallback if a T1 or SIP trunk fails, or use PSTN for SCN fallback).

By configuring ARS, calls may be routed via the optimum carrier. Time profiles can also be used to allow customers to take advantage of cheaper rates or better quality at specific times of day.

Multiple carriers are supported. For example, local calls are to go through one carrier between specific hours and international calls through an alternative carrier. Carrier selection using 2-stage call set up via in-band DTMF is possible. It is possible to assign specific routes on a per user basis, e.g. only allow expensive routes to be used by critical staff.

Note: Existing LCR configurations are automatically converted to ARS when upgrading to 4.0

Maximum Call Length

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

PIN Restricted Calling

See **Account Codes**.

Forwarding

Forwarding

This is the ability to forward a user's calls to another extension or external number such as a Mobile/Cell Phone. Calls can be forwarded in a number of ways and if the call is not answered at the forward destination it will go to IP Office voicemail if enabled for the user and call supervision is available. There are three separate forward destinations, one for forwarding on busy one for no answer and one for forward unconditional. Once the numbers have been entered, the user can toggle the forwarding to be active or not as required without having to re-enter the numbers.

If the user is a member of a hunt group, some types of Hunt Group calls can also follow forward unconditional. Users can select if forwarding is applied to external calls only, or all calls. Call forwarding is processed after Do Not Disturb and Follow-Me conditions are tested.

Associated Features	Precedence
• Do Not Disturb (DND)	• Forward Unconditional
• Voice Mail (VM)	• Forward Busy
• Follow Me	• Forward No Answer
• Hunt Groups	
• No Answer Interval	

Forward on Busy

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

Forward on No Answer

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

Forward Unconditional

This sends all calls for the user to the forward unconditional number, but if the call is not answered within a user's timeout period the call will be sent to IP Office voicemail, if enabled.

Forward Hunt Group

Calls for a hunt group that the user belongs to can also follow forward unconditional. The hunt group must be set for either hunt or rotary ring type and if the call is not answered at the forward destination it will follow the hunt group call handling instead of going to voicemail. This can be particularly useful in a sales or support environments where a number of people may be out of the office on Mobile/Cell Phones and still participate in the hunt group as if in the office.

Follow Me

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

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

Avaya Digital and IP Phones

Programmable Buttons

As well as the usual dialing keys, Avaya digital and IP phones have dedicated function buttons like Mute, Volume, Hold, Conference and Transfer. In addition to these, on many Avaya digital and IP phones there are keys that can be programmed with a range of selected special functions. These keys can be used for calling other extensions on the system (Direct Station Select or DSS keys), or can be used for options from speed dialing numbers to controlling features such as Do Not Disturb. Many features use an indicator to show whether a feature is enabled. Button programming is done through the IP Office Manager application as part of the system configuration, although some phones allow the user to program buttons and functions where given administration rights.

Busy Lamp Field (BLF) Indicators

Feature

- Status indicators which show the status of a programmable buttons associated feature or function.

Benefit

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

Description

Avaya digital and IP phones have programmable buttons which can be assigned to various features. When those buttons include some form of BLF indicator, the button can also be used to indicate when the feature is active. For example, a button associated with another user will indicate when that user is active on a call. A button associated with a group will indicate when the group has calls waiting to be answered.

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

Phone Manager and SoftConsole show these conditions:

-  **Busy**
-  **Message**
-  **Forward All**
-  **Do Not Disturb**

This feature is supported across the IP Office Small Community Network.

Call History

Feature

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

Description

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

The IP Office Phone Manager application maintains a call history record of the users last 100 calls. The application must be running to record call history. Phone Manager Lite can display call history for all calls and missed calls only. Phone Manager Pro can display call histories for all calls, missed calls, inbound calls and outbound calls. Entries in the call history can be used for return calls, sorted and added to the Phone Managers local directory or speed dials.

Language

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

Directory

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

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

Self-Administration

The IP Office administrator may give select users the ability to change some of the phone settings themselves. For example, button programming. The range of changes that the user can make depends on the phone type in use.

On Hook Dialing

Avaya digital and IP phones allow the user to make calls by just dialing the number on the keypad, without having to lift the handset or pressing a speaker button. Usually the call progress can be monitored using the speaker in the phone, on phones that support hands free the whole conversation can be had without having to lift the handset.

Inbound Call Handling

Inbound Call Handling

IP Office offers several features to provide versatile inbound call processing, including PC based applications, and a standards-based TAPI interface for 3rd party applications.

Incoming Call Routing

Incoming calls can to be presented to an Operator who then decides where to pass the call, but IP Office supports intelligent call routing capable of making routing decisions based on a number of criteria.

The system currently supports routing based on;

- Call presentation digits from the exchange such as DDI/DID or ISDN MSN.
- Calling telephone number or Caller ID (This could even be part of the number received such as an area code).
- ISDN sub-address.
- ISDN/PRI 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, as well as calendar-based routing for specific dates.

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, but note that calls already ringing a free extension are not considered queuing and are not affected by a high priority call joining a queue.

An optional tag can be added to calls on the Incoming Call Route, which can be displayed on the alerting telephone.

Hunt Groups

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

Four modes of call presentation are supported on IP Office;

- **Sequential**
  One extension at a time sequentially always starting at the top of the list.
- **Collective**
  All extensions in the Hunt Group simultaneously.
- **Rotary**
  Start with extension next in list to extension that was answered the last Hunt Group call.
- **Longest Waiting**
  Start with extension that has been free for the longest time.

If all extensions in the Hunt Group are busy or not answered, another Hunt Group, called an Overflow Group, can be used to take the calls. An overflow time can be set to stipulate how long a call will queue before being passed to the Overflow Group. The system can change the status of users who do not answer a hunt group call presented to them. The user can be put into busy wrap-up, busy not available or logged off. The change of status can be set per user and the use of this option can be set per hunt 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.

Night service fallback using a time profile is no longer applied to a hunt group already set to Out of Service.

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

Voicemail can also be used in conjunction with Hunt Groups to take all group related messages, play an announcement when the Hunt Group is in Night Service or Out of Service mode and give announcements while a call is held in a queue. For internal voicemail use a broadcast option is provided. This feature will alter the voicemail box operation so that the message notification will only be turned off for each hunt group member when they retrieve their own copy of the message.

Small Community Networking (SCN) Distributed Hunt Groups

Small Community Networking (SCN) Distributed Hunt Groups

Hunt groups in a Small Community Network can include members located on other systems within the network. This feature requires entry of an Advanced Small Community Networking license in each system in the network.

Note: Distributed Hunt Groups are not supported for use with CBC and CCC.

Night Service

When a Hunt Group is in Night Service mode the Hunt Group is temporarily disabled. Callers to this Hunt Group will:

- Pass to a Night Service Fallback group used to provide cover, e.g. pass calls to a manned extension or an external number
- Be played the Out of Hours greeting if Voicemail is operational
- Receive the busy tone

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

Time Profiles

Time Profiles can be used to define when a Service, Hunt Group, Least Cost Route, Conference Bridge 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 of day to take advantage of cheaper call rates. Multiple Time Entries can be created so that a Time Profile can be used to define 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. From Release 4.1, Time Profiles can also be based on specific calendar dates to make allowance for public holidays or other events.

Queuing

Queuing allows calls to a Hunt Group to be held in a queue when all extensions in the group extension List are busy. When an extension becomes free the queued call is then presented. The definition of queued calls now includes ringing calls and calls waiting to be presented for ringing. The queue limit can be set to control the maximum number of calls to wait against a hunt group.

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

Announcements

With IP Office 4.0, Hunt group announcements are separated from hunt group queuing and can be used even when queuing is off. Hunt group announcements are now supported by Embedded Voicemail in addition to Voicemail Pro and Voicemail Lite.

Further, times for the first announcement, second announcement, and between repeated announcements are configurable.

Contact Center Features

Contact Center Features

Contact Centers have specific needs for reporting on how calls are handled and these are covered in a separate section of the Product Description. Basic handling of telephony requirements for a Call Center is a standard part of IP Office from Automatic Call Distribution (ACD), Call Queuing to agents logging on and selecting the groups that they service.

Login

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

Monitor Calls

A user can monitor other peoples' calls by listening in. This feature is not available by default; it must be specifically enabled in the system configuration. An option exists to have a beep tone indicate when monitoring is in use. The user is only able to listen; they cannot speak into the conversation being monitored.

Note that all phone types can be used to monitor, however calls to and from IP phones cannot be monitored.

Acquire Call

Feature

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

Benefit

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

Description

The Acquire Call function can be setup as a special short code or programmed against a button on an Avaya digital or IP phone with programmable buttons. Use of the feature is subject to IP Office intrusion control settings, the user acquiring the call must be set to be able to intrude and the user whose call is being acquired must be set to can be intruded. Acquire call works in two ways, invoked with or without a number:

Without a value in the number field

- This allows a user to reclaim a call that was ringing on their phone but has now gone elsewhere, for example to Voicemail or Forward No Answer destination. The Intrude settings are not checked and the call can be reclaimed even if it has been answered.
- If the last call to ring this User is no longer ringing or connected on the system, the feature will fail.

With a number, where the number is the telephone number of a user who currently has the call to be acquired.

- If the user has a call ringing or waiting Acquire Call will act like the Call PickUp Extension short code and the user executing Acquire Call will be connected to the oldest ringing/waiting call.
- If the User has a connected call with no call waiting and the Intrude settings of the two Users allow it, the call will be connected to the user executing the Acquire Call and the other user will be disconnected.
- If the User does not have a call the feature will fail.

Queue Threshold Alert

When the number of calls queued against a Hunt Group exceed a threshold, the system can be configured to alert at a selected analog extension port. Typically the User to Alert will be a loud ringer or other alerting device. The alert does not present a real call and if answered the phone presents dial tone.

Miscellaneous Features

Conference Calls

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

The IP Office - Small Office Edition supports 24 conference parties with a maximum of 6 parties in any single conference.

The IP406 and IP500 can support multiple conference calls totaling up to 64 parties. For example one conference of 64 calls or 21 conferences of 3 calls each.

The IP412 has two 64-party conference bridges giving any combination from 2 x 64-party conferences to 42 x 3-party capacity.

Only two calls connecting through analog trunks are permitted in any single conference.

For more information on managing conference calls, refer to Chapter 12 where IP Office Conferencing Center is described

Dial On Pickup

Also known as "Hotline". Automatically dials a specified extension when the phone is taken off hook. This facility is commonly used in unmanned reception areas or for door entry systems to allow visitors to easily gain assistance.

Off Hook Operation

Off-Hook Station is designed for users who want their analog phone to operate like digital or IP feature phone, to isolate the user's phone idle state from the Hook state. This is a useful feature when using Phone Manager or SoftConsole to control the phone state when using a headset on an analog telephone and with call control and dialing from Phone Manager or SoftConsole.

External Control Port

The IP Office system unit has two electronic switches, similar to relays, which can be normally open, normally closed, pulsed open or pulsed closed and activated by dialing a short code or through Phone Manager, SoftConsole or Voicemail Pro action.

These switches can be used for several purposes, for example as a means to control an electronic door release. 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.

E911

This is a specific service for North America. When an emergency call is connected, IP Office 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 is clearly identified.

System Short Codes

System Short Codes

Short Codes are used as commands the IP Office to make changes for the user, group or system, so need to set up with consideration to security. The command may need additional information included with it, such as for forward, the phone number forwarded to. Short codes are a flexible and quick way of setting up certain features. IP Office has short codes provided by default on the system, or more advanced codes that need programming by the system administrator.

The full set of short code commands are listed below; please see product configuration documents for more detail on how to set them up.

AOC Previous Call
AOC Reset Total
AOC Total
Auto Attendant
Break Out
Busy
Busy On Held
Call Intrude
Call List
Call Listen
Call Pickup Any
Call Pickup Extn
Call Pickup Line
Call Pickup Group
Call Pickup Members
Call Pickup User
Call Queue
Call Record
Call Steal
Call Waiting On
Call Waiting Off
Call Waiting Suspend
Cancel All Forwarding
Cancel Ring Back When Free
Channel Monitor
Clear Call
Clear CW
Clear Hunt Group Night Service
Clear Hunt Group Out Of Service
Clear Quota
Conference Add
Conference Meet Me
CW
Dial
Dial 3K1
Dial 56K
Dial 64K
Dial CW
Dial Direct
Dial Direct Hot Line
Dial Emergency
Dial Extn
Dial Inclusion
Dial Paging
DialPhysicalExtensionByNumber
DialPhysicalNumberByID
Dial Speech
Dial V110
Dial V120
Dial Video
Disable ARS Form
Disable Internal Forwards
Disable Internal Forward Unconditional
Disable Internal Forward Busy or No Answer
Display Msg
Do Not Disturb Exception Add
Do Not Disturb Exception Delete
Do Not Disturb On
Do Not Disturb Off
Enable ARS Form
Enable Internal Forwards
Enable Internal Forward Unconditional
Enable Internal Forward Busy or No Answer
Extn Login
Extn Logout
Flash Hook
Follow Me Here
Follow Me Here Cancel
Follow Me To
Forward Hunt Group Calls On
Forward Hunt Group Calls Off
Forward Number
Forward On Busy Number
Forward On Busy On
Forward On Busy Off
Forward On No Answer On
Forward On No Answer Off
Forward Unconditional On
Forward Unconditional Off
Group Listen Off
Group Listen On
Headset Toggle
Hold Call
Hold CW
Hold Music
Hunt Group Disable
Hunt Group Enable
Last Number Redial
MCID Activate
Mobile Twinned Call Pickup
Off Hook Station
Park Call
Private Call
Private Call Off
Private Call On
Priority Call
Record Message
Relay On
Relay Off
Relay Pulse
Resume Call
Retrieve Call
Ring Back When Free
Secondary Dial Tone
Set Absent Text
Set Account Code
Set Authorization Code
Set Hunt Group Night Service
Set Hunt Group Out Of Service
Set Inside Call Seq
Set No Answer Time
Set Mobile Twinning Number
Set Mobile Twinning On
Set Mobile Twinning Off
Set Outside Call Seq
Set Ringback Seq
Set Wrap Up Time
Shut Down Embedded Voicemail
Suspend Call
Suspend CW
Toggle Calls
Unpark Call
Voicemail Collect
Voicemail Node
Voicemail On
Voicemail Off
Voicemail Ringback On
Voicemail Ringback Off

5. IP Telephony

Introduction to IP Telephony

Technological innovation is changing the way we communicate. This time it is coming in the form of changing the way telephone calls are transmitted. It brings with it several new capabilities that change the meaning of the phrase telephone call through the use of Voice over Internet Protocol (VoIP). Basically, VoIP means "voice transmitted over a packet data network." VoIP is often referred to as IP Telephony because it uses the IP protocols to make possible enhanced voice communications throughout the world, wherever IP connections have been delivered. IP Telephony unites a company's many locations—including mobile workers— into a single converged communications network. Telephony calls using VoIP go above and beyond what's been possible in the past. When it comes to placing telephone calls, VoIP provides a range of support services and features unequalled in the world of telephony, but above all deliver them at low cost.

How Does VoIP Work?

Voice over Internet Protocol means basically what the acronym states: Voice travels over an Internet Protocol. Internet Protocol refers to the type of rules that the network uses to send and receive signals. IP Telephony works by converting voice communications into data packets. Conveniently, it runs on the popular Ethernet LAN (local area network) technology, which currently supports over 96 percent of the worlds companies' LANs.

Circuit-switched or Time-Division Multiplexed Telephony

Before digital networking with the Internet took off, everyone had to use the "Plain Old Telephone Services" (POTS). These run over a network called the Public Switched Telephone Network (PSTN). The PSTN has been around since the telephone was invented in either analog or digital form using circuit switched technology where the telephone call gets exclusive bi-directional use of a wire – or circuit – while the call is in progress. Because the circuit is exclusive to each conversation, PSTN and private branch exchanges (PBXs) must be sized to cope with peak demand and have enough circuits available for all expected conversations. This is not a flexible approach and results in a lot of infrastructure investment that the telephone companies need to recoup, via the cost of access charges and calls. The Internet has changed this – where data services have driven down access charges and allowed voice to "travel for free" over a multipurpose data network.

Packet-Switched Telephony

Unlike circuit-switched connections, which always require use of dedicated bi-directional circuit for the duration of a call, VoIP technology has enabled telephony and other new and novel features and services to run over fixed and wireless networks including private local area networks. These newer network types use packet-switched protocols. Packet-switched VoIP puts voice signals into packets. Along with the voice signals, VoIP packets include both the sender's and receiver's network addresses. VoIP packets can traverse any VoIP-compatible network. Along the way, they can choose alternate, shared paths because the destination address is included in the packet. The routing of the packets is not dependent on any particular network route which means the network provides can provide a reliable service at a fraction of the cost of circuit switched providers.

What Advantage Does IP Office Have?

IP Office can provide support of PSTN, POTs, digital time division multiplexed phones AND digital IP phones all on the same system. This means you don't have to abandon the past to embrace the future, IP Office allows all the technologies to co-exist. IP Office connects to the PSTN and to IP trunks (the VoIP equivalent) so providing a "Hybrid" PBX function – where both legacy and future technologies can be used together to minimize operating costs and offer optimize business communications through both voice and data.

IP Office has digital telephones built on both TDM and IP technology that provide the same user interface offering a flexible choice of solution that can mix, for example TDM phones in the office and IP phones at a remote site of at home. With the choice of IP phones including real and virtual (software) phones, IP Office can take communications to a new level.

Buying IP Office allows you choice – you can use the pure POTs or the pure VoIP capabilities of IP Office, or use both at the same time to allow seamless technology transition of your business without the disruption of having to choose between them now.

IP Office Turns VoIP into IP Telephony

In order to make use of VoIP, IP Office uses signaling protocols called H.323 right now, and Session Initiation Protocol (SIP) which allow IP Office to establish end-to-end connections for the voice path through the IP network. It ensures each end of the connection is able to transmit and receive voice and provides the network addressing for end to end packet transmission. IP Office also allows for connecting between the different technologies by translating the signals they use, for example an analog phone may wish to connect to a VoIP destination. This requires both the signaling and voice transmission to be translated – IP Office does this easily as it contains technology elements called gateways and gatekeepers that enable translations to happen.

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

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

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

When making use of IP telephony, there are a number of data centric considerations such as which data types have priority on the IP network when there is contention. This is set with IP/TCP “quality of service” and should not be ignored. In situations where LAN Bandwidth is limited, a quality of service capable LAN switch should be used to ensure voice packets are transmitted with the required priority on the network. If not, the conversation carried over IP appears as broken up (due to packet loss) or has unacceptable delays introduced in the conversation (latency & jitter). With IP hardphones there is need for Power over Ethernet (PoE) or “midspan power” to be provided to the phones as the IP phones are no longer powered by IP Office – a list of Avaya approved PoE options is available at the end of this section.

Gateways, Gatekeepers and H.323 - Technology Overview

IP Office uses the H.323 signaling protocol which has the following architectural components

- Telephones are H.323 service endpoint devices that can support Audio calls. Other types of H.323 devices can support video as part of H.323
- Gateways provide media translation to allow calls to be made to non-H.323 devices, for instance an analog telephone or the public network to connect with a H.323 device
- Gatekeepers control the call processing and security for H.323 devices
- Multipoint Connection Units (MCU) for conferences by adding together media streams

These elements 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, IP telephones, Gateways and MCU make registration requests to a Gatekeeper which then authenticates (accepts or rejects) their request to become a member of the zone. Once accepted, a telephone wishing to make a call sends a call set-up message to the Gatekeeper which then determines how to route the call and will then send an alert to the called telephone, or if the call is to a non-H.323 telephone establish the call via a Gateway within the zone.

The design of IP Telephony systems has been driven by open standards. Digital IP Phones, Gateways and Gatekeepers all support the H.323 standard and it is this that allows devices from different manufacturers to work together. IP Office has an optional integral Gateway (Voice Compression Modules) and Gatekeeper functionality required to provide a fully functional IP Telephony solution.

IP Telephony Features

- **Gatekeeper**
  The IP Office gatekeeper allows the registration of up to 16 IP extensions on the Small Office Edition, 190 IP extensions on the IP406, 360 IP extensions on the IP412 and 272 IP extensions on the IP500, less the number of analog and digital TDM telephones already configured on the system.
- **Gateway**
  The Voice Compression Module provides the H.323 gateway function that allows IP extensions to make calls to other non-IP devices. The maximum number of simultaneous calls is limited by the number of channels available on the Voice Compression Module. IP Office must be fitted with an optional Voice Compression Module to enable IP telephony.
- **Silence Suppression**
  Silence suppression is a technique used to make the best use of available bandwidth, such as the connection over which the caller is listening, not speaking. Silence suppression works by sending descriptions of the background noise, rather than the actual noise itself, during gaps in conversation thereby reducing the number and frequency of voice packets sent on the network. 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.729a and G.723.1. 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 fast start is supported by an IP extension, this facility reduces the protocol overhead allowing an audio path to be established more quickly.
- **Out of Band DTMF**
  When out of Band DTMF is configured on an IP extension, the extension will signal to the other end of the connection which digits need to be regenerated by a local DTMF generator on behalf of the sending 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 up voice compression resources after establishing the end to end connection, allowing the resources to be used in the most efficient way.
- **Auto-Create Extensions**
  IP Office can automatically create an extension entry for new IP phones added onto the local area network. In cases where the local area network is not secure this facility can be disabled, but simplifies installation of IP telephone systems
- **Fax Transport**
  Fax Transport allows fax calls to be routed over VoIP trunks between IP Office systems on an IP network using a proprietary IP Office transport protocol. This is different from the T.38 protocol which is not supported.

LAN Switch Support

Avaya recommend the use of Extreme Alpine Series switches for IP telephony applications. For more information, contact Extreme Networks.

Power Options for IP Telephones

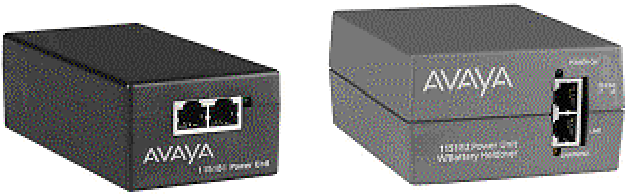
Avaya supports the IEEE 802.3af, standard for Power over Ethernet (PoE) on its range of IP telephones. With Power over Ethernet, both power and data are carried over one CAT 5 Ethernet cable. Deploying IP telephones utilizing Power over Ethernet eliminates the need for local power supplies, AC adapters and cables, and allows power to be provided from the wiring closet/switch room where it can be easily connected to a UPS system.

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

Avaya Individual Power Supply

Avaya provides individual power supplies that can be used to power each IP phone which provides a single 48 volt output. The power supply can operate globally within a wide range of Alternating Current (AC) input voltages: 90 - 264 Volts Alternating Current (VAC), 47-63 Hz. This power supply has a green indicator (LED) that shows the unit has power to the PHONE socket on pins 7&8 of the CAT5 cable.

This item is available in two different versions, with and without an internal battery for uninterrupted power to the phone.



1151 local power supply, without battery and with battery backup

Avaya Mid-Span Power Distribution Units

These power devices are designed for IP-telephony and provide power over Ethernet (PoE) for up to 24 IP telephones or wireless LAN (WLAN) access points in one unit. The Mid Span Power units are designed to mount in a 19-inch rack with the data equipment or they can be stacked up to four units high using the optional rubber feet. The mid-span is 1U in height (1.75 inches) and has up to twenty-four RJ45 sockets on the bottom row and twenty-four data and power output RJ45 sockets on the top row. The units provide a maximum of 200 Watts or a peak of 16.8 watts per port. Data is unaffected by power delivery, if the device does not require power. The mid-span power units are also referred to as PDU (Powered Data Unit) devices. Power over the LAN will simplify the installation and support of IP telephones for our customers and are available in 3 sizes; 6, 12 or 24 ports with optional SNMP management capability.

Mid-Span power supply

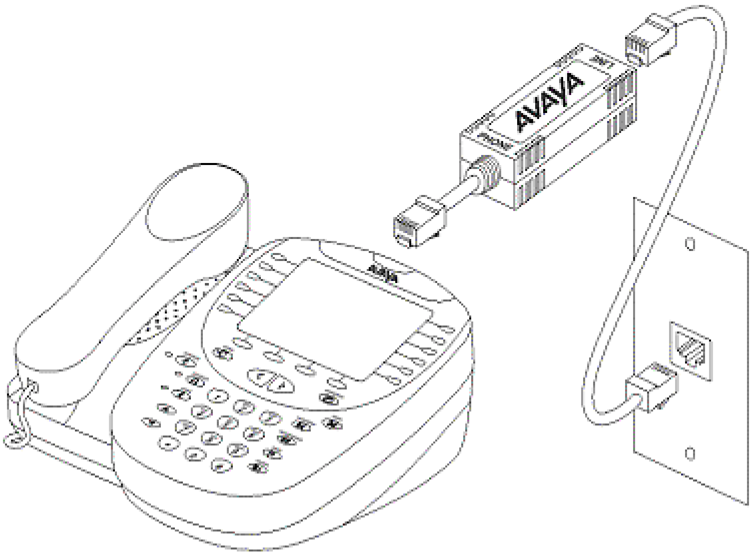
Avaya IP Phone Power Adapter

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

The Avaya IP Phone Power Adapter is supported on the following:

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

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



Avaya IP Phone Adapter

IP Telephone Power Consumption

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

Phone	Typical	Worst Case	IEEE 802.3af
4601, 4602, 5601, 5602	3.5W	4.6W	Class 2
4602SW, 5602SW	4.1W	5.0W	Class 2
4610SW, 5610	5.0W	6.4W	Class 0
4620, 5620	4.0W	6.0W	Class 2
4620SW	7.7W	9.9W	Class 3
4621SW	5.9W	8.0W	Class 3
4625SW	4.9W	6.45W	Class 3

Note: Typical is measured off-hook sample size 1. Worst Case is analytical. Except the 4601, 4602, 5601 and 5602 all telephones had a PC attached at 100Mbps. The EU24/EU24BL adds less than 1W to the 4620, 4620SW and 5620 numbers.

VoIP FAQ

Network Requirements

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

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

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

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

What are Voice Compression Modules (VCM's) for?

VCMs are required to support the following scenarios:

- Usage of Embedded Voicemail on the Small Office Edition (used as a memory boost by compressing the voice files)
- Internal phone calls between an IP device and a non-IP device
- Analog/digital phones to IP trunks (SIP/H.323) including managed Frame Relay and managed IP VPN (provides echo cancellation)
- IP phones to ISDN or PSTN trunks (convert IP to TDM and vice-versa)
- Call set up between IP phones (VCM resource will be released after call set up if direct media is used) to provide dial tone, busy tone etc. Direct media is a VoIP concept within the system that allows direct connection of the media steam (IP packets containing voice samples of the telephone call) between the two IP devices on the network.

VCMs are NOT required for:

- Calls between IP phones on the same system after call set-up (“Direct Media”), unless call recording is enabled

“Direct Media” is a VoIP concept that circumvents resources (TDM bus, Gateway) within the system and improves the voice quality. If two IP devices are connected on the same system, a direct LAN connection between them will be established once the call has been set up (as long as they use the same Codecs).

It is possible for an IP device to be physically located at one site while being registered at a different site. In this case, even for VoIP across the WAN the VCM would not be used, as long as the two IP devices involved in a phone call are registered on the same system.

Data Channels

A Data Channel is only required for Remote Access (RAS), Internet Access, and Voicemail connections:

- 10 PCs accessing the Internet over a single line = 1 Data Channel. If multiple lines are used (Multi-Link PPP) then as many data channels are required (e.g. 128k i.e. 2B channels requires 2 data channels)
- 10 users dialing in from home on 10 separate lines onto the LAN = 10 Data Channels
- Voicemail is an IP application on the LAN (i.e. one data channel is required for each voicemail port used)

Note: IP end-points do NOT require data channels

Bandwidth Required For Each Voice Call?

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

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

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

Acceptable Delay?

End-to-end delay should be 150 milliseconds or below.

How Many Simultaneous Calls Can I Get Down My Link?

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

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

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

What is the Maximum Number of Simultaneous VoIP Calls?

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

- The IP406 can be fitted with a single module offering up to 30 simultaneous calls.
- The IP412 is capable of supporting two modules of all types, allowing up to 60 simultaneous calls.
- The IP500 is capable of supporting two VCM 32/64 modules allowing up to 128 simultaneous calls.

Does the IP Office Support Fax over IP ?

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

Network Assessment

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

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

A network assessment should normally include:

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

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

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

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

IP Packet Flow Control

While a high-performance switch forwards data packets at full wire speed to and from its ports simultaneously, there may be times when a switch port may not be able to accept packets at the rate it is receiving them.

For example, the switch port may be receiving packets from multiple ports at the same time, or the switch port may be receiving packets from a port operating at a faster speed. For instance, the sending port might be operating at 100 Mbps, while the receiving port operates at 10 Mbps; or the sending port might operate at 1000 Mbps, while the receiving port operates at 100 or 10 Mbps. If data packets arrive for a port that is saturated with other packets, the packets may overflow the port’s buffer, resulting in dropped packets and lost data.

Flow control is a congestion-control mechanism that prevents data loss at congested ports. Flow control prevents packet loss by controlling the flow of data from the transmitting device to ensure that the receiving device can handle all of the incoming data.

IEEE 802.3 flow control is used on Avaya IP telephones operating in full-duplex mode. If the receiving device becomes congested, it sends a pause frame to the transmitting device. The pause frame instructs the transmitting device to stop sending packets for a specific period of time. The transmitting device waits the requested time before sending more data.

VoIP Standards Supported

IP Office supports the following protocols and standards:

- H.323 V2 (1998), Packet-based multimedia communications systems.
- Q.931, ISDN user-network interface layer 3 specification for basic call control.
- H.225.0 (1998), Call signaling protocols and media stream packetization for packet-based multimedia communication systems.
- H.245 (1998), Control protocol for multimedia communication.
- Session Initiation Protocol.
- 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.
    - RFC 3261 - Session Initiation Protocol (SIP).

6. Public and Private Voice Networks

Public and Private Voice Networks

With Avaya IP Office you can be networked via T1, PRI and BRI ISDN, including VoIP on the company WAN. Networking maximizes the current potential of your branch maximizes the current potential of your branch office and remote workers—while building the office and remote workers—while building the best possible foundation for your future growth. IP Office provides each location with a scalable (up to 360 users) telephony solution that supports voice networking, and offers:

- A uniform dialing plan, making it easy to call co-workers anywhere on the network and improve customer service
- Consistent user experience by sharing the same phones and messaging interface as in headquarters
- A user-defined central directory that is automatically synchronized
- Least cost routing and bandwidth on demand
- Centralized voicemail and/or the ability to network voicemail systems together

The benefits of networking:

- Operate a network of branch offices with a consistent set of communications and services across all locations; gain the efficiencies of universal functions and end-user familiarity.
- Leverage any existing investment in Avaya systems at other sites
- Centralize services (e.g. operator, voicemail) as well as management and administration to reduce costs
- Speed deployment of remote offices—respond more quickly to market demands.
- Improve inter-site communication to simplify information exchange and enhance customer service.

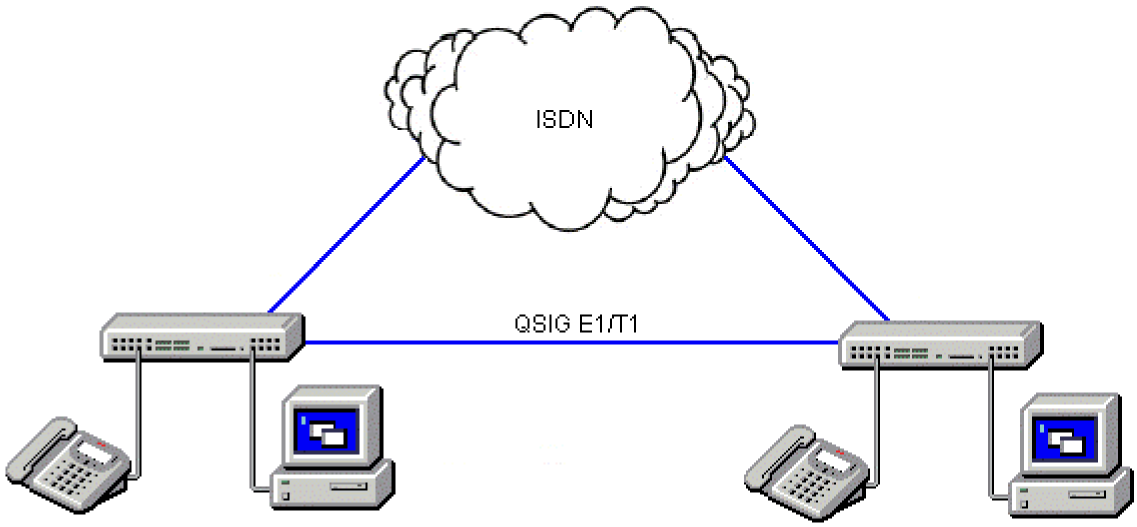
Private Circuit Switched Voice Networking

Private voice networks are built using structured leased line circuits (E1 or T1) or by establishing permanently connected 'B' channels over ISDN circuits between IP Office systems. Each channel within the E1 or T1 interface can provide a single voice or 64K/56K data call. Where leased line circuits are used within a private networking scenario, these E1 or T1 interfaces are typically configured to use QSIG signaling between sites.

QSIG provides a level of voice feature transparency between PBXs and is the favored signaling standard within multiple vendor and international voice networks. The IP Office E1 or T1 module terminates a QSIG connection with a 120 ohm RJ45 interface.

IP Office supports the following QSIG services across this network:

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



Circuit Switched Voice Networking

Public Voice Networking

The IP Office platform supports a range of trunks and signaling modes for connection to the public switched telephone network (Central Office). Some of these lines are only available in certain territories; please check with your distributor for local availability. Primary rate trunks are available with either a single (24/30 channels) or dual trunk (48/60 channels).

ISDN Primary Rate (ETSI CTR4)

Provided by the IP400 PRI E1 and IP500 Universal PRI cards.

ISDN Primary Rate provides 30 x 64K PCM speech channels over an E1 circuit and one signaling channel. 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) prevents 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)

Provided by the IP400 Quad BRI and IP500 BRI cards.

ISDN Basic rate provides 2 x 64K PCM speech channels and one signaling channel 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.

Additional ISDN features

The following ISDN features are supported by IP Office 4.0+ on both PRI and BRI trunks. Note that availability of these features is dependant on their also being supported and available from the ISDN service provider, for which there may be charges.

- **Malicious Call Identification – MCID**
  (24xx, 46xx, 54xx, 56xx, T3, T3 IP, DECT phones; Phone Manager)
  Short codes and button programming features are available so that users can trigger this activity at the ISDN exchange when required. This feature is NOT available on standard ISDN DSS1 phones.
- **Advice of Charge – AOC**
  (T3 digital and IP phones only; Phone Manager)
  Advice of charge during a call (AOC-D) and at the end of a call (AOC-E) is supported for outgoing ISDN calls other than QSIG. The call cost is displayable on T3 phones as well as Phone Manager and included in the IP Office Delta Server SMDR output for call accounting purposes. The IP Office allows configuration of call cost currency and a call cost mark-up for each user.
- **Call Completion to Busy Subscriber – CCBS**
  (2400, 4600, 5400, 5600, T3, T3 IP, DECT phones; Phone Manager)
  CCBS can be used where provided by the ISDN service provider. It allows a callback to be set on external ISDN calls that return busy. It can also be used by incoming ISDN calls to a busy user. This feature is NOT available on standard ISDN DSS1 phones.
- **Partial Rerouting – PR**
  (2400, 4600, 5400, 5600, T3, T3 IP, DECT phones; Phone Manager)
  When forwarding a call on an ISDN channel to an external number using another ISDN channel, partial rerouting informs the ISDN exchange to perform the forward, thus freeing the channels to the IP Office. This feature is NOT available on standard ISDN DSS1 phones and it is NOT supported on QSIG.
- **Explicit Call Transfer – ECT**
  (The normal usage of this feature is by a third party application)
  ECT is supported on the S0 interface. A Call to an S0 Endpoint can be transferred to any other device such as an analog, digital or IP endpoint or to any trunk. The normal usage of this feature is by a third party application connected via one or more S0 interfaces to IP Office. One example is the VoiceDirector, an automatic call assistant.

North American T1

Provided by the IP400 PRI T1 and IP500 Universal PRI cards.

T1 Primary Rate provides up to 24 64K 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

Provided by the IP400 PRI T1 and IP500 Universal PRI cards.

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

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

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

Analog Trunks

- **Loop Start**
  Loop start trunks are available on the IP Office Analog Trunk cards installed within the IP Office control unit, or on the Analog Trunk 16-port expansion modules (ATM16). The first two trunks on the ATM16 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**
  Ground Start trunks are only available on the ATM16, configured through IP Office Manager. 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. Not available in all territories.

E1R2 Channel Associated Signaling

Provided by the IP400 PRI E1R2 and IP500 Universal PRI cards.

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

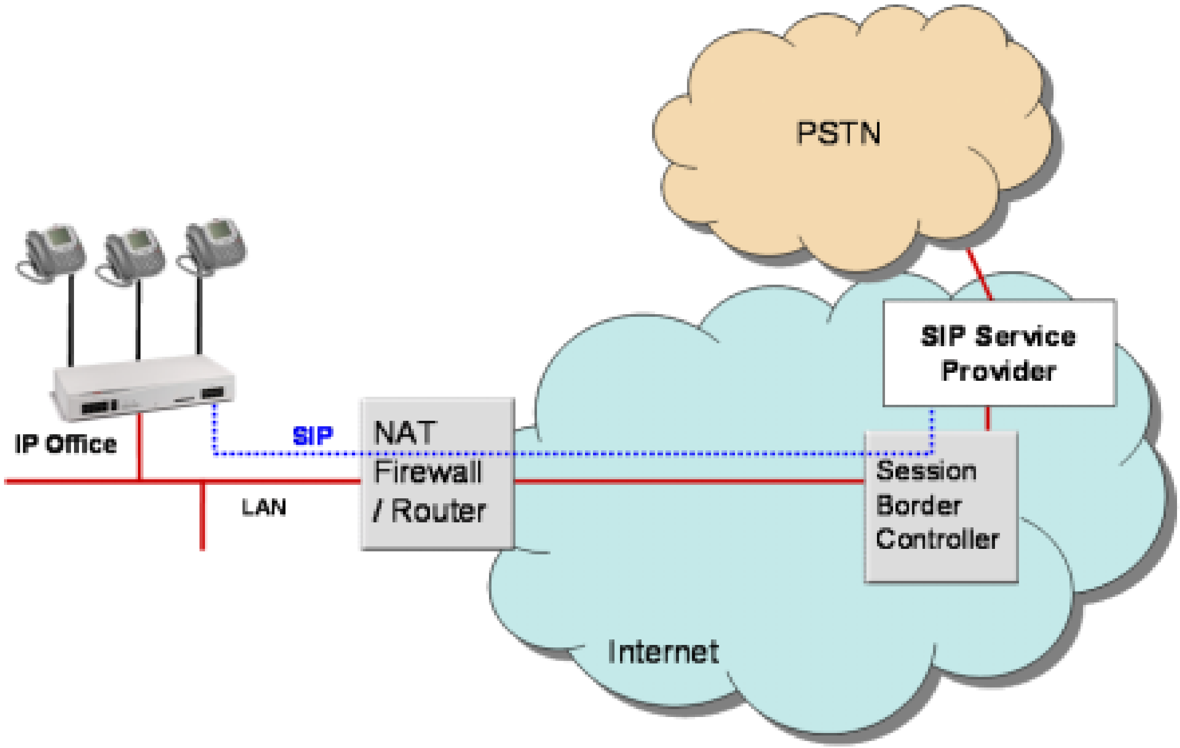
Session Initiation Protocol (SIP)

IP Office 4.0+ introduces SIP trunking. SIP trunks allow IP Office users to take advantage of new telephony services being offered by 'Internet Telephony Service Providers (ITSPs)'. In many cases, these telephony services can offer substantial savings in comparison to traditional exchange lines. The IP Office solution allows all users, regardless of their phone type, to make and receive SIP calls. SIP trunks are handled like any other line on IP Office, affording all the call routing and toll control needed to manage inbound and outbound calls.

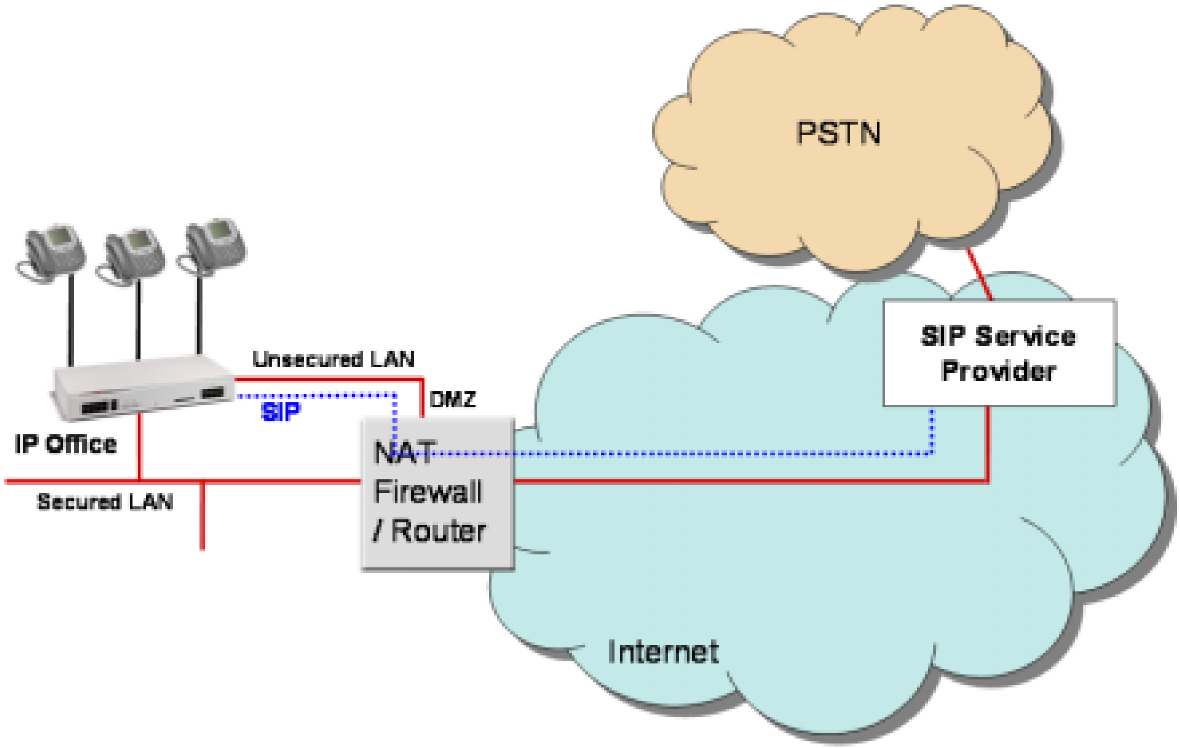
SIP trunks on IP Office require the provisioning of voice compression channels through the installation of VCM modules within the control unit. RTP Relay is also supported to allow the IP stream through SIP after call setup. A license for the maximum required number of simultaneous SIP calls is also needed.

There are several possible network topologies for SIP trunk systems, as shown in the following diagrams.

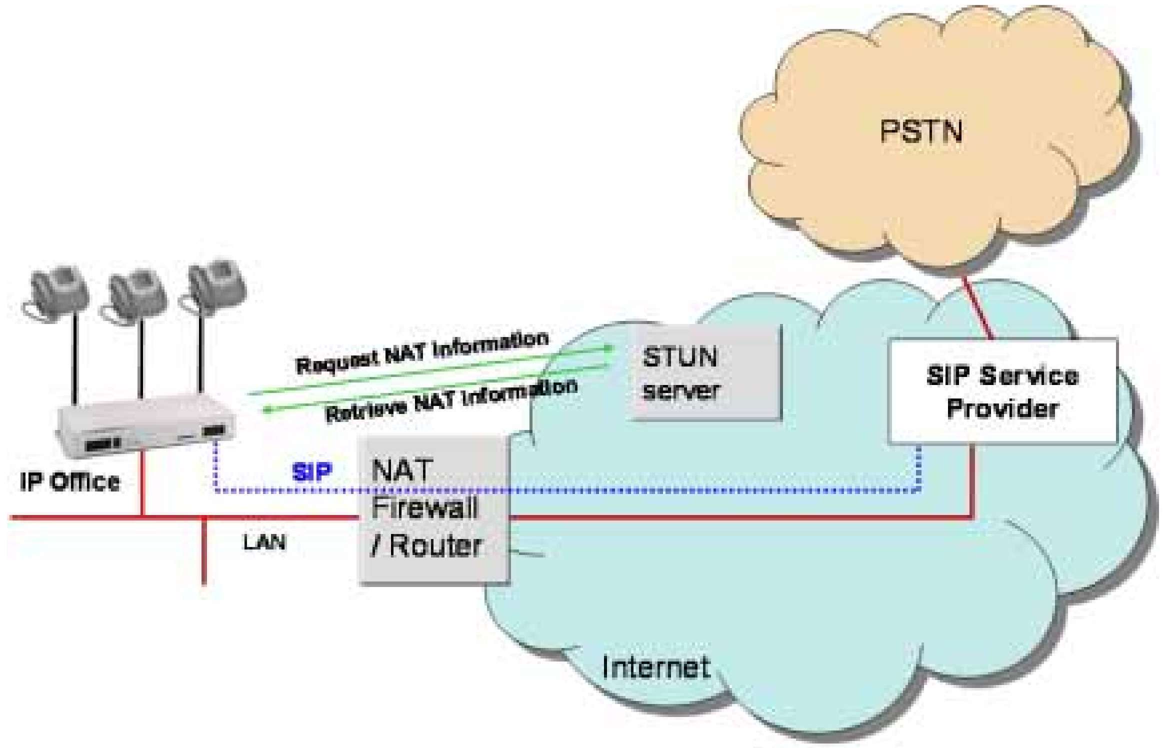
Option 1: Service provider with a Session Border Controller (SBC), which solves NAT traversal issues – this is the most reliable and preferred method.



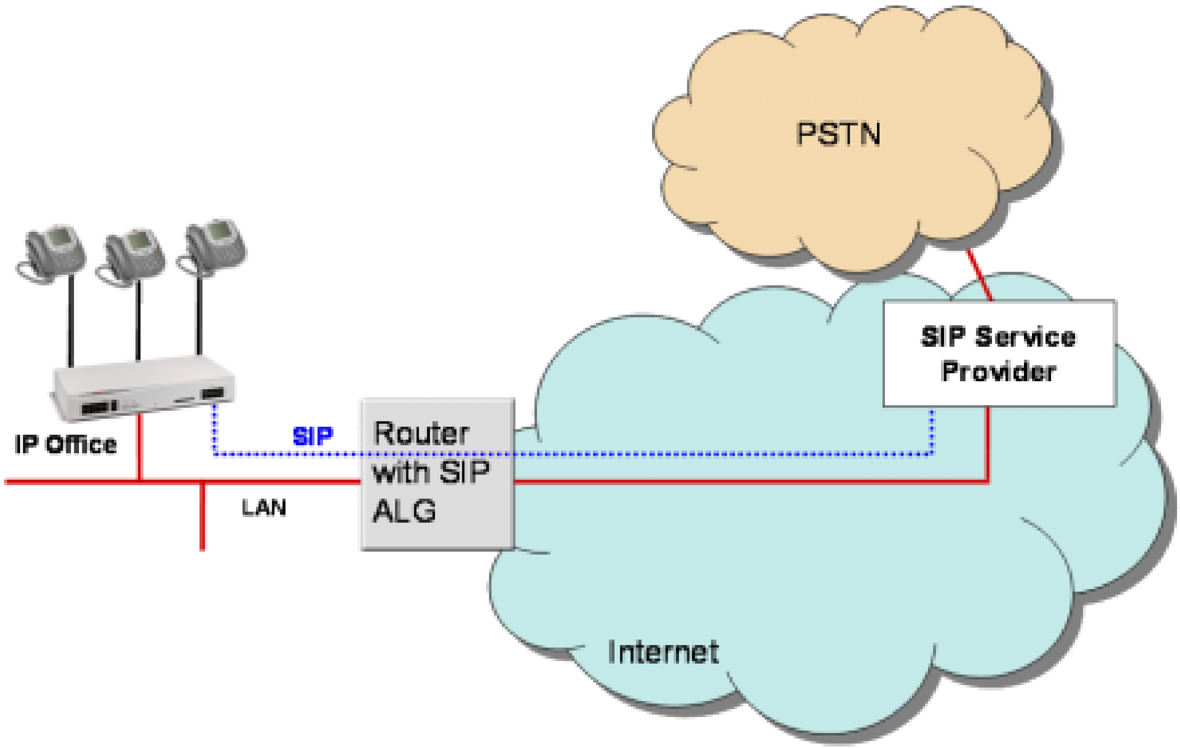
Option 2: Direct connection from the IP Office's second Ethernet port to the internet via a DMZ (demilitarized zone) port on the router. To make this configuration secure, the IP Office firewall is set to drop all packets except SIP.



Option 3: Connection to the ITSP over NAT using 3rd party STUN (Simple Traversal of UDP through NAT) servers in the network to discover the NAT mechanism being used.



Option 4: Connection to the ITSP through a router equipped with an Application Level Gateway (ALG) which transparently resolves all NAT issues.



For details on SIP ITSPs which have been tested by Avaya, please see the Technical Bulletin for the IP Office 4.0 release and/or IP Office Knowledge Base at http://www.avaya.com/ipoffice/knowledgebase.

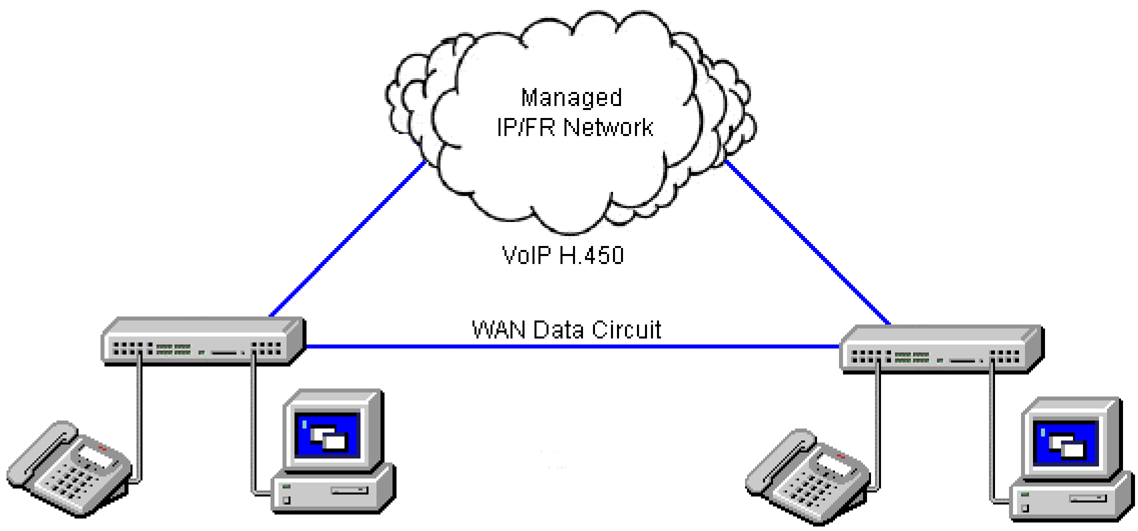
Packet Based Voice Networking

This section describes the options available for businesses that are able to utilize data networks to support voice solutions such as Voice over IP (VoIP). IP Office offers networked voice and data communications, providing:

- Built-in IP router
- One link for voice and data networking
- Common access to the Internet; share files and send e-mails to other sites
- Support for RIP-2 protocol for dynamic data routing; IPSec VPN, firewall and NAT (Network Address Translation) and for security; Centralized management and proactive fault management via SNMP,

IP 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 campus LAN.
- VoIP across the public network.



VoIP networking across IP network or WAN

VoIP over an Unstructured Private Circuit

Data networks can be constructed with IP Office using unstructured point-to-point data circuits (X.21, V.35) at speeds of up to 2 Mbps. These data circuits are accessed via optional Wide Area Expansion modules (one port is included on the IP Office system unit) and Voice Compression Modules (VCM). This approach can realize significant savings by allowing packetized VoIP calls to be interleaved with data on up to 7 leased data circuits with spare bandwidth. Depending on required solution sizing, IP Office supports from 3 to 128 concurrent VoIP calls.

VoIP over a Managed Frame Relay Network

Frame Relay is a high-speed, packet switching WAN protocol that enables the interconnection of LANs and is usually offered as a service by a public network provider. 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 circuit 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

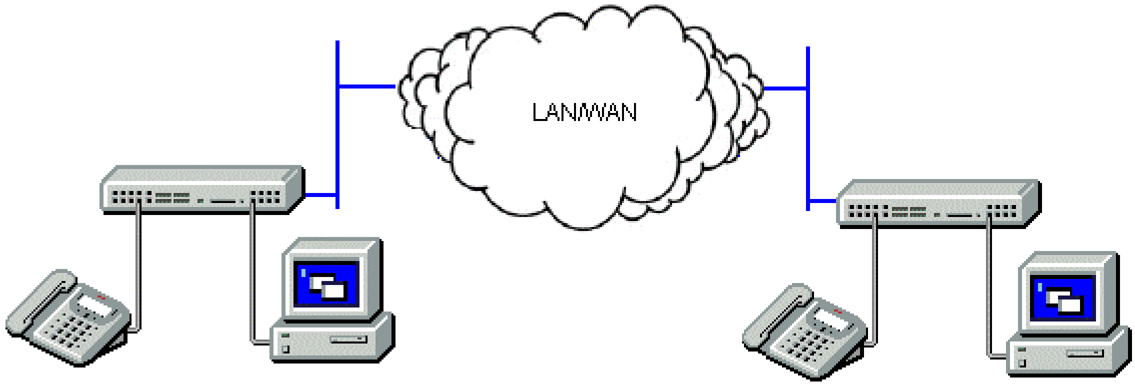
Even though IP Office can operate as a pure circuit switched system with analog and digital TDM handsets, because IP Office includes an integrated Voice over IP (VoIP) Gateway significant cost savings can be made by sending voice and data over a single managed IP VPN.

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

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. Avaya do not recommend networking IP Office systems over a unmanaged public IP networks where neither QoS nor service levels can be guaranteed by the provider.

VoIP across the LAN

In a factory or campus environment, voice calls can sent over 10/100 Mbps LAN connections on systems equipped with optional Voice Compression Modules (VCM). In order to avoid bandwidth contention issues, VoIP across the LAN will require some form of bandwidth management through Diffserve.



VoIP networking across the LAN

VoIP across the Public Network

IP Office is capable of realizing the benefits of Q.931 and H.450 supplementary service support across a public connection where an appropriate QoS connection can be established.

Supplementary Services within IP Networks

Supplementary services within an IP environment are provided via Q.931 and H.323. IP Office provides the same rich services as enjoyed within a traditional network environment. Our standards based approach allows interoperability within mixed vendor networks.

The basic supplementary service features supported by H.323 on IP Office to IP Office IP trunk links are listed below.

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

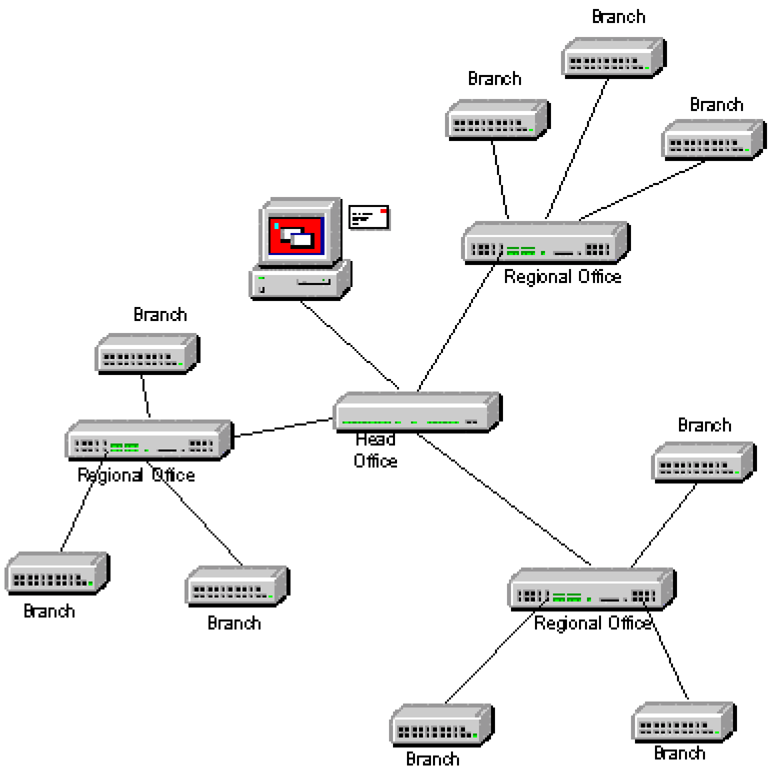
Additional features can be added through the use of IP Office Small Community Networking (see the following section).

On IP trunks to non-IP Office systems the Supplementary Service will depend on those also supported by the non-IP Office system.

Small Community Networking

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

- **Busy Lamp Field.**
- **Camp-on.**
- **Call Back When Free.**
- **Paging.**
- **Call Pick-up.**
- **Centralized Voice Mail (Voicemail Pro).**
  Support for mailboxes, call recording, dial by name and auto attendants. Remote queuing on remote systems is also supported with the Advanced Small Community Networking license (see below).
- **Internal Directory.**
- **Absent Text Message.**
- **Anti-Tromboning.**



For Small Community Networks VCM modules are required in all systems being connected. The IP lines should be configured to connect the IP Offices in a star configuration, however the data network itself can be meshed. Also the names and numbers (groups, line, services, etc) on the separate IP Office systems should be unique to reduce potential maintenance confusion.

Each IP Office system broadcasts UDP messages on Port 50795. These broadcasts typically recur every 30 seconds but BLF updates are potentially more frequent. There are no updates if there is no activity and the overall level of traffic is very low – typically less than 1 kbps per system.

From IP Office Release 2.1(35) and higher, SCN is supported between IP Office systems with differing software levels but network features will be based on the lowest level of software within the network. This option is intended to allow the phased upgrading of sites within a SCN and it is still recommended that all systems within a network are upgraded to the same level where possible. Always refer to the IP Office Technical Bulletin for the latest SCN compatibility matrix

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.

Note: on IP500 systems, Small Community Networking requires one or more additional licenses.

Small Community Networking - Advanced Networking Features

IP Office R4.0 allows a number of additional feature to be enabled by addition of an Advanced Small Community Networking license. Those features are:

- **Distributed Hunt Groups**
  Hunt groups can include users located on other IP Office systems within the network.
- **Remote Hot Desking**
  Users can hot desk between IP Office systems within the network. The system on which the user configured is termed their 'home' IP Office; all other systems are 'remote' IP Offices. To log on at a remote IP Office requires that IP Office to have an Advanced Small Community Networking license. A license is not necessary on the user's home IP Office.
- **Breakout Dialing**
  This feature allows the user to select an IP Office system in the network from a displayed list and then dial a subsequent number as if dialing locally on the select system. This feature is triggered either by a programmable button or short code.

Note that both Distributed Hunt Groups and Remote Hot Desking are not supported for use with CBC and CCC.

The Advanced Small Community Networking license is required in every IP Office site where remote workers are expected to hot desk to as well as on every site where members are included in distributed groups.

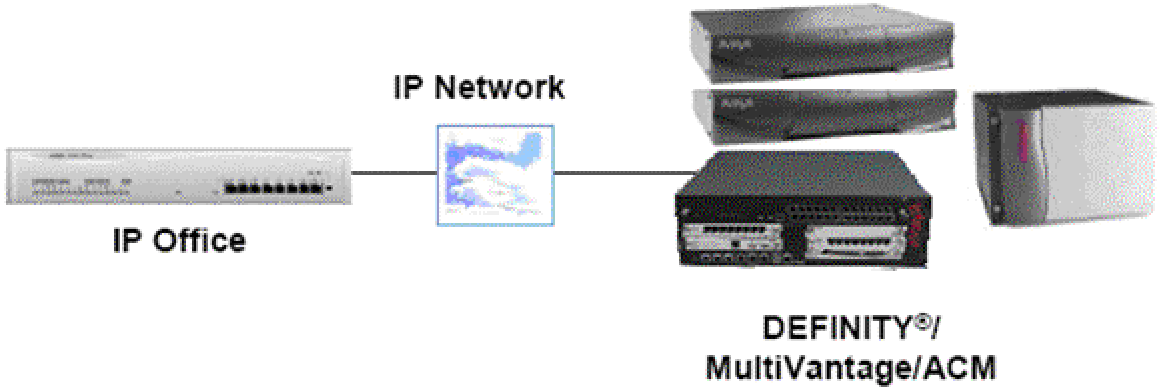
Internetworking with Other Avaya Products

IP Office will support the most appropriate way for communication with any other existing PBXs in a customer network, whether TDM or IP-based. With Avaya DEFINITY, MultiVantage, Avaya Tenovis I55, or Avaya Communication Manager (ACM), the protocols used will be QSIG or H.323 over T1, E1 or IP links

VoIP networking using H.323

IP Office (since release 1.1 in US and release 1.2 in EMEA) has been successfully tested to be interoperable over IP trunks with DEFINITY G3si (release 10) and IP600 (release 9.5). The protocol supported is H.323 using industry-standard compression codecs (types G.711A, G.711MU, G.729A and G.723.1-6K3). The features currently supported are as follows:

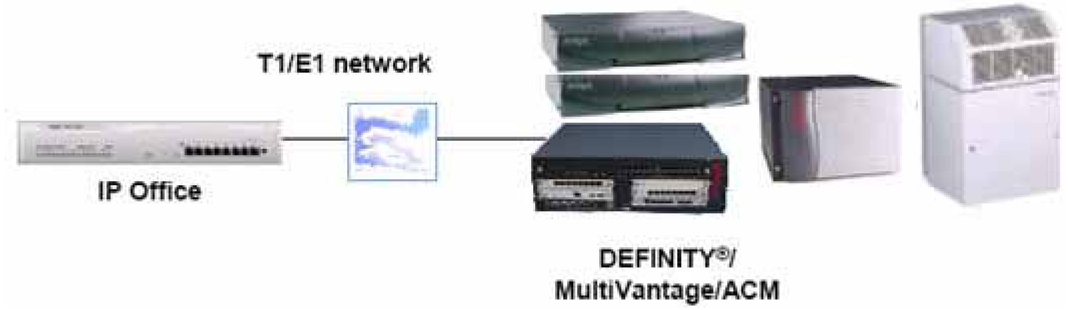
- Desk to desk dialing (basic voice call)
- Calling/Connected Party ID number
- Calling/Connected Name Presentation
- Call Hold
- Call Transfer



These features allow for simple networking needs between IP Office remote branches to a DEFINITY/ACM at the main site.

QSIG networking using T1/E1 links (TDM)

Alternatively QSIG may be favored as the chosen signaling standard within multiple vendor environments and provides the following supplementary services which are also available between IP Office and DEFINITY / MultiVantage/ I55 /ACM (equipped with the relevant RFA licenses):

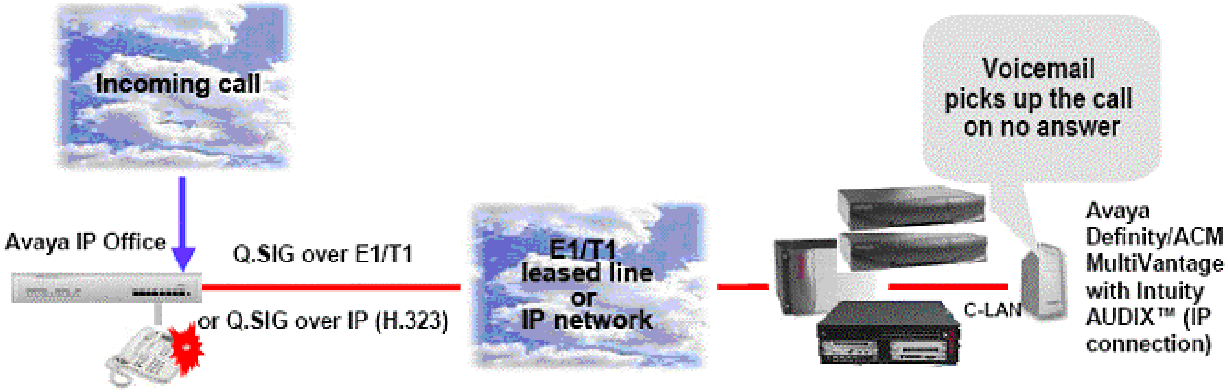


- Simple Telephony Call/Basic call (ETS 300 171/172)
- Circuit Switched Data Call/Basic call (ETS 300 171/172)
- Calling/Connected Line Identity Presentation (ETS 300 173)
- Calling/Connected Name Presentation (ETS 300 237/238)
- Message Waiting Indication (ETS 301 260/255)

Messaging Networking

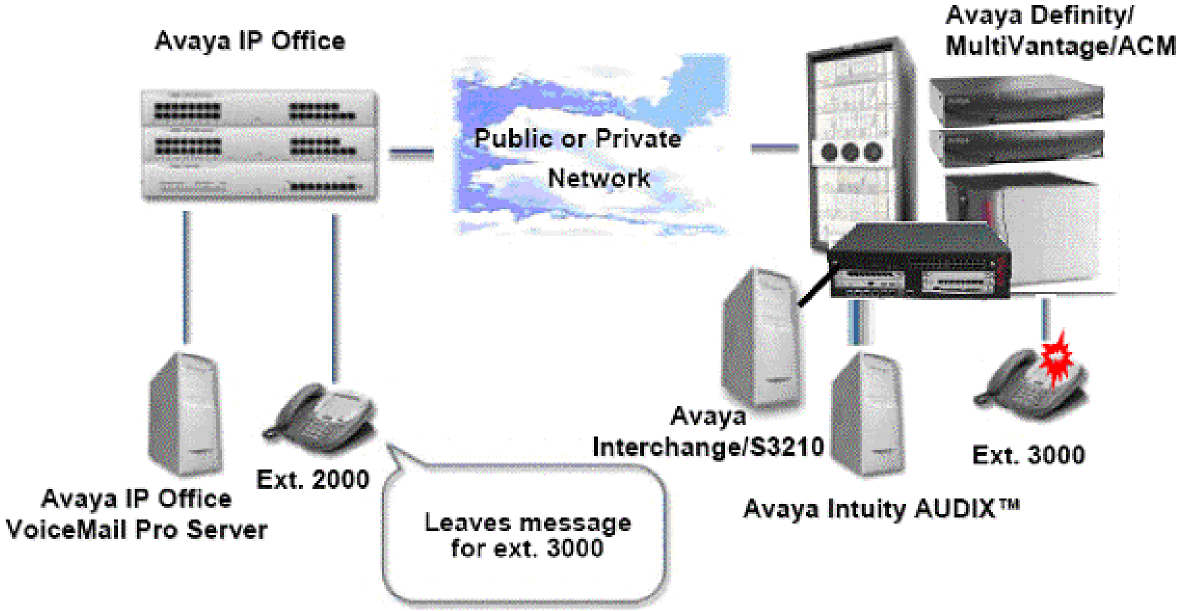
There are 2 options available today to provide messaging interoperability between IP Office and Definity / MultiVantage / ACM. The first option provides Centralized Voicemail while the second allows Avaya voicemail systems to be networked. The requirements, functionality and restrictions are summarized below:

- IP Office to DEFINITY / MultiVantage / ACM connected to Intuity AUDIX™ over a QSIG link (E1/T1 or IP)



  - No local Voicemail required on remote branch IP Office but AUDIX RFA required on every IP Office
  - Requires Intuity Audix 4.4+ connected via C-LAN to DEFINITY 9.5+ (see IP Office Offer Announcement dated August 2003 for more information on compatibility)
  - Maximum of 19 IP Offices can be supported on 1 INTUITY AUDIX™ server (20 total with DEFINITY/ACM occupying one slot)
  - Requires QSIG and Private Networking licenses on DEFINITY / MultiVantage / ACM
  - User mailbox with Message Waiting Light support
  - Forward voicemails between users
  - No auto attendant (enhancement currently being investigated)
  - No call recording
  - No queuing at remote sites
  - No Fax over IP to AUDIX™
  - No Small Community Networking support when AUDIX™ is configured on IP Office.

- Avaya IP Office Voicemail Pro networked to Avaya Modular Messaging / Octel / Intuity AUDIX™ via Interchange / S3210



- Requires local Voicemail Pro on every branch IP Office licensed with Voicemail Pro Networked Messaging RFA
- Requires Avaya Interchange/S3210 on Modular Messaging, Octel or Intuity Audix
- Provides 2,000 remote mailboxes per Voicemail Pro server i.e. per branch office (to be extended to 10,000 remote mailboxes by next Voicemail Pro maintenance release)
- User mailbox with Message Waiting Light support
- Forward voicemails between known remote users
- Fully-featured Voicemail Pro at every branch office
- Voicemail Pro Networked Messaging will only accept an incoming voicemail message for a local mailbox. It will NOT forward it to a remote Voicemail server. If required, this facility is available through Avaya Interchange.
- Voicemail Pro Networked Messaging RFA is currently in extended trial and limited to Avaya Messaging Servers (not third-party messaging platforms)

Common Networking Features

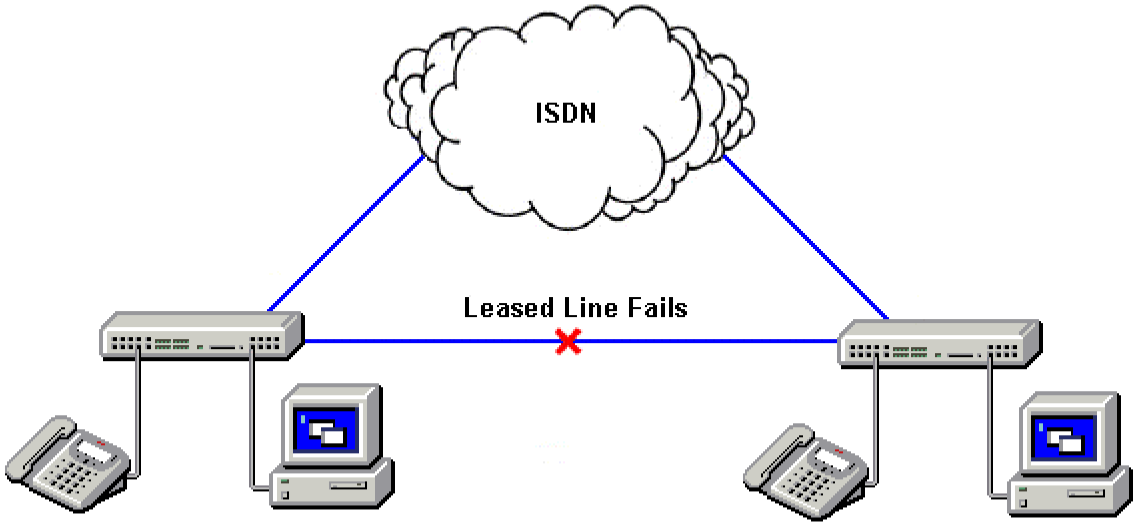
Alternate Route Selection

Alternate Route Selection (ARS) allows calls to be routed via the optimum carrier. Time profiles can also be used to allow customers to take advantage of cheaper rates or better quality at specific times of day.

If a primary trunk is unavailable or congested, then ARS provides automatic fallback to an available trunk (e.g., analog trunk fallback if a T1 or SIP trunk fails, or use PSTN for SCN fallback).

Multiple carriers are supported. For example, local calls are to go through one carrier between specific hours and international calls through an alternative carrier. Carrier selection using 2-stage call set up via in-band DTMF is possible. It is possible to assign specific routes on a per user basis, e.g. only allow expensive routes to be used by critical staff.

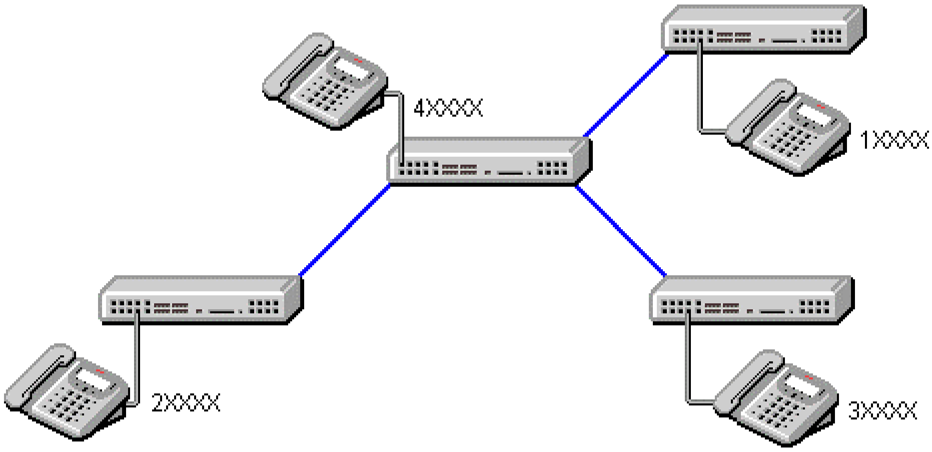
Note: Existing Least Cost Routing (LCR) configurations are automatically converted to ARS when upgrading to 4.1



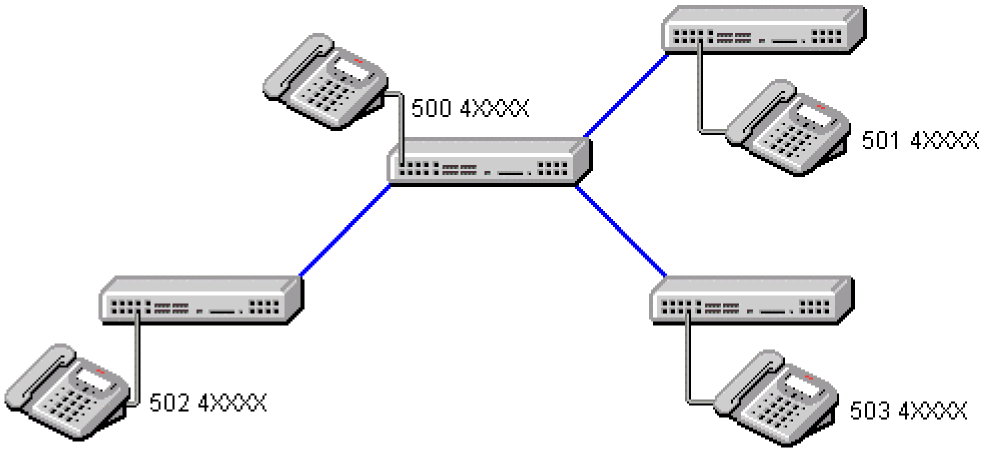
Network Numbering Schemes

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

7. Data Networking Services

LAN / WAN Services

Computers connected to an IP network in an office communicate via the LAN (Local Area Network). To support small networks both Small Office Edition and IP406 incorporate a Layer 2 Ethernet switch. The Small Office Edition supports 4 ports (with a fifth Ethernet port as a firewalled Layer 3 switch), the IP406 supports 8 ports. The IP412 and IP500 support a firewalled 2 port Layer 3 Ethernet Switch only.

When computers on the LAN communicate they do not care where the destination is, they just send messages with the address of the destination. These messages are likely to be received at all other computers on the same network but only one – the target destination – will act on the message. Where the destination is on another network, the router is needed to be the "gateway" to the rest of the world and find the optimum route to send the message on to the destination. The router alleviates the need to establish and hold a call for the duration of a communication session (when messages or IP packets are being sent between source and destination) 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). The IP Office system supports all of these types of network connections.

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 systems have an integral router with support for bandwidth on demand that allows the negotiation of extra bandwidth dynamically over time. Where connection is over ISDN, IP Office initiates extra data connections 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.

Quality of Service

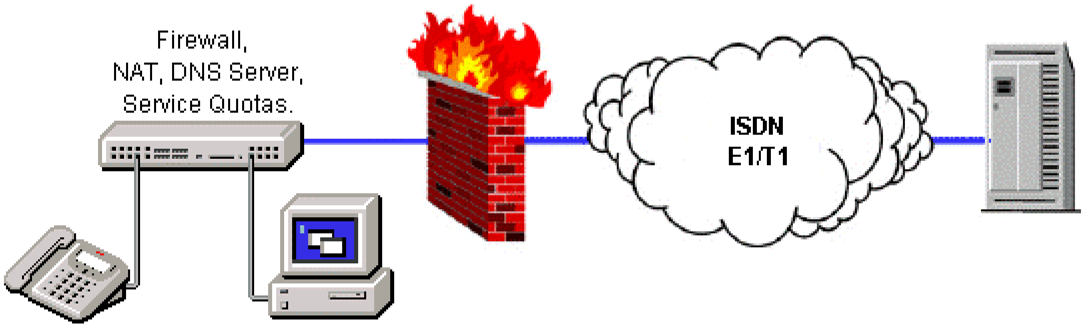
IP Office supports 802.1p packet prioritization. 802.1p is a specification for giving Layer 2 switches the ability to prioritize traffic (and perform dynamic multicast filtering). The prioritization specification works at the media access control (MAC) framing layer of the OSI model. To be compliant with 802.1p, Layer 2 switches must be capable of grouping incoming LAN packets into separate traffic classes. Eight classes are defined by 802.1p. Although network managers must determine actual mappings, IEEE has made broad recommendations. The highest priority is seven, which might go to network-critical traffic such as Routing Information Protocol and Open Shortest Path First table updates. Values five and six might be for delay-sensitive applications such as interactive video and voice. Data classes four through one range from controlled-load applications such as streaming multimedia and business-critical traffic - carrying SAP data, for instance - down to "loss eligible" traffic. The zero value is used as a best-effort default, invoked automatically when no other value has been set. In operation, 802.1p calls for the use of priority fields within the packet to signal the switch of the priority-handling requirements.

Internet Access

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

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

Internet security concerns are addressed through the provision of an integrated firewall so removing the need for a standalone firewall. 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. The firewall 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 a remote access service to ensure authorized users can gain access. Service Quotas place a time limit on outgoing calls to a particular IP Service so limiting costs. Each service can be configured with an alternative fall back, 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 can be supplemented by CHAP (encrypted passwords) to verify the end users, 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 for dial in. 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.

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

LAN to LAN Routing

All businesses now have a need for data routing 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. 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 multiplexers 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.

Data Networking Features

Integral 10/100 Mbit Layer 2 Ethernet Switch

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

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

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

Integral 10/100 Mbit Layer 3 Ethernet Switch

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.

It is possible to set up a firewall between two LAN segments using the IP Office layer 3 switch. Small Office Edition offers a firewall between its four port Layer 2 Ethernet switch and its Layer 3 Ethernet WAN port, while IP412 and IP500 support a two-port Layer 3 Ethernet switch with the firewall between them. Both of these switched ports have their own IP addresses (LAN1 and LAN2) and in order for traffic to pass from one port to the other, a route is configured in the system’s routing tables.

From Release 4.1 onwards, port 8 of the IP406 V2 Ethernet Switch can optionally be configured as LAN2.

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 DHCP server or be set with its own static IP address. Both IP Office - Small Office Edition and IP412 have two independent DHCP servers, one dedicated to each of the Layer 3 switched LANs.

Leased Line Support

All IP Office systems 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 ISDN dial-up circuits using its E1/T1 or Basic Rate trunks. Where data speeds greater than a single channel are required (64K/56K), additional channels can be added to the call as and when they are needed.

Point-to-Point Protocol (PPP)

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

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

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

Frame Relay

Frame relay is a wide area networking protocol based on 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 can be configured to limit the maximum number of minutes that a service, such as Internet Access, is available for each user. 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 times for service. For example, a time profile could be set up 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. This feature also applies to days of the week or specific calendar dates.

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, but adds to transmission latency.

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 remote end of the data call to request additional calls to be made to improve aggregate 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 names such as www.avaya.com to the domain's IP address required to establish a connection. IP Office provides this service to PCs on the network by proxy.

Network Address Translation (NAT)

Network Address Translation is a mechanism that allows you to use different IP address on your private network behind a router with a public IP Address. 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 and providing additional security to the internal users as their address in hidden to the public.

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

IP Office integrated firewall provides packet 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 through IP Office Manager. 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 IP Office to be synchronized with the information on an LDAP server (limited to 500 entries). Although targeted for interoperation with 'Windows 2000 Server Active Directory', the feature is sufficiently configurable to interoperate with any server that supports LDAP version 2 or higher.

Remote Access Server (RAS)

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

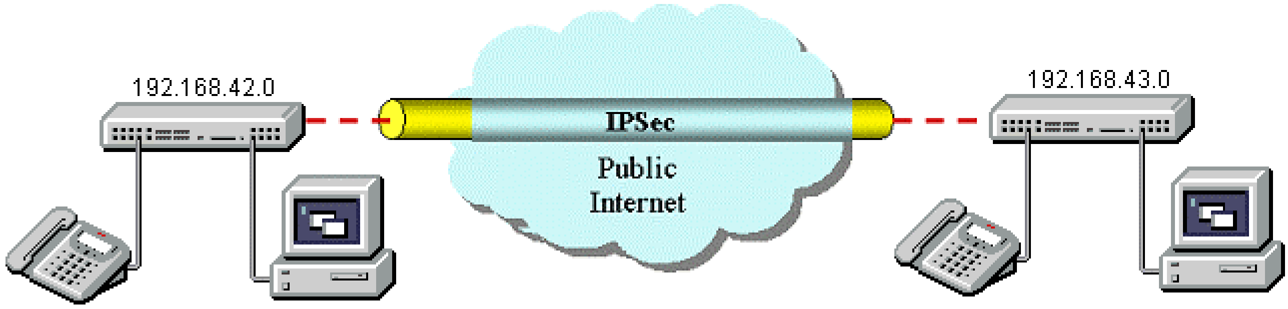
Transaction Packet Assembler Dissembler (TPAD)

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

Routing Information Protocol (RIP)

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

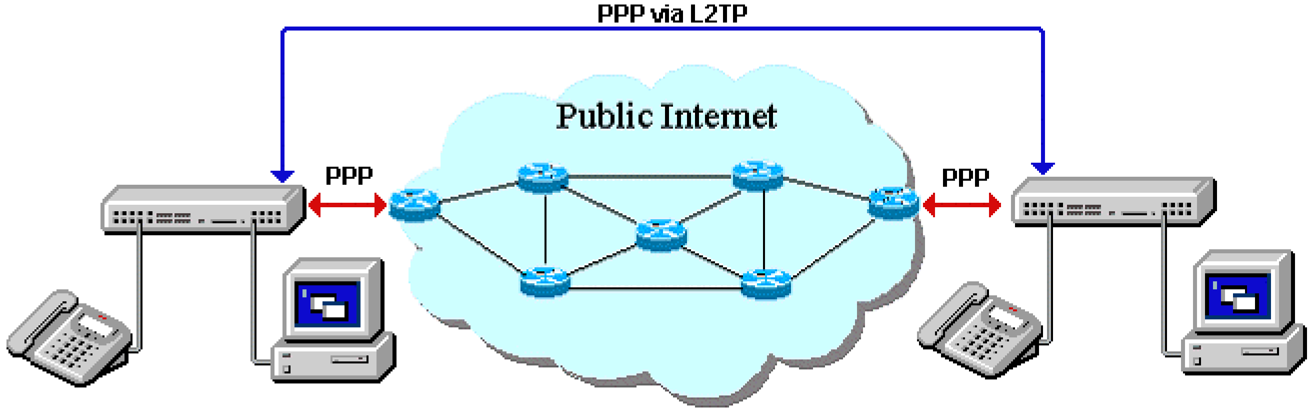
VPN: IPSec Tunneling



IPSec tunnels allow a company to pass data between locations over unsecured IP networks such as the public internet. The company 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 IP Office systems 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 optional and enabled on IP Office through a License Key.

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

VPN: Layer 2 Tunneling Protocol



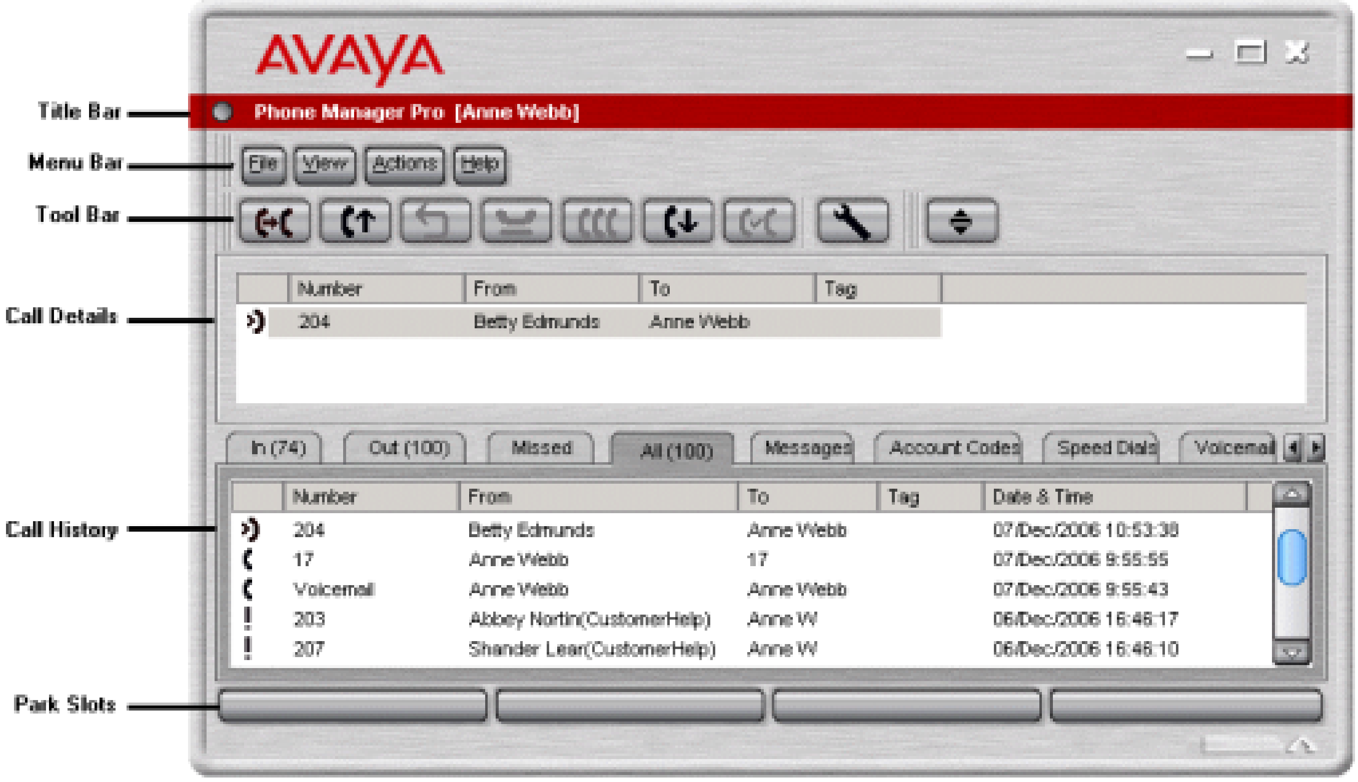
PPP authentication using PAP or CHAP takes place between directly connected routers only. When using a public IP Network to connect sites this authentication takes place between the customers router and the service provide router that it is connected to. In some circumstances it is desirable to authenticate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

8. Phone Manager

Phone Manager

The IP Office Phone Manager application provides users control of their telephone from a networked PC.

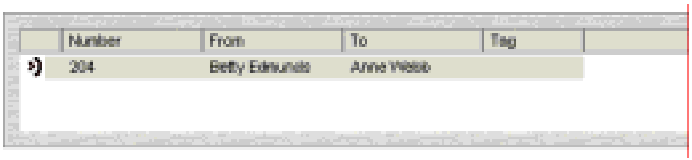
Phone Manager can be used with any IP Office extension; analog, digital or any IP telephones, wired or wireless, and is available in three versions: Phone Manager Lite, Phone Manager Pro and Phone Manager PC Softphone subject to
licensing.



Phone Manager Lite

Phone Manager Lite is included as part of the IP Office solution free of charge for every user and provides easy access to telephony features, call information and call control. Phone Manager will normally run in the Windows system tray once the user is logged on, minimizing screen space when not in use.

Caller ID/Name Presentation



Caller ID is presented as standard (where provided) allowing users to see who's calling before answering. The caller's phone number and name (if known to IP Office) are clearly shown in the call status area of the Phone Manager screen. For incoming calls, the dialed destination is also visible, for example the user’s Direct Dial number, or a specific department, for example switchboard, sales, support or administration.

This feature allows users to answer the call appropriately and gives the flexibility to participate in multiple hunt groups, particularly important for small businesses. The same information is also displayed should a second incoming call be presented, allowing users to easily switch between calls or allow the second call to go to voicemail. Users can choose to have the information pop-up on their PC automatically as soon as a call is presented, or when the call is answered.

Call History

Phone Manager's call history keeps a combined record of up to 100 calls while the application is active. Double-clicking any logged call dials that number. If Advice Of Charge service is available from the ISDN service provider, this will also be displayed for outgoing calls.

Voicemail Access

Phone Manager Lite provides notification of any new voicemails received and provides access into the user or group’s mailbox allowing messages to be played.

Desktop PC Telephony Controls

Phone Manager has telephony buttons on a tool bar that activate standard telephone functions such as Answer, Transfer, Hold, Account codes and Conference etc. so that users don’t need to remember IP Office specific feature codes. Personal settings such as Do Not Disturb (including exceptions list), call forwarding, mobile twinning and voicemail transfer option settings can be easily set up using Phone Manager.

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

Personal Productivity & Collaboration

All versions of Phone Manager feature a Busy Lamp Field (BLF) and Speed Dials. This allows users to customize the application to reflect the status of their department, immediate colleagues or the whole company as desired. The Direct Station Select allows you to dial regularly used internal and external numbers via a single-click. A single Direct Station Select icon allows you to dial their work, mobile/cell phone and home numbers. The Busy Lamp Field feature allows you to see at a glance, who is available to take a call, who is already on a call and who has placed their phone on Do Not Disturb. BLF information is also available on remote users as long as they are on a Small Community Network (SCN). Phone Manager Lite supports up to 15 Speed-Dial/BLF entries.

Internal User	
	Busy
	Message
	Divert
	Do Not Disturb
	Not logged into LCS
	Logged into LCS

External Number	
	Work
	Mobile
	Home
	Fax

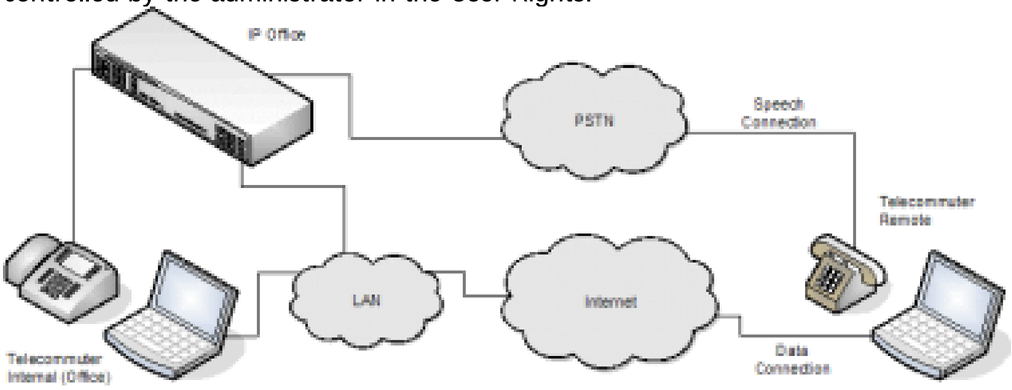
Where Microsoft Live Communications Server (LCS) is also available within the user's business, Phone Manager users can view colleague's presence (online, offline) as well as send Instant Messages (IM) via Phone Manager. For example users can send an IM to alert a colleague that an important call is waiting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

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

Phone Manager Pro

Phone Manager Pro is licensed on a per-user basis and provides all of the Phone Manager Lite features plus the following:

- **Personal Productivity and Collaboration**
  Phone Manager Pro offers increased capacity from 15 to 1000 Speed Dial/BLF entries. These are distributed across 10 tabs to allow users to group speed-dial/Busy Lamp Field icons by department or location, for example Sales, Support, etc. Each speed-dial tab supports up to 100 speed dial/BLF entries.
- **Telecommuter Mode**
  Phone Manager Pro allows making and receiving calls and retrieving voicemails on an external phone number as if they were in the office, with Phone Manager providing the call control. It also provides billing convenience and potential cost savings for remote workers and mobile work force. Access to the feature is controlled by the administrator in the User Rights.



  When logging on, a Telecommuter user will be asked to choose the phone number they can be reached at. This number is either entered directly or is part of a previously saved profile. Once selected, Phone Manager will use this number to make and receive calls and retrieve voicemails for the duration of the session.

- **Integration with Contact Management packages**
  To facilitate screen popping of the contact details of an incoming caller, dialing from the contact record with a simple mouse click and simple creation of new contact records with auto-insertion of the telephone number while on a call. The user can select which Contact Management should be popped:
  - **Outlook**
  - **GoldMine**
  - **ACT!** *(*ACT! 7.0 and higher requires the TAPI.NET add-on from various providers plus the IP Office TAPI driver from Avaya)*
  - **Maximizer.**

- **Voicemail Pro mailbox control**
    - **Manage voicemails**
      Phone Manager Pro allows users to play, rewind, fast-forward, save or delete their voice messages.
    - **Manage Personal Distribution Lists**
      Phone Manager Pro allows users to configure their Personal Distribution Lists (Voicemail Pro Intuity mode only).
    - **Manage voicemail greetings**
      Users can record & select which of the personal greetings is active (Voicemail Pro Intuity mode only).
- **Personal Directory**
  Personal phone number directory which allows further personalization and improves productivity:
- **Name matching**
  If the Caller ID is recognized in the local PC directory, the caller's name can be displayed. Up to 1000 entries are supported.
- **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 message to use.
- **Distinctive ringing**
  Allows the configuration of distinct ringing on a per caller basis. WAV 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, clients, and unknown callers.
- **Compact Mode**
  Compact mode minimizes the screen space required to run the Phone Manager Pro application. While in compact mode, a notification slider alerts new calls and allows the user to view the caller ID or associated caller's name and answer the call. Users can easily switch between standard and compact modes.

- **Agent Mode**
  Agent mode operation 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. Agent-mode users can set their phone to "Busy" or "Wrap-Up" and select which hunt group they are member of via simple button clicks. Access to this feature is controlled by the administrator via User Rights.

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

- **Account Codes tab**
  Users can easily activate Account codes (before or during the call) through the 'Account Codes' tab. This tags calls with an alphanumeric account code via a single-click. Note: Lite users can enter account codes but cannot view the Account Codes tab.
- **Queue monitoring**
  Queue monitoring allows the user to see the number of calls waiting in up to 2 queues. The Phone Manager Pro user does not need to be part of the hunt groups being monitored.

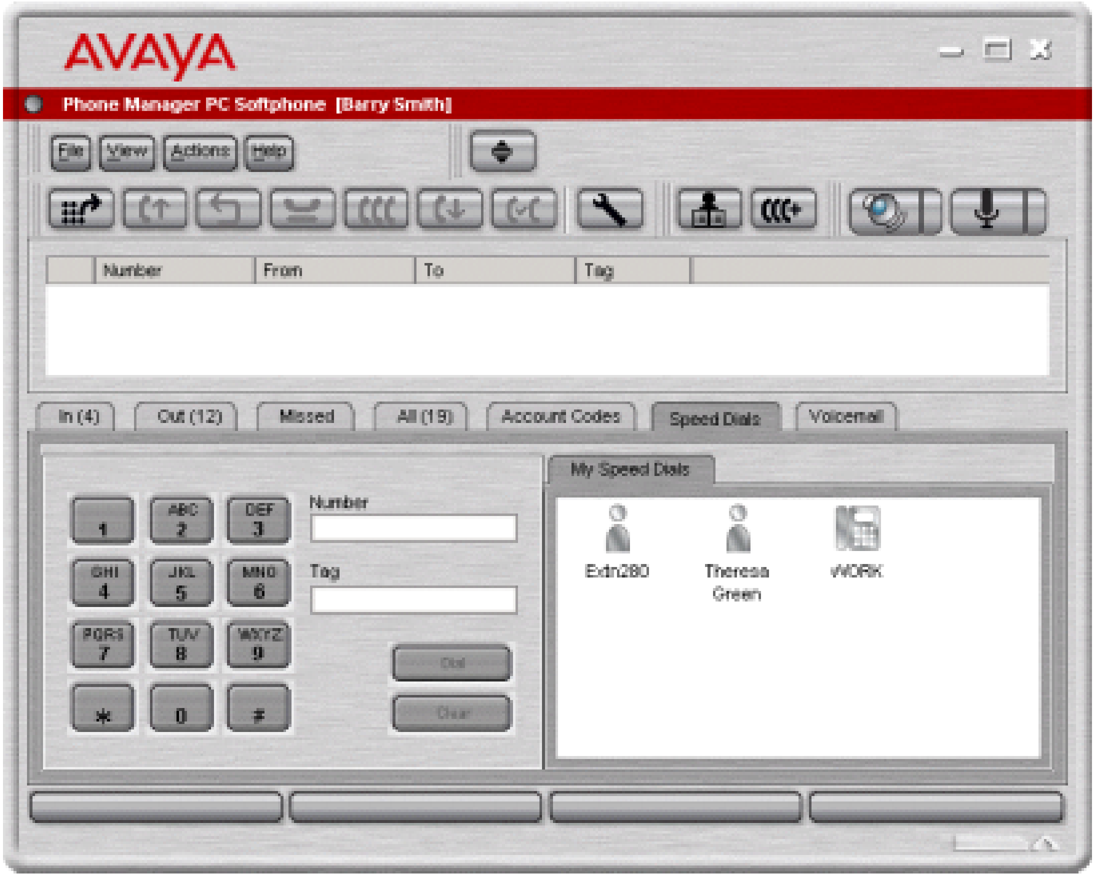
- **Door entry control**
  Door entry control allows the user to open or close the two external relays in the IP Office system. This can be used to activate an external system such as door-entry or security camera.
  

- **Call History**
  Phone Manager Pro provides separate tabs for Incoming, Outgoing, Missed and All Calls. Each call log tab will store the last 100 entries which can be sorted by date & time, caller ID and call duration if required.

Phone Manager PC Softphone (IP Softphone)

Phone Manager PC Softphone is licensed on a per-user basis and provides all of the Phone Manager Pro functionality. In PC Softphone mode, both audio and call control operations are handled on the PC so no additional telephone is needed. When using PC Softphone, the user will need an audio device such as a headset or USB handset, both USB and soundcard interfaces can be used with PC Softphone.

PC Softphone can be twinned with another IP Office extension offering mobility and choice so that calls can be answered on either endpoint.



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

Phone Manager Feature Summary

Feature	Phone Manager Lite	Phone Manager Pro and PC SoftPhone
Inbound/outbound call handling.	Yes	Yes
Phone call control.	Yes	Yes
Configure phone preferences.	Yes	Yes
Configure keyboard short cuts.	Yes	Yes
CLI (ANI) / Name display.	Yes	Yes
Speed dial / Busy Lamp Field management.	Yes - 15 icons maximum.	Yes - 100 icons maximum per tab.
Speed Dial tabs (to group Busy Lamp Field icons)	Yes - 1 tab.	Yes - 10 tabs maximum.
Microsoft Live Communications Server (LCS) Integration	Yes	Yes
View internal users' presence via LCS	Yes	Yes
Send Instant Messages (IM) to internal users via LCS	Yes	Yes
Telecommuter mode	–	Yes (not PC SoftPhone)
Compact mode	–	Yes
Local Phone Directory.	–	Yes - 1000 entries maximum.
Call history log – all, missed, messages.	Yes	Yes
Separated incoming/outgoing call log.	–	Yes
Collect new voicemail messages.	Yes	Yes
Voicemail box control (Intuity and IP Office modes).	–	Yes
Personal Distribution List set up (Intuity mode)	–	Yes
Incoming call scripting.	–	Yes
Time on call.	–	Yes
Advice of Charge (ISDN service provider dependent)	Yes	Yes
Door opening control.	–	Yes
Queue monitoring.	–	Yes - 2 Queues
Conference Control Display.	Yes	Yes
Conferencing Center action buttons	Yes	Yes
'Screen pop' contacts (Outlook, Goldmine, ACT! and Maximizer).	–	Yes
Simple Outlook contact record creation.	–	Yes
Agent Mode.	–	Yes
Distinctive Ringing (WAV file).	–	Yes
Post Connect dial (sending DTMF while connected to another party).	Yes	Yes
VoIP mode (to run as an PC Softphone)	–	Optional license

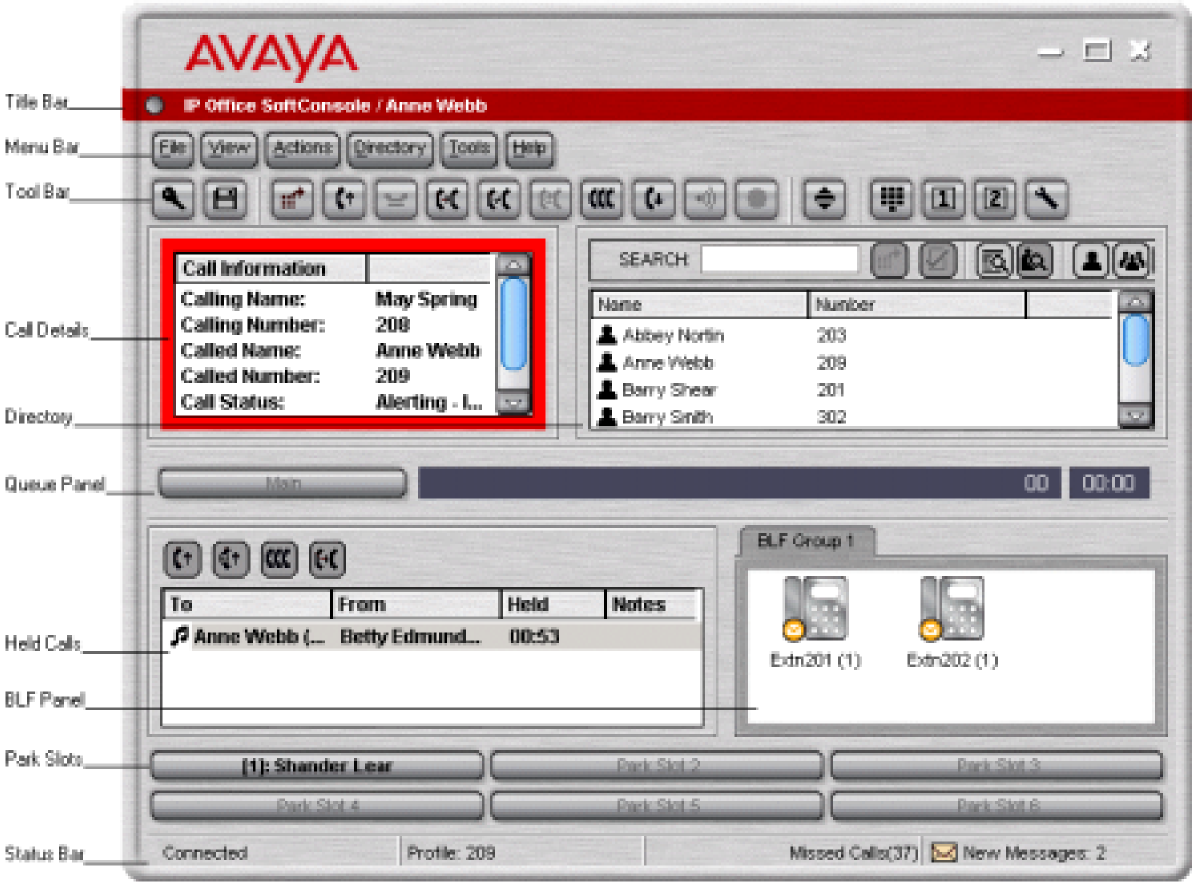
Phone Manager System Requirements

- **Phone support:**
  Any telephones connected to IP Office, although hands-free operation is only supported on suitable Avaya Digital and IP telephones.
- **PC requirements:**
  - Always refer to the latest Avaya IP Office Technical Tip or Technical Bulletin for any updated information with regard to Operating Systems, Service Packs or PC hardware
  - Refer to Technical Specifications section of the Product Description for Operating System and Hardware requirements
- **Licensing:**
  - **Phone Manager Pro:**
    Requires a Phone Manager Pro license for each user.
  - **Phone Manager PC Softphone:**
    Requires an IP Office PC SoftPhone license in addition to the Phone Manager Pro user license. There must be equal or greater Phone Manager Pro licenses than PC SoftPhone licenses. The use of a headset is strongly recommended. Operation through standard speakers and integral PC microphones is possible but not recommended.
- Phone Manager Pro screen popping provides integration with Microsoft Outlook 2000/2003/XP, Act! 6.0 and 2005, Maximizer 7.5 and 8.0 Enterprise, Goldmine 6.0 and 6.7.
- Phone Manager PC Softphone supports QoS in the form of DiffServ for both Windows XP/2000.
- Phone Manager PC Softphone can be used over a wireless LAN; in this configuration no more than 3 simultaneous calls can be supported per access point.
- Up to 360 Phone Manager users can be supported on the same LAN subnet as IP Office. Where remote subnet Phone Manager users are deployed, up to 10 remote users will receive BLF updates.
- Up to 5 Phone Manager users can be supported on a single Citrix thin client server
- Instant Messaging options require the network to have a Microsoft Live Communication Server (LCS) with both a server license and client licenses for each user. Phone Manager has been verified as compatible with Microsoft LCS 2003 and 2005. No additional license is required in IP Office.

9. SoftConsole

SoftConsole

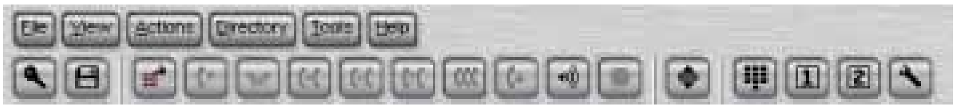
SoftConsole is the PC based Windows Operator Console for IP Office. SoftConsole has been designed to improve operator service by providing the operator with call information and available call actions to simplify call handling and give the appropriate response to the caller. With this easy to use software tool the operator can maintain visibility of the number and type of calls waiting and so ensure that clients are greeted in a professional manner. SoftConsole has a similar look and feel to the Phone Manger application and can be minimized in the Windows system tray when not in use, but will pop up on the screen when a call is received.



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

The SoftConsole screen is divided into the following areas:

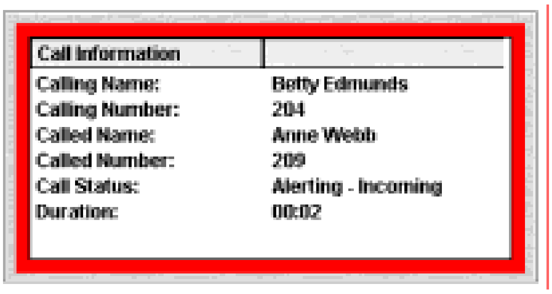
- **Main Menu Bar**



Commands & actions are available through menus. Some features can only be used when the right conditions. If they are not available, the feature will be “grayed out” until conditions change that allow the feature to be used. The following features are available on the tool bar:

- Login.
- Save Profile.
- New call.
- Answer call.
- Hold call.
- Transfer call.
- Transfer complete.
- Reattempt transfer.
- Conference.
- Hang up.
- Page.
- Record call.
- Compact view.
- Dial Pad.
- Access conference room 1.
- Access conference room 2.
- Options.

- **Call Details Panel**

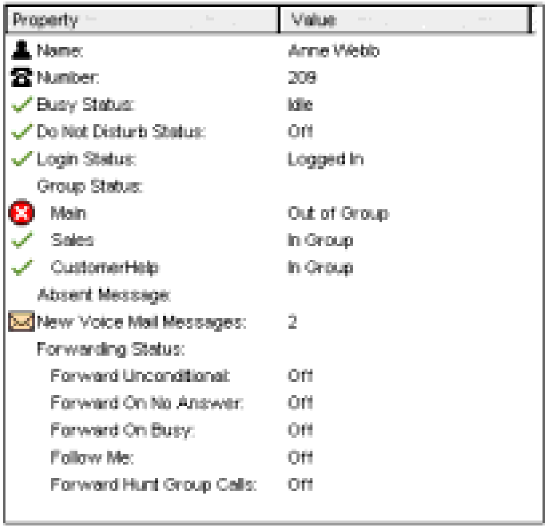


The call details panel on the left shows details of the current call which will include the following information:

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

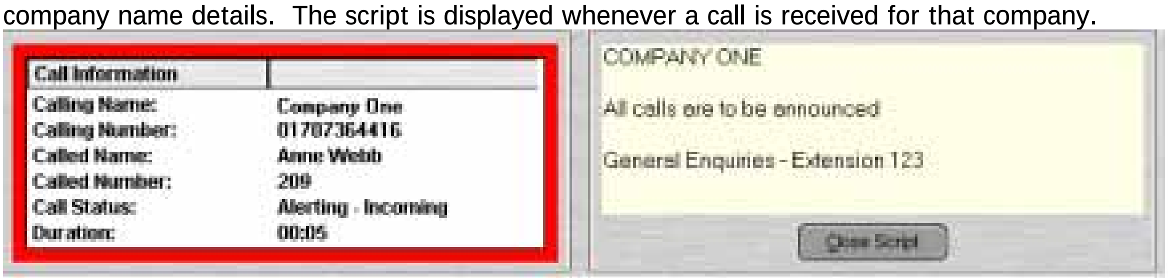
If a new call arrives, the call details panel will display the calls waiting to alert the operator and allow answering of the call based on the Caller ID.

- **Directory Panel**



The directory panel on the right shows information on following:

- **Directory entries**
  Including IP Office users, hunt groups and external directory user (non IP Office extensions)
- **Single directory entry details**
  Including IP Office users, Hunt Groups and external directory user (non IP Office user).
- **Script**
  When a script has been configured for either the calling or called number, the script is displayed in this panel. For example, an operator may be answering calls on behalf of more than one company. To ensure the call is answered with the correct company name a script file can be created with the company name details. The script is displayed whenever a call is received for that company.



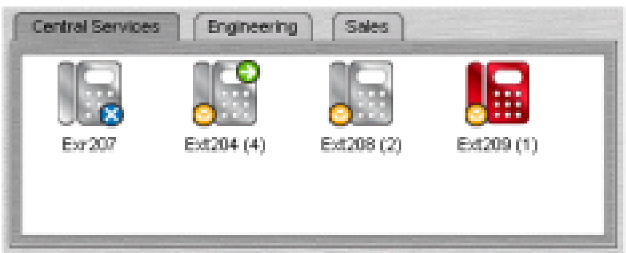
- **Conferencing**
  Within SoftConsole, calls can be conferenced when held, or a conference can be created through the two conference rooms:
  - **Conference Held Calls**
    An operator can conference calls that are in the Held Panel. All calls in the Held Panel will be conferenced.
  - **Conference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

Not Invited. Invited. Joined. Declined. Unavailable.

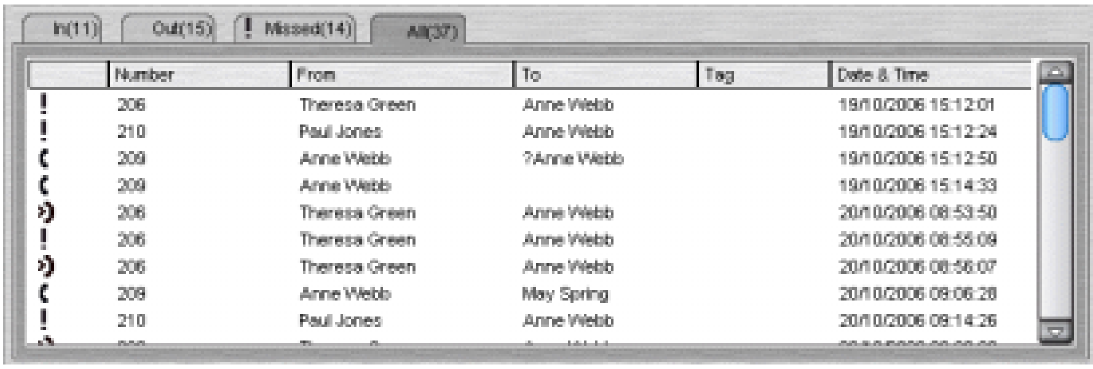
- **Queue Panel**
  The queue panel displays a bar graph of the number and the status of external calls held in a particular queue. Up to 8 Call queues can be configured and labeled to reflect incoming calls for specific Hunt Groups.

- **Held Calls Panel**
  The held call panel enables the operator to manage all calls held at the operator station. These calls 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, for example Busy, DND and Forwarded is indicated by the various icons used. Up to 10 tabs with 100 icons on each tab are supported.



- **Park Slot Panel**
  The park slot panel can contain up to 16 system-wide park slots with specific Park ID's for each slot.
- **Call History**
  SoftConsole's call history keeps a combined record of up to 100 (incoming, outgoing and missed) calls while the application is active Double-clicking any logged call dials that number.



- **Status Bar**
  This Shows current status of the system and is divided into four sections that display current connection status, current Profile name, information messages and The number of new voice mail messages for the operator. Information messages include any alarm conditions that are present within the system.

SoftConsole Options

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

- **Incoming Calls**
  This tab enables the operator to manage the local SoftConsole directory by creating, editing and deleting entries from the selected directory. Also the operator is able to associate a script or media file with each specific entry.
- **Queue Mode**
  This tab enables the operator to configure the queue window with up to 8 hunt group queues, which will include a recall queue. Queues can be created, edited and deleted while also providing the operator with the additional benefit of positioning them in the queue window in order of operator preference. Management by exception is used to monitor queue status by enabling the operator to set up various alarm thresholds such as the Number of calls in queue and Longest waiting call time. A WAV media file can be associated with an alarm for further customization.
- **Park Slots**
  This tab enables the operator to define which park slots are accessible on a system wide basis up to a maximum of 16. The operator is also able to assign which numbers are used to access each park slot and where the slot appears in the park slot panel.
- **BLF Groups**
  This tab allows the operator to create and edit BLF groups.
- **Door Entry**
  This tab allows the operator to configure up to two door entries.
- **Directories**
  This tab enables the operator to choose access to the following directories: SoftConsole local directory, IP Office system directory and Microsoft Outlook contacts. Once chosen, the operator is able to map fields to directory entries.
- **Conferencing**
  This tab allows the operator to set up the names of the two conference rooms. The names will appear on the telephone displays of users in the conference room (maximum of 10 characters).
- **Keyboard Mapping**
  This tab allows the operator to assign keyboard short cut keys for SoftConsole functions.
- **Keyboard Actions**
  This tab allows the operator to specify the default action when alphabetic or numeric characters are pressed.
  - **Alphabetic Keystrokes:** Begin directory search or Open call annotation window
  - **Numeric Keystrokes:** Begin directory search or Open pop-up dial pad
- **Appearance**
  This tab allows the operator to change the appearance of SoftConsole fonts, skins and the call information window color.
- **SoftConsole**
  This tab allows the operator to save the changes made to the configuration of SoftConsole either automatically or manually to a local configuration file on the PC.

SoftConsole Administration

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

- **Control panel views**
  The BLF panel, call history panel, held calls panel and park slot panel can be hidden or made visible.
- **Change the Administrator password**
- **Edit operator profiles**
  Each operator can have a personalized profile, which can be configured by the administrator.
- **Create and modify templates**
  SoftConsole comes with three predefined templates, which can be modified, or new templates can be created.
- **Specify the maximum length of call notes**
  IP Office supports a wide range of different telephone types. These have different display sizes so the operator can define the character length of messages sent to each user according to the type of phone they use.
- **System Tray working**
  The application can be minimized and left running in the system tray so that it can pop on received calls.

SoftConsole Telephone Requirements

- SoftConsole provides extensive call management, but it still requires an IP Office telephone to provide the speech path. SoftConsole has been tested and is certified to work with all Avaya wired digital and IP phones that are listed in chapter 4.
- SoftConsole cannot be used with IP DECT 3700 series telephones.

SoftConsole PC Requirements

- IP Office software release 2.0 or later.
- PC requirements:
  - Always refer to the latest Avaya SMB Technical Tip or Technical Bulletin for any updated information with regard to Operating Systems, Service Packs or PC hardware
  - Refer to Technical Specifications section of the Product Description for Operating System and Hardware requirements
- A maximum of four SoftConsole applications can be run per system. An IP Office license controls the number of simultaneous SoftConsole users.

10. Voicemail

Voicemail

Voicemail provides a telephone answering machine with a personalized greeting on every employee's desk and allows callers to leave spoken messages when the user cannot answer a telephone call. Voicemail messages are retrieved either locally or remotely via any telephone (users are prompted for a PIN if they are using any telephone other than their allocated extension or a trusted location e.g. mobile telephone).

For users that prefer to have email as their main message store, they can forward their voice messages to their email and collect them via their email account.

The voicemail server is multi-lingual and can offer different prompts depending on the user's preferred language, independently of the default system setup. Similarly, external callers can hear prompts in their own language depending on their incoming call route (e.g. based on caller ID).

Four voicemail options are available:

- **Voicemail Lite**
- **Embedded Voicemail** (IP406 V2, IP Office 500 and Small Office Edition only)
- **Voicemail Pro**
- **Centralized INTUITY Audix / Modular Messaging Voicemail**

Positioning Summary

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

Feature	Embedded Voicemail	Voicemail Lite	Voicemail Pro
Supported IP Office Systems	Small Office Edition, IP406 V2, IP500.	PC Based - All IP Office systems.	
Mailboxes	IP Office automatically creates mailboxes for each user and hunt group on the system.		
Message Storage Capacity	Small Office = up to 10 hours. IP406 V2 = up to 15 hours. IP500 = up to 15 hours.	1MB per minute up to hard disk capacity.	
Maximum Simultaneous Calls	Small Office VoIP 3 = 3. Small Office VoIP 16 = 10. IP406 V2 = 4. IP500 = 4.	4.	Requires licenses: Small Office Edition = 10. IP406 V2 = 20. IP412 = 30. IP500 = 30.
Centralized operation.	No.	No.	Yes.
Queue Announcements	Yes.	Yes.	Yes.
Auto Attendant	Yes.	No.	Yes.
Call Recording	No.	No.	Yes.
Intuity Emulation	No.	No.	Yes.

Voicemail Lite

Voicemail Lite is the IP Office basic Voicemail application and can handle up to 4 simultaneous calls. Each user has the option of turning their Voicemail on or off. When on, the system automatically answers their telephone when they are not available to take a call, plays a personal greeting to confirm that the intended recipient will actually receive the message, and records a message.

When a message has been left, the user will see a message-waiting lamp lit on their telephone and can press a retrieval button to collect their messages.

Voicemail Lite can ring the extension to deliver any new messages. When voicemail messages are left they are time & date stamped and the caller's number noted. Once listened to, old messages are automatically deleted 36 hours after being left, unless the user chooses to save the message permanently.

Voicemail can be collected remotely by calling into the Voicemail Lite server. If the number the user is dialing from is recognized (home number or Mobile/Cell Phone for example), they will listen to their voicemail straight away. 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 listen to voicemail. Users have the ability to set and change their own PIN codes.

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

- Voicemails can be forwarded to another mailbox, or group of mailboxes
- Recipients can add their comment to the voicemail before forwarding to another mailbox or mailboxes.
- Voicemails can be forwarded as email WAV attachments.

Voicemail Lite telephony user interface (TUI) only operates in IP Office mode, not INTUITY mode.

Note: On the IP500, Voicemail Lite is only supported after upgrading to IP Office Professional Edition.

Embedded Voicemail

(IP500, IP406 V2 and IP Office - Small Office Edition only)

In environments like retail or home office, where space, noise or cost considerations rule out using a PC for voicemail, Embedded Voicemail will be the preferred option for an entry-level voicemail service. With the Small Office Edition Embedded Voicemail makes use of the voice compression resources to optimize the message storage by compressing messages before storing, and expanding them during playback. By doing this up to 10 hours of messages can be stored for all users of the system. Neither the IP500 nor the IP406 V2 require voice compression modules for storing messages and both support up to 15 hours of storage.

Key features of Embedded Voicemail include:

- 3 Port voicemail as standard on Small Office Edition (10 ports with 16VC variants of SOE), 4 port voicemail for IP500 and IP406 V2.
- Up to 10 hours storage on SOE, 15 hours message storage on the IP406V2.
- Configurable record time: Default value 2 minutes, maximum value 3 minutes.
- Mailbox security codes ensure a minimum of 4 characters to be set.
- Multiple languages stored on the Flash Memory card.
- Help menus (via *4). Greetings & Mailbox Navigation.
- Voicemail Breakout/Personal Auto-Attendant: Up to 3 breakout numbers can be set up. When callers are directed to your mailbox, they can either leave a message or choose to be transferred to one of three numbers (e.g. Operator, mobile/cell phone, colleague, etc).
- Configurable system-wide short code for Voicemail collect (e.g. *17).
- 4 independent Auto Attendants (AA) with 3 time profiles per AA.
- Up to 12 menu items per Auto Attendant with automatic time-out to fallback number.
- Access and control of voicemail via the digital or IP terminal display (Visual Voice). This feature is supported on the 2410, 2420, 4610, 4620, 4621, 4625, 5410, 5420, 5610, 5620 and 5621 phones.
- Reply to a message to either an internal or external number (if Caller ID available).
- Support for Hunt group announcements.
- Fax option for rerouting fax calls via the auto-attendant menu.
- Support for Fast Forward (#), Rewind (*), Skip message (9) and Call Sender (**) when listening to messages.
- No License Key required.

Voicemail Pro

IP Office Voicemail Pro offers all the features and facilities of Voicemail Lite and can be tailored to meet the individual needs of a business. It has higher call capacity by scaling up from 4 to 30 simultaneous calls. All options are available in a choice of languages; both spoken voice prompts and graphical programming interfaces and have the choice of IP Office TUI and INTUITY emulation TUI.

At the heart of Voicemail Pro is the ability to construct call flows from a series of different building blocks. These building blocks allow automation over tasks like answer a call, listen for tone-dialed digits, make a call etc. Voicemail Pro call flows allow far more than just guiding a user to the group or extension they require. Call flows allow Voicemail Pro to dial back users as soon as a voicemail message is left for them, it provides remote access to phone forwarding settings should a user wish to change their Forwarding or Follow Me number from an external telephone. Voicemail Pro provides message handling for individuals or groups, audio information to callers so assisting the operator during periods of heavy call activity and links to business applications through services such as Text-to-Speech. Voicemail Pro provides a full telephony applications environment where call flows can be set up and interact in real time with business workflow – callers can interact via menus and data entry and Voicemail Pro applications can speak back results. For example, users can listen to their email messages through the telephone

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

Other uses for Voicemail Pro include:

- Whisper Announce that prompts callers for information (usually their name) which is recorded and passed on to the user's extension on answer, 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.
- Conditional routing of calls. 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.
- Call modules. Modules allow you to create sequences of actions that you want to share between a number of different call routing scenarios – like a “macro” in PC applications. 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.
- Activation of the external relays on the IP Office system. 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.
- Finally, a Speaking Clock, that takes its time from the Voicemail server, is built into Voicemail Pro to minimize call charges.

Key features of Voicemail Pro include:

- Voicemail Pro client, a graphical user interface for programming and configuring applications both locally and remotely.
- IVR for individual business requirements.
- Personal Numbering.
- Broadcast group messages.
- Audiotex and Auto-Attendant services (including dial by name).
- Sophisticated Queue Announcement facilities.
- Conditions (e.g. test if 'out of hours').
- Automatic and On Demand Call Recording with an option for ContactStore Search and replay of saved messages
- Voice Forms/Questionnaire Mailboxes (Campaign Manager).
- Personal distribution lists.
- Access to Database information for building Interactive Voice Response (IVR) systems.
- Tag information retrieved from a database to a call and delivers it with the call to an agent.
- Visual Basic (VB) Script support to allow the configuration of the Voice system through VB Scripts rather than Voicemail Pro call flows.
- Extended Personal Greetings to customize the information presented to a caller based upon the availability of a user.
- Text To Speech facilities to allow emails to be read out over the telephone and/or for database information to be read to a caller in 14 languages.
- Housekeeping facilities for the management of messages.
- Automatic detection and routing of Fax calls within Auto Attendants and within a subscriber's voicemail box.
- Forwarding of voicemail messages to Email systems via SMTP.
- Support for a range of the INTUITY telephone user interface features in INTUITY emulation mode.
- Recording of system prompts through the telephone handset or using multimedia facilities on a PC.
- Speaking Clock.
- 22 supported prompt languages: Chinese (Mandarin), Danish, Dutch, English (UK), English (US), Finnish, French (France), French (Canadian), German, Greek, Hungarian, Japanese, Italian, Korean, Norwegian, Polish, Portuguese (European), Portuguese (Brazilian), Russian, Spanish (Castilian), Spanish (Latin American), Swedish
- Support for TTY hearing impaired text phone
- Centralized voicemail within a multi-site IP Office environment.
- Networked Messaging with other Avaya voicemail systems.
- Capacity of up to 30 ports (depending on IP Office Control Unit).
- Voicemail channels between Voicemail Pro and the IP Office can be reserved for business critical functions or left unreserved for any function.
- Access and control of voicemail via the digital or IP terminal display (Visual Voice).
- Improved voice recording, including recording of calls made over IP endpoints (including those using Direct Media); automatic call recording triggered by Incoming Call Routes; pausing recording when call is parked or placed on hold.
- User start points in Voicemail Pro now include Queued and Still Queued options.

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

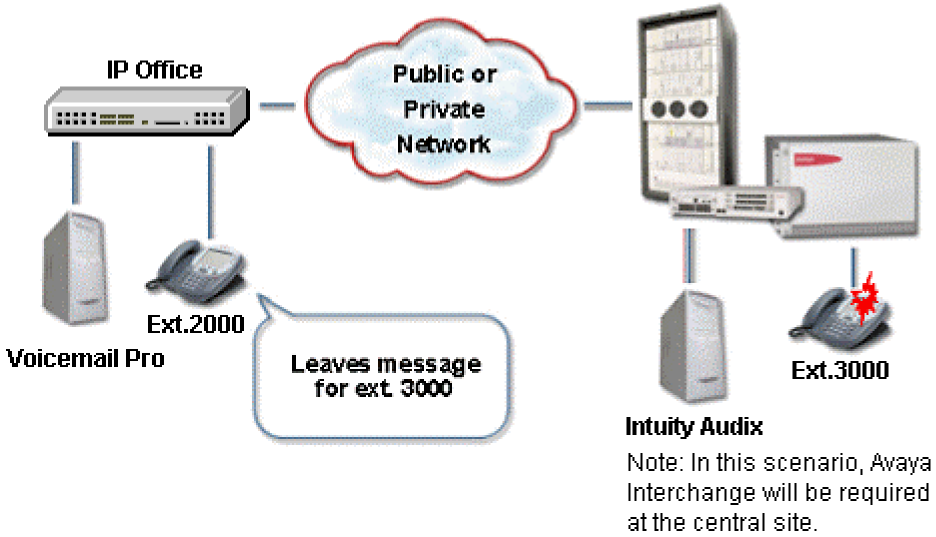
Note: on the IP500, Voicemail Pro is only supported after upgrading to IP Office Professional Edition.

Networked Messaging

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

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

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

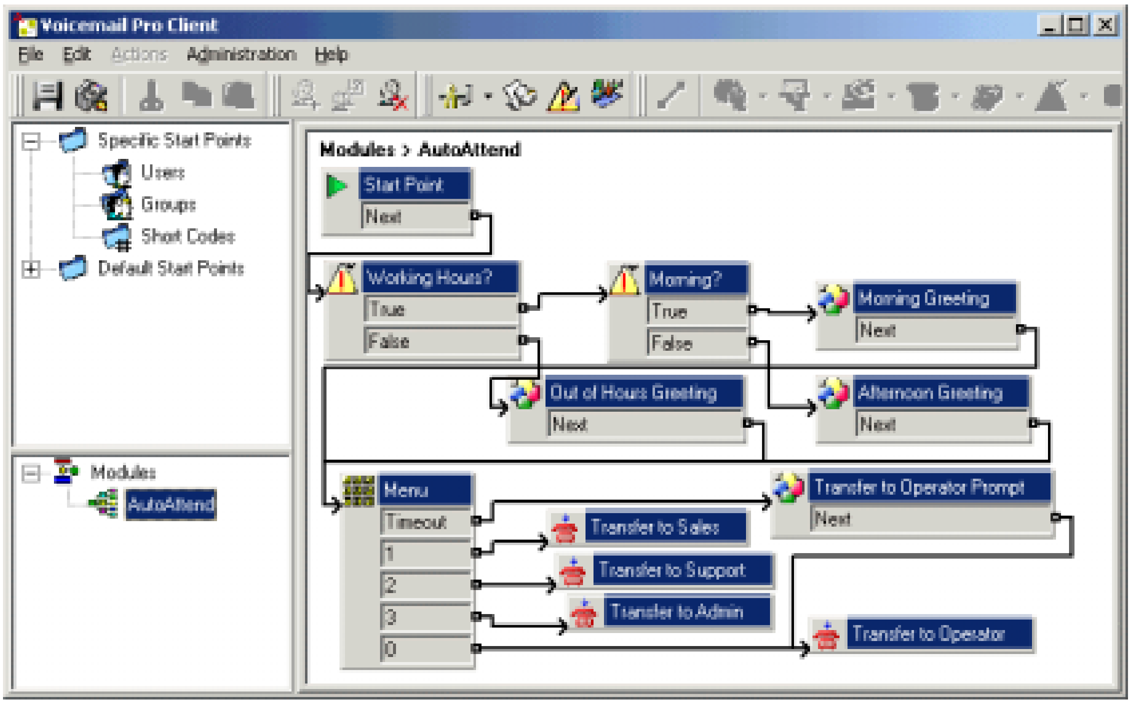


Auto Attendant

Voicemail Pro provides an easy-to-use, multi-level configuration tool (the Voicemail Pro client) which allows network managers and system administrators to construct an interactive menu system, based upon DTMF telephone key entry. This allows an Auto-Attendant system to be built and configured to suit business needs, be that on its own or as a back-up for the regular operator when call volumes are high. Voicemail Pro offers the caller the ability to dial the name of a person via the phone keypad (like “Text” messaging on cell/mobile phones). In response the auto-attendant offers the caller a best match name or if there is more than one, a selection list is offered and the caller can select which one they want 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 name 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 to Direct Dialing (DDI/DID), allowing callers to "learn" the required extension number from the prompting of Voicemail Pro, and then in future dial the extension number, or other pre-defined variables, directly. Auto-Attendant operation is also ideal where multiple languages are required, for example "Dial 1 for English, 2 for German, 3 for French, ...".



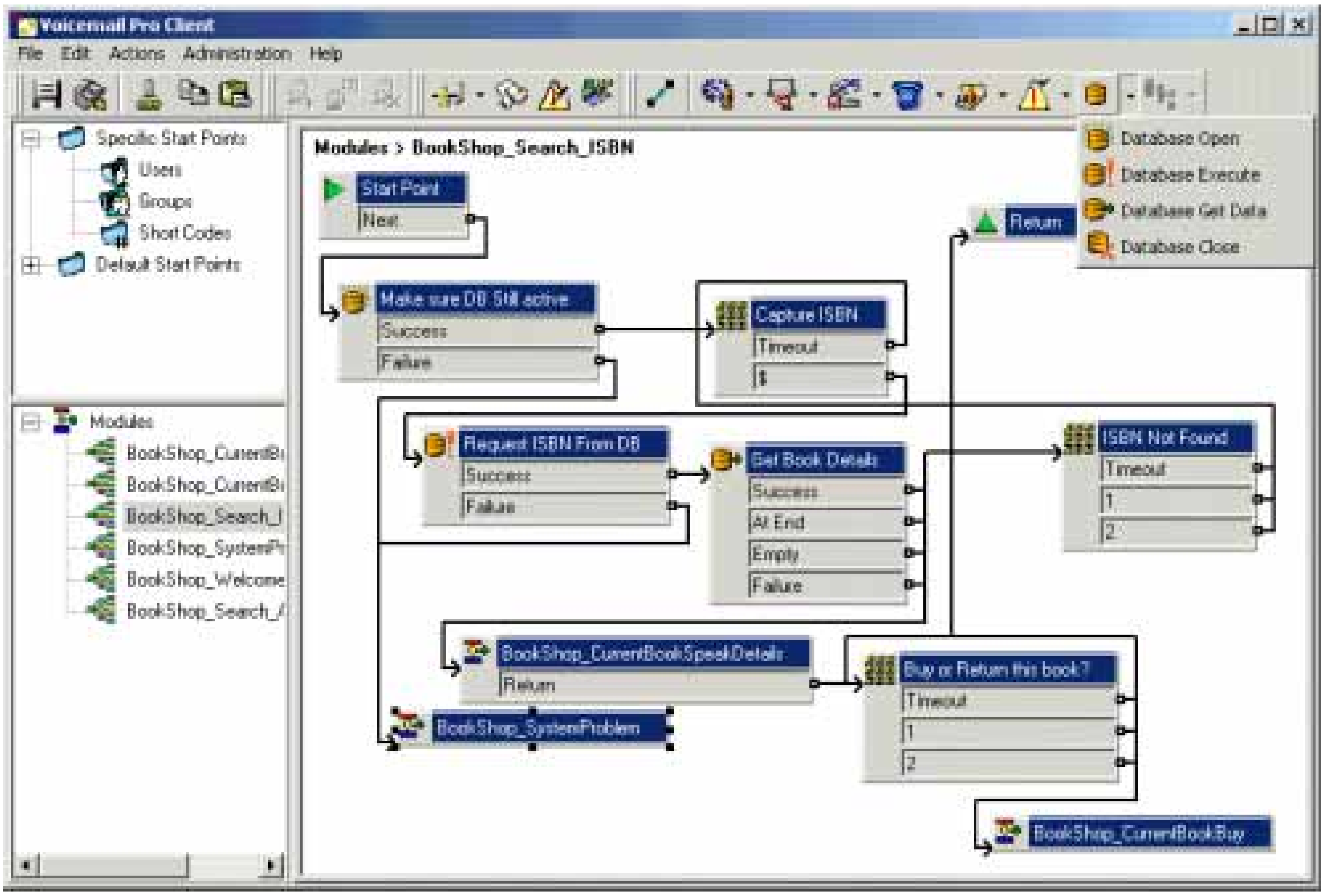
Auto-Attendant created using Voicemail Pro Manager

Accessing Database Information within Call Flows (IVR)

Voicemail Pro 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 using 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 delivered to specifically meet the requirements of the business and the customer experience that is required.

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

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



Example Call Flow Utilizing Database Actions

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

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

As with other Voicemail Pro call flow actions, the database actions include the ability to communicate with the Avaya Compact Contact Center for reporting purposes, the Voicemail Pro installation includes Microsoft Data Access Components (MDAC) to simplify connection to most common databases.

Interaction with the opened database is done through 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 created automatically through the use of a SQL Query Builder Wizard.

Using Text To Speech (TTS) Facilities within a Call Flow

A Text To Speech (TTS) engine can be added to further enhance IP Office IVR capabilities; TTS facilities can enhance the callers experience by allowing the system to read back to them any information that has been extracted from a database. For example, in a Book Shop, the caller dials into the system and is asked for an ISBN number of the book they require. The caller enters the ISBN through the telephone keypad and the system locates the title of the book from the database. As well as finding the title, 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".

The languages currently supported by the Avaya TTS engine are:

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

TTS Licensing

TTS is an optional licensed component of Voicemail Pro, and adds a TTS resource pool for Voicemail Pro to use and release as required. TTS licenses are independent of Voicemail Pro licenses. If a system integrator wants to use a different TTS language set from those supplied by Avaya this can be done by using the 3rd party TTS license instead of the Avaya language TTS. Both license types are based on a concurrent usage model

Visual Basic (VB) Scripting

The Voicemail Pro call flow programming 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 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.

VB Scripting on IP Office Voicemail Pro is an optional licensed component.

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 Voicemail 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.

Users with Mobile Twinning are able to remotely activate their twinning capabilities through Voicemail Pro call flow.

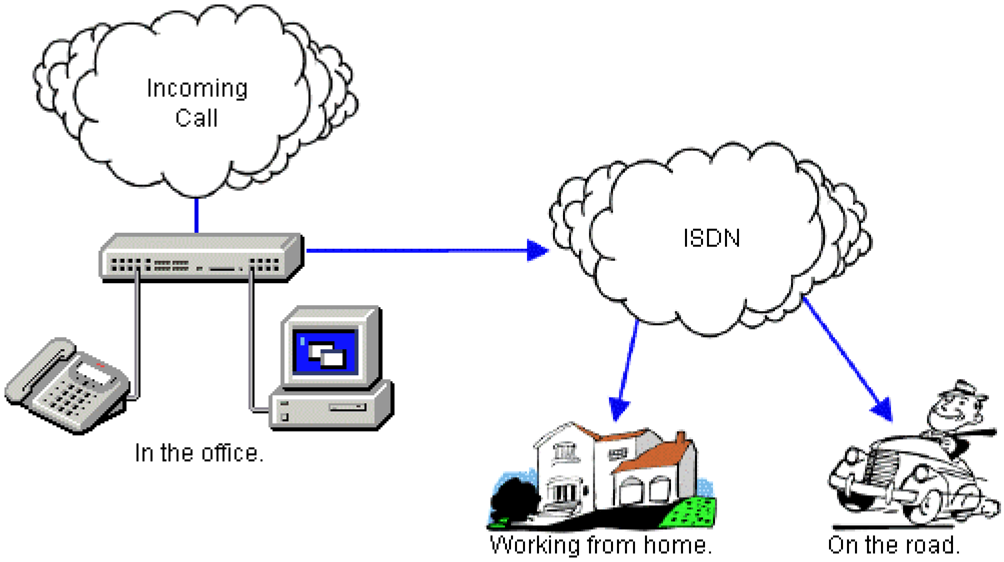


Diagram illustrating personal numbering

Extended Personal Greetings

In INTUITY emulation mode, the Voicemail Pro system has the ability to hold a number of greetings within each user's 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 played back to the caller, based upon the reason the caller was routed to the Voicemail. The supported call states are:

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

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:

1. Out of hours (Hunt group mailboxes only).
2. Internal/External greeting.
3. Busy/Engaged.
4. 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. Phone Manager Pro users can record and manage their voicemail greetings through the Phone Manager GUI

Hunt Group Broadcast Messages

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

- **Hunt group mode**
  Messages are stored in the Hunt Group mailbox and Message Waiting only informs those individuals configured for message waiting indication from that group. This is ideal for scenarios where only a few people such as a call center supervisor need to be initially aware of group messages. Any message waiting light lit by this is extinguished when the new hunt group message is accessed by a user. This is the default mode of operation.
- **Broadcast mode**
  Messages are not stored in the hunt group mailbox. Instead they are broadcast (copied and forwarded) to the individual mailboxes of the entire hunt group membership. This lights the individual messages waiting light of each user of the Hunt Group until they access their mailbox.

Personal Distribution Lists

Personal Distribution Lists are only available with Voicemail Pro when operating in INTUITY emulation mode. The feature provides the ability for a user to distribute a voicemail message to a list of recipients simultaneously. Lists can be configured by a voicemail box subscriber either through their voicemail box telephone user interface (TUI) or through the desktop PC application Phone Manager Pro.

The features available to a voicemail box subscriber include:

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

Cascaded Out-Calling

Voicemail Pro can send a notification, with an escalation capability, that a new voice message has been received in a user's mailbox to specified phone number(s). This is particularly useful in environments such as healthcare and support where important voice messages are left and need to be answered promptly - even outside of office hours.

For example should a patient leave an important message to the main number of the doctor's office, the voicemail system can ring the doctor at the office then on no response escalates to the doctor's mobile/cell phone, his/her home phone or the doctor on duty after a programmable timeout. This avoids having to rely on an external answering service and allows mobile/cell and home phone numbers to remain private.

The voicemail notification can be sent for:

- Any new voice messages
- Any new priority voice messages

Mailbox owners can configure their own options from their handset (Telephone User Interface or TUI)

- Create own Time Profile – defining when notification should take place (e.g. office hours only)
- Out-calling destinations – defining where notification should take place and in which priority order

Five destinations can be defined by the mailbox owner through the TUI (Telephone User Interface). The destinations selected in the escalation list are called in sequence. The possible destinations are:

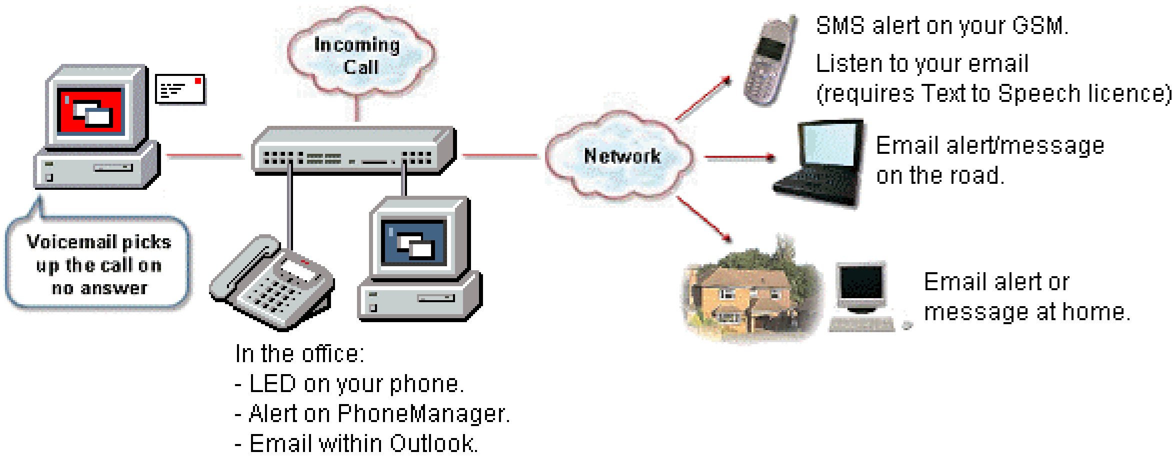
- Desk
- Mobile/Cell
- Home
- Delegate
- Other

Each time an outcall event occurs, each number in the escalation list will be called until either the call is answered, or the end of the list is reached. This process will be repeated on each retry attempt, for the number of retries set.

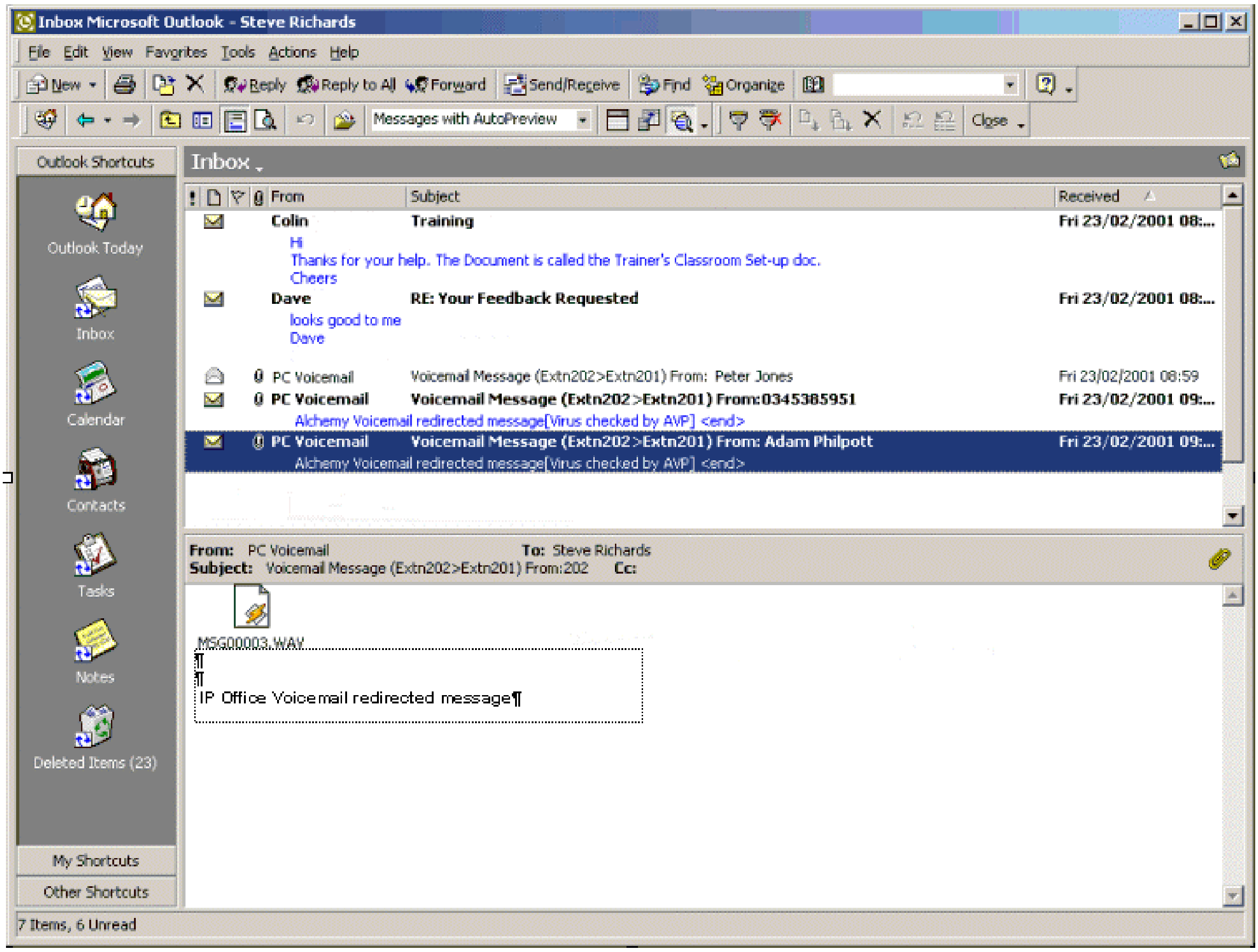
Out-calling preferences are set for global operation via the Voice Mail Pro Client. Out-calling is only available in INTUITY Mode. The administrator sets the number of retries and time interval between retries on a system-wide level.

Interaction of Voicemail with Email Systems

As standard, Voicemail Lite and Pro allow for a simple voicemail alert where the entire voicemail is forwarded (copied) as a .WAV attachment to any MAPI or SMTP compliant Email application. (Microsoft Outlook, Exchange, Lotus Notes, etc.) Forwarding allows emails and voicemails to be unified and collected from a single source. This simple alert option that forwards only the caller's number in the subject of the email, and is ideal for use with commercial Short Message System (SMS) or paging 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 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

Fax Messages

While not directly supplying or supporting fax software, integration with fax to the desktop or client fax applications can be done through the use of 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 is provided for all incoming fax calls. In addition to, or as an alternative to, the Voicemail Pro 'Menu' action or a subscriber's voicemail box (INTUITY mode) can automatically detect any incoming fax calls and then direct the call to a predefined location. The benefit to a business or user is that only one number is required for either voice or fax calls.

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

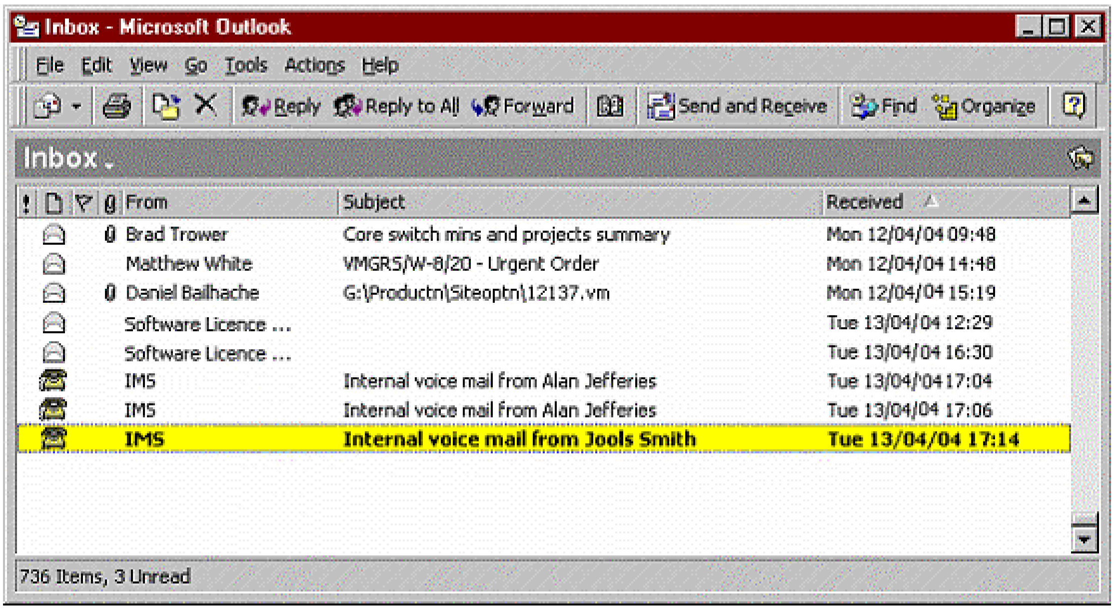
Most fax solutions can be used in conjunction with IP Office, however the following products have been tested and verified to operate in the above scenarios:

- **Equisys - Zetafax**
  Zetafax for Networks provides versatile network fax software solutions for small businesses, corporate offices and distributed enterprise businesses. It enables employees to send and receive faxes at their desktop, without the need to print fax communications, take them to a fax machine and send them manually. Zetafax can be seamlessly integrated into market leading email systems like Exchange allowing users to send and receive faxes directly from their Outlook client. In addition Zetafax can be integrated with other existing applications, such as accounting or CRM systems, for fast, automated faxing from the desktop or back office. Zetafax for networks is already used by more than 60,000 customers worldwide.
  - Further product information available from www.equisys.com
- **Captaris - RightFax**
  RightFax offers a broad, scalable product line that integrates with email, desktop, CRM, ERP, document management, imaging, archival, call center, copier/scanner systems, as well as host, legacy and mainframe applications–virtually all business applications.
  - Further product information available from www.captaris.com
- **Fenestrae – Faxination**
  Fenestrae Faxination Server for Microsoft Exchange integrates fax into email technology. Create faxes on your desktop and deliver them to your chosen fax machine at the click of a mouse.
  - Further product information available from www.fenestrae.com
- **GFI – GFI FaxMaker**
  GFI FAXmaker for Exchange/SMTP allows users to send and receive faxes and SMS/text messages directly from their email client It integrates with Active Directory and therefore does not require the administration of a separate fax user database. GFI FAXmaker integrates via the SMTP/POP3 protocol with Lotus Notes and any SMTP/POP3 server. .
  - Further product information available from www.gfi.com
- **Avaya C3000 *(Germany only)***
  The C3000 can run as a fax server only and be integrated with Voicemail Pro. This variant of C3000 is known as FaxMail Pro.
- **Castelle Fax**
  Faxes routed to a user's mailbox by Castelle fax servers will be recognized by Voicemail Pro as faxes, and will be supported by Voicemail Pro Fax features.

Integrated Messaging Pro (Microsoft Exchange & Outlook only)

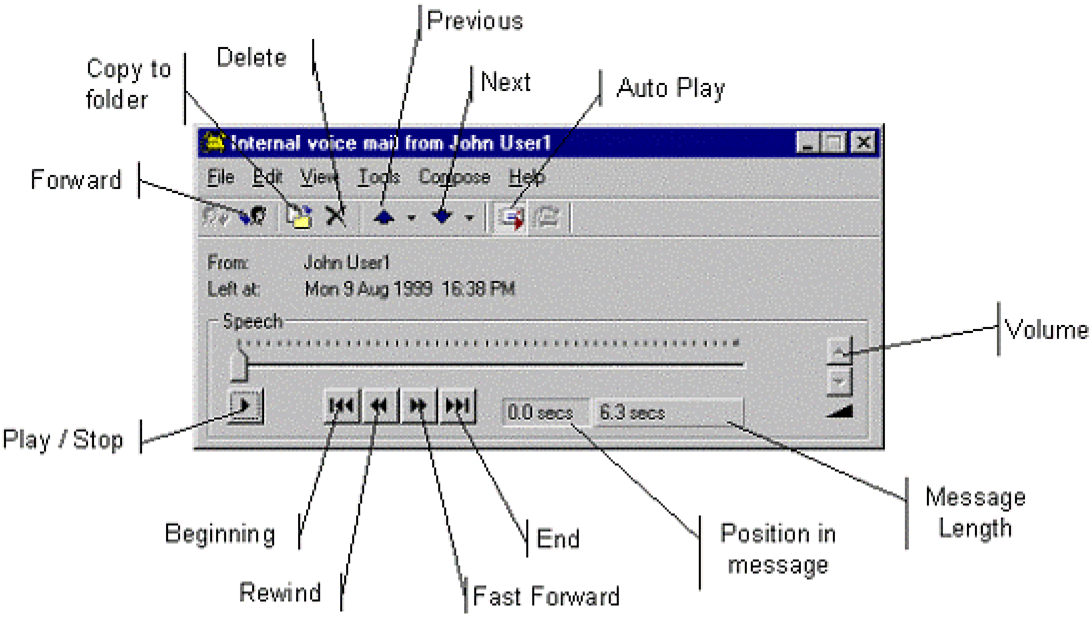
Integrated Messaging Pro (IMS) allows easy management and prioritization of email and voicemail messages through one inbox. This optional application integrates IP Office Voicemail Pro and Microsoft Exchange Server and Outlook client email systems.

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 telephone, at home or on a Mobile/Cell Phone if diverts are set at the desktop. The latter option is useful when working from home or on the road as it avoids downloading large voicemail files for playback on a multimedia PC.



Integrated Messaging Pro user interface

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

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

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

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

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 Email messages through the telephone. When operating in INTUITY mode and with the system licensed for Text To Speech (TTS) facilities the user will be presented with a list of both Voicemail messages and Email messages. The emails can be read out over the telephone in any of the supported 14 languages, based upon the system or user localization settings. The benefit to the user is that their messages are now accessible while in and out of the office through any telephone.

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

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 TTS 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.

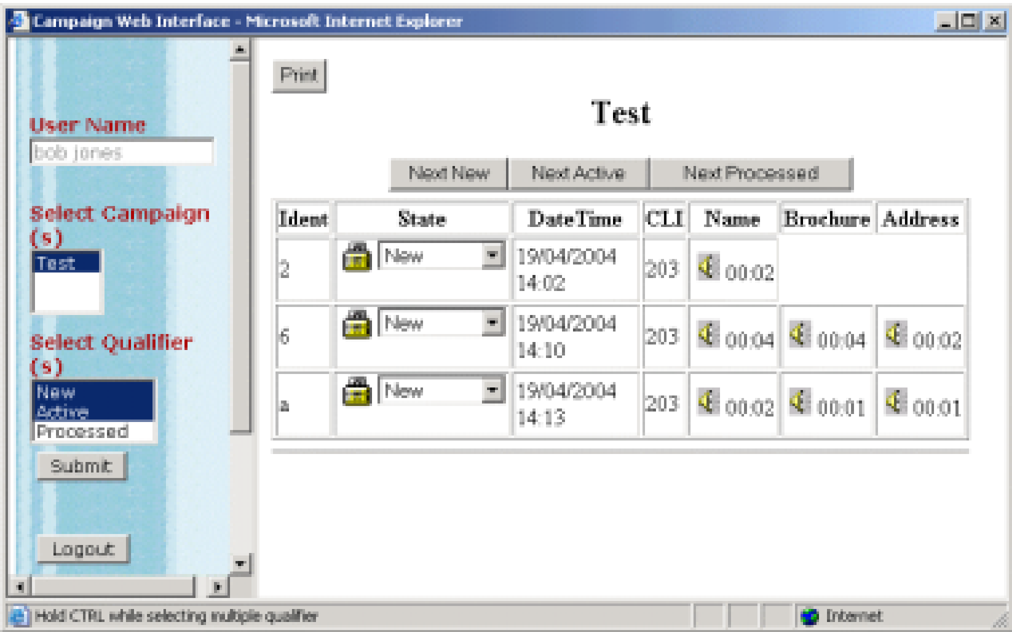
Where the user has email reading in their voicemail box, they will be able to record a voice reply to the email, and send it as a .WAV attachment to a reply email to the person who sent the email.

Campaign Manager

As part of Voicemail Pro, Campaign Manager enables the gathering of repetitive information form inbound calls (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.

In a Contact Center environment, when agents are busy, an overflow to Campaign Manager relieves congestion and pressure on agent groups. An agent can collect the completed transaction via a web browser 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.



Call Recording

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

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

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

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

- **User Recording:**
  The calls to and/or from a particular user can be automatically recorded. By default the recordings are placed in the user's mailbox
- **Hunt Group Recording:**
  The calls to a particular hunt group can be automatically recorded. By default the recordings are placed in the hunt group's mailbox, but there is the ability to select a target mailbox made for or on behalf of a subscriber.
- **Account Code Recording:**
  An account code can be applied to a call by the user before it is made. This can be used to trigger recording of outgoing calls.
- **Caller ID Recording:**
  Account codes can be assigned to a call by Caller ID matching. This allows recording to be based on a Caller ID match.
- **Time Profiles:**
  For each user, hunt group and/or account code, an IP Office time profile can be used to determine when auto-recording is used.
- **Incoming Call Routes**
  Incoming Call Routes can trigger automatic call recording.

Note: It is possible for several recordings to be made of the same call. For example, if both automatic hunt group recording and automatic user recording are applicable to the same call, separate recordings are produced for both the hunt group and the user. Recording only continues while the party triggering the recording is part of the call, for example:

- Recording triggered by a user stops when that call is transferred to another user.
- Recording triggered by a hunt group continues if the call is transferred to another member of the same group.
- Recordings triggered by an incoming call route last until the call is cleared from the system.

Call recording uses the conference facility and so is subject to the conference restrictions of the IP Office system. For some situations, it may be a requirement that call parties are advised that their call is about to be recorded. This is done by switching on the Play Advice on Call Recording option via the Voicemail Pro client. The maximum length of any call recording is 60 minutes

IP Office ContactStore

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

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

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

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

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

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

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

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

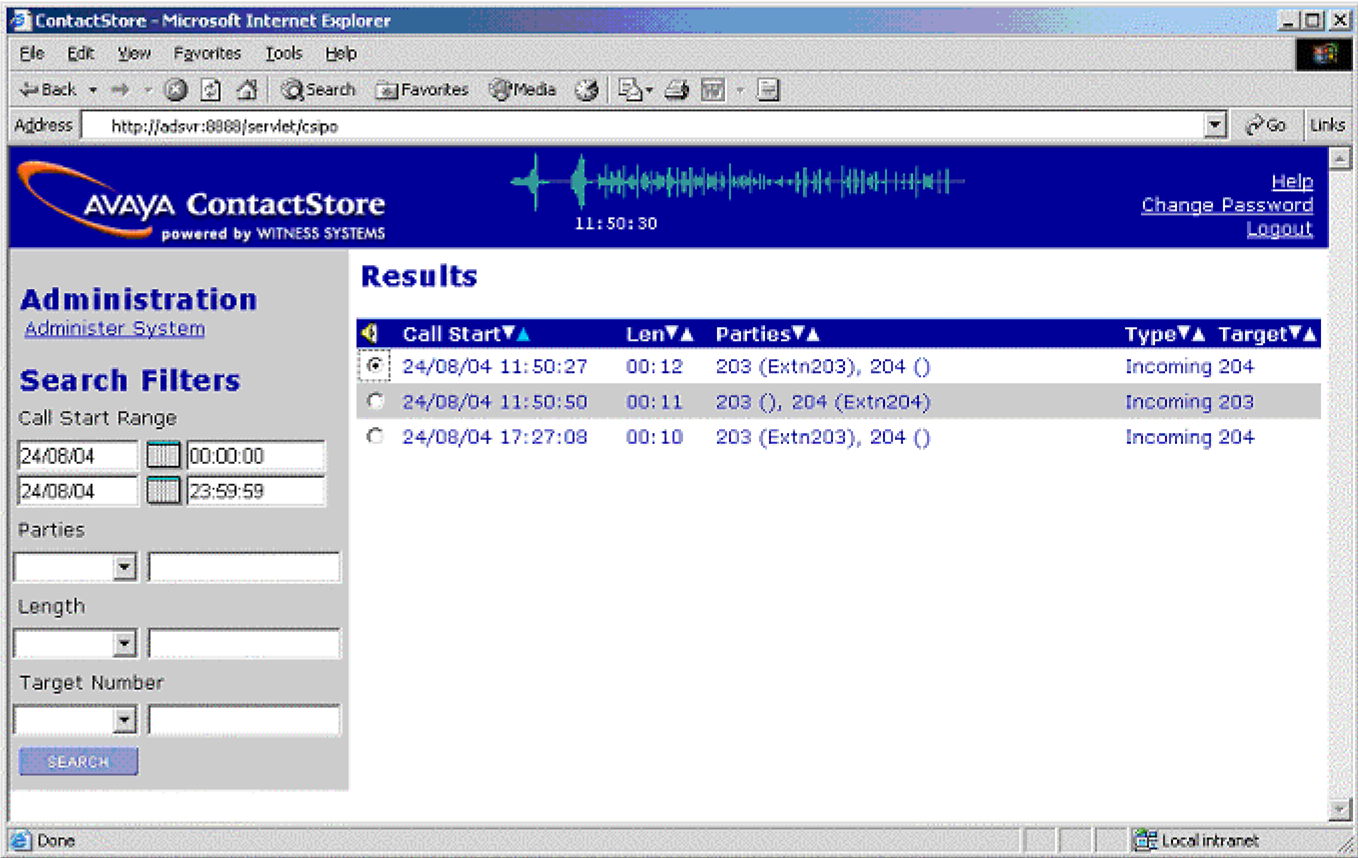
IP Office ContactStore stores recorded calls with certain security in place. Access to recordings is strictly controlled according to the security constraints configured within the System Administration pages. Each recording has an owner; the call owner is the number of the extension that recorded the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

The system will automatically generate alarms showing system warnings. Alarms are logged to IP Office ContactStore's database and held for a month before being purged. The administrator can define specific Email addresses for alarms to be automatically forwarded to. The email recipient could be a local system administrator, a manned help-desk and/or suppliers' support desks if you have a support agreement that includes this facility. The system sends an email message each time an alarm occurs or is cleared. It also sends an email once per day as a "heartbeat" to let you know it is still operating. Failure to receive the daily heartbeat message should be investigated; it could indicate that the server has failed.

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

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

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



Centralized Messaging with Avaya Communication Manager

Where IP Office is deployed in an Avaya Communication Manager (ACM) Environment, it may be desirable to use one centrally managed voicemail system (INTUITY or Modular Messaging) to provide voicemail services to IP Office users. IP Office can be configured to use an INTUITY or Modular Messaging system over a remote connection such that all messaging calls divert to this location and message waiting indications are provided from the remote location and are displayed correctly on IP Office extensions. Connectivity must be either an E1 or T1 circuit or an IP trunk running QSIG services. In addition to the IP Office license Key (Centralized VM with ACM RFA) that enables this service, further license keys may be required on the ACM system.

Voicemail Feature Comparison

Platform Support

	Embedded Voicemail	Voicemail Lite	Voicemail Pro
IP Office - Small Office Edition	Yes (uses in built VCM resources)	Yes	Yes
IP406 V2	Yes (does not use VCM resources)	Yes	Yes
IP412	No	Yes	Yes
IP500	Yes (does not use VCM resources)	Yes*	Yes*

*IP500 running in IP Office Professional Edition mode only.

Capacities

Voicemail	Embedded Voicemail	Voicemail Lite	Voicemail Pro
Number of Mailboxes supported	No specific limit on IP Office - Small Office Edition or IP406 V2. Limited only by IP Office configuration.	No Limit - Limited only by IP Office configuration.	No Limit - Limited only by IP Office configuration.
Maximum Number of Concurrent Calls (ports)	4 simultaneous calls on IP500 and IP406 V2. From 1-10 simultaneous calls on IP Office - Small Office Edition depending up on available VCM resources	4 simultaneous calls on IP Office - Small Office Edition, IP406 V2 and IP412	Up to 30 dependent on license & platform (IP Office - Small Office Edition=10, IP406 V2 =20, IP412=30, IP500 = 30).
Recording Time	IP Office 50 and IP406 V2: Approximately 15 hours IP Office - Small Office Edition: 10 hours minimum	PC dependent (Requires 1MB per minute)	PC dependent (Requires 1MB per minute)

Features

	Embedded Voicemail	Voicemail Lite	Voicemail Pro
Runs as a service	No	No	Yes
Multi-lingual support	Yes	Yes	Yes
Voicemail for Individual users	Yes	Yes	Yes
Voicemail for Virtual users	Yes	Yes	Yes
Voicemail for Hunt Groups	Yes	Yes	Yes
Centralized Voicemail Services	No	No	Yes
Voicemail Ringback	Internal only	Internal only	Internal and external
Voicemail Help TUI	No	Yes	Yes
Message Waiting Indication	Yes	Yes	Yes
Visual Voice (interactive menu on phone display)	Yes	No	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	Yes	Yes	Yes
Forward to Multiple Mailboxes	Yes	Yes	Yes
Forward with a Header Message	Yes	Yes	Yes
Repeat Message	Yes	Yes	Yes
Rewind Message	Yes	Yes	Yes
Fast Forward Message	Yes	Yes	Yes
Pause Message	No	No	Yes
Skip Message	Yes	Yes	Yes
LIFO/FIFO Message Playback Option	No	No	Yes
Set Message Priority	No	No	Yes*
Set automatic message deletion timeframe	No	No	Yes
Alphanumeric Data Collection	No	No	Yes*
Callers Caller ID, time & date announced	Yes	Yes	Yes
Call Back Sender (if Caller ID available)	Yes	Internal only	Yes
Remote Access to Mail Box	Yes**	Yes	Yes
User Definable PIN Code	Yes	Yes	Yes
Known Caller ID PIN Code By-Pass	Yes	Yes	Yes
Breakout to Reception	Internal and external.	Internal only	Internal and external.

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

In-Queue Announcements

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

Auto-Attendant / Audiotex

	Embedded Voicemail	Voicemail Lite	Voicemail Pro
Multi-Level Tree Structure	Yes	No	Yes
Message Announcements	No	No	Yes
Whisper Announce	No	No	Yes
Alarm Calls	No	No	Yes
Assisted Transfers	No	No	Yes

Other Features

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

IP Office Voicemail Pro Intuity Audix Emulation Features

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

PC Requirements

General Requirements

- An IP Office Feature Key is required for Voicemail Pro.
- License for Voicemail Pro and any additional ports required. If Voicemail Pro server is installed without a license it will run for 2 hours and then shutdown.
- License for all options of Voicemail Pro being installed.
- IP Office Voicemail Pro CD.
- Installation on the same PC as being used for IP Office Manager is recommended.
- Switch off any PC and hard disk sleep, power down, suspend, hibernation modes.

PC Specification

- Always refer to the latest Avaya IP Office Technical Tip or Technical Bulletin for any updated information with regard to Operating Systems, Service Packs or PC hardware
- Refer to Technical Specifications section of the Product Description for Operating System and Hardware requirements

Network

- The Voicemail PC must be configured and tested for TCP/IP networking.
- The Voicemail PC must have a fixed IP address.

Disk Space

A compact or typical installation requires 500MB for the Voicemail Pro software. A full installation requires up to 2GB of disk space. However prompts and recorded messages consume an additional 1MB of disk space per minute.

- For Avaya IP Office - Small Office Edition, you can expect to require at least 200 minutes of message recording space, that is 200MB.
- For a busy environment you can expect to require at least 1,000 minutes of message recording space, that is 1GB.

Web Server Operation

If web browser access to campaigns is required Microsoft IIS Web Server must be installed on the Voicemail PC before Voicemail Pro. Both applications must run as a service.

Voicemail Email Connection

Voicemail Email operation is supported using either MAPI or SMTP. MAPI requires the Voicemail Pro server PC to have a MAPI compliant email client install. See Voicemail Email Integration.

If Text to Speech is installed, email text to speech is supported using MAPI.

In both cases above, full email sending from the server PC to users PC should be configured and tested before Voicemail Pro installation using the same PC user account under Voicemail Pro will be installed.

IMS Pro Connection

IMS requires the Voicemail server to use MAPI.

- Integrated Messaging Pro (IMS) is supported on Microsoft Exchange 5.5, 2000 and 2003.
- An Exchange User account for user 'IMSAdmin' will be needed to as part of IMS installation.
- Must be a member of the same Domain as Voicemail Pro Server.
- A list equating Exchange User account names with voicemail box users.

Voice Recording Library Management

IP Office Voice Recording Library (VRL) application is IP Office ContactStore. This application and its installation are documented separately. However:

- Avaya ContactStore for IP Office should be installed after Voicemail Pro has been installed and its operation verified.
- Avaya ContactStore for IP Office must use a separate hard disk partition for its message archiving from that used by Voicemail Pro for current mailbox messages. Use of a separate hard disk or installation onto a separate server PC are alternatives.
- The use of RAID 1 or RAID 5 are recommended.
- The use of a DVD recorder for long-term archiving is recommended.
- A figure of 7.2MB per hour of archived recordings is given.
- The archived messages held by IP Office ContactStore are accessed via web browser using the port address 8888. This port address is not configurable and so it is necessary to ensure that it does not conflict with any other web server service running on the same server PC.

11. 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 employees are planning and working together effectively, and regularly keeping in touch when separated by time and distance? In addition, 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 in decision making 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.

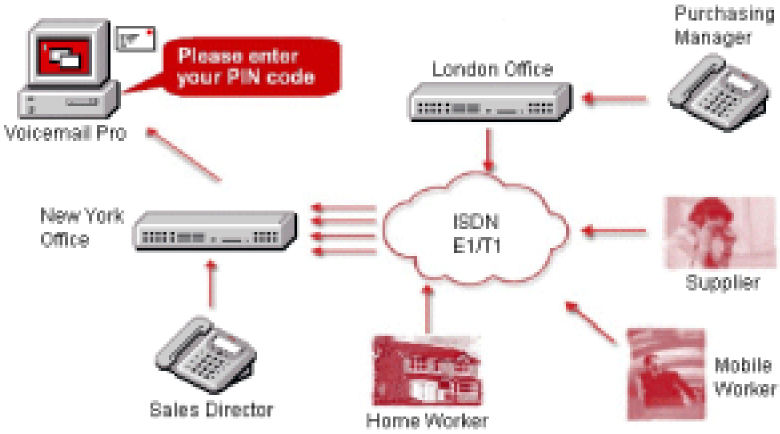
As a result of using conferencing, the benefits gained are:

- 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 Provider conferencing services based upon 2 hourly conferences with 5 participants per week.

IP Office Meet-Me Conferencing Solution

The conferencing solution built-in to IP Office enables multiple callers to talk in an audio conference. Callers can be on-site personnel as well as external parties whether field-based engineers, sales staff on the road, customers or suppliers. Conference calls can be planned in advance or established ad-hoc as and when required.



IP Office Voicemail Pro complements the built-in meet-me conference bridge facility on IP Office systems by adding guidance prompts as well as requesting PIN codes as participants enter the conference for security. For example, if conference calls are regularly scheduled, Voicemail Pro can have pre-programmed Call Flows for weekly conference calls e.g.: every Tuesday between 2pm and 5pm using PIN code 1234 for a sales call, etc. If multiple conference calls are scheduled, users can select which one they want to attend via a simple menu. Should users encounter any issues, calls can be automatically routed to the operator for assistance. For additional security, if Caller ID information is provided by the network Voicemail Pro can make CallerID checks before allowing calls into a conference.

IP Office Conferencing Capacity

IP Office 406 and 412 provide a flexible conferencing solution for 3 to 64 way calling over 64 conference resources or a IP406 or 128 conference resources on IP412. IP Office Small Office Edition provides 2 to 6 way calling with a maximum of 24 conference resources. This means that several conferences of different sizes can all run at the same time if the total calls do not exceed the systems conference resources. IP Office does not impose limits on the mix of internal and external calls in conference, but if all except one call disconnects from the conference bridge, the last calls is disconnected automatically by the system for added security.

Control Unit Conference Capabilities

The following tables show the maximum number of conference parties when calling via the different types of interface available on IP Office:

Maximum Participants	Small Office Edition	IP406 V2	IP412	IP Office 500
E1 ISDN (Rest of World)	6	64	120	64
T1/PRI-T1	6	64/64	96/92	64/64
IP	6	30	60	64
Internal users	6	64	2x64	64
Total max.	**24**	**64**	**2x64**	**64**

Notes:

1. **Analog Trunk Restriction**
   In conferences that include external analog line calls, a maximum of two analog line calls are allowed 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).
5. **Meet-Me Conferencing on IP500 requires Professional Edition**
   IP Office Standard Edition supports 64-way basic conferencing, but if Meet-Me capabilities are required the Upgrade License to IP Office Professional Edition should be purchased.
6. **IP Office Conferencing Center**
   If IP Office Conferencing Center is installed, 5 resources are reserved for use by the system. The maximum number of callers in any one conference and the total number of people on conference calls is reduced by 5. The maximum number of conferences on the system for IP406 V2, IP412 and IP Office 500 is reduced by 2.

IP Office Standard Conferencing Features

The IP Office provides the following features and benefits relating to conferencing:

- **No special conferencing equipment required**
  You only need an IP Office system unit with as many digital trunks/VoIP channels as external participants (as well as Voicemail Pro should PIN code/menu prompts be required).
- **Ease of use**
  Simply dial the direct number allocated to the conference bridge, type in the PIN if required and you have joined the conference (PIN codes require Voicemail Pro).
- **Conference control from IP Office Phone Manager Lite and Pro**
  For ad-hoc conferences with a few participants, users can easily set up immediate conferences by calling all parties and bringing them to the conference bridge. Thanks to IP Office Phone Manager, the instigator of the conference can keep control: the Caller ID 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.
- **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, Caller ID 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**
  Allows 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 is enhanced by adding Conferencing Center. This optional licensed 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 visual presentation web interface.

The scheduler is independent of the web client, either or both can be used. Conferencing Center also interacts with SoftConsole and Phone Manager.

Note: Conferencing Center on the IP500 requires a license for IP Office Professional Edition.

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 conference participants required. The conference is 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 to callers while waiting for the conference to start.

Access to the Web Scheduler requires a user to be granted a user logon and password 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.

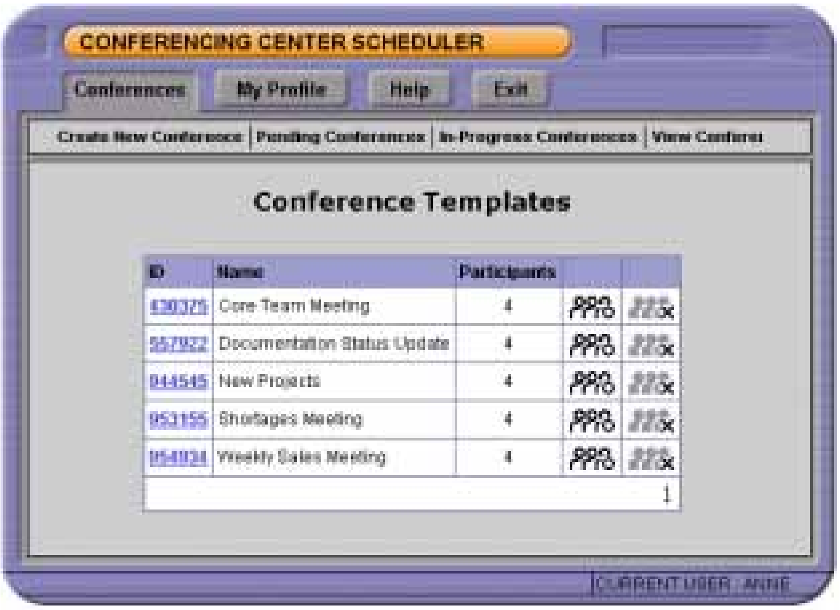
The user setting up the conference can then add participant details including their email address and their telephone number. This allows email notification to all participants confirming the conference call details including the conference name, description, host contact details, bridge number, conference 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). At any time prior to the start of the conference, Participants' details can be changed.

Voice Conferencing Notification (VCN) can 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 caller announcements are required, Voicemail Pro can announce each participant by asking them for their name which is then announced to all participants already on the bridge. Similarly at the end of the conference, each participant leaving the conference will be announced.

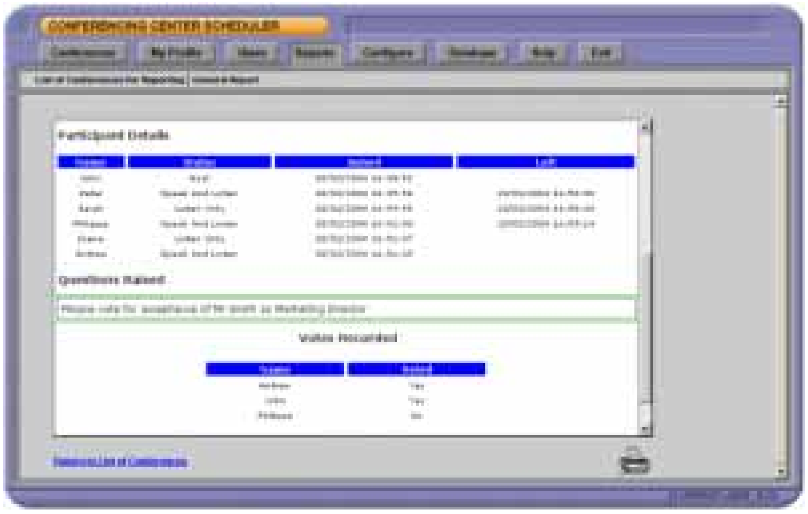
A local address book facility is available to provide a convenient method of managing conference contacts and using these contacts when booking a conference. The address book can be accessed in two ways, either from the 'My Profile' tab or from the Add/Update Conference Participants process.

Conference templates can be used to book recurring conferences, all booking information including the conference ID and participants PINs are retained, except for the conference date. Using a conference template in this way can save re-entering of repetitive information thus saving time and effort. Once a template has been created they can be accessed via the 'My Conference Template' tab:



Conferencing Center Reporting

The System Administrator can generate reports regarding conference usage and individual conference reports. This will detail the conference name and ID, the start date and time, duration and number of participants. If PIN codes were used, individual reports can be run listing participant details and when they joined/left the conference. Finally, 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 share information over the Internet. The Web Client offers a browser 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. A conference host has the ability to pose questions, modify participant speak/listen settings and whisper to a single participant connected into the conference. When in listen-only mode, participants can request the right to speak through their Web Client (raise hand function). A Web Chat service is available between Host and Participants and the dialog is recorded and sent via email to the Host after the conference. Two modes of communication between Host and Participant is supported, either private or public. Public allows all participants to see the dialog

The host can present a document on the Web Client with all participants. (for example a PowerPoint presentation, Word document or an Excel spreadsheet) or simply a website URL. Files can be loaded on demand using the Web Client, or in advance using the Web Scheduler. 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 IP 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 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 (up to 64 maximum per conference). For security, access to the Web Client requires the participant to logon using the Conference ID and their unique PIN number. This 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 participant’s 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.
- Web Chat between Host and Participant
- Whisper facility for the host to have a private conversation with one of the participants.
- Viewing area for reviewing PowerPoint presentations, Word documents and Excel spreadsheets.
- 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 using the speed dials. Voicemail Pro will then contact the participants and bring them to the conference. External participants need to be 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 to a conference created via Conferencing 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 Conferencing Center application by clicking 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 the system administrator and if the Conferencing Center system is installed and available.

System Requirements for Conferencing Center

Conferencing Center Server PC Specification

- Always refer to the latest Avaya SMB Technical Tip or Technical Bulletin for any updated information with regard to operating systems, service packs or PC hardware.
- Refer to the Technical Specifications section of the Product Description for operating system and hardware requirements.

Conferencing Center web client:

- Internet Explorer 6.0 or higher.
- No download required.

12. The Contact Center

IP Office Contact Center/CRM Solutions Overview

Avaya provides Customer Contact solutions that meet the needs of the small to medium business. From the smallest company that requires basic system performance reporting to the larger businesses that need advanced routing and multimedia integration with the Customer Contact Center of up to 75 agents. Avaya provides an appropriate solution on the IP Office communications platform:

- **Compact Business Center**
- **Compact Contact Center**

Compact Business Center

Compact Business Center

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

Key Benefits

- **Lower TCO**
  Provides small businesses with basic 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 Business Center shows a maximum of 4 real time graphs, in any of 6 different graph types e.g. bar, pie, etc. These real time graphs display statistics for either the entire system or any three departments/hunt groups.

Compact Business Center Example

CBC Real Time Information

In order to define the real time graphs the user may select three variables of their choice. The following variables are available:

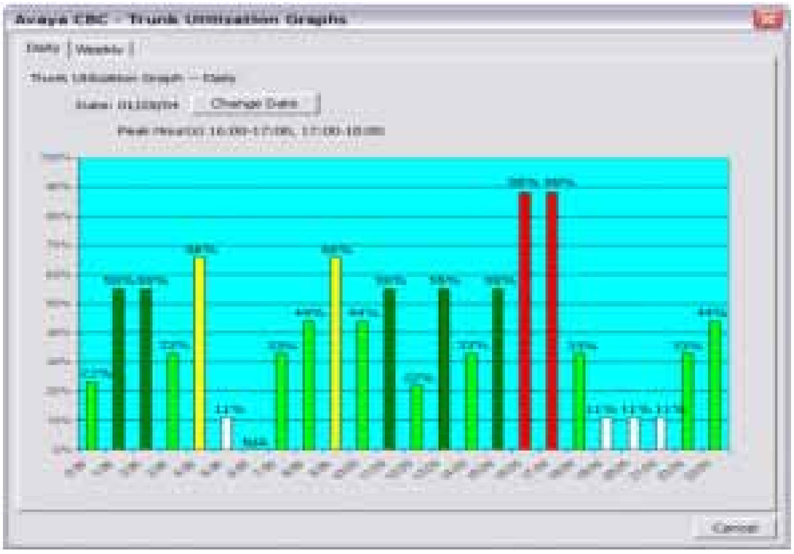
- **Total Calls Presented**
- **Total Calls Answered**
- **Total Calls Lost**
- **Total Outgoing Answered**
- **Active incoming/outgoing Calls**
  The number of calls currently in progress across the entire system highlighting a snap shot view of call activity. This allows the user to have some insight into the balance between agent resource availability and call traffic load.
- **Caller satisfaction level** – the average call wait time before answering
- **Number of available 'Logged-on agents'**
- **Trunk Utilization**
- **Calls waiting**

It is possible to group these variables into two categories i.e. incoming and outgoing calls. These figures can be displayed both in a numerical format and as a percentage of the total calls presented on the incoming side and all variables associated with outgoing side. 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.

Trunk Utilization Graph

With the Trunk Utilization Graph, a business can see hour by hour how much usage there is on trunks, when all trunks are in use and what their busiest times of the day are. It even integrates with the email notification feature described below, so if all trunks in a business become used, key people know immediately.



CBC Alarms & Email Notification

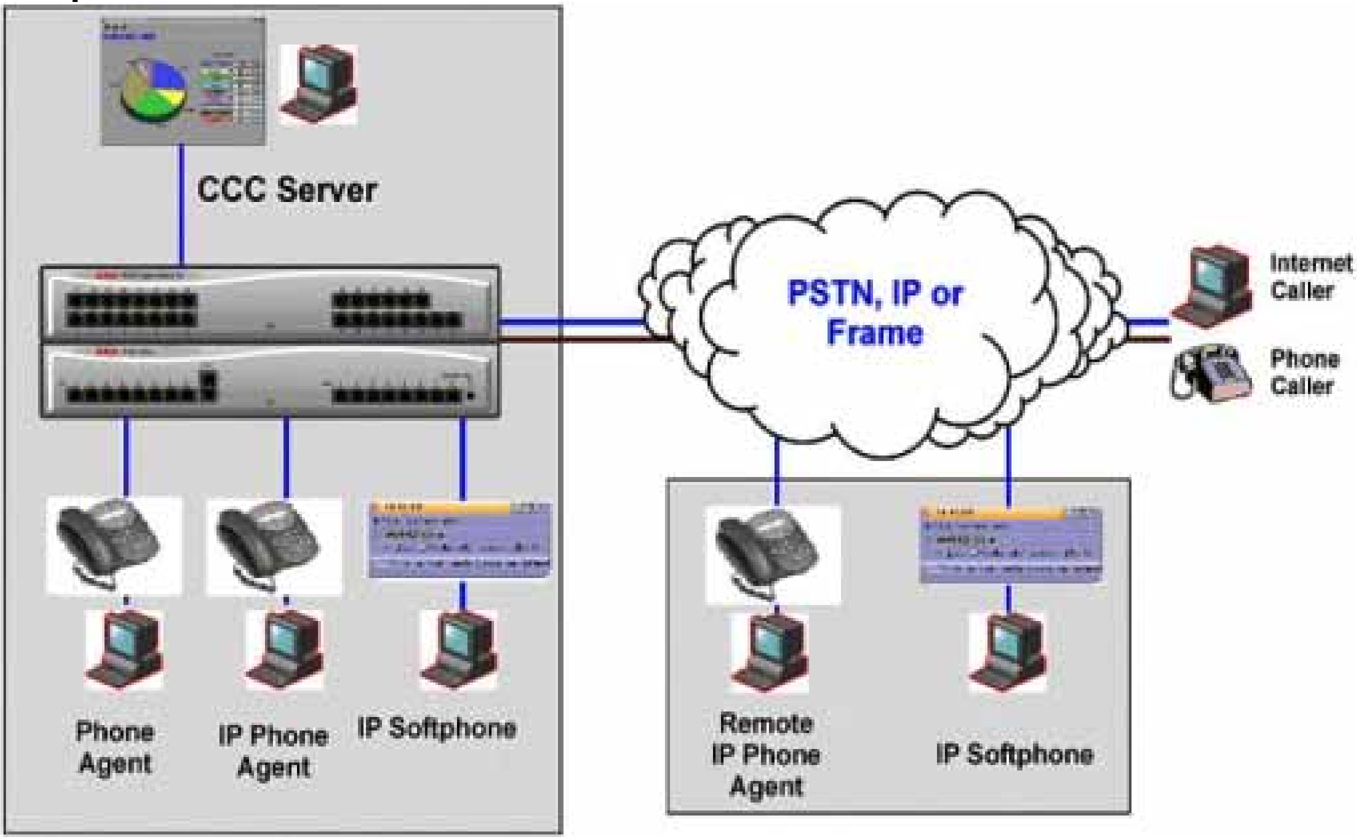
In order to warn the business of developing situations, Compact Business Center provides alarms on the following pre-defined parameters:

- Lost Calls.
- Trunk Utilization (Available Lines).
- Calls Queued.
- Available Agents.

In addition to providing these visual alarms, CBC also provides email notification to key contacts in both the business and the system maintainer, providing up to the minute status on the business. This feature is extremely useful for determining whether an increase in trunk capacity is needed, or more agents need to be logged in to cover call volume.

Compact Contact Center

Compact Contact Center



IP Office Compact Contact Center is a modular contact center solution catering for all contact center sizes from 2 to 75 agents. The following modules are available as part of the CCC software application:

- **Compact Call Center (CCC) Server - Base System**
  Provides one supervisor position with real-time information view, management by exception, and historical reports for any aspect of the contact center. Up to 73 standard reports can viewed or printed. Also included are reporting capabilities on 5 agents and one license for a PC Wallboard (PCWB) application.
- **Agent & Site Management (Real Time)**
  - **Real Time Supervisor Monitoring - Call Center View**
    As many as 21 supervisor CCV positions can be used in CCC (please note: MSDE installations can only be supported up to 5 supervisor positions). This provides a supervisor with the ability to monitor in real time the service being provided to callers. There are up to 12 separate real-time graphs that can be viewed by the supervisor. Alarms also appear in real time prompting the supervisor to acknowledge them as they occur.
  - **Phone Manger Pro: Agent Enabled**
    Provides agents with a PC CTI application where they can log in, join groups, and go into busy status when they are unable to accept calls for short periods so no special turrets are needed – CCC and Phone Manager allow Agent working on any wired IP Office extension type. Phone Manager PC Softphone can be used in agent mode as well, without the need for a physical telephone. Please refer to the applications section for more information on Phone Manager Pro.
  - **Alarm Reporter**
    Alarm Reporter is designed to enhance the exception management used by Call Center View (CCV). The Alarm Reporter enables the contact center supervisor to look back on the performance of the contact center, on a daily or weekly basis, by reporting on certain criteria predefined by the contact center supervisor.

- **Historical Reporting**
  The Compact Contact Center archives all call center interactions (telephony or multimedia) to a central database (MSDE or SQL). This provides the data source for a set of standard reports to the business, and the capability to create custom reports.
  - **CCC Reporter**
    The system allows up to 20 separate Report Viewers within the contact center (for MSDE installations, up to 5 viewers are supported). Access to the standard reports is a thin client application based on Crystal Reports. Up to 73 standard reports are available, with the ability to create 3 more custom reports, see custom reports section below. Reports can be exported to a variety of formats, including Excel, CSV, HTML, and PDF.
  - **Report Scheduler**
    All historical reports created within CCC can be scheduled for individual delivery to anyone via email or sent to multiple network printers.
  - **Custom Reports**
    All CCC reports are created through Crystal Reports™. This application provides a much richer experience for the small to mid-market customer, and creates an environment where custom reporting is more accessible. To create more than 3 CCC custom reports requires the designer license (IPO CCC DESIGNER RFA) AND a compatible version of Crystal Reporting software (Crystal version 9).
  - **Wallboards**
    - **Fixed Wallboards**
      Fixed scrolling wallboards enable key statistics and messages to be displayed for everyone in the contact center to see. Supervisors can send ad-hoc messages to wallboards to broadcast important information, or to make announcements.
    - **PC Wallboards**
      PC-based wallboards allow individual agents to see their own individual statistics, those for their group, or for the whole contact center. Agents can customize their view so that information is presented in the way most useful to them. In additional, supervisors can set particular messages to appear on PC Wallboards, as a motivational or informational tool. Please refer to the CCC System Administration manual for a complete list of variables available.
  - **3rd Party Integration**
    - **Microsoft TAPI Integration**
      By utilizing either the 1st party or 3rd party TAPI support on IP Office businesses can link their contact management to their telephony (e.g. ACT! Goldmine) and increase the productivity of their agents and the profitability of the contact center.

Call Center View - Real Time Reporting

Supervisors in a contact center are there to manage workload. Call Center View provides the Supervisors with the combination of real time service monitoring and resource management, allowing them to balance and manage their resources (i.e. staffing levels against the traffic levels of incoming calls) and therefore improve customer service and reduce costs. Call Center View contains 18 real time screens showing all aspects of the Contact Center activity. Alarms may be set on up to 16 parameters per device, with three levels per alarms available, ensuring that a supervisor will be informed should an exception occur, thus freeing the supervisor to continue with other, more productive activities.

CCV Supervisory Screens

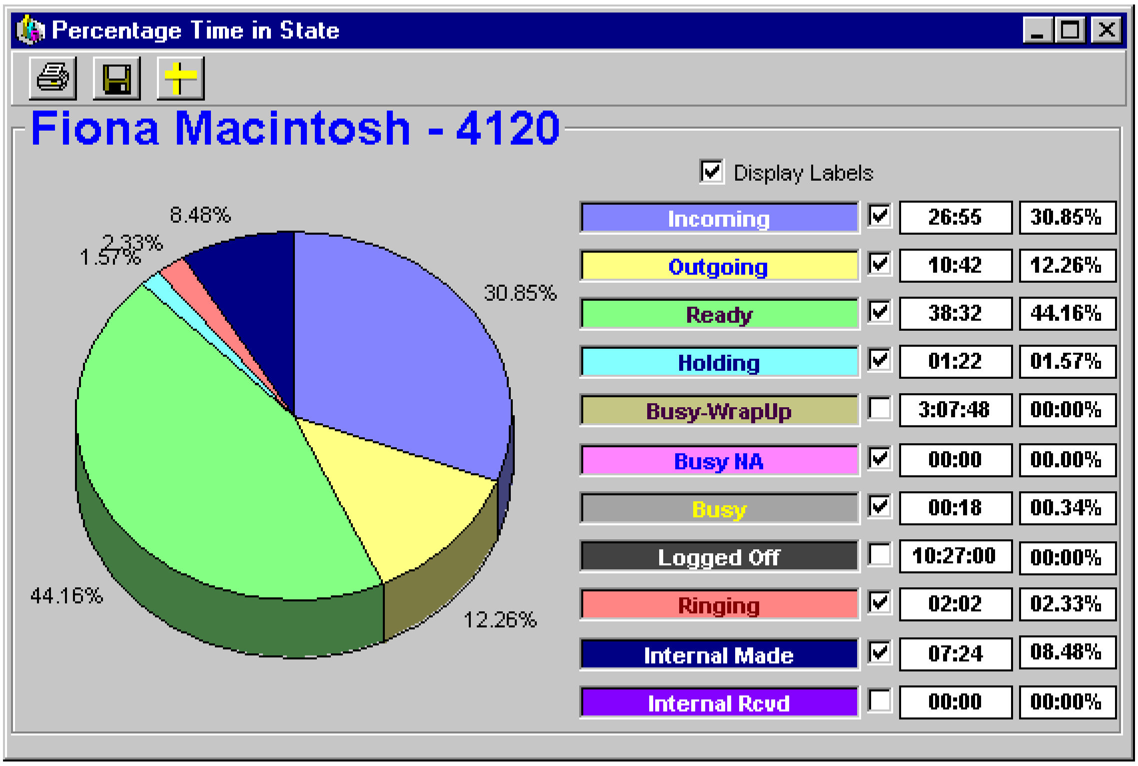
- Alarm Handling.
- BLF Details.
- Extension Activity.
- Callback Request.

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

CCC Reporter - Historical Reporting

CCC Reporter provides in depth historical reporting on the customer facing department's activity. Report Manager provides standard reports for measuring overall contact center call handling and individual/team performance. Data is retrieved from the database, which provides a source of data limited only by the hard disk space available (SQL only). These standard report templates may be formatted by the user to provide reports daily, weekly, monthly, or any defined time period and by individual, group, or trunk. CCC uses Crystal Reports™ format, which provides ease of use and thin client operation for reporting.

Standard Reports List

- Account Code Log by Agent Group (Graphical)
- Account Code Log by Agent Group
- Account Code Log by DDI (Graphical).
- Account Code Log by DDI.
- Account Code Log by Pilot (Graphical)
- Account Code Log by Pilot.
- Account Code Log by Target (Graphical).
- Account Code Log by Target.
- Agent Activity Trace.
- Agent Activity
- Agent Callback Request.
- Agent Group Busy Status.
- Agent Group Graphical Summary (All Calls).
- Agent Group Graphical Summary.
- Agent Group Member Call Duration Report (All Calls).
- Agent Group Member Duration.
- Agent Group Tabular Summary (All Calls).
- Agent Group Tabular Summary.
- Agent Group Tabular.
- Agent Individual.
- Agent Tabular.
- Customer Tracking by Call Identifier.
- Customer Tracking by CLI.
- DDI Call Duration.
- DDI Distribution by Target.
- DDI Distribution
- DDI Response
- DDI Routing
- DDI Summary.
- External Transferred Account Code.
- Incoming Duration Summary.
- Incoming Pilot Summary.
- Lost Call CLI.
- Outgoing Account Code Costing Log
- Outgoing Account Code Log (Graphical).
- Outgoing Account Code Log.
- Outgoing Most Common Destination by Agent Group.
- Pilot Call Duration.
- Pilot Distribution by Target.
- Pilot Distribution.
- Pilot Response.
- Pilot Routing.
- Pilot Summary (All Calls).
- Pilot Summary
- System Summary.
- Target Graphical Summary.
- Target Member Duration (All Media).
- Target Member Duration.
- Transfer Call Tracking Detail by Agent.
- Trunk Group Activity
- Trunk Group Busy.
- Trunk Group Call Duration.
- Trunk Group Response.
- Trunk Group Summary.
- VM Call Flow Monitor by Call Flow Name.
- VM Call Flow Monitor by Topic.
- VM Call Flow Monitor.
- VM Summary
- Incoming Calls By Target Group
- Plus 3 custom reports.

Report Scheduler

Report Scheduler allows reports to be scheduled to run at a specified date and time, or repeated at regular intervals. Supervisors can schedule reports to be delivered to various places within the contact center. Reports can also be delivered to multiple recipients via email in the following formats; PDF, CSV, XLS, RTF, RPT and Word format. Reports can even be scheduled for delivery to multiple printers within the network at the same time.

Custom Reporting

Custom Reporting allows the business to create reports tailored specifically to the needs of the individual business, providing greater flexibility in the presentation of traffic and agent information. This capability is aimed at the contact center manager who wants to take the statistics to a deeper level in order to make better-informed decisions.

Within Compact Contact Center, custom reporting is available, but requires the purchase of Crystal Reports or Crystal Design software from an authorized Crystal/Business Objects software reseller or distributor. With this software, the designer has the ability to create and load 3 custom reports into the CCC Reporter (no additional license required). Custom reports can be added and subtracted as required. If the business requires greater than 3 custom reports, the following license is required:

- IPO LIC IP 400 CCC DESIGNER RFA LIC:CU

Designing Reports Using Crystal Reports

CCC is designed to work with Crystal Reports™ reporting software package (using Crystal version 9). Crystal Reports is available in four different editions to meet the needs of application developers, IT professionals, and business users. The following is an overview of the types of Crystal products that can be used:

Application Development Solutions

- Advanced Developer – Web development and deployment bundle for integrating and deploying dynamic report creation and viewing capabilities into web applications.
- Developer Edition – For integrating report viewing, printing, and exporting capabilities into applications.

Report Design Solutions

- Professional Edition – For report creation and maintenance based on a large variety of data sources plus out-of-the-box web report delivery for workgroups.
- Standard Edition – For basic report design based on PC-based data sources.

The chart below illustrates some of the key feature differences between the various Crystal Reports 9 editions:

		S	P	D	A
Report Design	**DATA CONNECTIVITY**	S	P	D	A
	PC-based and Microsoft® ODBC/OLE DB for MS Access and SQL Server	●	●	●	●
	XML		●	●	●
	OLAP		●	●	●
	Enterprise database servers (ODBC, native)		●	●	●
	Custom, user-defined data through JavaBeans™, ADO, .NET and COM			●	●
	DATA CONNECTIVITY	S	P	D	A
	Visual report designer for rapid data access and formatting	●	●	●	●
	Customizable templates for faster, more consistent formatting	●	●	●	●
	Repository for reusing common report objects across multiple reports		●	●	●
	WEB REPORT DELIVERY	S	P	D	A
	Crystal Enterprise Express for rapid web report delivery (Introductory workgroup offer)		●		
Application Development	**APPLICATION INTEGRATION**	S	P	D	A
	Report viewing APIs (Java, .NET and COM SDKs)			●	●
	Report creation APIs (Java, .NET and COM SDKs) for end user report creation and modification at runtime				●
	Custom Java Tag Library for easy customization of the end user report viewing experience			●	●
	APPLICATION DEPLOYMENT	S	P	D	A
	Crystal Reports Java, .NET and COM reporting components[3] for embedded report viewing, printing and exporting			●	●
	Crystal Enterprise Embedded for offloading report processing from Web Server				●

For more information on how to purchase Crystal Reports products, go to:

www.businessobjects.com/products/reporting/crystalreports

Crystal Reports Training

Training is available from a number of providers; the following is a sample list.

1. Learning Tree International - www.learningtree.com
2. World-Wide Source for Crystal Training - www.crystal-reports.com
3. Stafford Technology - www.crystaltraining.com

Microsoft CRM™ Reporting Integration New for CCC Version 5

Microsoft CRM™ was introduced in January 2003 and has quickly become the premier CRM application for the Small and Medium Enterprise (SME). Avaya and Microsoft are working together to provide a complete CRM, Communications, and Networking solution for any size of business.

In Compact Contact Center Version 5, in conjunction with the introduction of the IP Office Customer Management solution, Avaya has taken this integration one step further by integrating several Microsoft CRM reports with CCC. Supervisors who operate both systems can now drive any of the 73 CCC reports from the MS-CRM interface, and there are 7 combined reports that utilize both systems data to present a 360˚ view of the contact center. The 7 MS-CRM reports are listed below:

- Microsoft CRM Sales Reports
    - Opportunity Activity & Notes
    - Contact Activity & Notes
    - Account Activity & Notes
    - Contact Center Summary by State/Province
    - Contact Center Summary by Zip Code/Postal Code
- Microsoft CRM Service Reports
    - Account Activity & Notes
    - Account Service Report

Wallboard Server/Client

Wallboard Manager

Two types of wallboards are available – traditional wall mounted units and PC based wallboards on the agent's PC desktop. Both types of wallboards are managed from Wallboard Manager/Wallboard Server.

Wall Mounted Wallboards are not available in all territories; please check with your Avaya representative for more information.

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 connected serialy. For those using the Wallboard/22, the communications card is shipped with a single cable able to drive the wallboards.

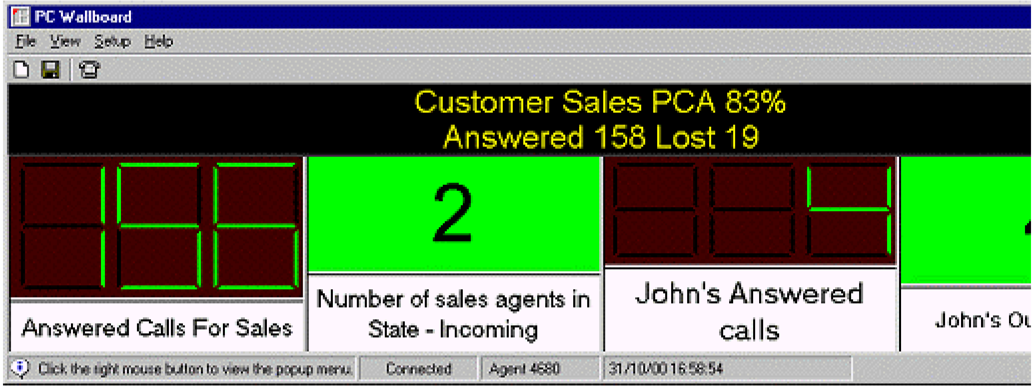
In addition to the physical Spectrum wall-mounted wallboard an IP Office license is required when being used with CCC. This IP Office license supports 4 x Spectrum wall-mounted wallboards. If more than 4 wall-mounted wallboards are required additional license keys must be purchased (each license key supports 4 wallboards at a time). A maximum of 16 wall-mounted wallboards can be supported.

Description	Short code	Material code
Wallboard/22	IND DISP CCM WALLBRD 22 GB	700040173
Wallboard Manager Communications card	IND CP CCM WALLBRD	700038854
IP 400 CCC Wallboard 4 RFA License key required supporting 4 wallboards.	IPO LIC IP400 CCC WALLBRD 4 RFA LIC:CU	176196

PC Wallboard

The PC Wallboard delivers wallboard functionality to the contact center manager and contact center agent's desktop, but with the benefit of each agent being able to configure and monitor a personalized view of the contact center via their own PC wallboard. Supervisors can provide one template for all users in order to standardize the view that agents obtain when starting PC Wallboard.

A CCC agent is able to split their PC Wallboard into twenty (20) different variables that allow different measures of groups and agents in real-time. The data that is presented is identical to that of the physical wallboard. Examples of this are Answered Calls, Longest Call Waiting, Agents logged in, and Lost Calls.

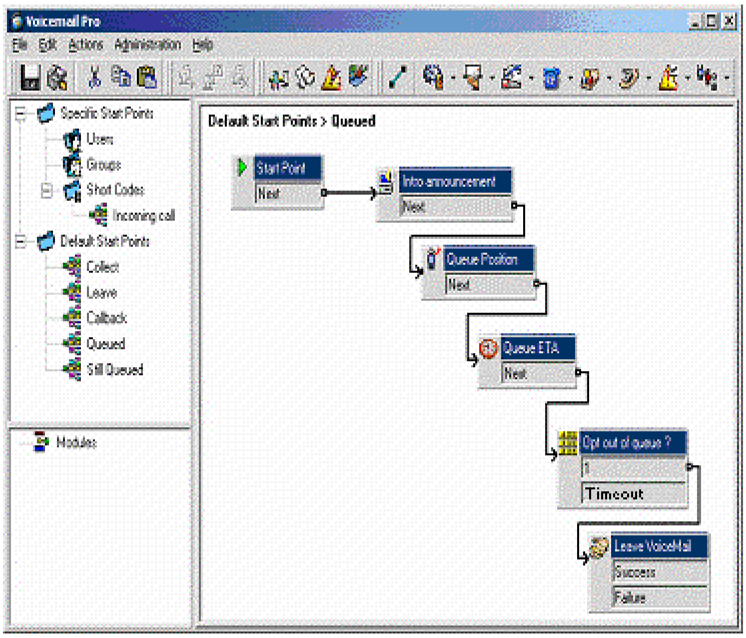


PC Wallboard Example

Queuing Announcements

Voicemail Pro provides system wide messages and announcements programmed by Voicemail Pro call flows. Through call flows it is possible tailor the pre-connection call experience that a customer receives when calling in. By using 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.

The Voicemail Pro application provides Queue Handling facilities, allowing incoming Hunt Group calls to be answered when department, group or individual telephones are busy. Customers entering a queue are played a message informing them of the situation and then hear hold music (internally generated or from an external source), while being regularly updated. Two unique messages may be recorded for each Hunt Group (queue entry and queue update message). Queue announcements can also provide time in system, time in queue, position in queue and estimated time to answer to the caller. It always gives the caller the option to opt out of the queue and leave a message at any time if desired.



CBC / CCC

Compact Business / Contact Center SCBC CCC Summary

Feature	CBC	CCC
Real time screens	1	18
Real time graphs	4	By Group/Agent
Variables	3 of 13	N/A
Reporting period	24 hours	24 hours
Historical data	31 days	Hard disk dependant
Pre-defined reports	None	73
Call Center View	Not available	Included
Report Manager	Not available	Included
Wallboard Manager	Not available	Included
Networked Administrator	Not available	Included
Remote Management	Not available	Via RAS
System (Note: Both systems require Delta Server, see HW requirements).	Windows 2000 Windows XP	Windows 2000 Windows XP
PC Wallboard	Not available	Optional
Report Designer	Not available	Optional
WFM Interface	Not available	Optional
Agents	Not Applicable	75
Supervisor	Not Applicable	21

CCC / CBC Technical Specification

See Product Description appendix Technical Specification section for supported PC operating systems and minimum hardware requirements.

All CCC & CBC applications are based on industry standards and exploit the resilient Windows 2000/2003/XP operating systems and Microsoft's MSDE and SQL technology. Openness and data export are achieved through standard SQL tools and ODBC drivers, as well as a very powerful Report Designer module. This sections sets out the minimum recommended requirements for both the server and client platforms.

- Always refer to the latest Avaya SMB Technical Tip or Technical Bulletin for any updated information with regard to Operating Systems, Service Packs or PC hardware.

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 businesses 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 multiple telephones and gives access to advanced call center operation.

Because IP networking is integrated into 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.

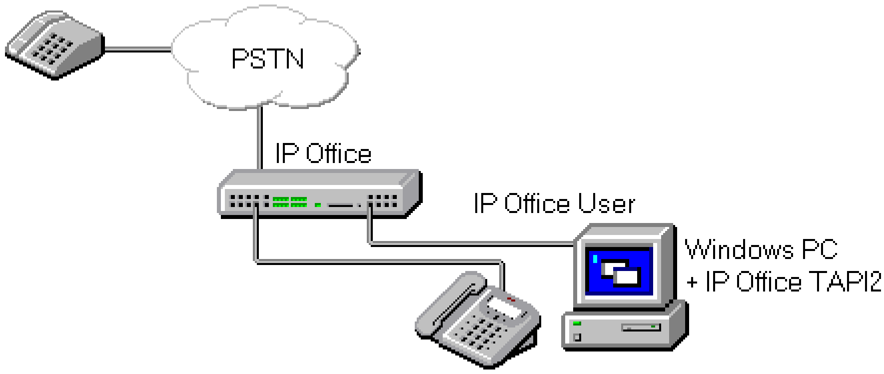
Computer Telephony Integration with IP Office

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

- **TAPILink Lite**
  Provides first-party CTI support for Microsoft TAPI 2.1 and TAPI 3.0, so each PC can control or monitor one handset device. The software components are supplied with the IP Office system on the User CD-Rom, and do not required a license key for use.
- **TAPILink Pro**
  Provides third-party CTI support for TAPI 2.1 and 3.0. These components are identical to their first-party 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. 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. The CTI Link Pro is licensed and enables 4 ports of voice processing; additional ports can be purchased in 4 port increments.
- **DevLink Pro**
  Provides a real-time event stream in addition to the SMDR interface provided in IP Office SMDR. 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. IP Office SMDR is available free of charge, and distributed on the IP Office Admin CD-ROM.
- **Software Development Kit**
  This toolkit is delivered on 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 become familiar with IP Office CTI interfaces.

TAPILink Lite (1st Party TAPI Support)

TAPILink 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)

TAPILink Pro provides all of the features and functionality of TAPILink 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, TAPILink 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

The Developer Connection Program ("DevConnect") is the Avaya developer partner program, and is designed for third-party companies who are creating a product for sale, and who wish to receive technical support. Membership of the program is at the sole discretion of Avaya.

DeveloperConnect members pay an annual fee, for which they receive technical support directly from Avaya. In addition, Avaya will perform interoperability testing between IP Office and the member's product, and may also create opportunities for joint marketing, including exhibitions, use of Avaya's logo, and other benefits.

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

13. CRM Integration

IP Office Microsoft CRM Integration

Introduction

Avaya and Microsoft enjoy a global partnership. Avaya's innovative voice communications and applications based on Microsoft Windows .NET and Dynamics CRM platform, are enabling small medium business to become more effective and profitable. As a Gold Certified Partner and thought leader, Avaya, in partnership with Microsoft, continues to deliver a broad spectrum of technologies that are reliable, scalable and secure.

Avaya – Microsoft Dynamics® CRM 3.0 Integration

The Avaya Microsoft™ CRM Integration Solution allows a business to connect Microsoft Dynamics® CRM 3.0 to Avaya IP Office. It integrates contact points in such a way that will transform the way your business interacts with your customers, this is accomplished by integrating incoming calls directly to the desktop of the user through the use of screen pop technology and by providing outbound dial capability directly from the Microsoft CRM entity.

The Avaya Microsoft CRM Integration Solution requires the following applications to be installed on the Server PC prior to installing the Avaya Microsoft CRM Integration Solution.

- Microsoft Dynamics® CRM 3.0
- IIS 5.0 +

The Avaya Microsoft CRM Integration Solution requires the following applications installed on the Client PC. The client machine will be checked at installation for these components and they will be installed if not found.

- Microsoft .NET 2.0
- IP Office TAPI 2.1 Driver (1.0.0.27)

The Avaya Microsoft CRM Integration Solution is supported on the following client operating systems:

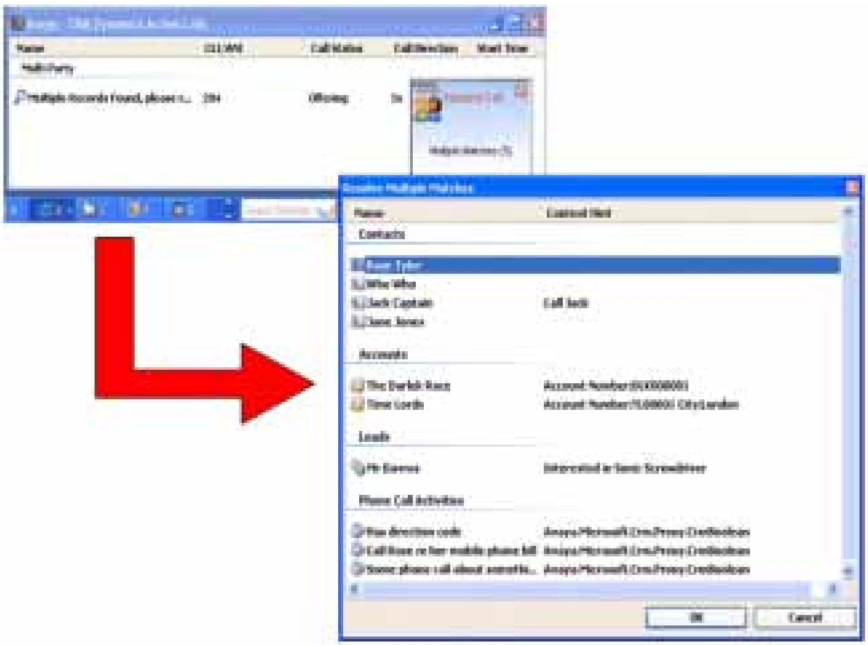
- Microsoft Windows 2000™ Professional
- Microsoft Windows XP™ Professional

Inbound Call Operation

A user can set up their integration to provide inbound screen pops for the following screens within Microsoft CRM™:

- **Contacts**
- **Accounts**
- **Leads**
- **Phone Call Activities**

The user can define what actions to take when an inbound call matches multiple screens, this is accomplished through the use of an Answer Bar that allows the user to select which screen to "pop" into, as identified below:



Outbound Call Operation

Outbound calls are tightly integrated with the Microsoft CRM screen for quick, easy dialing directly from the application.

Customer Benefits

- Link customer information with the touch points used to interact with them
- Handling calls more effectively—reducing and eliminating long hold times, multiple transfers, abandoned calls
- Support employees across the business—everyone working off the same customer information
- Getting calls to the right person at the right time with the right information
- Remembering every customer interaction

14. Common Management Utilities

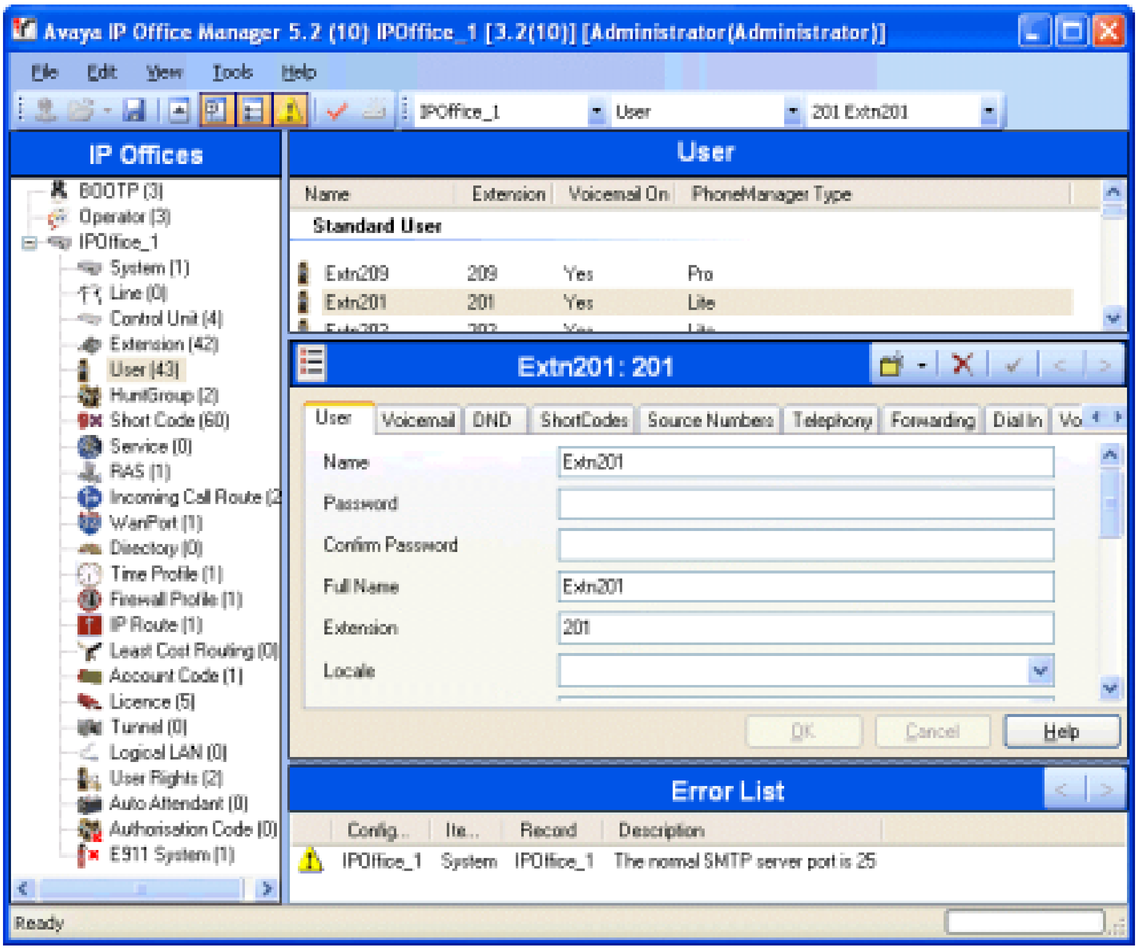
Introduction to IP Office Management Utilities

This section gives an overview of the management applications that are common to all IP Office platforms.

- **IP Office Manager**
  IP Office's main configuration tool.
- **Monitor**
  A trace utility for trouble shooting.
- **SNMP**
  Alerts and alarms from IP Office systems to SNMP tools or to SMTP email.
- **CDR**
  Outputs call detail records direct to an attached printer or separate PC.
- **IP Office SMDR**
  Outputs call detail records for off switch processing.
- **System Status Application (SSA)**
  Outputs call detail records for off A diagnostic tool to monitor and check the status of IP Office systems.

IP Office Manager

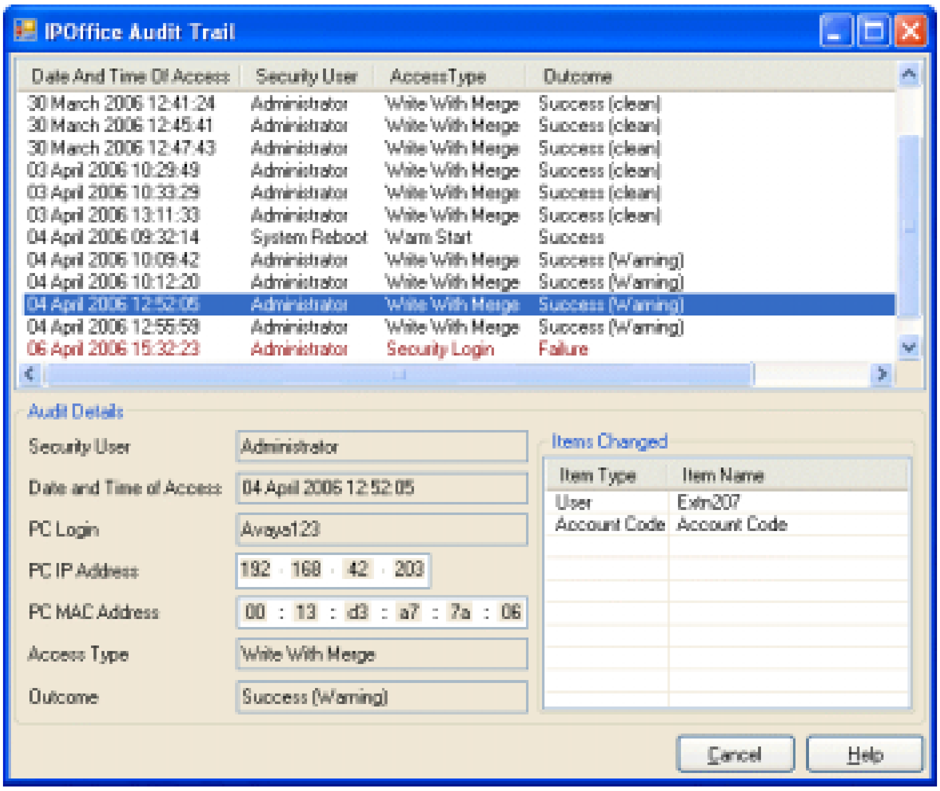
This application is IP Office's main configuration tool. Using a Windows Graphical User Interface, Manager provides an intuitive interface for installation, configuration and subsequent moves and changes. As with all IP Office applications, the Manager is multi-lingual and coupled with the ability to use the application both locally and remotely, it is possible for an administrator to manage any of their IP Offices from any country using their local language preference. Access to each IP Office is protected by passwords and definable user rights. This allows Manager to operate according to the individual administrator's level of expertise.



The IP Office Manager operates on a local copy of the IP Office configuration file. 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.

IP Office has a built-in audit trail that tracks changes to the system configuration, and who has made them. Manager can display the audit trail to assist with problem resolution. The Audit trail records the last 15 changes in the configuration and records the following elements:

- Configuration Changed - For configuration changes, the log will report at a high level on all configuration categories (users, hunt group...) that have been changed.
- Configuration Erased
- Configuration merged
- Reboot – user instigated reboot.
- Upgrade
- Cold Start
- Warm Start
- Write at HH:MM – This is when the administrator saved the configuration via the schedule option
- Write with Immediate Reboot
- Write with Reboot When Free



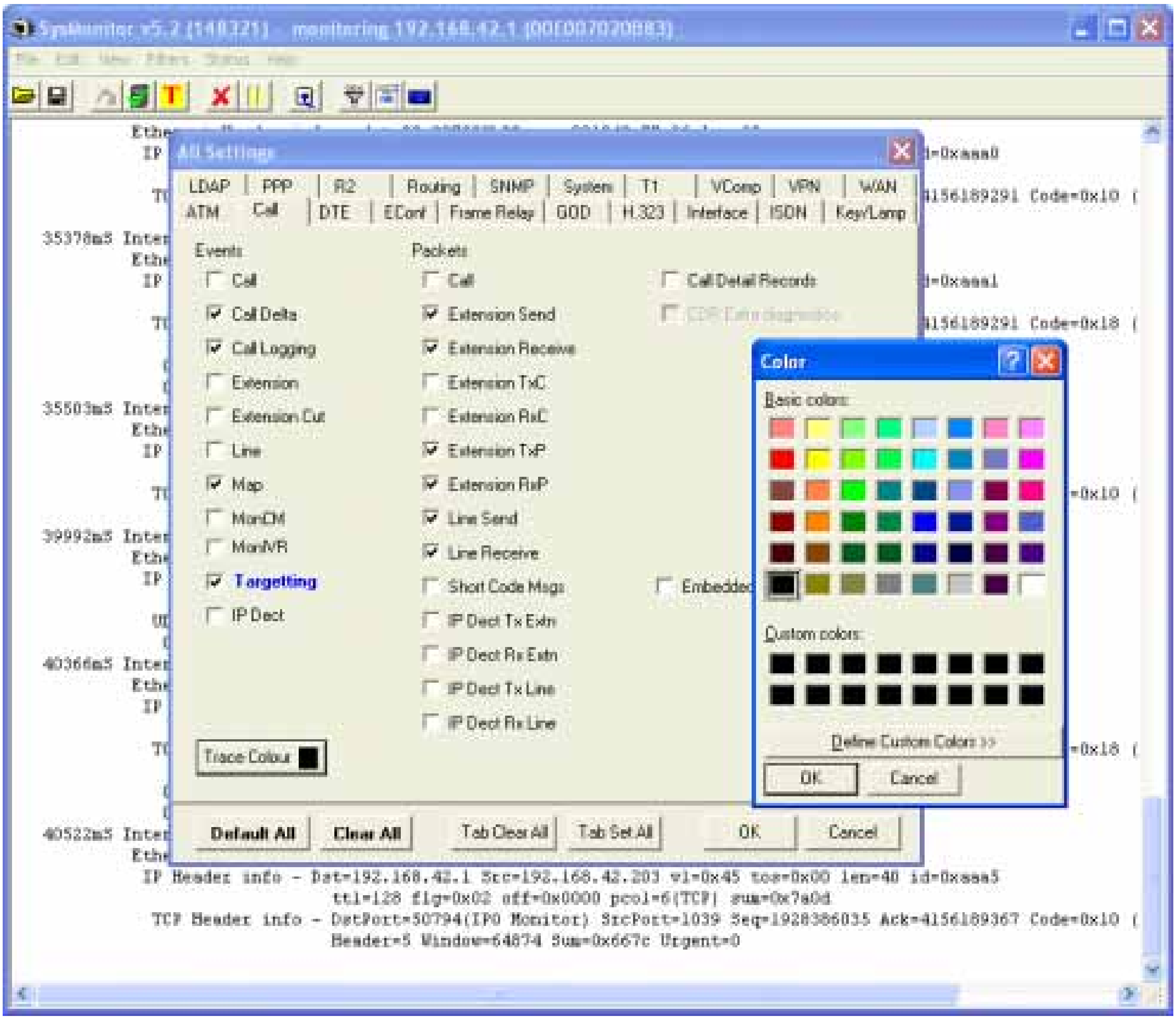
Manager is also used for maintenance functions such as:

- Upgrade to the IP Office system software.
    - Systems running 2.1 or later have the added benefit of being able to send software over an IP network link to a system and have it validated before committing to the upgrade
- IP Office Manager 3.2 is backwards compatible with systems from release 2.1 onwards to allow a single management application.
- Importing and Exporting IP Office configuration information in ACSII-CSV files. Manager will create files for the following data
    - Configuration.csv which is a complete list of items as per Manager 5.1 and earlier
    - Directory.csv containing fields NAME, NUMBER
    - HuntGroup.csv containing fields HUNT GROUP NAME, HUNT GROUP EXTENSION, GROUP, HUNT, ROTARY, IDLE, QUEUING, VOICEMAIL, BROADCAST MESSAGES, EMAIL ADDRESS
    - License.csv is import only containing fields LICENCE OPTION, LICENCE KEY
    - ShortCode.csv containing fields SHORT CODE, TELEPHONE NUMBER, FEATURE NAME
    - User.csv containing fields NAME, EXTENSION NUMBER, USER RIGHT, EMAIL ADDRESS
- User templates for rapid programming and user rights for setting up user access levels.

Monitor

The IP Office Monitor application 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).

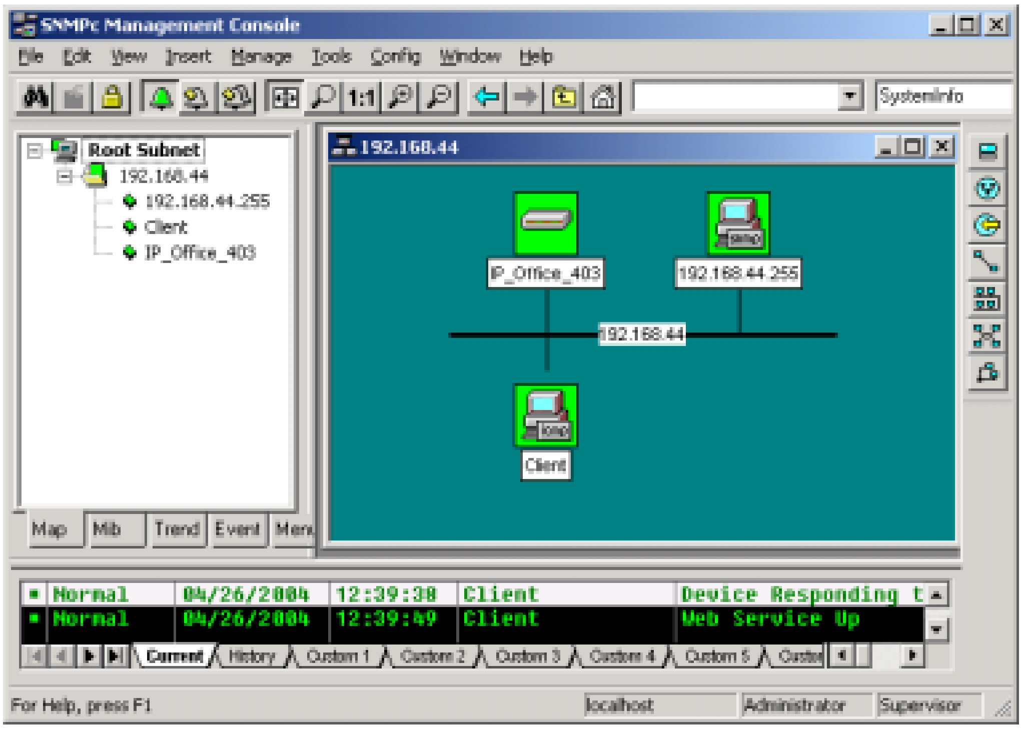
A simple interface allows an engineer to select which protocols and interfaces are to be monitored and decoded. The trace can either be captured directly to screen or as a log file for later analysis. Traces from different protocols can be color coded to improve the clarity of large log files. In addition to monitoring, the application captures system alarms and will display an activity log of the last 20 alarms that have occurred.



Simple Network Management Protocol (SNMP)

SNMP is an industry standard designed to allow the management of data equipment from different vendors using a single Network Manager application. 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 monitors the state of its Extensions, Trunk cards, Expansion Modules (except WAN3 module) and Media cards so that if an error is detected IP Office will notify the Network Manager. IP Office allows two separate Network Managers to be configured so that both a customers Network Manager and a Maintainers Network Manager to be notified of the same alarm condition. As the IP Office solution comprises many applications, the core software notifies SNMP events from both Voicemail Pro and Embedded voicemail to warn of approaching storage capacity limits.

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.



On customer sites where SNMP management is not available, IP Office can email events using up to 3 email addresses each containing a different set of alarms. The following system event categories can be chosen for email notification, if installed on the system:

- **Generic**
- **Trunk lines**
- **Embedded Messaging Card**
- **VCM**
- **Expansion modules**
- **Applications**
- **License**
- **Phone change**
- **CSU Loop-Back**

IP Office sends email notifications directly to the email server; no additional PC client is needed.

CDR

For IP Office customers that choose not to have a separate server for capturing call details (see SMDR below), the system can output Call Detail Records (CDR) direct to an attached printer or separate PC. The records that are detailed by the IP Office CDR are displayed below:

- **Date Records**
  A date record is sent each time a CDR connection is started and then once a day (at midnight). The date can be in month/day or day/month format, as selected on the System | CDR tab.
- **Call Detail Records**
  Call detail records are sent at the termination of a call (in 5 second increments). For some formats, additional fields can be selected using the Normal, Enhanced, or ISDN options on the System | CDR tab.

Depending upon the selected report format and options, there are a number of different fields available within the CDR, they are listed as follows (please review the IP Office Manager documentation for further information):

- Access Code Dialed
- Access Code Used
- Account Code
- BCC (Bearer Capability Class)
- Calling Number
- Calling Number/Incoming Trunk Access Code
- Carriage Return
- Condition Code
- Dialed Number
- Duration
- Feature Flag
- Incoming Circuit ID
- Incoming Trunk Access Code
- Line Feed
- Null
- Outgoing Circuit ID
- Space
- Time

IP Office SMDR

For more formal call logging and reporting, the IP Office SMDR is used by third party applications for many call accounting applications. IP Office SMDR provides much greater details of the call, including duration, ring time, hold time, and transfer information.

IPO SMDR runs as a Windows service included in the Delta Server. The IP Office SMDR application is provided on the Admin portion of the IPO CD/DVD set. It allows the detail of all calls to be sent to a file on the PC, over an IP network to a TCP/IP port, or to a serial port for printing.

Third party applications use this data to allocate costs to departments, analyze trunk capacity, report usage against account codes etc. One IP Office SMDR (Delta Server) is required for each site requiring the use of call accounting software. Please refer to the Technical Specifications section for the Delta Server requirements.

SMDR Diagnostics

Time Of Call Arrival	Call Duration	Ring Time	CLI	Dir.	DDI	DDI	Account Code	Internal	Call ID	More	P1 ID	P1 Name	P2 ID	P2 Name	Hold Time	Park Time
2004/10/19 07:47:07	00:00:00	0	211	O	215	215		1	6	0	E215	Extn215	E215	Extn215	0	0
2004/10/19 07:47:07	00:00:00	0		O				1	100	0	E-1	No Name			0	0
2004/10/19 07:46:56	00:00:10	0	215	I	215	215		0	6	0	V9551	CO Channel 1	E215	Extn215	0	0
2004/10/19 07:46:54	00:00:09	1	211	I	369	369		0	7	0	V9551	CO Channel 1	E211	Extn211	0	0
2004/10/19 07:46:56	00:00:07	0	211	I	9551	9551		0	7	0	V9551	CO Channel 1	E369	Extn369	0	0

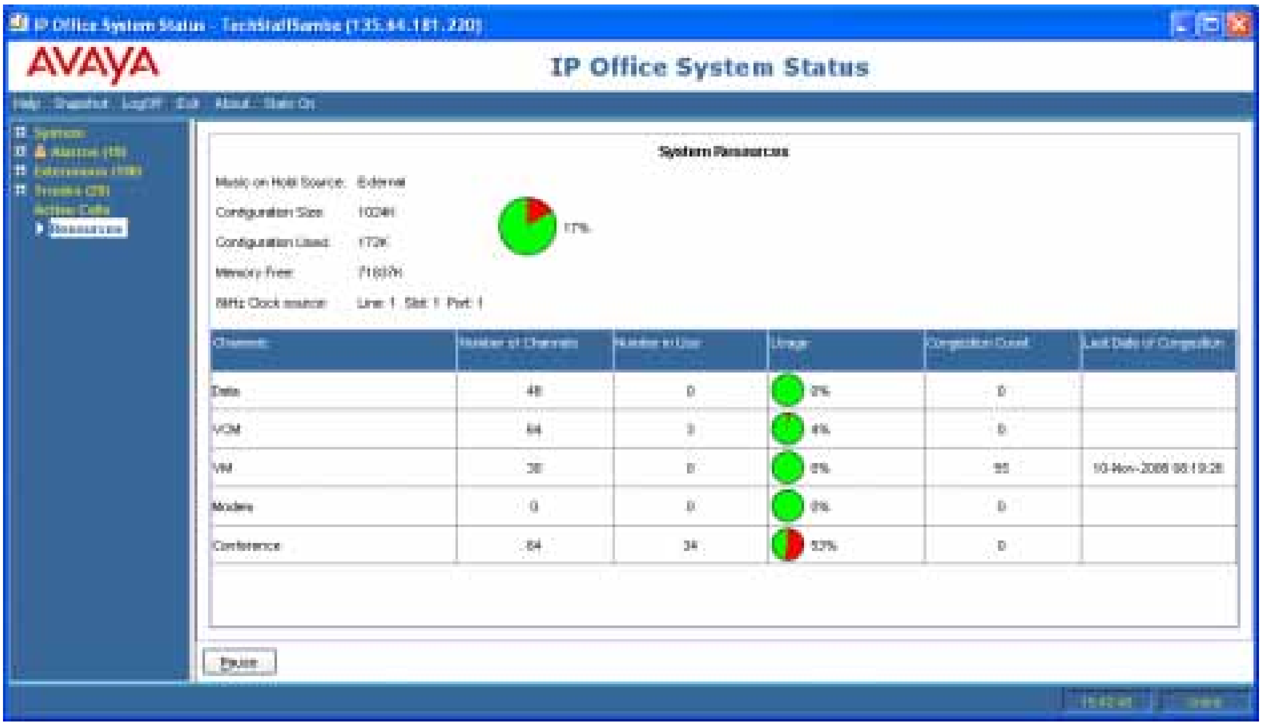
Sample IP Office SMDR Information Output

System Status Application

The System Status Application (SSA) is a diagnostic tool for system managers and administrators to monitor and check the status of IP Office systems locally or remotely. SSA shows both the current state of an IP Office system and details of any problems that have occurred. The information reported is a combination of real-time events, historical events, status and configuration data to assist fault finding and diagnosis. SSA provides real-time status, historic utilization and alarm information for ports, modules and expansion cards on the system. SSA connects to all variants of IP Office running release 4.0, using an IP connection that can be remote or local. Modem connections at 14.4kbps or above are supported for remote diagnostics.

SSA provides information on the following:

- **Alarms**
  SSA displays all alarms which are recorded within IP Office for each device in error. The number, date and time of the occurrence is recorded. The last 50 alarms are stored within IP Office to avoid need for local PC.
- **Call Details**
  Information on incoming and outgoing calls, including call length, call ID and routing information.
- **Extensions**
  SSA details all extensions (including device type and port location) on the IP Office system. Information on the current status of a device is also displayed.
- **Trunks**
  IP Office trunks and connections (VoIP, analog and digital) and their current status are displayed. For VoIP trunks, QoS information is also displayed (e.g. round trip delay, jitter and packet loss)
- **System Resources**
  IP Office includes central resources that are utilized to perform various functions. Diagnosing these resources is often critical to the successful operation of the system. This includes details on resources for VCM, Voicemail and conferencing.



SSA can be launched independently or from IP Office Manager and there can be up to two (2) SSA clients connected to an IP Office unit at one time.

Note: SSA is not a configuration tool for IP Office systems. For information on configuration, refer to IP Office Manager

A: Configurations

Product Configurations

Small Office Control Units

All Small Office Edition control units include twin PCMCIA slot for embedded voicemail and wireless access point options, four port Ethernet switch, single Ethernet WAN port and a slot for optional V24/V35/X21 or T1 WAN option modules.

- **Avaya IP Office Small Office Edition - 4T+4A+ 8DS (3 VC) US *(700350424)***
  Providing four US specification analog trunks, four analog extensions and eight Digital Station ports. Complete with three voice compression resources as standard for VoIP applications.
- **Avaya IP Office Small Office Edition - 4T+4A+8DS (3 VC) INT *(700280209)***
  Providing four analog trunks (not US), four analog extensions and eight Digital Stations. Complete with three voice compression resources as standard for VoIP applications.
- **Avaya IP Office Small Office Edition - 4T+4A+8DS (16 VC) US *(700350432)***
  Providing four US specification analog trunks, four analog extensions and eight Digital Station ports. Complete with sixteen voice compression resources as standard for VoIP applications.
- **Avaya IP Office Small Office Edition - 4T+4A+8DS (16 VC) INT *(700280217)***
  Providing four analog trunks (not US), four analog extensions and eight Digital Stations. Complete with sixteen voice compression resources as standard for VoIP applications.

Avaya IP Office - Small Office Edition Expansion Cards

- **Avaya IP Office Small Office Edition - WAN Expansion Kit *(700289713)***
  Optional card for connection to private circuits and network terminating devices with V.24, V.35 and X.21 interfaces.
- **Avaya IP Office Small Office Edition - Embedded Voicemail *(700289721)***
  PCMCIA format memory card with embedded auto-attendant and voicemail applications installed.
- **Avaya IP Office Small Office Edition - Wireless LAN Card *(700289739)***
  PCMCIA Wireless card providing IEEE 802.11b Access Point functionality when used with IP400 Access Point RFA license.

IP406 Control Units

Includes: 8 x Digital Station ports, 2 x analog station (POTS) ports, 1 x compact flash slot for embedded voicemail option, 8-port Layer-2 LAN switch, 9-pin DTE serial port for license feature key and system diagnostics, 37-pin WAN port, 3.5 mm jack for Music-on-Hold audio input and 2-switch external door-relay control port. Internal expansion slots to support 1 x 12-port remote access modem module and 1 x Voice Compression Module (up to VCM30 for non-blocking IP/PRI applications). 6 x external expansion module ports to support additional analog trunks, WAN interfaces, digital or analog extensions. Includes 60W earthed external power supply. Regional power cord and software/documentation CD pack not included.

- **IP406 V2 Office Mu-Law *(700359946)***
  Mu-law voice encoding base unit pre-configured for US locale settings. 2 x trunk module slots to support US T1 PRI and 4-port analog trunk cards.
- **IP406 Office V2 A-Law *(700343536)***
  A-law voice encoding base unit pre-configured for multi-country locale settings. 2 x trunk module slots to support Euro-ISDN BRI, E1/PRI and 4-port analog trunk cards.

IP412 Control Units

Includes: 2-port Layer-2 LAN switch, 9-pin DTE serial port for license feature key and system diagnostics, 37-pin WAN port, 3.5 mm jack for Music-on-Hold audio input and 2-switch external door-relay control port. Internal expansion slots to support 1 x 12-port remote access modem module and 2 x Voice Compression Modules (including VCM24 and 30 for non-blocking IP/dual-PRI applications). 12 x external expansion module ports to support additional analog trunks, WAN interfaces, digital or analog extensions. Includes 60W earthed external power supply. Regional power cord and software/documentation CD pack not included.

- **IP412 Office Mu-Law Base Unit *(700350408)***
  Mu-law voice encoding base unit pre-configured for US locale settings. 2 x trunk module slots to support US T1 PRI and 4-port analog trunk cards.
- **IP412 Office A-Law Base Unit *(700234479)***
  A-law voice encoding base unit pre-configured for multi-country locale settings. 2 x trunk module slots to support Euro-ISDN BRI, E1/PRI and 4-port analog trunk cards.

IP500 Control Unit (700417207)

Includes: 4 x front slots for combinations of extension/VCM cards and trunk daughter cards, 1 x smart card slot for locale settings and license feature key, 1 x compact flash slot for embedded voicemail option, 2-port Layer-3 LAN switch, 9-pin DTE serial port for system diagnostics, 3.5 mm jack for Music-on-Hold audio input and 2-switch external door-relay control port. 8 x external expansion module ports to support additional analog trunks, digital or analog extensions. Includes auto ranging internal power supply. Regional power cord and software/documentation CD pack not included. Only one variant of control unit is available, but regional locale is determined by the appropriate smart card feature key (mandatory):

- **IP Office 500 Software License Feature Key Mu-Law *(700417470)***
  Configures the control unit for Mu-law voice encoding and US locale settings.
- **IP Office 500 Software License Feature Key A-Law *(700417488)***
  Configures the control unit for A-law voice encoding and multi-country locale settings.

IP Office External Expansion Modules

Except where noted, all the following are supported by the IP406 V2, IP412 and IP500 control units. Note that external expansion modules are only supported by the IP500 when running in IP Office Professional Edition mode.

- **Phone 8 Module V2** ***(700359896)***
  Adds an additional 8 analog Plain Ordinary Telephone ports to control units.
- **Phone 16 Module V2** ***(700359904)***
  Adds an additional 16 analog Plain Ordinary Telephone ports to control units.
- **Phone 30 Module V2** ***(700359912)***
  Adds an additional 30 analog Plain Ordinary Telephone ports to control units.
- **Digital Station 16 Module V2** ***(700359839)***
  Adds an additional 16 Digital Station ports to control units.
- **Digital Station 30 Module V2** ***(700359847)***
  Adds an additional 30 Digital Station ports to control units.
- **IP Office 500 Expansion Module Phone 30** ***(700426224)***
  Adds an additional 30 analog Plain Ordinary Telephone ports to control units.
- **IP Office 500 Expansion Module Digital Station 30** ***(700426216)***
  Add an additional 30 Digital Station ports to control units.
- **So8 Module** ***(700185077)***
  Provides 8 ISDN BRI S-interface device lines to the desktop.
- **Analog Trunk 16 - North America only** ***(700211360)***
  Provides an additional 16 Analog trunks (loop start or ground start) and two power fail sockets.
- **Analog Trunk 16 EU** ***(700241680)***
  Provides an additional 16 Analog trunks (loop start) and two power fail sockets. European CTR21 specification.
- **Analog Trunk 16 NZ** ***(700241698)***
  Provides an additional 16 Analog trunks (loop start) and two power fail sockets. New Zealand specification.
- **WAN3 10/100 Module** ***(700262009)***
  Provides an additional three V.24/V.35/X.21 ports. This expansion module is connected to the IP406 and IP412 control unit using the LAN and does not impact on the maximum number of external expansion modules supported. This module is not supported on the IP500.

IP400 Voice Compression Modules

All of the following can be installed in the IP Office 500 using the IP500 Legacy Card Carrier *(700417215)*.

- **Voice Compression Module 4** ***(700359854)***
  4 Channel Voice Compression module required for IP trunks and extensions. Includes 64ms echo cancellation.
- **Voice Compression Module 8** ***(700359862)***
  8 Channel Voice Compression module required for IP trunks and extensions. Includes 64ms echo cancellation.
- **Voice Compression Module 16** ***(700359870)***
  16 Channel Voice Compression module required for IP trunks and extensions. Includes 64ms echo cancellation.
- **Voice Compression Module 24** ***(700359888)***
  24 Channel Voice Compression module required for IP trunks and extensions. Includes 64ms echo cancellation.
- **Voice Compression Module 30** ***(700293939)***
  30 Channel Voice Compression module required for IP trunks and extensions. Includes 25ms echo cancellation.

IP500 Voice Compression Modules

Only supported in the IP500.

- **IP500 Media Card Voice Compression Module 32 *(700417389)***
  Voice Compression Module required for IP trunks and extensions. 4 channels are enabled by default. Additional channels up to the maximum of 32 are enabled through license keys. Includes 128ms echo cancellation.
- **IP500 Media Card Voice Compression Module 64 *(700417397)***
  Voice Compression Module required for IP trunks and extensions. 4 channels are enabled by default. Additional channels up to the maximum of 64 are enabled through license keys. Includes 128ms echo cancellation.

IP400 Modems cards

- **IP400 Modem 12 (700343452)**
  Internally fitted card allowing twelve simultaneous V.90 modem calls. Not supported on the IP500.

IP400 Trunk Interface Cards

Except where noted, all of the following can be installed in the IP500 using the IP500 Legacy Card Carrier *(700417215)*.

- **IP400 BRI-8 (UNI) (700262017)**
  Interface card for the Small Office Edition, IP406 and IP412 providing 4 x ISDN T-Bus Basic Rate Interface ports (8 lines).
- **IP400 PRI 30 E1 (1.4) (700272461)**
  Interface card for the IP406 and IP412 providing 1 x ISDN Primary rate port (30 lines).
- **IP400 PRI 30 E1R2 RJ45 - CALA (700241631)**
  Interface card for the IP406 and IP412 providing 1 x E1R2 Primary rate port (30 lines). RJ45 termination.
- **IP400 PRI 30 E1R2 COAX - CALA (700241656)**
  Interface card for the IP406 and IP412 providing 1 x E1R2 Primary rate port (30 lines). Co-Ax termination. Not supported on the IP500.
- **IP400 Dual PRI E1 (700185184)**
  Interface card for the IP406 and IP412 providing 2 x ISDN Primary rate ports (60 lines).
- **IP400 PRI T1 (700185200)**
  Interface card for the IP406 and IP412 providing 1 x T1/PRI port (24 lines).
- **IP400 Dual PRI T1 (700185218)**
  Interface card for the IP406 and IP412 providing 2 x T1/PRI (48 lines).
- **IP400 Quad Analog Trunk (Universal) (700359938)**
  Interface card for the IP406 and IP412 providing 4 x Loop start analog trunks. Universal variant supports specifications for North America, Europe and New Zealand.

Spares

The following are orderable spares available from Avaya.

5400, 5600, 2400 and 4600 series telephones

Item	Color	Material Code
Replacement Handset	Dark Grey	700203797
HDST HIP QD CORD- 4606/16/24/30 SETS		700212442
Amplified Handset	Dark Grey	700229735
Noisy Location Handset	Dark Grey	700229743
Push to Talk Handset	Dark Grey	700229727
24 Button expansion module for 5620/5420/4620/2420	Grey	700203656
Handset Cords 25ft	Dark Grey	700217417
1151C1 Power supply	–	700356447
1151C2 Power supply with battery backup	–	700356454
Power Cord INPUT 10A - European - 106336 CRD31	–	106336
Power Cord 98IN European 12013S	–	407786623
Power Cord 98IN United Kingdom 14012	–	407786599
Power Cord US Plug (15A, 120V) 17505	–	405362641

5600 and 4600 Series only

Item	Color	Material Code
Cat 5 Cable specific to 4620		700261613
IP PHONE MOD CORD 1 FT CAT5	–	408406932
IP PHONE MOD CORD 7 FT CAT5	–	408406957
IP PHONE MOD CORD 14 FT CAT5	–	408406940
IP PHONES Power 1152A1 Mid-Span	–	700180433

IP Office Control and Expansion Units

Item	Color	Material Code
60W in line power supply.	Black	700357387

Country Availability

IP Office is available from distribution partners in the following countries. Please refer to your country price list for the availability of individual items.

North America

Canada USA Mexico

South America

Argentina	Chile	Peru
Brazil	Colombia	

Europe, Middle East and Africa

Austria	France	Latvia	Slovenia
Belgium	Germany	Lithuania	South Africa
Croatia	Greece	Luxembourg	Spain
Cyprus	Hungary	Netherlands	Sweden
Czech Republic	Iceland	Norway	Switzerland
Denmark	Ireland	Poland	Turkey
Estonia	Israel	Portugal	UAE
Finland	Italy	Russia	United Kingdom
		Saudi Arabia	

Asia Pacific

Australia	Hong Kong	New Zealand	South Korea
China	India	Pakistan	

Sample Configurations

IP406 Office

Scenario 1:

A customer in Europe with complex telephony requirements, needing 30 exchange lines and 80 digital extensions.

This configuration provides support for up to 98 Avaya digital extensions (18 spare for growth) and a single Primary Rate Euro-ISDN connection (30 channels). If growth beyond 98 users or additional trunk capacity is anticipated, up to 3 more external expansion modules (another 90 extensions) and another trunk card (up to 60 additional channels) can be fitted. Typically, a business of this size has a data network that interconnects its users and provides access to business applications, front and back office systems as well as internet resources. The IP406 Office can be connected to this network through its integrated 8-port LAN switch. This provides all users with access to the business communications and personal productivity applications supported by IP Office.

Kit List

- 1 x IP406 Office DS control unit.
- 4 x Region specific power cords.
- 1 x PRI 30 E1 trunk card.
- 3 x Digital Station 30 external expansion modules.
- 80 x Avaya 5410 digital feature phones.

Scenario 2:

A business in the USA needs 32 analog telephones and one PRI (23+1D channels) for basic telephony

The IP406 Office with a single T1 PRI card and two Phone 16 external expansion modules provides the required line and extension capacity. The Phone Manager Lite application enhances the capabilities of each analog telephone, by enabling each user to handle calls and control their extension settings through a PC-based interface. For future growth, the system can support a further 4 external expansion modules and one additional internal trunk card.

Kit List

- 1 x IP406 Office DS control unit.
- 2 x Region specific power cords.
- 1 x Single T1 PRI trunk card.
- 1 x IP400 Phone 16 external expansion module.

IP412

Scenario 1:

A US business requiring 180 display phones and 96 digital trunks with 20 analog lines for fallback purposes.

This configuration uses a IP412 providing 180 extensions and 96 digital trunks (4 x T1) and two IP400 Analog Trunk 16 modules offering capacity of up to 32 analog trunk lines . With the addition of a single Dual PRI T1 interface, the system is fitted with an extra trunk card in its spare slot to provide the additional 48 lines.

Kit List

- 1 x IP412 control unit.
- 9 x Region specific power cords.
- 2 x PRI 48 T1 trunk cards.
- 6 X IP400 Digital Station 30 external expansion modules.
- 2 x IP400 Analog Trunk 16 external expansion modules.
- 180 x Avaya 5410 digital phones.

Scenario 2:

A Business moving to a pure IP Telephony solution with 90 IP hardphones, 90 IP softphones and 60 external trunk lines for its main location and the ability to network with other sites using IP trunking.

This configuration uses an IP412 PRI 60 E1 fitted with two 30-channel Voice Compression Modules (VCMs). These two internally fitted cards allow up 60 simultaneous calls to external parties (IP extension calling a non-IP telephone or line). For IP to IP calls, VCM resources are only required for initial call set-up. Depending on the typical utilization of external trunks, a lower capacity VCM variant could be employed, as appropriate.

The IP Office softphone is 'Phone Manager Pro PC Softphone' which is an enhanced version of the standard Phone Manager Pro application enabled for each user using two License Keys as listed below.

Kit List

- 1 x IP412 control unit.
- 1 x PRI 60 E1 trunk card.
- 1 x Region specific power cord.
- 2 x IP400 VCM 30 cards.
- 60 x 5610 IP phones.
- 1 x IP Office Feature Key
- 1 x IP400 Phone Manager Pro RFA 50.
- 1 x IP400 Phone Manager Pro RFA 10.
- 1 x IP400 Phone Manager PC SoftPhone RFA 50.
- 1 x IP400 Phone Manager PC SoftPhone RFA 10.

IP500

Scenario 1:

A US business requiring 190 display phones and 96 digital trunks with 20 analog lines for fallback purposes.

This configuration uses an IP500 providing 196 extensions and 96 digital trunks (4 x T1) and two IP400 Analog Trunk 16 modules offering capacity of up to 32 analog trunk lines .

Kit List

- 1 x IP500 control unit.
- 1 x IP500 Feature Key
- 1 x IP Office Standard Edition upgrade to Professional Edition license
- 9 x Region specific power cords.
- 2 x IP500 Digital Station 8 cards
- 2 x IP500 Legacy Card Carriers
- 2 x PRI 48 T1 trunk cards.
- 6 X IP400 Digital Station 30 external expansion modules.
- 2 x IP400 Analog Trunk 16 external expansion modules.
- 190 x Avaya 5410 digital phones.

Scenario 2:

A Business moving to a pure IP Telephony solution with 90 IP hardphones, 90 IP softphones and 60 external trunk lines for its main location and the ability to network with other sites using IP trunking.

This configuration uses an IP500 fitted with a 64-channel Voice Compression Module (VCM). This card allows up to 64 simultaneous calls to external parties (IP extension calling a non-IP telephone or line). For IP to IP calls, VCM resources are only required for initial call set-up. Depending on the typical utilization of external trunks, a lower capacity VCM variant could be employed, as appropriate.

The IP Office softphone is 'Phone Manager Pro PC Softphone' which is an enhanced version of the standard Phone Manager Pro application enabled for each user using two License Keys as listed below.

Kit List

- 1 x IP500 control unit.
- 1 x IP Office Standard Edition upgrade to Professional Edition license
- 1 x IP500 Legacy Card Carrier
- 1 x PRI 60 E1 trunk card.
- 1 x Region specific power cord.
- 1 x IP500 VCM 64 card (4 channels enabled by default).
- 1 x IP500 VCM 60 channel license
- 60 x 5610 IP phones.
- 1 x IP500 Feature Key A-Law
- 1 x IP400 Phone Manager Pro RFA 50.
- 1 x IP400 Phone Manager Pro RFA 10.
- 1 x IP400 Phone Manager PC SoftPhone RFA 50.
- 1 x IP400 Phone Manager PC SoftPhone RFA 10.

B: TAPI Functions Supported by IP Office

TAPI 2.1 Functions Supported

TAPI*Link* Lite provides the following functionality for TAPI 2.1:

- lineAddToConference
- lineAnswer
- lineBlindtransfer
- lineCompleteTransfer
- lineConfigDialog
- lineClose
- lineDeallocateCall
- lineDial
- lineDrop
- lineGetAddressCaps
- lineGetAddressID
- lineGetAddressStatus
- lineGetAppPriority
- lineGetCallInfo
- lineGetCallStatus
- lineGetDevCaps
- lineGetID
- lineHold
- lineInitialiseEx
- lineMakeCall
- lineNegotiateTAPIVersion
- lineOpen
- linePark
- lineRedirect
- lineRemoveFromConference
- lineSetAppPriority
- lineSetAppSpecific
- lineSetCallPrivilege
- lineSetStatusMessages
- lineSetupTransfer
- lineShutdown
- lineSwapHold
- lineUnhold
- lineUnpark
- lineSetCallData
- lineDevSpecific
- lineGenerateDigits
- lineGenerateTone
- lineMonitorDigits
- lineMonitorTones

TAPI 3.0 functions supported

The following functions are supported using TAPI 3.0:

- **ITTAPI**
- Initialize
- Shutdown
- EnumerateAddresses
- RegisterCallNotifications
- Put_EventFilter

- **ITAddress**
- get_AddressName
- get_dialableAddress
- get_ServiceProviderName
- CreateCall

- **ITMediaSupport**
- get_MediaTypes

- **ITCallInfo**
- get_Address
- get_CallState
- get_CallInfoString
- SetCallInfoBuffer

- **ITBasicCallControl**
- Connect
- Answer
- Disconnect
- Hold
- SwapHold
- ParkDirect
- Unpark
- BlindTransfer
- Transfer

- **ITCallStateEvent**
- get_Cause
- get_State
- get_Call

- **ITCallNotificationEvent**
- get_Call

- **ITCallInfoChangeEvent**
- get_Call

- **ITCallHubEvent**
- get_Event
- get_Call

Notes:

- TAPI*Link* Lite can be used from C, C++ and Delphi. Visual Basic cannot directly use TAPI 2.1, but does support TAPI 3.0 without any third-party tools.
- TAPI*Link* Lite provides detailed information on telephony events, including the ability to screen-pop based on CLI and/or DDI.

Changes from previous versions of IP Office

TAPI Reserved Fields

TAPI fields that were previously reserved by IP Office for internal use have now been released for general use by developers. A full definition of theses fields are contained in the IP Office developers SDK CD. The following table shows the device specific data available via TAPI.

- Phone's extension number
- Forward on busy flag
- Forward on no answer flag
- Forward unconditional flag
- Forward hunt group flag
- Do not disturb flag
- Outgoing call bar flag
- Call waiting on flag
- Voicemail on flag
- Voicemail ring-back flag
- Number of voicemail messages
- Number of unread voicemail messages
- Outside call sequence number
- Inside call sequence number
- Ring back sequence number
- No answer timeout period
- Wrap up time period
- Can intrude flag
- Cannot be intruded upon flag
- X directory flag
- Force login flag
- Login code flag
- System phone flag
- Absent message id
- Absent message set flag
- Voicemail email mode
- User's extension number
- Users Locale
- Forward number
- Follow me number
- Absent text
- Do not disturb exception list
- Forward on busy number
- User's priority
- Number of groups the user is a member of
- Number of groups that the user is a member of that are currently outside their time profile
- Number of groups the user is currently disabled from
- Number of groups that the user is a member of that are currently out of service
- Number of groups that the user is a member of that are currently on night service

DevLink Reserved Fields

DevLink fields that were previously reserved by IP Office for internal use have now been released for general use by developers. A full definition of these fields is contained on the IP Office 2.0 developers SDK CD. The following table shows the device specific data available via DevLink. A "Y" in the column indicates that the field is already described in the DevLink manual.

#	Field Data (S Message)	#	Field Data (S Message)
1	A call id	26	Voicemail disallow
2	B call id	27	Sending complete
3	A state	28	Bc.tc,bc.tm
4	B state	29	Owner hunt group name
5	A connected	30	Original hunt group name
6	A is music	31	Original user name
7	B connected	32	Target hunt group name
8	B is music	33	Target user name
9	A name	34	Target RAS name
10	B name	35	Is internal call
11	B list (possible targets for the call)	36	Time stamp
12	A slot ,channel	37	Connected time
13	B slot , channel	38	Ring time
14	Called party presentation & type	39	Connected duration
15	Called party number	40	Ring duration
16	Calling party presentation & type	41	Locale
17	Calling party number	42	Park slot number
18	Called sub address	43	Call waiting
19	Calling sub address	44	Tag
20	Dialled party type	45	Transferring
21	Dialled party number	46	Sv active
22	Keypad type	47	Sv quota used
23	Keypad number	48	Sv quota time
24	Ring attempt count	49	Account code
25	Cause	50	Unique call identifier
#	**Field Data (D Message)**	**#**	**Field Data (A Message)**
1	A call id	1	A call id
2	B call id	2	B call id
3	Unique call identifier	3	Unique call identifier

C: Technical Specifications

General

Dimensions

Unit Dimensions (mm/inches)	Width	Height	Depth
IP406 V2, IP412 and all Expansion Modules	445mm/17.5"	71mm/2.8"	245mm/9.7"
IP Office - Small Office Edition	255mm/10.0"	76mm/3.0"	241mm/9.5"
IP500	445mmm/17.5"	73mm/2.9"	365mm/14.4"

- The recommended minimum clearance, front and rear, for the connection of cables and other devices is 75mm/3".

Weight

Unit	Weight
IP500 System Unit	3.2Kg/7.0lbs
IP406 V2 Control Unit	3.0Kg/6.7lbs
IP412 Control Unit	3.0Kg/6.7lbs
IP Office - Small Office Edition	1.2Kg/2.6lbs
Analog 16 Module	2.9Kg/6.5lbs
DS 16 Module	3.0Kg/6.7lbs
DS 30 Module	3.5Kg/7.8lbs
WAN3 Module	2.8Kg/6.3lbs
So8 Module	2.8Kg/6.3lbs
Phone 8 Module	2.8Kg/6.3lbs
Phone 16 Module	2.9Kg/6.5lbs
Phone 30 Module	3.1Kg/6.94lbs

Environmental

- 0°C to +40°C (32°F to 104°F). 95% relative humidity, non-condensing.

Telephone Extension Cable Lengths

The following table details the maximum cable lengths supported for the telephone ranges. These figures assume that standard twisted-pair telephone cable or CAT5 network cable is used.

Telephone	Unshielded Twisted-Pair (UTP) - 50nf/Km			CW1308
	AWG22 (0.65mm)	AWG24 (0.5mm)	AWG26 (0.4mm)	
2400/5400 Series	1200m/3937'.	1000m/3280'.	670m/2200'.	400m/1310'.
4406D Phone	1000m/3280'.	1000m/3280'.	400m/1310'.	400m/1310'.
4412D Phone	1000m/3280'.	700m/2295'.	400m/1310'.	400m/1310'.
4424D	500m/1640'.	500m/1640'.	400m/1310'.	400m/1310'.
6400 Series	1000m/3280'.	1000m/3280'.	400m/1310'.	400m/1310'.
T3 Series (Upn)	1000m/3280'.	1000m/3280'.	400m/1310'.	–
Analog Phones	1000m/3280'.	1000m/ 3280'.	400m/1640'.	800m/2620'.

Heat Dissipation

Note that the above numbers are for reference only. For practical purposes, for example the calculation of heat dissipation, it is recommended to base environmental requirements (for example air cooling or UPS ratings) on the maximum input rating of the power supplies of the planned IP Office configuration, as follows.

In order to calculate the maximum, that is worst case, amount of heat that can be generated by an IP Office system, it is assumed that all input power is converted to heat; whether from the PSU itself, the system unit, expansion module and/or cabling.

Heat dissipation is normally measured in British Thermal Units (BTU's). A heat value expressed in Watts can be converted to BTU/hr by multiplying by 3.41297. As indicated above, you should use the maximum power input of 115 VA of each power supply to calculate this most accurately

Using the conversion factor:

- Heat Dissipation = 115 x 3.41297 = 392.5 BTU/hour.

The metric equivalent to BTU is a Joule where 1 BTU = 1,055 Joules.

This calculates the BTU value per power supply. The maximum BTU per system is therefore calculated, based on total number of power supplies installed in the system. For example, for a IP412, this would be 1 for the base unit and up to 12 for the expansion modules.

- IP412 Maximum Heat Dissipation = 13 x 392.5 = 5,103 BTU/hr.

Remember to budget for the power requirements of any additional devices that are to be co-located with the IP Office such as server PC's (voicemail, etc).

Power Supply

- **Input**
  - **Small Office Edition:** 2.5mm DC inlet socket. 24Vdc power input. Rating 24V DC, 1.8A maximum.
  - **IP406 V2, IP412 and expansion modules:** 2.5mm DC inlet socket. 24Vdc power input. Rating 24V DC, 2A maximum.
  - **IP Office 500 System Unit:** IEC AC inlet socket. 100-240V AC, 50/60Hz, 81-115VA, 2.5A maximum.

- **Power Supply Units:** All CE/UL/Dentori Safety Approved.
  - **Standard 40W Power Supply Unit** (All control and expansion units unless otherwise indicated) Supplied with the control or expansion unit. 40W PSU with integral lead to the unit. Connection to switched mains supply requires separately supplied country specific IEC 60320 C7 power cord (2-wire figure 8 connector).
    - Input: 100-240V AC, 50/60Hz, 81-115VA, 2A maximum.
    - Output: 24Vdc, 1.875A, output power 45W maximum.
  - **Small Office 45W Power Supply Unit**
    Supplied with the unit. 45W PSU with integral lead to control unit. Connection to switched mains supply requires separately supplied country specific IEC 60320 C13 power cord (3-wire earthed cold kettle lead).
    - Input: 100-240V AC, 50/60Hz, 81-115VA, 1.5A maximum.
    - Output: 24V DC, 1.875A, output power 45W maximum.
  - **IP406 V2 60W Power Supply Unit**
    Supplied with the control or expansion unit. 60W PSU with integral lead to the unit. Connection to switched mains supply requires separately supplied country specific IEC 60320 C13 power cord (3-wire earthed cold kettle lead).
    - Input: 100-240V AC, 50/60Hz, 81-115VA, 2.5A maximum.
    - Output: 24V DC, 1.5A, output power 60W maximum.
  - **IP Office 500 80W internal Power Supply**
    Integral to the System Unit. Connection to switched mains supply requires separately supplied country specific IEC 60320 C13 power cord (3-wire earthed cold kettle lead).
    - Input: 100-240V AC, 50/60Hz, 81-115VA, 2.5A maximum.

Interfaces

Interface	Information
DTE Port	• 25 way D-Type female connector, V.24/V.28. • 9 way D-type on IP412, IP406 V2, IP500 and IP Office - Small Office Edition.
ISDN Ports	**EU Interfaces:** • **BRI:** RJ45 sockets. ETSI T-Bus Interface to CTR3 for Pan European Connection. • **PRI E1:** RJ45 socket. ETSI T-Bus Interface to CTR4 for Pan European Connection. • **PRI T1/J1:** RJ45 socket: FCC Part 68/JATE connection. **USA Interfaces:** • **PRI T1 Service:** Ground Start (GS) – Default, E&M, 56k data for 5ESS, 56/64/64 restricted for 4ESS. • **PRI ISDN Switch support:** 4ESS, 5ESS, DMS-100, DMS-250 (includes conformance to ANSI T1.607 & Bellcore Special Report SR4287, 1992). • **PRI ISDN Services:** AT&T Megacom 800, AT&T WATS (4ESS), AT&T SDS Accunet 56kB/s & 64kB/s (4ESS), AT&T Multiquest (4ESS).
Analog Trunk Ports	• **RJ45 sockets:** Loop start/Ground start (regional dependant)
Power Fail Ports	• **RJ45 sockets:**
ISDN Data Rates	• **BRI:** B-channel 64kbps or 56kbps, D-channel 16kbps. • **PRI:** B-channel 64kbps or 56kbps, D-channel 64kbps.
Analog Phone Ports	• **RJ45 sockets:** • **CLI Schemes:** DTMFA, DTMFC, DTMFD, FSK and UK20. • **REN:** 2. (External Bell via POT port: REN = 1) • **Off Hook Current:** 25mA. • **Ring Voltage:** 40V (nominal) RMS.
LAN	• RJ45 sockets. Auto-negotiating 10/100 BaseT Ethernet (10/100Mbps).
WAN	• Small Office Edition: RJ45 Ethernet socket. • IP406 V2 and IP412 (optional on Small Office Edition): 37 way D-Type female sockets. X.21 interface to 2048k bps, V.35 interface to 2048Kbps and V.24 Interface to 19.2Kbps.
Audio	• 3.5mm Stereo Jack socket. Input impedance - 10k /channel. • Maximum AC signal – 200mV rms.
External Output Port	• 3.5mm Stereo Jack socket. Switching Capacity - 0.7A. • Maximum Voltage - 55V DC. On state resistance - 0.7. • Short circuit current - 1A. Reverse circuit current capacity - 1.4A.
Wireless Module	• Small Office Edition only. • 16bit Type II PCMCIA format PC card. • IEEE 802.11b WiFi.
Embedded Voice Memory	• Small Office Edition: 64MB Flash memory, 16bit Type II PCMCIA card. • IP406 V2 and IP500: 512MB Compact Flash memory card.

Specification for IP Office Application PC's

Applications System Requirements

- Any IP Office system. (2.1 and above)
- Any IP Office supported desktop telephone.

Ethernet attached PC running as a recommended minimum, Microsoft Windows 2000/2003/XP Professional, with the following minimum supported specification

Product Key

- VM Lite = Voicemail Lite
- VM Pro = Voicemail Pro
- IMS = Integrated Messaging Pro
- CM = Campaign Manager
- TTS = Text To Speech
- IVR = Third Party Database Access
- CS = ContactStore
- CBC = Compact Business Center
- CCC = Compact Contact Center

Server Applications Dependencies

Applications	Minimum PC Resources	Intel Pentium	Intel Celeron	AMD	Notes
VM Lite	256MB RAM 2GB drive.*1	Any 1.4GHz	Any 1.7GHz	Any 1.4GHz	Attempting to run the applications on lower specification PC's may cause degradation of operation and will not be supported.
VM Pro	256MB RAM 2GB drive.*1	Any 1.4GHz.	Any 1.7GHz.	Any 1.4GHz.	To avoid replacing the server when adding new applications we recommend that a Pentium 4 2.8GHz (or equivalent) is used when possible.
VM Pro + IMS + CM	512MB RAM 2GB drive.*1	Pentium4 2.8GHz.	Not tested	Athlon XP 3000+ All Athlon64.	
VM Pro + IVR + TTS	512MB RAM 20GB drive.*1	Pentium4 2.8GHz	Not tested	Athlon XP 3000+ All Athlon64.	If the database being queried is located on the VM Pro server the query speed of the database will be affected by the amount of memory available. Please take into account the memory requirements of the database being queried.
VM Pro + CS	512MB RAM 20GB drive.*1	Pentium4 2.8GHz	Not tested	Athlon XP 3000+ All Athlon64.	
VM Pro + CCC	512MB RAM 30GB drive.*1	Pentium4 2.8GHz	Not tested	Athlon XP 3000+ All Athlon64.	VM Pro and CCC can be run on the same server OS up to a maximum of 25 agents, 8 ports of VM Pro.
VM Pro + CBC	512MB RAM 120GB drive.*1	Pentium4 2.8GHz.	Not tested	Athlon XP 3000+ All Athlon64.	The client PC needs to be Pentium III, 800MHz with 128MB RAM minimum.
CCC	512MB RAM 10GB drive.	Any 1.4GHz.	Any 1.7GHz.	Any 1.4GHz.	
Conferencing Center	512MB RAM 80GB drive.	Pentium4 2.8GHz.	Not tested	Athlon XP 3000+ All Athlon64.	Windows XP Professional or 2000 Professional can be used but would typically support a maximum of 10 web clients. To support more than 10 clients a server OS with IIS will be required.
CBC/SMDR	256MB RAM 10GB drive. IE6.0 or higher.	Pentium III 800MHz.	Celeron3 800Mhz	Athlon B 650MHz	The Delta Server and CBC can be installed on either the same PC or on separate PC's. In both cases these are the minimum PC specifications.
Feature Key Server PC	256MB RAM 1MB free disk space.	Pentium III 800MHz.	Celeron3 800Mhz.	Athlon B 650MHz.	

*1: For all voicemail servers, also allow 1MB per minute for message and greeting storage.

Client Applications Dependencies

Applications	Minimum PC Resources	Intel Pentium	Intel Celeron	AMD	Notes
Conferencing Web Client	Internet Explorer 6 or above.	Any.	Any.	Any.	Any desktop machine can be used as long as it is capable of running IE6.
Phone Manager Lite/Pro	64MB RAM 160MB free disk space.	Pentium III 800MHz.	Celeron3 800Mhz.	Athlon B 650MHz.	A sound card is needed if audio features are required.
Phone Manager PC SoftPhone	64MB RAM 1GB free disk space.	Pentium III 800MHz.	Celeron3 800Mhz.	Athlon B 650MHz.	A sound card is needed.
SoftConsole	128MB RAM with 1GB of free disk space	Pentium III 800MHz.	Celeron3 800Mhz.	Athlon B 650MHz.	A maximum of four SoftConsole applications can be run per system, a license controls the number of simultaneous SoftConsole users. A sound card is needed if audio features are required.
ContactStore Web client	Internet Explorer 6.0 or above.	Any	Any	Any	Any desktop machine can be used as long as it is capable of running IE6.
IP Office Manager	128MB RAM 1GB disk space	Pentium4 600Mhz.	Not tested	AMD Opteron, Athlon 64 or Athlon XP.	For Windows XP, minimum recommend RAM increases to 256MB.
Call Status	64MB RAM 50MB disk space	Pentium III 800MHz.	Celeron 3 800Mhz.	Athlon B 650MHz.	For OS of Windows XP, minimum RAM increases to 256MB
System Monitor	128MB RAM 10GB disk space	Pentium III 800MHz.	Celeron 3 800Mhz.	Athlon B 650MHz.	For OS of Windows XP, minimum RAM increases to 256MB
Contact Center View (CCV)	128MB RAM 10GB disk space	Pentium III 800MHz.	Celeron 3 800Mhz.	Athlon B 650MHz.	For OS of Windows XP, minimum RAM increases to 256MB
CCC Reporter	Internet Explorer 6.0 or above.	Any	Any	Any	Any desktop machine can be used as long as it is capable of running IE6.
Wallboard Server	128MB RAM 10GB free disk space.	Any 1.4GHz.	Any 1.7GHz.	Any 1.4GHz.	The Wallboard Server MUST reside on the same PC as the Delta Server
Wallboard Client	128MB RAM 10GB disk space.	Pentium III 800MHz.	Celeron3 800Mhz.	Athlon B 650MHz.	For OS of Windows XP, minimum RAM increases to 256MB
PC Wallboard	128MB RAM 10GB disk space.	Pentium III 800MHz.	Celeron3 800Mhz.	Athlon B 650MHz.	For OS of Windows XP, minimum RAM increases to 256MB

Operating Systems for IP Office 4.0

The following table gives a summary of the Server & Client Operating Systems (OS) on which various IP Office applications are tested and supported for IP Office 4.0.

Microsoft Server OS's[1,9]	IP Office Manager	CBC[2]	CCC v5 Server	VM Lite	VM Pro[3]	SMDR[6]	Conferencing Center Server
Windows 2000 server (SP4)	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Windows 2003 server[8]	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Windows XP Professional (SP2)	Yes	Yes	No	Yes	Yes	Yes	No

Microsoft Client OS's[1,9]	IP Office Manager	CBC[2]	CCC Clients	VM Lite	VM Pro[3]	Soft Console	Phone Manager	Conferencing Center Client[5]
Windows XP Professional (SP2)	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Windows 2000 Professional (SP4)	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes

Windows Operating System Service Pack Support

Operating System	Current Service Pack and Date of Availability	Next Update and Estimated Date of Availability	Notes
Windows 2000 Professional, Windows 2000 Server, Advanced Server and Datacenter Server	SP4 June 26th 2003	Dependant upon Microsoft release and Support schedule.	
Windows XP Home Edition	SP2 August 9, 2004		
Windows XP Professional	SP2 August 9, 2004		Details of how to configure IP Office applications for operation with SP2 are contained with in the IP Office Tech Tip Bulletin 49.
Windows Server 2003	N/A		Please see IP Office Tech Tip Bulletin 49.

Notes:

1. Windows ME, Windows 95 and NT4 Operating Systems are no longer supported by Avaya.
2. CBC requires the associated Delta Server application to be installed on a Windows 2000/XP workstation or a 2000/2003 server. Windows 2003 server requires Delta Server 4.0(33) or above.
3. IMS and Web Campaigns options within Voicemail Pro are only supported on Windows Servers. Aspects of operation such as Voicemail to E-mail, Integrated Messaging Pro (IMS), Web Campaigns, etc, are subject to further requirements. Please refer to the Voicemail Installation and Administration manual. Integrated Messaging Pro (IMS) is supported on Microsoft Exchange 5.5, 2000 and 2003. The R3.0GA release of Voicemail Pro does not support IMS operation with Outlook 2003 operating in cache mode. The R3.0 maintenance release will provide this support.
4. For Phone Manager/PC Softphone Avaya recommends the use of Windows XP/2000.
5. Conferencing Center Web Client simply requires Internet Explorer 6.0 or higher (no other application required).
6. Although a server application, IP Office SMDR can also run on a Windows 2000, 2003 and Windows XP client Operating Systems but should not run on the same PC as a CBC or CCC Delta Server.
7. Windows 98 is only supported on IP Office V2.1 and V3.0 applications; it is not supported on IP Office 3.1 applications and above. Systems that are upgraded to V3.1 should have also have any Windows 98 PCs that are running IP Office applications upgraded to use Windows 2000, Windows XP or later operating systems.
8. Windows Small Business Server 2003 is supported for the same applications as Windows 2003 Server.
9. 64-Bit versions of Microsoft operating systems are not currently supported with IP Office applications.

Protocols

Protocol	RFC	Information
V120	-	A standard Rate Adaptation mechanism.
V110	-	A standard Rate Adaptation mechanism.
PPP	RFC1661	Point to Point Protocol.
LCP	RFC1570	Link Control Protocol.
MP	RFC1990	Multi-Link (Point to Point) Protocol.
IPCP	RFC1332	Internet Protocol Control Protocol.
PAP	RFC1334	Password Authentication Protocol.
RTP/RTCP	RFC1889	Real Time and Real Time Control Protocol.
CHAP	RFC1994	Challenge Handshake Authentication Protocol.
CCP	RFC1962	Compression Control Protocol.
STAC	RFC1974	STAC LZS Compression Protocol.
MPPC	RFC2118	Microsoft Point to Point Compression (Protocol).
BACP	RFC2125	Bandwidth Allocation Control Protocol.
UDP	RFC768	User Datagram Protocol.
IP	RFC791	Internet Protocol.
TCP	RFC793	Transmission Control Protocol.
DHCP	RFC1533	Dynamic Host Control Protocol.
NAT	RFC1631	Network Address Translation.
BOOTP	RFC951	Bootstrap Protocol.
TFTP	RFC1350	Trivial File Transfer Protocol.
NTP	RFC868	Network Time Protocol.
SNMPv1	RFC1157	Simple Network Management Protocol. (STD15)
	RFC1155	Structure and identification of management information for TCP/IP based internets. (STD16)
	RFC1212	Concise MIB Definitions. (STD16)
	RFC1215	A convention for defining traps for use with SNMP.
MIB-II	RFC1213	Managment Information base for network management of TCP/IP based internets: MIB-II. (STD17)
ENTITY MIB	RFC2737	Entity MIB (Version 2).
RIP	RFC1058	Routing Information Protocol.
	RFC2453	RIP Version 2. (STD56)
	RFC1722	RIP Version 2 Protocol Applicability Statement. (STD57)
IPSec	RFC2401	Security Architecture for the Internet Protocol.
	RFC2402	IP Authentication Header.
	RFC2403	The Use of HMAC-MD5-96 within ESP and AH.
	RFC2404	The Use of HMAC-SHA-1-96 within ESP and AH.
	RFC2405	The ESP DES-CBC Cipher Algorithm with Explicit IV.
	RFC2406	IP Encapsulation Security Payload. (ESP)
	RFC2407	The Internet IP Security Domain of Interpolation for ISAKMP.
	RFC2408	Internet Security Association and Key Management Protocol.
	RFC2409	The Internet Key Exchange.
	RFC2410	The NULL Encryption Algorithm and its Use with IPSec.
	RFC2411	IP Security Document Roadmap.
L2TP	RFC2661	Layer Two Tunneling Protocol "L2TP".
	RFC3193	Securing L2TP using IPSec.
Header Compression	RFC2507	IP Header Compression (IPHC).
	RFC2508	Compressing IP/UDP/RTP Headers for Low-Speed Serial Links.
	RFC2509	IP Header Compression over PPP.
DiffServ	RFC2474	Definition of the Differentiated Services Field (DS Field) in the IPv4 and IPv6 Headers.
PPP MP	RFC1990	The PPP Multilink Protocol (MP).
Frame Relay Encapsulation	RFC1490	Multi protocol Interconnect over Frame Relay.
ML-PPP	RFC2686	The Multi-Class Extension to Multi-Link PPP.

Session Initiation Protocol

- Rec. E.164 [2] - ITU-T Recommendation E.164: The international public telecommunication numbering plan
- RFC 2833 [7] - RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
- RFC 3261 [8] - SIP: Session Initiation Protocol
- RFC 3263 [10] - Session Initiation Protocol (SIP): Locating SIP Servers
- RFC 3264 [11] - An Offer/Answer Model with Session Description Protocol (SDP)
- RFC 3323 [14] - A Privacy Mechanism for the Session Initiation Protocol (SIP)
- RFC 3489 [18] - STUN - Simple Traversal of User Datagram Protocol (UDP) Through Network Address Translators (NATs)
- RFC 3824 [24] - Using E.164 numbers with the Session Initiation Protocol (SIP)

Glossary

A

ANI: Automatic Number Identification (ANI). See CLIP

Assisted Transfer: A call transferred from voicemail, which if it returns again to voicemail, will return to the previous position.

B

BACP: Bandwidth Allocation Control Protocol (BACP) is a protocol specification for PPP that allows Multilink PPP routers to negotiate extra bandwidth dynamically over time. Using BACP, two routers can dynamically connect extra "B" channels at times of higher load, then can drop the channels when they are no longer needed. BACP is described in RFC2125.

BDC: Backup Domain Controller is a server in a network domain that keeps and uses a copy by a computer without interrupting its current or primary task. For Windows NT Server domains, BDC refers to a computer that receives a copy of the domain's security policy and domain database and authenticates logons.

Blind Transfer: A call transferred without waiting for the transfer destination to answer first.

BOOTP: This protocol was invented when it was expensive to store software or configurations in small hosts (and even more expensive to upgrade them) so when the host was switched on it would ask (broadcast) on the LAN for its software. A machine with a disk would reply and send the software. Typically the BOOTP Server would send a file to the host using Trivial File Transfer Protocol (TFTP). The main unit uses BOOTP to obtain new versions of its operational software (which it stores in its flash memory). The Manager program acts as the BOOTP server. The BOOTP server recognizes the main unit by its MAC address, this is a hardware address built into the unit at manufacture. This information is obtained from a BOOTP entry which must also include the unit's IP Address and name of the software file to be sent. BOOTP entries are created automatically and stored in the PC's registry.

C

Callflow: A general term for a sequence of actions used to determine what facilities are offered to a caller.

CAPI: Common Application Programming Interface.

CHAP: Challenge Handshake Authentication Protocol (CHAP). An authentication scheme used by PPP servers to validate the identity of the originator of a connection, upon connection or any time later.

CLI: Calling Line ID. Information passed from the telephone network exchange to the IP Office. Also called ICLID and CLID.

CLID: Calling Line ID. See CLI.

CLIP: Calling Line Identity Presentation. Displays the calling party's number to the called party. Variations include withholding CLI and displaying alternative presentation numbers. ANI (automatic Number Identification) is the USA equivalent.

CLIR: Calling Line Identification Restriction (CLIR) Inhibits the telephone number of the IP Office being presented on an outbound call.

COLP: Connected Line Identity Presentation (COLP). Displays the connected party's number to the calling party. Useful where the call has been diverted away from the originally dialed party.

COLR: Connected Line Identification Restriction (COLR) Inhibits the COLP service.

CSU: Channel Service Unit: Used to terminate an incoming digital trunk at the customer premises. Incorporates features to allow trunk testing and checking, including loop-back functions.

CTI: Computer Telephony Integration, a technology that acts as an electronic bridge connecting telephones or switches with computers. CTI controls or coordinates business processes and related applications through the exchange of commands and messages between computers and telephone systems.

D

DDI(DID)/MSN: Direct Dial In (DDI/DID) and Multiple Subscriber Numbering (MSN) are telephone company services that can be subscribed to. Call destinations can therefore be passed down the ISDN line and the system can use this information to deliver the calls to their final destination, perhaps individuals or departments.

DHCP: Dynamic Host Configuration Protocol, a standards-based protocol for dynamically allocating and managing IP addresses. DHCP runs between individual computers and a DHCP server to allocate and assign IP addresses to the computers and also limits the time computers can use the address. When time expires on the use of the IP address, the computers contact the DHCP server again to obtain an address.

DiffServ: DiffServ (RFC 2474) is a TCP/IP quality of Service mechanism used to ensure that IP packets are prioritized according to their importance, for example prioritization of voice packets over data packets. Prioritization is based upon the Type of Service (ToS) field in the IP header.

Digital Stations: Refers to Avaya telephones in the 2400, 4400, 5400 and 6400 series. Supported by DS sockets on IP Office control units and Digit Station modules. Note: Not all terminals in the above ranges are supported on IP Office.

Dn: Directory number.

DNIS: Dialed Number Identification Service (DNIS). Available in US markets. DNIS identifies to the called party the dialed number. Can be used to identify the purpose of inbound calls.

Domain: The part of the computer network in which the data processing resources are under common control.

DSS: Direct Station Select - A DSS key can be programmed with a number or feature code.

DSU: Data Service Unit: Normally incorporated within the CSU of digital trunk connections. The DSU allows the trunk to be shared between data and voice services.

E

Embedded Voicemail: A voicemail system stored on a memory card inserted into the IP Office telephone system's control unit.

ESP: Encapsulation Security Payload: A standard (RFC2406) that forms part of IPSec.

F

Frame Relay: Connections to private or public Frame Relay services, such as BT FrameStream, can be made via the WAN port on the rear of main unit, or the WAN port of an associated WAN 3 module. Both data and Voice over IP (requires the use of the Voice Compression Module) are supported across Frame Relay.

G

G.711 A-Law 64K: A VoIP compression mode. Each voice call is converted from analog to digital (refer to G.723) and uncompressed.

G.723.1 6K3 MP-MLQ: A VoIP compression mode. A real-time implementation of the ITU-T Multi-Pulse Maximum Likelihood Quantization (MP-MLQ) 6.4 Kbps and Algebraic Codebook Excited Linear Prediction (ACELP) 5.3 Kbps speech coding algorithms. The G.723.1 speech coder operates upon 30 ms frame of digitized, telephone bandwidth speech signals sampled at 8 kHz. The frames are divided into four 7.5 milli-second sub frames of 60 samples each. Each frame of 240 input samples is converted into 12 16-bit word of compressed data at the high rate or 10 16-bit words of compressed data at the low rate. The Voice Activity Detection/Comfort Noise Generation (VAD/CNG) specified in Annex A to ITU-T G.723.1 is fully implemented,and may be used to further reduce the average bit rate.

G.726 ADPCM 16K/32K: A VoIP compression mode. Each voice call is compressed using the standard ADPCM compression technique (refer to G.732). This algorithm uses 16,000 or 32,000 bits per second.

G.729(a) 8K CS-ACELP: A VoIP compression mode. A fully compliant, real-time implementation of the ITU-T fixed-point conjugate-structure, algebraic code-excited linear prediction (CS-ACELP) speech coding algorithm. The CS-ACELP operates at 8Kkbps. The coder processes 10 millisecond frames of speech sampled at an 8 kHz rate, which together with a 5 millisecond look-ahead results in a total algorithmic delay of 15 milliseconds. For each frame of 80 samples of 16-bit linear PCM data, the coder outputs five 16-bit words. Applications using the G.729 vocoder include digital telephony, satellite and wireless communications.

Gatekeeper: An H.323 entity that provides address translation, controls access, and sometimes bandwidth management to the LAN for H.323 terminals, Gateways, and Multipoint Control Units. IP Office units can register themselves with multiple external H.323 gatekeepers.

GUI: Graphical User Interface.

H

H.323 VoIP: Allows voice and data traffic to be networked between systems. Connections between platforms across the WAN, at speeds up to 2.048Mbps (in conjunction with the Voice Compression Module), or across the LAN at 10 or 100 Mbps. Multiple WAN links maybe supported utilizing the optional WAN3 modules. Also allows telephone calls to be made from PCs running Microsoft's NetMeeting when fitted with a sound card, speakers and microphone. Calls can be made between PCs or to standard analog or digital telephones. Please note that at this point in time, we do not consider NetMeeting to offer a Toll Quality voice service. The addition of the IP Telephony Extensions to the H.323 Gateway protocol allows physical H.323compliant IP "Hardphones" and PC based, IP "Softphone" applications to make and receive phone calls.

H.450: VoIP Supplementary Services H.450 provides extended features within H.323 based VoIP networks similar in concept to QSIG within ISDN.

HTML: Hyper Text Markup Language, the authoring language used to create hypertext documents for the World Wide Web.

HTTP: Hyper Text Transfer Protocol, the application protocol for moving hypertext files across the Internet. The protocol requires an HTTP client program on one end of a connection and an HTTP server program on the other.

I

ICLID: Incoming Caller ID. See CLI.

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.

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 successful 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 LDAP-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.

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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B

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

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

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

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