



General Description

Publication Information

Toshiba America Information Systems, Inc.
Telecommunication Systems Division

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IPe-GD-ALL-VK
Version K.1, August 2014
4010488

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IPedge General End User Information

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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. (Additional information is provided in the IPedge Install Manual.)



CP01, Issue 8, Part I Section 14.1

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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.

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Introduction

This General Description provides an overview of the *IPedge* business telephone systems, associated hardware, features, capabilities, and capacities. The features described in this document assume that the *IPedge* system has the current software release installed.

Organization

This document is divided into the following major topics:

- **Chapter 1 – Overview** is a brief introduction of the *IPedge* system, environmental and power considerations, related software, administration, configuration, and network requirements.
- **Chapter 2 – Telephones and Peripherals** describes the most recent Toshiba-proprietary stations and peripherals, customer-supplied peripherals, as well as cabling and connectors.
- **Chapter 3 – Unified Communications** describes the *IPedge* Messaging, Call Manager, Meeting, and Mobility Solutions which together form Toshiba's Unified Communications product suite.
- **Chapter 4 – Networking** describes the various network related configurations that need to be done when installing the *IPedge* system.
- **Chapter 5 – Contact Center** describes the ACD software that resides on the MAS, its basic capacities, system expansion, and remote maintenance.
- **Chapter 6 – 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

- IPedge Installation
- IPedge Feature Description and Implementation
- UCedge Client User Guide
- VIPedge General Description
- VIPedge Administration Manual

User Guides

- IPedge Telephone, Messaging, and Call Manager

Quick Reference Guide

- IPedge IP5000-Series Telephone

Internet Site

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

The IPedge[®] system is an advanced pure IP platform that provide sophisticated business communication features. The IPedge system performs call processing, voice mail, unified messaging, media processing that includes conferencing and paging, meet-me conferencing with web collaboration, centralized management, Call Manager unified communications, and more. IPedge uses Linux for the base operating system that provides a high level of scalability and security.

IPedge is a software centric Unified Communications platform designed for customers who want to deploy on an all IP network infrastructure to realize the savings of managing a single network.

Networking between IPedge servers and/or Strata CIX systems is enabled via IPedge Net using one IP address.

The IPedge solution is easy to install and is available in three different sizes depending on the number of users, applications, and conferencing needs. The EP server supports 8 to 40 users, the EC server supports applications up to 200 users and the EM server supports up to 1,000 users.

Available with the IPedge system is Toshiba's SIP Trunking I-VoIP service (SIP Trunking). Toshiba's integrated SIP Trunking service and equipment enable businesses to choose a single-vendor business telephone service and equipment solution.

IPedge can also connect to a MAS or MicroMAS via IP to run ACD and it can connect to uMobility and other external applications.



EC Server

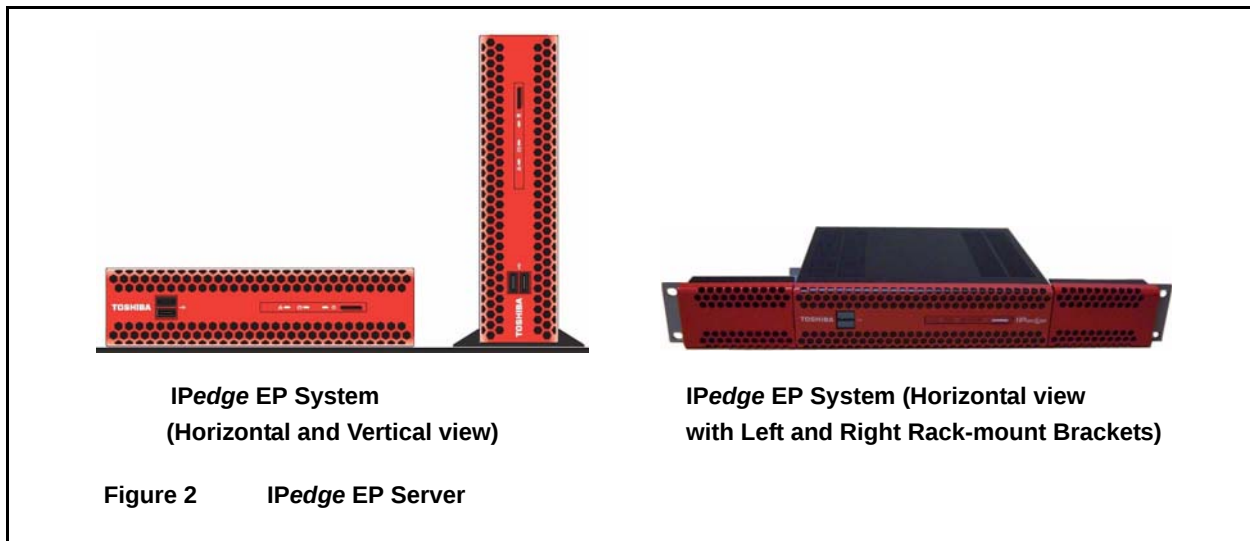


EM Server



Bezel (Optional)

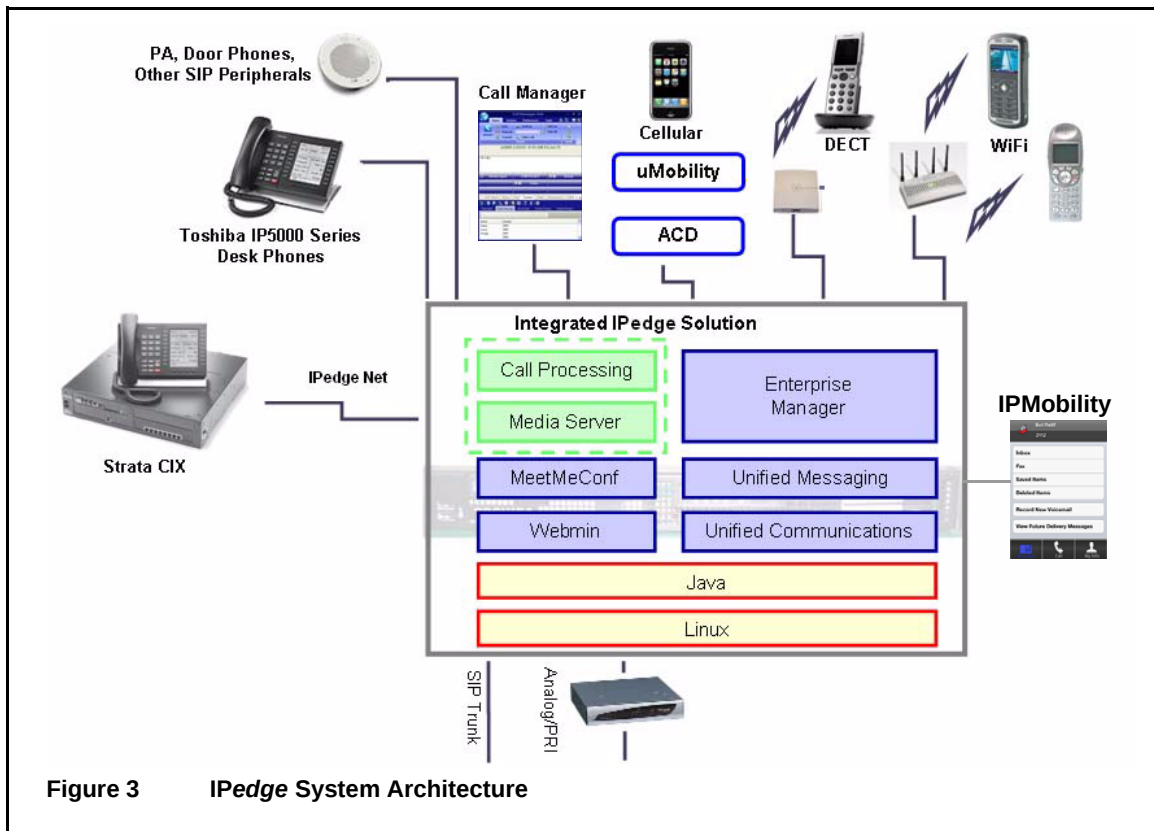
Figure 1 IPedge EC and EM Servers

**Table 1 Basic Specifications**

EC Server	EM Server	EP Server
Rackmount	Rackmount	Stand alone or 19" Rackmount
1U; 15" Deep; 19" Wide	1U; 25.6" Deep; 19" Wide	1.75U or 2.362" Height; 15" Deep; 8.12" Wide
1 x Core 2 Quad x 2.6GHz Processor, 4GB DRAM	2 x Quad Core x 2GHz Xeon Processors, 12GB DRAM	1 x Atom Dual Core x 1.80 GHz Processor, 4GB DRAM
1 x 250GB HDD (available RAID1 kit includes a second 250GB HDD)	2 x 300GB HDDs (RAID 1 standard) 4 x 300GB HDDs (RAID5 optional)	1 x 250GB HDD
Up to 200 Users	Up to 1,000 Users	8 to 40 Users

IPedge Solutions

The IPedge server integrates all the necessary customer centric applications as shown below. The IPedge reduces the need for multiple servers to support each application separately, therefore it dramatically decreases the cost and complexity of deploying multiple applications.



On a single server, IPedge provides the following:

- Call Processing – IPedge provides the basic and advanced call processing features with a single IP interface.
- Voice Mail / Unified Messaging – Voicemail is built in and can be configured as either a single centralized voicemail system for the entire enterprise or as a distributed voicemail system for each site.
- Unified Communications – Unified Communications is built in and provides Call Control from PC, Chat and Presence on the desktop (Call Manager)
- Meet-me Conference and Web Collaboration - Available on EC and EM systems only.
 - Having a built in conferencing and web collaboration eliminates costly monthly subscription fees. The integrated conferencing and web collaboration tool boasts an extensive list of features including the following all on a simple and easy-to-use GUI.
 - On Demand Conferencing
 - Scheduling One-time calls
 - Scheduling Recurring calls
 - Web-based Reporting
 - Telephony User Interface (TUI) for Moderator and Participants

- IPMobility application allows an iOS or Android client to make calls using the app which routes them through the host IPedge system, and without displaying the users mobile number to the called party.
- Centralized Management for multiple sites – The Enterprise Manager resides on the IPedge Server and enables an administrator to manage all trunks and stations in all the servers of the enterprise, using one consolidated view. From one central location, the administrator can backup and restore configurations of all sites, and update the firmware on any or all phones in the enterprise.

In addition to the above, the IPedge server can connect to a separate MAS or MicroMAS server to include:

- Strata ACD, Networked ACD (ACD + Unifier), and Call Center Reporting (TASKE or Insight)
- uMobility™

Other Advantages

The IPedge also provides the following benefits:

- Runs multiple communication applications built into one server platform
 - Call processing
 - Voice mail
 - Unified messaging
 - Meet-me conferencing and web collaboration
 - Call Manager unified communications with Presence, IM, call control from PC, CRM screen-pop integration, outbound dialing from any application, electronic document launch
 - Enterprise Manager web-based centralized system administration is integrated with browser access from your PC
 - Simplifies and integrates multiple forms of communications to optimize business processes
- Leverage server-based technologies
 - Low-profile chassis offers a sleek look and occupies minimum rack space
 - Standard Rack-Mount allows mounting on an existing standard 19 inch server rack
 - Survivability within or across the network providing business continuity when there is a hardware or network failure, and by allowing IP telephones to fail over to an active server at another location
 - Redundant Power Supplies and hard disk drives (RAID) on the EM model ensure business continuation after a single point of hardware failure
 - Expanded memory and Ethernet capacity to allow for multiple advanced applications
- LINUX Operating System
 - Provides a high level of scalability and security and is more resistant to virus attacks than common desktop operating systems. However, a secure network with proper monitoring capability is still recommended.
- Session Initiation Protocol (SIP)
 - Open interface to external devices and applications.
 - SIP based gateways are available to connect IPedge to analog and digital interfaces (FXS/FXO/T1/PRI). See [“Gateways” on page 22](#).

- Gateways are not required for SIP trunking.
- A variety of SIP endpoints are available for specific needs including wireless, smartphone, door phone, paging, etc. See [Chapter 2 – Telephones and Peripherals](#) for SIP approved endpoints.
- NAT Traversal allows VoIP calls to take place easily while each telephone device and the telephone system are all safely behind firewalls.
- Ease of installation and administration using Enterprise Manager
 - Administration software is built into the platform
 - Administration is accessed by web browser
 - Multi-user administration using Enterprise Manager Personal Administration (EMPA)
 - Centralized management of all locations, saving time, providing consistency, and eliminating potential mistakes.
 - One administration interface manages both IP PBX and voice mail, enabling new systems to be setup quickly with less labor and training required
- Strata CIX systems can be networked with IPedge systems via IPedge Net to either add a system to the network, or use the Strata CIX system as a gateway to IPedge for continued use of digital telephones and trunks connected to the Strata CIX.
 - Strata ACD running on a MAS or MicroMAS can connect to IPedge.

Operating Environment

The environmental requirements for the IPedge EC and EM systems are shown under [“Operating Environment.” on page 109.](#)

Software

The following software is included and installed on the IPedge server:

- Linux Operating System
- Java, Apache TomCat, PostgreSQL platform software
- IPedge Core (Call Processing, Media Server)
- Voicemail / Unified Messaging
- Net Server / Unified Communications (Call Manager)
- Meeting / Meet Me Conferencing / Web Collaboration
- Enterprise Manager / Web-based administration

Deployment

The administration software, Enterprise Manager is built into the platform which reduces installation time. The Enterprise Manager can be securely accessed from any PC with Microsoft™ Internet Explorer version 7 or later, or Mozilla® Firefox® version 5 or later. The administrator can view all the servers and all stations connected to each server in the enterprise in one consolidated view which aids in planning the numbering scheme.

A single IP Address is required for IPedge Net IP routing configuration. IPedge Net IP configuration is improved with the use of “Quick Access Guides” to all the programs required to configure IPedge Net IP. It guides the administrator step-by-step through the configuration. The Mozilla add-on ‘IE TAB’ is required to properly view some Enterprise Manager pages in Firefox.

Administration

IPedge Enterprise Manager provides a web interface for users to configure data, manage, control and maintain all components of these applications, and to coordinate the configuration of all IPedge Solutions in an Enterprise System. The system can be administered remotely over the Internet. No administration application is required on the user's PC.

In a networked multi-site system, all nodes are administered through a single location which provides centralized administration and database backup for the entire system.

Unified System Administration

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

- Enterprise Manager combines administration and management of the telephone system and voice mail into combined menus, allowing technicians and system administrators to program both together, and eliminate many duplicate steps.
- Several wizards support setup via integrated screens. A troubleshooting tool is also included.
- User Profile is designed to record equipment, user privileges, and authenticate the users during login. The User Profile allows organizing the equipment into domain groups of networked systems and presenting them in a tree. The user can select from a pull-down list of equipment they want to access for administration.
- Online Program Update makes updating to the latest version of software quick and easy.

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.

The Enterprise Manager application reduces the time required to manage multiple IPedge equipment sites. It can manage a group of IPedge systems in a single session. The possible situations when an administrator may need to configure multiple IPedge systems as a group include:

- Multiple IPedge systems in an IPedge Net configuration.
- Setting up several IPedge systems, at different sites, through the dealership LAN at the same time.
- Several sites, at different geographical locations, each with its own IPedge system, all managed centrally over a WAN.

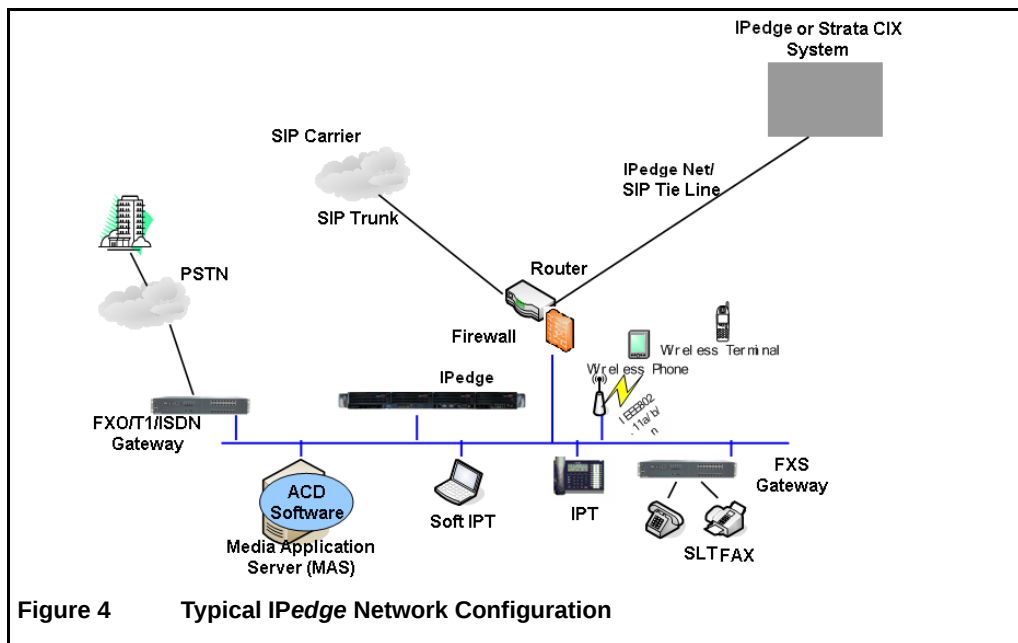
Toshiba's personal administration tool integrated in Enterprise Manager using a User login, puts telephone personalization in the hands of individual users, allowing them to easily program speed dial buttons, feature buttons, Do Not Disturb functions, and even the name on their LCD displays, using their 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.
- Enterprise Manager is also very useful to system administrators, who can administer changes for groups of users.

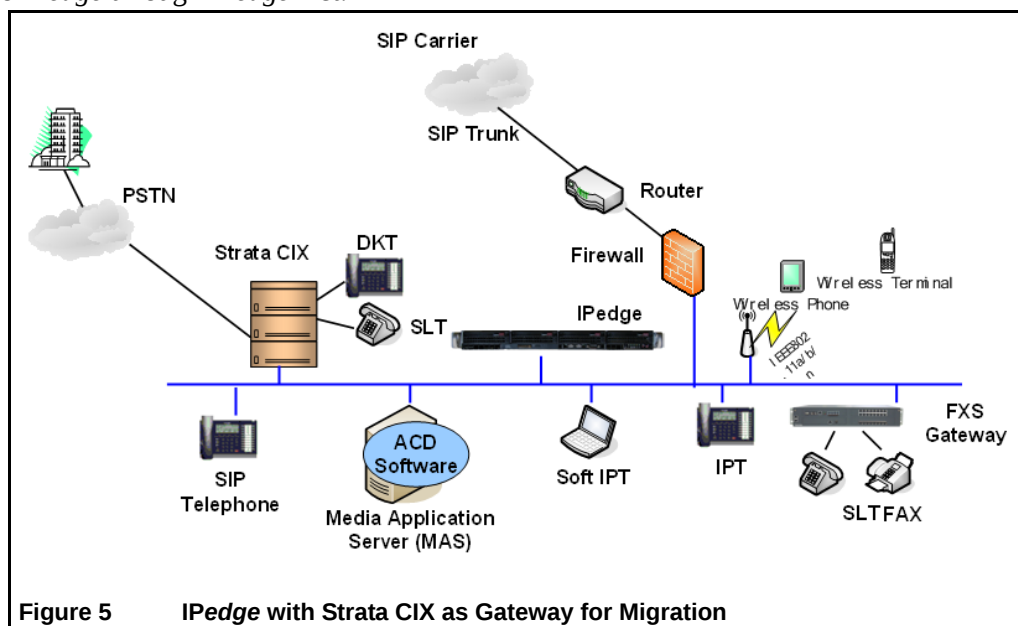
Configuration

In a typical network configuration with IPedge, the IPedge server is placed behind the NAT firewall and given a private IP address.

IPedge provides SIP trunking for incoming and outgoing calls. On the LAN, IPedge works with the SoftIPT and IP telephones. An FXO/T1/ISDN gateway is used to connect the all IP IPedge solution to PSTN networks.



Strata CIX system users can migrate to IPedge and minimize their investment by using their Strata CIX system as a gateway for PSTN interfaces and DKT telephones. The Strata CIX is networked to IPedge through IPedge Net.



Multi-node IPedge and Strata CIX systems can be networked together using IPedge Net.

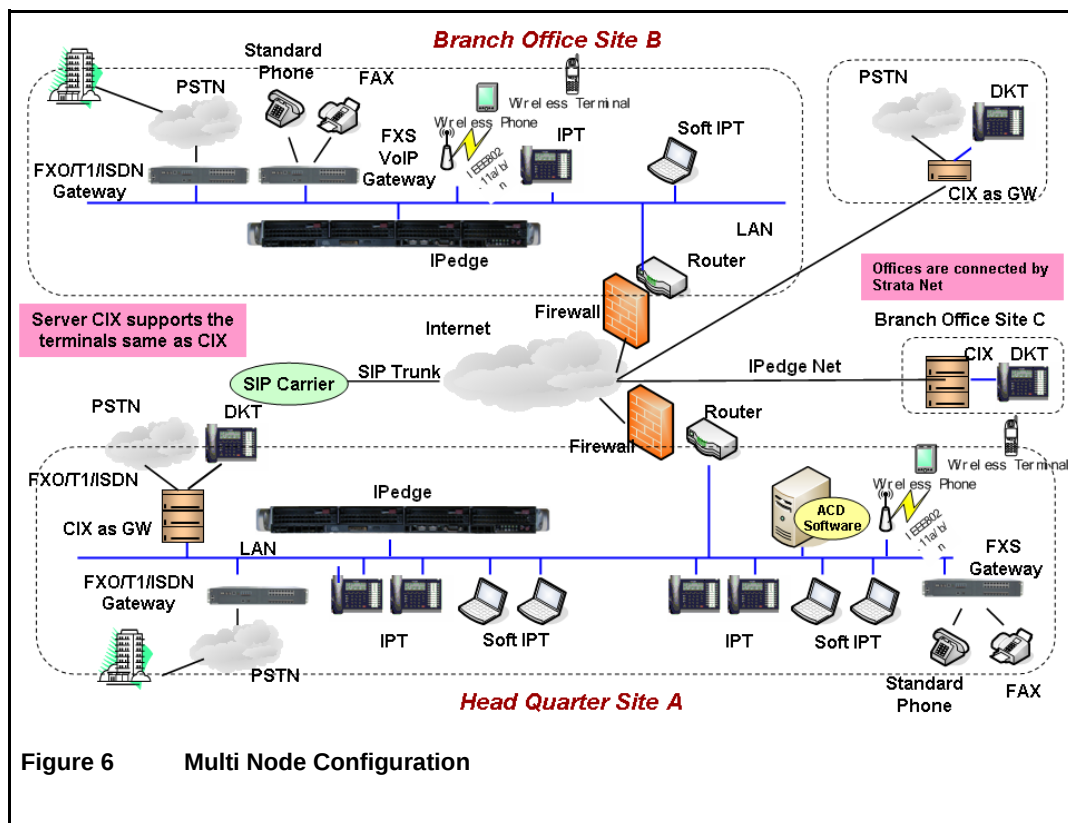


Figure 6 Multi Node Configuration

Note To network a Strata CIX system to an IPedge system via IPedge Net the Strata CIX system must use MIPU or GIPU interface cards. The LIPU and BIPU interface cards do not support IPedge Net operation.

IPedge Model Databases

Toshiba has developed “model databases” for the IPedge EC, EM, and EP systems. Model databases may be downloaded and installed using the Data Restore functionality from Webmin in Enterprise Manager. These programs are in addition to the current default values. The default database files are available for download from the [Toshiba FYI](#) website.

- IPedgeEPModel_mmddyyyy.zip
- IPedgeECModel_mmddyyyy.zip
- IPedgeEMModel_mmddyyyy.zip

For detailed installation and programming instructions on how to get the model database operational, refer to the IPedge Installation Manual located on the [Toshiba FYI](#) website.

Software Support and Upgrade Service

Toshiba's Software Support and Upgrade Service (SUS) plan for IPedge provides a great way to protect the investment in an IPedge system. It provides three important benefits: software updates, technical support, and license transfers.

Software Updates – While covered under this plan, software updates for enhancements, new features, and corrections may be applied to the IPedge system. Some new features may require additional licenses. Software updates are obtained through the Authorized Toshiba Dealer. If the SUS plan lapses, software updates may not be applied unless you pay additional charges to regain current status for maintenance.

Technical Support – Systems covered under the SUS plan are eligible for full technical support by the Authorized Toshiba Dealer. If the plan lapses, technical support is billed “per incident” and software updates are not available until the SUS plan is reinstated.

License Transfers – Systems covered under the SUS plan are eligible for license transfer when upgrading to larger systems. A small transfer fee and equivalent license price differences may apply. Licences are not eligible for transfer without a current SUS plan.

The first year of SUS is included with the purchase of a new IPedge system. At the time of purchase, the SUS plan term may be extended for up to 5 years with a discount for purchasing multiple years. Before the plan expires, the plan term again may be extended for 1 to 5 years, with multi-year discounts available. An IPedge system with a lapsed Software Support and Upgrade Service plan may reinstate the SUS plan for a fee of 125% of the annual rate over the lapsed period. The term begins after purchase, when licenses are generated for the IPedge system.

This Software Support and Upgrade Service plan is available to the original owner of the system and is not transferable. Registration and proof of purchase of the original owner of the IPedge system may be required.

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This chapter covers Toshiba's 5000-series Internet Protocol (IP) Telephones and peripherals that are compatible with *IPedge* telephone systems.

IP 5000-Series IP Telephones

The *IPedge* system supports the IP5000-series telephone product line.

Toshiba offers many IP Telephone models with backlight displays (except IP5022-SD), full-duplex speakerphones, and Gigabit Ethernet:

- **IP5122-SD & IP5622-SD** – 10-button IP speakerphone with 4-line x 24-character backlit LCD. The IP5622-SD does not support Gigabit Ethernet, ADM, DSS, BESCIB and carbon handset/headset.
- **IP5022-SD & IP5522-SD** – 10-button IP speakerphone with 4-line x 24-character LCD (without backlight)

The IP5022-SD & IP5522-SD have all the same characteristics as our IP5122-SD except the backlight for the display. These telephones still have full feature functionality with its 4-line display and 10 programmable buttons. However, the IP5022-SD supports Gigabit Ethernet, while the IP5522-SD does not support Gigabit Ethernet, ADM, DSS, BESCIB and carbon handset/headset.

- **IP5122-SDC** – Looks and functions similar to the IP5122-SD telephone when connected to the *IPedge* system. This telephone fully support all the *IPedge* features and services of a regular IP5122-SD telephone. However, this telephone can have a unique feature button called the Analog Central Office (ACO) button to connect directly to your local Central Office. This ACO feature enables you to make Emergency 911 calls and/or calls on your direct CO line by bypassing your *IPedge* system. Power over Ethernet (POE) or AC power is required for the telephone's analog local line connection to operate.
- **IP5132-SD** – 20-button IP speakerphone with 4-line x 24-character backlit LCD
- **IP5131-SDL, IP5531-SDL & IP5631-SDL** – 20-button IP speakerphone with 9-line x 24-character backlit LCD and HTML interface with navigation key. This telephone has 18 Soft Keys located on the sides of the large LCD to respond to system feature prompts, ten of these are programmable feature buttons. The Shift key (Purple key) toggles the LCD screen between flexible keys 1~10 and flexible keys 11~20. The IP5531-SDL and IP5631-SDL do not support Gigabit Ethernet and carbon handset/headset.

Note ACD and FeatureFlex application keys are not supported on programmable feature buttons 11~20 of the large 9-line display telephone.

IP5022-SD & IP5522-SD

10 programmable buttons, 4-line LCD

IP5122-SD, IP5122-SDC & IP5622-SD

10 programmable buttons, 4-line backlit LCD



IP5132-SD

20 programmable buttons, 4-line backlit LCD



IP5131-SDL, IP5531-SDL & IP5631-SDL

20 programmable buttons, large backlit LCD
with HTML support and navigation key



Features

The IP5000-series telephones include a speakerphone and are 802.3af standard compliant for Power-Over-Ethernet (PoE). The IP Telephony product family also includes matching Add-on Module and a DSS Console.

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

- Backlit Displays (except IP5022-SD)
- Gigabit Ethernet Switch (except IP5522-SD, IP5531-SDL, IP5622-SD and IP5631-SDL telephones)
- Analog Central Office (ACO) button (IP5122-SDC only)
- Busy Lamp Field (BLF) display of station status.
- Background Music through telephone speakers.
- Paging over telephone speakers.
- IPT Anywhere
- 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 (except IP5522-SD and 5622-SD)

In addition, the 5000-series IP telephones contain several important features, including:

- Full-duplex speakerphone capability.
- 802.3af power over Ethernet compliant
- Soft Keys to respond to the *IPedge* 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.

Connectivity

The IP telephones connect to the *IPedge* system via the LAN or WAN.

- These telephones do not use the System power supply, so there is no power restriction to limit the number of IP telephones that can connect to an individual *IPedge* system. The IP Telephones have built-in connectors. The back of the telephone has connector labels. The telephones can be powered by a local power supply or by PoE (Power over Ethernet).
- The RJ45 LAN jack connects the telephone to the network via the LAN cable supplied with the telephone or with an optional power brick that can be ordered per IP phone.. These telephones operate on the network at 10/100Mbps and 1000Mbps (IP5000-series with the exception of the IP5522-SD, IP5531-SDL, 5622-SD, 5631-SDL telephones) and can be connected to a fast switch hub, router, LAN, WAN, etc.
- The RJ45 PC jack can connect the IP telephones to the user's PC. The IP telephones can operate like a switch, as opposed to a hub, so the telephone can be connected directly to the LAN or Cable/DSL modem, and then a PC can be connected to the telephone PC jack to connect to the LAN through the telephone.
- The built-in headset jack enables headsets to be connected to the telephone. No optional headset interface is required.

Capabilities

The Toshiba IP Telephones also have the following capabilities:

- The IP telephones contain two types of codecs (coder/decoder): G.711 and G.729A. The codec determines the IP telephone voice quality and network bandwidth requirements. The G.711 requires the most bandwidth and provides the best voice quality. The G.729A requires less bandwidth, but it does not provide the best voice quality. The desired codec can be selected for each IP telephone in IP station administration using Enterprise Manager.
- The external ringer interface connector is mounted inside the telephone base. This enables connection of an BESC external speaker device to provide a loud ringer for the IP telephone.
- All telephones in the system can be IP telephones up to the system's maximum station capacity (see [Table 16 on page 118](#)).
- Terminal Authentication is an option that allows a particular IP telephone to keep a reserved directory number on a IPedge system. This prevents IP telephones from logging in with another telephone's directory number if the other telephone has been disconnected to be taken to another location. This feature uses the unique Media Access Control (MAC) address that is permanently coded into each telephone network interface circuit. The unique MAC address of the telephone is assigned to a particular directory number in system programming.
- IP telephone firmware can be updated locally or remotely using Enterprise Manager. This enables service personnel to update IP equipment with new features and enhancements as they become available. Updates require a brief interruption of IP telephone operation (a few minutes).
- IP telephones have a discovery retry timer to prevent network congestion when many IP telephones request services simultaneously. If network congestion is detected, the telephone will pause and then retry for service.
- IP telephones have loop back and ping capabilities for maintenance and fault finding purposes.
- IP telephones support Dynamic Host Configuration Protocol (DHCP) or static IP addressing.

Liquid Crystal Display (LCD) Models

The IP5022-SD, IP5122-SD, IP5522-SD, IP5622-SD, IP5122-SDC and IP5132-SD models display up to 24 characters times four lines of information and provide four Soft Keys.

The IP5131-SDL, IP5531-SDL, IP5631-SDL has 4 soft keys and a 9-line LCD. From the idle screen you can access telephone directories and speed dial lists of names or departments, internal or external to the telephone system. You can search by name or letter within a list.

The IP5131-SDL screen consists of Phone, Config, and Web-application screens. The screens change easily by pressing the tab soft key.

- Phone screen – This screen is a 150 x 168 pixel LCD screen.
- Config screen – This screen can be used for Telephone configuration settings. Using this screen, you can set the IP address, subnet mask, etc.
- Web screen – provides users with the access to the web page. This capability can be used to create the custom applications designed to match with IP5131-SDL phone screen so that the user can enter or retrieve the data from the IP5131-SDL phone.

All LCD telephone models can provide:

- Advisory Messages
- Automatic Number Identification (ANI)
- Caller ID, Name and Number with call history
- Contrast adjustment (13 levels)
- Backlight adjustment, except IP5022-SD (On/Off/Synchronized)
- Date/Time of Day
- Dialed Number Identification Service (DNIS Name and Number)
- Feature Prompting Soft Keys that are used as an alternative to access codes or feature buttons. Station users can access features by responding to LCD prompts.
- Called Number displays on outgoing calls (1~120 secs.)

IP4100 DECT Telephone

The IP4100 DECT telephone (shown right) supports 8~10 simultaneous call sessions per base and allows for seamless roaming between bases in a multi-base configuration. The High Definition voice enabled speaker and microphone allows for crystal clear speech. The handsets support more than 20 hours talk time and 200 hours standby time in normal operations. The IP4100 DECT telephones are administered via a powerful web-based management tool that simplifies administration and installation.



IP4100 DECT Telephone and Base

Handset Features

- 2.0 inch color TFT back-lit display
- Li-ion battery for long talk time and standby time
- Wideband Voice (HDSP) Basic and Extended operations
- Over-air feature upgrade support
- DECT 6.0 frequency support with encryption
- Back-lit keypad for easy operation in low-lit environments
- Laser-etched keypad numbers for extended life use
- Three soft keys for ease of operation
- Four programmable feature keys for flexibility in operations
- 2.5 mm headset jack
- Polyphonic ringer support
- Vibration ring support
- Additional battery charging station in handset charger
- High-quality speaker phone
- Belt clip

IP4100-BASE Features

- Attractive design
- Power over Ethernet support
- Omni-directional internal antenna for flexible installation options
- Antenna diversity switching to avoid interference
- Frequency support 1920-1930mhz with Wi-Fi avoidance technology
- Colored LED status indication for ease of installation and operation
- Easy installation via web GUI interface
- Built-in trouble-shooting tools.

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 (except the IP5522-SD and IP5622-SD telephones), these buttons can be assigned as CO line, Directory Number, DSS, One Touch, Speed Dial or any other flexible feature.

The LM5110 supports backlight and LCD labels, it can be connected to any IP5000-series telephone.



Key Module (KM5020)

The KM5020 adds 20 programmable feature buttons to the 5000-series telephones. These buttons can be assigned as CO line, Directory Number, DSS, One Touch, Speed Dial or any other flexible feature.



IP Direct Station Selection (IDM5060) Console

The IDM5060 console (shown right) is for system attendants.

The IDM5060 operates alongside an IP telephone and has 60 programmable feature buttons. These buttons can be assigned as CO Line, Directory Number, DSS, One Touch, Speed Dial or any other flexible feature.

Up to three consoles can operate with one IP telephone. However, the IP5522-SD and IP5622-SD telephones do not support an IDM.

The IDM5060 console uses LEDs to indicate call and feature status; the DSS has dual red and green LEDs to help further define status, such as station in DND status. The IDM5060 console connects directly to an IP5000-series station.



Attendant Console

The Attendant Console runs on a PC with Microsoft® Windows XP Professional or Windows 7 (32 bit) operating system. The Attendant Console PC offered by Toshiba is equipped with an Intel two gigahertz CPU in a small, compact desktop chassis that is just the right size for a receptionist's desk. If there isn't any room on the desk for the system to lay flat, it can also be stood on its side for an even smaller footprint. Add to that the powerful IPedge Attendant Console software, and you have a winning solution for any IPedge installation!

The Console has a built-in softphone for IPedge and uses a Computer Supported Telephony Application (CSTA) interface.

Each Attendant console requires the I-CP-AUX, the basic port license, and the I-CP-ATT license which is bundled with the Toshiba supplied Attendant Console PCs and software. The I-CP-ATT license must be ordered separately to enable the Console to be installed on a Dealer / Customer supplied PC. The latest Attendant Console software is available at no charge on Toshiba FYI.

The minimum PC requirements for the IPedge Attendant Console are:

- Operating Software – Microsoft Windows XP Professional or Windows 7 (32 bit)
- Processor – 2.0GHz Intel Pentium 4, Celeron or higher
- Memory – 1GB RAM
- CD-ROM Drive, CD-R/W Drive or DVD capable of reading CD-ROM data files to install Attendant Console software.
- Hard Drive: 500M space available
- Sound card with internal or external speakers
- I-CP-ATT must be ordered separately if using a Dealer supplied PC for the Attendant Console

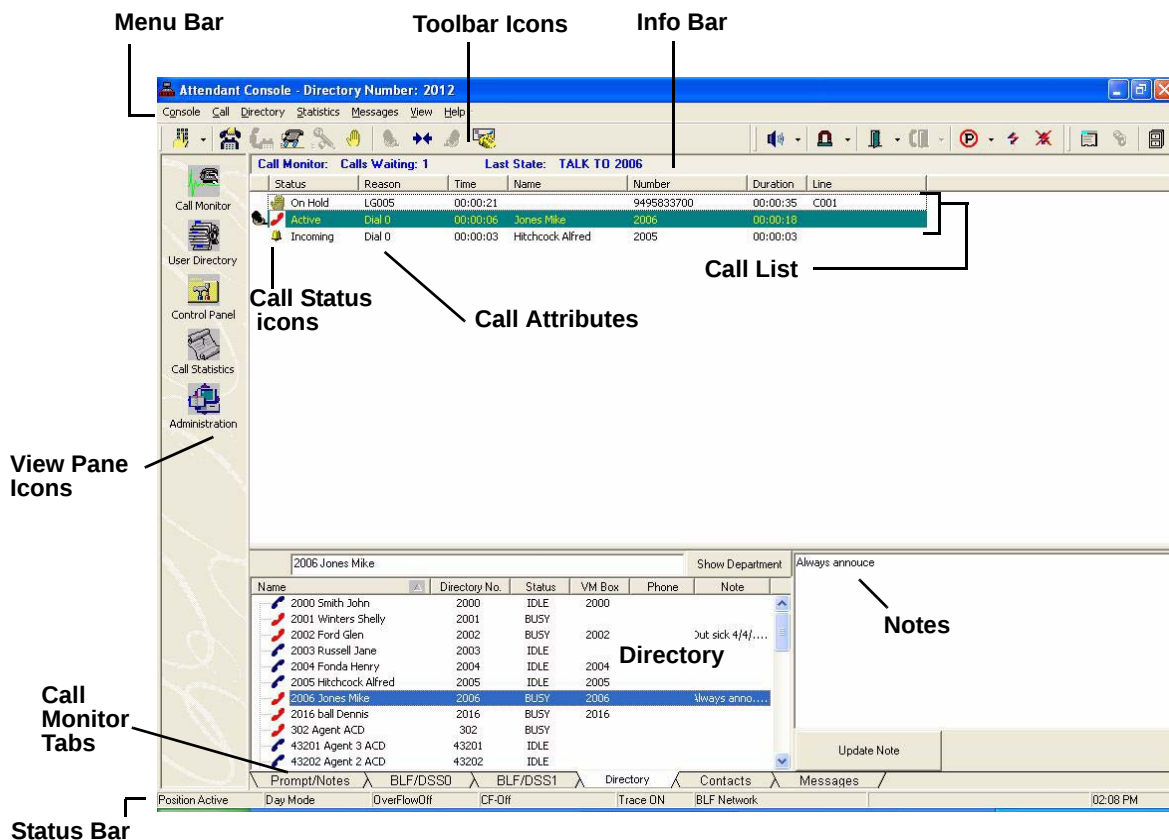
Toshiba offers IPATTCONS, a complete attendant console system with the following items:

- Attendant Console License (I-CP-ATT) for the IPedge system
- Intel Pentium D Dual Core 1.8GHz CPU
- 1GB Random Access Memory (RAM)
- DVD R/W drive
- Windows XP Professional (factory installed)
- IPedge Attendant Console software (factory installed)
- Comprehensive set of multimedia inputs and outputs
- Keyboard & Mouse
- PC Headset (Jabra 2000)
- Attendant Keyboard stickers (Black) CIX-BL-ATCON-VA

Optional item:

- 17" Flat screen Monitor – Toshiba offers a 17" Liquid Crystal Display (LCD) flat screen monitor; part number CTX-LCD-MONITOR.

The Attendant Console is designed to handle all call activity within a single Call Monitor screen, shown below. All calls will appear in a single list.



Calls are marked with icons to show the current status.

Features such as Paging, Call Pickup, Call Park offer many alternatives. The Administration window enables which option is the primary operation for that Attendant. For example, if two zones are used for paging, as well as the All Call, then an option pull down arrow is next to the icon. Clicking the icon starts the All Call Page, then the Attendant can select one of two page zones.

All other views available for the Attendant are for administrative and management use. They do not control any type of call handling except how chosen options affect the overall operation.

The Attendant Console also enables an attendant to manage console settings, maintain a user directory, and view call statistics. The Console provides a Name/Number search that works with automatic or manual call handling. Other features include Queuing, DSS, signaling, Emergency Call ID, keyboard or mouse operation, and headset or handset operation with volume control.

Peripherals

The *IPedge* supports a variety of third party peripherals in order to meet specific business needs.

Polycom End Points

The *IPedge* call control platform supports a variety of end points from Polycom. The following table lists end points (by type – desk phones, wireless, conference etc) that have been qualified to work with *IPedge*.

Table 2 Polycom End Points

Type	IPedge - End Points Supported	Image
Desk Phones	SoundPoint IP 321 SoundPoint IP 331 SoundPoint IP 450 SoundPoint IP 550 SoundPoint IP 560 SoundPoint IP 650 SoundPoint IP 670	
VoWLAN Phones	SpectraLink 8440 SpectraLink 8450	
WiFi Wireless Phones	SpectraLink 6020 (connects via a gateway) SpectraLink 8002 SpectraLink 8020 SpectraLink 8030 SpectraLink e340, h340, i640	
KIRK DECT Wireless Phones	KIRK 2010 KIRK 4020 KIRK 4040 KIRK 5020 KIRK 5040 Note: These digital telephones use Polycom gateways that connect to the <i>IPedge</i> system.	

Table 2 Polycom End Points *(continued)*

Type	IPedge - End Points Supported	Image
Conference Phones	SoundStation IP 7000 SoundStation IP 6000	

Note The list of supported end points changes as Polycom brings new products to market. For the most recent versions supported Polycom end points, Data sheets, etc, refer to Internet FYI > TSD 3rd Party Partners > Polycom page.

CyberData

The CyberData® Voice-over-IP (VoIP) Intercom product is targeted for door phone, emergency phone, and access control applications.

The CyberData VoIP Intercom product operates over a Local Area Network (LAN) and is Power-over-Ethernet (PoE) 802.3af enabled. SIP line extension dialing and features allow the user to use the IPedge capabilities for two-way communications or as a paging speaker as required.

The VoIP Intercom provides adaptive full-duplex voice operation, along with network adjustable speaker volume and microphone sensitivity. The product is easily configurable via a network web management utility.

Providing door closure and tamper alert signaling, the VoIP Intercom has a tamper-resistant design that allows the unit to be mounted securely and safely. Network downloadable firmware support allows for easy upgradeability.



The CyberData VoIP Intercom provides:

- SIP RFC 3261 compliance
- LAN Ethernet 10/100 Mbps
- G.711 (A-law and μ -law) codec support
- Single press call button
- Call activity indicator (light)
- POE 802.3af compliance
- One dry contact relay for auxiliary control
- Tamper-resistant design
- Network web management

Gateways

A VoIP gateway is hardware equipment that converts Time Division Multiplexing (TDM) telephony traffic from the Publicly Switched Telephone Network (PSTN) into digital IP packets for transport over an IP network. A VoIP gateway can also convert digital IP packets into TDM telephony traffic for transport across the PSTN. VoIP gateways have become a central, yet complex, component in most state-of-the-art VoIP systems. Other functions of a VoIP gateway include voice compression or decompression, control signaling, call routing, and packetization. VoIP gateways come in many different configurations. Toshiba sells some third party gateways.

Audiocodes

Mediant 1000 Series (Digital)

The Mediant 1000 (shown right) is Audiocodes' cost-effective, converged wireline VoIP media gateway. Intelligently packaged in a stackable 1U chassis, it is designed to interface between TDM & IP networks in enterprises or small-scale carrier locations. Incorporating AudioCodes' innovative Voice over Packet technology, the Mediant 1000 enables rapid time-to-market and reliable cost-effective deployment of next-generation networks.



The Mediant 1000 is based on VoIPerfect, AudioCodes' underlying, best-of-breed, media gateway core technology for all of its products. The Mediant 1000 provides superior voice-technology for connecting legacy telephone and PBX systems to IP networks, as well as seamless connection of the IP-PBX to the PSTN. The Mediant 1000 is fully interoperable with multiple vendor gateways, Softswitches, gatekeepers, proxy servers, IP phones, Session Border Controllers and firewalls.

The Mediant 1000 matches the density requirements for small locations while meeting enterprises' and service providers' demands for scalability. The compact Mediant 1000 Modular Gateway is extremely scalable and supports multiples of 1, 2, or 4 E1/T1/J1 spans, 4 to 20 BRI ports or 1 to 24 analog ports in various FXO/FXS configurations. The Mediant 1000 also supports mixed digital/analog with media processing capabilities such as conferencing, play/record configurations.

The Mediant 1000 can support a variety of telephony interfaces. The digital module can be configured as regular E1/T1/J1 interfaces, with up to 1 or 2 paired spans acting as life-line interfaces for switching to the PSTN in case of power failure or network problems. The analog module is available as regular FXS or FXO interfaces, where 1 FXS line can be used as a life-line interface for switching to the PSTN.

Interface Modules:

- Digital (E1/T1/J1) – connecting the PSTN or PBX to the IP-network
- Analog FXS – connecting analog phones to the IP-network
- Analog FXO – connecting analog trunks to a CO line

For Mediant 1000 series parts, refer to Audiocodes Parts, [Table 28 on page 126](#).

Mediapack Series (Analog)

The MediaPack™ Series Analog VoIP Gateways are cost-effective, best-of-breed technology products. These stand-alone analog VoIP Gateways provide superior voice technology for connecting legacy telephones, and PBX systems with IP-based telephony networks, as well as for integration with new IP-based PBX systems.

MediaPacks are well suited for commercial VoIP deployment because of their mature and field-proven voice technology. Their rich feature set allows integration with a wide range of Carriers and Enterprise network applications. MediaPack gateways are used by Carriers and Service Providers in Access networks for connecting Multi-Tenant Units (MTU), IP Centrex subscribers, pay phones and rural users over various wireless and satellite links.

Enterprises use MediaPack gateways to connect their legacy PBX systems over an IP infrastructure. In addition, in IP Centrex and central IP-PBX applications, the MediaPack increases the remote location availability and provides Stand Alone Survivability (SAS) when there is no IP connection between branch locations and the central SIP servers, SIP Proxy or central IP-PBX.

MediaPacks are third generation products that have been designed to meet real market needs. In addition to superior voice technology, the products provide advanced telephony features such as long-haul, metering tones generation, country dependent MWI and CID for true integration with the existing telephony infrastructure. A variety of management and provisioning tools, such as AudioCodes' EMS, embedded web server, Telnet and SNMP enable fast deployment and management of large and complex networks. MediaPacks are based on VolPerfect™ architecture, AudioCodes underlying, best-of-breed, core media gateway technology for all of its products.

MediaPacks are part of AudioCodes' complete family of stand-alone VoIP Gateways for OEM system integration. Support of multiple VoIP control protocols has been tested with leading Softswitch vendors. As a provider for OEMs, System Integrators and Network Equipment Providers, AudioCodes offers short time-to-market with field-proven products.

MediaPack Series Features

- Spans a range of 2 to 24 analog ports
- Supports PSTN/PBX analog telephone sets or analog trunk lines (FXS/FXO)
- Selectable, multiple LBR coders per channel
- T.38 compliant – Rich subscriber Feature Set including; 3-Way conference with local mixing, call pickup, hunt groups, call forwarding, call hold, call transfer
- Echo cancelation, Jitter Buffer, VAD and CNG
- Enhanced capabilities which include MWI, long-haul, Metering Tones, STUN, Security features, Generation, CID and outdoor protection
- Stand Alone Survivability (SAS) for SIP based IP Centrex and Central IP-PBX applications
- Web Management for easy configuration and installation
- EMS for comprehensive management operations (FCAPS)
- Automatic provisioning via TFTP/HTTP – Internal Access List firewall for network traffic filtering

For Mediapack series part numbers, refer to Audiocodes Parts, [Table 28 on page 126](#).

Epygi Gateways

The Epygi Gateways (shown right) are available in a sturdy metal rack-mountable housing that permits the inclusion of a built-in power source and cooling fan for heavy duty operation and extended life span.



These gateways include call routing and auto attendant capabilities, voice prioritization over data and sophisticated firewall and security elements. The QuadroM functions primarily to bridge traditional PBX traffic to the Internet, reducing costs and simplifying system administration.

QuadroM E1/T1 Single-RM Gateway – This model features 30 channels of compressed g.729 codec and conforms to a broad variety of voice codecs and signaling protocols. Architecture and design enhancements improve sound quality and application integration.

QuadroM Dual E1/T1 – This model doubles the call capacity of the QuadroM E1/T1, affording as many as 60 simultaneous calls (48 simultaneous calls for T1– US and Japan) from the two broadband trunks.

The QuadroM Dual E1/T1 30 features 30 channels of g.729 compression codec.

The QuadroM Dual E1/T1 60 allows for 60 channels.

Sound quality and application integration match the quality of the QuadroM E1/T1 with single port.

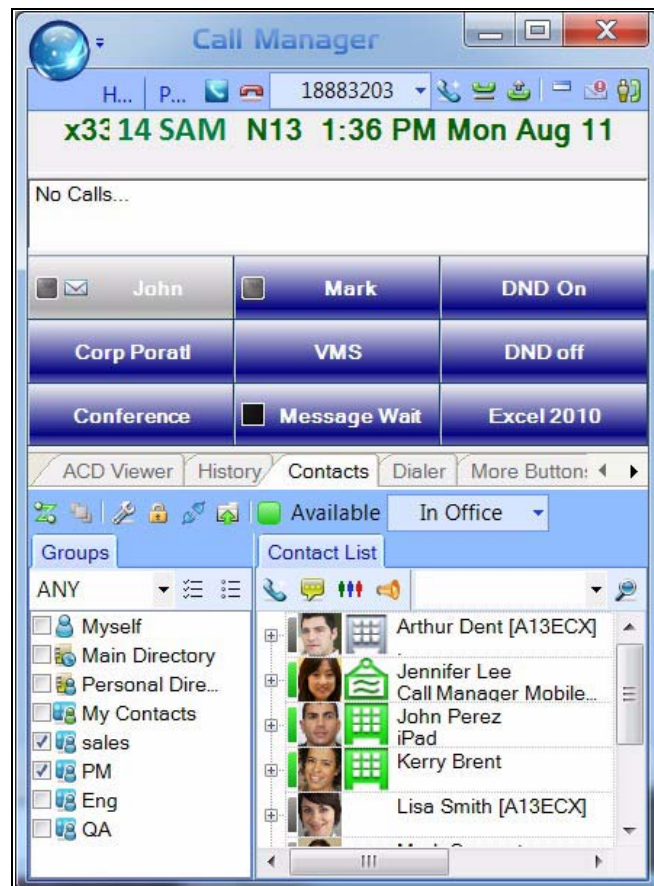
Table 3 Epygi Gateway Comparison Table

	E1/T1 Single-RM	E1/T1 Dual 30-RM	E1/T1 Dual 60-RM
Part Number	SKU-QM-0002	SKU-QM-0003	SKU-QM-0004
Ports for analog telephones or devices	1	1	1
Single/Dual E1/T1 ports	Single: 30 simultaneous calls (E1); 24 (T1)	Dual: 48 simultaneous calls (E1), 48 (T1)	Dual: 60 simultaneous calls (E1), 48 (T1)
E1 / T1 Ports	1	2	2
Ethernet LAN Ports	1	1	1
Ethernet WAN Ports	1	1	1

This chapter describes IPedge Messaging, Call Manager, Meeting, UCedge client, and Mobility Solutions which together form Toshiba's Unified Communications product suite. The IPedge system 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:

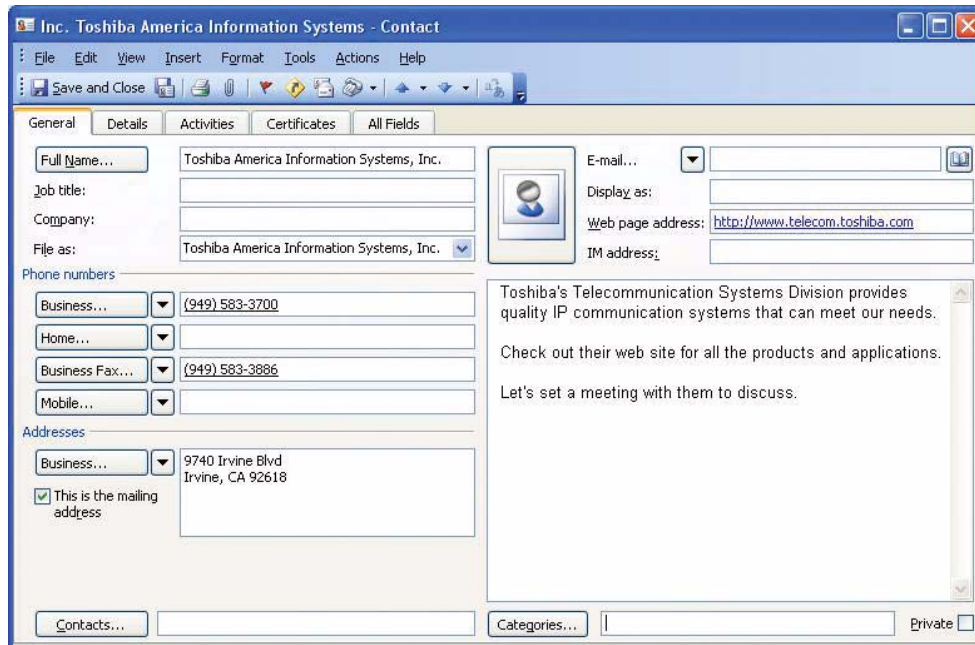
- **Presence and Instant Messaging** – With UCedge client and Call Manager you can see the status of other users, both their telephone busy/idle status and calendar status from Outlook integration, with the ability to click on the name to either call or instant message chat with them. You can decide 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 SCM 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



Call Manager Presence Screen

dialing, and other functions faster and easier. SCM 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.

- **Visual Voice Mail** – With UCedge you can use your mobile device, desktop or laptop to retrieve voice mail messages from the office phone system. See caller information and the time the message was left. Play back or forward the message. Touch to call back if the caller ID is available.
- **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 customer relationship management applications and databases. This saves you time and serves your customers better.



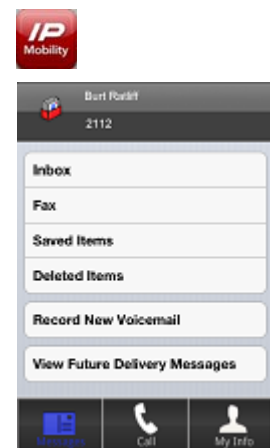
- **Off-premise Call Forwarding** – Enables your incoming calls to reach you when you're out of the office and enables you to change your forwarding destination from any remote location. You can stay in touch no matter where you are.
- **IPMobility for Android and iOS** – The IPMobility Application for Android™ and Apple iPhone™ allows a mobile device to act as an extension of the IPedge system by providing incoming and outgoing call features.

Users may also easily access key voice messaging functionality and manage administration of their voice mailbox without dialing into the voice mail system. IPMobility does not interfere with the ability to make a phone call or access the voice mail of the mobile device itself.

The IPMobility Application is available for both Android and Apple mobile platforms. See [“Mobile Device Support for IPMobility” on Page 130](#) for more details.

For incoming calls, the IPMobility app uses the host IPedge system's “Follow Me” feature to route the call to the users mobile phone.

For outgoing calls, the app uses IPedge phone services to callback the mobile phone, then call the preset destination and then connect the two calls. This process:



- Takes advantage of the host system's telephone service rates
 - Sends the Caller ID of the users office phone number - not the users mobile device number
- Users may also easily access voice messaging features and manage their voice mail without having to dial in to the system.

IPMobility users will incur per-minute usage on their cellular/wireless plan.

- **Fixed Mobile Convergence** – Toshiba's uMobility solution empowers mobile workers to make and answer their IPedge system calls from virtually anywhere. The user's smart cell phone functions as their IPedge system extension phone both while in the office via the wireless LAN and while out of the office via a cellular network. This is the most advanced level of mobility and insures the most expeditious handling of your incoming calls.

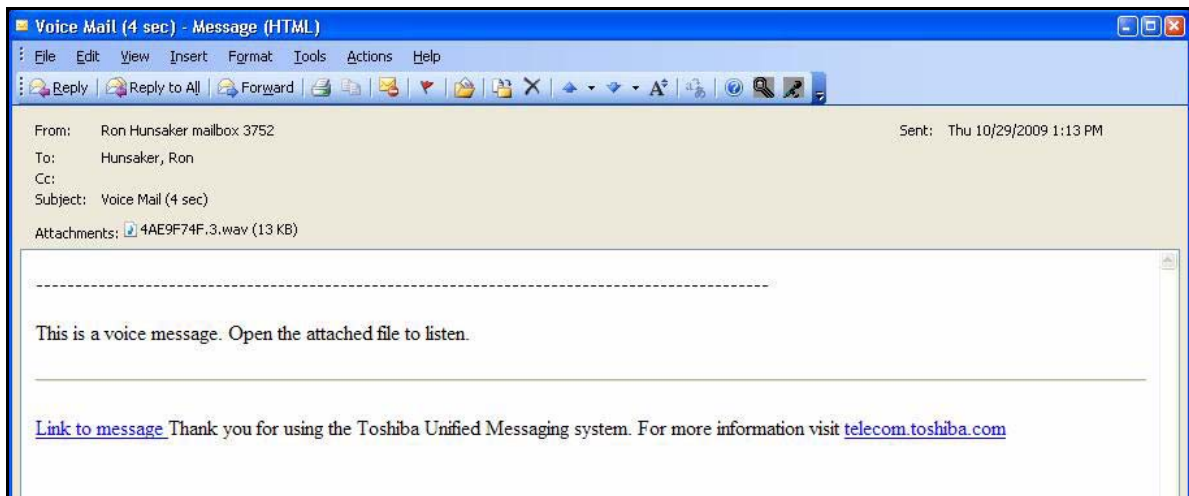


Android



BlackBerry

- **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.

IPedge Messaging

Messaging is an integrated voice processing application within the IPedge system 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.

Since Messaging is incorporated into the IPedge system, it delivers streamlined user administration and system management. With it, 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 through IPedge Net Server 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; Fax-to-Email, Print Emails to Fax, Redirect Fax Messages, 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 94](#).

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, Windows 7, or Windows 8 operating systems.

There are two levels of *IPedge* Call Manager:

- Call Manager Standard version is free to all users of the *IPedge* system. The license (I-CM-STD1) for Call Manager Standard is included in the user license bundle at no additional charge.
- Call Manager Advanced version provides enhanced functionality, including full Unified Communications (UC). Purchase Call Manager Advanced license (I-CM-1) when full UC capabilities are required.

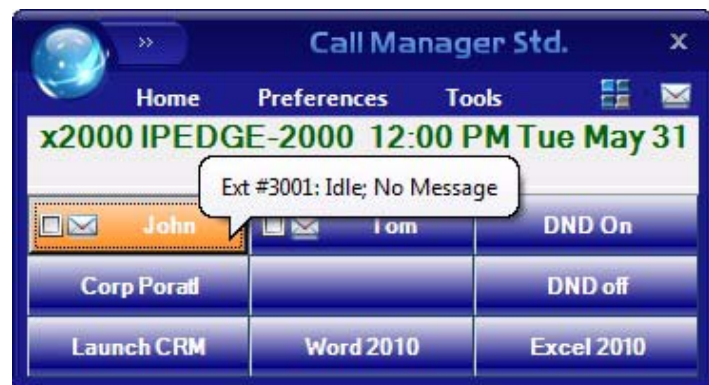
ACD customers and/or Network Call Manager customers must purchase the full featured Call Manager Advanced as the Call Manager Standard is not supported under this environment.

Call Manager clients connect to the Net Server running on the *IPedge* system with the appropriate license (I-CM-1/I-CM-V1) on *IPedge*. When Call Manager is used with ACD or Unifier, it must connect to the external server with ACD or Unifier with the appropriate license (LICMAS-NETSEAT) for the server.

Call Manager Standard

Call Manager Standard provides the following major functions:

- Call control support (dial, answer, transfer, with drag and drop operation)
- 9 configurable buttons for any of the following features:
 - DSS/BLF (status display)
 - Feature access code
 - Run Program
 - Speed Dial
 - System/PBX Command
 - Web URL
- LCD View
- Dialing from Microsoft Outlook Contacts



Call Manager Advanced

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.

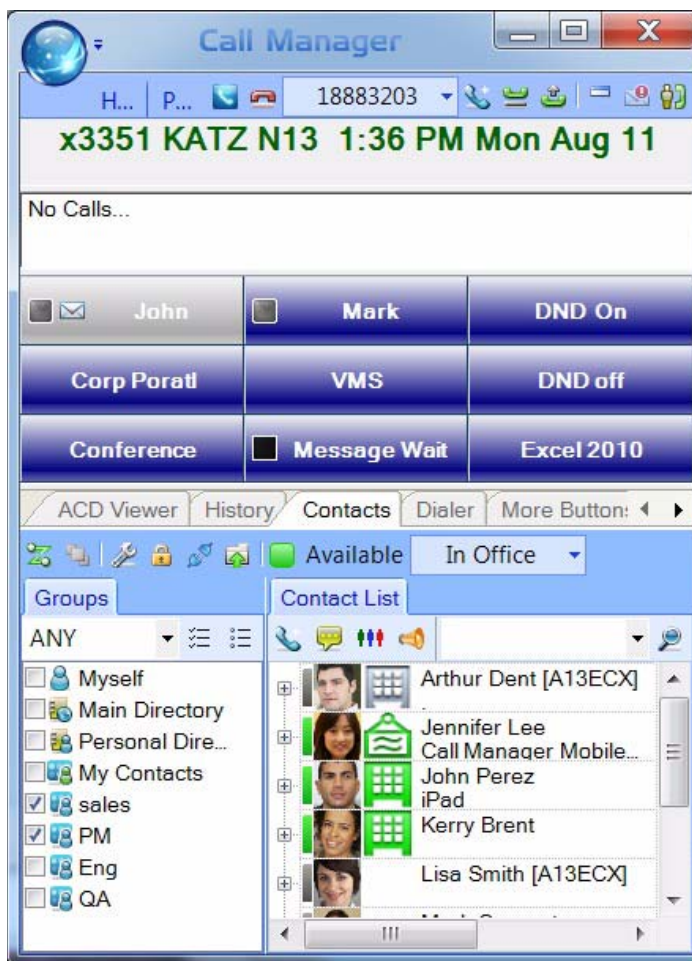


Figure 7 Call Manager Main Page

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

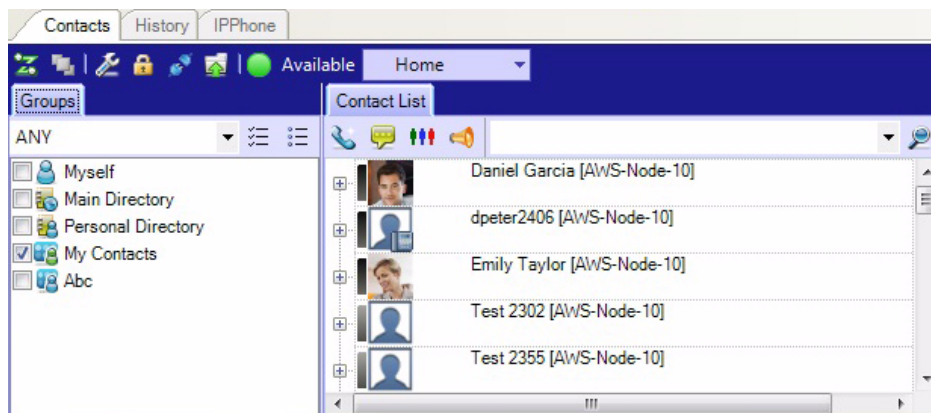
Microsoft's fluent user interface breaks the ribbon GUI down into multiple tabs. The tabs are broken down into groups. The ribbon groups all the common features and functionality together. Each tab has a specific function and all the buttons in that tab support that specific function. For example, the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

Call Manager Features

The Call Manager application supports some powerful features.

Contacts (Directory, Presence Viewer, Instant Message, and Speed Dial)

The Contacts tab (shown below) performs several features: Directory, Instant Messaging, Presence and Speed Dial. The Contacts provide a powerful set of directory features that allow you to look up and dial IPedge system extensions with a click of the mouse.



- **Directory** – 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.
- **Presence and Speed Dial** – 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.
- **Instant Messaging / Chat** – The Call Manager Chat application enables Call Manager users connected to the IPedge system 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.

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.

ACD Viewer

The Call Manager is tightly integrated with the ACD from Toshiba. The Call Manager ACD Viewer enables users connected to ACD to view the status of all ACD groups in which they belong. This additional functionality does not require MIS software to be installed. Call Manager shows the operating status of each group. Each group view can be expanded to see the number of calls and the status of each of the agents and supervisors in the group. Each group contains a “My Status” icon showing your own status in the group (logged in, logged out, busy, in wrap-up, etc), and the status can be changed by right-clicking on the icon.

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

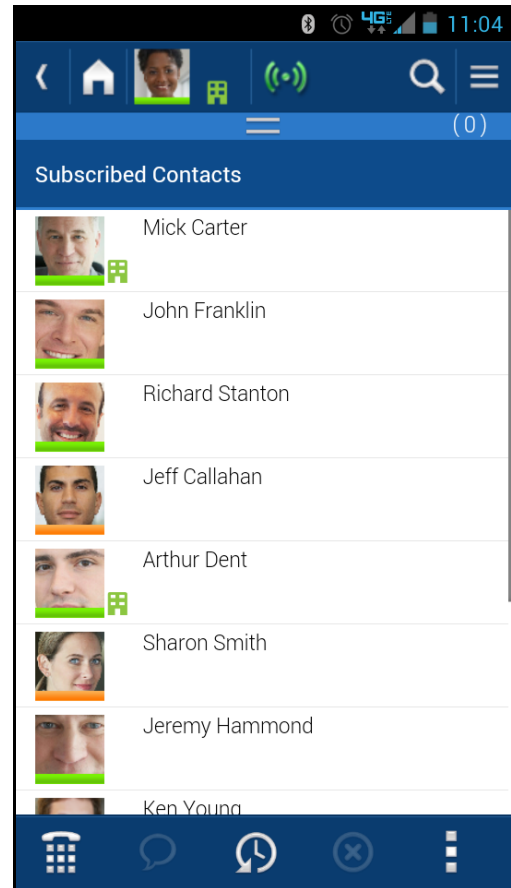
Call Manager has many different User Programmable buttons on the main screen and side window. These buttons can be easily configured by the user as DSS buttons, Feature buttons, Speed Dial buttons, User Action buttons, ACD buttons, etc. This provides the user with one-touch access to features, applications,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

UCedge Client

The UCedge Client is a UC solution for users of the IPedge system and VIPedge Application Services. The UCedge Client is a productivity tool that is integrated with the IPedge business telephone system.

The following are the benefits of having the UCedge Client:

- **Pairing** – Initiate dialing on your desk phone from your tablet or smartphone.
- **Softphone** – Make and receive calls on your Android device.
- **Call Thru/Call Back** – 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.
- **Follow Me** – Receive calls when you are away from your desk at the same extension and get things done sooner.
- **Call Thru or Call Back** – Make International Calls from your cell phone at land line rates and save money.
- **Call Back** – Be available on your business number with a local SIM or phone when traveling internationally and save money on cellular roaming rates.
- **Presence** – See the status of your VIPedge/IPedge colleagues before you call them and save time.
- **IM and Group IM** – Instant Message your VIPedge/IPedge/CIX colleagues who are on the phone or busy and get more done.
- **Contacts with Avatars** – Quickly find and call your VIPedge/IPedge/CIX colleagues without having to remember their internal extension.

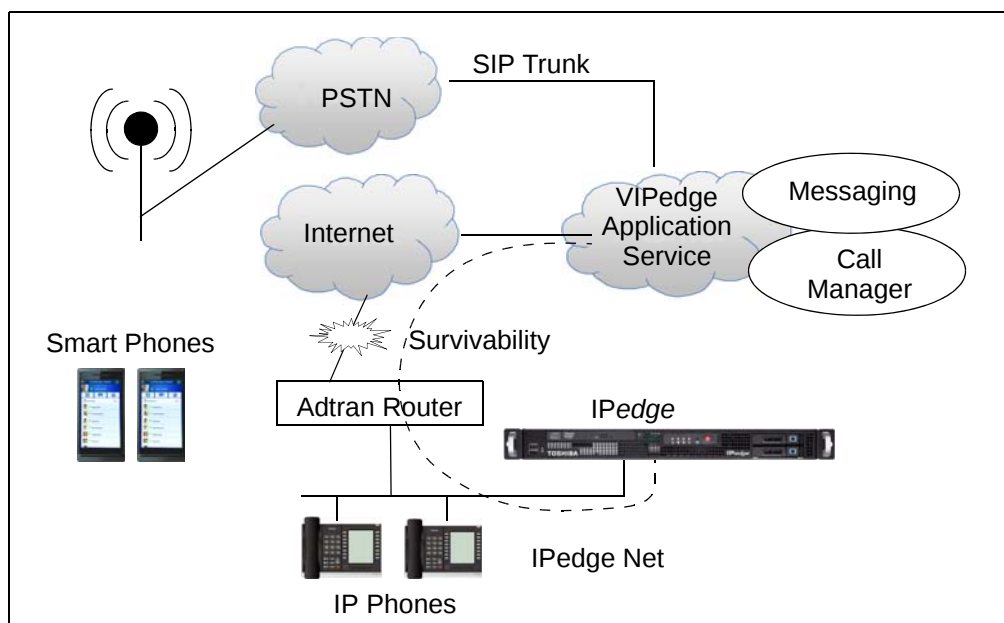


VIPedge Application Services

VIPedge Application Services allow premise based IPedge® systems to take advantage of unified messaging and unified communications applications and survivability in the cloud. IPedge systems can use cloud based voicemail so that even if their WAN connection is down, calls can still be answered by voicemail and follow their users to their call manager mobile clients on their cell phones. The VIPedge Application Services are supported using IPedge software version 1.6 point release 1 with GCP 1.6.0-5 or later.

SIP Trunks, Voicemail and Call Manager in the cloud

The VIPedge Application Service can be used with a premise based IPedge system to provide cloud based Messaging, Call Manager and UCedge client features. The IPedge system is connected to the Internet behind a router, and it is networked to the VIPedge Application Service using IPedge Net, creating a two-node network. Connected they operate as an integral system, but when separated by a WAN failure each is able to operate and act for the business independently answering calls in the cloud and allowing station to station dialing on the premise. The VIPedge Application Service is connected by SIP Trunks to the PSTN.



VIPedge Application Services offer the following features:

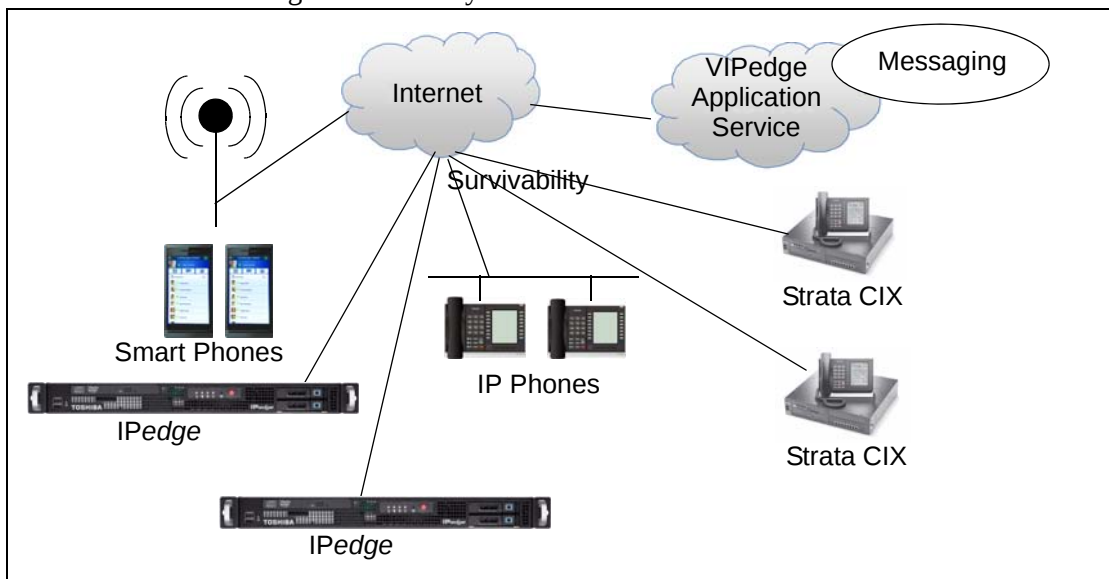
- **Messaging** – Calls that are not answered by the phone on premise are answered by VIPedge messaging in the cloud. A message can be taken and retrieved by phone, or it can be sent by email. The message waiting indicator on the on-premise phone can let you know when you have a message. Calls can be routed with the follow-me feature to a cell phone or call manager mobile client.
- **Call Manager** – The IPedge system on the premise can be configured to use the Call Manager in the Cloud. This allows call manager mobile for Android and iOS smartphones with presence and instant messaging to be used to provide reachability anywhere (there's wireless access).
- **Survivability** – If the WAN connection goes down, the station to station calling is still available on the premise. In the cloud, incoming calls will be answered and they can go to voicemail or to the user's cell phone or call manager mobile client using Follow-Me.

- **Configurations** – One IPedge system can be configured to one VIPedge Application server. One IPedge Net Channel is available for each SIP Trunk configured onto the VIPedge Application Server, including the burst channels. The on premise IPedge system can then be networked to other nodes on the premise for network calling throughout.
- **Fax**– Analog trunks connected directly to fax machines are recommended.
- **UCedge Client** – The UCedge Client is a productivity tool that is integrated with the VIPedge and IPedge business telephone solution. See “[UCedge Client](#)” on page 33 for more details.

IPedge Networking with VIPedge

Multiple CIX and IPedge systems can be networked with VIPedge.

- Configurations
 - Up to 127 IPedge and CIX systems can be networked with one VIPedge system.
 - On the VIPedge system, one IPedge Net Channel is available for each SIP Trunk configured onto the VIPedge Application Server, including the burst channels.
 - For FAX analog trunks directly to fax machines are recommended.



Part Numbers

See [Table 32 on page 130](#).

Microsoft® Lync® Integration

Toshiba has a plugin that is installed on a customer's PC to integrate with the Microsoft Lync client. This eliminates the complex server configuration that is required for server integration. This integration enables customers who adopt Lync as the Instant Messaging/Presence application to integrate with the IPedge system telephone features. Toshiba plugin requires the Call Manager Advanced license. In order to use the built-in softphone, Call Manager VoIP option license is also required.

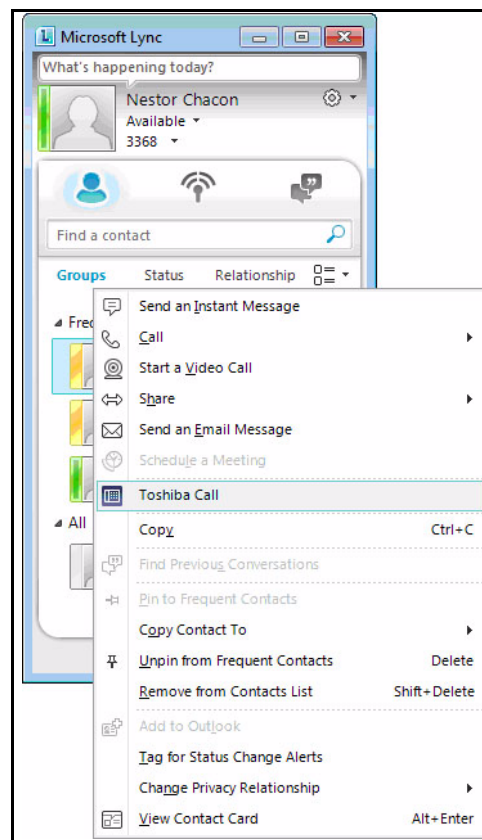
The following features are available through the integration:

- Lync Presence reflecting user's telephone status (In call).

When a Lync user is on a call using a Toshiba digital telephone, IP telephone, or built-in softphone; other users will see the user's status as Busy (in call).

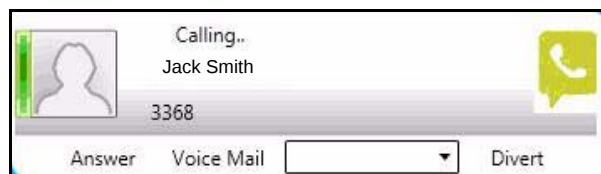
- Make Call from the Lync contact (shown right).

The Toshiba Plugin provides a menu to use the Toshiba digital telephones, IP telephones, or the built-in softphone from the Lync Contact by right-clicking a contact and selecting Toshiba Call. If the user selects Call, it will use Lync softphone when it is available.



- Pop up notification for a ringing call with Lync contact information (shown right).

When a call arrives, Toshiba Plugin pops up the notification and shows the contact name if available from the Lync Contact. The user can answer the call or route the call to the voice mail or other specified destination.

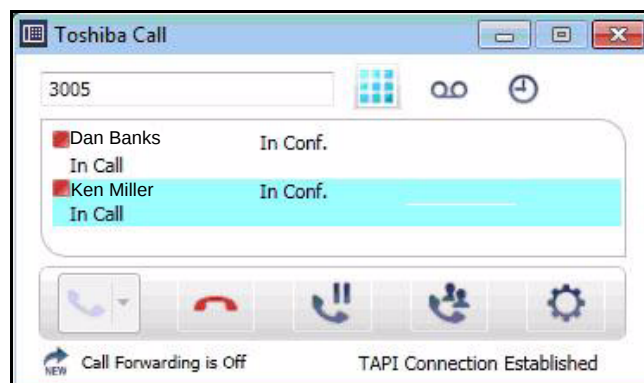


- Transfer and Conference Call

User can transfer or conference call from the Toshiba Plugin main window (shown right).

- Optional built-in softphone

Toshiba Plugin can be used together with Toshiba digital telephone, IP telephone, or SoftIPT for the user to control the telephone from Toshiba Plugin. In addition, as an option, the built-in softphone can be used with Toshiba Plugin.



Note Telephones to be used with Toshiba Plugin should be configured to have the Primary DN only, and Secondary DN/ Shared DN and other GCO/Pool line keys should not be used. When used, the Toshiba Plugin or popup notification may not work properly.

Meeting

The Meeting application is integrated into the IPedge system. Meeting allows participants to dial into a single conference or any combination of conferences. Meeting is web-based (shown below), so it's easy to set up conferences from anywhere, view conference participation during a call, and share a desktop screen. There can be up to four conferences with a total of 24 audio and web participants on the IPedge EC system; up to eight conferences with a total of 48 participants on IPedge EM system; and one conference with a total of four participants on IPedge EP system. Below is a list of features available with the Meeting application.



Audio Conference Features

- **Reservation-less and Scheduled Meet-me Conferencing** – enables conferences to start quickly and can be organized flexibly.
- **Web User Interface for Moderators** – enables moderators to schedule conferences.
- **Conference View** – shows moderators the participants that are in their conference and enables the management of individual participants. Participants can be muted, disconnected, or transferred to another conference for a side bar and conversation.
- **Telephone Portal for Moderator and Participants** – enables moderators and participants to exercise in-conference controls via DTMF.
- **Outlook Calendar Integration** – allows meetings to be easily scheduled and invitations distributed to all participants.

- **Web-based Reporting** – enables managers to have a view into the impact of audio conferences and web collaboration sessions in their business.
- **Moderator and Participants Codes** – adds security and control to who can manage and participate in conferences.
- **Exit and Entry Tone** – lets participants know when people are entering and leaving conferences in order to avoid surprises.
- **Audio Recording Capabilities** – enables audio conferences to be recorded for later playback or archived for record retention.
- **Flexible Configuration from 4 to 48 ports** – provides cost-effective small conferencing system for the SMB with room to grow and expand.
- **Dial Out** – Moderators can dial out (#31) to call participants into a conference.

Web Collaboration Features

- **Video** – Participants in web conferences can share video from the webcam on their PC.
- **Web based Desktop Sharing** – enables moderators to share documents, presentations and conversations in the meeting.
- **Web User Interface for Participants** – enables participants to join a web conference from any computer that is convenient at the time and does not – require a dedicated application to be installed.
- **Chat** – enables participants to exchange text messages to the group or individually while in a conference.
- **Computer Screen Sharing** – Sharing of a region from a primary or a second monitor.
- **Waiting Room** – Web conferences have a waiting room, where participants can wait until the moderator joins.

Mobility

The IPedge delivers virtually every feature to every user, regardless of the type of device they are using, whether they are stationary or mobile. Each individual user can choose the type of device that best meets their communication needs.

These devices can be used by local or remote users, so employees can work anywhere, with the same level of functionality and productivity.

The IPedge system supports IP phones, wireless IP telephones, IP softphones on laptops, PDAs, and tablet PCs.

uMobility Fixed Mobile Convergence (FMC)

The uMobility solution empowers mobile workers to cost-effectively make and answer their IPedge system calls from virtually anywhere. The user's smart cell phone functions as their IP telephone extension both while in the office via the wireless LAN and while out of the office via a cellular network.

By using the enterprise wireless LAN for voice calls rather than cellular minutes, enterprises get the best of both WiFi® and cellular coverage from a single device. The uMobility solution routes calls through the enterprise network for domestic and international connections. This solutions offers your enterprise a compelling way to optimize costs and improve accessibility for your mobile workforce.

uMobility is compatible with most cellular service providers and on select Android, BlackBerry, iPhone®, Windows® Mobile and Nokia® devices it enables users to make and receive calls as an extension of the IPedge system. uMobility solution contains two software components: client (smart phone) and server (uMC). The client communicates with the network-based uMobility Controller (uMC), selecting the best WiFi or cellular network available and joins the smart phone to the IPedge as an extension.

Users using the uMobility client have access to many IPedge telephony features. Refer to the Features chapter for more uMobility features.

Important! *Not all of the listed features are supported by all cell phones.*

- Make/Receive calls over WiFi and Cell (Answer/Release)
- Call Hold over WiFi and cell
- Call Mute and Volume control
- Blind Call Transfer
- Call Mute and Volume control
- Keypad dialing / DTMF support
- Viewable call status
- Missed call indication
- G.711 Audio Codec (SIP/WiFi)
- Personal directory
- Last Number Redial
- Voice Mail setting/access for SIP (WiFi) and Cellular using a common user interface
- Do Not Disturb (DND)
- Speed Dial
- Call History (Redial, Missed Call, Callers Lists)
- Access to contact list for dialing numbers
- WiFi/Cellular hand-off

Wireless Telephones

The IPedge also works with Toshiba certified SIP telephones, such as Polycom over WLAN. Refer to [“Polycom End Points” on page 20](#).

- The IP4100 DECT telephone on [page 16](#).
- The Polycom SpectraLink and Kirk Wireless Telephones fully integrate with the IPedge system.

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 three major features that allows the user to be mobile.

- Enables the user to log-off and log into any SoftIPT 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. 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.
- Allows the Administrator to “oversubscribe” when building IP extensions. The Administrator can create and build more IP stations using Enterprise Manager than there are licensed IP endpoints.

Advantages

- Multiple users can share one IP telephone (hot desk) or one SoftIPT.
- 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.
- IP User Mobility incorporates a primary and secondary server IP address. When an IP phone boots up, it first tries the primary IPU address. If the attempt is unsuccessful, the IP phone automatically tries the secondary IPU address.
- IP User Mobility works across multiple nodes. Any IP user can log into any IP phone in any node without manually entering a dedicated IPU address.

IPT Anywhere enables you to connect IP telephones remotely through the Internet and use all IP5000-series telephone features. IP telephone remote connections can be set with or without the use of Virtual Private Network (VPN). VPN connections provide increased security and are recommended for permanent type IP telephone remote connections. When moving IP telephones frequently to different locations (hotels, conferences, etc.), non-VPN connections are more practical.

When using home type xDSL or cable connections, only one or two IP telephones may be connected because of xDSL and cable bandwidth limitations. Broadband is required when installing two or more phones at a remote site.

An Internet configuration could use the following connections:

- No VPN, and thus, no security
- Third party VPN software residing on DHCP gateway server. To connect IP telephones over the Internet, using third party or Microsoft VPN software residing on a DHCP gateway server, see [Figure 8](#).
- ATM (IP over ATM virtualization by VC/VP)
- Broadband Ethernet virtualization by Virtual LAN (VLAN)
- IP-VPN (IP-VPN based on Multi-protocol Label Switching (MPLS))
- Private line connection

For an access line to link the user's location with the access point of the carrier or provider, using a private line, broadband line (xDSL, CATV), or fiber optics is recommended.

The IPT Anywhere feature enables remote IP telephone users working in branch offices or home offices to make full use of the extension features of the IPedge. The diagram below shows IPT Anywhere connections using the optional VPN connection.

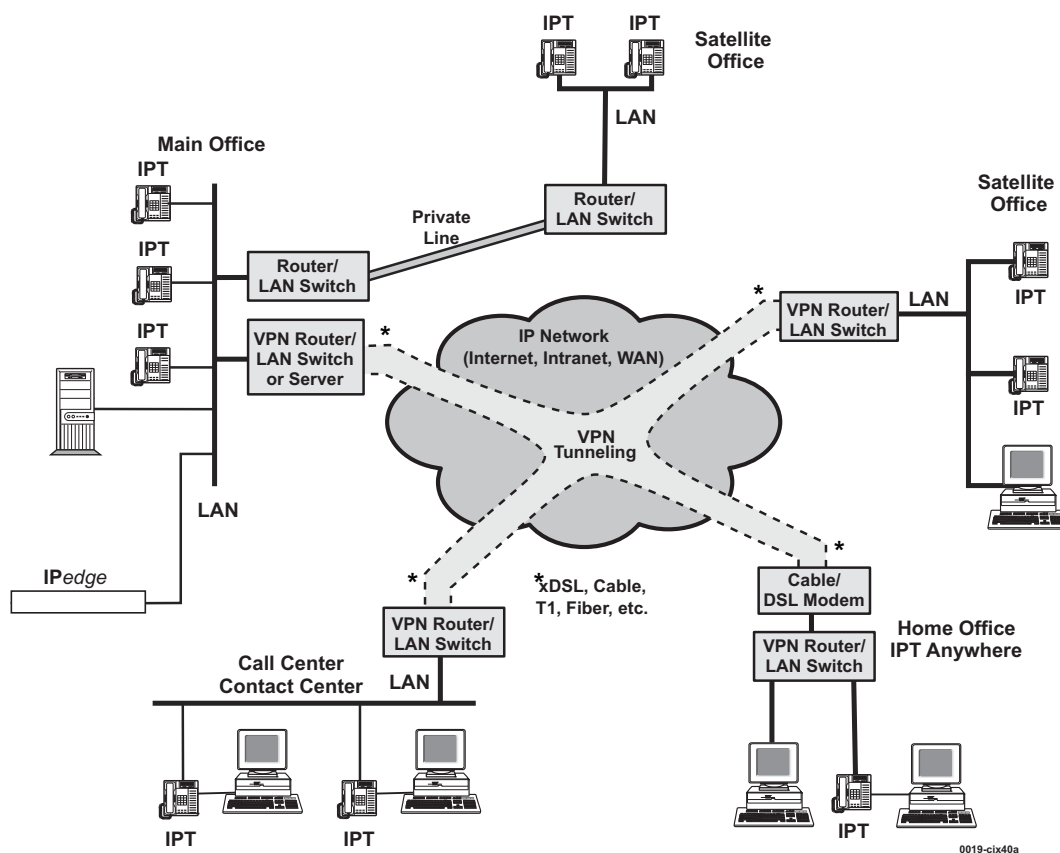


Figure 8 IPT Anywhere Example

SoftIPT Client

The Toshiba SoftIPT™ is an IP telephony client that works with a wired or wireless (Wi-Fi) tablet, laptop or desktop PC. The SoftIPT client on a PC integrates the power of the PC with most of the features available on an IP5000-series telephone.

With the Toshiba SoftIPT installed on a Wi-Fi laptop PC, users can have true mobility with access to voice mail, programmable feature buttons, and a directory that works with Microsoft® Outlook®.

The Toshiba Soft Phone works on desktop or laptop PC with Microsoft Windows Vista, Windows XP and Windows 2003 operating systems (OS).

The SoftIPT client looks similar to 5000-series telephones (shown below).

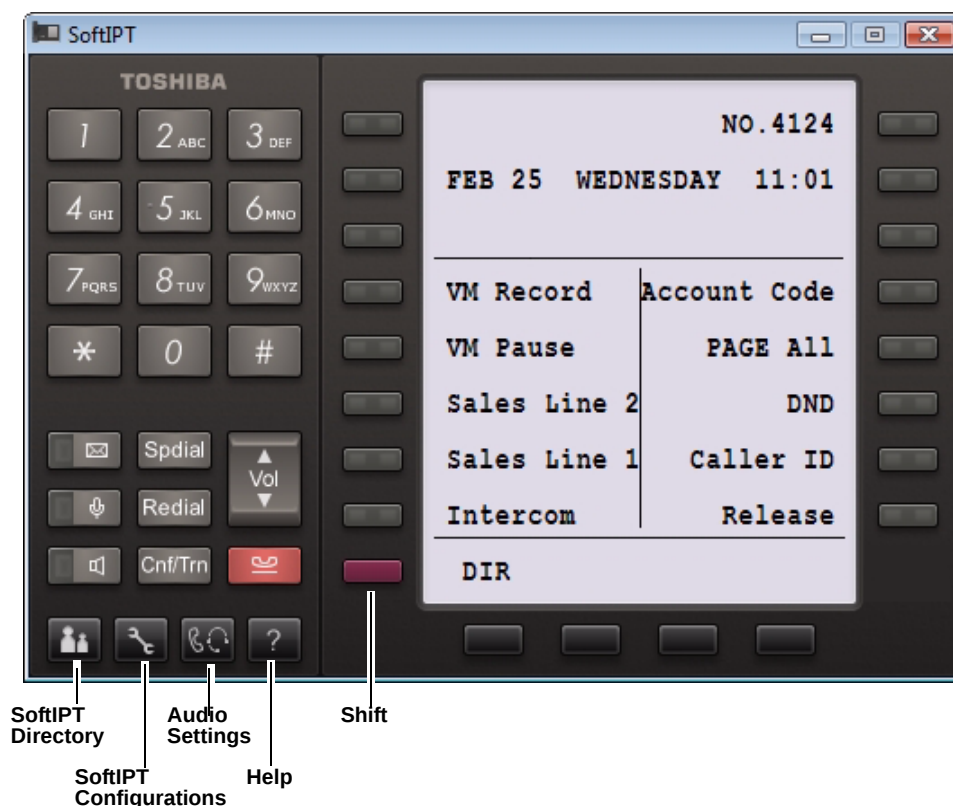


Figure 9 Toshiba SoftIPT Sample Screen

SoftIPT on a PC integrates the power of the PC with most of the features available on an IP5000-series telephone.

The features supported by SoftIPT are:

- Self-labeling feature keys
- User control over feature button labels
- Shift key toggles the LCD screen between flexible key 1~10 and flexible keys 11~20

A mouse or stylus is used to click or select the buttons. There are multiple feature buttons that can be customized from telephone programming mode.

With Microsoft Outlook, you can create a directory for the SoftIPT. Once a directory is created, the user can click on a name in the directory to automatically dial their number (see [Figure 1](#)).

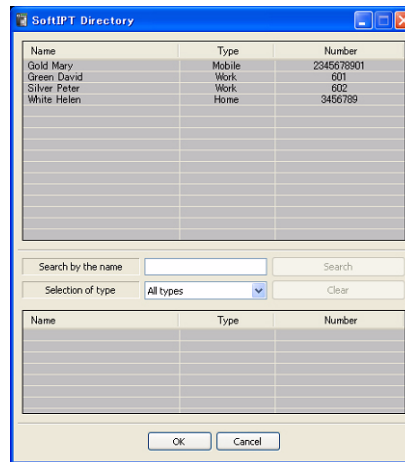


Figure 1 Example of SoftIPT Directory

The SoftIPT can be connected to the IPedge system in several different ways:

- Intranet – A wired or wireless PC can connect to the office LAN that connects to an IP telephone.
- Internet – A wired or wireless PC at a remote site can connect to a Cable or DSL modem, to an Internet Service Provider (ISP), to a router.
- Wireless – The wireless PCs or Toshiba Tablet PC need a Wi-Fi system. The SoftIPT wireless units can operate within range of an access point (dealer-supplied or use existing).

Licensing

Refer to [“IPedge Net IP and IP Telephone Quality of Service”](#) on page 116.

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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 installing the IPedge system.

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

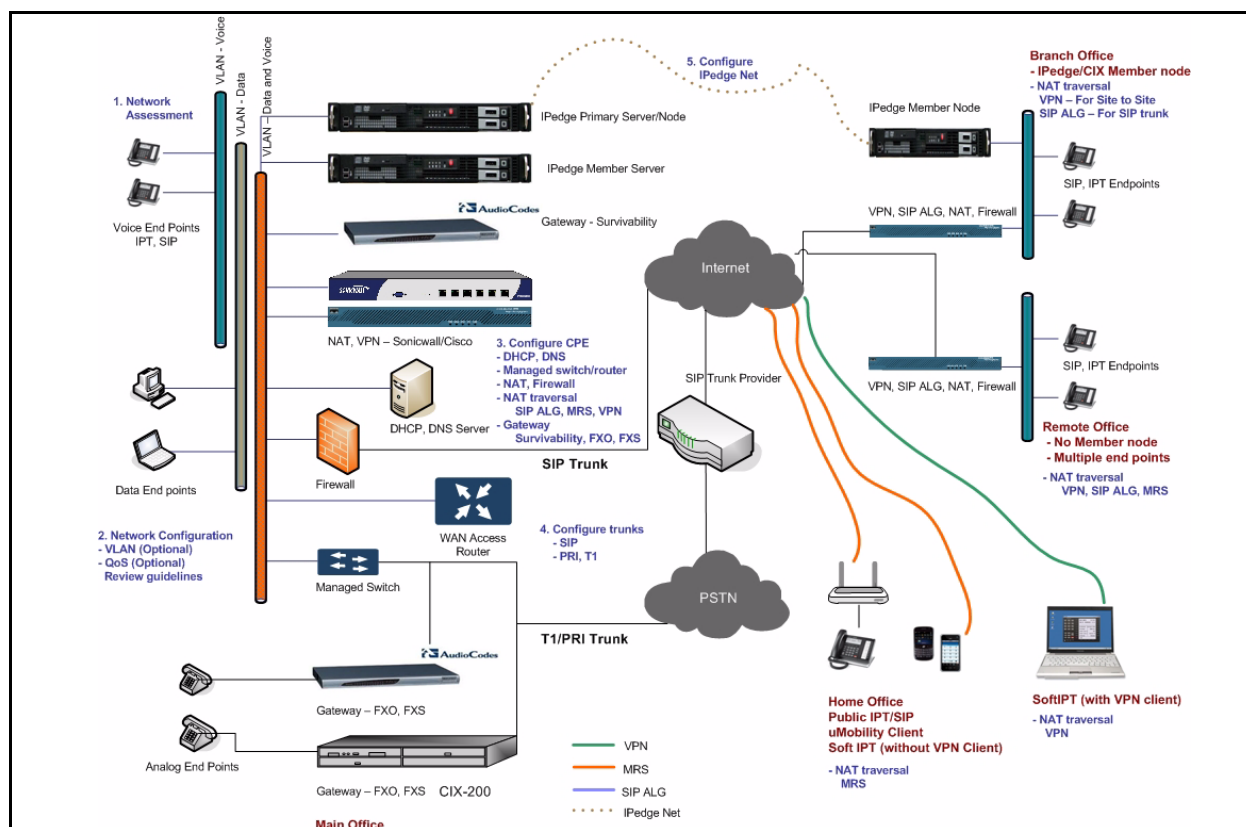
Network Assessment – A network assessment must be carried out to determine whether network or service upgrades are required to support a VoIP deployment. Toshiba recommends carrying out the network assessment with QoS enabled on the network.

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	Capacity
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

Network Device	Capability	Capacity
SIP ALG	Ability to successfully and accurately transform all SIP/SDP message headers.	Throughput, Latency, Jitter
VPN Gateway	Ability to create site to site tunnels. If required ability to create multi-site VPN tunnels.	Throughput, Latency, Jitter
Server Platform	DHCP, DNS, NTP Services	Capability to handle multiple concurrent requests at high performance.

Site Network Diagram & Configuration – For networked VoIP deployments with multiple IPedge or Strata CIX nodes, multiple trunk types or groups, one should document the network topology in the form of a diagram (example shown below) along with configuration settings for various network connectivity options (VLANs, IPs, IPedge Net, VPN, and Trunks etc). This topology diagram along with the system connectivity configuration will serve as reference for both, planning the initial deployment and for post-deployment maintenance, troubleshooting and debugging activities.



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.

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)	Wide Area Network (WAN)
Reliability	99.99% uptime	99.99% uptime
Latency	< 20 msec	< 50 msec
Jitter	< 10 (+/- 5) msec or less	< 20 (+/-10) msec or less
Packet Loss	< 0.1%	< 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.
- 802.1q (VLAN) – For larger deployments (> 100 IP Phones) create a separate Voice VLAN and a data VLAN to limit the broadcast domain and minimize impact of data traffic on voice 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 802.1q, 802.1p, 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.

Static IP Addressing for IPedge – In order to ensure availability and security of the system, the IPedge server and any gateways in the system must be provisioned with a static IP address on the LAN. This process must be carefully managed to ensure that there are no IP address conflicts in

the network. *IPedge* can be deployed with a private/static IP address, or it can be deployed with a public/static IP address as long as it is still behind a firewall.

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

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, SIP trunking configuration uses URIs and hostnames and these must be resolved to IP addresses using a DNS server. The *IPedge* server requires that one FQDN be configured, and if *IPedge* meeting is being deployed, a second FQDN is required.

NTP Server – It is critical that all networked devices (*IPedge*, gateways and IP phones) are synchronized to the same clock (system time). The best way to do this is configure all of the network devices to get and set their respective clocks from a public NTP server.

Firewall/NAT (Network Address Translation) Router – The *IPedge* server is designed to be deployed, primarily in a private address space behind a Firewall and NAT router.

- In order to successfully deploy and manage the *IPedge* system the firewall rules need to be configured to ensure that all valid incoming traffic is allowed by the firewall. For security reasons, the firewall should be configured to deny all other traffic.
- One must configure NAT policies (both inbound and outbound) to map traffic for all *IPedge* services between the WAN public IP address and the LAN *IPedge* private address. The details on how the NAT policies and Firewall rules need to be configured are described in the requirements that follow.
- Recommend network monitoring with a tool such as WhatsUp Gold, OpenNMS, and Zenoss®.

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). This is currently a system wide setting in *IPedge*. However, this will enable 802.1p on remote phones requiring the switches at all locations where remote phones are deployed to support 802.1p.

For IP telephone configuration, broadcast mode is not recommended for large deployments (greater than 100 phones).

Remote Administration

Benefits

Gives an administrator the ability to manage the system from a remote location, thereby saving time and money.

Requirements

In order to manage an IPedge system from a remote location, any of the following mechanisms can be used.

HTTPS – This is a standard web based method to securely access and administer the IPedge system from a remote location. In order to use this mechanism:

- Configure the Firewall to allow bi-directional HTTPS traffic.
- Configure NAT router to forward HTTPS traffic on a given port to the IPedge HTTPS service (TCP port 9443)
- Configure the Firewall to allow and NAT to forward TCP ports (10000, 9101-9103)
- Configure certificate on the IPedge system for HTTPS authentication
- Setup a FQDN for the IPedge server in the DNS server.

IPSec VPN – An IPSec VPN session is setup so an administrator can log on to the network and administer the system directly. The VPN session must be configured with sufficient privileges to enable access to the IPedge system.

Firewall and NAT Configuration – This method relies on manual configuration of the Firewall (allow inbound management traffic) and the NAT router (forward inbound management traffic to IPedge) to allow remote access to the IPedge system from a specific IP address or group of IP addresses. The following ports need to be opened and forwarded for remote administration purposes, from specified IP addresses:

- HTTP – TCP port 8080
- Webmin – TCP port 10000
- BACULA – TCP ports (9101 - 9103)

Interactions

Port Forwarding – In some instances and deployments the standard ports used by IPedge may also be used by other systems and/or services within a deployment. One can also encounter this situation if there are multiple IPedge nodes within a single LAN. When one encounters such situations it would be necessary to use NAPT (Network Address and Port Translation) instead of standard NAT (Network Address Translation only) translation, to enable remote administration.

Centralized Administration

Benefits

Centralized administration allows multiple nodes to be managed through a single point, reducing the time it takes to administer multiple servers and also reducing the possibility of miss-configuration.

Requirements

- **VPN** – A VPN is required between sites that implement Centralized Administration in order to provide for security of the administration information.
- **Multi-node Centralized Administration** – The IPedge EC and EM systems enables administration of a multi-node system from a single node called the Primary Server. In order to enable this unified administration the following SNMP ports need to be opened and forwarded on the Firewall and NAT router (at each site that is being administered) respectively.
 - SNMP – UDP ports (161-162)

Interactions

For most centralized operations an administrator will connect to the primary node. For some interactions, like setting the IP address on a remote node/server, the web browser will be redirected to the specific node/server. To allow these functions it is necessary to configure the Firewall and NAT rules at the remote node as documented in the “Remote Administration” section above.

IPedge Net

Benefits

A system with multiple independent IPedge nodes (and associated end points) can be logically unified by networking the individual nodes together using IPedge Net. The primary benefit for doing this is visibility and connectivity between any two end points irrespective of what node they are connected to. The IPedge Net wizard reduces installation time in IPedge systems. The wizard optimizes IPedge Net programming by providing the basic IPedge configuration and reduces manual input.

Requirements

- **IPedge Net** – From IPedge Enterprise Manager, IPedge Net connections need to be setup between all the connected nodes.
- **VPN Connection** – If the individual nodes to be connected are separated across a public network (Internet/WAN) a VPN connection needs to be setup for a successful IPedge Net connection to be established between these nodes.

Interactions

IP Packets from phones on different IPedge Net nodes are relayed through the IPedge server; IP packets for these communications are always sourced from one node's IP address and destined to the other node's IP address. If these nodes are behind a firewall this communication would need to be allowed.

Remote IP Telephones

Benefits

- Small office or Home office users can be connected to their work phone system.
- When using SoftIPT, enables Road warrior scenarios.

Requirements

In order to connect a remote IP telephone (or SoftIPT) to an IPedge node, either of the following mechanisms can be used:

Media Relay Server (MRS) – Within the Firewall and the NAT router at the main office, the following ports need to be opened and forwarded; in order for the remote IP telephone to communicate to the IPedge system:

- H.225 – UDP (1718 – 1719)
- Megaco – TCP 2944
- Media Relay Server – UDP port range (21000 - 22999)
- FTP, FTP Data – TCP ports (20-21)

IPSec VPN – An IPSec VPN session needs to be setup between the remote location and the main office. In case of a SoftIPT the IPSec VPN connection is established by a soft VPN client back to the main office.

Interactions

- When using MRS, most home office and small office routers require no configuration in the firewall and NAT locally as they tend to initiate the connection and save state for the inbound connection. In some cases this may not work and the ports references in Requirement 1 may also have to be opened on the home router.
- Using the MRS the IPedge system dynamically and automatically configures port forwarding rules for any NAT traversal related issues. If, for some reason a remote IP telephone still cannot register due to local NAT router issues, configure the remote IP telephone's station programming to use the Static MRS parameter. This will resolve any NAT issues that prevent registration of the IP telephone.

Remote SIP Phones

Benefits

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

Requirements

In order to connect a remote SIP end point to an *IPedge* node, any of the following mechanisms can be used:

- **SIP ALG (Application Layer Gateway)**
 - A SIP ALG device needs to be configured at both the main office and at the remote location to transform the SIP/SDP session header addressing information.
 - Firewall and NAT configurations to allow and forward SIP – UDP port 5060 to the *IPedge* system.
- **Media Relay Server (MRS)** – If a SIP ALG does not operate as desired, the MRS function within *IPedge* can be used to get a remote SIP end point registered with an *IPedge* node. To use this method, within the Firewall and the NAT router at the main office, the following ports need to be opened and forwarded; in order for the remote SIP end point to communicate to the *IPedge* system:
 - SIP – UDP port (5060)
 - Media Relay Server – UDP port range (21000 - 22999)
- **IPSec VPN** – A remote SIP end point can also connect to an *IPedge* node over an IPSec VPN connection.

Interactions

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 *IPedge* Static MRS function can be invoked to still support these end points.

Toshiba's SIP Trunking I-VoIP Service

Benefits

- SIP trunking is one of the most cost effective ways for an enterprise to provide voice services for their users.
- In an increasingly data networked environment, it offers enterprises the ability to use their existing data network to offer both voice and data services; thereby simplifying and consolidating the costs associated with maintenance and administration of two separate networks.
- IPedge system is able to support SIP Trunking with routers that cannot support SIP ALG.

Requirements

- SIP trunking service – SIP trunking service needs to be purchased from a Toshiba certified SIP trunk provider.
- Configure SIP trunking on IPedge node – SIP trunk parameters provided by the service provider need to be configured within the IPedge administration interface in order to register with and use the service.
- Within the Firewall and the NAT router the following ports and protocols need to be opened and forwarded; in order for local end points to communicate and make calls via the SIP trunk service:
 - SIP – UDP 5060
 - RTP/RTCP – Dynamically assigned
- When used behind a NAT firewall that does not support a SIP ALG, the IPedge system can still be given a private IP address. The IPedge server will use it's internal Media Relay Server (MRS) to route media packets between the WAN and the LAN to internally fix-up the NAT translated IP address in the packets and the IPedge system will fixup the SIP messages. Within the NAT router, port forwarding rules will need to be configured, and a range of ports opened for the Media Relay Server.
- When used with a NAT firewall that does support an enterprise grade SIP ALG (such as the Cisco ASA5500 product line), the SIP ALG feature needs to be enabled. In this configuration, the media packets will be routed directly from the LAN to the WAN and mid-call survivability of a PSTN call is possible.

Interactions

Different SIP trunk providers have different configuration requirements to enable the service. There are variations in requirements around registration, authentication, ports used, SRV records etc. For details and references on configuring SIP trunk programming for these different variations, refer to the IPedge Install Manual.

Web Conferencing

Benefits

Gives users across geographic boundaries the ability to do audio and web conferencing on demand. This is helpful for purposes of collaboration in distributed team environments; attendees from different locations can view and work on the same information in real time by using features such as desktop and document sharing.

Requirements

- Within the Firewall and the NAT router at the main office, the following ports need to be opened and forwarded, in order for a participant to access the *IPedge* Meeting conferences:
 - Admin – TCP 80
 - Secure Admin – TCP 8444
 - Web conferencing – TCP 443
 - Data port for desktop screen sharing service – TCP (1935, 1945)
- DNS Server Configuration – In order for the *IPedge* Meeting service to be easily and publicly (from the WAN) accessible, a couple of fully qualified domain names (FQDNs) need to be created. These FQDNs need to be mapped via the DNS service to the IP address of the *IPedge* Server. FQDN mappings within the DNS service (server) need to be created for:
 - Meeting administration service
 - Meeting web conferencing service

Note If the *IPedge* server is deployed behind a NAT firewall, the DNS service would need to be configured to map the FQDNs to the NAT public IP address instead.

Interactions

In some IT environments, public facing services and servers are configured in the DMZ. If that is the case in a customer deployment, firewall and NAT rules will need to be configured for *IPedge* meeting services to be accessible from both the WAN and the LAN.

An external Automatic Call Distribution (ACD) software option with the *IPedge* is provided by connection of an external PC-based CTI application server or as an application on the Media Application Server (MAS). See [“Media Application Server \(MAS\) and MicroMAS” on page 129](#) for specifications. The CTI server runs both the ACD call processing application and the separate Management Information System (MIS) application such as Insight, as well as other CTI applications.

The ACD application is available in Basic and Enhanced feature functionality, along with the number of groups and active agent size increments to provide cost-effective pricing levels according to the user’s needs. Enhanced ACD includes all basic capabilities plus multiple group login, skills-based routing, priority queuing, time scheduled ACD queues, agent and call priority escalation handling, and balanced call count agent search. For more information regarding ACD, ACD Licensing, etc., refer to the *IPedge Contact Center General Description*.

Basic ACD Features

Advanced Call Routing

The optional Call Router enables calls to be routed based upon parameters such as Caller ID, account numbers, private lists, time-of-day, day-of-week, day-of-year, and user entered data (account code, etc.). This is an optional feature that can be added to the ACD application.

Intelligent Announcements

The holding caller can be informed of call status, such as their place in queue or estimated time before an agent answers. The intelligent announcement function can also offer alternative options to continuing to hold, such as going to voice mail.

IVR Voice Assistant Open Database Connectivity (ODBC) Access

There are two options that can be used with customized services: Interactive Voice Response (IVR) Voice Assistant (VA) application can be used as a stand-alone product and/or as an IVR service to the ACD application. For example, an IVR port could be used to do an external page to alert agents to return and login to an ACD queue when it gets too overloaded with calls.

Other useful functions include gather and validate user input, play menus and act on response, and trigger other events. The IVR VA can also be used to provide low cost text to speech capabilities. The IVR VA is an optional feature that can be added to the ACD application.

Enhanced ACD Features

Agent Priority Routing

The Agent pool can be expanded when traffic gets heavy based upon agent priority levels. When all agents are busy at one level, calls automatically get distributed to agents at the next level. Calls can be distributed by agent priority, preferred agent treatment, or balanced call count.

Multiple Group Agent Login

ACD agents can be logged into multiple ACD groups, enabling agents to answer calls for multiple groups. This is very useful for back up coverage between groups. It is also the foundation for skills-based routing and agent priority routing, enabling many advanced call center applications.

Skills-based Routing

Based on the caller's input, the system can route the call to the agent best suited to handle the call. Calls can be routed to certain agents, based upon agent capabilities, in addition to Dialed Number Identification Service (DNIS), CO line, or Auto Attendant routing into different groups. With the capability for agents to log into multiple groups, calls can be routed to different agents based upon skills needed for each specific call.

Priority Queuing

ACD calls can optionally be tagged with a priority number before they are placed into the ACD group queue. The priority number assigned to the call determines where the call is placed in queue. This feature enables high priority calls to be answered sooner than low-priority calls. The escalation parameter ensures that no call is lost due to higher priority calls.

Web-based Contact Center

Multimedia Contact Center application uses web technology to increase the customer's reach by providing web surfing customers with the option to contact the contact center.

Web Callback

The Web Callback service enables users to place a callback request from the customer's website. This service gives users easy access to the contact center via the web. Most types of browsers are supported, including on the Mac®.

The Web Callback service easily integrates with a customer's existing web pages. This service is set up similar to the current voice based callback and the call center has the same ability to generate reports they currently use.

Chat Contact Center

Chat is a semi-realtime communication method which adds another media of the communication to users. For example, if a Technical Support person needs to spend some time to investigate a question from the user, the user can continue to surf the web while waiting for the response from the Technical Support person. Sales groups can assist users in placing orders while the user is on the web page.

This chapter contains the IPedge features. They are presented in alphabetical order to make it easy to locate each feature.

Account Codes

Account Codes are often used for cost allocation of the call or the time the caller was involved on a phone call. The codes are printed on a Station Message Detail Recording (SMDR) printout along with other call details so that the customer can identify all calls associated with a specific account code.

Account codes may be forced (required after dialing all or specific phone numbers) or voluntary (optionally entered anytime during calls). Codes can be as long as 15 digits and can be verified or non-verified by the system.

Active Directory Sync

IPedge Active Directory Synchronization (AD sync) allows an administrator to import user information such as the name, telephone number and email address to the IPedge system. It makes it easy for an administrator to install the new system by importing the user information from the existing Active Directory. It can also be used to automatically import changes such as new users, deleted users, and modifying existing users by scheduling synchronization.

AD sync supports networked IPedge systems, and users can be assigned to the proper IPedge system based on the value of the specified data field such as Office in the Active Directory.

Active Directory Sync also supports LDAP server.

Add-on Module (ADM)

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

One to two KM5020's (20 button) can be attached to the IP5000-series telephones to provide an additional 40 flexible buttons.

Note The KM5020 is not supported on the VIPedge system.

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

When the IP5000-series telephone or attendant console dials an extension number with an Advisory Message enabled, the message displays on LCD of the originating station. When a PhDN is dialed, the message set at the PhDN's owner station displays.

If the destination station answers the incoming call and starts a conversation, the LCD displays as an ordinary call. When the originating station goes on-hook or any other button is pressed, the LCD returns to the idle display. This status is also sent to Attendant Consoles.

Alarm Notification

IPedge can send alarm notifications to a Monitoring PC/Server or send an alarm notification to a telephone.

System alarms can be sent to up to three unique IP addresses from IPedge SNMP traps.

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 Call Distribution (ACD) Server

See [Chapter 5 – Contact Center](#) for details.

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.

Background Music (BGM)

Background music can be played through the speakers of IP5000-series telephones.

There are 15 music sources from which BGM may be selected.

Selection of the BGM source to be played can be done individually by each telephone user and for each external page zone using Enterprise Manager.

With the “random” feature for music sources, the music will start at a random point every time the BGM starts.

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 may be activated or deactivated for every Directory Number (DN), Primary Directory Number (PDN), Phantom Directory Number (PhDN), Pilot Number of a Multiple Call Group, and Station Hunting pilot number.

Call Forward **may not** be activated or deactivated for any of the Group CO (GCO) or pooled lines buttons. Call forward applies to the PDN of the station that owns the GCO or pooled line button.

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

The Call Forward feature may be programmed at IP Telephone base station, attendant console, or online 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 re-dialed 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

The Call Manager features are covered in [Chapter 3 – Unified Communications](#).

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 pre-programmed 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 predesignated External Paging circuit, a Telephone Paging group or both.

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.

Ring, 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. IPedge Net provides three means of transferring a call and three means of terminating transferred calls, 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. 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 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 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 voice mailbox does not need to be associated with an active telephone in the IPedge Net system. 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 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 Line or DN, two short beeps are issued to alert the telephone user of the call.

Call Waiting works for calls originating from within or outside the system. The length of the Call Waiting beeps are different for internal and external Call Waiting types.

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). Digital stations are assigned memory for creating Call History.

ISDN Calling ID Name and Number

Both Caller ID name and number are supported from the service provider (if available.)

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) and an attendant console have 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

The Conference Call feature enables a user to establish a conversation with two or more parties. These additional parties may be internal or external to the IPedge system. Any IP5000 Series telephone may establish a conference call with other stations or outside lines.

Some models of SIP telephones, including the Toshiba IP4100 Series, also have their own built-in conference feature. Please refer to the device documentation for description and programming instructions.

A conference is defined as three or more parties joining into one conversation. A maximum of eight parties are allowed into a conference with up to six from outside lines or eight parties on internal stations.

The originator of the first conference is the “conference master” and controls the adding and dropping of conference parties. The conference master may drop the last added party by pressing the **Cancel** button.

Note Some third-party SIP devices are able to master conferences - please refer to those systems’ respective documentation for more information.

Conference on-Hold

A conference call may be placed on hold where callers remain connected and no Music-on-Hold is applied. The station placing the conference on hold may rejoin the conference by pressing the **Line** button.

“Conference Master” status may be transferred to another station by placing the conference on hold, and pressing **Line** on that station. 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 caller conference may leave (Split) the conference for a private conversation. During this time, other conference members will remain connected. The conference master may “join” both of the “split” callers back into the conference, or the master may disconnect (Drop) the split member they are connected to.

Releasing Tandem SIP Trunk Connections

This feature disconnects unattended line-to-line connections for the IPedge system, 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 may then be used to make or receive calls from the originating telephone.

Telephones and/or Voice Mail (VM) devices may establish and release tandem SIP trunk connections without disconnecting the tandem connection in the IPedge system. After releasing from a tandem call, reconnecting is accomplished by dialing an access code.

Tandem line buttons may or may not appear on a telephone. The user may 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.

Credit Card Calling

Callers can make “0+” telephone credit card calls from selected toll restricted stations. When dialing from toll restricted stations, if the caller does not enter a credit card number after dialing “0,” the call will be disconnected.

Calls are billed to the credit card instead of the IPedge Net line. The “0+” credit card calling feature can be enabled selectively or assigned to stations and CO lines capable of supporting this service.

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 attendant, 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 attendant or 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.

Dialed Number Identification Service (DNIS)

DNIS lines receive 800 and 900 type telephone calls that provide the number the caller dialed to reach IPedge Net. The IPedge Net translates the DNIS number into a name that displays on the telephone's LCD. This allows the user to identify the caller before answering.

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.

DNIS use cases 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 allows one group of lines 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.

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.

The DID format used in IPedge must follow specific entry rule and exclude any formatting:

- 1-949-583-3000 should be entered as: 19495833000
- +011-23-4567-8901 should be entered as: 0112345678901

DID numbers can vary between 1~15 digits in length for each DID line group on systems running IPedge R.1.6 and later software. Earlier systems support 1~7 digits. Each DID number may be assigned to 1 of 15 possible music-on-hold sources.

Directory Numbers

Primary DN Buttons

Primary Directory Number (PDN) buttons 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 (PhDN) buttons 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 IPedge Net, a time-out and overflow service is provided to redirect the call when the link is down. Calls being held on the Pilot DN using the CTI link can specify any of the 15 on-hold music sources that are possible on the IPedge Net.

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

Direct Station Selection (DSS)

Direct Station Selection DSS buttons can be placed on IP telephones, add-on modules, and DSS consoles. When placed on one of these devices, these buttons serve two functions: to make direct calls or transfer calls to other stations; and to display the status of other stations and PDNs.

The DSS button is numbered with a station's PDN and when pressed, calls that PDN. DSS buttons are not DN buttons, so they do not provide a dial tone when pressed. The DSS button's LED shows the status of the station and the PDN it represents (idle, busy, DND or ringing). The DSS' LED will turn on steady or flash at a unique rate, depending on the status (refer to the DSS Button Status Display list below).

The DSS LED displays the telephone's status for any type of call on any button, including PhDN and **Line** buttons. The DSS LED also indicates the status of the station's PDN. If the PDN appears as a Secondary DN (SDN) on multiple telephones, the DSS status will display Busy if any telephone is using the SDN.

DSS Button Status Display

Red, steady: Busy on a call not connected to your telephone

Green, steady: Busy on a call connected to your telephone

Red, quick flash rate: Ringing

Red, slow flash rate: Do Not Disturb

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

IP Telephone users sometimes need to distinguish the ringing of one key on their phone from another key 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 key on each telephone.

You can set up to ten different incoming ringing tones for internal, as well as external 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 DNs. Telephones with DND Override capability can ring DND telephones.

Emergency Call

An emergency call access code may be established in the system to route calls to specified emergency destinations and to prioritize their delivery to those destinations. Up to four emergency destinations may be programmed for each mode of operation: Day, Day 2, 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.

Emergency Call Feature Code default system setting is “#911”.

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, and Network Coordinated Numbering can be established uniquely in each system.

Coordinated Numbering Plan

IPedge Net may be configured to allow users to call each other across network nodes with simple network directory numbers. This eliminates the user's need for access codes and network maps. Calls that encounter a busy or unanswered destination can be forwarded to any node in the network, including a centralized voice mail system or attendant.

Handsfree Answerback

When a voice-announced internal DN call comes in to a telephone, users can answer without lifting the handset. SIP, cordless, single line telephones, and standard telephones are not compatible with this feature.

Headset

IP telephones have a built-in headset adapter and therefore require only the headset.

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 may then alternate between the new and the old call by pressing the desired **Line** or DN.

Note 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 feature. Call Hold temporarily suspends a call, allowing the station user to perform additional tasks, including using the phone. Callers on hold may 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 may be placed on Exclusive Hold to ensure the privacy of the connection and that the call may only be retrieved by you, even if the held call appears on buttons on other telephones.

Hold

A call placed on hold may be retrieved by anyone, if the held call appears on buttons on their 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 call on hold. Hold recall time is set independently for each telephone (from 0~255 secs.). Hold recall time may 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. This feature is compatible with standard and digital telephones.

IPedge Net

IPedge Net is a private networking application based on QSIG, an international standard for interconnecting telephone systems. IPedge Net delivers a rich set of calling features across multiple systems throughout the enterprise. Users benefit from transparent dialing and simple feature operation.

Multi-system networking can be implemented over an IP network using IPedge systems, VIPedge systems, and Strata CIX systems with MIPU or GIPU IP interface circuit cards.

Advanced networking features include Centralized Voice Mail, Centralized Attendant, Network SMDR, and Station DSS button appearances across all nodes. Alternate routing provides for toll bypass configurations and automatic recovery from network disruptions.

IPedge Net provides full connectivity and capabilities over an IP network (VPN WAN, Internet, intranet Frame Relay, fiber, or wireless).

Up to 128 nodes can be accommodated within the IPedge Net numbering plan. As with any network design, transport delay, speech volume and other issues must be carefully considered.

You can set up network DN tables across nodes. Through system programming, you can attach a node ID to non-redundant DNs, PhDNs, and Pilot numbers. This enables someone in one node to call an extension in another node without having to dial the node ID number. The caller dials the extension and the system automatically routes the call to the node in which the called extension is located and rings the called extension.

IPMobility

IPMobility is an IPedge Messaging application for the Android and iOS that allows a mobile device to perform as an extension of the office desk telephone. For devices that support

IPMobility, refer to [“Mobile Device Support for IPMobility” on page 130](#). IPMobility provides the following features:

- Support for the IPedge Follow Me (twinning) feature.
- Outbound calling through the host IPedge system.
- Visual Voice Mail.

Follow Me (Twinning)

The IPedge Messaging Follow Me (twinning) feature enables a single phone number to reach a user's chosen devices, e.g., desk phone, mobile phone, or both (simultaneous ring). Once answered, IPMobility offers call management providing users with a popup screen within the application to transfer the call to another extension or transfer the call to voice mail. IPMobility also gives users the ability to designate how to handle incoming calls if busy or out of the office for an extended absence.

Important! *The incoming call management described above requires the mobile phone service to support simultaneous voice and data (characterized by the ability to access the internet while talking on the phone). Administrators need to check with their specific service provider to confirm simultaneous voice and data.*

Making Calls

For outgoing calls, Toshiba's IPMobility application uses the host IPedge system's phone services to reach intended destinations. This feature not only takes advantage of the host system's telephone service rates, but also masks the user's cell phone number with the IPedge system office phone number.

IPMobility uses either a Call-thru or Callback process to set up the call.

- Call-thru – IPMobility sends a data command to the host IPedge system to notify the system that a user wishes to make a call. IPMobility then dials a specific DID number into the IPedge system. The calling party identification of the mobile phone is compared with the previously received data command, and then calls the destination number and bridges the two calls together.
- Callback – With Callback, after the same data command is sent, the IPedge system calls the mobile phone back, then calls the defined destination and then connects the two calls.

IPMobility does not conflict with the mobile device's ability to make a phone call or access the service provider's voice mail. Users can dial within the IPMobility application by typing in the phone number or extension directly or use the mobile phone's built-in contacts.

Visual Voice Mail

Users can also easily access key voice messaging functionality and manage administration of their voice mailbox without dialing into the voice mail system and navigating key presses or voice commands. Now, users can view, play, forward, and reply to their voice and fax messages mail from within the IPMobility application. Users can also;

- Manage mailbox personal greeting and name recordings
- Manage mailbox password.
- Setup IPMobility's Make Call functionality, e.g. Call-thru, Callback.

Line Buttons

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.

Note A DSS Button is NOT a line button

Live System Programming

Programming IPedge from an on-site or off-site location does not interrupt the operation of the system, in most cases. It is interrupted for hardware upgrades and may be interrupted for some software upgrades.

Meeting

Refer to the [“Meeting” on page 37](#) covered in [Chapter 3 – Unified Communications](#).

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 or by dialing an access code from a standard telephone.

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

The Mobility features are covered in [Chapter 3 – Unified Communications](#).

Multi User Administration in EMPA

Enterprise Manager is used to administer the database for IPedge system. The Role assigned to each User defines the level of access to the system database of each User. There are two types of administration roles.

- **Enterprise Manager Personal Administration (EMPA)** administrators have access only to their own telephone configuration. EMPA Super Users can administer their own telephone and the telephones of users assigned to them.
- **Enterprise Manager** administrators have access to the system database. Each role is defined as a list of permission items (access rights) that determine the user's access level in Enterprise Manager.

The IPedge Enterprise Manager has technician roles (Administrator) and telephone user (EMPA) roles defined by default which may not be changed. New roles may be added to create custom definitions.

EMPA User Roles

- **Normal User** - Normal Users can administer the following allowed items on their own telephones:
 - Settings – Name to Display, Do Not Disturb On or Off, Feature Button assignments
 - Call Forwarding – Setup call forwarding destination, timers, On/Off
 - Station Speed Dial – Assign Station Speed Dialing
 - System Speed Dialing – View only
 - System Info – Display (View only) system Flexible Access Code and Advisory Messages
 - Meeting – Meeting application configuration

- Messaging – Voice Mail setup
- User Guide – Display IPedge Telephone, Messaging, and Call Manager User Guide
- About – Display the Enterprise Manager software version and links to the EULA and OSS Attribution list.
- **Super User** – A Super User can administer the following allowed items on their own telephones and the allowed functions on the telephones in their ‘other EMPA users’ list.
 - Settings – Same as Normal User
 - Call Forwarding – Same as Normal User
 - Station Speed Dial – Same as Normal User
 - System Speed Dial – Same as Normal User
 - System Info – Same as Normal User
 - Account Code – Same as Normal User
 - Meeting – Same as Normal User
 - Messaging – Same as Normal User
 - Preference – Same as Normal User
 - User Guide – Same as Normal User
 - About – Same as Normal User
 - Change Directory Name Listing – Control over whether the user name appears in the system Directory.
 - Manage Other Account – When included, this User can administer the stations in the ‘Manage Other’ list.

The Super User can manage:

- The stations of all other types of EMPA Users
- Only the stations assigned by the System Administrator
- Only the features assigned in the Super User’s own account

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 Media Server has a total of fifteen (15) music sources plus Quiet Tone. The system administrator selects from these 15 internal WAV files on the IPedge/VIPedge system music sources, and quiet tone. Administrators may upload their own WAV files to the system.

IPedge systems with 1.5.1 or later software have the “Random” option for Music on Hold (MOH) that allows callers, when placed on hold to start listening from a different location within the recorded music on hold file every time they are put on hold, rather than always starting at the beginning. IPedge default to always starting at the beginning. This selection can be chosen when uploading the MOH file. Starting at the beginning allows the record-

ing to begin with “thank you for waiting.” Random, similar to the continuous loop, allows the recording to contain a series of messages, and callers can hear different parts of the messages when they are put on hold at different times. The file will play to the end, then “loop” to the beginning of the file.

Random start is not available on VIPedge systems.

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 a Single Line Telephone.

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 or DSS button to make a call. It also applies to DSS buttons on DSS consoles or add-on modules associated with the Call Forward Overriding telephone.

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 are 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 defined separately from regular paging. An emergency page may be answered in the same manner as a regular page.

External Paging

Paging over external speakers requires a SIP based paging device. The external paging device is treated as a SIP station by the IPedge system.

For external paging solutions refer to the IPedge Accessories manual.

Power Failure Protection

Reserve Power Battery Backup

An Uninterruptible Power Supply (UPS) is required for power backup on an IPedge system. The UPS is similar to the ones used for Computer systems and Networking equipment.

See [“Power Considerations” on page 110](#).

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.

Remote Update

Remote Program update is administered using Enterprise Manager to update the IPedge software remotely.

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 Ringing and Distinctive Ringing.

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 IPedge system to get PRI-like services from an Internet Telephony Service Provider using SIP.

A SIP Trunk allows an IPedge system to connect internal voice and private data traffic to the outside public network (PSTN and public data) via IP.

Media Relay Server Overview

In a NAT environment use the NAT Traversal capability of the IPedge server (Release 1.3 and later) or a SIP ALG router along with the IPedge Media Relay Server (MRS).

The MRS is configured by defining the Public IP address of the IPedge server and the port range to be used for calls. Each call requires two UDP ports for the audio streams (one port RTP, one port RTCP).

Refer to the IPedge Install manual for Media Relay Server setup instructions.

SIP Trunk NAT Traversal

In order to support SIP Trunking on R1.2 and earlier IPedge systems in a NAT environment, the router needs to support an enterprise grade SIP ALG (Application Layer Gateway). In Release 1.3 and later, the IPedge system can support SIP Trunking with routers that do not have a SIP ALG.

When used behind a NAT firewall that does not support a SIP ALG, the IPedge server can still be given a private IP address. The SIP Trunk NAT Traversal capability (Release 1.3 and later) along with the MRS will allow the IPedge server to:

- Use its internal Media Relay Server to route media packets between the WAN and the LAN and
- Apply the correct IP address to SIP signaling messages so that when they are sent out through a NAT firewall, the SIP Trunk service provider will be able to send responses to the correct IP address.

Within the NAT router, port forwarding rules will need to be configured, and a range of ports opened for the Media Relay Server.

When used with a NAT firewall that does support an enterprise grade SIP ALG (such as the Cisco ASA5500 product line) the SIP ALG feature needs to be enabled. In this configuration the media packets will be routed directly from the LAN to the WAN and mid-call survivability of a PSTN call is possible.

Security

The IPedge system is built using Red Hat Linux® operating system. Linux has a secure file structure making it less susceptible to viruses when compared to other popular operating systems. Furthermore, there are fewer viruses targeting the Linux operating system.

The IPedge system is a purpose built solution with pre-installed applications. Toshiba has taken numerous steps to inhibit virus attacks on the IPedge system.

- Unused software components have been disabled or removed.
- Unused IP ports have been closed.
- Industry standard vulnerability scans are run on IPedge and its software components are updated as required.

Provided that Toshiba's IPedge system is used explicitly as it is intended and as described in Toshiba's documentation, the IPedge is a minimum security risk for virus attacks. There are no guarantees against all threats that may arise in the future. Therefore it is still necessary for each customer to install the IPedge server using good security practices. In light of the IPedge server's security architecture, Toshiba does not require the use of antivirus software on the system.

Specified Caller ID

The Specified Caller ID feature allows applications, such as Messaging and uMobility, to send the phone 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 IPedge Messaging routes the call to the cell phone. Likewise, uMobility users can see the true Caller ID when they receive the call on their smart phone.

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. IPedge/VIPedge systems 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 **CnflTrn**, 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) may 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 call coverage situations are typical for hunt group applications. Hunt group members can remove themselves from the group by placing their station into 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 specific series. 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 may 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 Automatic Camp On, 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 DNs, camped on calls queue onto all DNs in the group and will connect to any DN in the group that becomes available. When not using Pilot DNs, 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.

Station Message Detail Recording (SMDR)

For each incoming, outgoing or tandem call, the IPedge system can generate a record that includes details of the call, including the originating station or trunk, the start time of the call, its duration, authorization codes, etc. If a station user dials “911,” the IPedge system will also generate a record at the beginning of the call as part of its internal notification that an emergency call is in progress. SMDR requires a connected Call Accounting system connected by TCP.

Survivability

Survivability is based on the capability of the IPT to send Register messages to two IPedge servers. This makes it possible to provide telephone service in the event that an IPedge server or the link to that server goes down.

The IPT can register to the backup survivability server (Secondary Server) automatically when sever goes down or the linkage between server and IPT is disconnected. Survivability is provided by the following.

- IPT speech path Survivability — If the user is on a peer-to-peer call when the failure occurs the fail-over will occur after the call is finished.
- Fail-over — The IPT can fail-over to the Secondary server by sending a Register message to the Secondary server.

- Fail-back — When the primary server becomes operational, the IPT phones will automatically re-register to the primary) when each IPT has no active call and is not in use by the station user.
- Telephony Service Survivability — The IPedge system is survivability-aware and will automatically select the appropriate destination server according to IPT Fail-over or Fail back status without complex system data programming or user setting.

Enterprise Manager Survivability

In a multi-node system Enterprise Manager Survivability allows a member server to become the primary server and Enterprise Manager on that server to take the place of the failed or off line primary server.

Only IPedge EC and EM systems can be primary servers. The server configured to be the redundant or fail-over primary server must also be an EC or EM server.

Call Manager Survivability

The survivability feature enables Call Manager to connect to a member server when the primary server is down.

Both VoIP Option/SoftIPT and IPT relies on its connections to the Call Processing module to determine whether or not to switch over, and Call Manager relies on the connection to Net Server to determine whether or not to switch over.

Note If a component failure such as Net Server module or Call Processing module shutdown takes place instead of the complete server failure, Call Manager and the phone may connect to different IPedge systems. In that case, the user can manually override the survivability to reconnect to the same IPedge system.

Call Manager survivability is supported for the IPedge built-in Net Server. ACD or Unifier based Net Server is not supported.

Messaging Survivability

The IPedge Messaging application can be licensed and configured with a feature called Direct Cluster Networking (DCN). DCN allows the joining of two or more IPedge systems (individually referred to as a Node) into a cluster. These clusters act in unison to maintain the integrity of the messaging database of the entire network. Each node that is configured into the cluster has a copy of the database of the other participating nodes. If one node fails, then when IPedge telephones register into another IPedge system, that is a node participant, all of that user's greetings and messages are available.

Nodes can be geographically distributed in various configurations. Each node contains the complete database for the entire cluster, and the Messaging application residing on each node only uses the local copy of the database. Each node is identified by a NODE ID. In addition all files, including system greetings, user greetings and messages can be replicated to all nodes (standard cluster) or replicated to a designated subset of nodes (hybrid cluster), depending on cluster size and network capability. Note that all of the nodes in a cluster must be running the same version of the IPedge system software.

System Fault Finding and Diagnostics

IPedge can detect problems in the system. These conditions can be detected, alerted, logged, and traced. IPedge includes many useful diagnostic tools.

Alarm Indication of System Faults

Visual Alarms are presented to Enterprise Manager.

Fault Detection and Error Logs

The IPedge system detects and logs abnormalities that it encounters during operation. All error and trace logs are stored on the hard drive and are monitored by Enterprise Manager. Examples are trunk failure detection and auto busy-out, IP telephone channel failure detection and auto busy-out plus error log, etc.

Event and System Administration Logs

Events such as station buttons pushed or lines accessed are stored in an Event Log. All actions made by the System Administration user are logged. Both logs may be called up at a later time.

Automatic Fault Recovery

The system can automatically correct certain conditions detected during operation. This enables the system to continue operating normally without requiring correction.

Backup and Restore

The customer database can be backed up and restored automatically or manually scheduled. The customer database can be moved to a network drive or can be moved to another location using FTP. The backup and restore functions can be performed locally or remotely.

Maintenance and Administration

The Enterprise Manager terminal can be connected directly to the IPedge system or via the customer's LAN as well as remotely over the Internet over the public network.

Software Upgrade

A regular IPedge system software upgrade can be performed. You can upgrade the Operating System without affecting your customer database.

Tenant Services

This feature enables the IPedge system to provide separate service to multiple companies or departments (tenants). PDNs, DIDs and incoming line groups (ILGs) can be assigned to one through eight tenants. Each tenant can have different attendant or night bell assignments for day-of-the-week schedules using Day 1, Day 2, or Day 3 modes. This feature is enabled through system programming.

Traffic Measurement

Technicians and System Administrators can monitor the effectiveness of the system resources for proper traffic balance. These traffic statistics are necessary for the system administrator to both monitor the effectiveness of the system and determine whether the system has enough resources or improper traffic balance. No additional hardware is needed to support Traffic Measurement.

Traffic Measurement setup and reporting is done using Enterprise Manager. Approximately five days of Traffic Measurement reports can be stored.

Traffic Reports

New traffic reports include outgoing and incoming trunk group usage, “all circuits busy” reporting and media server resources. The reports are stored on the IPedge server and can be downloaded through Enterprise Manager or sent to a remote device. Traffic reporting is set up based on day of week and time of day. Reports are easy to read, time-stamped files that are generated and sent out hourly. No additional software application is required.

Reports include traffic intensity on incoming/outgoing line groups and system resources such as Media Server resources. Reports can measure traffic in Centi Call Seconds (CCS) or Erlangs. All circuits busy and Abandoned calls are also reported. In IPedge, media resource usage, consumption of media resource channels for conferences and general call processing, such as paging, service tone generation, MOH will be reported.

Transfer Direct To Voice Mail

Transfer Direct To Voice Mail is the ability to redirect a connected call to new destination. The IPedge system provides three means of transferring a call and three means of terminating transferred calls, 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 regular 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 in the *IPedge* system. 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 SMDI VM integration and does not require Toshiba proprietary VM integration.

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

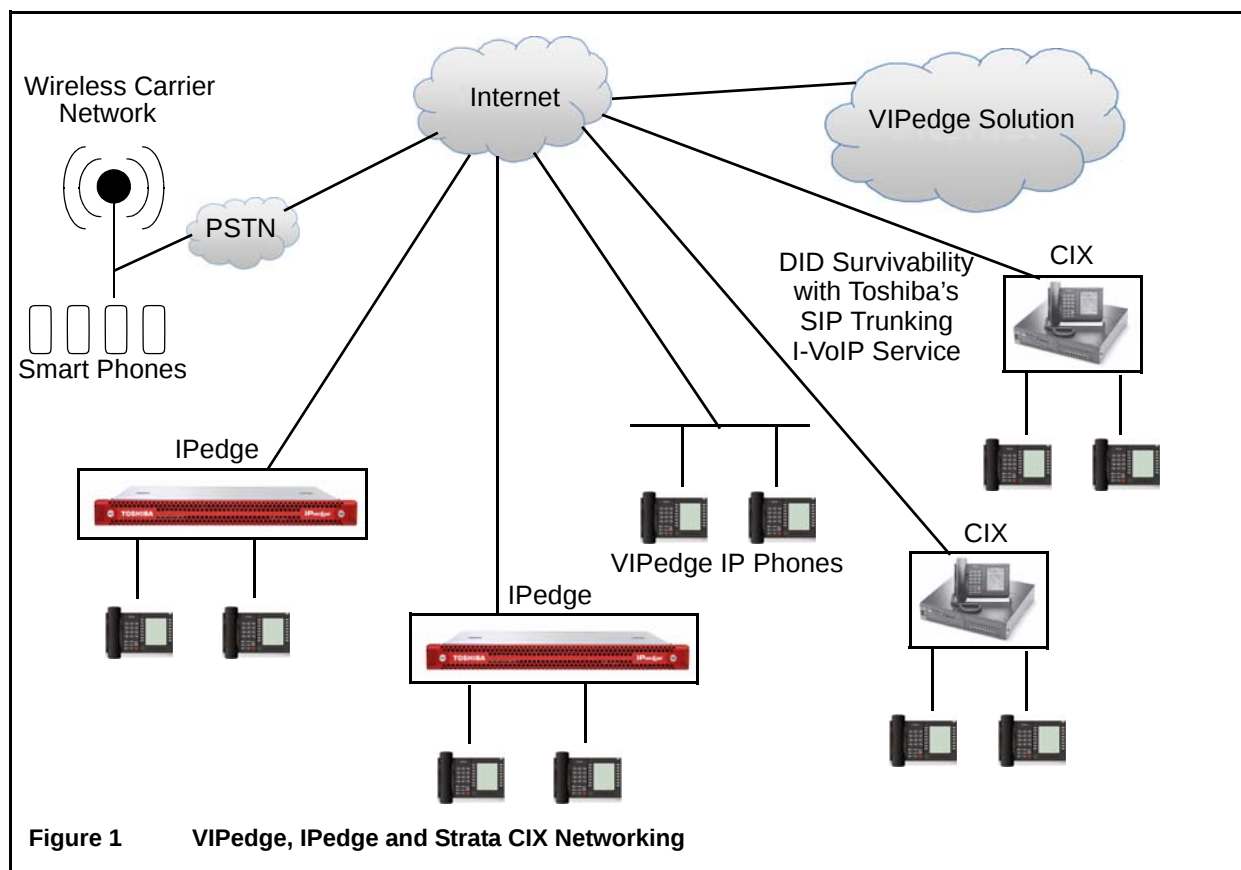
Refer to the [Chapter 3 – Unified Communications](#).

VIPedge and IPedge Networking

Multiple IPedge systems can be networked with the VIPedge Cloud-based Business Telephone Solution.

Multiple IPedge and Strata CIX systems can be networked with VIPedge.

- Up to 128 systems can be networked together including one VIPedge system and multiple IPedge and CIX systems.
- VIPedge Net channels are included free of charge with service. Separate IPedge Net licenses are still required, same as current offerings.
- Premise based Call Manager clients do not currently intercommunicate with cloud based Call Manager clients. Only premise-to-premise Call Manager intercommunication and cloud-to-cloud Call Manager intercommunication supported. Road map to premise and cloud Call Manager intercommunication in June 2014.



VLAN Tagging

The IPedge system supports 802.1Q Virtual Local Area Network (VLAN) technologies. For sites that have thousands of IP devices, VLANs may 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 (IPT) and data port may be programmed manually using the base station or remotely via Enterprise Manager. There are no settings to set on the IPedge server, however, ensure that the data switch port connected to the IPedge server is configured to be in the same VLAN ID as the IPTs. For IPT configuration, broadcast mode is not recommended for large deployments (100+ phones).

With or without VLANs, 802.1p and Diffserv protocols may be used to provide Quality of Service for voice by allowing voice packets to be prioritized over data packets.

To provide additional prioritization for voice services, it is possible to enable 802.1p in conjunction with 802.1Q (VLANs). This is currently a system wide setting in IPedge.

However, this will enable 802.1p on remote phones requiring the switches are all locations where remote phones are deployed to support 802.1p.

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 one or both protocols.

Reasons 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 may 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 IPedge 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. Each IPedge system ships with one department. Additional departments can be enabled with licensing.

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).

Fax

All IPedge system models support T.38 communication when the end-to-end communications are entirely SIP. Fax features are licensed on a user level, not a system level basis. An Advanced User license is required for a user to take advantage of the fax mail and personal fax features.

Fax from Desktop

Provides the ability to send faxes from the mailbox owner's desktop.

Fax Format

Fax documents sent from the mailbox owner's desktop may be formatted as PDF, TIF or DCX.

Fax Log

A web-based report displays the mailbox owner's outbound faxes. The fax log includes date, time, status of an outbound fax, fax destination, account and billing codes.

Fax-on-Demand

This component allows incoming callers to access a library of documents and select a specific fax document to be faxed to them. Fax on demand applications are created using the Messaging Script mailbox. A Script license is required for this feature.

Fax Mail

Fax mail allows a mailbox owner to receive faxes in his voice mailbox and view them via unified messaging (an email attachment) or use the telephone interface to re-route the incoming fax to a physical fax machine.

Fax Queue

A web-based report displays the mailbox owner's outbound faxes currently queued for transmission.

Fax Settings

The mailbox owner may set personal outbound fax settings, such as number of times to retry fax delivery based on busy or no answer and how long to wait between each try. Each fax user can transmit its own name and number (CSID) on outbound fax.

Incoming Fax DID

For inbound fax messages, a DID number may be associated with the mailbox. An incoming fax to this number will automatically trigger a fax tone and the fax will be stored in the mailbox.

Incoming Fax Target

Faxes may be re-routed from an incoming mailbox to a secondary mailbox.

Personal Fax

With the use of a custom printer driver, Messaging allows users to send documents as faxes to remote locations, using the IPedge system. Just select the print option, as you would print a document, and choose the Messaging Fax printer. A web applet will be presented to accept addressing options and to add a fax cover page.

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 98](#)).

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. IPedge 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 in the IPedge system. 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.

Key Ahead

Bypass a voice prompt by selecting a key press.

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 IPedge 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 from 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

IPedge 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 IPedge 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 IPedge 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

IPedge 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.

Fax-to-Email

Fax-to-email allows the mailbox owner to review fax information directly from the email inbox (including fax sender and number of pages), view fax messages onscreen with any TIFF or PDF image viewer and forward fax messages to any email address directly from the email inbox.

Print Emails to Fax

Forward emails to a fax machine so that they may be printed.

Redirect Fax Messages

Redirect fax messages from the voicemail box to any fax machine when the email inbox is not available for fax viewing.

Integration with Email Clients

IPedge 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. Messages are synchronized with IPedge Messaging.

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.

Msync

Msync is actually a Microsoft® Exchange Web Services connector, which allows the IPedge system to access a Microsoft Exchange Server, in order to manage IPedge Messaging users' voice and fax messages within the email message store without requiring them to have to enter and maintain their email log on credentials within Messaging. Msync uses a service account to log into the Active Directory, and therefore does not require individual users to enter their mail credentials.

Msync provides the following capabilities:

- Stores new voice / fax messages in a user's inbox as they are received in IPedge Messaging.
- Change message status to read in the email client when it is saved through the Messaging Telephone User Interface (TUI).
- Deletes a message from the email inbox when it is deleted using the TUI.
- Changes the message status to saved on the TUI when a message is opened on the email client.
- Changes message status to delete on the TUI when a message is deleted from the inbox of the email client.
- Voice messages that are forwarded using email from one user to another will be received as voice messages in Messaging.
- Reading / Saving / Deleting email messages.

Enhanced Msync Feature

A feature provided by Enhanced Msync is Presence. Presence allows you to define a schedule based on Presence status in Microsoft Outlook. For this option to be effective, the system must have the Msync connector activated and connected to Microsoft Exchange. Available states are: Free, Busy, Out Of Office, and Out Of Office Assistant (Automatic replies).

Msync requires the minimum software requirement for the host Exchange server to be one of the following configurations:

- Microsoft Exchange 2007 SP1 (Running on Windows 2008 SP2 64-bit)
- Microsoft Exchange 2010 SP1 (Running on Windows 2008 R2 standard 64-bit)
- Standard Msync requires IPedge Messaging versions: 10.5.4.11 and higher
- Enhanced Msync requires IPedge Messaging versions: 10.5.4.24 and higher.

Multi-site Networking

VPIM

Using the industry standard VPIM protocol, mailbox owners using Messaging can transparently send and reply to messages from mailbox users located on dissimilar, but VPIM-enabled voicemail systems.

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 IPedge 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 by IPedge 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

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.

Messaging Survivability

The IPedge Messaging application can be licensed and configured with a feature called Direct Cluster Networking (DCN). DCN allows joining the Messaging application of two or more IPedge systems (individually referred to as Nodes) into a cluster. These clusters act in unison to maintain the integrity of the messaging database of the entire network. Each node that is configured into the cluster has a copy of the database of the other participating nodes. If one node fails, then when IPedge telephones register into another IPedge system, that is a node participant, all of that user’s greetings and messages are available.

Nodes can be geographically distributed in various configurations. Each node contains the complete database for the entire cluster, and the Messaging application residing on each node only uses the local copy of the database. Each node is identified by a Node ID. In addition all files, including system greetings, user greetings and messages can be replicated to all nodes (standard cluster) or replicated to a designated subset of nodes (hybrid cluster), depending on cluster size and network capability.

Functional Considerations

Although DCN provides a robust voice mail survivability solution, there are some functional considerations that need to be understood and communicated to customer users.

- If a telephone has a Message Waiting Indicator (MWI) illuminated and the system that supports that telephone fails, the MWI will not be reinstated until another new message is received. The telephone survives over to another system that is in the cluster and has its mailbox intact, but the Message Waiting light will not light until a new message is received.
- The voice mail hunt group pilot number should be the same on the different nodes. If the voice mail hunt group pilot number is different on the different nodes incorrect voice mail forwarding after a node failure will occur. For example, station 201 on IPedge Node 11 (DCN Node 1) is set to system call forward to voicemail hunt group pilot 300. The DN on IPedge Node 12 (DCN Node 2) are set to system call forward to voicemail hunt group pilot 400. If IPedge Node 11 fails and station 201 re-registers with IPedge Node 12, station 201 will not properly forward to voicemail when a call is presented to it.

Note The Messaging application must be running on every IPedge system that will run DCN.

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.

Appendix – Specifications

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

- [Operating Environment.](#)
- [Power Considerations](#)
- [Station Dimensions](#)
- [IP Telephone Power Consumption](#)
- [IPedge Component Compatibility](#)
- [System Tones](#)
- [System Tones](#)
- [IPedge Net and IP Telephone Bandwidth Requirements](#)
- [IPedge Net and IP Telephone Bandwidth Requirements](#)
- [Capacities](#)

For further details, refer to the *IPedge I&M Manual*.

Operating Environment.

Table 4 **Operating Environment**

	EC Server	EM Server	EP Server
Operating Temperature	50°F ~ 95°F; 10°C ~ 35°C	50°F ~ 95°F; 10°C ~ 35°C	50°F ~ 95°F; 10°C ~ 35°C
Operating Humidity	20% ~ 80% (non condensing)	20% ~ 80% (non condensing)	20% ~ 80% (non condensing)
Storage Temperature	-20 ~ +60°C	-20 ~ +60°C	-20 ~ +60°C
Power	100 ~ 240 VAC; 50 ~ 60 Hz; 1.2 Amp at 120 VAC; 130 Watts Inrush Current: 11.5 Amp	100 ~ 240 VAC; 50 ~ 60 Hz, 2.1 Amp at 120 VAC; 250 Watts Inrush Current: 10.4 Amp The IPedge EM server has two redundant, hot-swap power supplies. The server can run indefinitely on one supply.	100 ~ 240 VAC; 50 ~ 60 Hz; In-rush: Maximum 10 Amp (cold start) Peak Load: 0.4 Amp
Heat	785 BTUs	778 BTU/hour; 867 BTUs max.	106 BTUs

Power Considerations

The IPedge server should have a dedicated AC power circuit. The specific input voltage and current requirements for each server is listed the specifications for each model.

UPS Recommendation

Toshiba recommends an Uninterruptible Power Supply (UPS) with power conditioning for the IPedge server. The recommended UPS from ONEAC are shown in the [Table 5](#) below. The UPS shown in the table include power conditioning.

Table 5 IPedge System Power Conditioner and UPS

		IPedge Server		
		EC	EM	EP
Power Conditioner		PC180A-S2S	PC360A-S4S	PC075A-S2S
Battery Backup Time	30 Minutes	ON700XAU-SN	ON700XAU-SN	ONE254AG-SE
	1 Hour	ON700XAU-SN1	ON700XAU-SN1	ONE254AG-SE
	2 Hours	ON700XAU-SN1	ON700XAU-SN1	ONE604AG-SE
	4 Hours	ON700XAU-SN1	ON700XAU-SN2	ONE300XAU-W-SV1
	8 Hours	ON700XAU-SN2	ON700XAU-SN4	ONE300XAU-W-SV1

Station Dimensions

Dimensions for the 5000-series, IP telephones and related equipment are listed in [Table 6](#).

Table 6 Station Dimensions

Device	Height		Width		Depth	
	Inches	mm	Inches	mm	Inches	mm
IP5022-SD, IP5122-SD, IP5122-SDC, IP5132-SD, IP5522-SD, IP5622-SD, IP5131-SDL [15-degree tilt], IP5531-SDL, IP5631-SDL	5.1	129	10.16	258	6.10	155
IP Direct Station Selection (DSS) Console - IDM5060 [15-degree tilt]	4.0	102	10.16	258	6.10	155
Add-on Module (KM5020, LM5110) [15-degree tilt]	4.0	102	3.54	90	6.10	155
Direct Station Selection (DSS) Console (DDM5060) [15-degree tilt]			10.16	258		

IP Telephone Power Consumption

The power consumption for the IP5000-series telephones and the Add-on modules is shown in [Table 7](#). Use this information to calculate the Power over Ethernet (PoE) requirements and UPS capacity.

Table 7 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					
IP5122-SD	none	--	7.4	0.15	6.2	0.13	0
IP5122-SDC	none	--	7.4	0.15	6.2	0.13	0
IP5132-SD	none	--	7.4	0.15	6.2	0.13	0
IP5131-SDL	none	--	7.4	0.15	6.2	0.13	0
IP51xx +	IDM5060	3	10.3	0.21	8.6	0.18	0
IP51xx +	IDM5060	2	9.4	0.20	7.8	0.16	0
IP51xx +	IDM5060	1	8.4	0.18	7.0	0.15	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
IP51xx +	KM5020	2	8.9	0.19	7.4	0.15	0
IP51xx +	KM5020	1	8.2	0.17	6.8	0.14	0
IP5622-SD	none	--	3.7	0.08	3.0	0.06	1
IP5631-SDL	none	--	4.1	0.08	3.3	0.07	2
IP5631-SDL	IDM5060	3	6.4	0.13	5.4	0.11	2
IP5631-SDL	IDM5060	2	5.6	0.12	4.7	0.10	2
IP5631-SDL	IDM5060	1	4.8	0.10	4.0	0.08	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
IP5631-SDL	KM5020	2	5.2	0.11	4.3	0.09	2
IP5631-SDL	KM5020	1	4.6	0.10	3.9	0.08	2
IP5531-SDL	none	--	3.6	0.08	3.0	0.06	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

IPedge Component Compatibility

The IPedge system supports all types of Toshiba IP and third party provided SIP telephones, it provides the configuration flexibility to build the communications system you need, in addition to the investment protection from re-using devices from other Strata systems. It's a unified communications environment that supports many types of client devices.

Using the IPedge system, your telephone can have peer-to-peer IP communication and Strata Media Application Server compatibility, and even use the Strata CIX systems as gateways.

Table 8 Component Compatibility

System Software and Features	EC Server	EM Server	EP Server
IPedge Software and features	X	X	X
Linux Operating System	X	X	X
Contact Center	EC Server	EM Server	EP Server
ACD	X	X	X
Insight Call Center Reporting	X	X	X
TASKE Call Center Reporting	X	X	X
Administration	EC Server	EM Server	EP Server
Enterprise Manager [Browser-based Unified Admin]	X	X	X
Enterprise Manager Centralized Management Primary Server	X	X	NA
Stations and Terminal Equipment	EC Server	EM Server	EP Server
IP5000-series telephones	X	X	X
IP Attendant Console	X	X	X
Wireless IP Telephones	X	X	X
Simultaneous Ringing Desk / Mobile phones	X	X	X
SIP (3 rd party) IP telephones	X	X	X
Messaging and Collaboration	EC Server	EM Server	EP Server
Messaging (integrated)	Built-in	Built-in	Built-in
Mobility	EC Server	EM Server	EP Server
SoftIPT softphone client for Laptop	X	X	X
uMobility	X	X	X
Unified Communications	EC Server	EM Server	EP Server
Call Manager	X	X	X
Gateways	EC Server	EM Server	EP Server
Strata CIX system (with IPedge Net)	X	X	X
Audiocodes gateways	X	X	X
Epygi Gateways	X	X	X
Other Telephone Systems	EC Server	EM Server	EP Server
Strata CIX system (with IPedge Net)	X	X	X

System Tones

Tones which can be heard from speaker or handset are described in [Table 9](#).

Table 9 Call Progress Tones

Tone Name	Conditions	Ringing 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.
External Call Waiting Tone for Standard Telephone	An external call is waiting. This tone is sent to the receive party only.	1209 Hz, 2 bursts of 0.16 sec. apart 0.16 sec., twice, 3 sec. apart.
Internal Call Waiting Tone for Standard Telephone	An internal call is waiting or somebody is listening to (monitoring) the conversation.	1209 Hz, 2 bursts of 0.5 sec. On, apart 3.0 sec.

Ring tones are described, along with their cadences in [Table 10](#). Due to the limitation in the tone generation algorithm, the listed tone duration is slightly different from the actual one.

Table 10 Ring Tones

Tone Name	Description	Ringing 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
External/Internal Ring for Standard Telephones	Internal and External Ringing Cadence: For Release 1.3 and higher, two types of ringing cadences can be selected in system programming.	Ringing Type1: External Ring: 20Hz, 1sec. On - 3 sec. Off Internal Ring: 20Hz, 0.4 sec. On - 0.2 sec. Off - 0.4 sec. On - 3 sec. Off Recall: 20Hz, 1sec. On - 1sec. Off
		Ringing Type 2: Same Ringing Cadence as DK. External Ring: 20Hz, 0.4 sec. On - 0.2 sec. Off - 0.4 sec. On - 3 sec. Off Internal Ring: 20Hz, 1 sec. On - 3 sec. Off Recall: 20Hz, 1 sec. On - 3 sec. Off
Recall	A call is returned & needs to be answered.	2 kHz interrupted at 10 Hz, 1 sec. On, 1 sec. Off, repeat.
Recall for Standard Telephone	A call is returned & needs to be answered.	20 Hz, 1 sec. On, 1 sec. Off, repeat.
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 11](#).

Table 11 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.

IPedge Net and 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 IPedge Net channels and IP telephones depends heavily on the LAN parameters as shown in [Table 12](#). This table shows the amount of bandwidth required for each IPedge Net IP voice call (without data traffic) based on the interval and the CODEC.

Table 12 IPedge Net IP and 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 13](#) 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 13 IPedge Net 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 14](#).

Table 14 IPedge Net IP Class Definitions

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

[Table 15](#) shows the amount of bandwidth required for setting up and tearing down calls independent of the amount of voice traffic.

Table 15 IPedge Net 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.

Capacities

The following tables contain IPedge capacities.

Table 16 Station/Peripherals System Capacities

Stations	EC Server	EM Server	EP Server
PC Attendant consoles	2	6	2
IP5000-series stations / SIP stations	200 per System ¹	1,000 per System ¹	40
DSS Consoles	3 per Station 24 per System	3 per Station 24 per System	3 per Station 5 per System
Add-on modules - LM5110, KM5020	160	800	80
Simultaneous calls	148	576	40

1. Capacity could be reduced by the addition of Messaging and or Meeting applications.

Table 17 Trunk Capacities

Trunks	EC Server	EM Server	EP Server
IPedge Net IP channels	96	440	20
SIP Trunk channels	96	440	20
Total IPedge Net IP Channels, SIP Trunk Channels, Analog, T1 and ISDN trunk channels.	96	440	20
Channel Groups ¹	2	2	2

1. One group for SIP trunks and one group for IPedge Net.

Table 18 IP Telephone Station Buttons

Station Buttons per System	EC Server	EM Server	EP Server
Call Forward, Personal CF Buttons	160	1,000	40
Caller ID (CLID) button (IP telephone only)	160	1,000	40
Group CO Line (GCO) Buttons ¹	96	440	20
Pooled CO Line Buttons ²	50	220	20
CO Group and Pooled Line Buttons ²	440	440	40
Flexible Telephone Buttons	48,000	48,000	3,720
Line and DN Buttons in use at the same time	6,000	6,000	3,720
Message Waiting Registration (DNs with MW)	1,344	1,344	1,344
Multiple Appearances of DNs on Telephones	27,000	27,000	3,720
Night Transfer Buttons	192	192	40
One Touch Buttons	24,000	24,000	3,720
Primary Directory Numbers [PDNs] per system	200	1,000	56
Phantom Directory Numbers [PhDNs] per system	4,000	4,000	3,720
[PhDNs] with Message Waiting Indication LED	192	192	192

1. This is the total number of all GCO or Pooled Line Buttons allowed in a system. Example: If the GCO1 button appears on 10 telephones, it counts as 10 buttons.

Table 19 System Feature Capacities

Features	EC Server	EM Server	EP Server
Pilot DNs	1,024	1,024	256
Advisory LCD Messages (Set on a Telephone)	1	1	1
Advisory LCD Messages Lists (per System)	10	10	10
Attendant Groups	1	1	1
Call Accounting SMDR Interface ¹	1	1	1
Call Forward, System CF Patterns	48	48	48
Call Park Orbits (General)	96	96	96
Call Park Orbits (Individual)	576	576	576
Minimum / Maximum Caller ID per Station	Min:0, Step:5	Min:0, Step:5	Min:0, Step:5
Max means max records of a station	Max:200	Max:200	Max:200
Maximum number of Stations that can have Caller ID/ ANI/DNIS Numbers stored (Call History records)	600	600	600
	3,000	3,000	3,000
CO Line Groups - Incoming Line Groups (ILG)	220	220	220
CO Line Groups - Outgoing Line Groups (OLG)	220	220	220
Outgoing Line Groups (OLG) Members per system (Trunks + ISDN Line Service Index)	660	660	660
Conference Channels (Depends on system configuration)	24	96	12
Conferencing (three-parties simultaneously) ²	installed channels/3	installed channels/3	installed channels/3
Conferencing (eight-parties simultaneously) ²	installed channels/8	installed channels/8	installed channels/8
Conference Party types (up to 8 total lines + stations) ²	8	8	8
Two-CO Line simultaneous Connection ² (Two party only, no telephone or VM channel)	48	220	10
DID Numbers for Calling Number ID/system	1,500	1,500	1,500
DNIS/DID Incoming Numbers (1–7)	3,000	3,000	3,000
DNIS/DID Routing Destination Numbers (1–6 digits) ³	9,000	9,000	9,000
DNIS/DID Routing Destination Numbers (7–32 digits) ³	1,500	1,500	1,500
Network DNs	6,000	6,000	6,000
Uniform Numbering Plan	30,000	30,000	30,000
E911 Groups	128	128	56
Emergency Call Groups	128	128	56
Hunt Groups (Serial/Circular/Distributed combined)	1,100	1,100	1,100
Hunt Group Size (DNs per group)	160	1,000	56
Hunt Group Stations (per system)	5,000	5,000	56
ISDN Line Service Indexes	220	220	220
Multiple Call Ring Group	96	96	96
Off-hook Call Announce to Telephone Speakers ⁴	200	1,000	40
Page Groups (Phones with or without External Zones)	24	24	24
Paging – (Group Page – simultaneous stations paged) ⁵	120	120	12
Pickup Groups	48	48	48
Ring Tones (External Call Ring Tones for IP telephones)	10	10	10

Table 19 System Feature Capacities*(continued)*

Features	EC Server	EM Server	EP Server
Ring Tones (Internal Call Ring Tones for IP telephones)	10	10	10
Speed Dial - Station SD numbers per system ⁶	8,400	8,400	8,400
Speed Dial - System SD numbers per system	800	800	800
Tenants	8	8	8
Destination Restriction Level (DRL) Classes	16	16	16
Verified Account Codes	4,000	4,000	4,000
Voice Mail SMDI Interface ¹	1	1	1
SIP URI per system	3,000	3,000	3,000
SIP Trunk service Index	128	128	128
LCR Exception Table Size	2,500	2,500	2,500
MAX digit number for LCR Route Plans	19	19	19
MAX Number of LCR Route Plans	128	128	128
CSTA Device Monitors	2,304	2,304	56
CSTA Call Monitors	560	560	56

1. SMDI and SMDR require a LAN interface.
2. Conference channels are used dynamically, so the maximum number of simultaneous conferences is affected by the number of conference members in each conference. The total number of members in simultaneous conferences cannot exceed the total number of conference channels. The total number of conference channels available is configured through Enterprise Manager, and each conference channel requires one Media Server Resource. Each conference can have up to eight members. Third party SIP endpoints cannot be the originator of a conference call.
3. 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.
4. 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).
5. The maximum number of paged stations is limited to the lesser of the number of Media Server generic channels available at the time of paging and to this maximum.
6. Up to 100 Station SD numbers, allocated in increments of 10, can be programmed per station.

Application Capacities

Table 20 Enterprise Manager

	EC Server	EM Server	EP Server
Enterprise Manager Simultaneous Sessions	16	32	4
Web Based Station Admin Simultaneous Sessions	64	128	4

Table 21 Media Server

	EC Server	EM Server	EP Server
Resources	174	480	22

Table 22 Meeting

	EC Server	EM Server	EP Server
Audio Channels	24	48	4
Web Sessions	24	48	4
Video Sessions	24	48	4
Conference Record	4	8	1

Table 23 Call Manager

	EC Server	EM Server	EP Server
Users with Call Manager	200	800	40

Table 24 Messaging

	EC Server	EM Server	EP Server
Departments	999	999	999
Mailboxes (basic or UM)	5,000	10,000	1,000
Script Mailboxes	1,000	1,000	1,000
Simultaneous Calls	32	80	8
Hours of Storage	4,000 hours	7,000 hours	4,000 hours

Mean Time Between Failures (MTBF)

Table 25 MTBF

	EC Server	EM Server		EP Server
		I-EM-1A	I-EM-1B	
MTBF ¹	4.02 years	2.71 years	2.60 years	9.95 years
MTBF ²	4.29 years	19.45 years	17.97 years	NA

1. I-EM-1A and I-EM-1B refer to the IPedge EM server with RAID1 and RAID5 respectively. The calculated value is based on any failure even though there are redundant components.
2. The IPedge EC server refers to I-EC-1A with RAID1 option installed. The calculated value is based on at least one component of each redundant system continuing to operate.

Device Monitor Capacities for IPedge Systems

Applications including Strata ACD, Call Manager, Tracer, Taske, and System TAPI send requests to the IPedge system to monitor the status of the telephones using the respective applications. These requests are sent over the CSTA ethernet link connecting the application and the IPedge system. These requests can produce a heavy load on the IPedge and LAN so there is a limit to the number of telephones and devices that can be setup for monitoring and how many can be active on a monitored call simultaneously. The capacity limits and a table listing how the telephone and device capacities are counted is provided below:

CSTA Device Monitor Capacity Limits

The limits below apply to the IPedge EC and EM servers.

- Total number of devices that can be monitored: 1152
- Total number of simultaneous device monitor calls: 560

Table 26 Applications using CSTA Device Monitors

Device Category		Number of CSTA device monitors required
1	ACD Agent or Supervisor only.	1 CSTA device monitor per agent or supervisor.
2	ACD Agent or Supervisor with Call Manager and/or Tracer or both.	1 CSTA device monitor per agent or supervisor.
3	Normal User with Call Manager and/or Tracer or both.	1 CSTA device monitor per user.
4	ACD Groups.	1 CSTA device monitor per group.
5	ACD Voice Assistant ports.	1 CSTA device monitor per port.
6	Extensions to be monitored by Call Manager or Taske.	1 CSTA device monitor each.
7	Attendant Consoles	1 CSTA device monitor per console.
8	System TAPI Service Provider application.	1 CSTA device monitor per TSP application user.
Note The total CSTA Device Monitors used is equal to the sum of the devices in each Device Category.		

IPedge Software License Requirements

The following table describes the content and use for each IPedge license.

Table 27 IPedge License Part Numbers

Part Number	Description
I-EC-1A	IPedge EC model rack mount server. Factory equipped with single 250GB SATA hard drive, 4GB RAM, and all the necessary software to support IPedge features. Mounting rails required, see the 4-post (I-EC-RL4-1A) or 2-post (I-EC-RL2-1A) rail kits sold separately. Optional RAID1 kit (I-EC-RAID1-KIT) includes second 250GB hard drive, factory installed in a drive drawer. Requires I-SYS-EC or I-SYS-EC-DSCNT license.
I-EC-BZL-1A	Optional custom red front bezel with Toshiba and IPedge logos for I-EC-1A server. Provides a sleek look and secures accessibility. Includes two keys. (Buy extra bezel keys using the I-BEZEL-KEY part number).
I-EC-HDD ¹	Spare 160GB SATA hard drive for IPedge model I-EC-1A servers.
I-EC-PWR SUPPLY	Spare power supply for IPedge model I-EC-1A server.
I-EC-RL2-1A	Two post rail kit to mount IPedge server model I-EC-1A server in a two post server rack.
I-EC-RL4-1A	Four post rail kit to mount IPedge server model I-EC-1A server in a four post server rack.
I-EM-1A	IPedge EM model rack mount server. Factory equipped with a Linux operating system, two 300GB SAS hard drives in RAID1 configuration, 12GB RAM, dual redundant power supplies, and all the necessary software to support IPedge features. System ships with one 4-post rail kit. Requires I-SYS-EM license.
I-EM-1B	IPedge EM model rack mount server. Factory equipped with a Linux operating system, four 300GB SAS hard drives in RAID5 configuration, 12GB RAM, dual redundant power supplies, and all the necessary software to support IPedge features. System ships with one 4-post rail kit. Requires I-SYS-EM license.
I-EM-BZL-1A	Optional custom red front bezel with Toshiba and IPedge logos for I-EM-1A server. Provides a sleek look and secures accessibility. Includes two keys. (Buy extra bezel keys using the I-BEZEL-KEY part number).
I-EM-HDD	Spare 300GB SAS hard drive for IPedge model I-EM-1A and I-EM-1B servers.
I-EM-PWR SUPPLY	Spare power supply module for IPedge model I-EM-1A and I-EM-1B servers.
I-EC-HDD	Spare 160GB SATA hard drive for IPedge model I-EC-1A servers.
I-EC-PWR SUPPLY	Spare power supply for IPedge model I-EC-1A server.
I-EM-HDD	Spare 300GB SAS hard drive for IPedge model I-EM-1A servers.
I-EM-PWR SUPPLY	Spare power supply module for IPedge model I-EM-1A server.
I-BEZEL-KEY	One spare key for the bezel locks for I-EC and I-EM custom bezels. Key fits both locks. As needed for spares.
I-EP-1A	IPedge EP server with AC adaptor. Requires I-SYS-EP license.
I-EP-RL2-1A	Optional I-EP-1A mounting bracket, used for installing system in 2 or 4 post server racks.
License Part Numbers	
I-CM-1	IPedge Call Manager Advanced License - per user. VoIP voice plug-in is sold separately.
I-CM-STD1	IPedge Call Manager Standard version provides the screen based telephony and Outlook Contact dialing. Bundled with IPedge user license and not required to purchase.
I-CM-V1	IPedge Call Manager Voice add-on License (voice plug-in license to add VoIP per user). Requires I-CM-1
I-CM-STD1-SUR	Call Manager Standard survivability license.
I-CM-1-SUR	Call Manager Advanced survivability license
I-CM-V1-SUR	Call Manager VoIP option survivability license
I-CP-AUX	IPedge Auxiliary Channel license for each channel of Attendant Console, or ACD Voice Announcement channel.

Table 27 IPedge License Part Numbers (continued)

I-CP-CSTA	IPedge CSTA license - per application
I-CP-SOFTIPT	One required for each SoftIPT user on IPedge. Also requires a user license (I-CP-USR-EM or I-CP-USR-EC).
I-CP-IPENET	One required for each IPedge Net channel.
I-CP-TRUNK	IPedge Trunk License - per channel of SIP, PRI or Analog Gateway
I-CP-TRUNK -DISC	IPedge Call Processing Trunk License. This license enables one channel of SIP trunking at a reduced price, up to a maximum of 20 trunks on a system including what is in the base license. Maximums are EP:17, EC:8, EM:4. There after, IPedge EC and EM require I-CP-TRUNK.
I-CP-USR2	Provides IPedge license for 2 SIP Lines on a single device.
I-CP-USR-EC	IPedge User or Endpoint License - per endpoint on EC server
I-CP-USR-EM	IPedge User or Endpoint License - per endpoint on EM server
I-CP-USR-EP	IPedge user or endpoint license - per endpoint on EP server
I-CP-USR-SUR-EC	Survivable IPedge IP Phone or IP Application Port. Used only when creating a survivable station on EC Servers.
I-CP-USR-SUR-EM	Survivable IPedge IP Phone or IP Application Port. Used only when creating a survivable station on EM Servers.
I-CP-USR-SUR-EP	Survivable IPedge IP Phone or IP Application Port. Used only when creating a survivable station on EP Servers.
I-MS-BASE	IPedge Media Server Resource Base License. This is purchased as part of an I-SYS package and is not usually purchased separately.
I-MSG-BASE - EC	IPedge EC Messaging Base License - Base Messaging configuration license for the EC system, includes use of departments and scripts. One required per system. This is included in the I-SYS-EC system license and is not usually purchased separately.
I-MSG-BASE - EM	IPedge EM Messaging Base License - Base Messaging configuration license for the EM system, includes use of departments and scripts. One required per system. This is included in the I-SYS-EM system license and is not usually purchased separately.
I-MSG-BASE - EP	IPedge EP Messaging Base License - Base Messaging configuration license for the EP system, includes use of departments and scripts. One required per system. This is included in the I-SYS-EP system license and is not usually purchased separately.
I-MSG-ADV	IPedge IP Messaging Advanced User License with Unified Messaging per user. This license includes basic voicemail features plus unified messaging.
I-MSG-ADV UP	IPedge IP Messaging Advanced User upgrade- per user. This license requires the I-MSG-BAS license and adds unified messaging.
I-MSG-CH	One IPedge Messaging simultaneous channel license required for each simultaneous call into voicemail.
I-MT-A	IPedge Meeting meet-me conferencing audio channel License. One required for each simultaneous meet-me audio conferencing participant. Minimum 4.
I-MT-RCD	IPedge Meeting Audio Conference Record License. One required for each simultaneous channel of audio conference recording.
I-MT-W	IPedge Meeting Web Conference Application - per concurrent user IPedge Meeting meet-me conference web collaboration channel license. One required for each simultaneous web collaboration session participant.
I-MT-V	IPedge Meeting Video channel license
I-MT-WV	IPedge Meeting Web plus Video channel license
I-SYS-EC-DSCNT	IPedge System License for EC server. One required for each system. Specially priced bundle includes the IPedge EC system License and Call Manager Standard and one Recovery DVD, for a total of: 24 I-CM-STD1, 24 I-CP-USR-EC, 12 I-CP-TRUNK, 1 I-MS-BASE, 24 I-MSG-ADV, 1 I-MSG-BASE-EC, 6 I-MSG-CH, 1 I-SYS-PLTFM-EC and 1 I-RCVY-DVD-VB.

Table 27 IPedge License Part Numbers (continued)

I-SYS-EC	IPedge System License for EC server. One required for each system. Bundled licenses include: 24 of I-CP-USR-EC, 12 of I-CP-TRUNK, 1 of I-MS-BASE, 24 of I-MSG-ADV, 1 of I-MS-BASE-EC, 6 of I-MSG-CH, and 1 of I-SYS-PLTFM-EC. This is bundled with the I-SYS-EC-DSCNT license and not usually purchased separately.
I-SYS-EM-DSCNT	IPedge System License for EM server. One required for each system. Specially priced bundle includes the IPedge EM system License and Call Manager Standard and one Recovery DVD, for a total of: 32 I-CM-STD1, 32 I-CP-USR-EM, 16 I-CP-TRUNK, 1 I-MS-BASE, 32 I-MSG-ADV, 1 I-MSG-BASE-EM, 8 I-MSG-CH, 1 I-SYS-PLTFM-EM and 1 I-RCVY-DVD-VB.
I-SYS-EM	IPedge System License for EM server. One required for each system. Bundled licenses include: 32 of I-CP-USR-EM, 16 of I-CP-TRUNK, 1 of I-MS-BASE, 32 of I-MSG-ADV, 8 of I-MSG-CH, 1 of I-MSG-BASE-EM, and 1 of I-SYS-PLTFM-EM. This license is part of I-SYS-EM-DSCNT and is not usually purchased separately.
I-SYS-EP-DSCNT	IPedge System License for EP server. One required for each system. Specially priced bundle includes the IPedge EP system License and Call Manager Standard and one Recovery DVD, for a total of: 6 I-CM-STD1, 6 I-CP-USR-EP, 3 I-CP-TRUNK-DSCNT, 1 I-MS-BASE, 6 I-MSG-ADV, 1 I-MSG-BASE-P, 4 I-MSG-CH, 1 I-SYS-PLTFM-EP and 1 I-RCVY-DVD-VB.
I-SYS-EP	IPedge System License for EP server. One required for each system. Bundled licenses include: 6 of I-CP-USR-EP, 3 of I-CP-TRUNK, 1 of I-MS-BASE, 6 of I-MSG-ADV, 1 of I-MS-BASE-EP, 4 of I-MSG-CH, and 1 of I-SYS-PLTFM-EP. This is bundled with the I-SYS-EC-DSCNT license and not usually purchased separately.
I-SYS-EP-PKG16	IPedge EP system 16 user package. Includes I-SYS-EP.
I-SYS-PLTFM-EC	IPedge EC system Linux OS, database, and base platform software license. One required per system. Usually purchased as part of the I-SYS-EC bundle. (Not purchasable in FYI as it is included in the bundles).
I-SYS-PLTFM-EM	IPedge EM system Linux OS, database, and base platform software license. One required per system. Usually purchased as part of the I-SYS-EM bundle. (Not purchasable in FYI as it is included in the bundles).
I-SYS-PLTFM-EP	IPedge EP system Linux OS, database, and base platform software license. One required per system. Usually purchased as part of the I-SYS-EC bundle. (Not purchasable in FYI as it is included in the bundles).
LIC-ACD	ACD license. Required to activate ACD support in an IPedge or Strata CIX system (one license is included in ACD turnkey packages and software packages). One license is required for each system in the network ACD system. Also required for Tracer and Talkument system if ACD or Net Phone is not installed. On IPedge an I-CP-AUX license is required for each voice announce channel. On Strata CIX, a basic port license is required for each voice announce channel.
LIC-ATT	One license is required to activate each Attendant Console on an IPedge or Strata CIX or CTX system. This license is also bundled with Toshiba supplied Attendant Console PCs IPATTCNS, CIX-IPATTCNS and CTX-ATTCONSOLE2 so it is NOT necessary to order the LIC-ATT for these parts. LIC-ATT should only to be ordered separately when using a Dealer or customer supplied PC for the Attendant Console. This license can be used for the Attendant Console with an IP or Digital talk path on Strata CIX And CTX. Attendant Console Software and Documentation is available for download on TSD FYI. On IPedge I-CP-AUX is also required, on Strata CIX or CTX a Basic Port license is required.
SUS	Software Support and Upgrade Service (SUS) token. This part is required when purchasing Software Support and Upgrade Service for a system which has a Multi-Year Support Agreement. Quote and FYI will calculate the quantity to be purchased and FYI will prompt the Dealer at time of license generation. Dealer will use this "SUS" license part number to complete the license generation process.

Table 27 IPedge License Part Numbers (continued)

User License Bundles	
I-USR-STD-EC	IPedge EC User bundle with Call Manager Standard. Includes I-CP-USR-EC and I-CM-STD1.
I-USR-STD-EM	IPedge EM User bundle with Call Manager Standard. Includes I-CP-USR-EM, I-CM-STD1.
I-USR-STD-EP	IPedge EP User bundle with Call Manager Standard. Includes I-CP-USR-EP, I-CM-STD1.

1. Toshiba strongly recommends that dealers carry at least one spare hard drive to support their installed base. Hard drives are field replaceable.

Table 28 Audiocodes Part Numbers

Part Number	Description
MP1122SSIP-BDL	Bundles the MP112/2S/SIP gateway + Audiocodes Enhanced Technical Support + Hardware Replacement Warranty. MediaPack 112 Analog VoIP Gateway, 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. Bundle-includes MP112/2S/SIP Gateway + Ext. Support + AWR. (Can only be purchased as a bundle which includes Enhanced Technical Support and Hardware Warranty Replacement. THERE IS NO OUT-OF-WARRANTY REPAIR available for this item. Unit must be in warranty to qualify for replacement coverage in case of defects. FYI will send a reminder to the dealer to reactivate the warranty for an additional year).
MP1142SOSIP-BDL	Bundles the MP114/2S/2O/SIP gateway + Audiocodes Enhanced Technical Support + Hardware Replacement Warranty. MediaPack 114 Analog VoIP Gateway, 2 FXS, 2 FXO SIP Package including 2 FXO and 2 FXS analog lines, single 100/10 BaseT, AC power supply, including G.711/723.1/726/727/729AB Vocoders, SIP. (Can only be purchased as a bundle which includes Enhanced Technical Support and Hardware Warranty Replacement. THERE IS NO OUT-OF-WARRANTY REPAIR available for this item. Unit must be in warranty to qualify for replacement coverage in case of defects. FYI will send a reminder to the dealer to reactivate the warranty for an additional year).Bundle-includes MP114/2S/2O/SIP Gateway + Ext. Support + AWR
MP114/4SSIP-BDL	Bundles the MP114/4S/SIP gateway + Audiocodes Enhanced Technical Support + Hardware Replacement Warranty. MediaPack 114 Analog VoIP Gateway, 4 FXS, SIP Package including 4 FXS analog lines, single 10/100 BaseT, AC power supply, life line support (requires additional life line cable), G.711/723.1/726/727/729AB Vocoders, SIP. (Can only be purchased as a bundle which includes Enhanced Technical Support and Hardware Warranty Replacement. THERE IS NO OUT-OF-WARRANTY REPAIR available for this item. Unit must be in warranty to qualify for replacement coverage in case of defects. FYI will send a reminder to the dealer to reactivate the warranty for an additional year).
MP1144OSIP-BDL	Bundles the MP114/4O/SIP gateway + Audiocodes Enhanced Technical Support + Hardware Replacement Warranty. MediaPack 114 Analog VoIP Gateway, 4 FXO SIP Package including 4 FXO analog lines, single 10/100 BaseT, AC power supply, life line support (requires additional life line cable), G.711/723.1/726/727/729AB Vocoders, SIP. (Can only be purchased as a bundle which includes Enhanced Technical Support and Hardware Warranty Replacement. THERE IS NO OUT-OF-WARRANTY REPAIR available for this item. Unit must be in warranty to qualify for replacement coverage in case of defects. FYI will send a reminder to the dealer to reactivate the warranty for an additional year).

Table 28 Audiocodes Part Numbers *(continued)*

MP1184S4O-BDL	Bundles the MP118/4S/4O/SIP gateway + Audiocodes Ext. Technical Support + Advance Warranty Replacement. MediaPack 118 Analog VoIP Gateway, 4 FXS, 4 FXO SIP Package including 4 FXO and 4 FXS analog lines, single 100/10 BaseT, AC power supply, including G.711/723.1/726/727/729AB Vocoders, SIP. (Can only be purchased as a bundle which includes Enhanced Technical Support and Hardware Warranty Replacement. THERE IS NO OUT-OF-WARRANTY REPAIR available for this item. Unit must be in warranty to qualify for replacement coverage in case of defects. FYI will send a reminder to the dealer to reactivate the warranty for an additional year).
MP1188SSIP-BDL	Bundles the MP118/8S/SIP gateway + Audiocodes Enhanced Technical Support + Hardware Replacement Warranty. MediaPack 118 Analog VoIP Gateway, 8 FXS, SIP Package including 8 FXS analog lines, single 10/100 BaseT, AC power supply, life line support (requires additional life line cable), including G.711/723.1/726/727/729AB Vocoders, SIP. (Can only be purchased as a bundle which includes Enhanced Technical Support and Hardware Warranty Replacement. THERE IS NO OUT-OF-WARRANTY REPAIR available for this item. Unit must be in warranty to qualify for replacement coverage in case of defects. FYI will send a reminder to the dealer to reactivate the warranty for an additional year).
MP1188OSIP-BDL	Bundles the MP1188OSIP gateway + Audiocodes Enhanced Technical Support + Hardware Replacement Warranty. MediaPack 118 Analog VoIP Gateway, 8 FXO, SIP Package including 8 FXO analog lines, single 10/100 BaseT, AC power supply, life line support (requires additional life line cable), including G.711/723.1/726/727/729AB Vocoders, SIP. (Can only be purchased as a bundle which includes Enhanced Technical Support and Hardware Warranty Replacement. THERE IS NO OUT-OF-WARRANTY REPAIR available for this item. Unit must be in warranty to qualify for replacement coverage in case of defects. FYI will send a reminder to the dealer to reactivate the warranty for an additional year).
M1K-D1-BDL	Bundles the M1K-D1 gateway + Audiocodes Enhanced Technical Support + Hardware Replacement Warranty. Mediant 1000 VoIP Gateway, 1 E1/T1, SIP package including single module of 1 span E1/T1, dual 10/100BaseT Ethernet, and single AC power supply. Supports mixed configurations of Analog and Digital voice modules. Control protocol: SIP, including G.711/723 (Can only be purchased as a bundle which includes Enhanced Technical Support and Hardware Warranty Replacement. THERE IS NO OUT-OF-WARRANTY REPAIR available for this item. Unit must be in warranty to qualify for replacement coverage in case of defects. FYI will send a reminder to the dealer to reactivate the warranty for an additional year).
M1K-D2-BDL	Bundles the M1K-D1 gateway + Audiocodes Enhanced Technical Support + Hardware Replacement Warranty. Mediant 1000 VoIP Gateway, 1 E1/T1, SIP package including single module of 1 span E1/T1, dual 10/100BaseT Ethernet, and single AC power supply. Supports mixed configurations of Analog and Digital voice modules. Control protocol: SIP, including G.711/723 (Can only be purchased as a bundle which includes Enhanced Technical Support and Hardware Warranty Replacement. THERE IS NO OUT-OF-WARRANTY REPAIR available for this item. Unit must be in warranty to qualify for replacement coverage in case of defects. FYI will send a reminder to the dealer to reactivate the warranty for an additional year).

Table 28 Audiocodes Part Numbers *(continued)*

M1K-D3-BDL	Bundles the M1K-D3 gateway + Audiocodes Enhanced Technical Support + Hardware Replacement Warranty. Mediant 1000 VoIP Gateway, 4 E1/T1, SIP Package including single module of 4 spans E1/T1, dual 10/100BaseT Ethernet, and single AC power supply. Supports mixed configurations of Analog and Digital voice modules. Control protocol: SIP, including G.711/723.1/726/727/729AB Vocoders. (Can only be purchased as a bundle which includes Enhanced Technical Support and Hardware Warranty Replacement. THERE IS NO OUT-OF-WARRANTY REPAIR available for this item. Unit must be in warranty to qualify for replacement coverage in case of defects. FYI will send a reminder to the dealer to reactivate the warranty for an additional year).
M1K-D4-BDL	Bundles the M1K-D4 gateway + Audiocodes Enhanced Technical Support + Hardware Replacement Warranty. Mediant 1000 VoIP Gateway, 1 Fractional Span SIP Package including a single module of 1 Fractional span E1/T1 (15 voice channels), dual 10/100BaseT Ethernet, and single AC power supply. Supports mixed configurations of Analog and Digital voice modules. Control protocol: SIP, including G.711/723.1/726/727/729AB Vocoders. (Can only be purchased as a bundle which includes Enhanced Technical Support and Hardware Warranty Replacement. THERE IS NO OUT-OF-WARRANTY REPAIR available for this item. Unit must be in warranty to qualify for replacement coverage in case of defects. FYI will send a reminder to the dealer to reactivate the warranty for an additional year).

Table 29 Adtran Part Numbers

Part Number	Description
4212908L1	Total Access 908 - T1 network interface, 8 FXS ports, DSX-1 port, 10/100 BaseT and IP Router. Includes G.168 Echo Cancellation and support for G.711 and G.729a CODECs. Supports VoIP applications using SIP. Router features include OSPF, BGP, RIP V1 and V2, Static routes, 802.1d Bridging (all protocols), CLI, Frame Relay and PPP WAN Protocols, SNMP, Telnet, Craft/Console port, TFTP support and stateful inspection firewall.
4212924L2	Total Access 908e - Four T1 network interfaces (two of the T1's can be configured for DSX-1 applications), 8 FXS ports, two 10/100 BaseT interfaces, single FXO interface and IP Router. Includes G.168 Echo Cancellation and support for G.711 and G.729a CODECs and supports up to 60 simultaneous TDM to VoIP call conversions. Supports VoIP applications using SIP. Router features include OSPF BGP, RIP V1 and V2, Static routes, 802.1d Bridging (all protocols), CLI; FR, MLFR, PPP and MLPPP WAN Protocols; SNMP, Telnet, Craft/Console port, TFTP support and stateful inspection firewall.

Table 30 Epygi Part Numbers

Part Number	Description
SKU-QM-0002	Quadro E1/T1 Single-RM rack mountable gateway enclosed in a rugged metal 1U chassis. It can be mounted in a 19-inch rack. Supports 24 channels of G.729 or G711 using a T1. Supports 30 channels of G.729 or 30 channels of G.711 using E1.
SKU-QM-0003	Quadro E1/T1 Single-RM rack mountable gateway enclosed in a rugged metal 1U chassis. It can be mounted in a 19-inch rack. Supports 30 channels of G.729 or 48 channels of G711 using a T1. Supports 30 channels of G.729 or 60 channels of G.711 using E1.
SKU-QM-0004	Quadro E1/T1 Single-RM rack mountable gateway enclosed in a rugged metal 1U chassis. It can be mounted in a 19-inch rack. Supports 48 channels of G.729 or 48 channels of G711 using a T1. Supports 60 channels of G.729 or 48 channels of G.711 using E1.

Media Application Server (MAS) and MicroMAS

Table 31 Specifications

Specification	MAS	MicroMAS
CPU	Pentium Dual Core, 1.8 GHz	Pentium 2.9GHz
Operating Software	Microsoft Windows 7 or Windows 2008 See table below.	Microsoft Windows 7
Memory	2 x 2 GB DDR2	2 x 2GB DDR3
Hard Drive	See table below.	250 GB Serial ATA
SATA DVD Drive	Serial ATA Combo Drive CD, CD/RW, DVD-ROM	DVD/RW Optical Drive
PCI Slots	2 Full length (Note: RAID 5 chassis support one PCI slot)	2 Half length
USB	2 Ports - Front side connection	USB 3.0 x2, USB 2.0x4
Serial I/O	1 Port - Front side connection	2 ports
Chassis	2U rack mount chassis	Desktop chasis
Dimensions (HxWxD)	3.47" x 17.32" x 20"	13" x 12.6" x 3.7"
Operating Temperature	32° ~ 122° F	32° ~ 104° F
Fans	2 fixed rear fans, 1 hard drive chamber fan	1 rear fan
Power Supply	350 Watts	80 Plus 270W

MAS No RAID	250 GB Hard Drive	Windows 7 Operating Software
MAS with RAID 1	250 GB x 2 Hard Drive	Windows 2008 Operating Software
MAS with RAID 5	300 GB x 3 Hard Drive	Windows 2008 Operating Software

Mobile Device Support for IPMobility

The IPMobility client application is supported on Android OS versions; 2.x, 3.x, & 4.x, and has been tested on the following devices:

Table 32 IPMobility Mobile Device Support

Device Category	Requirements
HTC	with Android 2.2.2
	EVO 4G LTE with Android 4.0.3
LG-P350	with Android 2.2.2
Motorola	Droid 2, 3, & X with Android 2.3.4
	MB 520 with Android 2.2.1
Samsung	Galaxy II with Android 2.3.5
	SGH I997 with Android 2.2.1
Sony Xperia™ E15i	with Android 2.1

There are no material differences in the functionality of these devices requiring changes to the application. There are two known variances:

1. Automatic answer of Callback calls from IPedge system is not available after Android 2.3.5
2. Not all devices allow override of the default answering screen with a custom answering screen (app call screening function). One example of this is the Samsung SGH I997 with Android 2.2.1.

With the above information in mind, and considering the array of differences among mobile devices in the marketplace including best practices for mobile application development - Toshiba elected to test the IPMobility application with a sampling of popular devices.

VIPedge Application Services Part Numbers

Part Number	Description
V-APP-ACC	VIPedge Application Service Accessory Service. Includes 1 DID number. This enables an accessory only for use on the VIPedge Accessory platform. This could be used for door phones or external paging for IPedge and CIX Systems. (subject to VIPedge Terms of Service)
V-APP-STD	VIPedge Application Service Standard User Service. Includes 1 DID number and 1 advanced voicemail box with unified messaging and follow me and 1 Call Manager Mobile service.
V-APP-CHCC	VIPedge Application Service Channel - Unlimited Local / Long Distance Minutes per VIPedge Terms of Service. International and toll free calling charged on per minute basis.
V-APP-CHU	VIPedge Application Service Channel - Unlimited Local / Long Distance Minutes. International and Toll Free calling charged on per minute basis. (Subject to VIPedge Terms of Service)
V-APP-DIDPK	VIPedge DID Parking fee - monthly recurring charge.
V-APP-AA	VIPedge Application Service license - One per system/container. This is automatically added by the quoting system.

End User License Agreement

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Telecommunication Systems Division
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TOSHIBA AMERICA INFORMATION SYSTEMS, INC.
Telecommunication Systems Division
End User Standard Limited Warranty

For Telecommunication Equipment

Purchased Within the Fifty (50) United States and District of Columbia,
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This limited warranty applies to the telecommunication equipment (except for fuses, lamps, and other consumables) sold by Toshiba America Information System, Inc. (“**Toshiba**”) or a Toshiba authorized dealer to an end user within the Fifty (50) United States and District of Columbia, United States Territories, Puerto Rico, Latin America, and the Caribbean for such end user’s own use and not for resale (“**End User**”) and described in the below table (the “**Product**”). The limited warranty period for each Product is for the period described in the below table and begins on the date Toshiba or its authorized dealer delivers the Product to End User (the “**Limited Warranty Period**”).

Product	Limited Warranty Period
IPedge™ New Telecommunication Equipment	One (1) year
Toshiba-branded New Telecommunication Equipment (excluding IPedge™)	Two (2) years
Toshiba-branded Refurbished Telecommunication Equipment	Ninety (90) days

End User may be required to provide proof of purchase as a condition of receiving warranty service.

Toshiba warrants that the Product is free from defects in materials and workmanship under normal use.

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- Service made necessary by installing or using Product in combination or in assembly with third party products that are incompatible or of inferior quality, design or performance.
- Service on Product on which the Toshiba label or logo or serial number is defaced or missing.
- On-site service and repair of the Product.
- Modifications to the Product not approved in writing by Toshiba.
- Replacement of missing parts, providing retrofits or preventive maintenance.
- Third party products. The Product may be subject to warranty provisions provided by a third party provider.
- Toshiba and third party software and any related documentation pre-installed on or shipped with the Product or otherwise made available by Toshiba in whatever form or media.

PROTECTION OF STORED DATA

For an End User's important data, please make periodic back-up copies of all data stored on the Product as a precaution against possible failures, alteration, or loss of the data. IF AN END USER'S DATA IS ALTERED OR LOST DUE TO ANY TROUBLE, FAILURE OR MALFUNCTION OF THE PRODUCT AND THE DATA CANNOT BE RECOVERED, TOSHIBA WILL NOT BE LIABLE FOR ANY DAMAGE OR LOSS OF DATA OR ANY OTHER DAMAGE RESULTING THEREFROM. WHEN COPYING OR TRANSFERRING AN END USER'S DATA, PLEASE CONFIRM WHETHER THE DATA HAS BEEN SUCCESSFULLY COPIED OR TRANSFERRED. TOSHIBA DISCLAIMS ANY LIABILITY FOR THE FAILURE TO COPY OR TRANSFER THE DATA CORRECTLY.

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