

VIPedge™

General Description

Publication Information

**Toshiba America Information Systems, Inc.
Telecommunication Systems Division**

Publication Information

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VIPedge General End User Information

FCC Requirements

Means of Connection: The VIPedge does not connect directly to the telephone network. All direct connections are made to a gateway. Please refer to the gateway manufacturer's documentation

Radio Frequency Interference

Warning: This equipment generates, uses, and can radiate radio frequency energy and if not installed and used in accordance with the manufacturer's instruction manual, may cause interference to radio communications. It has been tested and found to comply with the limits for a Class A computing device pursuant to Subpart J of Part 15 of FCC Rules, which are designed to provide reasonable protection against such interference when operated in a commercial environment. Operation of this equipment in a residential area is likely to cause interference, in which case, the user, at his/her own expense, will be required to take whatever measures may be required to correct the interference.

Underwriters Laboratory

This system is listed with Underwriters Laboratory (UL). Secondary protection is required, on any wiring from any telephone that exits the building or is subject to lightning or other electrical surges, and on DID, OPS, and Tie lines.



CP01, Issue 8, Part I Section 14.1

Notice: The Industry Canada label identifies certified equipment. This certification means that the equipment meets certain telecommunications network protective, operational and safety requirements as prescribed in the appropriate Terminal Equipment Technical Requirements document(s). The Department does not guarantee the Equipment will operate to the user's satisfaction.

Repairs to Certified Equipment should be coordinated by a representative designated by the supplier. Any repairs or alterations made by the user to this equipment, or equipment malfunctions, may give the telecommunications company cause to request the user to disconnect the equipment.

Users should ensure for their own protection that the electrical ground connections of the power utility, telephone lines and internal metallic water pipe system, if present, are connected together. This precaution may be particularly important in rural areas.

Before installing this equipment, users should ensure that it is permissible to be connected to the facilities of the local telecommunications company. The equipment must also be installed using an acceptable method of connection. The customer should be aware that compliance with the above conditions may not prevent degradation of service in some situations.

CAUTION! Users should not attempt to make such connections themselves, but should contact the appropriate electric inspection authority, or electrician, as appropriate.

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Refer to http://www.telecom.toshiba.com/Telephone_Systems_Support/VIPedge-Terms/terms-and-conditions.pdf

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Contents

Introduction

- Organization xi
- Conventions xii
- Related Documents/Media xii

Chapter 1 – Overview

- Product Line 2
- Advantages 2
- Deployment 2
- Administration 3
 - Enterprise Manager 3
- Ordering and Billing 4
 - Local/Long Distance 4
 - Toll Free 4
 - International Calling 4

Chapter 2 – Unified Communications

- Messaging 7
 - Follow-Me 7
 - Unified Messaging 7
 - Other Messaging Features 7
- Call Manager 8
 - Companion Applications 9
- Call Manager Mobile 11
 - Benefits 11
 - Features 12
 - Devices Supported 14
- IP User Mobility 14

Chapter 3 – Telephones

- IP Telephones 15
 - Features 15
- Telephone Button Expansion Options 17
 - LCD Add-on Module (LM5110) 17

Chapter 4 – Networking

- Preplanning for VoIP Deployment 19
 - Benefits 19
 - Requirements 19

Interactions	20
Wide Area Network (WAN)	20
Benefits	20
Requirements	20
Interactions	20
Router/Firewall	21
Benefits	21
Requirements	21
Interactions	21
LAN Deployment	21
Benefits	21
Requirements	21
Interactions	22
Remote Administration	22
Benefits	22
Requirements	22
Interactions	22
Remote IP Telephones	23
Benefits	23
Requirements	23
Interactions	23
Remote SIP Telephones	23
Benefits	23
Requirements	23
Interactions	23
Survivability	23

Chapter 5 – Features

Add-on Module (ADM)	25
Advisory Message	25
Alarm Notification	25
Alternate Answer Point	25
Automatic Busy Redial	25
Automatic Callback (ACB)	26
Automatic Line Selection (ALS)	26
Call Completion	26
Call Forward	27
Call History	27
Call Manager	27
Call Park	28
Call Pickup	28
Call Transfer	29
Call Waiting	30
Caller Identification	31
Class of Service	31
Conference Call	32
Day/Night Mode	33
Dial Directory	34
Direct Inward Dialing (DID)	34

DID Parking	34
Dialed Number Identification Service (DNIS)	35
Directory Numbers	35
Distinctive LED Indicator	36
Distinctive Ringing	36
Do Not Disturb (DND)	36
Emergency Call	37
Enhanced 911 (E911)	37
Feature Prompting With Soft Keys	37
Flexible Line Ringing	37
Flexible Numbering	38
Handsfree Answerback	38
Headset	38
Hearing Aid Compatible	38
High Call Volume Buttons	38
Hold	38
Automatic Hold	38
Call Hold	38
Consultation Hold	39
Exclusive Hold	39
Hold Recall	39
Hot Dialing	39
Hotline Service	39
Line Buttons	39
Pooled CO Line Button	39
Group CO Line Button	40
Message Waiting	40
Mobility	40
Music/Messages On Hold	41
Multiple Call/Delayed Ringing	41
Off-hook Camp On	41
Override	42
Call Forward Override	42
Class Of Service Override	42
Do Not Disturb (DND) Override	42
Executive Override	42
Privacy Override	42
Paging	43
Telephone Group Paging	43
Emergency Page	43
Privacy	43
Repeat Last Number Dialed	43
Ringling	44
Ring Over Busy	44
SIP Trunk	44
SIP Trunk Wizard	44
Specified Caller ID	44
Speed Dial	45
One Touch Buttons	45
Station Hunting	46

Serial Hunting	46
Circular Hunting	46
Distributed Hunting	46
Camp on to Hunt Groups	46
Transfer Direct To Voice Mail	47
Uniform Call Distribution	48
Unified Communications	48
VLAN Tagging	49
Voice / Tone Signaling	49
Volume Control	50
Messaging	50
Automated Attendant	50
Voice Messaging	52
Unified Messaging	57
Administration	58
Reporting	60
Security	61
Appendix – Specifications	
Datacenter Specifications	63
Connectivity	63
Power	63
Security	63
Environmental	63
Network Operations Center	63
Physical Isolation	63
Station Dimensions	64
IP Telephone Power Consumption	64
System Tones	65
IP Telephone Bandwidth Requirements	67
VIPedge Capacities	69
Application Capacities	71
Part Numbers	72
Index	75

Introduction

This General Description provides an overview of the VIPedge cloud-based business telephone solution, associated hardware, features, capabilities, and capacities.

Organization

This document is divided into the following major topics:

- **Chapter 1 – Overview** is a brief introduction of the VIPedge system, environmental and power considerations, related software, administration, configuration, and network requirements.
- **Chapter 2 – Unified Communications** describes the Messaging, Call Manager, Meeting, and Mobility Solutions which together form Toshiba's Unified Communications product suite.
- **Chapter 3 – Telephones** describes the most recent Toshiba-proprietary stations and peripherals, customer-supplied peripherals, as well as cabling and connectors.
- **Chapter 4 – Networking** describes the various network related configurations that need to be done when deploying a VIPedge solution.
- **Chapter 5 – Features** describes the features which are available system-wide, as well as stations features.
- **Appendix – Specifications** includes detailed information on network requirements, station dimensions, system tones, hardware compatibility, software license requirements, and capacities.

Conventions

Conventions	Description
Note	Elaborates specific items or references other information. Within some tables, general notes apply to the entire table and numbered notes apply to specific items.
Important!	<i>Calls attention to important instructions or information.</i>
Courier	Shows a computer keyboard entry or screen display.
“Type”	Indicates entry of a string of text.
“Press”	Indicates entry of a single key. For example: Type prog then press Enter .
Plus (+)	Shows a multiple PC keyboard or telephone button entry. Entries without spaces between them show a simultaneous entry. Example: Esc+Enter . Entries with spaces between them show a sequential entry. Example: # 5.
Tilde (~)	Means “through.” Example: 350 ~ 640 Hz frequency range.
Start > Settings > Printers	Denotes a progression of buttons and/or menu options on the screen you should select.
See Figure 10	Grey/Blue words within the printed text denote cross-references. In the electronic version of this document (Library CD-ROM or FYI Internet download), cross-references appear in blue hypertext.

Related Documents/Media

Installation and Programming Manuals

- VIPedge Installation
- VIPedge Feature Description and Implementation

User Guides

- VIPedge Telephone, Messaging, and Call Manager

Quick Reference Guide

- VIPedge IP5000-Series Telephone

Internet Site

For *authorized users*, Internet site FYI (<http://fyi.tsd.toshiba.com>) contains all current VIPedge documentation and enables you to view, print and download current publications.

Toshiba's VIPedge cloud-based business telephone solution is designed for organizations needing robust unified communication features without having to own and maintain a telephone system.

The VIPedge solution is based on the feature-rich functionality as an on-premise Toshiba IPedge® business telephone system. VIPedge supports the high-quality Toshiba endpoint's, including IP5131-SDL, IP5531-SDL, IP5631-SDL desk telephones, and IP4100 DECT wireless telephone.

VIPedge offers a solution alternative to buying a telephone system. Customers pay a recurring monthly service fee, based upon size and usage, instead of purchasing or leasing the telephone system.

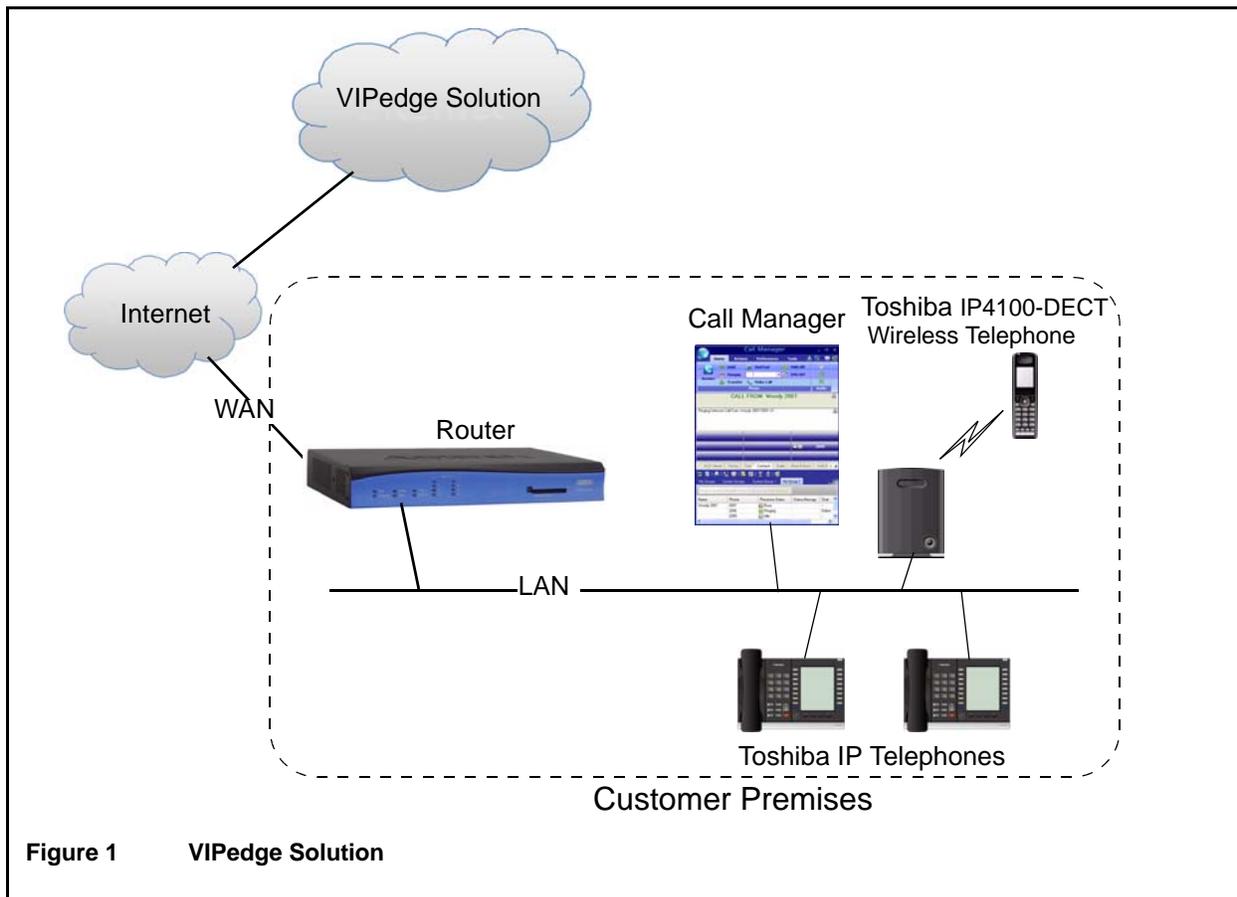


Figure 1 VIPedge Solution

Product Line

VIPedge, Toshiba's cloud-based business telephone solution is based on the technology and software centric capabilities of the Toshiba IPedge telephone system. VIPedge targets small to medium enterprises and branch locations of larger enterprises. This provides a reduced capital alternative to buying or leasing an on-premise system. VIPedge expands Toshiba's product line providing customers more choices of how their communications capabilities are delivered. Customers can choose from a VIPedge cloud telephone solution, an on-premise IPedge pure IP communication system, or a Strata® CIX® hybrid converged IP and digital telephone system, whichever best fits their needs.

Customers get similar capabilities regardless of which delivery method they choose. Best of all, customers who start with the VIPedge solution can easily migrate to their own on-premise IPedge system in the future, while retaining their investment in Toshiba endpoints.

Advantages

The key advantages of the VIPedge solution are as follows:

- Comprehensive capabilities at your fingertips.
 - Competitive business telephone system features
 - Voicemail with Unified Messaging
 - Mobility with Find-me Follow-me
 - Call Manager unified communications with presence, IM/chat, and call control from your PC
 - And much more
- Full-Featured IP-PBX Functionality in the Cloud
 - IPedge system has been virtualized in the Cloud
- Unified Messaging and Unified Communications are standard
- Toshiba Quality of Service
- Strong TSD Dealer Channel
- Highly Competitive Price/Feature Ratio
- Toshiba IP5131-SDL, IP5531-SDL, IP5631-SDL, and IP4100 interoperability and easy migration to premise-based IPedge solution.

Deployment

The VIPedge solution runs in the cloud which makes deployment quick. These are the basic steps:

- Provision the customer's telephones in the VIPedge Management Portal.
- Configure the customer's telephone system through the dealer's login to the VIPedge solution Enterprise Manager.
- Install and Configure the router.
- Configure the LAN for QoS.
- Install the telephones on the LAN.
- Verify the Installation.
- Train the customers.

Each VIPedge system starts up with a default database. This default includes a 3-digit station numbering plan and IP Telephone button programming. Dealers can add customer specific information, such as names and email addresses, or they have the option of entering a custom configuration.

Administration

VIPedge Enterprise Manager provides a web interface for dealers to configure data, manage, control and maintain all components of the solution. Since the VIPedge solution is cloud based, management and administration are always accessible remotely over the internet. An authorized TSD Dealer is able to log in and view the VIPedge solutions of their customers, then select a system and manage it.

Enterprise Manager

VIPedge Enterprise Manager is a web-browser based administration tool that unifies the programming of both the Call Processing features and the Messaging Voice Processing features.

- Enterprise Manager combines administration and management of the two previously separate applications, telephone system and voice mail into combined menus, allowing technicians and system administrators to program both together, and eliminate many duplicate steps.
- Wizards are added to support setup on integrated screens.

Enterprise Manager can be accessed locally over the LAN or, with proper network security, remotely over the Internet, and because it is used from the user's Web browser, no special software is required to be loaded on the user's client PC. The Web browser must be Windows Internet Explorer 7.0 or Firefox 5.0 and above. The Mozilla add-on 'IE TAB' is required to properly view some Enterprise Manager pages in Firefox.

Toshiba's personal administration tool integrated in Enterprise Manager using a User login, puts telephone personalization in the hands of individual users, allowing them to easily program speed dial buttons, feature buttons, Do Not Disturb functions, and even the name on their LCD displays, using their PC's Web browser. The benefits of this tool are as follows:

- No special software is required to be loaded on the user's client PC.
- Administrator support is reduced because individual station users can program their own telephones themselves.
- Using Enterprise Manager, every person within the organization can customize their telephone to incorporate the features they use the most.

Ordering and Billing

An ordering portal is used for VIPedge orders (www.VIPedge.com). VIPedge solutions are billed directly to the end customer. Dealers and customers work together to create and submit the customer order. When the order is submitted, the customer will be emailed with a request to provide credit card information. VIPedge solutions and equipment must be pre-paid with a credit or debit card by the customer. After this information has been provided and verified, the dealer will be notified via email that the order has been completed.

Local/Long Distance

Local/Long Distance for the continuous 48 states for Call Center type customers will be billed on a per minute basis for incoming and outgoing calls.

Toll Free

Toll Free is available for \$0.039 per minute for incoming calls from all 50 states. Toll Free Numbers can be assigned to VIPedge Business Standard Users or VIPedge Auto Attendants.

International Calling

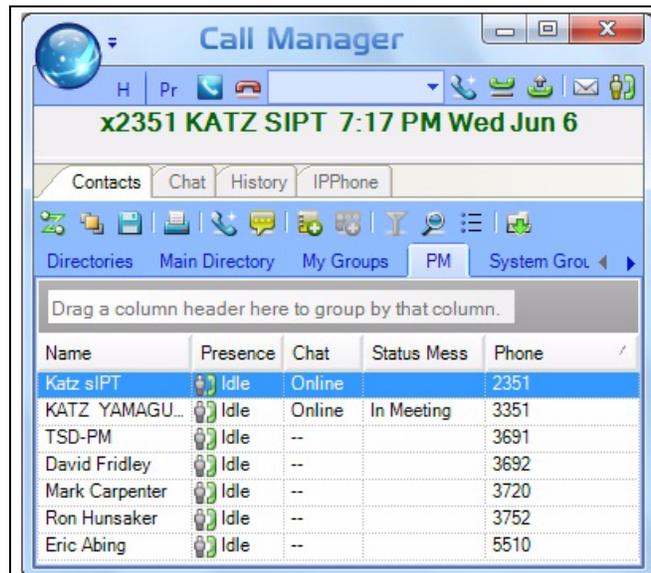
International calling can be enabled or disabled and will be billed on a per minute basis based on current rates. Here are some current examples

Area Code	Country – City	Price (\$ per minute)
8610	China - Beijing	\$0.0192
1289	Canada - Ontario	\$0.02
331	France - Paris	\$0.0204
813	Japan - Tokyo	\$0.0522
5255	Mexico - Mexico City	\$0.0248
Note Rates are subject to change. See VIPedge.com for latest.		

This chapter describes Messaging, Call Manager (CM), Meeting, and Mobility Solutions which together form Toshiba's Unified Communications product suite. This solution supports all Unified Communications (UC) applications on one platform, dramatically decreasing the cost and complexity of deploying multiple applications. This includes Presence, IM/Chat, PC call control, Auto Attendant, Voice Mail, Unified Messaging, Interactive Voice Response (IVR), and Enterprise Manager system administration.

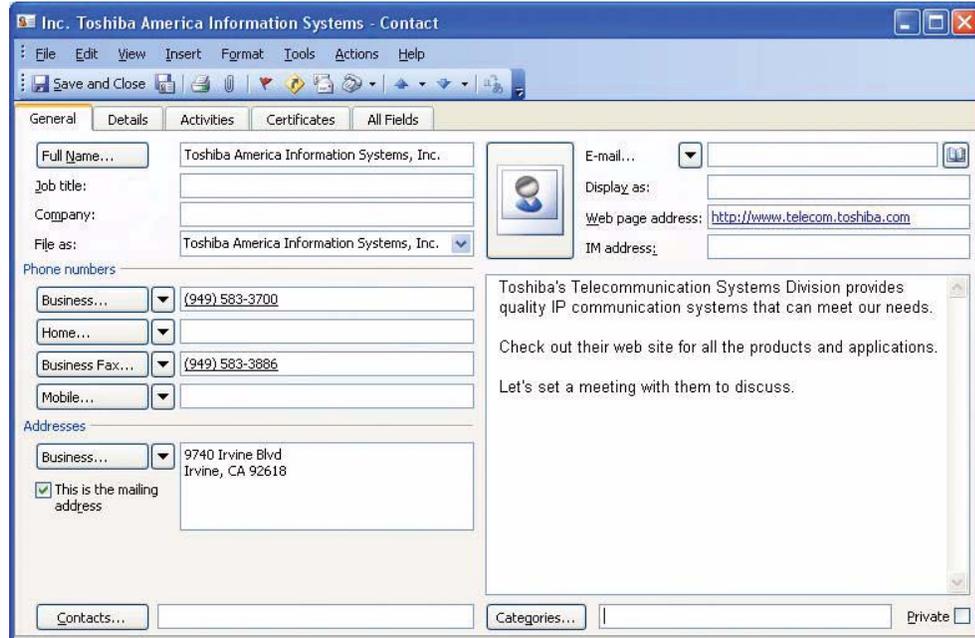
The Unified Communications product suite is easy to acquire, deploy, manage, and use. You can select the mix of modular capabilities that meet the specific needs of your business. Some of the key features and benefits of these tools are mentioned below:

- **Presence and Instant Messaging** – With a presence viewer you can see the status of other users, both their telephone busy/idle status and calendar status from Outlook integration, with the ability to click on the name to either call or instant message chat with them. You can decide the best way to contact someone to maximize efficiency.
- **Outbound Dialing from Any Application** – Making a call is as easy as highlighting a number and clicking the mouse. You can also launch electronic documents, applications and web pages directly from the CM interface for quick access to the most frequently used communications tools. This saves you valuable time.
- **Desktop Call Control** – Using the Toshiba Call Manager (CM) desktop call manager application, you can combine the capabilities of your computer and telephone to dial, answer, or transfer calls, and more, using your mouse without ever picking up the telephone. Clicking on features make call transfer, speed dialing, and other functions faster and easier. CM can be used at your desk with your desk telephone or as a stand-alone IP soft phone providing mobility and remote access. You get the efficiency of combining your telephone and computer into one integrated communication tool.
- **CRM Integration and Screen-pops** – Your call answering personnel can provide better service by immediately knowing which customer is calling with screen-pop integration to your

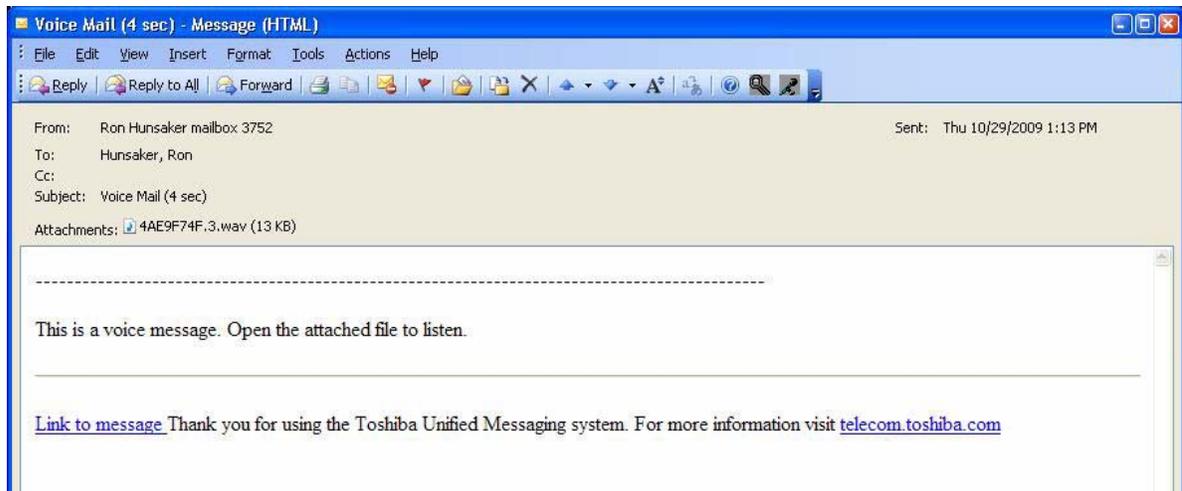


Call Manager Presence Screen

customer relationship management applications and databases. This saves you time and serves your customers better.



- **Off-premise Call Forwarding** – Enables your incoming calls to reach you when you're out of the office and enables you to change your forwarding destination from any remote location. You can stay in touch no matter where you are.
- **Unified Messaging** – You can access your voice messages from your email inbox, providing the convenience of checking all your messages from one place. Web-based unified messaging adds mobility allowing users access to their email, voice by using only an Internet browser, without even needing to first access their email providers. This is especially useful when away from the office; enabling mobile users to access and manage their voice from anywhere they have internet access.



- **Remote Connection and Mobility** – You may have a mix of on-site employees, telecommuters who work at home, mobile employees, and personnel in remote branch offices. It's important to improve employee productivity for all of them no matter where they are. Toshiba provides the tools for remote connectivity and mobility to make them all operate as if they were right there in the office.

Messaging

Messaging is an integrated voice processing application that provides standard voice mail and Automated Attendant features as well as Unified Messaging capabilities, Follow Me, Message Notifications, Soft Key navigation of mailbox menus, and Call Recording.

Messaging delivers streamlined user administration and system management. With Messaging, users can easily and conveniently manage their voice messages with intuitive on screen prompts. Users can program different types of greetings, call routing, and message notifications.

Some key features include:

Follow-Me

A mailbox can be set up to forward a call to an external phone number before the call is transferred to voicemail. When using supervised follow-me, the mailbox owner can perform functions such as record the call, conference in another subscriber, or send the caller back to the mailbox owner's voicemail box. This feature is a part of the UC features that allow users to flexibly control the call based on a user's requirement as follows:

- Caller ID based call handling
- Calendar based call handling
- Sequential ring and simultaneous ring
- Present the actual calling party's number on the cell phone or other destinations
- Routing to last answer device
- Follow-Me Connect Verification – The mailbox owner can positively accept the follow-me calls by pressing a key to prevent calls from ending up in cell phone voicemail or other telephone answering devices.
- Follow-Me Record to Mailbox – Allows the mailbox owner to record a conversation that has been answered at the follow-me number. The conversation is saved and sent to the mailbox owner's voicemail box as a new message.
- Follow-Me Transfer Back – After the mailbox owner receives the call to the external device he can redirect the caller to another internal extension.

Follow Me feature provides better phone operation integration and provides the following capabilities.

- Follow Me feature control button on the phone – User can assign the button for the Follow Me feature and activate and deactivate the feature from the button on the IP Telephone to easily change the operation when users are in the office or on the road.
- Hand-off – When the user takes the call from the cell phone and return to the office, the call can be easily handed off to the desk phone by pressing the same button.
- Status Indication – The button has the LED to show the status of the Follow Me feature.

Unified Messaging

Unified messaging allows a mailbox owner to access voice messages directly through an email inbox. Emails may also be listened to from the voicemail box. Unified Messaging provides users with the following features; Integration with Email Clients, Messaging as an IMAP Server, and Messaging as a POP Server. Details on these features can be found in the Features chapter.

Other Messaging Features

Refer to [“Messaging” on page 50](#).

Call Manager

Call Manager (CM) is a powerful unified communications tool, a PC soft phone designed to enhance productivity for mobile and office users.

The Call Manager application runs on a PC with Microsoft® Windows XP, Windows Vista, the Terminal server on Windows Server 2003 ~ Windows Server 2008 R2, or Windows 7 operating systems.

The Call Manager Advanced version that comes with the VIPedge solution provides enhanced functionality, including full Unified Communications (UC).

Call Manager Advanced provides the following major functions:

- Desktop call control from your PC
- Customized call handling – CM allows you to place, answer, handle, view, and manage phone calls using your computer screen, keyboard, and mouse.
- Outbound dialing from any application
- CRM integration with screen pops – CM can easily interface with many popular programs (like Microsoft Outlook, Salesforce CRM, ACT, etc.). This allows you to dial from and “screen-pop” into these programs or the Internet / Intranet.
- Presence Viewer to display the status of other users
- Instant Messaging / Chat
- Using the VoIP Audio capabilities can provide a complete soft phone speech path when using a PC with the proper speech component support.
- Rules and actions can be set up to automatically activate when calls arrive even while you are away.



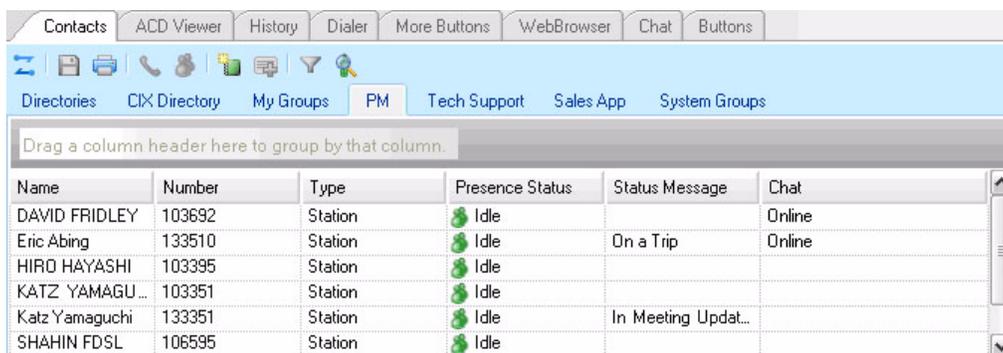
Figure 1 Call Manager Softphone Client

The Call Manager is based on the Microsoft Fluent User Interface which is easy to use and manage.

Microsoft’s fluent user interface breaks the ribbon GUI down into multiple tabs. The tabs are broken down into groups. The ribbon groups all the common features and functionality together. Each tab has a specific function and all the buttons in that tab support that specific function. For example, the Home tab encompasses all the basic telephony functions including: Hold, Transfer, Hang up, and Make Call.

Companion Applications

The Call Manager application supports some powerful companion applications. The Companion application tabs are shown below.



Contacts (Directory, Presence Viewer and Speed Dial)

The Contacts companion application performs three features: Directory, Presence and Speed Dial. The Contacts application provides a powerful set of directory features that allow you to look up and dial extensions with a click of the mouse.

The directory functionality in Contacts is generated by the system so it is always up to date with every extension. The directory can be easily searched and sorted by name and number. In addition to sorting by column name, the Contacts application now has a grouping feature where you can drag a column name into the grouping section and the resulting list will be grouped by the column name.

The Contacts application also has a Presence status column so you can quickly view the current status of the user's phone. Right-clicking on any user brings up a window that enables you to either call, chat, send broadcast, edit or delete.

History

The History companion application automatically creates a log of calls dialed, received and missed on the local telephone extension. The Call History can be searched for specific calls by date, telephone number, name or account code. Calls can be automatically dialed by double clicking the call in the Call History window.

The entire Call History or a search result can easily be printed or exported to a file. In addition to sorting by column name, the History application also has the new grouping feature where you can drag a column name into the grouping section and the resulting list will be grouped by the column name.

Chat

The Call Manager Chat application enables Call Manager users connected to interactively have chat conversations. Chat also supports whiteboard and canned messages. This program enables employees in an enterprise to communicate using real time text-based communications.

Using Call Manager Chat you can have individual conversations with anyone else on the server with the same feature installed. Chat can also be used to send a broadcast message to an individual or to an entire group. A broadcast message is a one-time message that will appear on the recipient's Chat window.

Dialer

Call Manager Dialer enables users to easily schedule phone calls to be placed later. For example, when a sales representative arrives in the morning he may know he needs to make calls to 15 of his customers, so from Microsoft® Outlook® he can drag and drop the contact information of all 15 of the customers into Dialer. When the designated time arrives for each call to be placed the user will be presented with a pop-up screen alerting him that it is time to place the call. Once the call is finished, the user will be presented with another pop-up screen requesting information about whether the call was successful and if not whether it should call again later (for example if the party was busy). The Personal Power Dialer displays all calls yet to be placed as well as calls that have recently been completed, plus as the status of the dialers and the phone.

Web Browser

The Call Manager provides an integrated web browser window for access to Internet or Intranet locations, or direct access to local HTML files. In addition to basic browsing, you can create custom web applications to extend the functionality of Call manager for your business. For example, your call center could have incoming calls automatically open the Call Manager browser window to a “Caller Survey” page. The Call Survey page could display information about the caller and display an answering script the agent could use to guide the conversation. The web page might also allow the agent to enter answers from the caller into HTML forms and submit the results to a company web server.

More Buttons

Call Manager has many different User Programmable buttons on the main screen and side window. These buttons can be easily configured by the user as Feature buttons, Speed Dial buttons, User Action buttons, ACD buttons, etc. This provides the user with one-touch access to features, applications, files, phone numbers, employees and more.

Call Manager Mobile

Call Manager Mobile on iPhone and Android smartphones (shown below) provides users with a productivity tool that is integrated with the VIPedge cloud based business telephone solution.



iPhone



Android Phone

Benefits

- Make calls from your cell phone using your Business Caller ID – and make yourself and your business easier to reach by only giving out one number. Use Call Thru/Call Back. Receive calls when you are away from your desk – at the same extension and get things done sooner. Use Follow Me.
- Make International Calls from your cell phone at landline rates and save money. Use Call Thru or Call Back.
- Be available on your business number when traveling internationally and save money on cellular roaming rates. Use Call Back and a local SIM or phone.
- See the status of your VIPedge colleagues before you call them and save time. Use Presence.
- Instant Message your VIPedge colleagues who are on the phone or busy and get more done. Use IM and Group IM.
- Quickly find and call your VIPedge colleagues without having to remember their internal extension. Use Contacts with Avatars.

Features

Features supported by Call Manager Mobile are as follows:

Contacts with Avatars

Call Manager Mobile gets its contact list from the VIPedge system so the subscriber does not have to create contacts for fellow VIPedge users. It also lets users upload avatars / images of themselves – making it easier to find people quickly.

Presence with Detail Information

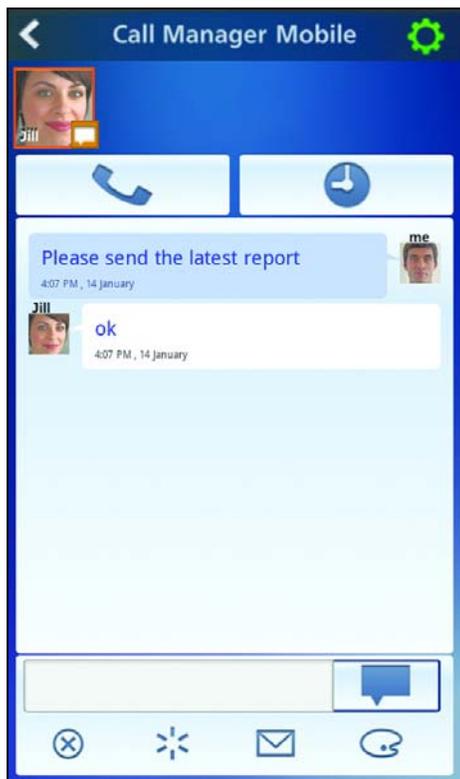
Presence enables you to see the presence status of other users before he/she begins communication. General presence information includes Available, Busy, and Away; and the detail information includes the user specified location and/or Instant Messaging (IM) status.

Instant Message (IM) / Group Instant Message (GIM)

A user can send or receive an instant message (shown below) to or from another user and can have multiple sessions at a time. It is also possible to broadcast the messages to multiple people. In addition, IM sessions can be held among multiple users who are invited by the coordinator.

History

Instant Messaging history is stored in the device so that it can be reviewed later. History records can be filtered by the user so that it can be found easily (shown below).



Instant Messaging



History

Mobility Features

Call Manager Mobile allows subscribers to use their cell phone as an extension on their VIPedge while making and receiving calls over the cellular network.

Follow Me

VIPedge provides Follow Me service where the call to an extension can be routed to a user's cell phone. After the call is answered, the user can transfer the call or record the conversation.

Call Back

Call Manager Mobile allows the user to dial a contact using the cell phone. When the user is located away from the office and needs to pay a high call charge, Call Back option is appropriate. Call Back option calls the user's cell phone first and then calls the destination. This way, VIPedge takes all the call charges for the local/long distance/international calls to user's cell phone and destination while the user is only responsible for the data usage. The destination user will see the DID number of the user instead of cell phone. After the call is connected, the user can transfer the call or record the conversation.

Call Thru

Call Manager Mobile provides another way to make a call which is more straight forward operation to the user. The user can make a call from a contact, and the call is made to the destination through VIPedge system. In this option, user may need to pay for the call charge to VIPedge system and data usage while local/long distance/international fee is charged to VIPedge system. This feature requires the auto attendant service for the DID number to be assigned to Voicemail. The destination user will see the outbound user's DID number, and the user can transfer the call or record the conversation.

Multiple Devices for Single User

The system allows the user to login from Call Manager Mobile and Call Manager on Windows at the same time so that the UC capabilities are provided regardless of the user's location.

Visual Voice Mail

The user can retrieve the voice mail left in the VIPedge Messaging from the Android phone. The list of voice mails are shown with the caller information and time (shown right), and the user can play back the message. The user can also call back the person who leaves the voice mail when the Caller ID from PSTN is available.



Visual Voice Mail

Devices Supported

Call Manager Mobile has been tested to work with the following devices. Depending on the device and the carrier, the user experience may be different.

- Apple iPhone 4S, 5
- Samsung® Galaxy Nexus
- Samsung Galaxy S® III
- Samsung Galaxy Note 2
- HTC Desire HD
- HTC One
- HTC Droid Incredible
- HTC Droid DNA
- Motorola Atrix™
- Motorola Razr
- Motorola Droid 3
- Motorola Droid X2
- Motorola Photon
- Sony Experia
- LG VS840

IP User Mobility

IP User Mobility is a set of features designed to give the user more flexibility as to where they use their IP phone. IP User Mobility consists of two major features that allows the user to be mobile.

- Enables the user to log-off and log into any Call Manager with voice or IP telephone without having to make any configuration changes. This is similar to “Hot Desking” where the user can go to any existing IP phone and use his or her corporate directory number (DN) to log in. All button programming assigned to that DN will be applied to the extension.
- Enables the user to transfer registration of an extension that is currently in use and is idle. In essence, the user can log into another IP phone with his or her extension even if his extension is already in use by transferring the registration from one IP telephone to another.

Advantages

- Multiple users can share one IP telephone (hot desk).
- Logout function allows a user to log-off his/her extension to free up resources or log in at that IP telephone.
- Better security when logging in is provided by using the Password field. The password is programmed on a per-DN basis.

IPT Anywhere enables you to connect IP telephones remotely through the Internet and use all IP5000-series telephone features.

This chapter covers Toshiba's IP5131-SDL, IP5531-SDL and IP5631-SDL Internet Protocol (IP) desk telephones and peripherals that are compatible with VIPedge solution.

IP Telephones

The VIPedge solution supports the IP5131-SDL, IP5531-SDL and IP5631-SDL telephones.

Features

These telephones include a speakerphone and are 802.3af Power-Over-Ethernet (PoE) compliant so that only one cable is required on the desktop. The IP Telephony product family also includes a matching Add-on Module.

The IP5000-series telephones support a very comprehensive and powerful feature set including:

- Backlit Displays
- 100mbit and Gigabit Ethernet telephone models
- Paging over telephone speakers.
- Automatic Configuration
- Terminal Authentication (security)
- Off-hook Call Announce (OCA) over telephone speaker while the user is talking using the handset.
- Built-in headset interface for headsets and external speaker connection (BESCB)
- IP Add-on Modules
- Full-duplex speakerphone capability.
- 802.3af power over Ethernet compliant
- Soft Keys to respond to feature prompts.
- Additional feature adjustments, such as setting button beeps, room noise sensitivity and handset busy override tone.
- An adjustable tilt stand base is built-in, providing flexible angle adjustment of the entire telephone.



IP5131-SDL, IP5531-SDL, IP5631-SDL

- Two codecs (coder/decoder) G.711 for toll quality voice and G.729A for reduced bandwidth. The external ringer interface enables connection of an BESC external speaker device to provide a loud ringer for the IP telephone.
- IP telephone firmware updates will be done by Toshiba Technical Support only. Dealers or users will be notified when the firmware update is scheduled. Updates require a brief interruption of IP telephone operation (a few minutes).
- Discovery retry timer to prevent network congestion.
- Loop back and ping capabilities for maintenance and fault finding purposes.
- Dynamic Host Configuration Protocol (DHCP) addressing.
- VLAN support with separately configurable VLANs for voice and PC data.

Features	IP5131-SDL	IP5531-SDL	IP5631-SDL
LAN	1G	10/100	10/100
LCD Characters (Pixels)	9x24 (168 x 150)	9x24 (168 x 150)	9x24 (168 x 150)
LCD Backlight	Y	N	Y
LCD Key Strip	Y	Y	Y
Number of Programmable Feature Buttons	10+10 ¹	10+10	10+10
Off-hook Call Announce	Y	Y	Y
PC LAN Interface	1G	10/100	10/100
AC Adaptor Input	Y	Y	Y
Speaker Phone	Y	Y	Y
Headset Interface ²	Y	Y	Y
Hearing Aid Compatible	Y	Y	Y
ADM Interface	Y	N	Y
External Speaker Interface	Y	N	Y
PoE Class	0	2	2

1. 10+10: 10 programmable keys are displayed at a time and the shift key can be used to shift between 2 sets of 10 keys.
2. When using a GN9350 or GN9330 headset, make sure your headset base is configured for GN RHL mode and the clear dialtone switch is set to position A .

When Power Over Ethernet (POE) cannot be provided, such as in home applications, the LADP2000-3A external power module can be used to power the telephone. Purchased separately.

Model Name	Features	Appearance
LADP2000-3A	Input AC120V Output 12VDC/1A	

Telephone Button Expansion Options

Upgrade options for the Toshiba IP telephones are described below.

LCD Add-on Module (LM5110)

The LM5110 adds 10 programmable LCD feature buttons to the 5000-series telephones with ADM interface, these buttons can be assigned as CO line, Directory Number, One Touch, Speed Dial or any other flexible feature.

The LM5110 supports backlight and LCD labels, it can be connected to the IP5131-SDL or IP5631-SDL telephone.



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This chapter provides a description of the various network related configuration pieces that need to be done when using VoIP networks and deploying the VIPedge solution.

Preplanning for VoIP Deployment

Benefits

- More accuracy and predictability in estimating budget requirements (and TCO) for the VoIP deployment by identifying network infrastructure needs up-front.
- Reduce cost of deployment by reducing trouble shooting costs.
- Reduce cost of post deployment maintenance and support by having the necessary information before hand.
- Improve Project Management – All requirements and conditions for a successful VoIP deployment are articulated, considered and factored prior to deployment.

Requirements

Network Assessment – A network assessment must be carried out to determine whether network or service upgrades are required to support a VoIP deployment. Toshiba recommends carrying out the network assessment with QoS enabled on the network.

Site Inventory Analysis – A site survey must be carried out to determine the list of network devices required for a given deployment. This survey must include considerations for the minimum required capability (feature set) and capacity for any networking device. A Gap analysis must then be performed to determine what upgrades or purchases would be required to support the deployment. The following table can be used for reference when doing Site Inventory Analysis.

Network Device	Capability	Considerations
Switches	QoS, VLANs, Autosensing, PoE	Port Capacity Port Bandwidth (GigE) Throughput, Latency, Jitter
Routers	QoS – DiffServ, DSCP 46	Uplink port Bandwidth Throughput, Latency, Jitter
Firewall and NAT	Ability to set firewall rules granularly. Flexibility to translate based on address and ports if required.	Throughput, Latency, Jitter
Server Platform	DHCP, DNS, NTP Services	Capability to handle multiple concurrent requests at high performance.

Interactions

While most end customer deployments fit the Toshiba recommended network deployment model, there may be instances where an end customer has unique network infrastructure or security policies which necessitate custom configuration and deployment. As this can potentially increase deployment time and effort it is critical to review end customer deployment environment and policies as part of the planning process.

Wide Area Network (WAN)

Benefits

Using the WAN to carry voice and data is more cost economical than paying for separate voice and data circuits. The WAN bandwidth can be used to carry voice when there is a call, and the bandwidth can be used for data when there is no call. This allows the most efficient allocation of communication circuit charges. Customers can select the best WAN service for their situation; the VIPedge service does not require a specific service provider. The service provider can also be changed when prudent.

Typical WAN services are:

- T1 data service
- Business-class Cable service
- Metropolitan Ethernet data service

Requirements

There are important requirements that must be met when selecting and using a WAN service for voice over IP. A VoIP Quality WAN service must deliver at least 99% of the voice packets between the WAN and the LAN. Over provisioned services specify maximum data rates but during peak times will drop large amounts of data on the WAN. For this reason DSL services are not supported.

Network Requirements (VoIP)	Wide Area Network (WAN)
Reliability	99.99% uptime
Latency	< 50 msec
Jitter	< 20 (+/-10) msec or less
Packet Loss	< 1%

For whatever service is selected, a network assessment is required. The network assessment will check the path from the site all the way to Toshiba's servers in the data center and will measure packet loss, latency and jitter to determine if the WAN connection is sufficient for carrying voice.

The minimum WAN requirement is 88Kbps in each direction per channel of voice and there must be at least 25% additional bandwidth available to be used for data. So a service with 10 channels of voice, needs 880kbps (for voice) and 220kbps available for data or 1.1mbps of WAN service. The 25% additional is required to ensure that bandwidth management and traffic shaping algorithms for voice and data to work properly. Often, people want even more bandwidth for data.

Interactions

In order to ensure that 99% of voice packets get through even when there is a high data communications requirement, a properly configured firewall with traffic shaping setup for both the up-link and the down-link is required.

It is important to make sure that there is enough bandwidth for voice on both the up-link and the down-link. In many offerings, the downlink rate will be higher than the up-link rate and people may only focus on the down-link rate.

If there is insufficient bandwidth for voice, or if the carrier's network is so over provisioned that 99% of voice packets don't get through, voice calls will experience drop-out.

Router/Firewall

Benefits

The VoIP capable firewall works to ensure VoIP quality even during periods of heavy web surfing, downloads, and social networking by managing bandwidth.

Bandwidth management basically involves prioritizing the voice traffic over the data traffic during times when there is more data and voice being carried between the LAN and the Internet than the WAN can handle, and either buffering or dropping the data packets in order to make sure the voice packets get delivered. However, even if the data packets are dropped, basic internet protocol will ensure they are re-sent. This is better than having voice packets dropped and sent again later.

Requirements

Bandwidth management of both the uplink and the downlink is required.

Toshiba offers and supports Adtran NetVanta routers sold as part of the VIPedge solution.

Interactions

If bandwidth management in both the uplink and downlink are not in place voice packets will not get through, during periods of high data traffic and voice calls will experience drop-out.

LAN Deployment

Benefits

Cost savings from using and administering a single IP network infrastructure for both voice and data communications.

Requirements

Core Network Characteristics for VoIP – In order to maintain voice quality, the underlying IP network must satisfy the characteristics that are listed in the following table. The table lists requirements for delivery over both Local Area Networks (LANs and WLANs) and for delivery over Wide Area Networks (WANs).

Network Requirements (VoIP)	Local Area Network (LAN/WLAN)
Reliability	99.99% uptime
Latency	< 20 msec
Jitter	< 10 (+/- 5) msec or less
Packet Loss	< 0.1%

Network Bandwidth Capacity – The underlying IP network has to be provisioned so it can handle the anticipated “maximum call volume”. To estimate this bandwidth, multiply the total number of voice streams by the bandwidth per stream. Note each voice call is composed of two voice streams, one in each direction. Bandwidth per stream = 88 kbps (G.711)

Network Assessments (MOS > 4.0) – A Network Assessment is required for every VoIP deployment to confirm that the underlying IP network satisfies the requirement mentioned above.

The result of the Network Assessment must be a test report which qualifies the network for VoIP readiness in terms of MOS scores. The minimum acceptable MOS score for acceptable voice quality on an IP network is 4.0.

QoS Mechanisms – It is required that the network be designed to prioritize voice traffic over data traffic as voice quality is very sensitive to packet loss, delay and jitter in the network. Depending on the size of the implementation, one or both of these mechanisms is required.

- DiffServ (DSCP 46) – Enable DiffServ in the network switches and routers and in the phones to prioritize voice traffic over data traffic.

Managed Switch or Router – It is necessary to ensure that the network is provisioned with managed switches and routers which provide the following capabilities:

- Autosensing Capability – All L2 port connections must be configured for maximum possible link (1000Mbps/100Mbps/10Mbps) speed and with full-duplex settings. The simplest way to ensure this is to deploy switches and phones with “Autosensing” capability.
- QoS Mechanisms like DiffServ (DSCP 46)
- Power over Ethernet (PoE) – It is recommended that IP telephones be powered using the PoE (IEEE 802.3af) technology. If PoE is not used, power bricks must be purchased separately for each phone.

DHCP Server – A DHCP server must be installed and configured at each site in order to automatically provision IP addresses for the IP telephones. This is included in the Adtran NetVanta router.

DNS Server – In an IP system a lot of the underlying communication relies on addressing hosts by their fully qualified domain names (FQDN). In order for this communication to be successful host names have to be resolved to IP addresses. For example, telephones and web browsers must be able to access VIPedge.com.

Interactions

Network Bandwidth Capacity – On a shared network infrastructure (both data and voice services), one needs to consider bandwidth requirements for both data and voice.

To provide additional prioritization for voice services, it is possible to enable 802.1p in conjunction with 802.1q (VLANs). However, this will enable 802.1p on remote phones requiring the switches are all locations where remote phones are deployed to support 802.1p.

Remote Administration

Benefits

Because VIPedge is in the cloud, dealers and users can easily login and manage the system or their personal settings remotely using a web browser from any location, not just while in the office.

Requirements

- Internet Explorer 8 or above, or Firefox.
- Internet connectivity.

Interactions

None

Remote IP Telephones

Remote IP5000 telephones are supported with VIPedge. This means that after a phone is configured, it can be relocated to a home, or another location where there is a good quality internet connection and still be part of the VIPedge system. The telephone will still be able to register with the VIPedge server.

Benefits

Users can take an IP5000 telephone to their home or small office and still use it as part of the business telephone system. Also road warriors can use their Call Manager client with Voice.

Requirements

In order to connect a remote IP telephone (or Call Manager with voice) to VIPedge, the following are necessary:

- A good quality broadband network connection and DHCP server offering phone's IP address and DNS server address so that the remote IP telephone is accessible to VIPedge.com domain.

Interactions

- The 911 emergency call destination must be setup in order for 911 calls to work properly.

Remote SIP Telephones

Benefits

SIP end points provide the ability to use application specific devices such as door phones, wireless devices, etc.

Requirements

- IP4100 SIP DECT wireless telephone

Interactions

Devices not supported must not be connected. When connecting SIP devices across routers, some home routers implement SIP application layer gateways. To work successfully, these gateways need to be configured to enable SIP transformation. However, not all application layer gateways work consistently and if a given gateway does not work, it should be disabled and the Static MRS function can be invoked to still support these end points.

Survivability

With the VIPedge solution running in the cloud on virtual servers, survivability is built in. Survivability is provided at the data center, server, and WAN level.

The data center is designed for survivability with multiple ISP connections, a building with a building design and even on-site power generators. For more details see the [“Datacenter Specifications” on page 63](#).

For the servers that are running the VIPedge application, there are stand by servers ready to take over when a failure is detected.

For survivability of WAN network outages that a customer's site may experience there is the Follow-me feature of VIPedge. This allows customers to receive their VIPedge calls on their cell phone even when they are experiencing a network outage.

Note Long distance and cellular toll charges still apply).

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This chapter contains the descriptions of various VIPedge features. They are presented in alphabetical order to make it easy to locate each feature.

Add-on Module (ADM)

One to two LM5110's (10 button) can be attached to IP5000-series telephones to provide an additional 10 or 20 programmable buttons.

ADM buttons can be programmed with outside line or Directory Number buttons, Direct Station Selection, One Touch Speed Dial or any other flexible feature button.

Advisory Message

Any telephone user can set a message on his or her telephone. Whenever another IP5000-series telephone with a display calls a station with a message set, the information in the message displays on the calling station's LCD. This feature allows users to define their current status and make that status available to others attempting to call that person. This status is also sent to Attendant Consoles.

Alarm Notification

VIPedge can send alarm notifications to a telephone.

Alternate Answer Point

Users can answer a transferred internal or outside line call from any station that has a Directory Number button appearance of the "transferred to" Directory Number.

Automatic Busy Redial

Automatic Busy Redial (ABR) enables a telephone user to automatically redial a busy outside number multiple times at programmed intervals. Each station may only have one call registered with ABR at any time.

Automatic Callback (ACB)

When a station user dials a busy station DN or outside line access code and receives a busy tone, Automatic Callback (ACB) can be activated by pressing an ACB feature Soft Key or by dialing 4. When the busy DN or outside line becomes available, the station will be automatically called back and be connected to and ring the originally called station or receive a dial tone from an outgoing line.

When ACB is activated, the calling station receives a success tone followed by a busy tone. Once ACB is activated, the caller can hang up. ACB can be canceled any time using an access code. It will also cancel automatically after a predetermined time.

Automatic Line Selection (ALS)

This feature automatically connects a telephone to a specific line or extension button when the user lifts the handset off-hook, presses the **Spkr** (speaker) button, or presses a digit on the dial pad (Hot Dialing). This feature is necessary to make telephone operation consistent for the user because a telephone can have up to 20 line and extension buttons. Each telephone can be assigned in system programming with various options that determine what type of line or extension button is selected when the user takes the handset off-hook to make or answer calls.

When answering calls, this option can be set to answer the call or not when a call rings the telephone and the user takes the handset off-hook. If the option is set to not answer automatically, the user can press the ringing button on the telephone to answer manually. With Automatic line selection, if more than one type of call is ringing simultaneously on the telephone, this option selects which type of call should be answered as a priority, then the longest ringing call in that call type is answered first.

Automatic line selection options are set independently for each telephone, for originating new calls and answering ringing calls. This feature can also be disabled on all or selected telephones to allow users to manually press a button to originate or answer calls.

Call Completion

This feature applies when calls are not completed because the station does not answer, is busy, or is in Do Not Disturb. A series of options are available to the user when encountering these conditions. They include changing the calling signal from Voice Announce to Ringing or vice versa, setting Automatic Callback, setting the Message Waiting light, Camp on Busy, Overriding the condition with Privacy/DND/Executive Overrides, or using Off-Hook Call Announce.

These options are easily activated by dialing a single digit code or pressing a soft key when the condition is recognized. These options are individually set for each telephone to be able to activate the call completion feature and to permit the feature to be activated when called.

Call Forward

Call Forward can be activated and deactivated for every DN (PDN, PhDN, Station Hunting pilot number). Call Forward cannot be activated and deactivated for the GCO key, but the call forward is applicable to the PDN of the station having respective ownership.

When an incoming trunk call terminates at a DN key or GCO key and Call Forward activates at both keys, the call forward at the DN key has priority. The call forward (that is, PDN for the station having Ownership) on the GCO key is ignored, regardless of the type of Call Forward activated.

This feature can be set and reset for the DN (PDN, PhDN, Station Loop, extension pilot number).

This feature can be set and reset (local/remote) from an IP telephone or using Enterprise Manager.

Call History

Incoming calls with Caller ID or ANI information may be optionally recorded into a rolling list for the station where the call is ringing. The call is placed in the list along with the number, name (if provided), time and date of the call, and status of the call (answered, abandoned, or redirected). This list is accessible by the user from the telephone LCD and any call may be selected and redialed using the flexible Caller ID or Hist button.

When calls ring a button (Line or DN) that appears on multiple stations, the number is stored on the telephone that is designated as the owner of the Line or DN and on the telephone that answers the call. If an incoming call is directed to a telephone, but the call is not answered by that telephone because it hunts or forwards to another destination, the call record is still stored on that telephone as “redirected” and on the telephone that answers the call as “answered.” If a call is not answered, it is stored on the line or DN owner’s telephone as “abandoned.”

To store call records, a telephone must be allocated Call History memory by the System Administrator.

Call Manager

Call Manager allows you to manage your calls from your PC. See [Chapter 2 – Unified Communications](#) for more details.

Call Park

Call Park gives any station, regardless of type, a method for holding calls. By parking a call, you are free to make other calls and retrieve the call at a later time or use the paging system to announce a call to be picked up by someone else on the system. Any call can be parked. Parking a call to your phone is known as Local Park, parking a call on someone else's phone is known as Remote Park, and if a general orbit is used, it is called Auto Park.

Call Park Orbits

The Call Park feature enables a station user to place a call temporarily in an orbit so that the call can be retrieved by any user, either from the same station or from a different station. Personal Park Orbits are available to any type of telephone, including standard telephones. If a call is parked, but not retrieved within a preprogrammed time period, it will recall the parking telephone. The Park Recall timer is a system wide timer setting.

Park and Page

This feature enables station users to park a call (in a General or Personal Park Orbit), enter a Page Zone or Group access code, and then announce the orbit number of the waiting call to the Paged party. A pre-programmed One Touch button can be assigned to telephones to automatically connect to a Telephone Paging group.

Call Pickup

Call Pickup enables station users to pick up all types of ringing or held calls including internal, PDN or PhDN calls ringing or on hold at other stations. When you pick up an internal call, the calling station and the called station displays on your LCD.

Group Pickup

Two or more stations can be assigned to a pickup group, there are a total of 48 pickup groups available. You can easily pick up ringing calls on other extensions. Ringing calls include: new, transferred, internal, or external calls. You will have the ability to pick up calls for other extensions in your group and other groups as well. See your system administrator for group assignments.

Ringing, Page, or Held Call Pickup

This feature picks up ringing or held calls, including Group Page and All Call Page calls. If these types of calls occur at the same time, the pickup priority is station-to-station and then Page calls in the order of occurrence. In some systems, this feature can be applied to pick up All Call Page exclusively.

Call Transfer

Call Transfer is the ability to redirect a connected call to a new destination. Three means of transferring a call and three means of terminating transferred calls is available, depending on the calling state of the destination. The transferring features and the terminating features may be used in combination to serve most needs that arise.

Music or Ringing Option

This feature enables ringing or music to be heard by the caller when their call is transferred, depending on system programming.

Transfer With Camp On

This feature enables the transfer of a call to a busy destination. The transferred party automatically camps on to the busy destination when the transferring party releases the call.

Transfer Immediate

Call Transfer Immediate simplifies the transfer of calls for users of IP telephones. With a conversation in progress, the display phone user presses the **TRNS** Soft Key and dials the transfer destination. The calling party is placed on Consultation Hold, the call immediately transfers and the transferring phone returns to idle.

Transfer Privacy

An outside call that has been transferred can only be answered at the station where the call has been transferred. Another station cannot pick up the transferred call unless it is another station using the Directed Call Pickup feature or a station that has a DN appearance of the “transferred to” DN. A flashing red LED indicates the call is transferred. Privacy for transferred calls can be disabled in programming.

Transfer (Screened)

The transferring party can talk privately with the receiving party before connecting the party to be transferred. While that conversation is going on, the transferred party is on Consultation Hold listening to Music-on-hold. When the receiving party agrees to accept the call, the transferring party can use the switch hook or feature button to include the original party in the conversation. At this point, the transferring party can hang up and the other two parties remain connected.

Transfer (Unscreened)

Unscreened Transfer allows the transferring party to exit the connection before the transfer destination answers. After the destination answers the call, the system treats it as a regular call. If the destination does not answer the call within the predetermined time period, the transferring party is recalled.

Transfer to Voice Mail

The transferring party can transfer a call directly to a person’s voice mailbox without waiting for the call to forward from the called party’s telephone.

The transferring party presses Direct Transfer to VM and dials the mailbox number, and the call transfers immediately on receipt of the last digit. The transferred party hears the greeting associated with the specified mailbox and can then leave a message.

Direct Transfer to Voice Mailbox simplifies getting a call for a busy or absent employee to his/her mailbox. It eliminates the need for the caller to enter the desired mailbox number after being connected to the voice mail system. This feature is available using standard DTMF VM integration and does not require Toshiba proprietary VM integration.

Call Waiting

When a station is busy with a call and another call is directed to that station's busy Line or DN button, two short beeps are issued to alert the telephone user of the pending call.

Call Waiting works for calls originating from within or outside the system. The length of the Call Waiting beeps is different for internal and external Call Waiting. The different beeps distinguish which type of call is waiting.

Caller ID, DNIS, or ANI information appears on LCD telephones for 10 seconds. If Caller ID information is not available, the device name, such as the SIP trunk or DNIS name or number is shown.

IP telephones receive a Call Waiting tone twice from the handset receiver. Call Waiting tones can be turned off on each station by the System Administrator.

When a station is busy with a call and another call is incoming, a tone alerts the caller of a pending call. On LCD telephones, the Caller ID information displays for 10 seconds. The combined effect of the Call Waiting alert tone with the displayed information enables users to identify whether or not they want to interrupt their current call for the waiting call.

To answer the Call Waiting, the current call must be parked, terminated, or transferred. Multiple calls can be queued to a single station, all waiting for that station to become free; the call at the head of the queue provides the Call Waiting signal and LCD indication.

The tone (two beeps) signaling Call Waiting tone is provided through the speaker of the phone. Caller ID display is not available with standard telephones.

Caller Identification

Caller Identification (Caller ID) is the general term for the information provided identifying the originating party of a public network call. The name and telephone number of the calling party displays on the ringing telephone's LCD. Incoming calls with Caller ID or Automatic Number Identification (ANI) information may be optionally recorded into a rolling list for individual stations. Station users with LCD displays can access this list to select and redial these calls (Call History).

Caller ID lists can include the number, name (if provided), time and date of the call, and status of the call (answered, abandoned, or redirected). Stations are assigned memory for creating the Call History.

Class of Service

The Class Of Service allows or denies a user to access the particular feature or determines to what degree the user is serviced. A station (IPT/ SIP terminal/ SLT) their own Class Of Service. Regarding an incoming call, the Class Of Service is either assigned to a line on programming base or assigned to a call on call by call basis.

The Class of Service consists of four specific parameters:

- Class of Service (COS) is used to check if a station or port is entitled to access a feature.
- Destination Restriction Level (DRL) is used by the Toll Restriction feature to check if the calling party is allowed to make a call to the dialed external destination.
- Facility Restriction Level (FRL) is used by the Station CO Line Access to check if the calling party is entitled to access the trunk facility.
- Queuing Priority Level (QPL) is used by the Automatic Callback feature and the Off Hook Camp On feature to determine the queuing priority of the calling party.

COS is used to check if a station or port is entitled to access the feature. DRL is used by the Toll Restriction feature to check if the calling party is allowed to make a call to the dialed external destination. FRL is used by the Station CO Line Access feature to check if the calling party is entitled to access the trunk facility. QPL is used by the Automatic Callback feature and the Off-hook Camp On feature to determine the queuing priority of the calling party.

The COS is always referred against the configuration table, while the DRL/FRL/QPL are referred with their numeric values. Thus, the numeric value of COS is just an index. In addition to the COS index, the station profile may be used to represent more detailed and complex information of allowance/denial of stations. As for the DRL/FRL/QPL, the larger value has higher priority than the smaller value. The COS takes indexes 1 through 32 while DRL/FRL/QPL take levels 1 through 16.

Conference Call

Conference calling enables other people to join your conversation. Any station can set up a conference with other stations or outside lines. A conference is defined as any time three or more parties join into one conversation. A maximum of eight parties are allowed into a conference with up to six from outside lines or standard stations. The originator of the first conference is the “master” and controls adding and deleting conference parties. The conference “master” can drop off the last added party by using the Cancel button.

Conference On-Hold

A conference call may be put on Hold so that all the remaining conferees remain connected and no Music-on-hold is applied. The person putting the conference on hold may rejoin the conference by pressing the Line button on their phone. The Hold state of the conference can be released from another station by pressing the Line button of that station. At this time, the station that released the Hold state becomes Conference Master. This enables one person to establish a conference call for others.

Split/Join/Drop

This feature enables the conference master to add (Join) other phones to a conference. The conference master and another member of the conference can leave (Split) the conference for a private conversation. During this time, other conference members remain connected. The conference master can then Join both of the Split callers back into the conference, or the master can Drop (disconnect) the Split member he/she is connected to.

Releasing from Conference Tandem SIP Trunk Connections

This feature enables unattended line-to-line connections, freeing the conferencing station or voice mail port for other calls and important tasks. When a tandem connection is set up with a DN button, the DN button will go idle after releasing from the connection. The DN can then be used to make or receive calls from the originating telephone.

Telephones and/or VM devices can establish tandem SIP trunk connections and then release from them without disconnecting the tandem connection in the system. After releasing from a tandem call, reconnecting to the call can be accomplished by dialing an access code.

Whether or not tandem line buttons appear on a telephone, the telephone user can enter the connection and release the line that was connected to the original line or release both lines by pressing the Cancel button.

Voice Mail Conference

Voice Mail ports may be included in conference calls. This enables all members of the conference to listen to and play voice mail messages.

Day/Night Mode

Auto Schedule – The system has three operating modes that are based on the time-of-day, day-of-the-week, and up to 128 holiday schedules. The operating modes are Day1, Day2, and Night. Each mode controls the routing of incoming line calls and settings for station and line Class Of Service restrictions. The system can be programmed to use all three modes, Day/Night mode only, or just the Day mode. The system switches automatically from one mode to the next based on the system's time-of-day clock.

Example Day/Night Mode Applications:

Incoming Calls – Incoming line call including ground/loop start lines converted to SIP trunks via a gateway, SIP trunk URIs, and individual DID and DNIS numbers can change their ringing destinations automatically according to the date and time of day.

For example: On workdays, calls are routed to the individual telephones, ACD groups, etc., until 5:00 p.m. After 5:00 p.m., calls are routed off-premise to another office, to the Night Bell, or to night announcements and voice mail message boxes. On holidays and weekends, calls are routed independently to the appropriate holiday announcements or voice mail message boxes.

Class Of Service – Station, lines, and DID numbers are assigned options in Class of Service. These include Toll Restriction, Override privileges, allowed tandem connection, security code administration, etc. Any of these options can be changed independently for each telephone, line and DID number when the system switches from one operating mode to another.

For example: When the system changes from the Day to Night mode, selected stations can be automatically restricted from dialing outside or long distance calls. Note that outgoing route selections set in LCR are switched using a route selection schedule that is independent of the Day/Night mode schedule.

System Call Forward – The System Call Forward settings for stations can be changed automatically when the system changes from the Day to Night mode. For example: During the day, a telephone can forward to a person's car or cell phone, and at night automatically forward to the person's voice mailbox.

Tenant Services – Up to eight different tenants can each have different night bell assignments for day-of-the week schedules using Day 1, Day 2, or Day 3 modes. This feature is enabled in system programming.

Manual Mode Change – The system also enables users to manually change the Day/Night operating mode, even if the system is using the Auto Schedule feature. A Night Transfer key can be set on telephones for manually switching at any time from one mode to another. The key's LED flash rate indicates the system's operating mode.

If used with the System Auto Schedule operation, the Night Transfer key overrides the current Auto Schedule mode. However, when it is time for the system to switch to another mode per the Auto Schedule timer, the system will switch to the mode set by the Auto Schedule.

Example: If the system is switched from Day to Night at 2 p.m. manually with the Night Transfer key (Auto Schedule is set to switch from Day to Night at 6 p.m.), the system will still automatically switch back to the Day mode at 8 a.m. the next morning per the Auto Schedule.

Dial Directory

Station users can dial by name using Toshiba's IP5000-series LCD telephones. The Dial by Name feature searches for names much like a cell phone directory and then allows the user to press one button to dial. This feature includes speed dialing and internal directory names. Your telephone will display names (First Name, Last Name or vice versa) depending on the way they have been programmed in the telephone system. All directory names are sorted alphabetically. Individuals stations and speed dials can be omitted from the Dial by Name directory if desired.

Direct Inward Dialing (DID)

This feature allows external callers to dial directly to individual extensions or groups of telephones without intervention by an operator, IVR, or auto attendant. Each incoming DID number can be routed individually to an extension or other resources, such as pooled or group line button, night bell, voice mail box, or back out over the public or private telephone network. DID routing assignments can change automatically when the system switches between the Day and Night Modes.

DID numbers can vary between 1 ~ 7 digits in length for each DID line group. Each DID number can be assigned to 1 of 15 possible music-on-hold sources.

DID service is provided by the SIP trunk interface and works with ISDN trunks and T1 gateways that support it.

DID Parking

DID parking allows a number of DID's to be parked and then used later. DID parking is a line item on the ordering portal. Numbers can be ported or new numbers can be ordered and then parked. There is a minimal per DID monthly recurring charge for this service.

Dialed Number Identification Service (DNIS)

DNIS lines receive 800- and 900- type telephone calls and provide the number the caller dialed. The DNIS number is translated into a name that displays on the telephone's LCD. This allows the user to identify where the call is coming from and the purpose of the call before it is answered.

The DNIS Name/Number routes calls to specific telephones, departments or ACD groups. When the call is ringing and after it is answered, the DNIS Name/Number displays on the telephone's LCD. The name and number display can also be sent to an agent computer to be used by a CTI application.

Applications include sending DNIS calls to a group of agents that take orders for a number of different companies and products. The agents know how to answer the calls from the DNIS display. Using DNIS capabilities allows one group of lines to be used to serve multiple applications. DNIS service is provided by DID, T1 (via a gateway), or ISDN trunk (via a gateway) interfaces and provides the same call routing options and destinations as DID calls.

Directory Numbers

A Directory Number DN, sometimes called an "extension number," is the number someone must call to reach a destination within the system. Each DN is assigned as the main directory number of a telephone. To maximize call coverage flexibility any DN can appear on multiple telephones. Also, individual telephones can have multiple DN buttons with different Directory Numbers.

The system provides Primary and Phantom DN buttons on telephones. All DN buttons can be used to originate and answer calls.

Primary DN Buttons

Primary Directory Number buttons PDNs are needed to make and receive calls. It is a telephone's main extension number. Each telephone is assigned only one PDN and that telephone is designated as the owner of the PDN. This PDN button can be made to appear multiple times on other telephones (see Secondary DN Buttons). Features, Class of Service, etc., are associated only with the station assigned as the owner of the PDN. PDNs that appear on telephones other than the owner telephone are referred to as Secondary DNs or SDNs.

Phantom DN Buttons

Phantom DN buttons PhDNs are additional directory numbers appearing on telephones as extension buttons. PhDNs can be used as independent extensions on the phone or can appear on multiple phones to be used to allow call handling for departments or groups of telephones.

PhDNs can be used to make a telephone appear to have multiple PDN extension buttons. When assigning a PhDN for use as another appearance of the PDN, the display properties are set the same as the PDN and a hunting sequence is set up to roll the calls from the PDN over to the other PhDNs associated with the PDN. With this arrangement, Call Forward sends calls to the PDN's destinations and Voice Mailbox.

Pilot DN

A Pilot DN is a pseudo-location that is assigned a DN where calls may be directed. Unlike PDNs and PhDNs, a Pilot DN is *not* a button on any telephone. A Pilot DN is used as a device where calls can ring and be held while an external application using the Computer Telephony Integration (CTI) can control the call. To ensure calls do not get lost in the system, a time-out and overflow service is provided to redirect the call when the link is down.

Pilot DN's are also assigned to Station Hunt Groups (for details, refer to Station Hunting).

Distinctive LED Indicator

Distinctive LED indicators provide a method for quickly identifying the status of a line or feature button. The LED color or flash pattern can identify the call you are currently on, as well as other calls you are controlling, versus other calls that may appear on your telephone. Each telephone uses dual-color LEDs: green for lines you are using; red for lines used by someone else.

Distinctive Ringing

Users sometimes need to distinguish the ringing of one button on their phone from another button and sometimes stations in close proximity to one another need to distinguish the calls on one desk from another. Typically, multiple sounds are used to provide this distinction. Distinctive ringing can be assigned to each Line or DN button on each telephone.

You can set up to ten different incoming ringing tones for internal, as well as external calls. Previously, distinctive ring was not provided for internal calls.

Do Not Disturb (DND)

Station users with IP telephones can activate DND to prevent any calls from ringing their telephone. Callers will hear a fast busy tone when calling stations in the DND mode. Stations in DND mode can originate calls normally; however, they receive DND stutter dial tone (optional) when originating calls.

Call Forward-Busy will forward calls directed to a telephone with DND set, even if the telephone has idle DN's. Telephones with DND Override capability can ring DND telephones.

Emergency Call

An Emergency Call access code can be established to route calls to specified emergency destinations and to prioritize their delivery to those destinations. Up to four emergency destinations can be programmed for each mode of operation: Day, Day2, and Night. This is particularly useful in applications where employees, patients, or guests are not expected to know where to call for help at different times of the day.

Enhanced 911 (E911)

Enhanced 911 calling means the routing of a call to the appropriate Public Safety Answering Position (PSAP) accompanied by Caller Emergency Services Identifier (CESID). The CESID identifies the location where emergency services are to be sent.

The system can use either SIP trunks or SIP PRI gateways - which requires carrier and gateway support. Internal emergency destinations can also be automatically included in an emergency call.

When the IP telephones are set up in the office properly, 911 will work as intended. However, when the IP telephone is moved to an off-site location, the following warning applies because the call may not connect to the correct PSAP. When the IP telephone is moved 911 will not work correctly, until the appropriate action to update 911 emergency response address is completed. This involves the customer notifying the dealer about the location change.

WARNING! You may NOT be able to contact emergency personnel by dialing 9-1-1 from a telephone or from Call Manager. Use an alternate service, e.g., a mobile phone, to dial 9-1-1 if there is no dial tone; emergency personnel do not answer when you dial 9-1-1; or you reach the wrong emergency call center unless instructed otherwise.

Feature Prompting With Soft Keys

As an alternative to dialing access codes and using feature buttons, station users with LCD IP telephones use Soft Keys (shown on their LCD) to access features. Abbreviated feature names display during a call (when the telephone is in the ring or talk state) on the LCD above the fixed keys. Users can select a feature by pressing the associated key. The LCD feature selections change according to the call state to provide the most logical options.

Flexible Line Ringing

CO line ringing can be assigned to ring a specified DN on a station, a DN appearing on multiple stations, a Pilot DN, a direct appearance of the CO line, a Pooled appearance of CO lines, or Group CO line appearance. These assignments direct the ringing of the incoming call based upon the three Day/Night Modes of operation and offer immediate and two delayed ringing parameters. The delay parameters are assigned for each incoming Line Group.

Flexible Numbering

The system-numbering plan can be customized for the user's needs – Directory numbers, line and feature access codes.

Handsfree Answerback

When a voice-announced internal DN call comes in to a telephone, users can answer without lifting the handset.

Headset

Many of Toshiba's IP5000 telephones have a built-in headset jack which allow a wireless or wired headset to be attached.

Hearing Aid Compatible

Toshiba's IP telephone product line includes telephones that support hearing aids.

High Call Volume Buttons

Release, Release/Answer, and Cancel buttons can be assigned to telephones. They enable a busy user to handle calls quickly and efficiently in high call volume situations.

With one touch of the Release button, a user can disconnect from a call. This is especially useful in headset applications. The Release/Answer button disconnects or transfers the current call, and answers the next. The Cancel button voids the last operation, such as disconnecting internal or external parties from conference or tandem calls.

Hold

There are several variations of Hold:

Automatic Hold

This option enables a user to place a call on Hold by pressing another DN button. The user can then alternate between the new and the old call by pressing the desired Line or DN. If this feature is not activated, users must press Hold before accessing another line and switching between calls.

Call Hold

This is the most commonly used. Call Hold temporarily suspends a call, allowing the station user to do other things, including using the phone. Callers on hold can receive music or announcements as described in the Music-on-hold feature.

Consultation Hold

This is used when invoking other call features, such as Call Transfer or Conference.

Exclusive Hold

A call can be placed on Exclusive Hold to ensure the privacy of the connection and that the call can only be retrieved by you, even if the held call appears on buttons on other telephones.

Hold Recall

After placing a call on hold, it will recall the holding telephone after a predetermined time to remind the user of the held call. The hold recall time is set independently for each telephone (from 0~255 secs.). Hold recall time can also be disabled.

Hot Dialing

Hot dialing enables the telephone user to begin on-hook dialing without pressing a Line or DN button. The station can be programmed to automatically select a Line or PDN button when the dial pad is pressed while the station is idle. This saves a keystroke by not requiring the station user to press a DN or Line button to begin on-hook dialing. On-hook dialing saves time by not requiring the station user to lift the handset to begin dialing.

Hotline Service

If a station remains off hook for a programmable period, it can automatically be directed (immediately or with a delay) to a pre-programmed destination. The station may have partially dialed a number or have dialed no digits at all.

Each station is programmed with its specific ring down destination. This is particularly useful in applications where employees, patients, or guests are not expected to know where to call for help at different times of the day.

Line Buttons

Telephone buttons that are used for making and receiving outside calls are referred to as Line (or CO Line) buttons. The following types of line buttons are supported:

Pooled CO Line Button

Pooled line keys are used to provide a key appearance for a single URI, DNIS, or DID number expected to handle one call at a time.

Group CO Line Button

Group CO keys are used to provide several key appearances for one or more URI, DNIS, or DID numbers expected to handle multiple calls at one time. Group CO line buttons are like individual CO line buttons except these buttons represent all the lines for a particular Channel Group. This enables channels to operate similar to analog CO lines on a key telephone system. These buttons may have appearances on multiple telephones providing call coverage across several telephones. Multiple appearances of the same Group CO line button is possible on each phone to allow multiple call handling for that group from each station.

Message Waiting

Any station and most voice mail devices can turn on a message waiting indicator for a designated IP5000-series telephone. This feature can be disabled in station programming.

LED Indication

Message waiting lights can be activated when a voice mail message has been left or they can be turned on by a calling station. The station user can retrieve messages by pressing the button next to the message waiting light.

The telephone main **Msg** light indicates a message is waiting for the telephone PDN. Up to four PhDNs per telephone can also have individual MW LEDs assigned to flexible buttons.

Stutter Dial Tone

Stutter dial tone is also used to indicate a message is waiting or that your telephone is in the DND mode. When a station user goes off-hook, two different available stuttered dial tones indicate whether a Message Waiting (MW) or DND condition exists.

- The MW-stutter dial tone indicates a message is waiting for the station.
- DND-stutter dial tone indicates DND is set at the station. (DND provides a fast busy tone burst as stutter dial tone.)

If both conditions exist simultaneously, the MW-stutter dial tone has priority. This is very valuable to station users that do not have a MW Light Emitting Diode (LED) or **DND** button LED on their telephone.

Users can disable (in programming) stutter dial tone for message waiting and when in the DND mode. If stutter dial tone is disabled, they will hear a normal dial tone when off-hook.

Mobility

Mobility enables users to login to different stations and to take their stations off the company LAN. See [Chapter 2 – Unified Communications](#) for more details.

Music/Messages On Hold

This feature provides music or a tone to a station or line that is held by a station with Line Hold or Consultation Hold and the speech path is released.

The system has a total of fifteen music sources plus Quiet Tone and Beep Tone on the Media server. The system administrator selects from these 15 internal WAV files of music sources, one internal electronic tone (beep tone) and quiet tone. Users can upload their own custom WAV files to the system using EMPA.

Multiple Call/Delayed Ringing

If an incoming external or internal Directory Number (DN) call rings a station DN and is unanswered, alternate DNs can be programmed to ring at a later time. You can also assign Delayed Ringing to voice mail and auto attendants.

The Multiple Calling feature enables you to ring two or more telephone numbers simultaneously or after a predetermined time.

This feature is effective as follows:

- Enable ringing two or more standard telephones or SIP stations that cannot have multiple appearances.
- Enable ringing different kind of stations, such as an IPT and SIP stations.

If an incoming external or internal call rings to a station DN and is unanswered, alternate DNs can be programmed to ring at a later time. A separate delayed ring time can be set for each CO line group. The stations that were ringing initially will continue to ring after the Delayed Ringing begins. This feature is assigned for each line or DN button independently for each DN.

You can assign Delayed Ringing to voice mail and auto attendants. This feature can also be used to ring multiple (25 max.) telephones immediately or with a delay by dialing a group pilot number. Each group member can have Immediate, Delayed Ring 1, or Delayed Ring 2. Delayed Ring times are adjustable (1~180 seconds) for each Multiple Call Group.

Off-hook Camp On

When a dialed station designated external line or line group is busy, Off-hook Camp On allows the user to wait until the destination becomes idle.

The system automatically monitors objects of Off-hook Camp On and connects them to the waiting user immediately after they become idle. The user does not need to redial the destination number.

Override

Call Forward Override

Stations with this feature will not forward when they call stations that have System or Station Call Forward activated. This applies when using the telephone dial pad to make a call.

Class Of Service Override

By dialing a Class of Service Override (COS) code, a user can change a station's class of service to one associated with the override code. When the call is terminated and another is attempted from the same station, the original Class of Service is applied. This allows selected users to override toll restriction or other restrictions that are placed on any telephone in the system.

Do Not Disturb (DND) Override

A privileged caller may invoke the DND Override feature after dialing an internal station and receiving a DND indication. If that privilege is granted to the calling station and the called station permits its DND to be overridden, the call will ring on that phone.

Executive Override

Stations with this feature allowed by COS can enter any conversation in the system by dialing a 3 or pressing a Feature Prompting Soft Key after dialing a busy station. An optional warning tone notifies the parties that another party is about to conference into their conversation. Executive Override can be blocked selectively to any station in the system. Executive Override must be allowed in system programming for the called and calling station.

The Do Not Disturb feature can also be used to block Executive Override; however, stations that are allowed DND Override can use Executive Override on stations in the DND mode. The Privacy button does not block Executive Override.

Privacy Override

Privacy override controls the ability of multiple station users with a shared (common) Line or (DN) button appearance to join in each other's conversation by pressing the busy button appearance. A station must be programmed with Privacy Override to permit the intrusion on a shared Line or on (DN) buttons.

In the case where Privacy Override is normally allowed, a telephone can have a "Privacy" button to block Privacy Override (intrusion) to the call. The Do Not Disturb feature does *not* block Privacy Override. In the case where Privacy Override is not normally allowed, a telephone can have a "Privacy Release" button to allow intrusion to the call by any station with the shared button appearance. (See "Privacy" for more information.)

Paging

Telephone Group Paging

Paging is activated from an extension by specifying a Page Group. Paging is broadcast through the IP5000-series telephone speaker.

Emergency Page

Designated stations can be permitted to place an Emergency Page to ensure they can reach all concerned with an important announcement. An Emergency Page is one that will supersede any current page to allow this privileged station to take over the IPTs. Like other forms of paging, an Emergency Page can be an All Page or directed to a specific Page Group.

Emergency Page groups follow the regular Group Paging. The list for Emergency All Call Paging is a separately defined list from regular paging. An emergency page may be answered in the same manner as a regular page.

Privacy

Privacy prevents intrusion on calls that appear on shared (common) DN or line buttons. If a telephone has a call on a DN or line button that appears on other telephones, the other telephones cannot intrude on the call by pressing the shared button unless the intruding telephone has the Privacy Override feature or the telephone with the call activates the Privacy Release button.

Privacy Override

Privacy override controls the ability of multiple station users with a shared (common) Line or DN button appearance to join in each other's conversation by pressing the busy button appearance. A station must be programmed with Privacy Override to permit the intrusion on a shared Line or on DN buttons.

Repeat Last Number Dialed

This feature enables a digital station to automatically redial the last number dialed from their station by selecting an outgoing line and pressing the Redial button or by dialing an access code.

Ringling

See also Multiple Calls/Delayed Ringling and Distinctive Ringling.

Ring Over Busy

When a digital telephone is busy on a call and then receives an internal or external call on an idle DN or line button, the button will automatically flash and ring with Ring Over Busy tone. The tone burst can either be sent two times (three seconds apart) or repeated continuously every three seconds or not sent as a station option. To answer a Ring Over Busy call, the user can hold, transfer, or disconnect the existing call.

On Voice First calls to a busy telephone that has an idle DN, the caller will get busy tone. The caller can then dial the digit 1 to cause the idle DN to Ring Over Busy.

SIP Trunk

Session Initiation Protocol (SIP) is an application layer protocol used for establishing sessions in an IP network. SIP trunks allow the VIPedge solution to get PRI-like services from an Internet Telephony Service Provider using SIP.

Within each VIPedge system SIP trunks are used to communicate with the PSTN. The SIP trunk will support the number of channels provisioned for the system.

SIP Trunk Wizard

Toshiba has made programming SIP Trunks easier by giving the administrator the ability to enter URI ranges. The URI field range allows the administrator to enter a range of the user portion of the SIP Trunk URIs for the selected SIP Trunk service in the targeted server. The SIP Trunk Wizard also allows for making URI ranges from DID numbers. The administrator can also define a prefix which will be added to the front of each selected DID number to compose the URI. For more details, refer to the SIP Trunk Feature Description.

Specified Caller ID

The Specified Caller ID feature allows applications, such as Messaging, to send the telephone number (Caller ID) of the calling party to the cell phone or other phones when the application routes a call.

With this feature, the called party can see the actual Caller ID on the cell phone when the Follow Me feature of VIPedge Messaging routes the call to the cell phone.

Important! *Usage of this feature must comply with Toshiba's Accepted User Policy.*

Speed Dial

This feature, sometimes known as automatic dialing or one-touch dialing, enables the customer to assign dialing codes to telephone numbers that are frequently called. The VIPedge solution offer three forms of Speed Dial: System Speed Dial (up to 800 max. per system), Station Speed Dial (100 max. per station), and One Touch buttons. Station Speed Dial numbers and One Touch buttons are unique for each station and cannot be used by other stations. System Speed Dial numbers can be used by any station in the system.

To dial System and Station Speed Dial numbers, the user presses the Spdial button and then dials the appropriate three-digit code for the telephone number to be dialed. To dial a telephone number assigned to a One Touch button, the user simply presses the One Touch button. Users can program Station Speed Dial and One Touch buttons from their telephones.

The Web-based User Administration application is required to program System Speed Dial numbers and can also be used to program Station Speed Dial numbers, but not One Touch button numbers. Each Station and System Speed Dial number can be assigned a nine-character name using Enterprise Manager. This name appears in the LCD when using System Speed Dial and Personal Speed Dial directories (accessed through DIR softkey).

One Touch Buttons

One touch buttons enable users to store speed dial and custom feature access sequences on a single button. When this button is pressed, the stored number is dialed or the feature is accessed.

You can store frequently dialed numbers, such as three-digit System Speed Dial codes, onto a One Touch button. This eliminates the need to enter the three-digit code to dial a System Speed Dial number. Complete telephone numbers up to 32 digits can also be stored on a One Touch button.

These buttons make it easy to access features that usually require pressing multiple buttons and/or dialing special access codes. For example, a user may have to dial an access code (#31) plus a zone number (5) to page the warehouse. This sequence can be set on a One Touch button labeled "Page Warehouse." Another button can be set to page a particular group of telephones.

The One Touch button also has a "stop" function that can be entered between two numbers, such as a telephone number and security code. When the One Touch button is pressed, it can speed dial a telephone number, then pause (LED flashes). When the call is answered, it prompts for a security code. The user can then press the flashing button and enter the security code. Any number of "stops" can be set to enable dialing multiple numbers.

Multiple feature buttons such as Cnf\Trn, DN, CO line, etc., can be set on One Touch buttons to allow multiple button presses to be stored under one button. This enables tandem line connections and other call setup sequences to be dialed easily by pressing one button.

Station Hunting

A series of Directory Numbers (DNs) can be organized in groups in such a way that if a called DN is busy the call will try to ring another DN in the group. If that DN is busy it will hunt to a third DN, etc. Telephones in the same department, voice mail ports, and boss/secretary call coverage situations are typical applications for hunt groups. Hunt group members can remove themselves from the group by placing their station into the Do Not Disturb mode. The system supports three types of station hunting.

Serial Hunting

In this type of hunt group, calls hunt DNs in a series from first to last in a specific order. When any DN in the series is called, the system will ring the first idle DN in the series, starting with the called DN, hunting to the last DN in the series. As an option, this type of hunt group can have a unique Pilot DN assigned to it. When callers dial the Pilot DN to reach a telephone in the group, calls will hunt all DNs from first to last.

Circular Hunting

In this type of hunt group, calls hunt DNs in a series in a specific order. However, the series form a loop, which enables the last DN to hunt to the first DN. When any DN in the series is called, the system will ring the first idle DN in the series, starting with the called DN, hunting to all DNs in the series. As an option, this type of hunt group can have a unique Pilot DN assigned to it. When callers dial the Pilot DN to reach a telephone in the group, calls will hunt all DNs from first to last.

Distributed Hunting

This type of hunt group always has a unique Pilot DN assigned to it. Callers dial the pilot DN to reach a telephone in the group. Calls hunt so that the calls distribute evenly to each DN in the group. Hunting rotation always starts in sequence with the DN that follows the DN that received the last call – even if all other DNs are idle.

Camp on to Hunt Groups

On incoming CO line calls to busy hunt groups, the caller automatically camps on to the called DN or Pilot DN and the caller receives ring-back-tone.

On internal calls to busy hunt groups, the caller may get a busy tone. The caller can then dial a digit to initiate Camp On-Busy to the called, busy DN or the Pilot DN, if used. As an option, for each hunt group that uses a Pilot DN, calls will automatically camp on to the called Pilot DN.

With the Automatic Camp On option, the caller does not get a busy tone, instead the caller receives a confirmation tone followed by a ring-back-tone. When using hunt group Pilot DN's, camped on calls queue onto all DN's in the group and will connect to any DN in the group that becomes available. When not using Pilot DN's, Camp On is only applied to the called DN.

When more than one party is camped on (queued) to a hunt group, the party with the highest Queuing Priority Level (QPL) will connect first when the destination becomes available. If the parties have the same QPL, the longest waiting call will connect first.

Transfer Direct To Voice Mail

Transfer Direct To Voice Mail is the ability to redirect a connected call to new destination. Three means of transferring a call and three means of terminating transferred calls are available depending on the calling state of the destination. The transferring features and the terminating features may be used in combination to serve most needs that arise.

Transfer with Camp On

This feature enables the transfer of a call to a busy destination. The transferred party automatically camps on to the busy destination when the transferring party releases the call.

Transfer Immediate

Call Transfer Immediate simplifies the transfer of calls for users of digital display telephones. With a conversation in progress, the display phone user presses the **TRNS** Soft Key and dials the transfer destination. The calling party is placed on Consultation Hold, the call immediately transfers and the transferring phone returns to idle. This feature does not apply to network calls.

Transfer Privacy

An outside call that has been transferred can only be answered at the station where the call has been transferred. Another station cannot pick up the transferred call using a common CO line button unless it is another station using the Directed Call Pickup feature or a station that has a DN appearance of the "transferred to" DN. A flashing red LED indicates the call is transferred. Privacy for transferred calls can be disabled in programming.

Transfer (Screened)

The transferring party can talk privately with the receiving party before connecting the party to be transferred. While that conversation is going on, the transferred party is on Consultation Hold listening to Music-on-hold. When the receiving party agrees to accept the call, the transferring party can use the switch hook or feature button to include the original party in the conversation. At this point, the transferring party can hang up and the other two parties remain connected.

Transfer (Unscreened)

Unscreened Transfer allows the transferring party to exit the connection before the transfer destination answers. After the destination answers the call, the system treats it as a reg-

ular call. If the destination does not answer the call within the predetermined time of period, the transferring party is recalled.

Transfer Direct To Voice Mailbox

The transferring party can transfer a call directly to a person's voice mailbox without waiting for the call to forward from the called party's telephone. The voice mailbox does not need to be associated with an active telephone. Direct transfer to voice mail (VM) can be performed to a centralized VM system connected to a network node other than the user's node.

The transferring party presses Direct Transfer to VM and dials the mailbox number, and the call transfers immediately on receipt of the last digit. The transferred party hears the greeting associated with the specified mailbox and can then leave a message.

Direct Transfer to Voice Mailbox simplifies getting a call for a busy or absent employee to his/her mailbox. It eliminates the need for the caller to enter the desired mailbox number after being connected to the voice mail system.

Music or Ringing Option

This feature enables ringing or music to be heard by the caller when their call is transferred, depending on system programming.

Uniform Call Distribution

Uniform Call Distribution (UCD) functionality provides call flow to distribute calls more efficiently through a call center. UCD enables calls to be answered by the auto attendant, which prompts the caller to dial the correct UCD group number or, calls can ring directly to UCD groups. The call is then sent to the UCD agent or queue if all agents are busy or logged out, but never to a busy number. Incoming calls can also be directed directly to UCD groups without the use of an Auto-attendant.

Calls sent to agents are managed by distributed hunt groups to find the next available agent. Callers in queue can receive music and announcements embedded in one of the systems music-on-hold sources, and each UCD group can share or have a separate music sources. The announcements must be recorded on the music source. Overflow timing is controlled by a unique overflow timer for each UCD group. Agent log-in and log-out buttons make it easy for agents to sign in and out of the system so that calls can be routed appropriately.

Unified Communications

Refer to the [Chapter 2 – Unified Communications](#).

VLAN Tagging

The VIPedge solution supports 802.1Q Virtual Local Area Network (VLAN) technologies. For sites that have thousands of IP devices, VLANs can be used to separate the network virtually rather than physically, to prevent the broadcast and other traffic from one virtual LAN (typically a data LAN) from impairing the performance of equipment on another virtual LAN (for example, a VoIP LAN) even though the devices are plugged into the same physical network.

VLAN for the IP Telephone and the telephone's data port can each be programmed manually in the telephone or remotely through Enterprise Manager. Ensure that data switch port that the system is plugged into is configured to be in the same VLAN ID as the IP telephones.

Within or without VLANs, 802.1P and Diffserv can be used to provide Quality of Service for voice by allowing voice packets to be prioritized over data packets.

Note When using 802.1Q or 802.1P it is important to ensure that all the network of the ethernet switches and routers are capable of supporting this.

Reasons why a company might want VLANs:

- Security – Separating systems that have sensitive data from the rest of the network decreases the chances that people will gain access to information they are not authorized to see.
- Projects/Special Applications – Managing a project or working with a specialized application can be simplified by using a VLAN that brings all the required nodes together.
- Performance/Bandwidth – Careful monitoring of network use enables the network administrator to create VLANs that reduce the number of router hops and increase the apparent bandwidth for network users.
- Access Lists – Provides the network administrator with a way to control who sees the different types of network traffic. An access list is a table the network administrator creates that lists which addresses have access to that network.
- Broadcasts/Traffic flow – Since a principle element of a VLAN is the fact that it does not pass broadcast traffic to nodes that are not part of the VLAN, it automatically reduces broadcast traffic.

Voice / Tone Signaling

Each DN button can be programmed for either Voice or Tone Signaling as the standard method of internal incoming call signaling. Tone Signaling rings the telephone when a call comes in and ensures better privacy. With Tone Signaling, the called telephone receives a one-second ring tone every three seconds. There are programming settings to adjust/modify the ringing of both internal and external ringing. 10 possible settings for intercom external ring signals can be assigned to a PDN.

With Voice Signaling, station users will hear a tone burst followed by the caller's voice over their telephone speaker when called by another station user locally or over the private

network. Voice signaling allows handsfree talkback from the called telephone on internal and private network Tie line calls.

After calling a directory number that has Voice Signaling, the caller can switch to Tone Signaling by dialing **1**. The signaling method can also be switched from Tone to Voice Signaling by dialing **2**. Whether a call is initiated with Tone or Voice Signaling, it can always be switched back and forth by dialing **1** or **2**.

Note A call to a Voice Signaling DN will not Call Forward No Answer unless the signaling is switched from Voice to Tone Signaling.

Volume Control

Telephone users can independently adjust their handset hearing volume, speaker hearing volume including BGM, speaker incoming tone volume, and beep tone volume.

Messaging

The following is a list of Messaging features. Messaging is categorized into the following feature sets: Automated Attendant, Voice Messaging, Unified Messaging, Networking, Administration, Reporting, and Security.

Automated Attendant

Automated attendant routes incoming calls to the appropriate system extension without operator assistance. One of the benefits of an automated attendant is that it eliminates the bottleneck of calls at the operator's console, particularly during peak hours, and allows callers to reach their desired destination quickly. If a caller is not familiar with the telephone system's extension number, the automated attendant offers the caller the option of accessing a directory assistance function. The function prompts the caller to dial a number up to nine digits that corresponds to the letters in the party's name. The system then performs a lookup and announces the available options.

Departments

In Messaging, Automated Attendant features are configured in Departments. Each department's automated attendant functions can be configured separately. Up to 999 separate departments can be created, each with its own automated attendant greetings, day of week and time of day timers, operator, incomplete call destination and directory assistance.

Department Partitioning

Department partitioning allows for complete separation between departments or companies using one Messaging system, allowing for complete "tenant" functionality.

Departmental Time Zone

Departmental time zone is a configurable setting that defines the appropriate time zone for programmable departmental parameters, such as time of day-based greetings and call routing rules.

Directory Assistance

Messaging allows for incoming calls to the auto attendant to dial the first letters of the called party's first or last name.

Do Not Disturb

A mailbox owner can set "Do not disturb" to have calls sent directly to voicemail.

Follow-Me

A mailbox can be set up to forward a call to an external phone number before the call is transferred to voicemail. When using supervised follow-me, the mailbox owner can perform functions such as record the call, conference in another subscriber, or send the caller back to the mailbox owner's voicemail box.

Follow-Me Connect Verification

The mailbox owner can positively accept the follow-me calls by pressing a key to prevent calls from ending up in cell phone voicemail or other telephone answering devices.

Follow-Me Record to Mailbox

Allows the mailbox owner to record a conversation that has been answered at the follow-me number. The conversation is saved and sent to the mailbox owner's voicemail box as a new message.

Follow-Me Transfer Back

After the mailbox owner receives the call to the external device he can redirect the caller to another internal extension.

Holiday/Date-Based Greeting

Holiday messages and their dates can be pre-programmed into the system. When the internal calendar matches one of these dates, the appropriate holiday greeting will replace the main greeting.

No Response Destination

A destination that incoming callers will be transferred to if they do not respond when prompted by the auto attendant. The system will validate if a caller is still connected to the system before a call is transferred to the no response mailbox. This enables the filtering of calls that were dropped by the caller, but were not disconnected by the central office or the telephone system.

Operation Mode

Operation modes allow a department to operate under different modes such as day, night, emergency, lunch, or holiday. Each mode can have different conditions to handle calls (e.g., different greetings, operators, scripting routings). Operation modes can be set to change automatically or manually.

Simple Single-Digit Dialing

The Messaging departmental conversion tables allow the incoming caller to easily navigate by using single-digit DTMF keystrokes to reach specific company departments, services or extensions.

Time of Day Greeting

Time of day greeting is a time-dependent greeting (e.g., good morning, good afternoon, good evening).

Voice Messaging

Ad-Hoc Groups

A mailbox owner can send or forward a message to a group of mailboxes created on the fly, as opposed to predefined groups. See [“Distribution Groups” on page 53](#)).

Archive Mailbox

Messages can be archived by automatically copying from an originating mailbox to an archive mailbox. For example, hotel reception can access the archive mailbox to allow guests to recover messages after they have already checked out. Archived messages are stored by mailbox number and date for easy access.

Automatic Message Copy

Messages can be copied automatically from an originating mailbox to a destination mailbox. Specific types of messages, such as priority or group can be selected for automatic message copy, and the automatic message copy can happen immediately or be assigned to copy only after a pre-selected amount of time.

Call Queuing

When the automated attendant detects a busy event from an extension it can be set to put all callers on hold in a queue and let each caller know his position in the queue. Messaging will attempt to transfer the caller to the extension after a certain period of time and if the extension is still busy the system will announce to the caller their position in the queue. While holding, Messaging can play promotional announcements to the caller.

Call Record to Voice Mail

The mailbox owner can record an incoming call by using a key press on the telephone key pad. While on an active call, a telephone user can record the conversation and store it in their voice mailbox. Users can replay recorded messages by calling the voice mailbox that has the stored recording and play it back as any other message. Recording to Voice Mail (VM) is available on two-party and multi-party conference calls.

Call Screening

Call screening allows a mailbox owner to require that a caller state her name before a call is transferred to the requested extension. The name is played back to the mailbox owner and the owner can either accept or reject (i.e., send directly to voicemail) the call.

Caller ID (CID) Routing

Calls can be routed, based on caller ID information, to a mailbox or application. A complete or partial number (which includes only the area code, or area code + exchange) can be used. Caller ID routing tables are available at the system level, departmental level and for every voicemail box.

Cancel Operation

Allows a mailbox owner to cancel out of the current action and be brought back to the previous menu.

Change Message Time

The date and time of a message can be automatically updated when re-saved by a mailbox owner in order to extend message end-of-life.

Check Message Count

The mailbox owner can check how many new and saved messages are in his mailbox.

Codec Support

Codec support is built-in support for G.711 (ulaw and alaw) and G.729.

Confidential Message

A message may be marked as confidential and the recipient will be informed that it is confidential before the message plays.

Delete from Subscriber's Mailbox

A message may be deleted from another subscriber's mailbox by the subscriber who sent it, if it has not yet been listened to.

Direct Transfer to Voice Mailbox

The transferring party can transfer a call directly to a person's voice mailbox without waiting for the call to forward from the called party's telephone. The voice mailbox does not need to be associated with an active telephone. Direct transfer to voice mail (VM) can be performed to a centralized VM system connected to a network node other than the user's node.

The transferring party presses **Direct Transfer to VM** and dials the mailbox number, and the call transfers immediately on receipt of the last digit. The transferred party hears the greeting associated with the specified mailbox and can then leave a message.

Direct Transfer to Voice Mailbox simplifies getting a call for a busy or absent employee to his/her mailbox. It eliminates the need for the caller to enter the desired mailbox number after being connected to the voice mail system. This feature is available using standard DTMF or SMDI VM integration and does not require Toshiba proprietary VM integration.

Distribution Groups

A new message can be sent, or a message can be redirected to multiple individuals, without having to input individual mailbox numbers. Distribution groups are either global (available to all mailboxes) or private (each mailbox owner can establish their own groups). The system can manage up to 99,999 distribution groups (private and global) with unlimited members and groups within groups.

End Recording Key

The administrator can define a specific key that callers must press to stop their recording (for example, #). This is useful to prevent accidental termination of a recording.

Envelope Information

Envelope Information includes time and date information, caller ID, sensitivity and urgency of the message. Envelope information can be programmed to automatically play with a new message or only play when requested by the mailbox owner. If set to play automatically, it can be programmed to play either before or after the voicemail message.

External Message Notification

The mailbox owner can schedule notification to external devices when a message is received, such as text message to cell, notification to pager, and call-out to another phone number.

First-time User Tutorial (Mailbox Set-up)

Assists the mailbox owner with the set-up of her voicemail box (change password, set up personal greeting).

Forward/Rewind

A configurable timer that defines how far backward or forward a message will skip when the mailbox owner uses the skip backward/ forward key press during message playback.

Future Delivery

A mailbox owner can input a time and date to schedule a message for future delivery.

Hospitality Mailbox

A hospitality mailbox is a streamlined mailbox that allows guests (users) to retrieve room messages from any phone on or off the property and access voicemail through a web browser. The front desk can also retrieve messages for a guest as well as retrieve messages from the archive for a guest that has already checked out.

Mailbox Owner Language Selection

A default language can be set for each mailbox owner. This is the language of the prompts that a mailbox owner will hear when calling into his mailbox. If this feature is not set, the mailbox owner will hear the language identified in department settings.

Mailbox Time Zone

This configurable setting defines a time zone for the mailbox owner which is used during envelope information message playback. The owner will hear the message delivery time relative to their time zone.

Message Call Back

While listening to a message, a mailbox owner can initiate a call back to the caller (based on caller ID). In a supervised call back the Messaging remains on the call, allowing the use of functions such as call record, transfer to voicemail, or transfer to another mailbox owner.

Message Cascading

An administrator can create a set of independent rules to determine what happens to a message after it is received in a mailbox. For example, when a message comes in to a sales group mailbox it is automatically copied to all members of that group. The administrator can also define cascade

rules that will delete or save the messages for all the members as soon as one member has listened to the message.

Message Delete Confirmation

Message delete confirmation requires the mailbox owner to confirm message deletion by pressing an additional key. This option can be enabled or disabled by the system administrator.

Message Waiting Indication

The system will trigger a light on a phone when a new message is received. In addition, an indication on the phone display shows the mailbox owner how many phone messages are in the mailbox.

Notification of Non-Receipt

A mailbox owner may request notification when another mailbox owner does not listen to a specific message.

Octel® Prompt Emulation

In addition to the Messaging telephone user interface, the system includes a prompt set that mimics the Octel's system. The Octel prompt emulation can be used on a mailbox-by-mailbox basis or system-wide.

Park and Page

A caller is notified that the called party does not answer and asks if the caller wishes to page the called party. This feature can be set to be used at all times or only during night and/or day mode.

Pause Message

A configurable timer that defines how long a message will pause when a mailbox owner uses the pause key press during message playback.

Personal Assistant

Personal assistant allows the caller to press a single digit during the mailbox owner's mailbox greeting to be transferred to another extension.

Personal Automated Attendant

The Messaging mailbox conversion table allows the mailbox owner to provide a caller with directives to perform certain functions, such as transfer to assistant, replay greeting, contact pager, transfer to follow-me number, record a message, page mailbox owner, send caller's telephone number to email.

Play New Messages Automatically

Play new messages automatically is a programmable parameter that allows new messages to be played automatically when a mailbox owner logs in (without pressing any digit to begin message playback).

Priority Message

A message may be marked as priority to be sent to the front of the mailbox owner's message inbox.

Programmable Menu Timeout

A configurable timer that defines the number of seconds the system waits for an entry from the mailbox owner before it times out.

Redirecting Messages

A mailbox owner can forward a message to another subscriber's mailbox or to a group of mailboxes.

Retrieve a Deleted Message

A mailbox owner can retrieve a deleted message and move it back to his saved messages folder up to one day after being deleted (or a longer period of time, as defined by administrator).

Return Receipt

A message may be marked as return receipt to request confirmation that the recipient received and listened to the message.

Review Saved Messages

A mailbox owner may listen to messages already moved to the saved folder.

Speed Control

Allows the mailbox owner to increase and decrease the speed of message play back.

Soft Key Control of Voice Mail

The Liquid Crystal Display (LCD) of IP telephones connected to the *VIPedge* system, provides a visual presentation of the options within Messaging mailbox menus. Depending on the size of the LCD screen, some or all of the menu options are available by pressing corresponding soft keys located next to the desired option or function. When the phone is idle and a message arrives for an extension on the phone, the Msg LED is activated and the LCD shows the number of new messages that are currently in the mailbox. If any of the messages are marked as priority, the LCD shows the number of new and priority messages.

After a successful login to a mailbox, the LCD presents the mailbox Subscribers Menu options—listen to messages, record messages, and personal options. Selecting any one of these options presents a new LCD with the next available menu options.

Note For general information on using Soft Keys on your phone, refer to the appropriate Telephone User Guide. See the *VIPedge* Telephone, Messaging, and Call Manager User Guide for a sample list of available Soft Keys.

Subscriber's Menu

The subscriber's menu provides the mailbox owner access to all available features of the voicemail system.

System and Department Language Selection

Messaging supports multiple languages and can be used independently or simultaneously per system department group.

Additional languages available by request. Contact Toshiba Sales Applications Desk for details.

Variable Extension Length

Variable extension length is a configurable option that sets the number of digits that make up a valid extension number.

Variable Mailbox Length

Variable mailbox length is a configurable option that sets the number of digits that make up a valid mailbox.

Voice Mail Call Monitor

This optional feature enables a mailbox user to monitor a message while it is being recorded in his mailbox. This feature is active when the User's telephone is idle or for calls that are forwarded to voicemail and when a message recording begins. If the mailbox owner is present when the call comes in, he can press the "Call Monitor Button" to hear the caller leaving the message.

When the caller stops the recording process (by hanging up) the monitoring ends and the mailbox user hears the prompt, "The caller has finished. Good bye." If more than one caller is leaving a message at the same time, then the mailbox user is able to monitor the last caller.

Volume Control

Allows a mailbox owner to decrease or increase volume during message playback.

Wake-Up Call

A mailbox can be programmed to make two types of wake-up calls:

- System makes daily wake-up call until deactivated by mailbox owner.
- System makes a one-time wake-up call and is then deactivated. Can be set to enable or disable by the system administrator.

Unified Messaging

Unified messaging allows a mailbox owner to access voice messages directly through an email inbox. Emails may also be listened to and can be managed from the voicemail box.

Integration with Email Clients

The Messaging unified messaging provides seamless and fully synchronized integration with existing email clients without the requirement of a desktop client. This allows Messaging unified messaging to be desktop operating system-independent and greatly minimizes administration and deployment workload.

Messaging as an IMAP Server

This is an independent mail server configuration where voice and deleted messages appear in a separate folder from the mailbox owner's primary inbox.

Messaging as a POP Server

This is an independent mail server configuration where voice messages are displayed in the mailbox owner's primary inbox. Messages are not synchronized.

Administration

System administration is done using a web-based application named Enterprise Manager. An administrator's password is required for access to all system administrator functions.

Callout Length

A definable maximum length for a number the system is allowed to callout.

Class of Service (COS)

Class of service controls each specific mailbox's activities including personal options, incoming calls, transfer supervision, ringer and housekeeping. Messaging can accommodate up to 999 COS of service definitions for maximum system flexibility.

Housekeeping

A configurable length of time that defines how long a new, saved or deleted message will be stored. Each COS definition has its own housekeeping timers.

Import Data

New mailboxes or caller ID routing numbers can be batch imported via a CSV file.

Mailbox Mapping

An incoming DNIS/DID can be mapped to a mailbox number.

Mailbox Password

A mailbox owner's mailbox is protected by a numeric security code. Maximum password length is nine digits.

Mailbox Role

The mailbox owner/administrator's interface is controlled by roles that manage mailbox owners' and administrators' viewing and administration permissions.

Mailbox Search

An administrator can search for specific mailboxes based on mailbox owner's name, department, class of service, etc.

Mailbox Status

A real-time report showing all mailboxes in the system that currently contain messages. This report can be displayed on an overhead projector to show mailbox owners their message status when they have no access to a physical phone with a message waiting light.

Mailbox Swap

Mailbox swap is a database swap between mailboxes that includes all feature programming, messages and greetings.

Mailbox Transfer

A single box or range of boxes may be moved to a new numbering plan. The transfer includes all feature programming, messages and greetings.

Maximum Greeting Length

A configurable option to set a maximum mailbox greeting length. Options are also available for those mailboxes requiring an unlimited greeting length.

Maximum Message Length

Mailboxes may be assigned a maximum message length that determines the length of a message the incoming caller can leave for that mailbox. Options are also available for those mailboxes requiring an unlimited message length.

Maximum Messages

Mailboxes can be set with the maximum number of messages they may receive. If the maximum is reached the caller will be notified there is no room in the mailbox.

Maximum Silence Timer

Maximum silence timer is a configurable option that sets the maximum silence duration within a message. If reached, the message recording will terminate and the caller will be offered additional options (send message, continue recording, rerecord, etc.).

Message Playback Order

Messaging playback order allows each mailbox type (new messages, saved messages, email and deleted messages) to be independently assigned as first-in-first-out or first-in-last-out.

Minimum Message Length

Minimum message length can be set to prevent “hang-up” messages.

Push Mailbox

A range of mailboxes can be updated with a field change.

Quick Glance

Allows the administrator to see a list of all mailboxes with the following information: mailbox, extension, first name, last name, class of service, department, mailbox type, message waiting indicator, transfer mode, email client and call control client.

System Backup

The system can perform a daily or weekly backup of all system data including messages, greetings and configuration. The system can also automatically upload a backup to a remote FTP site and create multiple stored backup files.

System Monitor

Monitors the activity of the channels to display which channel is in use or on stand-by, which mailbox is in use and which mode the Messaging is using.

Transfer Supervision

Automated Attendant calls can be set to transfer supervision type (none, partial or full). If fully supervised, the number of rings for no-answer result can be defined.

Variable Password Length

Variable password length is a configurable number of digits that make up a valid password number. Each department may have a different variable password length.

WebController

All administration can be managed through a web-based interface. Administrators can create different roles for sub-administrators and mailbox owners to manage subsets of the system. The WebController can be used on a secure or non-secure http port.

Reporting

Messaging records all activity from calls coming in or out of Messaging. By collecting this information, administrators can generate different reports. These reports help the system administrator manage and maintain the system to ensure optimum performance. Reports are available for viewing, printing or emailing and can be accessed from the reports menu using Enterprise Manager.

Full Report

This comprehensive report includes the following information: date, channel, time, department, mailbox number, duration of call, type of call (external caller or internal user), incoming or outgoing call, call result (answered or unanswered) and caller ID.

Mailbox List

This report displays a detailed list of all mailboxes and includes mailbox, extension, subscriber name, department, COS, usage, new messages, saved messages, email messages, deleted messages and total messages.

Mailbox Usage by Date

This report displays the mailbox usage by date. The usage report records any activity made from the mailbox extension, which includes any calls received or made, whether they are external or internal.

Mailbox Usage Daily

This report displays mailbox usage information by date.

Message by Mailbox

This report provides a history of all messages by mailbox.

Message Activity

This report displays message activity by mailbox.

Outbound calls

This report provides information on all outbound calls placed using Messaging. The report includes mailbox number, date, time, result (answered/ unanswered), call duration and number dialed.

Port Statistics

This report indicates summary activity per port on specified dates. Information includes the port or channel number, number of internal versus external calls, total number of calls, total duration, number of transfers and completions.

Scripts

Messaging creates customized routines or scripts for directing callers around the system. Scripts programming is a centralized application that can create various choices to a caller as well as being the standard tool for setting up “Audio Text” mailboxes and building custom applications. Scripts offers many different applications, including:

- Intelligent call routing, whereby callers are routed based on time of day, day of week, and other criteria such as caller ID.
- Interactive questionnaires
- Recorded information

Scripts requires a license for each application desired.

Script Logging Reports

This report displays a list of all the calls to a script mailbox including time, date, caller information and key presses.

System Group List

This report displays all broadcast groups in the system and shows if they are system groups or personal groups and whether they have recorded the group name.

System Hourly Statistics

This report displays the total activity of Messaging on an hourly basis for the dates specified.

System Statistics

This summary report displays the total activity of the voicemail for the dates specified.

Unattended Mailboxes

This report lists all the mailboxes that have been created but not yet activated through the subscriber’s menu.

Security

Limited Dial-Out Digits

A limited number of digits are allowed in a dial-out according to class of service to prevent international toll fraud.

Limited Password Entry Attempts

When a certain number of password entry attempts per call is detected, the Messaging will immediately hang up the call to prevent automated dialers which try to expose passwords by “brute force” attacks.

Mailbox Lock and Administrator Notification

When a certain number of password entry attempts per mailbox is detected Messaging locks the mailbox to prevent further use and notifies the system administrator via email.

Secure Authentication for Outgoing Email

Outgoing emails sent from Messaging are SSL encrypted and can be configured to use secure authentication.

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Appendix – Specifications

This appendix includes detailed information on the items listed below. The sections in this appendix apply to the VIPedge solution, unless otherwise stated.

Datacenter Specifications

The VIPedge solution is housed in a major data center, here are some of the specifics:

Connectivity

- Redundant dedicated connections to Tier 1 Internet backbones

Power

- 2000KW Base Building Standby Generator.
- Generator Plug is installed directly outside our building to connect a Portable Standby Generator.

Security

- 24/7/365 security.
- Onsite network monitoring.
- Strategically placed access scanners.
- Extensive use of physical man traps.
- Dual authentication access doors.
- State-of-the-art CCTV system to cameras with archived storage.

Environmental

- 3 independent HVAC units providing a fully redundant N+1 system.

Network Operations Center

- 2,500 square foot facility.
- 24/7 manned network control and operations center.

Physical Isolation

- Built as a “building within a building” to minimize the risk of natural disasters.

Station Dimensions

Dimensions for the 5000-series, IP telephones and related equipment are listed in [Table 1](#).

Table 1 Station Dimensions

Device	Height		Width		Depth	
	Inches	mm	Inches	mm	Inches	mm
IP5131-SDL [15-degree tilt], IP5531-SDL, IP5631-SDL	5.1	129	10.16	258	6.10	155
Add-on Module (LM5110) [15-degree tilt]	4.0	102	3.54	90	6.10	155

IP Telephone Power Consumption

The power consumption for IP5000-series telephones and the Add-on modules is shown. Use this information to calculate the Power over Ethernet (PoE) requirements and UPS capacity.

Table 1-1 IP Telephone and Add-On Module Power Consumption

Telephone Model ¹	Option		Power Rating (Watts)	Current (A) ²	Typical (Watts) ³	Typical Current (A) ⁴	IEEE802.3af PD Class
	Model	Qty					
IP5131-SDL	none	--	7.4	0.15	6.2	0.13	0
IP51xx +	LM5110	2	10.3	0.21	8.6	0.18	0
IP51xx +	LM5110	1	9.4	0.20	7.8	0.16	0
IP5531-SDL	none	--	3.6	0.08	3.0	0.06	2
IP5631-SDL	none	--	4.1	0.08	3.3	0.07	2
IP5631-SDL	LM5110	2	6.4	0.13	5.3	0.11	2
IP5631-SDL	LM5110	1	5.6	0.12	4.7	0.10	2

1. Power ratings are only telephone and option modules consumption. The values do not include LAN cable power loss, and apply to PoE, not local power supplies.
2. Power ratings are only telephone and option modules consumption. The values do not include LAN cable power loss, and apply to PoE, not local power supplies.
3. Typical means that it is only an example and there is no guarantee implied. The “typical” value might be used for a calculation of actual UPS backup time in an average installation.
4. Typical Current (A) = Typical Watts / 48 v

System Tones

Tones which can be heard from speaker or handset are described in [Table 2](#).

Table 2 Call Progress Tones

Tone Name	Conditions	Ringling Cadence
Prime Dial Tone	Prompting to dial [DN] or access code or to press a feature button or to dial 9 + number.	350/440Hz continuously On.
Secondary Dial Tone (optional)	Prompting to dial [DN] or access code or to press a feature button, with someone on Consultation Hold.	
DND-Stuttered Dial Tone (optional)	Same as Prime Dial Tone with implication of DND activated. MW-Stutter dial tone has priority over this tone.	480/620Hz 0.125 sec. 4 bursts apart 0.125 sec., 350/440Hz 3 sec. On, repeat.
MW-Stuttered Dial Tone	Same as Prime Dial Tone with implication of MW received. This tone has a priority over DND-Stutter dial tone.	350/440Hz 0.1 sec. 5 bursts apart 0.1 sec., 3 sec. On, repeat.
Entry Tone	More digits are required such as account codes, some indexes, etc.	350/440Hz, 0.1 sec. 3 bursts apart 0.1 sec.
Ring Back Tone	Ringling the destination	440/480 Hz 1 sec. On, 3 sec. Off, repeat.
Success Tone (Confirmation Tone)	Operation was successfully accepted.	350/440 Hz, 3 bursts of 0.125 sec., apart 0.125 sec.
Reject Tone	Operation was rejected. After this tone is done, the original conversation is resumed.	1209 Hz 0.25 sec., 500 Hz 0.25 sec., 3 times.
Busy Tone	Destination is busy. Invoke desired feature or retry later.	480/620 Hz, 0.5 sec. On, 0.5 sec. Off, repeat.
Reorder Tone	Either the operation failed or the call is terminated. Hang up.	480/620 Hz, 0.25 sec. On, 0.25 sec. Off, repeat.
DND Tone	The destination is in the Do Not Disturb mode.	480/620 Hz, 0.125 sec. On, 0.125 sec. Off, repeat.
Splash Tone	Voice calling starts. Applicable to Voice Paging and Speaker OCA.	500 Hz, 1.0 sec. On.
Barge-in Warning Tone	Somebody is listening to (monitoring) the conversation.	440 Hz 1.0 sec. On.

Ring tones are described, along with their cadences in [Table 3](#). Due to the limitation in the tone generation algorithm, the listed tone duration is slightly different from the actual one.

Table 3 Ring Tones

Tone Name	Description	Ringling Cadence
Internal/External Ring 1	Incoming call from internal or external party to IP telephones. (10 different ring tones are available with R1.3 or higher software.)	500 Hz 1 sec. On, 3 sec. Off, repeat
Internal/External Ring 2		1300 Hz 1 sec. On, 1 sec. Off, repeat
Internal/External Ring 11		500/640 Hz 1 sec. On, 3 sec. Off, repeat
Internal/External Ring 12		500/640 Hz 1 sec. On, 1 sec. Off, repeat
Internal/External Ring 13		860/1180 Hz 1 sec. On, 3 sec. Off, repeat
Internal/External Ring 14		860/1180 Hz 1 sec. On, 1 sec. Off, repeat
Internal/External Ring 15		1300/1780 Hz 1 sec. On, 3 sec. Off, repeat
Internal/External Ring 16		1300/1780 Hz 1 sec. On, 1 sec. Off, repeat
Internal/External Ring 17		860/1180 Hz 0.5 sec. On, 1300/1780 Hz 3 sec. Off, repeat
Internal/External Ring 18		860/1180 Hz 0.5 sec. On, 1300/1780 Hz 1 sec. Off, repeat
Recall		A call is returned & needs to be answered.
Ring Over Busy (Internal)	Call rings an idle [DN] or CO line button while phone is busy. Another internal call offered to an idle button while the station is busy.	2 kHz interrupted at 10 Hz, 1 sec. On, 3 sec. Off, twice or repeat (For Call Waiting, twice only).
Call Waiting (Internal)	Internal call is waiting for the busy button. A call is camped-on to a busy [DN] or CO line button.	
Ring Over Busy (External)	Call rings an idle [DN] or CO line button while phone is busy. Another incoming call is offered to an idle button while phone is busy.	2 kHz interrupted at 10 Hz, 2 bursts of 0.25 sec. apart 0.25 sec., twice apart 3 sec. or continuous (For Call Waiting, twice only). Standard telephones with Caller ID also receive an 80 ms burst of CAS tone at -14 to 32dBm
Call Waiting (External)	External call is waiting for busy station. A call is camped-on to a busy [DN] or CO line button.	
Volume Control - Ringing Speaker	Adjusts speaker volume for ringing state.	500/640 Hz continuous.

Other types of tones that do not fit in the previous categories are listed in [Table 4](#).

Table 4 Administration/Programming Tones

Tone Name	Description	Ringling Cadence
Confirmation Tone	During user programming or administration mode, indicates the acceptance of input.	2 kHz two bursts of 0.125 sec. apart 0.125 sec.
Denial Tone	During user programming or administration mode, indicates the denial of input.	2 kHz 0.75 sec. On.
Volume Control - Beep	To adjust the beep volume.	2 kHz interrupted 10 Hz, continuous.

IP Telephone Bandwidth Requirements

The amount of bandwidth required for communications over a particular IP network segment depends on the number of voice channels supported, the anticipated call setup traffic, and how much other data network traffic is present.

The quality of service (Excellent, Good, Fair, and Poor) provided by the IP telephones depends heavily on the LAN parameters as shown in [Table 5](#). This table shows the amount of bandwidth required for each IP voice call (without data traffic) based on the interval and the CODEC.

Table 5 IP Telephone Quality of Service

IP Network Quality Parameters		Speech			
		Excellent: No one perceives delay. ¹	Good: Very few people perceive delay. ¹	Fair: Some people may perceive delay. ¹	Poor: Many people may perceive delay. ¹ IP telephones is usable even with a “Poor” rating if delay is acceptable.
Latency (Round trip delay) ²		20ms or less	50ms or less	100ms or less	200ms or less
Jitter ²		20ms or less (-10 ms ~ +10ms)	50ms or less (-25ms ~ +25ms)	50ms or less (-25ms ~ +25ms)	50ms or less (-25ms ~ +25ms)
Packet loss ²		1×10 ⁻³ or less	1×10 ⁻³ or less	1×10 ⁻³ or less	1×10 ⁻³ or less
Packet error ²		1×10 ⁻⁴ or less	1×10 ⁻⁴ or less	1×10 ⁻⁴ or less	1×10 ⁻⁴ or less
Speech quality dependency on CODEC parameters		Speech quality as the combination of the above network environment and the CODEC parameters.			
CODEC and packet interval in ms	Bandwidth per channel (Single direction, control channel included)				
G.711 at 20ms i	88kbps ³	Excellent	Good	Fair	Poor
G.711 at 40ms i	76kbps ³	Excellent	Good	Fair	Poor
G.729A at 40ms	20kbps ³	Good	Good	Fair	Poor
G.729A at 80ms	14kbps ³	Good	Fair	Poor	Poor

1. Ratings of Excellent, Good, Fair, Poor were based on the tester in a quiet room and the tester could not see the other call party.
2. When selecting router equipment, the Latency, Jitter, Packet loss and Packet error conditions above should be considered as well as the bandwidth. Bandwidth can be calculated with the CODEC and packet size. For better results, more bandwidth may be required, depending on the amount of overall data traffic. For more details on QoS refer to “A Handbook for Successful VoIP Deployment: Network Testing, QoS, and More” by John Q. Walker, NetIQ Corporation on www.netiq.com.
3. Use this number to estimate the bandwidth needed for the CODEC and IP headers required to achieve an expected Quality of Service (Excellent, Good, etc.). When planning you should allow extra bandwidth, especially when mixing voice and data.

When sharing voice and data on the same network segment, the data will cause some jitter in voice communications, especially on slower segments. [Table 6](#) shows calculations of the amount of jitter assuming a worst case data packet size of 1500 bytes (Maximum Transmission Unit (MTU) = 1500) based on a segment’s bandwidth. This also requires that the routers connecting the segment through the WAN support Diffserv.

Note A router that doesn’t support Diffserv may stack multiple data packets together increasing the jitter perhaps indefinitely. And the voice quality will be indeterminate.

Table 6 IP Jitter on Mixed Voice and Data WAN

No. of B-Channels of WAN	Bandwidth (kbps)	Time to transmit max. MTU (ms)	Expected Jitter (ms)	Class
2	128	93.75	100	Poor
4	256	46.88	50	Fair
6	384	23.44	30	Fair
8	512	15.63	20	Good
24	1536	1.00	1	Excellent

Class definition categories are shown in [Table 7](#).

Table 7 IP Class Definitions

Class	Delay (ms)	Jitter (ms)
Excellent	< 20	< 10
Good	< 50	< 20
Fair	< 100	< 50
Poor	< 200	< 100

[Table 8](#) shows the amount of bandwidth required for setting up and tearing down calls independent of the amount of voice traffic.

Table 8 IP Bandwidth Required for Call Setup

Traffic Rate (BHCA ¹)	Required Bandwidth
1000	6
2000	12
4000	23
6000	36

1. BHC = Busy Hour Call Attempts

So the amount of bandwidth that is required on a segment to support a specific number of calls is the sum of the number of channels multiplied by the bandwidth for the selected CODEC and interval, plus the bandwidth required for the selected number of busy hour call attempts. And the jitter is determined by the bandwidth of the WAN segment.

Example: If you want to support 4 calls using the G.711 CODEC with a 20 msec. interval, this requires 4 x 88 kbps = 352 kbps of bandwidth. In addition, to support 1000 busy hour call attempts, 6 kbps must be added for a total of 358 kbps. If only voice is going to be carried on the segment, then a 384 kbps segment (6 B-channels) is sufficient.

If voice and data are going to be mixed on the segment, then at least 25% (89.5 kbps) should be added, or more, based on the amount of data traffic desired. In this case, a total of 447 kbps will be required which would best be supported by a 512 kbps segment (8 B-channels). This would result in an expected jitter of 20 ms in the voice traffic.

When using the MEGACO+ protocol with IP telephones, keep alive packets are exchanged between the IPU and the phones. This traffic amounts to 3 kbps per phone.

VIPedge Capacities

The following tables contain VIPedge capacities.

Note These capacities are maximum values, however these values may vary depending on the configuration.

Table 1 Station/Peripherals System Capacities

Stations	Less than or equal to 60 Users	More than 60 Users
IP5000-series stations, IP4100 DECT (total)	60	500
IADM on IP Telephone	53 per system	400 per system
Simultaneous calls	96	413

Table 2 Trunk Capacities

Trunks	Less than or equal to 60 Users	More than 60 Users
SIP Trunk channels	60	260
Channel Groups	2	2

Table 3 IP Telephone Station Buttons

Station Buttons per System	Less than or equal to 60 Users	More than 60 Users
Call Forward, Personal CF Buttons	72	560
Caller ID (CLID) button (IP telephone only)	72	560
Group CO (GCO) Line Buttons	64	264
Pooled CO Line Buttons ²	64	128
CO Group and Pooled Line Buttons ²	128	264
Flexible Telephone Buttons (depends on system configuration)		
Line and DN Buttons in use at the same time	1,440	3,360
Message Waiting Registration (DNs with MW)	130	800
Multiple Appearances of DN on Telephones	2,300	15,000
Night Transfer Buttons	32	128
One Touch Buttons	1,600	12,000
Primary Directory Numbers [PDNs] per system	72	560
Phantom Directory Numbers [PhDNs] per system	288	2,240
[PhDNs] with Message Waiting Indication LED	18	128

Table 4 System Feature Capacities

Features	Less than or equal to 60 Users	More than 60 Users
Pilot DNs	90	256
Advisory LCD Messages (Set on a Telephone)	1	1
Advisory LCD Messages Lists (per System)	10	10
Call Forward, System CF Patterns	4	32
Call Park Orbits (General)	14	64
Call Park Orbits (Individual)	64	412
Minimum / Maximum Caller ID per Station	Min:0, Step:5	Min:0, Step:5

Table 4 System Feature Capacities (continued)

Features	Less than or equal to 60 Users	More than 60 Users
Maximum is the maximum records of a station	Max:100	Max:100
Maximum number of Stations that can have Caller ID/ANI/DNIS Numbers stored (Call History records)	72	560
	1,440	2,000
CO Line Groups - Incoming Line Groups (ILG)	32	128
CO Line Groups - Outgoing Line Groups (OLG)	32	128
Outgoing Line Groups (OLG) Members per system (Trunks + ISDN Line Service Index)	64	264
Conference Channels (depends on system configuration)		
Conference Party types (up to 8 total lines + stations) ²	8	16
Two-CO Line simultaneous Connection ² (Two party only, no telephone or VM channel)	32	132
DID Numbers for Calling Number ID/system	360	1,000
DNIS/DID Incoming Numbers (1~7)	360	1,000
DNIS/DID Routing Destination Numbers (1~6 digits) ¹	360	1,000
DNIS/DID Routing Destination Numbers (7~32 digits) ³	720	2,000
E911 Groups	8	128
Emergency Call Groups	8	8
Hunt Groups (Serial/Circular/Distributed combined)	90	640
Hunt Group Size (DNs per group)	72	560
Hunt Group Stations (depends on system configuration)		
Multiple Call Ring Group	16	64
Off-hook Call Announce to Telephone Speakers ²	20	560
Page Groups (Phones with or without External Zones)	4	16
Paging – (Group Page – simultaneous stations paged) (depends on system configuration)		
Pickup Groups	5	32
Ring Tones (External Call Ring Tones for IP telephones)	10	10
Ring Tones (Internal Call Ring Tones for IP telephones)	10	10
Speed Dial - Station SD numbers per system ³	1080	1080
Speed Dial - System SD numbers per system	800	800
Destination Restriction Level (DRL) Classes	16	16
Verified Account Codes	135	1,000
SIP URI per system	360	1,000
SIP Trunk service Index	32	128
LCR Exception Table Size	1,280	2,500
MAX digit number for LCR Route Plans	19	19
MAX Number of LCR Route Plans	64	128
CSTA Device Monitors	512	1,152
CSTA Call Monitors	320	560

1. Each DNIS/DID Number uses up to three Routing Destination Numbers (Day1, Day2 and Night) in any combination of (1~6) and (7~32) digit numbers.

2. This is not the number of simultaneous OCA but the terminal number of OCA available. (Simultaneous number is limited by maximum capacity of line and call).

3. Up to 100 Station SD numbers, allocated in increments of 10, can be programmed per station.

Application Capacities

Table 5 Media Server

	VIPedge
Resources	Dynamic - Depends on the number of VIPedge Users.

Table 6 Call Manager

	VIPedge
Users with Call Manager	60

Table 7 Messaging

	VIPedge
Hours of Storage	75 minutes average per user; 30 days message retention

Part Numbers

The following are VIPedge part numbers.

Services

Part Number	Part Name	Part Description
V-BIZ-SYS	VIPedge Solution	VIPedge Solution License - One per system/ container. This is automatically added by the quoting system. This is used on Unlimited ¹ as well as Call Center Systems.
V-BIZ-CHU	VIPedge Channel- Unlimited ¹	VIPedge Channel - Unlimited ¹ Local / Long Distance Minutes. International Calling charged separately on Per Minute basis.
V-BIZ-CHCC	VIPedge Channel- Call Center	VIPedge Channel - Call Center Per Minute Local / Long Distance. First 1200 Minutes Local / Long Distance included, then billed Per Minute. Additional Local / Long Distance calling is \$0.019 per minute. International Calling not included.
V-BIZ-CHCT	VIPedge Channel- Call Center Toll Free	VIPedge Channel - Call Center Toll Free Per Minute Local / Long Distance. First 1200 Minutes Local / Long Distance included. International Calling not included.
V-BIZ-DIDPK	VIPedge DID Parking Fee	VIPedge DID Parking Fee - a monthly recurring charge
V-BIZ-STD	VIPedge Business Standard User	VIPedge Standard User Features including DID, PBX, UM with Follow-Me, and Call Manager Advanced with VoIP Plug-in.
V-BIZ-MAIN	VIPedge Auto Attendant	VIPedge Main Number DID - Main Number or Department Number with Auto Attendant Feature.

1. Unlimited Local / Long Distance service is subject to Toshiba's Acceptable User Policy. Businesses that average more than 1,200 minutes per month per channel for local and long distance calls will be classified as a Call Center. If a business fits the Call Center classification for one or more months, the business will be re-configured with "V-BIZ-CHCC - VIPedge Channel - Call Center" channels and will be charged based on the standard Call Center per minute rate plan after the initial 1,200 included minutes are used.

Equipment

Part Number	Part Name	Part Description
IP5531-SDL	IP5531-SDL IP Phone	IP Phone, 9-line black and white display, 100 Megabit Ethernet, POE Support, No Backlight, Add-on Module Port, or External Speaker Port support.
IP5631-SDL	IP5631-SDL IP Phone	IP Phone, 9-line black and white display, 100 Megabit Ethernet, With Backlight, Add-on Module Port, External Speaker Port, and POE Support.
IP5131-SDL	IP5131-SDL IP Phone	IP Phone, 9-line black and white display, Gigabit Ethernet, With Backlight, Add-on Module Port, External Speaker Port and POE Support.
LM5110	IP5000 10-Button LCD Expansion Module	10-Button Backlit LCD Expansion Module for IP5000 Series. Up to two LM5110's supported per phone.
LADP2000-3A	IP5000 Series AC Adapter	IP5000 Series AC Adapter for use with IP5000 Series Telephones. Optional.
IP4100-DECT	IP4100-DECT IP Phone	IP Phone, SIP DECT, Color Display
IP4100-BASE	IP4100-BASE Base Unit	Base Unit for IP4100-DECT SIP DECT Phone.
1700601G2	Adtran 3120 Router	Adtran NetVanta 3120 Router Firewall - Fixed-port Ethernet secure access IP router with integral 4-port switch. Security features include stateful inspection firewall and IPSec VPN support. Routing features include IP routing, static routes, RIP v1/v2, OSPF, and BGP. Management features include Command Line Interface (Telnet and SSH) and Web GUI (HTTP and HTTPS). Access Controller capable for managing up to 8 NetVanta Access Points. Standalone desktop housing with external power supply.
1202820G1	Adtran 3430 Router	Adtran NetVanta 3430 Router Firewall - Modular access router that includes one (1) network interface slot and two (2) integral 10/100BaseT Ethernet ports. Supports all current NetVanta Interface Modules. Desktop, wallmount, or rackmount 1U metal chassis. 19" rackmount brackets and wallmount bracket included.
4200821E2	Adtran 3448 Router	Adtran NetVanta 3448 Router Firewall - Access router for frame relay and point-to-point connectivity. Includes modular network interface and 2 integral 10/100BaseT Ethernet ports. Also includes integrated 8 Port Fully-Managed Layer 2 Ethernet Switch. Supports all current NetVanta Interface Modules. Features include IP routing, OSPF, BGP, RIP V1 and V2, Static routes, 802.1d Bridging (all protocols), Command Line Interface, Frame Relay and PPP WAN Protocols, optional Modular Dial Backup.

Appendix – Specifications

Part Numbers

Equipment (continued)

Part Number	Part Name	Part Description
ES-2010G	LG Ericsson 8 Port Switch	8 Port 10/100/1000 Advanced Smart Switch plus 2 SFP uplink ports.
ES-3024GP	LG Ericsson 24 Port Switch	24 port 10/100/1000 Managed switch with PoE
ES-3052G	LG Ericsson 48-Port Switch	48 port 10/100/1000 Managed switch plus 4 SFP ports.
MP1122SSIP-BDL	Audiocodes SIP Analog VoIP Gateway	Audiocodes MP112/2S/SIP Analog VoIP Gateway + Audiocodes Enhanced Technical Support + Hardware Replacement Warranty. 2 FXS, SIP Package including 2 FXS analog lines, single 10/100 BaseT, AC power supply, G.711/723.1/726/727/729AB Vocoders, SIP.
V-DID-USR	User Installation	Installation of User DID Phone Number
V-DID-LNP	Line Number Portability	Line Number Portability of existing DID from previous service to VIPedge solution Important! Local Number Portability will take up to three calendar weeks (7 to 21 days).
V-IN-USR	User Installation	Installation of User Service including IP Phone and/or UC Client, DID Setup, and Feature Configuration
V-INSTALL-OD	General Install Service - Originating Dealer - \$10 Increment	General Install Service by Originating Dealer. \$10 Increments.
V-INSTALL-ID	General Install Service – Installing Dealer - \$10 Increment	General Install Service by Installing Dealer. \$10 Increments.
V-PERIPHERAL-OD	General Install Peripheral – Originating Dealer - \$10 Increment	General Install Peripherals and Materials by Originating Dealer. \$10 Increments.
V-PERIPHERAL-ID	General Install Peripheral – Installing Dealer - \$10 Increment	General Install Peripherals and Materials by Installing Dealer. \$10 Increments.

Index

A

- abandoned call numbers, 70
- ACT, 8
- administration, 58
- advisory messages, 25
- ANI, 70
- automated attendant, 50
- automatic
 - camp on, 47
 - hold, 38
- automatic message copy, 52

B

- buttons, 69

C

- call
 - completion, 26
 - park orbits, 69
 - park and page, 28
- call manager, 71
- call queuing, 52
- call record, 52
- call screening, 52
- caller ID, 70
- caller identification, 52
- capacities, 69
- class of service override, 42
- CNIS, 70
- CO lines
 - groups, 70
 - tandem, 32
- codec, 53
- conference
 - capacities, 70
 - on-hold, 32
 - ports, 70
 - split/join/drop, 32
- consultation hold, 39

D

- delayed ringing, 41

- destination restriction, 70
- DHCP, 16
- DND, 42
- DNIS, 70
- do not disturb
 - override, 42

E

- exclusive hold, 39
- executive override, 42
- external
 - page zones, 70

G

- G.711 and G.729A, 16
- Goldmine, 8

H

- hold
 - automatic, 38
 - consultation, 39
 - exclusive, 39
- hospitality mailbox, 54
- hunt groups, 70

I

- instant messaging, 8
- IP
 - user mobility, 14

L

- language selection, 54
- line
 - buttons, 69

M

- mailbox, 52
- media server, 71
- meeting, xi, 5
- message
 - count, 53
 - delete, 55

- waiting, 55
- message waiting, 40
- messaging, xi, xii, 3, 5, 6, 8, 50, 55, 56, 57, 59, 61, 71
 - automated attendant, 50
 - voice, 52
- Microsoft, 8
 - Outlook, 8
 - Windows '98, 8
 - Windows 2000, 8
 - Windows NT 4.0, 8
 - Windows Vista, 8
 - Windows XP, 8
- multiple call, 41
 - ring group, 70

O

- off-hook call announce (OCA), 70
- one touch buttons, 69
- override
 - class of service, 42
 - do not disturb, 42
 - executive, 42
 - privacy, 42

P

- paging
 - external zones, 70
 - max. simultaneous stations, 70
 - telephone group, 43
- park and page, 28
- peripherals, 15
- phantom directory numbers, 69
- pickup groups
 - capacities, 70
- primary directory numbers, 69
- privacy
 - override, 42
- privacy/non-privacy, 42

R

- reporting, 60
- ring over busy, 44
- ring tones, 66, 70

S

- security, 61
- SIP trunking, 69
- soft keys, 56
- speaker
 - OCA stations, 70
- speed dial, 45, 70
- split/join/drop, 32
- station

- buttons, 69
- dimensions, 65
- hunt groups, 70
- system
 - tones, 65

T

- tandem CO lines, 32
- telephone
 - group paging, 43
 - types, 15
- terminal authentication, 16
- time zone, 54

U

- unified messaging, 6, 7, 57

V

- verified account codes, 70
- voice messaging, 52