Hosted VoIP Feature List

Telephone: 01438 960 060

Flip Solutions Ltd t/a Flip Connect, Unit 5/6, 29 Mill Lane,
Welwyn, Hertfordshire, AL6 9EU

Station Features

Abbreviated Dial	This feature lets a user create, manage and use a personal speed dial list of up to 100 entries that is activated using a star code (*3), plus a two digit code. For example, if a phone number has been abbreviated to 55, then the user would dial *355
Call Park	This feature lets a user park a call. Parking a call is similar to placing a call on hold, except that when a call is parked, the associated line is freed for normal use (unlike hold, which ties-up the line). The user can then retrieve the parked call by pressing the Park button from any IP phone or by using the appropriate code from an analogue DTMF telephone.
Call Pick-up	Allows a user to answer another phone by pressing the Pick-up button. (Both phones must be members of the same Group Pick-up, as determined by the system administrator.)
Call Transfer	The Call Transfer feature lets a user transfer a call to another internal extension or external number.
Call Waiting	Lets a user receive and answer a call on the same line that is currently busy. If desired, the user can then use a feature button to switch back-and-forth between the two calls.
Call Waiting with Caller ID	Users with both the call waiting and caller ID capabilities activated can see the caller ID of a call waiting call if they have a display on their telephone
Call Waiting / ID Manager	While already on a phone call, this feature allows the user to view the Caller ID of a second incoming phone call and decide how the second call should be handled. The user has four (4) options: • Answer the new call and put the current on hold (already supported through the use of the Hold feature) • Immediately send the call to the user's Do Not Disturb (DND) destination. (The DND destination may be voicemail, another telephone, an announcement, etc.) • Send a "Please Hold" announcement to the incoming caller. • Send a "Call Me Back" announcement to the incoming caller.
Day, Date, and Time Display	The day, date, and current time is always displayed on display-enabled IP phones.
Denied Origination	Blocks calls from an extension based on calling rules set by the system administration (i.e. no international, no long-distance calls, etc.)
Direct Extension Assignment	This feature allows a user to re-program a telephone to be his/her telephone. For example, the user is visiting a remote office and wants to program a “guest” telephone to ring as his/her telephone
Directed Call Pick-up	The Directed Call Pick-up feature allows a user to pick-up an incoming call that is ringing on another Line or Hunt Group.
DDI (Direct Dial In)	This feature allows outside callers to directly dial an internal extension, bypassing an operator or auto-attendant.

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Do not Disturb	The Do Not Disturb feature allows a user to block incoming calls and still be able to make calls and use other telephone features. When the feature is activated, incoming calls are handled in one of the following ways: • Voicemail, if the user has a voice mailbox. • The Do Not Disturb destination programmed for that user in the Service Administrator. • If no action has been taken, the caller hears busy.
Do not Disturb Override	Allows numbers selected by the user to ignore Do not Disturb status and ring through to the speakerphone
End call	Ability to disconnect a call by pressing a phone button. This feature is mainly used when a headset is connected to the telephone through the handset jack to allow leaving the handset off the cradle while calls are made and answered using the other buttons on the phone. Calls arriving at the telephone will ring and when the line is selected, the call is connected through the headset.
External Line Access Code	Allows digit to be specified for obtaining outside line dial tone
Group Speed Dialling	This feature provides capabilities beyond those provided by speed dial buttons and the Abbreviated Dial feature. With Group Speed Dialling, a customer partition has a common speed dial list of up to 1,000 entries. Creating and managing the speed dial list is supported at the enterprise and system administrator level.
Hold	This feature lets a user place a call on hold. While a call is holding, the associated line is occupied by the holding party and it cannot be used for any other purpose. (To place a call on hold and free the line, the user can use Call Park, mentioned earlier.) After the party is on hold for one minute, the system rings the associated line to remind the user that the party is holding.
Hot Keypad Dialling	Allows users of some phones to press any number on the keypad to begin dialling a call. The user does not have to pick up the handset, select a line or turn on the speakerphone to make a call.
Hunt Groups	Calls are directed to groups of users for increased call coverage. Five different methods are used to distribute calls within a hunt group: • Top-To-Bottom • Bottom-To-Top • Longest Idle (aka Universal Call Distribution) • Round Robin (aka Circular) • Ring All
Intercom	The Intercom feature allows users to make extension-to-extension intercom calls between IP telephones with speakerphones. To use this feature, the user presses the Intercom button and dials the desired extension number. The system automatically turns on the called party's speakerphone with two fast beeps to identify the call as an intercom call and the two parties can then converse. NOTE: If the called party has a Cisco 7960 telephone, the user will have to pick up the handset to carry on a conversation.
Last Number Redial	This feature allows a user to redial the last number that was dialled.
LCD Feature Prompting	Phone display shows test prompts that match the phone state, such as reminders of voicemail codes when connected to the voicemail system
Message Waiting Indicator	A lit LED or stutter tone (special dial tone) at the phone notifies users they have a new message in voicemail
Multiple Call Appearance	Multiple extension numbers may be associated with a single physical phone. Two lines are supported for analogue phones. The number of lines supported by IP phones depend on the specific phone model (generally three)
Multiple Station Appearance	Allows a single phone number to ring multiple physical phones
Multiple Voice Mailboxes	Allows a single extension to have multiple voicemail boxes
Mute	Mute allows a user to disable and enable the speakerphone/handset microphones during a conversation to hold a private conversation that the other party cannot hear.

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Music on Hold	Flip Connect's Hosted VoIP system can play music, recorded announcements, or silence to callers on hold. Files provided with the system can be used for the music source or an analogue music source (such as a radio station) can be interfaced to the system. The music files provided with the system are: • Light Sounds Background • Easy listening Music • "Spring" Ireland • "Do Bridge" Japan • Jazz Music • "This Little Light of Mine" • "Spotlight Trance" (Harp Solo)
Night Service	Ability to route calls to alternate number (for example, directly to voice mail) during non-work hours
On-hook Dialling	Flip Connect's Hosted VoIP system supports the ability of users to dial a number while the handset is on-hook by doing one or all of the following: • Dial the phone number on the keypad and, if necessary, press a line or speaker button or softkey. • Press the desired line button and dial the number. • Turn on the speakerphone and dial the number. • Use a speed dial button. • Use the redial feature. • Use the intercom feature.
Paging	Allows users to make extension-to-extension calls between IP phones with speakerphones
Release	Release lets users disconnect from a call without having to hang up the handset.
Ringing Patterns	Depending on call conditions, different ring tones are generated. Standard Rings • One Ring -- Internal (Extension-to-Extension) Calls • Two Rings -- External Calls and system calls (reminder that a party is on hold, etc.)
Distinctive Rings	This feature allows adding up to two additional DDIs to a user's telephone to provide a different ring tone depending on the DDI number that is called.
Priority Ring	The call screening feature allows users to designate specific callers as important enough to ring through to their telephone even if they have Do Not Disturb activated. A different ring tone is provided for calls assigned the priority call option.
Selective Call Forwarding	Allows user to designate a unique call forwarding procedure to a specific, pre-defined incoming number
Selective Call Block	Allows users to permanently reject incoming calls from a specific, pre-defined incoming number
Single Call Arrangement	When DDI rings on multiple phones, only one user at a time can answer the call
Simultaneous Ringing	Allows calls to numbers assigned to multiple phones to simultaneously ring all assigned phones.
Soft Keys	Programmable phone buttons can be assigned particular features for single-button convenience
Speed Dial	This feature lets a user store a favorite phone number and then dial it by pressing the associated speed dial button.
Star Codes	Allows use of combination of the * and # and number keys to activate various phone and system features
Virtual Ring	Allows user to have callers from particular numbers to hear a ringing sound as if the call had not been answered. Ringing sound continues until the calling party disconnects.
Voicemail	Allows one-button access to voicemail, also accessible via direct dial and webportal
Voicemail Distribution List	Allows user to create up to 20 personal voicemail distribution groups. Accessed via star codes or WebPortal
Voicemail Notification	A lit LED or stutter tone at the phone notify users that they have a new message in voicemail
Volume up/down	Allows user to change the speakerphone/handset volume (also during a call)

System Features

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Alternate Routing	Allows for calls to be routed to an alternate number (for example) directly to voicemail during non-working hours
Analogue Station Support	Allows analogue phones to have most of the system features via IAD (integrated access devices) connectivity
Announcements	Play recorded announcements to the caller at various points in the call flow. Controlled by time-of-day settings.
Attendant Console Support	Supports a physical IP phone attendant console and offers also a software based solution
Automated Attendants	Virtually unlimited enterprise recordable voice messages guide callers to desired party with integrated touch-tone call routing based on programmable menus, dial by extension, time out handling, time-of-day routing, etc …
Automatic Phone Software Upgrades	Allows phone software to be automatically downloaded to a phone through the system connection.
Auto Station Relocation	Allows the phone system to properly identify and enable a phone even if it is plugged into a different physical location on the network
Browser Based Enterprise Level Administration	Dependent on the tools authorised by the service provider, allows the enterprise-level administrator to make user moves, adds and changes, class of service, corporate directory and auto attendant time-of-day changes.
Call Admission Control	A Quality of Service tool. When an outbound call is attempted, system checks a table of available bandwidth before the call is completed and returns a busy signal if there is not enough bandwidth for the call. May be used in conjunction with TOS and network engineering.
Call Log Details	Call log details to and from each extension
Codec Support	G.711 and G.729 µ-law
Conferencing Port Manager	Allows system administrator to set a limit (from 3-9) on the number of parties a user can add to an ad hoc (on demand) conference on a enterprise/partition
Company Directories	Shows extension information to all extensions in a partition/enterprise. Users may view the company directory via a PC user portal and on particular IP phones via the phone display.
Company Speed Dial Directories	Allows speed dial numbers to be created at the system administrator level and used across a group or company. Up to 1000 entries.
Configurable Star codes	Allows administrator to assign functions to star codes.
DHCP Server Compatibility	Uses a DHCP server to dynamically allocation IP addresses to IP Ethernet phones, gateways, IADS, etc. so that fixed IP addresses are not required.
Dial Tone Patterns	Allows for distinctive dial tone patterns based on internal versus external call origination, multiple DDI numbers assigned per phone, and for specific incoming number rules for priority set up by users.
Direct Extension Assignment by User	Allows user to change the extension assigned to a phone.
Direct Inward Dialling	Allows incoming calls to be routed directly to extensions.
Directory Look-up (via user portal)	Allows users to view the corporate directory and their personal directory via a PC user portal, including click to dial.
DTMF Support	Allows DTMF tones to pass successfully during a call.
999 Support	Routes 999 with CLI for termination to emergency services via the PSTN
Firewall compatibility	The Flip Connect's Hosted VoIP Proxy Firewall, located at the service provider premise, enables the system to work through the customers' enterprise firewalls and maintains security for the system.
Guaranteed Call Completion	Assuming voicemail services, allows call to be routed to voicemail if no answer.
Hold Recall	After a call has been on hold for one minute, system automatically rings the line as a reminder the call is still on hold.
Inbound Call Screening & Forwarding	Via user PC portal interface rules.
Integrated Access Device (IAD) Support	Supports a variety of industry-standard IAD devices through set-up wizards for simplified integration of the devices into the Flip Connect's Hosted VoIP system.
International Direct Dialling (IDD) Support	International dialling fully supported.

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IP Phone time Zone support	Support for multiple time zones on a single system.
Line Names	Allows names and titles to be assigned to extensions.
Multiple Administrator Access	Allows multiple administrators to have security access to the system at the same time.
Network Announcements	Provides standard network announcements for incomplete call attempts such as trunks busy, facility failures, prefix digits, area code reasons, changed numbers, etc.
Network Unified Wiring Support (IP)	Fully supported
On-Net Call Routing	Allows call originating on the system to remain on the system network when the destination is also on the system.
Out-of-Service Notification	If the system is out of service, display phones and user portal will show a related message.
Outbound Flex Routing	Outbound numbers are analysed to determine if they can be routed and delivered on-net or to an on-net PSTN gateway that can complete the call as a local call.
Partitions	Allows custom feature sets, dial plans, class of service, etc. per customer/enterprise on system. System can be set from one to thousands.
PBX/Key System Trunk Support	Allows legacy PBX or key systems to connect to CO trunk facilities through Flip Connect's Hosted VoIP' Solutions via our gateway service
Redundant, Reliable, Scalable configuration	Paired call servers and data-style infrastructure provide reliability and expandability.
Remote Maintenance	Administrative web interface enables remote maintenance from any browser.
Self-Provisioning	Enterprise can provision new lines real-time.
Standards-based	Flip Connect's Hosted VoIP system architecture is based on many industry standards: Ethernet, TCP/IP, HTTP, Cisco skinny, SIP, MGCP, G.711, G.729A
TAPI Compatibility	Provides a desktop application compatible with Microsoft's Telephony Application Programmer's Interface (TAPI). This compatibility allows users to dial phone numbers from Outlook.
TCP/IP	Complies with industry-standard TCP/IP protocols for all communication between system components.
Time-of-Day-Class of Service	Allows class of service to be set based on time of day and day of week. Example: blocking international calls at night.
Free Phone Calling	Allows Free Phone calls to be dialled.
Type of Service (TOS) Support	Supports TOS which allows system to priority voice packets sent to other network devices.
Unlimited customer Partitions	Allows custom feature sets, dial plans, class of service, etc. per customer/enterprise on system. System can be set from one to unlimited.
Wide Variety of Supported Phones	Supports certified SIP, MGCP, & SCCP phones.
Voice Virtual Private Network	Provides a single homogenous dial plan independent of location

Personal Phone Manager (PPM)

Flip Connect's Hosted VoIP PPM is web-browser-based application that allows the user to manage calls from your PC through an easy-to-use web browser interface. If the user is using the PPM on a PC that is connected to Flip Connect's Hosted VoIP platform, he can use point-and-click capabilities to control and manage calls.

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PPM Call Control Functionality

The PPM Call Control window provides information about the line on the user's phone and calls in progress as well as providing basic call control functionality.

• Line Information - This line of text displays the 10-digit number line currently is use under the time and date. It also has the Help link.
• Call Progress Display - When making or handling a call, this display shows progress messages related to the call.
• Dial Keypad - A standard telephone dial that can be used for dialling telephone number by clicking on the keypad buttons
• Call Control Buttons - These buttons allow you to perform call control functions from the PPM including hanging up a call (Release), placing a call on Hold or Park or sending the call to another user (Transfer). You can also activate and deactivate the Do Not Disturb (Dnd) feature. Also provided is a text box that you can use for typing in telephone numbers for transfers.
• Speed Dials List - If you have speed dial numbers on your telephone, they will appear in the Speed Dials area and can by used by clicking on them.

The following call control functionality is available:
• Making a call
• Transferring a call
• Hanging up a call
• Putting a call on Hold
• Parking a call
• Picking up a parked call
• Picking up a Multi-line parked call
• Using Do Not Disturb
• Using Speed Dials

PPM Call Management Functionality

The PPM provides several tools for managing calls. They include:
• Call Log - Provides a record of calls made and received from an specific telephone number including whether the call was answered or not.
• Find-Me Forwarding - Allows the user to configure his phone so that incoming calls are routed to other phone numbers if the user doesn't answer the phone.
• Call Treatment - This tab provides various methods of screening incoming calls and routing or forwarding calls.

PPM Directories

Flip Connect's Hosted VoIP PPM provides two directories "types" - Corporate and Personal. The corporate directory provides a listing for everyone in the company while the personal directory is where the user keeps all his personally important telephone numbers.
• Sorting the Directory - both directories allow users to sort using the column headings. In the corporate directory these are last name, first name, title and extension while in the personal directory these are last name, first name and company.
• Screening Calls - both directories also allow users to screen calls - apply call handling rules for a specific caller - with a few simple clicks of your mouse.
• Navigating the Directory- The Corporate Directory provides both a paging and search tool for finding company employees. The Personal Directory requires users to scroll through entries. When combined with the sort functions, users should be able to easily find the desired information.

Conferencing Scheduling using the PPM	The Meet-Me Conference feature provides a list of scheduled conferences that multiple callers can dial into and be connected. Anyone can participate in a Meet-Me conference, but to schedule a Meet-Me Conference, the user administrator must allow the user access to the scheduling function. If the user has not been granted scheduling permission, selecting the Conferences menu item will return a message that he is not authorised to schedule conferences. From the PPM the user can do the following: • Schedule (Add) a Meet-Me Conference (with the proper authorisation) • Find a Meet-Me Conference • Modify a Meet-Me Conference • Delete a Meet-Me Conference • Sort the conference listing
Voicemail Management using the PPM	PPM offers access to and management of a user's voicemail using the following tools: • Accessing Voicemail Inbox • Managing Saved Voicemail • Paging Notification • Managing Voicemail Distribution Groups • Changing the Voicemail Password
Using the PPM Options	The PPM Options menu provides functions for the PPM application as well as provides some configuration functions for the phone. • Change Password – Allows users to change the password for their telephone and PPM. • Remote Phone – Allows users to set up a remote phone to act as if it were the systems phone. Instead of placing and receiving calls from the system phone, calls are made and answered from the Remote Phone. The CallerID presented on outgoing calls and the features available to the user while connected remotely make it appear to all concerned like the user is calling from the system phone. • Reassign Phone – Allows users to reassign their phone number to a different physical phone. Users can also unassign the phone number and place the phone in "out of service" mode. • Modify User Profile – Allows users to configure specific features of the PPM.
PC Integration / TAPI Interface	Allows users to download and run Flip Connect's Hosted VoIP TAPI application that allows TAPI-enabled programs, such as Microsoft Office, to dial through Flip Connect's Hosted VoIP system
Import Contact Information (e.g. from Outlook)	Users can import an existing contact list which was exported from Outlook, Act or other "organiser" program as a comma separated value (CSV) file.
Voicemail management	Voice Messages, which are saved as a .wav file, can be played directly from the PC
Click-to-dial	Users can click on numbers in the directory, call log or voicemail to start a call.
PPM Branding	Service Provider can completely change the WebPortal "look and feel" by using Flip Connect's Hosted VoIP powerful APIs. For example, the user provider can change the color, logo, layout, functionality, etc. of the WebPortal

Conferencing Application

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Through Flip Connect's Hosted VoIP conferencing application carriers can deliver conferencing services to their customers without deploying conferencing hardware at the
customer's premises.

Flip Connect's Hosted VoIP system supports the following types of conferencing:

• On-demand or ad hoc - Users can create conferences using PPM's one-step conferencing, feature buttons or star codes within parameters set by system administrators.

• Meet-Me - Authorised users can reserve conference ports for a specific date and time when users, who have been provided with the proper telephone numbers and passwords, can call in and join the conference.

On-demand or ad hoc Conferencing	Flip Connect's Hosted VoIP platform provides support for On-Demand Conferencing via the conference button on an IP phone, by using flash hook on an analogue phone, or by clicking on entries in the PPM GUI (graphical user interface). On the WebPortal a user can, while in an existing call, add another party with one-click selection from the directory or by entering the other party's phone number. The conference is established automatically without putting the current call on hold and pressing the Conference button.
Meet-Me Conferencing	Meet-Me Conferencing provides the ability to reserve conference bridge ports for a conference call to be held at a schedule date and time. The moderator (who has control of the conference) and other conference call members dial into the conference and are connected (at the appropriate time) to the conference call. System administrators and authorised users can schedule Meet-Me Conferences. Administrators schedule Meet-Me Conferences through the Service Administrator while authorised users can use their phone or the PPM to schedule conferences. Users can create and participate in Meet-Me Conferences from both internal and external telephones if DID numbers for outside access have been created in VocalData's system. Other key features of the Meet-Me Conference feature include: • Base unit for schedule time and duration is 5 minute increments • Users can record a name that is announced after the join tone when they join a Meet-Me Conference • Roll call plays back a list of all conference participants to the moderator • Moderators can add additional ports (if available) to an active conference session • Moderators can extend conference calls in 5 minute increments if ports are available • A dedicated port is provided for each conference for announcements played to all participants • Class of Service settings (Moderator - No Restrictions) control which users have the ability to schedule Meet-Me Conferences • Meet-Me Conferences take precedence over ad hoc conferences (ad hoc conferences will be terminated to supply ports needed for a scheduled Meet-Me Conference) • Log file provides records of Meet-Me Conference calls for billing purpose
Conferencing Port Manager (Administration Feature)	This feature allows system administrators to set a limit (from 3-9) on the number parties a user can add to an ad hoc (on demand) conference on a customer (partition) basis. Setting the limit to zero allow users unlimited access – dependent on the availability of conferencing resources. All conferencing restrictions on whether conferences can include external parties remain in effect.

Advanced ACD

The Flip Connect's Hosted VoIP' Advanced ACD application can receive calls from outside callers (for example, to a main company number or a department) and evenly distribute these calls among agents trained in a particular product, business area, etc. While waiting to be answered by an agent or employee, callers can hear a mixture of
announcements, music on hold and/or advertisements. Different types of hunt groups are supported: multi-line hunt groups, directory number hunt groups, and PBX trunk hunt groups, each with a variety of available call distribution patterns.

Flexible Agent support	Agents in multiple queues - Agents sign-in/out of each queue
Call Routed to Agents	Top-to-Bottom and Bottom-to-Top | Longest Idle – The phone left idle the longest receives the next call in the queue. | Round Robin - The phone defined within the hunt group to be "next" receives the call .
Network API for development of 3rd party real-time applications	Provides for calling information, time/number in queue, abandoned calls, calls to voicemail, agent idle/call duration, etc.
Very flexible call treatment	Unlimited number of queues per system and per partition Unlimited number of Agents per queue Unlimited number of announcements based on waiting time, time-of-day, number in queue Early withdrawal to voicemail at anytime Overflow to voicemail or another queue

Console Assistant

Flip Connect's Hosted VoIP VoIP Console Assistant is an application that can be used to manage incoming calls to a main telephone number at a particular site. For example,
a company receptionist can use Console Assistant to answer and transfer calls to the extensions listed in the directory. Incoming calls can also be placed on hold or camped on an extension. Console Assistant also allows the receptionist to retrieve and manage voicemail for the main company number.

Console Assistant vs. PPM

Flip Connect's Hosted VoIP VoIP Console Assistant application provides call-handling features for an attendant in a call-intensive environment and provides buttons for commonly used features for Answering, Releasing, Transferring, Parking and Camping calls. PPM is designed for the individual VoIP phone user and has many more features that are specific to a user's VoIP personal phone.

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Console Assistant and Displays

Console Assistant provides a variety of information about the calls being handled. The information collected by various Console Assistant components and is organised into categories that are marked by tabs. The Console Assistant components and the information each provides are:

• Call Information - displayed by Console Assistant as a list of calls that are or on hold at the Console Assistant phone. Above and below the list are buttons for handling the calls.

• Call Log Tab - Incoming and outgoing calls are logged automatically by Console Assistant when it is active. The information displayed in the Call Log includes the direction of the call (In or Out), the phone number of the called or calling party (if available), the name of the called or calling party (if available), date and time of the call, and the duration of the call.

• Options Tab - allows the user to set operational parameters for Console Assistant as well as changing which functions display confirmation dialog boxes.

• Directory Tab - displays all of the phones in the Corporate Directory and is automatically populated when Console Assistant starts. The list is defined by Flip Connect's Hosted VoIP Administrator and cannot be modified from the Console Assistant application. The user may handle calls using the list, just the same as using the
Phones tab. The user can also add entries from the Directory Tab to the Phones List. by clicking a Directory entry and clicking Add.

• Phones Tab - accesses the Phones list. This is the primary screen that is displayed when Console Assistant starts. It displays extensions that have been added from the Directory Tab. This list will usually be a subset of the complete system directory.

• Voicemail Tab – presents the Voicemail message area with control buttons. The

Available Softbuttons

Several softbuttons are available to control system functionality:
• Announced Transfer
• Answer
• Blind Transfer
• Call
• Call Any Number
• Camp-On
• Hold
• Intercom
• Message Center
• Park
• Priority Transfer
• Redial
• Release
• Transfer Any Number

Console Assistant Attendant Customiszation

Console Assistant can be done to suit your needs and to make the program more "user friendly". Four Console Assistant components can be customised:
• toolbars,
• headers,
• the number of columns used in the Phones Tab display,
• call information colors.

Call Handling	How Console Assistant handles a call depends on whether the call is incoming (being answered), being transferred, or outgoing (being made). • Incoming calls - Console Assistant provides call information and allows you to handle incoming calls in the following ways: Answer a Call, Answer a Call Waiting Call (Double-click on incoming call), End a Call, Hold a Call, Park a Call • Transferred calls - Console Assistant allows user to perform the following call transfers: Announced Transfer, Blind Transfer, Priority Transfer, Message Center Transfer, Transfer to Any Number (, Camp-On a Call • Outgoing (or dialled) calls - Console Assistant allows you to place calls by using the following methods: Call Any Number, Call a Selected User, Transfer to Any Number
Call Log Tab	Incoming and outgoing calls are logged automatically by Console Assistant when it is active. The information displayed in the Call Log includes the direction of the call (In or Out), the phone number of the called or calling party (if available), the name of the called or calling party (if available), date and time of the call, and the duration of the call. Users can: • Sort Entries • Group Entries • Search Entries • Set Field Width • Delete a Call Log Entry • Dial Back a Call Log Entry
Voicemail Tab	The Voicemail Tab is the Console Assistant component that allows the user to operate the Voicemail function. From the Voicemail Tab, user can do the following: • Voicemail Tab Overview • Access the Voice Mailbox • Customise the Toolbar • Customise Headers • Listen to Messages • Save Messages • Delete Messages • Forward Messages • Forward Voice Messages Dialog Box Help • Callback Caller • Sort Entries • Group Entries • Search Entries • Set Field Width • Change the Voicemail Password • Use Intercom
Keyboard Shortcuts	Keyboard shortcuts can be used in place of mouse clicks to execute frequently used button presses. In command lists and paragraph titles, button presses that have a keyboard shortcut have the shortcut given in parentheses.

Enhanced Platform Functionality

Voicemail & Unified Messaging

Flip Connect’s Hosted VoIP voicemail is an integrated solution that includes all the necessary software to offer a variety of voicemail solutions including traditional voicemail, visual voicemail and unified messaging.

• Traditional Voicemail - allows voicemail to be accessed in the traditional fashion via desktop or other phone.

• Visual Voicemail - provides users with a whole new set of productivity enhancements. End-users can access their voicemail via the web portal using any standard web browser. Voicemail messages are converted into files that can be played through the computer, office phone, or forwarded to another user.

• Unified Messaging - enables end-users to combine their voicemail and email communications into one single messaging inbox. This gives the users one mailbox to check for all their communications needs. Voicemail messages can be treated the same way as email messages - forward, save, delete, print, reply, and file.

Remote Phone Feature

Extends the convenience of the office phone to remote phone users. When Remote Phone mode is enabled from the Web Portal or phone, incoming calls ring both the office phone and assigned remote number (much more flexible than call forwarding). Missed calls all go to office voicemail for centralised message handling, rather than having to check both remote and office mail systems. If a Remote Feature user forgets to change the remote number when changing locations, they can easily change it while on the move by phone or Web Portal.

Remote Phone also provides benefits for outgoing calls. End-users can choose to have their office phone caller ID show as the call originator when making calls from home or cell phone when dialling from the Web Portal in remote status. In addition to providing privacy, any long-distance or international calls are conveniently and automatically billed to the office phone. Flip Connect’s Hosted VoIP Remote Phone outbound dialling can also be accessed by any touch-tone phone when the Web Portal isn't available.

Softphone Support

Softphones are an ideal option for businesses with many mobile workers, because it allows travelers to take a laptop into a hotel room, log onto a Web portal and make and receive calls across the in-room high- speed Internet connection - just as if they were at their desk. Telecommuters can use the phones in a similar manner using their high-speed connections at home. Softphones is also an important productivity tool for call center agents.

Flip Connect’s Hosted VoIP VPN

Flip Connect’s Hosted VoIP can easily be used to provide a VoIP VPN.
Flip Connect’s Hosted VoIP VPN solution provides the benefits of private dial plans and cost savings through on-net voice calls within the paradigm of service provider's managed network. Whether an endpoint is a large office or a single user on an IP phone, all business locations can receive the same rich set of features without the need or cost of installing a PBX in each location.

Some of the Flip Connect’s Hosted VoIP features especially useful to a multi-office business customer include 4-digit dialling between locations, "on-net" phone charges for cross- location calls, and cross-location use of call transfer, call groups, conferencing, corporate directory.