Strata®

CIX™

Strata CIX40, CIX100-S, CIX100, CIX200, CIX670, and CIX1200 General Description

Publication Information

Toshiba America Information Systems, Inc.
Telecommunication Systems Division

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Strata CIX General End User Information

The Strata CIX40, CIX100, CIX200, CIX670, and CIX1200 Business Telephone System is registered in accordance with the provisions of Part 68 of the Federal Communications Commission's Rules and Regulations.

FCC Requirements

Means of Connection: The Federal Communications Commission (FCC) has established rules which permit the Strata CIX system to be connected directly to the telephone network. Connection points are provided by the telephone company—connections for this type of customer-provided equipment will not be provided on coin lines. Connections to party lines are subject to state tariffs.

Incidence of Harm: If the system is malfunctioning, it may also be disrupting the telephone network. The system should be disconnected until the problem can be determined and repaired. If this is not done, the telephone company may temporarily disconnect service. If possible, they will notify you in advance, but, if advance notice is not practical, you will be notified as soon as possible. You will be informed of your right to file a complaint with the FCC.

Service or Repair: For service or repair, contact your local Toshiba telecommunications distributor. To obtain the nearest Toshiba telecommunications distributor in your area, log onto www.toshiba.com/taistsd/pages/support_dealerlocator.html or call (800) 222-5805 and ask for a Toshiba Telecom Dealer.

Telephone Network Compatibility: The telephone company may make changes in its facilities, equipment, operations, and procedures. If such changes affect the compatibility or use of the Strata CIX40, CIX100, CIX200, CIX670 or CIX1200 system, the telephone company will notify you in advance to give you an opportunity to maintain uninterrupted service.

Notification of Telephone Company: Before connecting a Strata CIX system to the telephone network, the telephone company may request the following:

1. Your telephone number.
2. FCC and ACTA registration

- Strata CIX40, CIX100, CIX200, CIX670 or CIX1200 may be configured as a Key, Hybrid or PBX telephone system. The appropriate configuration for your system is dependent upon your operation of the system.
- If the operation of your system is only manual selection of outgoing lines, it may be registered as a Key telephone system.
- If your operation requires automatic selection of outgoing lines, such as dial access, Least Cost Routing, Pooled Line Buttons, etc., the system must be registered as a Hybrid telephone system. In addition to the above, certain features (tie Lines, Off-premises Stations, etc.) may also require Hybrid telephone system registration in some areas.
- If you are unsure of your type of operation and/or the appropriate FCC registration number, contact your local Toshiba telecommunications distributor for assistance.

FCC Registration Numbers			
SYSTEM	**PBX** Fully-protected PBXs	**Hybrid** Fully-protected multifunction systems	**KEY** Fully-protected telephone key systems
CIX40	CJ6-PF03BDTCHS402	CJ6-MF03BDTCHS402	CJ6-KD03BDTCHS402
CIX100, CIX670 and CIX1200	CJ6-MUL-35931-PF-E	CJ6-MUL-35930-MF-E	CJ6-MUL-35929-KF-E
CIX200	CJ6-PF03BDTCHS192	CJ6-MF03BDTCHS192	CJ6-KD03BDTCHS192

- Ringer equivalence number: 0.3B. The ringer equivalence number (REN) is useful to determine the quantity of devices which you may connect to your telephone line and still have all of those devices ring when your number is called. In most areas, but not all, the sum of the RENs of all devices connected to one line should not exceed five (5.0B). To be certain of the number of devices you may connect to your line, as determined by the REN, you should contact your local telephone company to ascertain the maximum REN for your calling area.

3. Network connection information USOC jack required: RJ11/14C, RJ21/2E/2F/2G/2HX/RJ49C (see Network Requirements in this document). Items 2, 3 and 4 are also indicated on the equipment label.
4. Authorized Network Parts: 02LS2/GS2, 02RV2-T/O, OL13C/B, T11/12/31/32M, 04DU9-BN/DN/1SN, 02IS5, 04DU9-BN/DN/1SN1ZN

Radio Frequency Interference

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

Underwriters Laboratory

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

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CP01, Issue 8, Part I Section 14.1
Notice: The Industry Canada label identifies certified equipment. This certification means that the equipment meets certain telecommunications network protective, operational and safety requirements as prescribed in the appropriate Terminal Equipment Technical Requirements document(s). The Department does not guarantee the Equipment will operate to the user's satisfaction.

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.

Repairs to certified equipment should be coordinated by a representative designated by the supplier. Any repairs or alterations made by the user to this equipment, or equipment malfunctions, may give the telecommunications company cause to request the user to disconnect the equipment.
Users should ensure for their own protection that the electrical ground connections of the power utility, telephone lines and internal metallic water pipe system, if present, are connected together. This precaution may be particularly important in rural areas.

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

CP01, Issue 8, Part I Section 14.2
Ringer Equivalence Notice: The Ringer Equivalence Number (REN) assigned to each terminal device provides an indication of the maximum number of terminals allowed to be connected to a telephone interface. The terminal on an interface may consist of any combination of devices subject only to the requirement that the sum of the Ringer Equivalence Numbers of all the Devices does not exceed 5.
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US5,787,391, US5,717,825, US5,708,757, US5,754,976, US5,701,392, US5,699,482, US5,444,816

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Telecommunication Systems Division
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Contents

Introduction

This General Description provides an overview of the Strata CIX40, CIX100-S, CIX100, CIX200, CIX670, and CIX1200 IP and digital business telephone systems, associated hardware and features. The features described in this document assume that the Strata CIX/CTX system has the current software release installed.

Organization

This document is divided into the following major topics:

- **Chapter 1 – Strata CIX Overview** is a brief introduction of the Strata CIX systems.
- **Chapter 2 – Strata CIX40** describes the system, its basic capacities and system expansion.
- **Chapter 3 – Strata CIX100-S / CIX100** describes the system, its basic capacities and system expansion.
- **Chapter 4 – Strata CIX200** describes the CIX200 system (hardware, software and circuit cards), its unique attributes and basic functionality.
- **Chapter 5 – Strata CIX670 and CIX1200** describes the system, its basic capacities, system expansion, and remote maintenance.
- **Chapter 6 – Universal Slot Circuit Cards** provide information about Printed Circuit Boards (PCBs) that can be installed in the universal slots of the Strata CIX/CTX systems.
- **Chapter 7 – Telephones and Peripherals** describes the most recent Toshiba-proprietary stations and peripherals, customer-supplied peripherals, as well as cabling and connectors.
- **Chapter 8 – IPedge Application Server** describes the features of the IPedge system that can be used with the Strata CIX system.
- **Chapter 9 – VIPedge Application Service** describes the VIPedge cloud based services available.
- **Chapter 10 – Features** describes the features which are available system-wide, as well as stations features.
- **Appendix – Specifications** includes detailed information on environmental characteristics, power considerations, hardware compatibility, network requirements, and station specifications and system 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!	*Calls attention to important instructions or information.*
`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 phone 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.
➤	Denotes a procedure.
➤	Denotes the step in a one-step procedure.
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

- Strata CIX Installation & Maintenance Manual
- Strata CIX Programming Manual (Volume 1)
- Strata CIX Programming Manual (Volume 2 – Stratagy ES Voice Mail Application)
- Strata CIX Programming Manual (Volume 3 – Application Implementation)

User Guides

- Strata CIX IP5000-Series Telephone
- Strata CIX DP5000-series Telephone
- Strata CIX IPT Telephone
- Strata CIX DP Telephone
- My Phone Manager™
- Strata CTX DKT3001/2001 Digital Single Line Telephone
- Strata CIX Standard Telephone
- Strata CIX DKT2404-DECT Cordless Telephone
- Strata DKT2204-CT/DKT2304-CT Cordless Telephones

Quick Reference Guide

- Strata CIX IP5000-Series Telephone
- Strata CIX DP5000-Series Telephone
- Strata CIX/CTX DKT/IPT Telephone

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

Strata CIX Overview 1

The Strata CIX family includes the CIX40, CIX100-S, CIX100, CIX200, CIX670, and CIX1200 Internet Protocol (IP) systems that provide sophisticated business communication features. These systems deliver on the promise of IP telephony by providing all the features and benefits of our traditional business communications systems on a converged IP platform. With the Strata CIX, you have a choice of mixing and matching technologies, creating a pure IP system or a converged solution, and changing based on the needs of your company. The Strata CIX systems can be networked to the IPedge system and VIPedge Cloud-based Business Telephone Solution. Refer to Private Networking Over IP to the IPedge System and Private Networking Over IP to the VIPedge.

Important! *The Strata CIX100-S uses the same hardware and configuration as the Strata CIX100, with a few exceptions. Whenever the CIX100 is mentioned in this book, it applies to both the CIX100-S and CIX100, unless specified otherwise.*

The Strata CIX IP business communications system provides pure IP peer-to-peer functionality and provides a smooth migration path from Toshiba Strata CTX and Strata DK digital business communication systems.

The Strata CIX supports many types of telephones and 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. That's why the Strata CIX is much more than just an IP system. It is a unified communications environment that supports many types of client devices.

Available with the Strata CIX 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.

Optionally available with the Strata CIX is the IPedge or VIPedge Application Servers, which allows multiple applications to be combined on a single device. Applications include Auto Attendant, Voice Mail, Automated Speech Recognition (ASR), Text-to-Speech, Unified Messaging, Interactive Voice Response (IVR), Automatic Call Distribution (ACD) and Reporting, Web browser-based Personal and System Administration, Web-based Telephone Applications, and third-party applications.

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Strata CIX40 2

This chapter provides a system overview of the Strata CIX40 telephone system.

The CIX40 is an IP communication system that brings powerful IP telephony and Strata Net networking capabilities to small customers and branch office locations.

The Strata CIX40 system is a compact system that provides large system features (for dimensions, see Table 1). The CIX40 is designed for wall mounting and occupies very little space.

An optional 4~8 port, 40-hour integrated plug and play Toshiba Voice Processing system (GVPH1A) with built-in modem plugs into a dedicated slot in the Strata CIX40 cabinet. This integrated combination of the Strata CIX40 and GVPH1A is a very powerful and cost-effective small business solution.

The plug and play CIX40 is easy and very cost effective to install. The default plug and play configuration is already programmed in the CIX40 and GVPH1A voice processing system. Any additional programming or changes to the CIX40 can be done by using the latest version of Network eManager software that is found on Internet FYI.

The CIX40 can be configured with up to 24 IP channels to support Toshiba IP Desktop and Soft phones; SIP telephones, SIP trunks and IP Strata Net channels to network multiple Strata CIX systems.

The CIX40 easily connects to outside public and private telephone lines. All of the telephones (stations) tied to the system can have direct access to each other, as well as to the public and private network. All lines, stations, and options are tied together through the cabinet.

The CIX40 does not require processor licenses for analog Caller ID CO Lines, the first four Voice Mail ports, digital or analog station ports, DTMF Receivers, etc. The only licenses needed are for Voice Mail ports 5~8, IP Strata Net networking channels, Toshiba IP telephones, SIP phones SIP trunks and Soft IP telephones connected to the optional GIPU8, GIPH-X1A, MIPU16-1A and MIPU24-1A IP interface cards.

Licenses are required on customer provided PCs for applications such as ACD, Strata Call Manager (SCM), TASKE, etc.

The CIX40 comes with built-in circuitry for eight digital telephone ports, four CO line ports with Caller ID Interface, one standard telephone port, built-in maintenance modem, LAN/NIC interface, and one power failure transfer relay.

The Strata CIX40 system capacity can be expanded by adding the following expansion components shown in parenthesis:

The fully expanded CIX40 system with optional interface cards provides the following capacities:

- 11 loop start CO line ports with Caller ID and 16 digital station ports (GCDU2A and GCOCIH1A).
- Two standard telephone ports (GSTU1A).
- Eight, 16 or 24 IP channels for IP for any combination of telephone or Strata Net connection (GIPU8, GIPH-X1A, MIPU16-1A or MIPU24-1A).
- 4, 6 or 8 voice mail ports with 40 hours of message storage and 360 mailboxes (GVPH1A, LIC-2 GVPH for 6 or 8 port).
- Four serial interface ports (BSIS).

The table below lists the CIX40 cabinet physical specifications

Table 1 CIX40 Cabinet Specifications

Cabinet	Weight	Height	Width	Depth
Cabinet (CHSU40A2) The system cabinet is off-black and contains the GMAU/GMAS motherboards	6.6 lbs.	18 in.	12.2 in.	3.5 in.

Processor

The Strata CIX40 processor (GCTU2A) is a standard part that plugs into a dedicated CIX40 cabinet slot.

The Strata CIX40 uses a high-speed 32-bit RISC processor, DRAM working memory, SRAM with lithium battery to back-up memory.

The Strata CIX40 processor has the following standard components built-in:

- Secure Digital (SD) memory card interface for program data backup, software upgrades, alarms, error logs, and admin logs done locally or remotely.
- Network Interface Card port provides one circuit for 100base-TX Ethernet LAN connection of the Network eManager computer, ACD Server, CIX Attendant Console, and CSTA applications.
- 33.6 Kbps/V.34 Factory installed AMDS1A maintenance modem for connecting the Network eManager administration tool locally or remotely.
- One Music-on-hold (MOH)/Background Music (BGM) RCA jack to interface with external MOH and/or BGM sources. The CIX40 can have up to three MOH/BGM source interfaces by adding two additional MOH/BGM input sources, connected to the Standard Telephone interfaces.

- One External Page RCA jack to interface with a Toshiba External Amplified Speaker (BESCB) or a customer-supplied page amplifier and speaker(s) for external paging, night ring over external page, and external BGM applications.
- One Relay control interface that provides an interface to a normally open relay contact which can be programmed to control a Night Bell, door lock or to mute BGM during an external page.
- Memory Protection Battery that protects data and the customers programmed configuration from memory loss. This information will be maintained in a powerless system for at least six years.
- 16 DTMF receivers.
- 16 Busy Tone (BT) detector circuits for Auto Busy Redial (ABR).
- 64 conference circuits.
- Digital volume PAD technology enables audio volume to be adjusted in eight steps to compensate for conference and/or CO line network losses.

The optional Serial Port Interface Subassembly (BSIS1A) can be installed on the processor to provides RS-232 interface ports for SMDR interface to Call Accounting devices, SMDI for external Voice Mail devices, and future applications. The BSIS1A is not needed for the GVPH1A voice mail system because the GVPH1A uses built-in SMDI through the back plane.

CIX40 Cabinet Slots

The CIX40 Cabinet has three dedicated slots used for the GCTU2A system processor card, GVPH optional Voice Mail circuit card and a slot that can add either four analog lines (GCOCIH1A) or 8, 16 or 24 IP channels (GIPU, GIPH, MIPU16 or MIPU24) optional VoIP interface cards. All other CIX40 optional interface cards plug onto the Processor or the Motherboard. The CIX40 does not support CIX1200/670/200/100 circuit cards.

Figure 1 CIX 40 Interior

Large Scale Integrated (LSI) Circuits (no licenses required)

The processor has an LSI circuit that supports the following:

- 16 DTMF receivers
- 16 Busy Tone (BT) detector circuits for Auto Busy Redial (ABR)
- 64 built-in conference circuits
- Built-in, adjustable, digital volume PAD technology enables audio volume to be adjusted in eight steps to compensate for conference and/or CO line network losses

GCTU2A Processor Interfaces

Memory Protection Battery

If commercial AC power is lost or if a system is moved or stored without power, the processor has an on-board battery that protects data and the customer's programmed configuration from memory loss. This information will be maintained in a powerless system for at least six years.

Relay Control Interface

An on-board terminal strip provides an interface to a normally open relay contact which can be programmed to control a Night Bell, door lock or to mute BGM during an external page.

External Page Interface

A 600 ohm RCA jack is built into the processor to interface with a Toshiba External Amplified Speaker Control box (BESCB) or a customer-supplied page amplifier and speaker(s) for external paging, night ring over external page, and external BGM applications.

Music-on-hold/Background Music Interface

One 600-ohm RCA jack is provided on the processor to interface with Music-on-hold and/or Background Music (BGM) sources. With the CIX40, you can have up to three MOH/BGM source interfaces. The CIX40 Standard Telephone interfaces can be used to provide up to two MOH/BGM input sources in addition to the processor MOH/BGM interface.

Secure Digital Memory

The processor has an on-board Secure Digital (SD) memory card slot. A SD memory card can be inserted into the slot to backup and restore customer program data. It also makes it easy to upload operating system data for software upgrades and is used to store maintenance log files.

CIX40 Processor Optional Subassembly

Optional subassembly can be attached to the GCTU2A processor to provide additional features. The subassembly is:

BSIS (Serial Port Interface) – Provides up to two RS-232 interface ports for SMDR interface to Call Accounting devices, SMDI for external Voice Mail devices, and two future applications.

Notes

- The GVPH does not require a BSIS SMDI port.
- The factory installed AMDS maintenance modem and Network Interface Card (NIC) come built into the GCTU2A processor.

GVPH – Integrated Voice Mail

With the release of Strata CIX40, Toshiba is also introducing a plug and play voice mail circuit card (GVPH1A) specifically for the Strata CIX40 cabinet. It plugs into a dedicated cabinet slot and requires no additional hardware to provide its full set of features.

The CIX40 is pre-programmed to match all related voice mail programming in the GVPH Voice Mail system. This includes station numbers, mailboxes, voice mail hunt groups, voice mail station IDs and many other items which are listed in the CIX40 installation documentation. The pre-programmed data is set when the processor is initialized even if the expansion cards or GVPH1A Voice Mail system is not installed prior to system initialization.

It provides comprehensive Auto Attendant/Voice Mail capabilities, including the following:

- 4-Voice Ports (default) and 6 or 8 ports with LIC-2 GVPH license.
- 40 hours voice storage or up to 10,000 total messages.
- 360 mailboxes.
- Voice Mail Call Monitor – answering machine like operation to monitor and optionally pick-up a call when someone is leaving a voice message.
- Voice Mail LCD Feature Prompting with Soft Key Operation (English and Spanish).
- Call Record – enables the user to record live calls.
- Installs in a dedicated CIX40 cabinet slot with SMDI integration built-in on backplane – no serial port or other interface hardware is needed for interface or for its full set of features.
- Built-in remote maintenance modem.
- Administration requires the Stratagy VM Manager Administration Software.
- The Backup and Restore function stores Names, Greetings, Mail box numbers, and all other database parameters. It does not back up/restore user messages.
- Pre-programmed with default mail boxes that match CIX40 default stations numbers, adding simplicity to any installation.

To program additional features beyond the default settings, refer to *Strata CIX40 Voice Processing Programming Manual* and UADMIN2 software.

Feature Compatibility

The Strata CIX40 uses the same Release 5.20 software as Strata CIX100, CIX200 and CIX670 systems, so functionality and features are the same between them, with some exceptions that are listed below.

The Strata CIX40 does not support the following capabilities:

- T1 and PRI digital trunk interfaces.
- DID, Ground start and Tie line analog trunk interfaces (supports Loop start with Caller ID).
- Dialed Number Identification Service (DNIS).
- Zone Page Interface.
- Analog Station: Caller ID, message waiting lamp control and off-premise stations

System Configuration

Following are Strata CIX40 configuration guidelines:

- Pre-programmed to initialize and match CIX40 extension numbers with GVPH1A voice mailboxes. So the system is ready for use when it is powered up the first time.
- Supports all CIX Release 5.2 and above features, except Speaker OCA and OCA on Digital Telephones; T1 and PRI digital trunk interfaces; DID, Ground start and Tie line analog trunk interfaces (supports Loop start with Caller ID), DNIS, and Zone Page Interface.
- Fully Licensed for 18 digital and analog telephones, 4 Voice Mail ports and 11 analog CO lines, 4 RS232 ports and 16-DTMF/ABR circuits. Licensing is only required for Strata Net networking channels, IP telephones, SIP phones, SoftIPTs, and 6 or 8 voice mail ports.
- Maximum Capacity is 16 Digital telephones, 11 Loop start lines with Caller ID Interface, 2 Standard telephone interfaces, 8 Voice Mail ports, 24 IP channels, and 4 RS-232 Serial Interface ports.
- Digital Telephone Compatibility – Supports Toshiba DP5000-series telephones, Add-on modules and DSS console. The DP5000-series telephones include a 10 button telephone set (DP5022-SDM) that is designed as a cost-effective telephone for the CIX40.
- IP Telephone Compatibility – Supports all Toshiba 5000-series IP telephones, Add-on-Modules and DSS consoles, SoftIPT soft phones on laptops and PDAs, and SIP telephones.
- CO Line Compatibility – Loop start lines with or without Caller ID.
- Up to 24 IP SIP and Strata Net trunks are supported.

GVPH1A Voice Mail (optional plug in voice mail card):

- 4, 6 or 8 ports, 40 hours storage for saved messages, 360 mailboxes (4 default, 6 or 8 require LIC-2 GVPH).
- Works similar to IVP8R2 with additional features of Voice Mail Call Monitor, Voice Mail LCD Feature Prompting with Soft Key Operation, and Call Record.
- Plugs into CIX40 dedicated cabinet slot.
- SMDI integration built-in on back plane – no serial port or other interface hardware needed.
- Built-in remote maintenance modem.
- Administration requires the GVPH UADM2 Administration Software. This is a new version of Stratagy Admin exclusively for the GVPH1A.
- Pre-programmed with default mail boxes that match CIX40 default stations numbers, adding simplicity to any installation.

Table 2 CIX40 System Configuration

Basic Configuration - No Licenses Needed	Basic Capacity
CHSU40A2 Cabinet, includes Power supply and Mother Board. The CIX40 cabinet and GCTU2A processor are boxed separately. The cabinet and processor are also warranted and repaired separately.	4 CO line w/CLID, 8 Digital Telephones and 1 Standard Telephone. Includes: 33.6 Kbps/V.34 Maintenance Modem, Page interface, MOH interface, Control relay, 16 DTMF/ABR circuit, 1 Power Failure Transfer interface, NIC port for 100base-TX Ethernet LAN connection of Network eManager computer, ACD Server, CIX Attendant Console, and CSTA applications, and a Secure Digital card interface.
Expanded Configuration – No Licenses Needed	**Capacity/Feature Option**
GCDU2A (3-CO line/8-DKT expansion card)	3 CO line w/CLID and 8 Digital telephones
GCOCIH1A (4 - CO expansion card)	4 CO lines with CLID
GVPH1A (voice mail plug-in card)	4-port Toshiba Plug-in Voice Mail system expandable to 6 or 8 ports with LIC-2 GVPH.
GSTU1A (standard telephone plug-on card)	1 Standard Telephone (total 2 standard telephones)
BSIS1A (4-port RS-232 interface plug-on card).	SMDR for Call Accounting or SMDI for external voice mail system
HPFB-6 one to two HPFB6 Battery with built-in Chargers	Reserve Power (Battery Backup)
Expanded Configuration – Licenses Required	**Expanded Capacity/Feature Option**
GIPU8-1A (8-channel GIPU IP plug-in card) GIPH-X1A (8-channel IP plug-in card). MIPU16-1A (16 channel IP plug-in card). MIPU24-1A (24 channel IP plug-in card). IP License requirements: LIC-CIX-STRN-CH – one license required for each IP Strata Net channel (Strata Net system license is not available). LIC-CIX-IP PORT – one license required for each IP Telephone, SIP phone and SoftIPT. LIC-SOFTIPT – one license required for each SoftIPT.	Maximum 8, 16 or 24 channels for any mix of IP telephones or Strata Net networking channels.

Note See the “CIX40 Functional Block Diagram” on page 12 to see how the circuit cards and interface connectors can be used for connecting stations and peripherals.

Table 3 CIX40 R2 Cabinet Configurations (CHSU40A2 Cabinet/R5.1 SW)

Analog CO Lines	Digital Telephones	IP Channels	Equipment
4	8	0	Cabinet (CHSU40A2)
7	16	0	Cabinet, GCDU2A
8	8	0	Cabinet, GCOCIH
11	16	0	Cabinet, GCDU2A, GCOCIH1A
4	8	8, 16, 24	Cabinet, GIPU, GIPH1A, MIPU161A, MIPU241A
7	16	8, 16, 24	Cabinet, GCDU2A, GIPU, GIPH1A, MIPU16, MIPU24
Voice Mail: GVPH1A can be installed in all of the above configurations to provide 4 default VM Ports and expand to 6 or 8 VM Ports with the LIC2-GVPH license			
Analog FX/Modem Ports: One analog station port is standard on the Base cabinet; a second can be added by installing GSTU1A.			

Table 4 CIX40 R1 Cabinet Configurations (CHSU40A Cabinet/R5.1 SW)

Analog CO Lines	Digital Telephones	IP Channels	Equipment
3	8	0	Cabinet (CHSU40A)
6	16	0	Cabinet, GCDU2A
7	8	0	Cabinet, GCOCIH
10	16	0	Cabinet, GCDU2A, GCOCIH1A
3	8	8, 16	Cabinet, GIPU8, GIPH1A, MIPU16
6	16	8, 16	Cabinet, GCDU2A, GIPU8, GIPH1A, MIPU16
Voice Mail: GVPH1A can be installed in all of the above configurations to provide 4 default VM Ports and expand to 6 or 8 VM Ports with the LIC2-GVPH license			
Analog FX/Modem Ports: One analog station port is standard on the Base cabinet; a second can be added by installing GSTU1A.			

Reserve Power

One or two HPFB-6 optional units can be added to the CIX40 to provide reserve power. The amount of reserve power time depends on the hardware (see Table 5). The table below is an estimate of battery backup time based on the premise that the HPFB-6 unit(s) are fully changed at the time of AC power failure. This estimated backup time is based on normal call traffic, the time estimates will be reduced by as much as half with extreme heavy traffic volumes.

Table 5 CIX40 Reserve Power Duration Estimates

Hardware	1 HPFB-6	2 HPFB-6
3CO/8DKT/DP - No GVPH	1 hr. 40 min.	3 hr. 20 min.
3CO/8DKT/DP - with GVPH	1 hr. 30 min	3 hr.
6CO/16DKT/DP - No GVPH	1 hr. 5 min.	2 hr. 10 min.
6CO/16DKT/DP - with GVPH	1 hr.	2 hr.
11CO/8DKT/DP - No GVPH	1 hr. 20 min.	2 hr. 40 min.
11CO/8DKT/DP - with GVPH	1 hr. 10 min.	2 hr. 20 min.
11CO/16DKT/DP - No GVPH	1 hr.	2 hr.
11CO/16DKT/DP - with GVPH	55 min.	1 hr. 50 min.

Note The CIX40 should be plugged into AC power and the DC power switch should be turned on when installing the HPFB-6. The HPFB-6 will not start to operate if AC power is not available during the initial installation.

Telephone and Console Support

The Strata CIX40 supports all current Toshiba 5000-series Telephones, Add-on Module, DSS Console, and CIX Attendant Console. see Chapter 7 – Telephones and Peripherals.

The Strata CIX40 supports 3000-series and 3200-series digital telephones, including the DKT3007-SD(W) and DKT3207-SD 7-button models, Add-on Module, DSS Console, CIX Attendant Console, and generic single-line telephones (2500-sets). For details on Telephones and Consoles, see Chapter 7 – Telephones and Peripherals.

The Strata CIX40 supports Toshiba 5000-series IP telephones, SoftIPT soft phones on laptops and PDAs, and SIP telephones.

The Strata CIX40 does not support analog electronic telephones (6500-series, 6000-series, etc.) or older IPT1020-SD IP telephones.

The Strata CIX40 supports CIX IP and Digital Attendant consoles.

CIX40 Functional Block Diagram

The Functional Block Diagram below shows the circuit cards and interface connectors used for connecting the stations and peripherals to the CIX40 R2 cabinet (CHSU40A2).

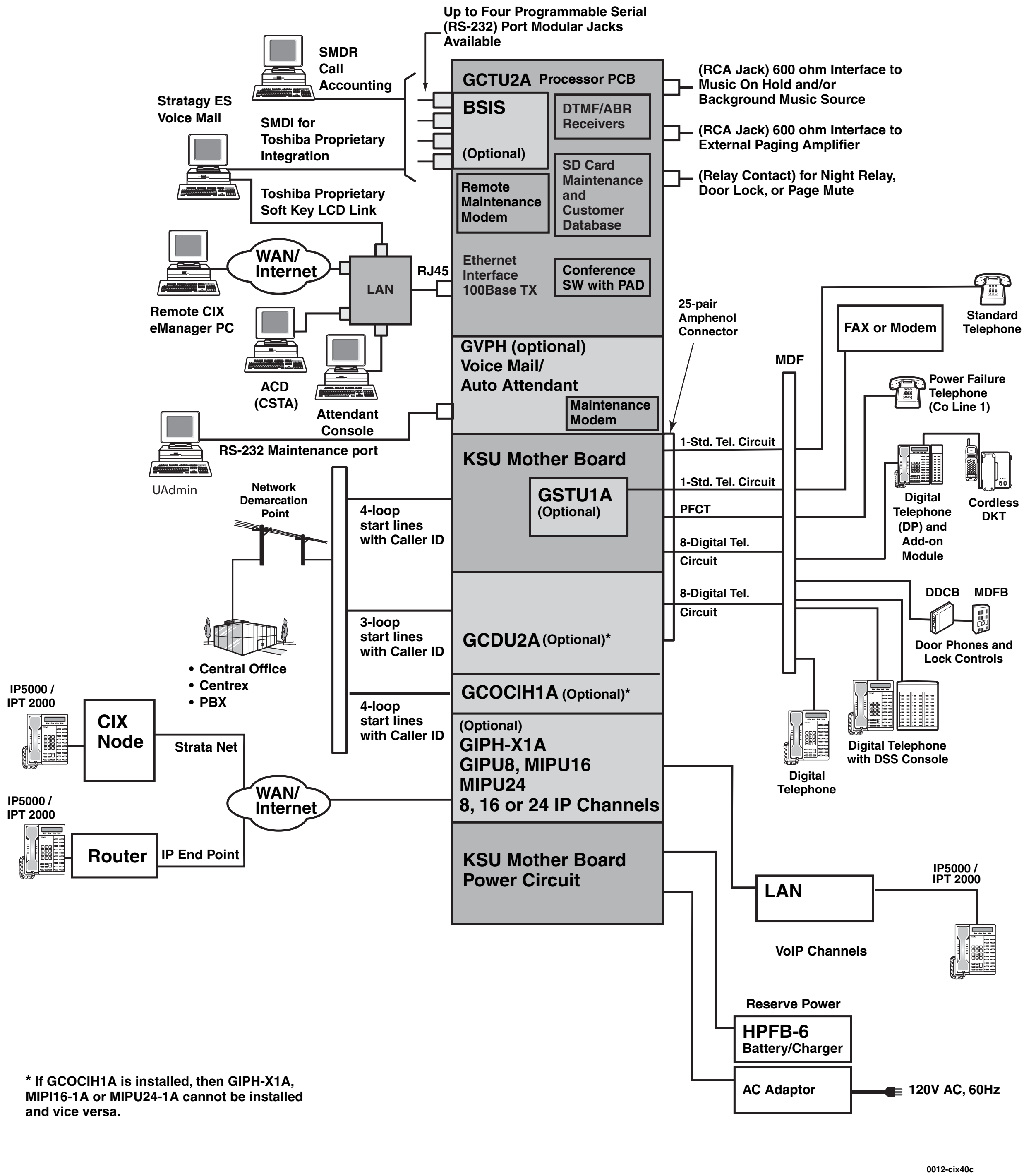


Figure 2 CIX40 Functional Block Diagram

Additional Specifications

Table 6 CIX40 Specifications

Category	Items	Specification	Remarks
Power	Input voltage rating	100-240VAC	
	Input current rating	1.3A max.	At full installation
	Input frequency	50/60 Hz	
	AC adapter output voltage	+15V/4.0A	
	Voltage of inside Power switch	-27.3V/+5.0V	
		DC Switch	Located inside cabinet
	Power failure back up	1 HPFB6: 1 hour 2 HPFB6: 2 hours See page 11	The amount of time depends on hardware configuration. One hour in HPFB6 means 6 CO+16DP with GVPH based on HPFB6 is fully charged.
	Grounding	Customer supplied Grounding wire.	Ground wire must connect the earth ground through the FG terminal on the GMAU for safety purposes and to protect noise.
	AC Plug	2-pin (American)	Ground wire must connect the earth ground through the FG terminal on the GMAU for safety purposes and to protect noise.
Environment	Operating Temperature	0 - 40 C	
	Operating humidity	20 - 80%	
	Storage temperature	-20 - 60 C	
Safety System	Maximum ports	40	16: DP phones
			6: Analog trunk
			8: IP channel
			4: Voicemail
			2: Analog devices
			4: Serial ports
Regulation	Safety	UL60950-1	USA
	EMC	FCC part15 class-A	
	Network Performance	FCC part68	
FCC/ACTA Registration Numbers	ACTA/FCC Part 68 Registration for Key System Code (KD): CJ6KD03BDTCHS402		
	ACTA/FCC Part 68 Registration for Multifunction Code (MF): CJ6MF03BDTCHS402		
	ACTA/FCC Part 68 Registration for Multifunction Code (PF): CJ6PF03BDTCHS402		

Station Loop Limits

The table below provides the maximum loop limits for connection of telephones, lines, peripheral equipment, and power supplies. The following information applies to only the Strata CIX40 system.

Digital Telephone/DIU/DDM Console/ADM/Loop Limits

Mode	CIX40 Cabinet or Battery Backup[1]	Maximum line length (24 AWG) 1 Pair feet	meters
DP5000, DP with BVSU[2] or DVSU or BHEU or HHEU.	CIX40 Cabinet	1000	303
	Battery Backup	695	204
DKT with BPCI	CIX40 Cabinet	1000	303
	Battery Backup	500	151
DKT with BPCI and BHEU	CIX40 Cabinet	1000	303
	Battery Backup	500	151
DSS3060 or DSS2060	CIX40 Cabinet	1000	303
	Battery Backup	675	204
DDCB3A	CIX40 Cabinet	165	50
	Battery Backup	500	151
BATI, RATI	CIX40 Cabinet	1000	303
	Battery Backup	1000	303
DKT with 1 ADM	CIX40 Cabinet	675	204
	Battery Backup	165	50
DKT with 2 ADMs	CIX40 Cabinet	500	151
	Battery Backup	33	10

1. Battery backup applies to instances when the system is being powered by batteries exclusively.
2. CIX40 Digital Telephones do not support Speaker OCA.

Strata CIX100-S / CIX100 3

This chapter provides a system overview of the Strata CIX100-S and CIX100 telephone systems.

The Strata CIX100-S/CIX100 systems are compact systems, yet they provide large system features (see Figure 3 and Table 7). They are designed for wall mounting and occupy very little space.

The CIX100 processor (ACTU3A) comes with 32 ports (licensed) and can grow to 112 ports by adding 4-port licenses.

The CIX100-S processor (ACTU3A-S) comes with 16 ports (licensed) and can grow to 32 ports by adding two eight-port licenses. Then, it can grow to 112 ports with four-port licenses.

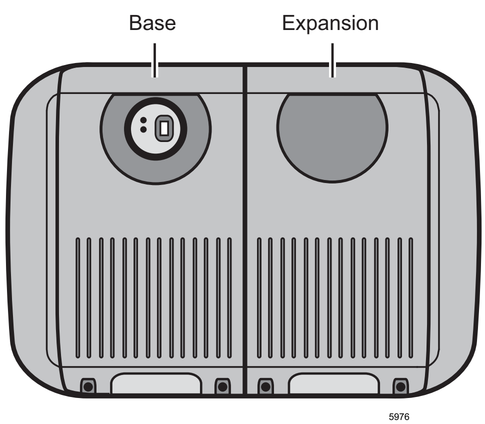


Figure 3 CIX100-S / CIX100 Base / Expansion Cabinets

Note The Strata CIX100-S and CIX100 system capacities depend on the licenses stored on the system processor and the hardware described in this chapter.

Important! *The Strata CIX100-S uses the same hardware and configuration as the Strata CIX100 with a few exceptions. Whenever the CIX100 is mentioned in this book, it applies to both the CIX100-S and CIX100 unless specified otherwise.*

Each ACTU3 basic processor can be configured with a one or two cabinet system. A single (Base) cabinet system supports a combination of up to 64 Central Office (CO) lines and stations, while a two cabinet system (Base and Expansion) can support up to 112 CO lines and stations.

System line and station capacity is expanded by adding CO line and station circuit cards and port licenses into its universal slot architecture.

The CIX100 easily connects to outside public and private telephone lines. All of the telephones (stations) tied to the system can have direct access to each other, as well as to the public and private network. All lines, stations, and options are tied together through the cabinets.

Table 7 CIX100-S/CIX100 Cabinet Specifications

Cabinet	Weight[1]	Height	Width	Depth
Base Cabinet (CHSUB112)	19.4 lbs.	14.6 in.	11.9 in.	10.2 in.
Base + Expansion Cabinet (CHSUE112)	34.6 lbs.	14.6 in.	19.9 in.	10.2 in.

1. Weight includes the processor card in the Base Cabinet and four universal circuit cards in each cabinet.

CIX100-S and CIX100 Processors

Each system operates with one processor card (ACTU3A-S for CIX100-S, ACTU3A for CIX100) that installs in a dedicated slot of the Base Cabinet. The processors incorporate the following hardware features.

CPU/Memory

Either processor card uses a high-speed, 32-bit, RISC processor, Dynamic Random Access Memory (DRAM) working memory, Static Random Access Memory (SRAM) with lithium battery for memory back-up, and flash program memory.

Large Scale Integrated (LSI) Circuits

The processors each have LSI circuits that support the following:

- 16 DTMF receiver hardware processor are built into the ACTU3. Five or more DTMF receivers require appropriate licenses.
- 16 Busy Tone (BT) detector circuits for Auto Busy Redial (ABR) are built into the ACTU3.
- 64 built-in conference circuits.
- Built-in, adjustable, digital, volume PAD technology enables audio volume to be adjusted in eight steps to compensate for conference and/or CO line network losses.

Memory Protection Battery

If commercial AC power is lost or if a system is moved or stored without power, the processor has an on-board battery that protects data and the customer's programmed configuration from memory loss. This information will be maintained in a powerless system for at least six years.

Relay Control Interface

An on-board terminal strip provides an interface to a normally open relay contact which can be programmed to control a Night Bell or door lock or to mute BGM during an external page.

External Page Interface

A 600 ohm RCA jack is built into each processor to interface with a Toshiba External Amplified Speaker (BESCB) or a customer-supplied page amplifier and speaker(s) for external paging, night ring over external page, and external BGM applications.

Music-on-hold/Background Music Interface

A 600-ohm RCA jack and volume controls are built into each processor to interface with Music-on-hold and/or Background Music (BGM) sources (one of the jacks is for future use). With the CIX100, you can have up to 15 MOH/BGM source interfaces by adding:

- Up to two BIOU circuit cards, each provides three MOH/BGM input sources
- An RSTU circuit card that provides up to eight MOH/BGM input sources

Secure Digital Memory

Each processor has an on-board SD memory card slot. A SD flash memory card can be inserted into the slot to backup and restore customer program data. It also makes it easy to upload operating system data for software upgrades and is used for maintenance functions (see "System Fault Finding and Diagnostics" on page 152 for more details).

Built-in Ethernet Connection

The ACTU processor has an on-board Ethernet 100Base-T Ethernet interface for connection to Open Architecture Computer Telephony Interface (CTI) applications. This provides extensive call control and telephone support for CTI applications, centralized Call Accounting, IPedge Application Server integration for Voice Mail, and the Meet-Me Audio Conference integration. The Ethernet Network Interface Card (NIC) port also enables connection to the following:

- CIX Attendant Console
- ACD server
- Local and Remote Network eManager PC
- Soft Key Control of Voice Mail features

CIX100 Processor Optional Subassemblies

Optional subassemblies can be attached to the ACTU3A-S or ACTU3A processors to provide additional features. The subassemblies are:

- **AMDS (Modem**) – Provides a 33.6Kbps/V.34 modem for point-to-point local or remote connection to the Network eManager administration PC.
- **BSIS (Serial Port Interface)** – Provides up to two RS-232 interface ports for SMDR interface to Call Accounting devices, SMDI or Toshiba Proprietary interface to Voice Mail devices, and two future applications.

CIX100 Cabinet Slots

Base Cabinet

The Base Cabinet has one dedicated slot used for the system processor card and four universal slots (S101~S104), that can accommodate station, line or option circuit cards. It also houses a power supply that is packaged with the cabinet.

Expansion Cabinets

One expansion cabinet has four universal circuit card slots (S105~S108) that can accommodate station, line or option circuit cards. It also houses a power supply that is packaged with the cabinet.

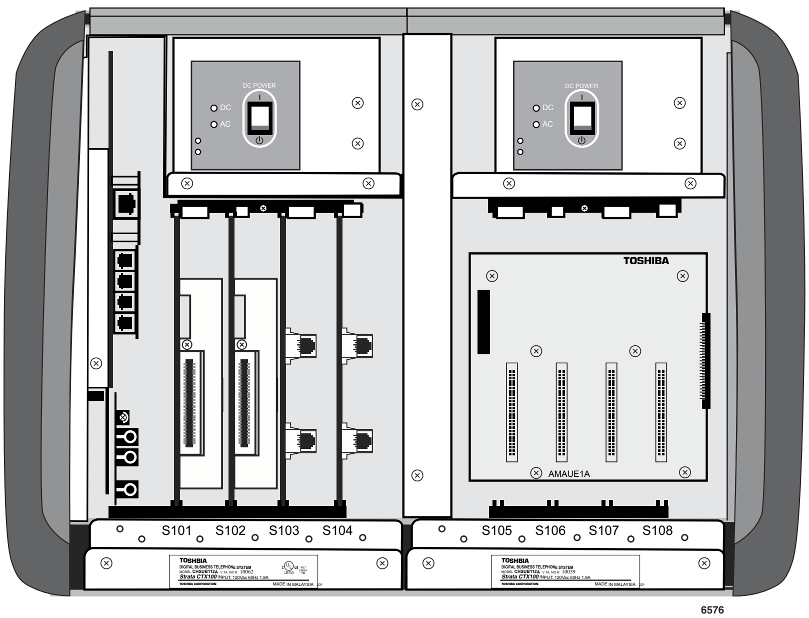


Figure 4 Base and Expansion Cabinet Interior

Strata CIX200 4

The Strata CIX200 supports 192 ports and can be configured as a one or two cabinet system, consisting of a Base Cabinet and one optional Expansion Cabinet. One processor model supports all Strata CIX200 configurations from smallest to largest.

These cabinets are designed specifically to fit standard (customer-supplied) 19” racks. The 19” rack and rack screws must be supplied by the dealer. Interface cables connect directly to the front panel or exit the cabinet through a port on the right side of the cabinet.

The CIX200 uses a single-card common control processor unit (LCTU2A), with a high-speed 32-bit RISC processor, 64 MB DRAM working memory, SRAM with lithium battery on-board flash program memory, and removable Toshiba Secure Digital flash memory.

Base Cabinet

The Base Cabinet (CHSUB192A) has four slots. The upper left slot, labeled B101 is reserved for the common control unit. The other three slots (labeled S101, S102 and S103) are universal card slots capable of hosting any of the station, line, and option interface PCBs compatible with the Strata CIX systems. The Base Cabinet also has a dedicated LSLU slot located at the back of the cabinet for two analog ports with two connectors on the front of the cabinet.

Expansion Cabinets

One Expansion Cabinet (CHSUE192A) can be added to increase the system station and CO line capacity. The Expansion Cabinet has four universal slots for interface cards (labeled S201 ~ S204). Compatible interface cards include the MIPU IP interface cards and the current Strata CIX interface cards that support CO lines, digital telephones, analog stations, etc.

Refer to “Capacities” on page 185 for cabinet slot and station/line capacities.

Basic Specifications

The following table lists the parts and basic specifications of the CIX200 Rack Mount Cabinets.

Dimensions of each cabinet – Base (CHSUB192A) and Expansion (CHSUE192A)	Height: 2U or 3.5 inches (88.9 mm) Width: 17.3 inches (440mm – without bracket); 19 inches (483 mm – with bracket) Depth: 16.1 inches (410mm) Weight: 16 lbs (7.2 kilograms)
Power Supply Unit (PSU)	Initially Built-in
19" Rack Installation Dimension	IEC297-1 (EIA RS 310-D) 465.1mm (front face screw pitch – width)
Installation	Rack-mount (preferred) or Table top. **Important!** *Do not place the equipment on the floor or in a dusty place.*

Power Backup

An Uninterruptible Power Supply (UPS) is required for power backup on a CIX200. The UPS is similar to the ones used for computer systems and networking equipment.

Configuration

The two primary considerations in system configuration are Card Slot Use and Power Factor.

Card Slot Use

The card slots can each support up to 32 ports. There are a few card position requirements based on cable lengths. These are discussed in detail in the *Strata CIX Installation Manual*. These requirements do not affect the system capacities.

In general, determine the number and type of stations needed, the number and type of trunks, and any additional requirements. Other requirements include voice mail, power fail transfer, tie lines, etc. The total number of ports required must not exceed 192.

Figure 5 Two Cabinet CIX

Power Factor

Power Factors for the CIX200 circuit cards are shown below. For all others please refer to the Installation and Maintenance Manual.

- LCTU1A: No effect on Power Factor calculations
- MIPU: PF = 0.2
- LSLU1A: PF = 1.5

There are no Power Factor considerations for the new IP Telephones (IPT2010/2020/2008), add-on modules (IADM2020 and IDSS2060), and DSS consoles because the power is not supplied from CIX200 cabinet.

The system (-24V) Power Factor is 67 per cabinet. There is no +5V power factor consideration for the CIX200. For an example on how to use the System Power Factor and for worksheets, refer to the *Strata CIX Installation and Maintenance* manual.

CIX200 PCBs

Two circuit cards are specific to the CIX200 systems; the LCTU processor card and the LSLU, a two-circuit, single-line, telephone interface card. The MIPU24 IP interface card with 24 channel works with the CIX200.

PCB Option Considerations

Strata CIX PCBs can be configured for a variety of hardware and software options. Hardware options are defined as either internal (generally related to optional PCB subassemblies) or external (related to connection of peripheral equipment such as background music, voice mail, etc.).

Some PCBs must have hardware options, such as jumpers or switches, set prior to installation of the PCB in the cabinet.

Common Control Processor Unit (LCTU)

The system operates with the LCTU2A processor PCB that installs in a dedicated slot of the Base Cabinet. The Strata CIX200 processor has the following standard components built in:

- Secure Digital (SD) memory card interface. A SD flash memory card can be inserted to backup and restore customer program data. It also makes it easy to upload operating system data for software upgrades and maintenance functions.

Note LCTU1A uses Smart Media (SM) flash memory.

- One Music-on-hold/Background Music Interface is provided by an RCA jack and a volume control that are built into the processor to interface with a Music-on-hold and/or Background Music source. Up to 15 MOH/BGM sources are supported by adding up to two BIOU option cards, each provides three MOH/BGM input sources, and a BSTU card that provides up to eight MOH/BGM input sources. MOH/BGM source volume adjustment is controlled by software programming.
- Network Interface Card (NIC) Ethernet 100Base-T Ethernet port for IP management and connection to Open Architecture Computer Telephony Interface (CSTA) applications. This provides extensive call control and telephone support for CTI applications.
- 16 built-in DTMF receivers (five or more DTMF receivers require licenses).
- 16 built-in Busy Tone (BT) detectors for Auto Busy Redial (ABR).
- 64 built-in conference circuits.
- Built-in, adjustable, digital volume PAD technology enables audio volume to be adjusted in eight steps to compensate for conference and/or CO line network losses.
- Page connector – External Page interface.
- Relay connector – Relay Control interface.
- Memory Protection Lithium Battery that protects data and the customer’s programmed configuration from memory loss. This information will be maintained in a powerless system for at least six years.

Optional subassemblies can be added to the LCTU processor PCB to enable system expansion and provide additional features. The subassemblies are:

BSIS interface PCB attaches to the LCTU to provide up to four RS-232 interface ports for SMDR Call Accounting, and SMDI or Toshiba Proprietary Voice Mail interface.

AMDS (Modem) – Provides a 33.6Kbps/V.34 modem for point-to-point local or remote connection to the Network eManager administration PC.

Analog Standard Station Interface Unit (LSLU1A)

The optional 2-circuit LSLU card provides for connection of two analog devices, such as faxes, modems, standard telephones, etc., without message waiting. The LSLU installs into a dedicated mount in rear of Strata CIX Base Cabinet behind the processor slot. The LSLU does not occupy any of the universal slots of the cabinet, so it does not reduce the maximum capacity of the system. The two analog station ports are accessible on the front of the base cabinet via modular connectors.

Modular or Amphenol Connections

The Strata CIX200 provides a choice of using either RJ45 modular connectors or amphenol connectors for terminating CO line and station interfaces.

- Modular connectors provide an IT friendly environment using RJ45 cables and patch panel connections in data racks.
- Amphenol connectors provide efficient connection to existing voice cables terminated on the wall on 110 or 66 blocks.
- Either way, the installation technician can exercise his preference or the customer's preference for the type of connection that will take less time to install or save the customer money.

Modular RJ45 connectors are provided by adding optional extenders and modular faceplates to CIX interface cards used in the Strata CIX200. This enables the modular interface cables to be connected at the front panel of the CIX200 cabinet.

Amphenol connectors are provided by using CIX interface cards in the Strata CIX200 without extender cards. In this configuration, amphenol interface cables are routed through the side of the cabinet and a flat face plate covers the slot housing the card. See the *Strata CIX Installation and Maintenance Manual* for more details on Modular or Amphenol connections.

The Strata CIX200 supports two different types of optional extenders for the other CIX interface cards, depending upon the type of connectors that exist on the card.

Extenders

The CIX system uses two types of extenders. One type is connected between the interface card and the cabinet backplane. This type of extender brings the interface connectors to the front of the cabinet. The correct front plate for the interface card is packaged with the extender. These extenders are optional.

The second type of extender plugs into the interface card front connector and presents different connectors to the front plate. Some of this type are optional, such as the extender that connects to the 25-pair Amphenol connector on the station card and presents eight or 16 modular connectors to the front panel. Refer to Table 8.

Power Failure Transfer Unit (LPFU1A)

The optional 8-circuit LPFU card when used with the RCOU and RSTU cards can automatically connect the CO line circuits to the Single Line Telephones (SLT) when system power fails. The LPFU card installs onto the CO line card as an extender, so it shares the same cabinet slot. The RCOU card must be in the slot above.

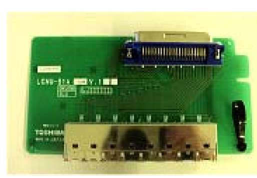
LCNU-A1A, LCNU-B1A, LCNU-C1A, LCNU-D1A, LEXU, and LCNU-E1A

Table 8 CIX Extender Use

Extender	Used For	Required	Comments
LEXU-A1A	MIPU	Optional	Extender installed between card and chassis backplane. Interface card connectors are bought to the cabinet face.
LEXU-B1A	BPTU1A RDTU3A	Optional	
LCNU-A1A	BSTCIU1A	Optional	Plugs into the 50-pin Amphenol socket on the interface card. Extends modular connectors to the CIX200 chassis face.
LCNU-B1A	BWDKU1A BDKU/ BDKS	Optional	
LCNU-C1A	RCOU3A RGLU3A RDDU2A BCOCIU1A	Optional (See Comments	Plugs into the modular connectors on the interface card. Extends modular connectors to the CIX200 chassis face. The LCNU-C1A connects the CO lines and Caller ID in one connector. If not used the connections must made on the MDF.
LCNU-D1A	REMU1A	Optional	Plugs into the modular connectors on the interface card. Extends modular connectors to the CIX200 chassis face.
LCNU-E1A	RCIU2A	Required (See Comments)	The LCNU-E1A is a set of small cables. These are required only if the LCNU-C1A is used. Connects RCIU2A to the LCNU-C1A to connect the Caller ID and CO line together in one modular socket.
LPFU1A	Power Fail Transfer	Required	Required for power fail transfer operation. The LPFU1A is two extender cards and two cables. The CO and Station cards must be in adjacent slots.

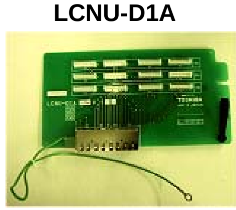
LCNU-A1A

LCNU-B1A

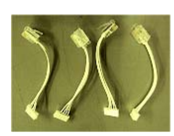


LCNU-C1A

LCNU-D1A



LCNU-E1A



Strata CIX670 and CIX1200 5

Strata CIX670

The Strata CIX670 system provides sophisticated telecommunication features in a modular system designed for growth. Its universal slot architecture enables you to select the combination of Central Office (CO) lines, stations, and peripheral options that best suit your needs.

The CIX670 BCTU basic processor can be configured for smaller systems as a one or two cabinet system with a capacity of up to 192 CO lines and stations combined. It can expand to support up to seven cabinets with a capacity of up to 672 CO lines and stations combined with the BCTU/BEXU expanded processor (see Figure 6).

System line and station capacity is expanded by adding processor expansion Circuit Cards, cabinets and line/ station circuit cards and station/line licenses.

The CIX670 easily connects to outside public and private telephone lines. All of the telephones (stations) tied to the system can have direct access to each other as well as to the public and private network.

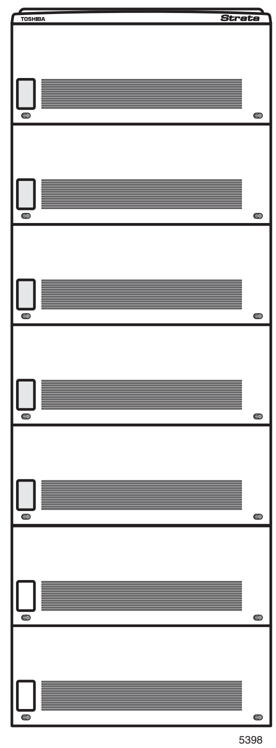


Figure 6 CIX 670 Base/ Expansion Cabinets

The Base Cabinet and optional Expansion Cabinets are the building blocks of the system. Each system has a Base Cabinet, and can have from one to six Expansion Cabinets. All lines, stations, and options are tied together through the cabinets. There are two types of cabinets, floor/wall mountable and rack mountable. See page 32 for details on rack mountable cabinets.

The overall weight and dimensions of the CIX670 cabinets are shown in Table 9.

Table 9 CIX670 Floor/Wall Mount Cabinet Specifications

Cabinet	Weight	Height	Width	Depth
Base Cabinet (CHSUB672)	31 lbs.	11.625 in.	26.5 in.	10.3 in.
Expansion Cabinet (CHSUE672)	29 lbs.	9.75 in.	26.5 in.	10.3 in.

CIX670 Processor Circuit Cards

The system operates with the BCTU only or the BCTU and BEXU processor circuit cards that install in dedicated slots of the Base Cabinet. The BCTU and BEXU processor incorporates the following on-board hardware features:

CPU/Memory

The CIX670 uses a high-speed, 32-bit, Reduced Instruction Set Computing (RISC) processor, Dynamic Random Access Memory (DRAM) working memory, Static Random Access Memory (SRAM) with lithium battery for back-up memory and flash program memory.

Large-scale Integrated (LSI) circuits

The processor has LSI circuits that support the following:

- BCTU provides 16 built-in DTMF receivers; 32 available using the BCTU and BEXU. For five or more DTMF receivers, appropriate licenses are required.
- BCTU provides 16 built-in Busy Tone (BT) detectors for Auto Busy Redial (ABR); 32 available using the BCTU and BEXU.
- BCTU provides 64 built-in conference circuits; up to 96 conference circuits are available using the BCTU and BEXU.
- Built-in, adjustable, digital volume PAD technology enables audio volume to be adjusted in eight steps to compensate for conference and/or CO line network losses.

Memory Protection Battery

If commercial AC power is lost or if a system is moved or stored without power, the processor has an internal battery that protects data and the customer’s programmed configuration from memory loss. This information will be maintained in a powerless system for at least six years.

Music-on-hold/Background Music Interface

An RCA jack and volume control are built into the processor to interface with a Music-on-hold and/or Background Music source. With the CIX670, you can have up to 15 MOH/BGM sources by adding:

- Up to two BIOU circuit cards, each provides three MOH/BGM input sources.
- A BSTU or any analog station card that provides up to eight MOH/BGM input sources.
- MOH/BGM source volume adjustment is controlled by software programming.

Secure Digital (SD) Memory

The BCTU processor has an on-board Secure Digital card slot. A SD flash memory card can be inserted to backup and restore customer program data. It also makes it easy to upload operating system data for software upgrades and is used for maintenance functions (see “System Fault Finding and Diagnostics” on page 152 for more details).

Network Interface

The BCTU processor has an on-board Ethernet 100Base-T Ethernet interface for connection to Open Architecture Computer Telephony Interface (CTI) applications. This provides extensive call control and telephone support for CTI applications. The Ethernet Network Interface Card (NIC) port also enables connection to the following:

- CIX Attendant Console
- ACD server
- Local and Remote Network eManager PC
- Soft Key Control of Voice Mail features

CIX670 Processor Optional Subassemblies

Optional subassemblies can be added to the BCTU processor card to enable system expansion and provide additional features. The subassemblies are:

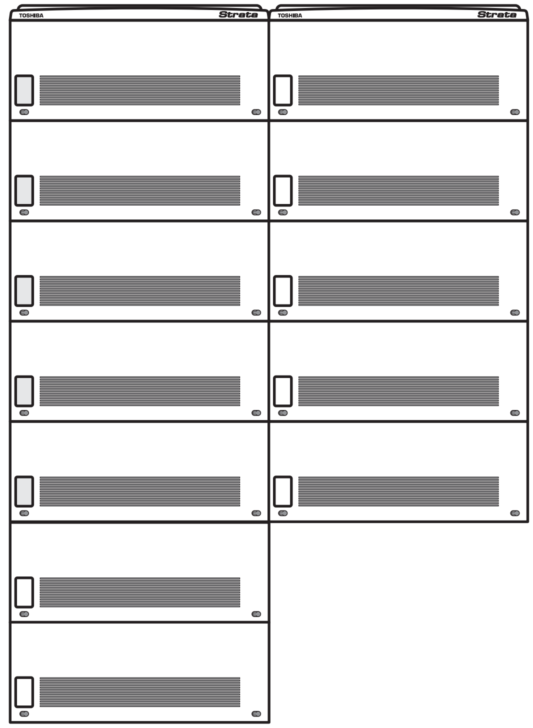
- AMDS (Modem) – Attaches to the BCTU to provide a 33.6Kbps/V.34 modem for point-to-point local or remote connection to the Network eManager administration PC.
- BSIS interface card which attaches to the BCTU to provide up to four RS-232 interface ports for SMDR Call Accounting and SMDI or Toshiba Proprietary Voice Mail interface.

Strata CIX1200

The Strata CIX1200 system provides sophisticated telecommunication features in a modular system designed for growth. Its universal slot architecture enables you to select the combination of Central Office (CO) lines, stations, and peripheral options that best suit your needs.

The CIX1200 HCTU basic processor can be configured for smaller systems as a one or two cabinet system with a capacity of up to 192 CO lines and stations combined. It can expand to support up to seven cabinets with a capacity of up to 672 CO lines and stations combined with the HCTU/HEXAU expanded processor. This configuration provides the same number of ports as the CIX670. However, it provides expanded feature capacity for items such as LCR tables, Network routing tables, etc.

The Strata CIX1200 can expand to support twelve cabinets (shown right) with a capacity of up to 1,152 CO lines and stations combined with the HCTU/HEXBU expanded processors.



CIX 1200 Base/Expansion Cabinets

System line and station capacity is expanded by adding processor expansion circuit cards, cabinets and line/station circuit cards and station/line licenses.

The CIX1200 easily connects to outside public and private telephone lines. All of the telephones (stations) tied to the system can have direct access to each other as well as to the public and private network.

The Base Cabinet and optional Expansion Cabinets are the building blocks of the system. Each system has a Base Cabinet, and can have from one to 11 Expansion Cabinets. All lines, stations, and options are tied together through the cabinets. There are two types of cabinets, floor/wall mountable and rack mountable. See page 32 for details on rack mountable cabinets.

The overall weight and dimensions of the CIX1200 cabinets are shown in Table 10.

Table 10 CIX1200 Floor/Wall Mount Cabinet Specifications

Cabinet	Weight	Height	Width	Depth
Base Cabinet (CHSUB672)	31 lbs.	11.625 in.	26.5 in.	10.3 in.
Expansion Cabinet (CHSUE672)	29 lbs.	9.75 in.	26.5 in.	10.3 in.

CIX1200 Processor Circuit Cards

The system operates with the HCTU (2 cabinets) only or the HCTU and HEXAU (7 cabinets) or HCTU and HEXBU (12 cabinets) processor circuit cards that install in dedicated slots of the Base Cabinet. The processor cards incorporate the following on-board hardware features:

CPU/Memory

The CIX1200 uses a high-speed, 32-bit, Reduced Instruction Set Computing (RISC) processor, Dynamic Random Access Memory (DRAM) working memory, Static Random Access Memory (SRAM) with lithium battery for back-up memory and flash program memory.

Large-scale Integrated (LSI) circuits

The processor has LSI circuits that support the following:

- HCTU provides 16 built-in DTMF receivers; 32 available using the HCTU and HEXAU and 48 using the HCTU and HEXBU. For five or more DTMF receivers, appropriate licenses are required.
- HCTU provides 16 built-in Busy Tone (BT) detectors for Auto Busy Redial (ABR); 32 available using the HCTU and BEXAU and 48 using HCTU and HEXBU.
- HCTU provides 64 built-in conference circuits; up to 96 conference circuits are available using the HCTU and HEXAU and up to 128 conference circuits using HCTU and HEXBU.
- Built-in, adjustable, digital volume PAD technology enables audio volume to be adjusted in eight steps to compensate for conference and/or CO line network losses.

Memory Protection Battery

If commercial AC power is lost or if a system is moved or stored without power, the processor has an internal battery that protects data and the customer's programmed configuration from memory loss. This information will be maintained in a powerless system for at least six years.

Music-on-hold/Background Music Interface

An RCA jack and volume control are built into the processor to interface with a Music-on-hold and/or Background Music source. With the CIX1200, you can have up to 15 MOH/BGM sources by adding:

- Up to two BIOU circuit cards, each provides three MOH/BGM input sources.
- A BSTU or any analog station card that provides up to eight MOH/BGM input sources.
- MOH/BGM source volume adjustment is controlled by software programming.

Secure Digital (SD) Memory

The HCTU processor has an on-board Secure Digital card slot. A SD flash memory card can be inserted to backup and restore customer program data. It also makes it easy to upload operating system data for software upgrades and is used for maintenance functions (see "System Fault Finding and Diagnostics" on page 152 for more details).

Network Interface

The HCTU processor has an on-board Ethernet 100Base-T Ethernet interface for connection to Open Architecture Computer Telephony Interface (CTI) applications. This provides extensive call control and telephone support for CTI applications. The Ethernet Network Interface Card (NIC) port also enables connection to the following:

- CIX Attendant Console
- ACD server
- Local and Remote Network eManager PC
- Soft Key Control of Voice Mail features

CIX1200 Processor Optional Subassemblies

Optional subassemblies can be added to the HCTU processor card to enable system expansion and provide additional features. The subassemblies are:

- AMDS (Modem) – Attaches to the HCTU to provide a 33.6Kbps/V.34 modem for point-to-point local or remote connection to the Network eManager administration PC.
- BSIS interface card which attaches to the HCTU to provide up to four RS-232 interface ports for SMDR Call Accounting and SMDI or Toshiba Proprietary Voice Mail interface.

CIX670 and CIX1200 Cabinet Slots

Base Cabinet

The Base Cabinet has two dedicated slots used for the system processor cards and eight universal slots, labeled "S101~S108," that can accommodate station, CO line or option circuit cards (see Figure 7). It also houses a power supply.

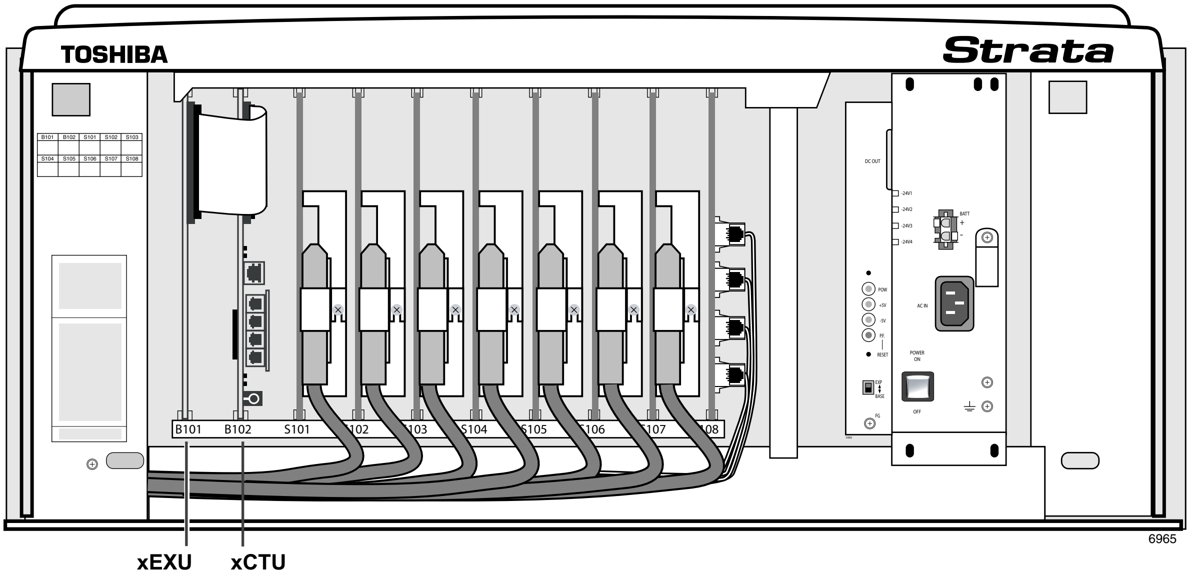


Figure 7 Base Cabinet Interior

Expansion Cabinets

One to twelve Expansion Cabinets can be added to increase the system station and CO line capacity. Each expansion cabinet provides 10 slots (S_01~S_10). Figure 8 shows an Expansion Cabinet.

Refer to the following section for cabinet slot and station/line capacities. Table 52 and 53 show the number of stations and CO lines allowed when additional cabinets and circuit cards are used.

Note CIX1200 expansion cabinets 8 through 12 must be connected with fiber optic cables (see page 34).

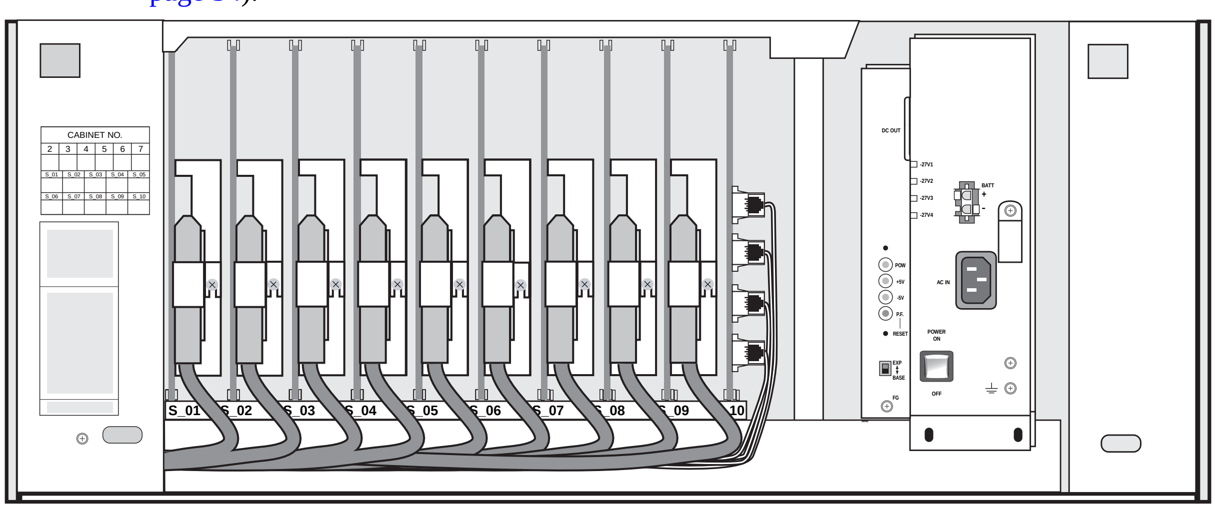


Figure 8 Expansion Cabinet Interior

Strata CIX670 and CIX1200 Rack Mount

The Strata CIX670 and CIX1200 Rack Mount cabinets consist of a Base cabinet (CRSUB672A) and Expansion cabinets (CRSUE672A). These cabinets are deeper, narrower, and designed specifically to fit standard (customer-supplied) 19" racks (shown right). The cabinets are made of plated sheet metal, dark gray in color with black cover plates. These cabinets support the same features as the floor/wall-mountable CIX670 and CIX1200 cabinets.

A CIX670 system can expand from one to seven cabinets using the same processors and interface cards as the CIX670 floor/wall-mountable cabinets. The CIX1200 system can expand from one to twelve cabinets using the same processors and interface cards as the CIX1200 floor/wall-mountable cabinets.The power factors and slot configuration rules are also the same as the floor/wall-mountable cabinets.

The difference between the floor/wall mount and the rack mount cabinets is the size for the Base and Expansion cabinets, the power supply that is installed in the cabinets, and a unique power strip for rack mount cabinets (BRPSB120A and the 240V version).

Toshiba does not support mixing the floor/wall-mountable cabinets with the rack mountable cabinets. The 19 inch-wide rack and rack screws must be supplied by the dealer. Interface cables plug into the front of the station and trunk cards and fold under each cabinet to exit the rear of the cabinets.

Basic Specifications

The following table lists the parts and basic specifications of the CIX1200 Rack Mount Cabinets.

Dimensions of Base (CSRUB672A) and Expansion (CSRUE672A) Cabinet	Height: 6U or 10.5 inches (267mm) Width: 1.58 feet (483mm—with bracket) Depth: 1.17 feet (358mm)	
Weight of Base and Expansion Cabinet	22.04 lbs. (10 kg)	
Power Supply Unit (PSU)	BRPSU672A (initially built in) – can also be ordered for spares	
19" Rack Installation Dimension	IEC297-1 (EIA RS 310-D) 465.1mm (front face screw pitch – width)	
Installation	Cannot be floor or wall mounted.	
Optional Equipment	Power Strip Box[1]	BRPSB120A
		BRPSB240A
	Reserve Power Battery Distribution Box[2]	BBDB1A
	From PSU to Battery Cable	PBTC1A-3M
	From BBDB to Battery Cable	BBTC1A-2.0M
	AC240V Power Supply Cord	BACL240A

1. Power strip boxes for floor/wall-mountable cabinets cannot be used for Rack Mount cabinets.
2. Reserve batteries are connected using the same battery distribution box and battery cables as the CIX670 and CIX1200 floor/wall-mountable cabinets.

Base Cabinet

The CIX670 and CIX1200 rack mount base cabinet is similar to the Strata CIX670 and CIX1200 wall mount base cabinet. It has two dedicated slots used for the system processor cards and eight universal slots, labeled "S101~S108," that can accommodate station, CO line or option circuit cards. It also houses a power supply. See Figure 9.

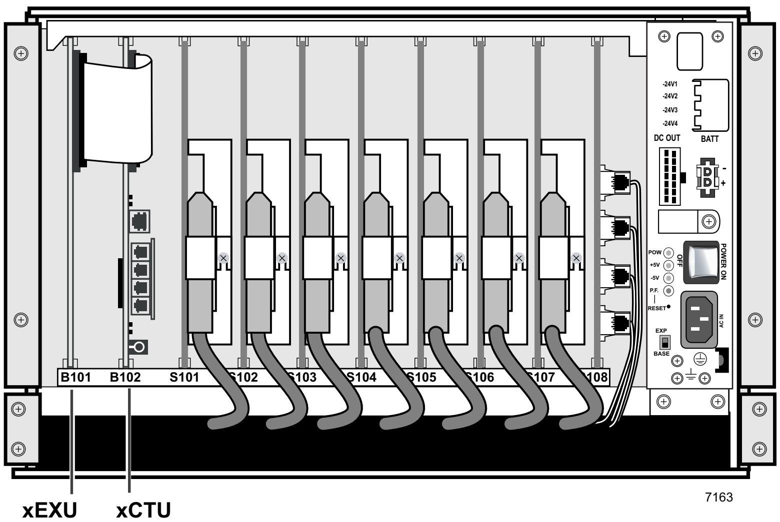


Figure 9 Strata CIX670 and CIX1200 Rackmount Base Cabinet Interior

Expansion cabinet

Each expansion cabinet provides 10 slots (S_01~S_10). Figure 10 shows an Expansion Cabinet.

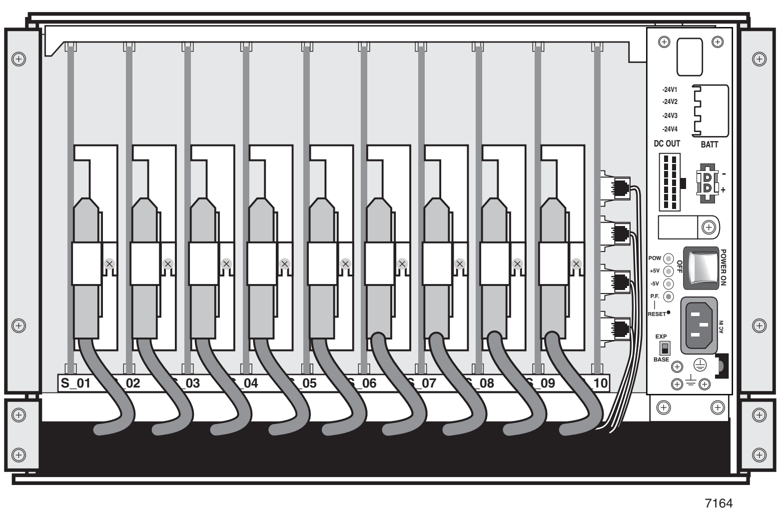


Figure 10 Strata CIX670 and CIX1200 Rackmount Expansion Cabinet Interior

CIX670 and CIX1200 Remote Expansion Cabinet

A CIX670 and CIX1200 Expansion Cabinet can be located between a few inches and up to three kilometers (1.86 miles) from its Base Cabinet. Remote Expansion Cabinets are enabled by the fiber optic RRCU interface cards. One RRCU connects to up to two Data Cables and applies the inter-cabinet signal to a multi-mode fiber-optic pair. One fiber pair can support one or two Expansion Cabinets in one remote location using one RRCU in the Base Cabinet and another in one of the Remote Expansion Cabinets.

CIX1200 expansion cabinets 8 through 12 must be connected as remote cabinets using RRCU interface cards and multi-fiber optic cables.

Fiber connected cabinets support the BIOU for external Page Zones, Night Bell, etc., and all CO line and trunk interface circuit cards. Network clock synchronization can only be derived from digital trunks installed in the Base Cabinet or a expansion cabinet connected with standard ribbon cables, not fiber cables.

A Remote Cabinet can support all circuit cards that can be installed in a local Expansion Cabinet, including digital trunk cards. However, the system cannot derive network clock synchronization from a digital trunk installed in a Remote Cabinet. This requires a digital trunk installed in the Base Cabinet or in a local Expansion Cabinet connected to the Base by a standard ribbon cable. For each Remote Cabinet location, local trunks may be required for correct 911 service.

Universal Slot Circuit Cards 6

Universal Printed Circuit Cards installed in the Strata CIX1200, CIX670, CIX200 or CIX100 cabinets provide interfaces for stations, lines, and peripherals. Each circuit card measures 7.5 x 5.5 inches (190 x 140 mm) and mounts in the slot with a 44-pin backplane connector. Circuit Card external connections to station equipment are made to the Main Distribution Frame (MDF) using industry-standard connectors.

Important! *This chapter does not apply to the Strata CIX40, except for GVPH, GIPU8, MIPU16 and MIPU24.*

Station, Line and Option Circuit Cards

The Circuit Cards are categorized as station, CO line or option cards (see Tables 11~13). Feature subassemblies that plug onto a universal slot circuit card are listed below the associated card. For details, see the *Strata CIX I&M Manual*.

Table 11 Station circuit cards

Internet Protocol Telephone (IPT) Interface Unit (MIPU, GIPU and GIPH)	
Provides IPT telephone circuits (IP5000-series, IPT2010-SD, IPT2020-SD and IPT2008-SDL) 1 100Base-TX RJ45 port Built-in Digital Signal Processor (DSP) 1 RS-232 maintenance port Network Address Translation (NAT) compatible for remote IP telephones when connected to a MIPU Enhanced version of MEGACO+ for Voice over IP Supports SIP telephones and UIP200 Simultaneously supports Line interfaces.	MIPU16 – supports 16 IP channels (supported by CIX40 R1 and R2 cabinets) MIPU24 – supports 24 IP channels (supported by CIX40 R2 cabinet) GIPU8, GIPH-X1A – supports 8 IP channels for CIX40 only Interface Options: LAN, Virtual Private Network (VPN) Internet, VPN WAN, Intranet.
Digital Telephone Interface Unit (ADKU, BDKU/BDKS, BWDKU1A)	
Provides digital telephone circuits for DP5000-series, DKT3000/3200-series digital telephones. Stand-alone digital cordless telephone DDM5060 console DDSS3260 console DDCB – Door Phone Stand-alone digital cordless telephone DDM console BATI – Digital Attendant Console	ADKU = 8 circuits (CIX100 only) BDKU = 8 circuits BDKS = 8 circuits BWDKU = 8 circuits

Table 11 Station circuit cards *(continued)*

Standard Telephone Interface Unit (LSLU - CIX200 only) (ASTU - CIX100 only)	
Provides two standard telephone circuits. Maximum number of ringers per circuit is one.	Interface Options: Standard telephones (no message waiting lamp control - no OPS (48V)) Other single-line devices Alternate MOH/BGM source Fax machines Voice mail devices
Standard Telephone Interface Unit (Card Slot Mount) (BSLU/BSLS - All CIX Systems)	
Provides eight (BSLU) or 16 (BSLU + BSLS) standard telephone circuits. Maximum number of ringers per circuit is one	Interface Options: Standard telephones (no message waiting lamp control) Other single-line devices Alternate MOH/BGM source Fax machines Voice mail devices No 48 Volt Off-premises Station support
Standard Telephone Interface Unit with Caller ID (Card Slot Mount) (BSTCIU - All CIX Systems, except CIX40)	
Provides eight standard telephone circuits with Caller ID. Maximum number of ringers per circuit is three. The BSTCIU2A offers disconnect supervision.	Interface Options: Standard telephones with Caller ID displays Message Waiting Lamp Control Off Premise Station (OPS) VM devices MOH / BGM
Standard Telephone Interface Unit (BSTU)	
Provides 8 standard telephone circuits. Stutter dial tone is provided for Message Waiting audible indication. The BSTU2A offers disconnect supervision.	**Interface Options:** Standard telephones Voice mail ports Off-premises stations Other similar devices Alternate MOH/BGM source Auto Attendant digital announcer Message Waiting Lamp Control Fax machines ACD announcer
-48 Volt Supply Internal Option (R48S)	
Attaches to BSTU and BSTCIU 48VDC circuit for up to 8 standard telephone circuits. Not available on LSLU, BSLU/BSLS, ASTU	Interface Options: Optionally interfaces to the BSTU and BSTCIU to extend loop length of standard telephones from 600 ohms to 1200 ohms. Required for OPS operation.
Voice Mail	
GVPH (CIX40 only)	Pre-programmed Voicemail Auto attendant circuit card. 4 Voice mail / Auto attendant ports default; 6 and 8 ports require LIC-2 GVPH 40 hours storage – 360 mail boxes Call record, Voice mail Call monitor and built-in modem
LVMU1A (CIX100, CIX200, CIX670, and CIX1200)	Pre-programmed Voicemail Auto attendant circuit card. 8 Voice mail / Auto attendant ports 40 hours storage – 360 mail boxes Call record, Voice mail Call monitor and built-in modem

Table 12 CO Line Circuit Cards

Strata Net Over VoIP Interface Unit (MIPU, GIPU and GIPH)	
Provides 32 IP Strata Net channels 1 100Base-TX RJ45 port 1 RS-232 maintenance port Based on IP Strata Net standard protocol (ECMA-336) Voice coding G.711/G.729A Built-in Digital Signal Processor (DSP) Simultaneously supports IPT Station interfaces	MIPU16 – supports 16 IP channels MIPU24 – supports 24 IP channels GIPU8, GIPH-X1A – supports 8 IP channels for CIX40 only **Interface Options:** LAN, Virtual Private Network (VPN) Internet, VPN WAN, Intranet.
SIP Trunking (MIPU)	
Provides 24 IP Strata Net channels 1 100Base-TX RJ45 port 1 RS-232 maintenance port Voice coding G.711/G.729A Built-in Digital Signal Processor (DSP) Simultaneously supports IPT Station interfaces	GIPU8 – supports 8 IP channels MIPU16 – supports 16 IP channels MIPU24 – supports 24 IP channels
Internet Protocol (IP) Interface Unit (BVPU)	
Provides 4 VoIP Circuits as E&M Tie lines 1 10Base-T port 1 RS-232 maintenance port H.323 standard for Voice over Internet Protocol (VoIP)	**Interface Options:** LAN, Internet, WAN.
Caller ID Interface Unit (RCIU2 / RCIS)	
Provides 4 Caller ID circuits. With RCIS: 8 circuits.	**Interface Options:** Provides Caller ID LCD display for analog loop or ground start lines with Caller ID. Requires: RCOU, RCOS, RGLU2, RGLU3 or PCOU. Not compatible with T1.
Caller ID Interface Subassembly (RCIS) Attaches to the RCIU2.	Same as RCIU2.
Direct Inward Dialing Interface Unit (RDDU)	
Provides 4 DID circuits.	**Interface Options:** DID analog lines.
Enhanced 911 CAMA Trunk Interface Unit (RMCU/RCMS)	
E911 CAMA circuits. Provides up to 4 CAMA trunk circuits. The RMCU/RCMS eliminates the need for connection of adjunct terminal adapter equipment to E911 CAMA trunks. Requires 1 or 2 RCMS circuit cards for 2 or 4 CAMA lines respectively.	E911 analog CAMA trunks.
CAMA Trunk Subassembly (RCMS) RCMS attaches to RMCU. Provides 2 E911 CAMA circuits. Up to 2 RCMSs per RMCU for 4 CAMA lines max. (1 RCMS comes packaged with the RMCU.)	Same as RMCU.
Ground/Loop Start Interface CO Line Interface Unit (RGLU3)	
Provides 4 ground or loop start line circuits. Each can be individually set for ground or loop start operation.	**Interface Options:** Analog loop or ground start analog lines.

Table 12 CO Line Circuit Cards *(continued)*

ISDN Primary Rate Interface Unit (RPTU and BPTU)	
Provides (1~8B + D), (1~16B + D), or (1~23B + D) channels (lines), depends on system programming. RPTU2 is required for Strata Net Networking.	**Interface Options:** ISDN PRI POTS FX Tie (senderized) Tie (cut through) OUTWATS (intra-LATA) and (inter-LATA) InWATS Strata Net
Loop Start CO Line Interface Unit (RCOU3A and RCOS3A)	
Provides 4 CO analog loop start line circuits. With RCOS, provides 8 CO analog loop start line circuits.	**Interface Options:** CO analog loop start lines
Loop Start CO Line Interface Subassembly (RCOS) Provides 4 additional Loop Start CO lines. 1 RCOS subassembly per RCOU.	Same as RCOU. RCOU/RCOS requires RCIU2/RCIS to support Caller ID.
Loop Start CO Line Interface with Caller ID Unit (BCOCIU1A)	
Provides four CO analog loop start line circuits. With BCOCIS1A, provides eight CO analog loop start line circuits. Caller ID interface is built-in, RCIU is not needed.	**Interface Options:** CO analog loop start lines (Does not support Type 2 Hook-Flash to answer a waiting call, does not provide dial pluse output)
Loop Start CO Line Interface Subassembly (BCOCIS1A) Provides 4 additional Loop Start CO lines with built-in Caller ID interface. Mounts on the BCOCIU1A.	Same as BCOCIU1A.
T1/DS-1 Interface Unit (RDTU)	
Provides T1 (DS1) Interface: 1~8, 1~16, or 1~24 channels (lines), depends on system programming.	**Interface Options:** T1 Loop start lines Ground start lines Tie lines (wink or immediate) DID/DOD lines (wink or immediate)
Remote Expansion Cabinet Unit (RRCU)	
Supports 2 CIX670 and CIX1200 remote cabinets. 62.5 mµ, multi-mode fiber.	Remote Cabinet not supported by main system reserve power.
Tie Line Unit (REMU2)	
Provides 4 analog Tie line circuits.	**Interface Options:** E&M Tie lines 2- or 4-wire transmission Type I and type II signaling Immediate start Wink start

Table 13 Option Circuit Cards

Option Interface Unit (BIOU)	**Interface Options:** Provides Paging output (600 ohm and 3-watt amp), 4 zone paging relays, three MOH interfaces and 4 control relays (Night Transfer and BGM mute).
Option Interface Unit (BSIS)	**Interface Options:** Provides four RS232 ports.

Voice Mail Cards

The LVMU1A and the GVPH are the only two circuit voice mail cards for the Strata CIX systems. The GVPH is a plug and play voice mail circuit card specifically for the Strata CIX40 cabinet. For more details on the GVPH refer to "GVPH – Integrated Voice Mail" on page 7. The LVMU1A is an integrated voice mail circuit card that installs in a slot (except a processor slot) of the Strata CIX100, CIX200 or CIX670 system. The LVMU1A has a 40 hour storage capacity and a maximum capacity of 8 ports.

The LVMU1A's features include:

- 2, 4, 6 or 8 port capacity with simplified licensing via eManager
- 40 hours of voice storage
- 360 mailboxes
- Messages per mailbox: New – 159, Saved – 160, Urgent – 159
- Call Monitor – Allows users to monitor callers leaving messages in their voice mailbox, with the option to retrieve the caller for a live conversation.
- Call Record – Gives users the ability to record live conversations from their Toshiba DKT or IP telephones which are subsequently saved as voice messages.
- LCD Soft Key Control – User control of voice mailbox prompting with the aid of soft keys on their Toshiba LCD telephone screen.
- Battery backup – Battery backup protection is from the Strata CIX
- On board remote access modem
- Bilingual Support (English / Spanish)

The LVMU1A has direct backplane communication with the host system. It does not require any additional hardware and is easy to install. Caller information is presented directly from the CIX processor. It comes pre-configured for installation into the Strata CIX100. This includes telephone system configurations as well as pre-programmed number of user mailboxes.

Licenses

The Strata CIX software that supports the LVMU1A comes equipped with two port licenses for the LVMU. Additional port licenses are available in two port increments for the CIX. These port licenses are licensed in the CIX and not the LVMU.

After installing the LVMU1A, dealers need only apply additional voice mail port licenses to increase the LVMU1A's capacity to the desired number of ports. All licensing is applied via the eManager administration terminal, while voice mail programming continues to be managed by the Stratagy UAdmin software.

Software

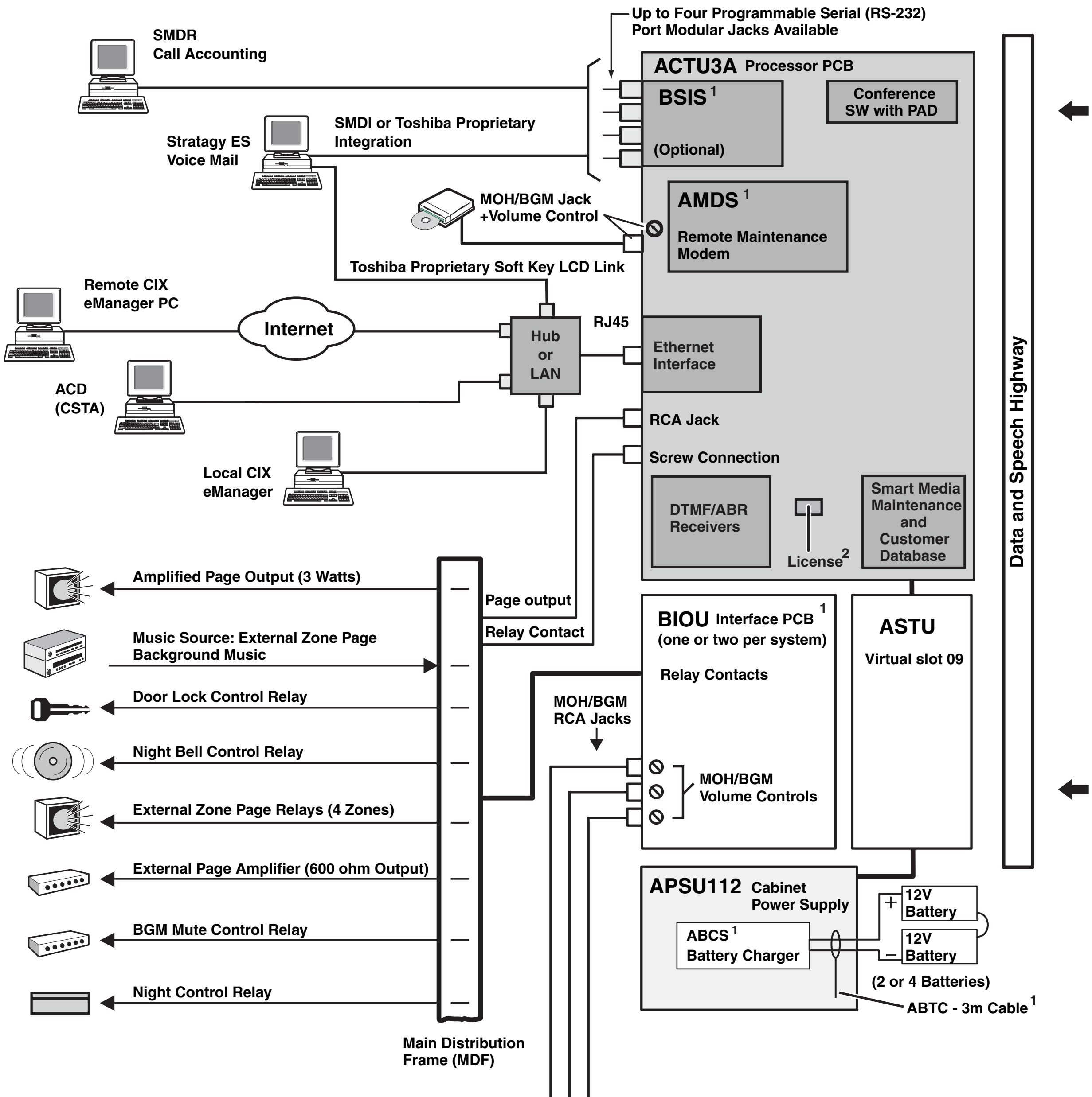
The software requirements for the LVMU1A are: Strata CIX Software R4.2 – MP026 or higher; eManager Version 4.20A09 or higher; Stratagy VM Manager or UAdmin Version 2.1

Port Upgrades

Each Strata CIX Release 4.2 processor is equipped with two LVMU port licenses. Additional voice mail ports require FYI licensing. Upgrading the number of voice ports on the LVMU1A is as easy as upgrading the CIX processor (R4.2 MP026 or higher is required). Upgrade licenses are purchased via normal processes and then activated through FYI Licensing. The activation code is applied to the Strata CIX Processor via eManager. The license is stored on the CIX processor rather than the LVMU1A.

Functional Block Diagrams

The Functional Block Diagrams show the circuit cards and interface connectors used for connecting the stations and peripherals (see Figures 11~15).



Notes

1. Optional.
2. License Control.

On this page, all equipment, except Stratagy ES, connected to the system processor PCBs and BIOU PCB is customer-supplied.

Figure 11 CIX100 System Processor and Optional Interface Circuit Cards

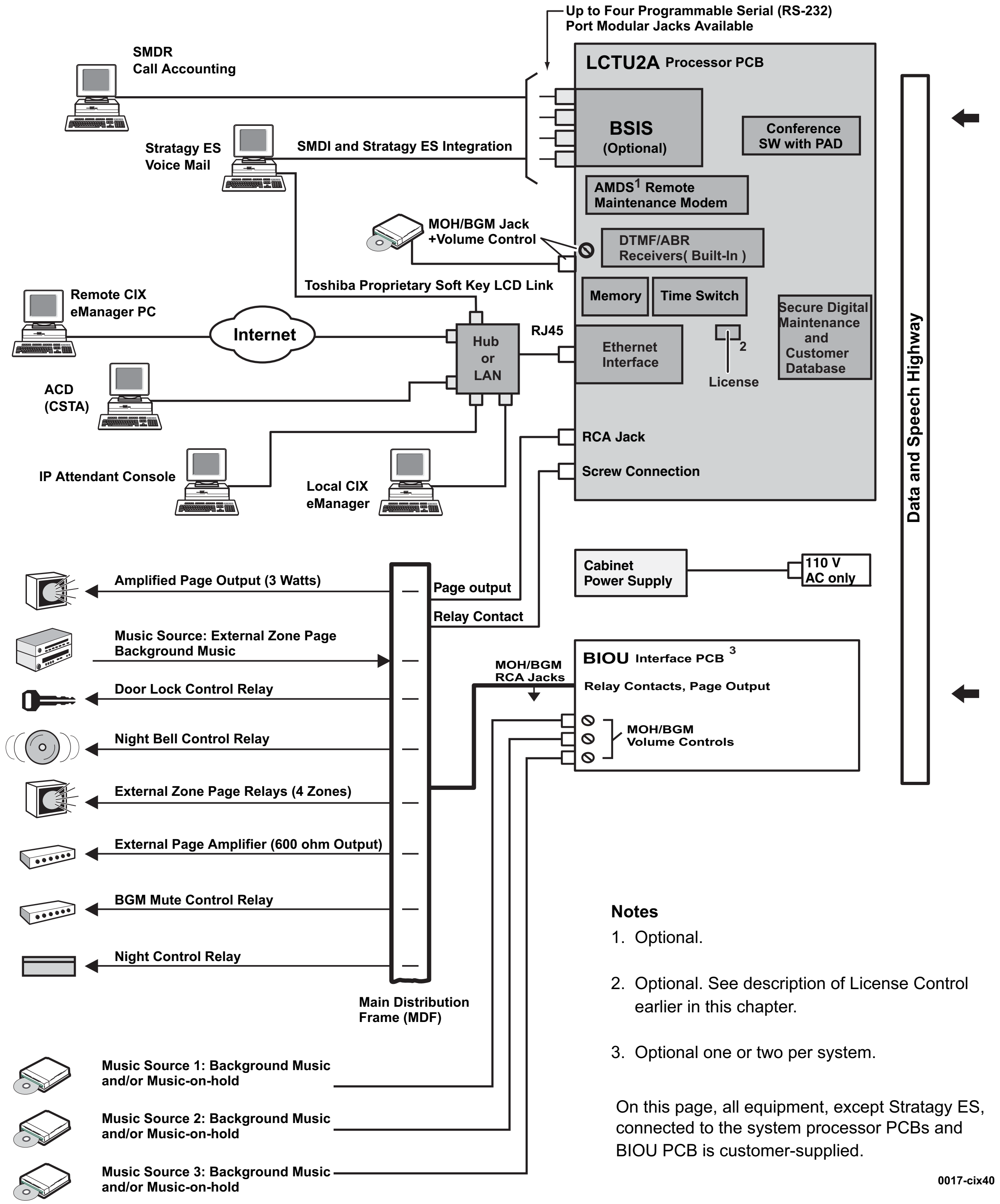
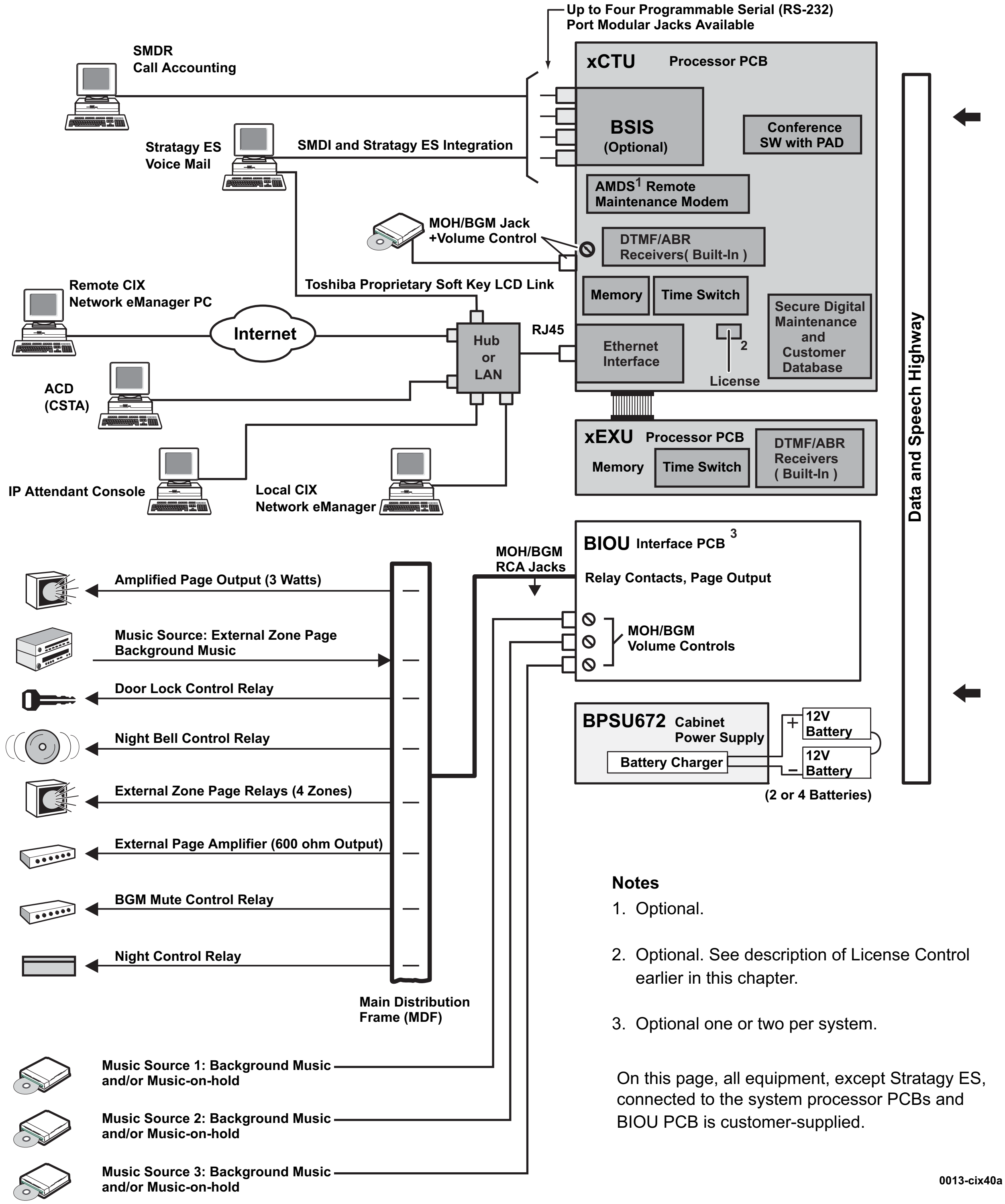


Figure 12 CIX200 System Processor and Optional Interface Circuit Cards



Notes

1. Optional.
2. Optional. See description of License Control earlier in this chapter.
3. Optional one or two per system.

On this page, all equipment, except Stratagy ES, connected to the system processor PCBs and BIOU PCB is customer-supplied.

Figure 13 CIX670 and CIX1200 System Processor and Option Interface Circuit Cards

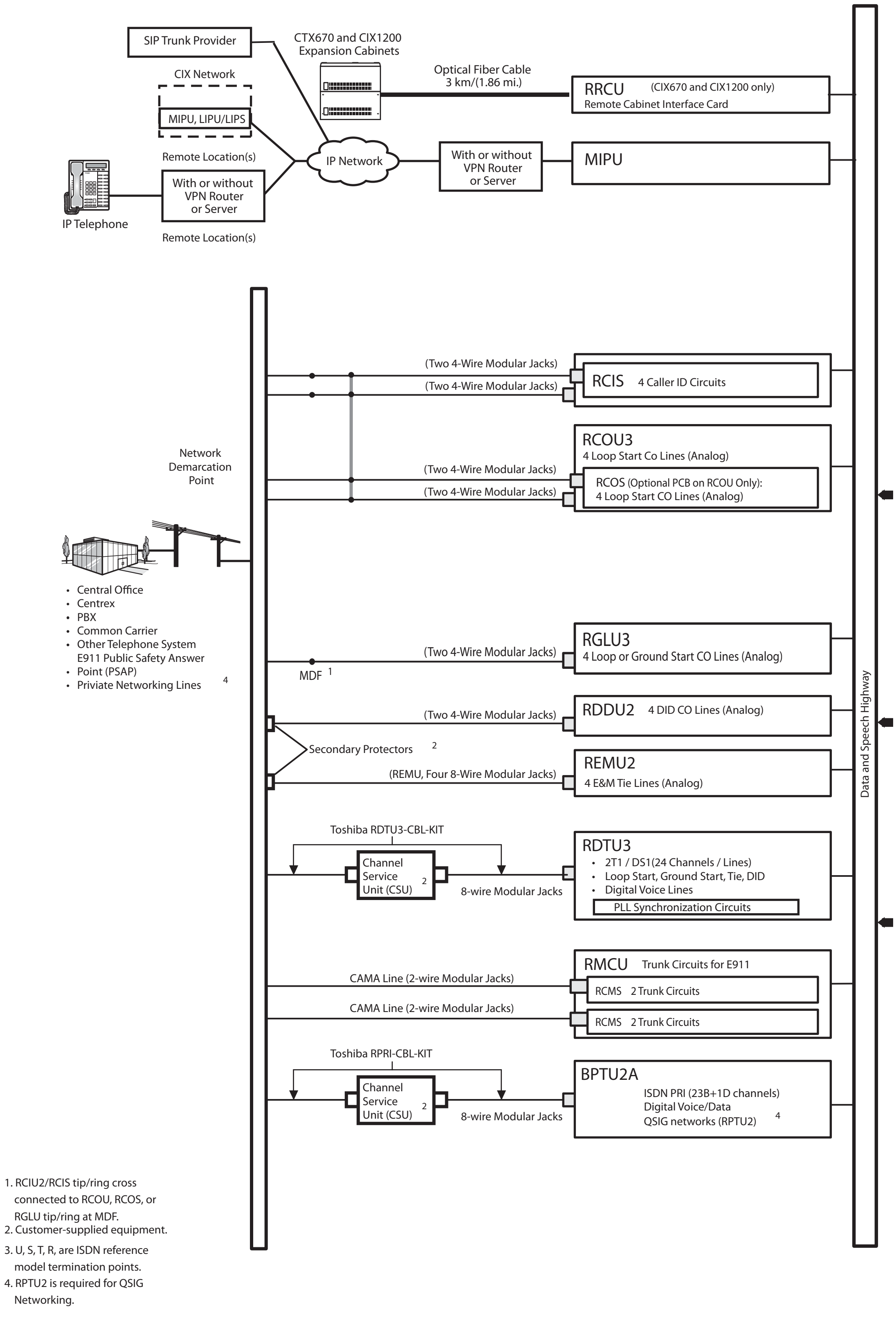
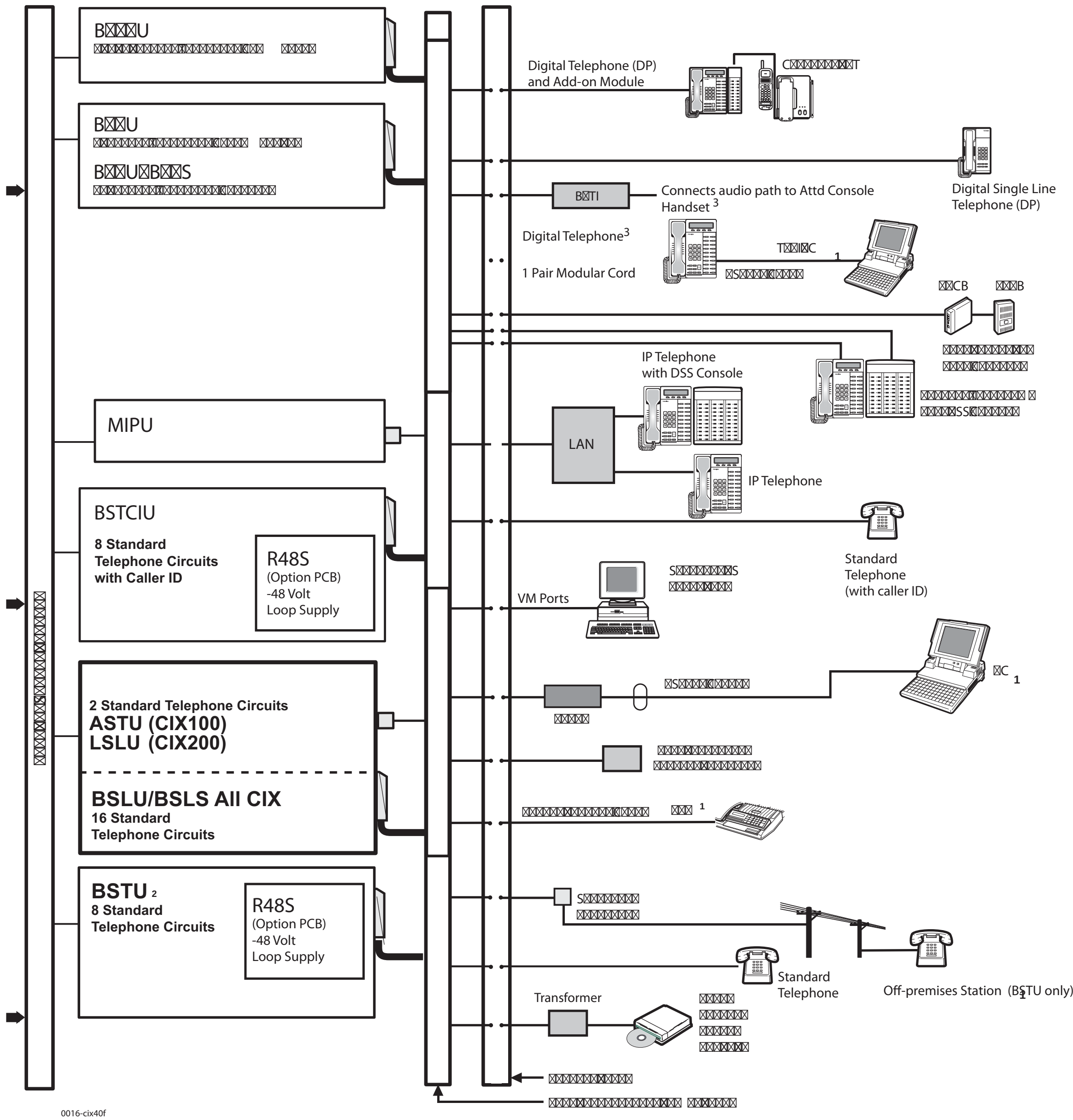


Figure 14 CIX100, CIX200, CIX670, and CIX1200 CO Line Side Functional Block Diagram



1. Customer-supplied equipment.
2. RSTU2 or above is required for standard telephone message waiting lamp.
3. PDKU and RDSU should only be used for 2000-series digital telephones. They do not support all of the 3000-series and 5000-series digital telephone features, including LCD. The PDKU also does not support BPCI, BATI and the CTX Attendant Console.

Figure 15 Strata CIX100, CIX200, CIX670, and CIX1200 Station Side Functional Block Diagram

Telephones and Peripherals 7

This chapter covers Toshiba's 5000-series Digital Telephones (DP), Internet Protocol Telephones (IP) and peripherals that are compatible with Strata CIX telephone systems.

DP5000-series Digital Telephones

The DP5000-series telephones replace the DKT3200-series telephones in the Toshiba product line.

All three generations of telephones, DKT2000, DKT3200 and DP5000, can co-exist with full functionality in one Strata CIX system with Release 5.1 software or above.

Although the DP5000-series telephones can be used on a CTX28 they will have limited functionality.

The DP5000 telephones have three operational modes:

- When connected to Strata CIX systems with Release 5.1 or above, they will automatically run in DP5000 Mode.
- When connected to Strata CIX systems with older than Release 5.1 (including Release 3) the DP5000-series telephones will automatically act as DKT3200 telephones; DP5000-series telephones can also be programmed to act as DKT2000 telephones.
- When connected to a PDKU card DP5000-series telephones automatically act as DKT2000 telephones.
- The DP5000 Backlight feature will function on all Strata CIX systems, regardless of the system software version.

DP5000-series Telephone Illustrations

20 Programmable Feature Buttons 4-Line LCD Telephone

Single Line Telephone 1 Programmable Button

Legend

A. Status LED (message and ringing)

B. LCD Display

C. Softkeys

D. Programmable Feature Buttons

E. Message Waiting LED Button

F. Microphone LED Button

G. Speaker LED Button

H. Volume

I. Hold Button

J. Microphone

K. Tilt stand

L. Off-Hook Button (Single Line Telephone)

DP5008

Digital Single Line Telephone

DP5018-S

Digital Speakerphone with 10 programmable buttons

DP5022-SDM (CIX40 only)
DP5022-SD
DP5122-SD

Digital Speakerphone with 10 programmable buttons 4-line LCD display

DP5032-SD
DP5132-SD

Digital Speakerphone with 20 programmable buttons, 4-line LCD display with backlight option (shown with KM5020)

DP5130-SDL and DP5130-FSDL

Digital half duplex and full duplex speakerphone options with 20 programmable buttons, 9-line LCD display with backlight.

The DP5000 Look

- Sleek low profile — less than 1-inch thick.
- Black body with dark gray keys.
- Metallic silver curved tilt stand with eight adjustable positions.

Large Display Area for Call Information

- New LCD Technology — High Contrast LCD panel.
- All LCDs are 24 characters wide.
- Four lines of LCD to display call information (on the DP5022-SDM, DP5022-SD, DP5122-SD, DP5032-SD, and DP5132-SD telephones).
- Nine lines of LCD on the DP5130-SDL and DP5130-FSDL telephones.
- Advisory Messages.
- Automatic Number Identification (ANI).
- Caller ID, Name and Number with call history.
- Contrast adjustment (13 levels).
- 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 seconds).

LCD Key-Strip

- Integrated in the DP5130-SDL and DP5130-FSDL telephones.
- Integrated in the LM5110 add-on module. This module is supported on all of the DP5000 series telephones except the DP5008.
- Key labels are programmable via Network eManager, MyPhone Manager (both local and remote), or user programming mode.
- User programmable line and trunk labels.
- Feature labels – user selectable.
- Ease of deployment and dynamic changes.

Backlight LCD

- Available on: DP5122-SD, DP5132-SD, DP5130-SDL, DP5130-FSDL, and LM5110.
- Place the telephones anywhere: regular office, reception area, high-end home studio, hotel lounge, low-light environments, etc.
- LCD backlight can be set to always on, always off or synchronized which turns backlight on anytime activity is sensed on the telephone and will automatically shut off after a period of time to conserve energy.

Mic Mute

Note Not supported on DP5008.

- All models have microphone and half duplex speaker capability.
- Enhanced Mute – the handset and microphone are muted simultaneously.
- The MIC key on the telephones toggles between Mic and Mute. When Mic key is lit, hands free communication is supported, when the Mic key is pressed and light is off, Mute is enabled.

Handset/Headset

- Built-In Headset Interface.
- Built-in Carbon Handset Interface controlled by software.
- Speaker Off-hook Call Announce (OCA) for all models except on the DP5008. Speaker OCA requires the DOCA interface.

Note The CIX40 supports Handset Off-hook Call Announce (OCA) but not Speaker OCA.

- On hook dialing.

Shift and Hist Keys

The bottom left and right keys on the large screen telephones are called the Shift and Hist keys. The Shift key toggles the LCD screen between flexible keys 1~10 and flexible keys 11~20. The Hist key accesses the Caller ID history. These buttons have no function when the telephone is connected to a Strata CIX system running on software released prior to R5.2 MT021.

Speaker Off-hook Call Announce (DOCA-1A)

Digital telephones equipped with DOCA-1A can receive Speaker OCA which enables stations to receive internal calls over their speaker while on another call using the handset. The DOCA-1A is not required in a telephone to originate OCA calls or in a digital telephone that receives OCA calls through the handset or headset.

Speaker OCA Interface (DOCA-1A) cannot be installed in DP5008 telephones.

DP5000-Series Telephone Tilt Angles

The DP5000-series telephones, except the DP5008, have three tilt positions built into the base. There is also a tilt stand extension that adds an additional 20 degrees of tilt. When the telephone is sitting on a desk or table there are a total of six different angles of tilt available. When wall mounted there are two angles available.

The DP5008 base is fixed at 15 degrees. All other DP5000-series telephones and add-on module can tilt at 15^0, 27.5^0, 35^0, 40^0, 47.5^0, and 60^0.

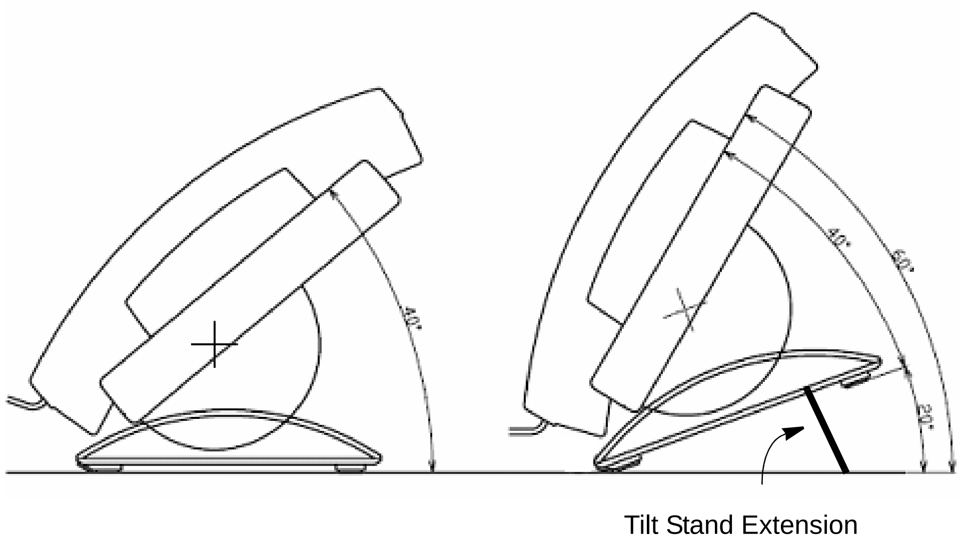


Figure 16 **DP5000-Series Desk-top Tilt Angle With Tilt-Stand Extension**

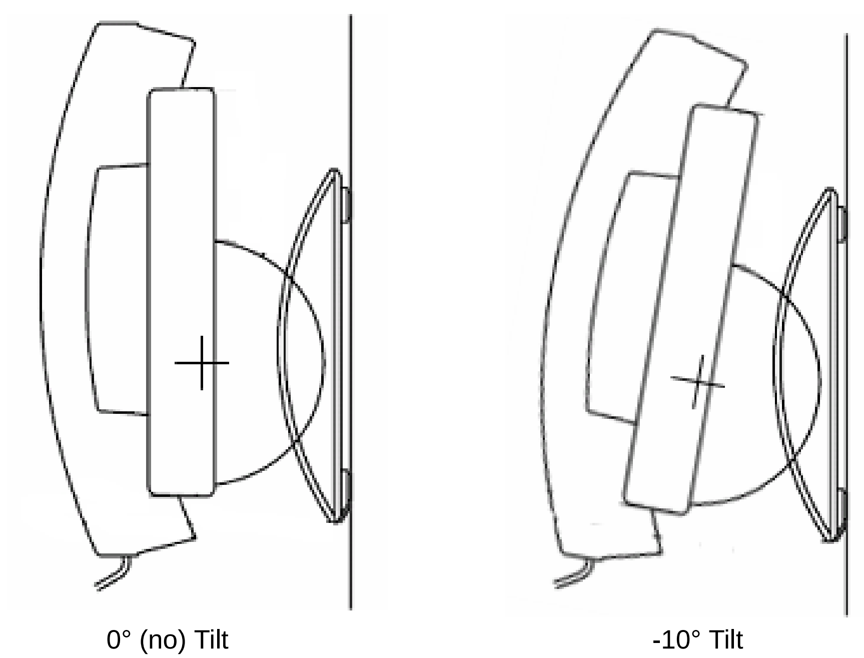


Figure 17 **DP5000-Series Wall Mount Angles**

Telephone Button Expansion Options

Add-on modules connect directly to the telephones and do not require an additional interface circuit (port). Up to two Add-on-Modules can be attached to a telephone to supplement the telephone's 10 or 20 buttons.

The CIX supports a limited number of Add-on-Modules per cabinet (see Table 53 on page 185 for the capacities of different common control units).

Expansion options for the Toshiba DP5000-series telephones are described below:

LCD Add-on -Module (LM5110)

The LM5110 adds 10 programmable LCD 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.

The LM5110 supports backlight and LCD labels, it can be connected to any 5000-series telephone (except DP5008).

LM5110 shown with DP5130 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.

KM5020 shown with DP5132-SD Telephone

Digital DSS Add on Module (DDM5060)

The DDM5060 console (shown right) is for system attendants.

The DDM5060 operates alongside a digital 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. The DDM5060 is not compatible with the DP5008.

Up to eight consoles can operate with one digital telephone depending on CIX system (See Table 53 on page 185).

DDM5060 shown with DP5132 Telephone

The DDM5060 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 DDM5060 console connects to a digital station port on the ADKU, BDKU, or BDKS card.

Cordless Telephones

DKT2404-DECT Digital Telephone

Toshiba offers the DKT2404-DECT cordless digital telephone model. This compact cordless digital telephones bring mobility and productivity to office telephones.

DKT2404-DECT shown with charger and Base

DKT2404-UDR100

The DKT2404-DECT delivers the DECT functionality for the Strata series of cordless digital telephones. DECT technology minimizes any interference between base stations in proximity or any surrounding WLAN access points.

This telephone provides a 24-character display for better compatibility with the Toshiba DP5000 Series desk phones. It provides four programmable keys with an additional four speed dial keys.

The digital cordless telephones operate from the same digital station port as the DP5000-series digital telephone. They can be attached to a Toshiba DKT3000- or 2000-series corded digital telephone or used as a stand-alone. If a cordless telephone is attached to a DP5000, the DP5000 must be put into 2000 mode.

Up to six repeaters (DKT2404-UDR100) can be linked per base station, with up to three capable of being daisy-chained in one direction, to extend operation range to greater than 1000 feet in open areas.

Some of the features this cordless model include:

- Two-line LCD with 24-character display, plus one line for icons.
- Dedicated Talk, Redial, Hold, Msg, Cnf/Trn, Menu/MUTE, Speaker keys.
- Four programmable keys, plus four speed dial keys.
- Handset measurements in inches: 2.25 wide x 1.0 deep x 6.5 tall. For base and charger measurements, see Table 41 on page 175.
- Transmission: 1.9 GHz DECT 6.0
- Number of Channels: 5 Channels
- Talk Time / battery life:16 Hours or 7 days
- Stand By Time: 168 Hours
- Battery Type: NiMH Battery
- Ringer and handset volume control
- Single button access to: Conference, Hold, Redial, Message and Transfer features
- Four programmable function buttons
- Charging stand
- AutoStandby
- AutoTalk
- Vibrate ringer alert

- Out-of-range protection
- Low-battery protection system
- Headset jack (2.5mm)
- Stand-alone or DKT operation
- High quality ultra-secure conversation with 32Kbps Adaptive Differential Pulse Code Modulation (ADPCM) voice code combination.
- Three ring tones

Notes

- They cannot receive Group Pages or All Call Pages.
- The handset and base unit of each cordless telephone is equipped with the same security code. In order for a handset to operate, it must be installed with the matching base unit.

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.

IP 5000-Series IP Telephones

The IP5000-series telephones replace the IPT2000-series telephones in the Toshiba product line.

IPT2000- and IP5000-series telephones can coexist with full functionality in one Strata CIX system, Release 5.2 software, or above.

The IP5000-series telephones have three operational modes:

- When connected to Release 5.2 CIX systems with MIPU interface cards, they will automatically run in IP5000-series mode.
- When connected to CIX systems with older than Release 5.1 (including Release 3), the IP5000-series telephones will automatically act as IPT2000-series telephones.

Strata CIX40, CIX100, CIX200, CIX670, and CIX1200 systems running R4 and later software can upgraded to Release 5.2 software. Systems running R3 must be updated to R4 processors then updated to R5.2.

Strata CTX28 systems do not support IP telephones.

Toshiba offers five 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, BESCB 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, BESCB and carbon handset/headset.
- **IP5122-SDC** – Looks and functions similar to the IP5122-SD telephone when connected to the Strata CIX. This telephones fully support all the Strata CIX 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 Strata CIX 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 & 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 the Strata CIX 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 IP5631-SDL does not support Gigabit Ethernet and carbon handset/headset.

Note ACD 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 & 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, 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 handset.
- Speaker OCA when using MIPU, GIPU or GIPH.

Note IP5000-series telephones enabled with Speaker OCA require two IP channels from the same IP interface card but only one end point license. The IP channel used for Speaker OCA is only used during the OCA call and may be shared by all IP5000-series telephones.

- 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 when using an MIPU interface card in a CIX system.
- 802.3af power over Ethernet compliant
- Integrated Application Processor for Voice Mail, Unified Messaging ACD, and other applications
- Soft Keys to respond to the Strata CIX 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.

IP Protocol

The Strata CIX uses RFC3015 Media Gateway Control (MEGACO) enabling Strata systems to provide all the feature functionality of digital-series telephones to IP telephone users.

Connectivity

IP telephones connect to either the MIPU, GIPU or GIPH in the Strata CIX system running software version 5.2 or above. However, the IPT2008-SDL can only be connected to the MIPU, GIPU or GIPH. These telephones must be connected to MIPU, GIPU or GIPH to use peer-to-peer Real Time Protocol (RTP) connection.

Note The Strata CIX also supports the IPT1020-SD. This telephone can be connected to a BIPU-M card only. It does not support peer-to-peer RTP connection.

- 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 Strata 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 on the PoE (Power over Ethernet).
- The RJ45 LAN jack connects the telephone to the network via the LAN cable supplied with the telephone. These telephones operate on the network at 10/100Mbps and 1000Mbps (IP5000-series with the exception of the IP5522-SD, 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 IPTs to the user's PC. The IPTs 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. The same carbon or ECM headset support used on Toshiba digital telephones can also be used on IP telephones.

Capabilities

The Toshiba IP Telephones also have the following capabilities:

- The IPTs 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 is selectable for each IP telephone in IP station administration using Network eManager.
- The external ringer interface connector is mounted inside the telephone base. This enables connection of an BESCB 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 53 on page 185).
- Terminal Authentication is an option that allows a particular IP telephone to keep a reserved directory number on a CIX 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.
- IPT firmware can be updated locally or remotely using Network eManager. 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, 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.)

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 5000-series telephone (except DP5008).

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 a 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 depending on CIX system. 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.

CIX Attendant Console

The Strata CIX Attendant Console runs on a PC with Microsoft® Windows® 7 Professional, Windows 2000 Professional or Windows XP Professional operating system. The Strata CIX Attendant Console PC 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 Strata CIX Attendant Console software, and you have a winning solution for any Strata CIX installation!

The Console connects to the Strata CIX processor via the LAN as a Computer Supported Telephony Application (CSTA). The Strata CIX system requires the processor NIC interface.

Each CIX Attendant console requires the LIC-ATT license which is bundled with the Toshiba supplied Attendant Console PCs and software. The LIC-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 CIX Attendant Console are:

- Operating Software – Microsoft Windows XP Pro or Windows 2000 Pro
- Processor – 2.0GHz Intel Pentium 4, Celeron or higher
- Memory – 512M 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

There are two types of CIX Attendant Consoles; each operates and functions the same but each provides a different type of talk-path.

The following Attendant Console hardware is available from Toshiba:

- CIX-DATTCONS – provides a digit talk path like a digital telephone and requires a BDKU interface port. Includes software and LIC-ATT license.
- CIX-IPATTCONS – provides an IP talk path like IP telephones and requires an MIPU interface port. Includes software and LIC-ATT license.

Each of the above CIX Attendant Consoles consists of the following *standard* items:

- Attendant Console License (LIC-ATT) for the CIX processor (bundled with Toshiba Console part number)
- LIC-ATT must be ordered separately if using a Dealer supplied PC for the Attendant Console
- Intel 2.8 GHz CPU
- 512M Random Access Memory (RAM)
- CD R/W drive
- Windows 2000 Professional (factory installed)
- CIX Attendant Console software (factory installed)
- Comprehensive set of multimedia inputs and outputs
- Keyboard & Mouse
- PC Headset (CIX-IPATTCONS only)
- Attendant Keyboard stickers (Black) CIX-BL-ATCON-VA

The *optional* items are:

- Printed documentation is available on Toshiba FYI, as well as Documentation downloads.
- 17" Flat screen Monitor – Toshiba offers a 17" Liquid Crystal Display (LCD) flat screen monitor; part number CTX-LCD-MONITOR.
- Attendant Console Interface Unit (BATI) and Handset/Cradle (BATHC) connects to a digital telephone interface port on the Strata CIX (CIX-DATTCONS only). An optional headset can be used in conjunction with the handset.

  ...or

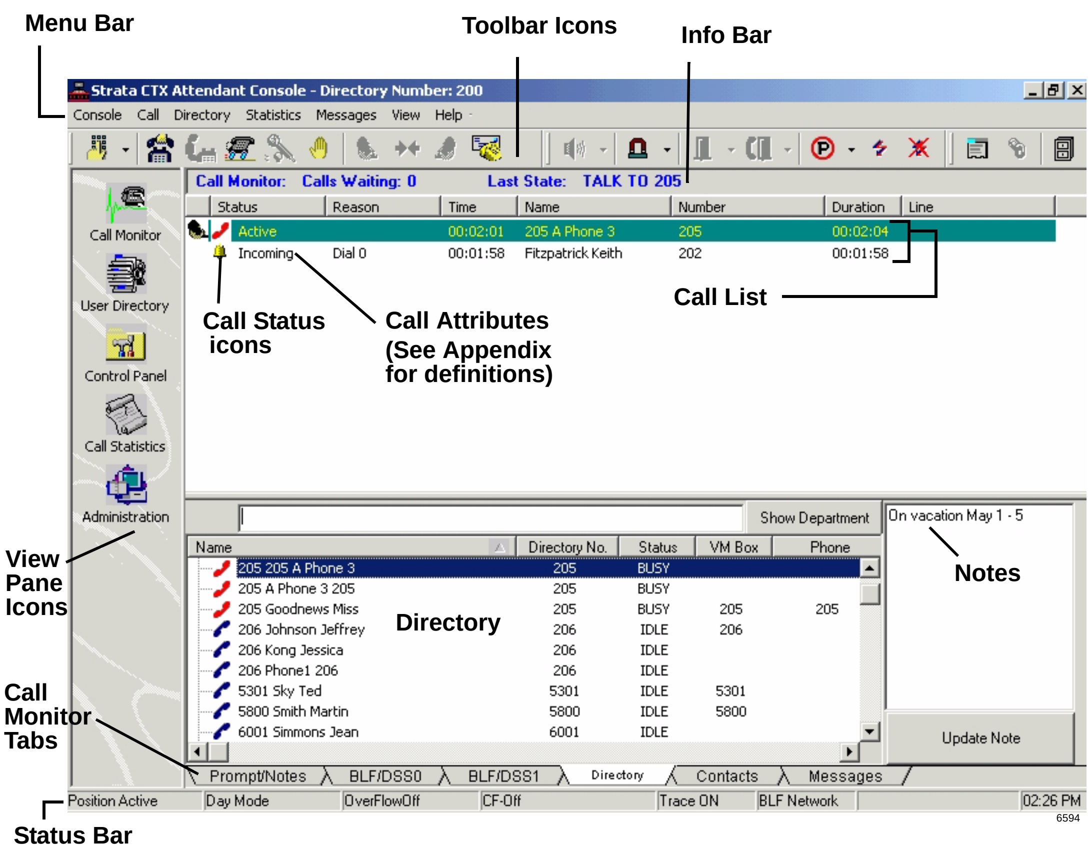
  a DKT3001 or DP5008 can be used as the console handset instead of the BATI/BATHC handset/cradle. If an optional headset is used, it connects to the optional BHEU installed in the DKT3001 or directly into the DP5008 built-in headset jack (CIX-DATTCONS only).

Important!

- *If a digital telephone is used in place of the BATI/Handset, it can be used as the Attendant Console Handset only. It cannot be used as a telephone to make or receive calls independent of the console. This includes when the console is in service or out of service.*

The Strata CIX1200 supports up to six, the CIX670 system supports up to four, and the CIX200, CIX100 and CIX40 supports up to two Attendant Consoles. Multiple consoles automatically share the incoming call load on a call-by-call rotation basis. Features such as Overflow and Position Busy add to the efficiency of single or multiple console applications.

The CIX 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 CIX 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 Strata CIX supports several types of stations and customer-supplied peripheral devices, such as door phones for visitor screening, a music source interface for MOH and BGM, a speaker for amplified ringer, Toshiba Voice Processing systems for voice mail/auto attendant applications, and more.

Toshiba Telecommunication Systems Division (TSD) does not provide ISDN or IP station equipment, such as ISDN IP telephones, fax machines, and computer interface devices for high speed Internet access or video conferencing. Toshiba does provide the interface circuit boards that support all of the above ISDN station equipment.

Door Phone (MDFB)

Door phones can be assigned to ring telephones when the button on the door phone is pressed. The Door Phone location displays on the called telephone's LCD. When the telephone answers, a two-way talk path exists between the telephone and door phone.

1873

Door phones can also be used as sound monitors. Station users can call the door phone (it will not ring) and listen to sounds from the surrounding area. Door phones also can operate as a "hot line." For example, a door phone can be used for calls between an office and a warehouse. Door Phones are often used with a door lock to screen building visitors. The door lock can be opened for a predetermined amount of time by pressing a button on a telephone.

Door Phone/Lock Control Unit (DDCB)

The DDCB can support as many as three door phones (MDFBs) or two door phones (MDFBs) and one door lock control relay. Using the door lock control, digital station users can unlock a customer-supplied electronic door lock at the touch of a button programmed on their digital telephone or by dialing a feature access code from any type of telephone. Each DDCB requires one digital station circuit.

1874

Each door lock can be programmed to remain open between three~30 seconds. The Door Lock button LED remains On while the lock is open. LCD telephones display "DOOR UNLOCKED" until the telephone releases or times out.

External Speaker (BESCB)

The BESCB is a multi-functional, external, speaker control box with a built-in three-watt amplifier. It can be used to control and drive a paging speaker, a paging amplifier, or a telephone's Loud Ringer.

Wireless Access Points

Toshiba has a strategic relationship with Adtran Inc., a leading global provider of networking and communications equipment.

The NetVanta 1550 series are Layer 3 Lite, Gigabit Ethernet Switches with Power over Ethernet (shown right). The NetVanta 1550-24P and NetVanta 1550-48P rackmount switches support 24 and 48 10/100/1000Base-T ports and four SFP+ uplink ports. Each SFP+ port supports up to 10 Gbps uplinks delivering up to 80 Gbps of bi-directional throughput. The NetVanta 1550 series switches have four Enhanced 1/10 Gbps SFP+ Ports.

The NetVanta 1550 series switches have aggressive SMB type pricing with Enterprise class features and functionality.

The key features/benefits of this product are:

- Faster Implementation
  - The VoIP setup wizard automates and simplifies the entire process of configuring NetVanta switches for your VoIP system and associated IP-phones
  - A built-in DHCP server eliminates the need for an external server and enables auto-provisioning of IP-phones, using DHCP "option 66"
  - Full LLDP/LLDP-MED support for zero-touch phone setup
  - Industry standard CLI–utilize a single script to bulk configure multiple switches for enterprise-wide VoIP deployments
- Faster Troubleshooting
  - On-demand VoIP report simplifies troubleshooting by providing a snapshot of the switch; Port status, Active endpoints, Power over Ethernet (PoE) status, and more.
  - Network forensics identifies connected endpoints and provides a single database for all device parameters.
  - Cable Diagnostics tool dramatically reduces time spent to trouble shoot cabling issues.
- Greater Security
  - Port scheduler disables ports and powers down devices during non-business hours, improving security and conserving power.
  - Denial of Service (DoS) protection enhances security by identifying and blocking common network DoS attacks.
  - AOS Security Audit tool assists in the Payment Card Industry - Data Security Standards (PCI-DSS) compliance testing and identifying potential vulnerabilities.
- Increased Uplink Bandwidth
  - Up to 80 Gbps for inter-switch connectivity, utilizing four 10G uplink ports
- PoE flexibility and Power Redundancy
  - PoE power redundancy up to 370 W (select NetVanta switches) utilizing an external

  - NV1131 RPS/EPS unit, minimizing disruptions due to power outages.
  - Enhanced PoE budget up to 370 W (select NetVanta switches) doubling the available PoE budget up to 740 W, utilizing an external NV1131 RPS/EPS unit.
  - 802.3af, 802.3at (PoE+) and Legacy PoE
  - Built in surge protection
- Industry Leading Warranties and Support
  - Comprehensive limited lifetime warranty on all NetVanta switches.
  - Included software updates and upgrades for the duration of the warranty, ensuring continued value for your switch investment.
  - Included Next Business Day advance hardware replacement and 24x7 phone support for the duration of the warranty, ensuring minimal disruptions to your business.

Cabling and Connectors

The Strata CIX uses industry standard cabling and connectors to interface with lines, stations, and peripherals. Stations use standard twisted-pair cabling to connect to the system via the MDF. Digital and standard telephones require just one pair-cabling. Two pairs may be required to achieve full distance when optional DP subassemblies are used.

Digital telephones connected to BDKS require an external power supply to reach maximum distance from KSU when the telephone has a DADM or DOCA-1A.

Station circuit cards connect to stations and peripherals with a 25-pair Amphenol connector via the MDF. Analog CO, DID, and Tie line circuits interface with the public telephone network via modular connectors. T1 and ISDN use industry-standard Amphenol and modular connectors (for details, see Table 37 on page 172).

Peripheral devices such as Network eManager maintenance PCs, etc., connect to a hub or LAN, which connects to the processor’s Ethernet LAN interface via an RJ45 connector and Category 5 wiring. Call Accounting and Voice Mail SMDI require RS-232 modular adapters and cords to connect to the processor BSIS interface.

IPedge Application Server 8

The IPedge Application Server for Strata CIX integrates three important applications into a single server saving you money rather than buying the applications separately and provides a pathway to full IP Telephony deployment in the future.

IPedge Messaging gives you voicemail, unified messaging, and an Android/iPhone client that lets you manage your messages and make and receive calls from your smartphone; enabling your effectiveness when you’re not at your desk.

IPedge Call Manager integrates your PC with call control on the Strata CIX so you can make calls from your contacts, text chat with your colleagues if you see they are on the phone, and take your calls at your home-office; enabling your effectiveness when you using your PC.

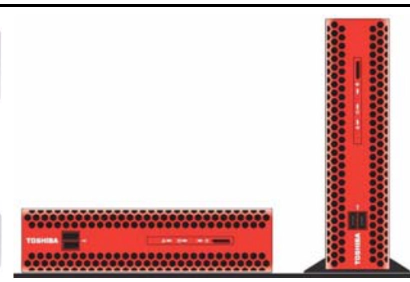
IPedge Meeting makes conference calls more like meetings by allowing you to share a PPT and even your video when you are talking remotely; making your conference calls more effective.

All three of these applications are integrated on the IPedge Application Server for Strata CIX which is available in three sizes so that you can choose the most cost effective model for your site.

Whether you start with only one application or all three, the IPedge Application Server for Strata CIX enables you and your business telephone system to be more effective.

IPedge EC Application Server

IPedge EM Application Server



IP*edge* EP Application Server (Horizontal and Vertical view)

Figure 1 IP*edge* EC, EM and EP Application Servers

Table 14 Basic Specifications

IPedge EC App Server	IPedge EM App Server	IPedge EP App 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)	4 x 300GB HDDs (RAID 1 standard) 4 x 300GB HDDs (RAID5 optional)	250GB HDD
Up to 200 Users	Up to 1,000 Users	8 to 40 Users

IPedge App Server Solution

On a single server, IPedge App Server provides the following:

- 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 (UCedge and Call Manager).
  - UCedge Client is a productivity tool that is integrated with the IPedge App Server. For more details, refer to “UCedge Client” on page 81.

  There are two levels of IPedge Call Manager:

  - 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.
  - Call Manager Standard – Existing users of the powerful Strata Call Manager (SCM) will continue to enjoy all the features of the SCM in the new IPedge Call Manager. The smart installation procedure will install CM or SCM based on which system the client will connect to.
- Meet-Me Audio Conference – Meet-Me Audio Conference application is built-in to IPedge Application Server. This conference application provides a simple, easy-to-use Meet-Me Audio Conference feature.
- Meet-me Conference and Web Collaboration – Having 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

Messaging

The following is a list of Messaging features. The Messaging features are classified into Basic and Advanced user features. Basic features do not require different licenses. However, Unified Messaging requires an Advanced license. Messaging within IP*edge* App Server for Strata CIX 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 IP*edge* 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 IP*edge* 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 IP*edge* 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 IP*edge* 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 71).

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. IP*edge* 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 IP*edge* 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 IP*edge* 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

IP*edge* 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 IP*edge* 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 IP*edge* 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

IP*edge* 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

IP*edge* 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 IP*edge* 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 IP*edge* system to access a Microsoft Exchange Server, in order to manage IP*edge* 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 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)

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.

Web Controller

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 Web Controller can be used on a secure or non-secure http port.

Reporting

Messaging records all activity from calls coming in or out of IP*edge* 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 IP*edge* Messaging. The report includes mailbox number, date, time, result (answered/ unanswered), call duration and number dialed.

Port Statistics

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

Scripts

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

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

Scripts requires a license for each application desired.

Script Logging Reports

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

System Group List

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

System Hourly Statistics

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

System Statistics

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

Unattended Mailboxes

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

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 DNs 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

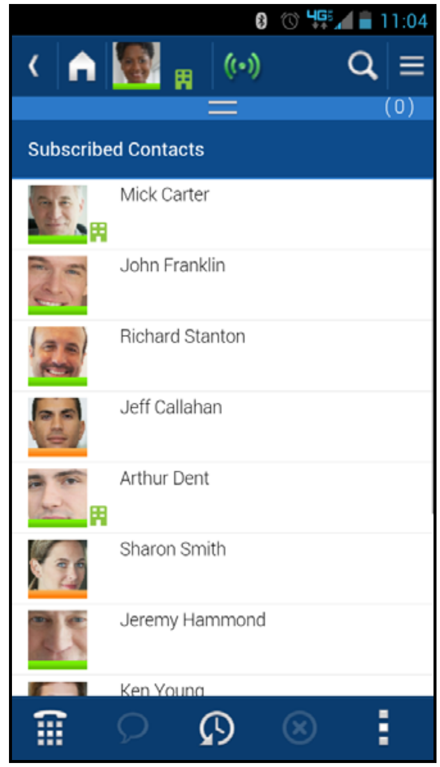
When a certain number of password entry attempts per mailbox is detected Messaging locks the mailbox to prevent further use and notifies the system administrator via email.

Secure Authentication for Outgoing Email

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

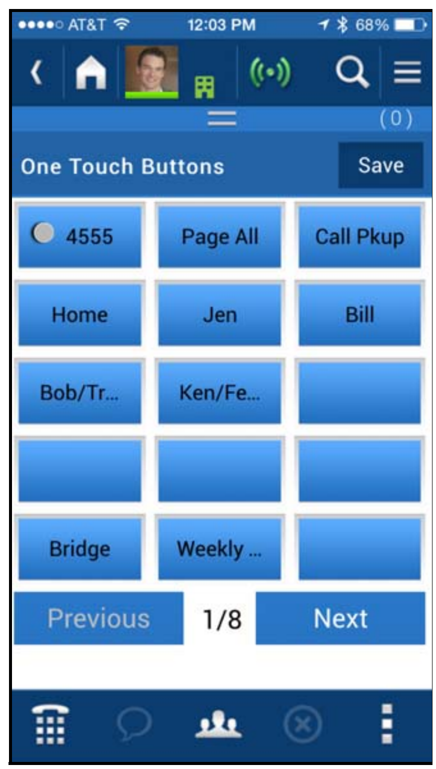
UCedge Client

The UCedge Client is a Unified Communication solution for the IPedge Application Server. It works on the iPhone, Android smartphones, PC's and Mac computers.



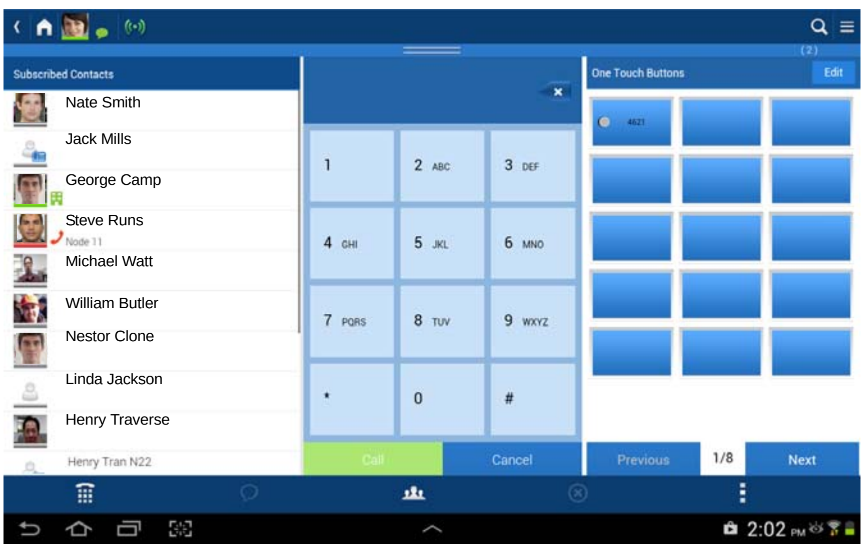
The following are the benefits of having the UCedge Client:

- **Pairing** – Initiate dialing on your desk phone from your tablet or smartphone.
- **Softphone** – Make and receive calls on your Android, iOS, Windows, and Mac devices.
- **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. Make International Calls from your cell phone at land line rates and save money.
- **Follow Me** – Receive calls when you are away from your desk at the same extension and get things done sooner.
- **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.
- **History** – Instant Messaging and calling history is stored in the device so that it may be reviewed later. Consolidated call and IM history is available on multiple mobile devices when sync to cloud is set. An action menu is available in history so that the user can start any communication from any history entry.
- **Buttons** – The user can have customizable buttons with one touch dial (speed dial) by configuring any number. It can be used for frequently dialed numbers, conference PINs, or any other dialable numbers (shown right).

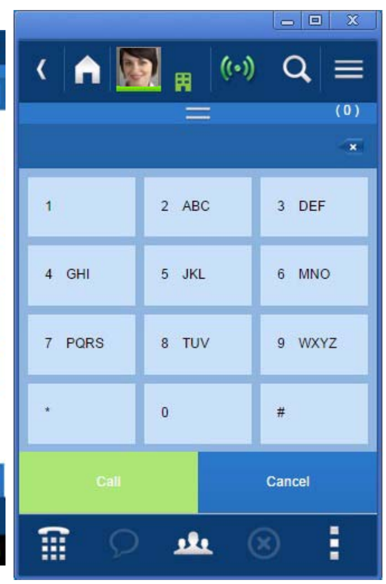


- **Initial Setup Wizard and New User Tutorial** – When the user logs into the device for the first time, the application starts a wizard so that the user can configure the device properly and start using it immediately. In addition, as the user accesses new features, it pops up the first time instructions so that the user can learn how to use the app.

- **Voice Mail / Fax Mail Support** – UCedge supports all mailbox features including Inbox, Saved message, and Deleted messages. The client can also receive the fax mail messages. Users can now reply to the voice mail, forward the voice mail to another mailbox or the email.
- **Greetings Management** – Greetings can be changed, recorded and managed using UCedge controls, as well as the routing and activation control for the Follow Me feature from the client.
- **Native Dialer Intercept** – This feature is available for Android devices only. By using this Native Dialer Intercept, the user can send the business phone number as the caller identity instead of the cellphone number. When using the dial pad on your smartphone, the UCedge application intercepts the operation and gives you the option to either dial using your business caller ID or your cell phone number. Once you choose UCedge, you have the option to use Callback/Callthru. iOS does not support this feature.
- **Call Alert / In Call Menu** – The user can get notification of a missed call which is forwarded to the mailbox so that the user can route the call to the cell phone or other destination. After the call is answered on the cell phone, the user can transfer the call to an extension.
- **Compact View Support** – This feature is available for PC's® and Mac® computers only. The UCedge client now supports the compact view (shown below) which is similar to the portrait view of the smartphone so that it does not occupy desktop space. The user can simply resize the window, and client will automatically switch to compact view.



Full View



Compact View

Call Accounting

The Call Accounting feature uses the Strata CIX SMDR output to record the call detail information, generate reports, search for specific call information, and send notifications based on the call information to the IPedge Application Server. The call information, the SMDR data is stored in a database in the IPedge Application Server.

Call accounting is accessed through Enterprise Manager. With the call accounting feature, an administrator can generate reports and setup automatic reports. A user also has the ability to search call accounting records and generate reports with the following parameters: Call Type, Station Number, Specific trunk or Line Group, Caller Id or Called Number, Account code, Call cost, Time, Call Duration, DISA Number, DNIS Number, and Network Node ID.

Rate tables can be used to calculate the cost of calls. Rate tables for specific area codes, international calls and, toll free calls can be created with Enterprise Manager. It is also possible to import rate tables from a CSV file.

Call Accounting can generate notifications for Emergency calls (ex. 911), inbound calls from specified callers, and/or outbound calls with the specific prefix code (for example: 01144 for international calls to the UK). An emergency call can be notified as soon as it is dialed, and other notifications are sent when the call is over to the specified email address.

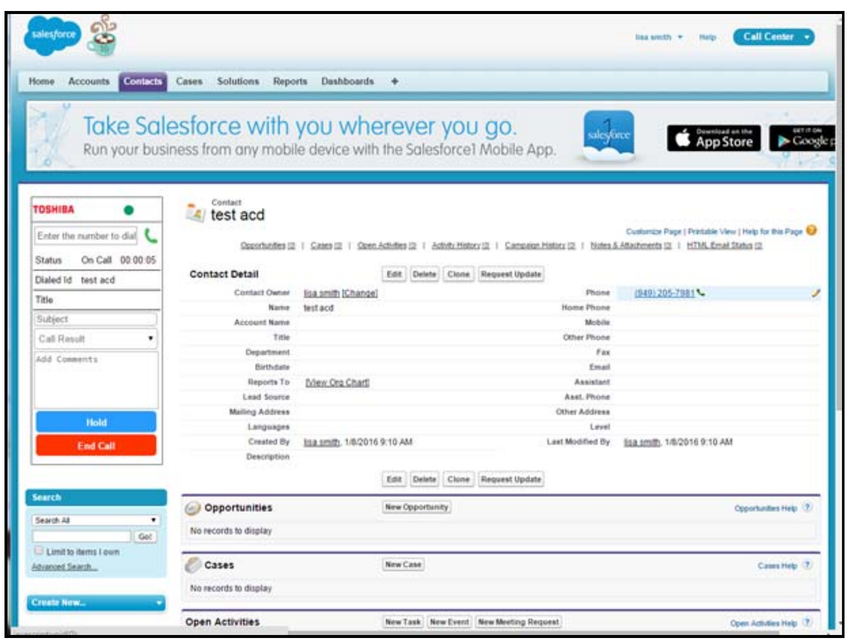
Salesforce.com Integration

The IPedge Application Server for Strata CIX provides Salesforce.com integration as a plug-in downloaded, at no charge, from the AppExchange™. Salesforce application users can make and receive calls and perform other call control functions using the Salesforce application. It provides users with click-to-call capability from the Salesforce Contact. It integrates call control and contact history. The user can add the call results and notes when necessary, and it is also recorded in the history. The Toshiba plug-in can work with IP telephone or softphone option on Call Manager or UCedge®.

In Contact Centers, the plug-in can be configured to automatically pop up the customer contact based on the phone number so that the agent can process the customer's request quickly and accurately.

If a matching contact is not found, Salesforce can be configured to open the new contact to create a new customer record.

Each Salesforce plug-in user requires the Salesforce UC license (I-UC-SFORCE).



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, Windows 8, Windows 10, the Terminal server on Windows Server 2003 ~ Windows Server 2012 R2, or Windows 7 operating systems.

There are two levels of IP*edge* Call Manager:

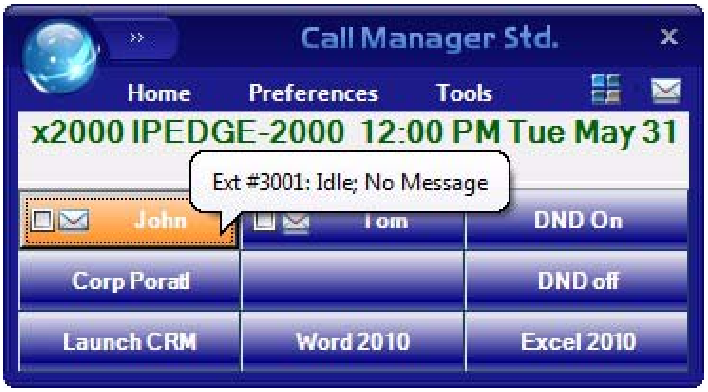
- Call Manager Standard version is free to all users of the IP*edge* system. The license (I-CM-STD1) for Call Manager Standard is included in the user license bundle at no additional charge.
- Call Manager Advanced version provides enhanced functionality, including full Unified Communications (UC). Purchase 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 IP*edge* system with the appropriate license (I-CM-1/I-CM-V1) on IP*edge*. 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.

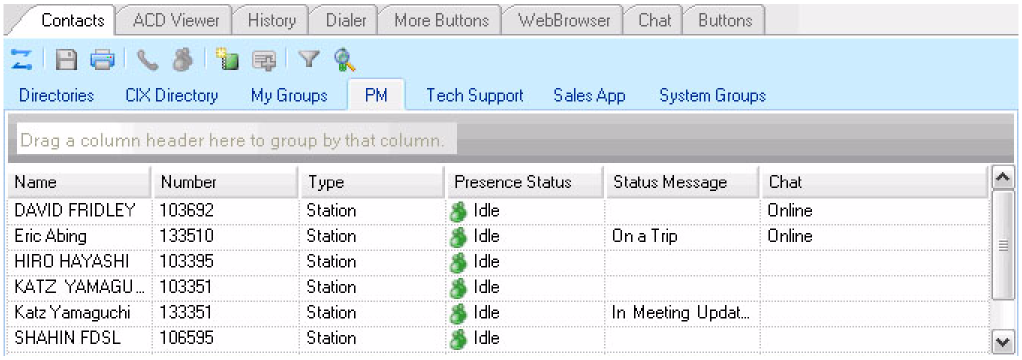
Figure 2 Call Manager Main Page

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.

Companion Applications

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



Contacts (Directory, Presence Viewer and Speed Dial)

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

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

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

History

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

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

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.

Chat

The Call Manager Chat application enables Call Manager users connected to the IP*edge* 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.

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.

Meet-Me Audio Conference Application

IPedge Application Server for Strata CIX has the Meet-Me Audio Conference application built in to server. This conference application provides a simple, easy-to-use Meet-Me Audio Conference feature. The administrator has to only apply the license to activate the feature. Four resources are included, additional licenses can be purchased.

On-Demand Meet-Me Audio Conference

The Meet-Me Audio Conference is a user controlled, PIN based, meet-me conference feature. The conference is setup by dialing the Meet-Me Audio Conference extension number. Each station with an IPedge messaging mailbox can be a conference owner.

- PIN Management – To establish a conference call, the owner sends the participant PIN to all participants. Any number of participants, up to the limit set by the administrator, can connect as long as a license is available. The conference owner can change the owner and participant PIN at any time through Enterprise Manager Personal Administration (EMPA).
- Waiting Room – As the participants join the conference they are put in the waiting mode and hear music on hold until the owner joins. When the owner hangs up, it can be configured to put all participants back into the waiting mode or disconnect all participants.
- Participants Control – The Administrator can set the limit of participants so that audio resources are not all used by one conference.
- COS Control – The administrator can control which users have access to the Meet-Me Audio Conference by assigning the appropriate Messaging Class of Service (COS) to users.

Capacity

Refer to Table 19 on page 92 for Meet-Me Audio Conference application capacities. The dealer is responsible for limiting the licenses to the capacities shown in the table.

Licenses

Four channel licenses (I-MTG-4CH-DISC) are included in all IPedge Virtual Server bundles. Additional channels can be purchased using part number: I-MTG-CH.

Meeting

The Meeting application is integrated into the IP*edge* Application Server for Strata CIX system (prior to IPedge 1.7.0 software release). IPedge Application Servers running release 1.7.0 software requiring Toshiba's Meeting feature will require a Meeting Application Server on separate physical server. 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 IP*edge* EC App server; up to eight conferences with a total of 48 participants on IP*edge* EM App server; and one conference with a total of four participants on IP*edge* EP App Server. 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.

IP*edge* Component Compatibility

The IP*edge* 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 IP*edge* 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 15 Component Compatibility

System Software and Features	EC App Server	EM App Server	EP App Server
Linux Operating System	X	X	X
Contact Center on Standalone	**EC App Server**	**EM App Server**	**EP App Server**
ACD	X	X	X
TASKE Call Center Reporting	X	X	X
Administration	**EC App Server**	**EM App Server**	**EP App Server**
Enterprise Manager [Browser-based Unified Admin]	X	X	X
Enterprise Manger Centralized Management Primary Server	X	X	NA
Messaging and Collaboration	**EC App Server**	**EM App Server**	**EP App Server**
Messaging (integrated)	X	X	X
Meeting (integrated)	X	X	X
Unified Communications	**EC App Server**	**EM App Server**	**EP App Server**
Call Manager	X	X	X
Other Telephone Systems	**EC App Server**	**EM App Server**	**EP App Server**
Strata CIX system - direct SIP registration	X	X	X

Application Capacities

Table 16 Enterprise Manager

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

Table 17 Media Server

	EC App Server	EM App Server	EP App Server
Resources	174	480	22

Table 18 Meeting

	EC App Server	EM App Server[1]	EP App Server
Audio Channels	24	24	4
Web Sessions	24	24	4
Video Sessions	24	24	4
Conference Record	4	8	1

1. Limit of 24 on IPedge App Server for Strata CIX configuration due to requirement for single MIPU on Strata CIX.

Table 19 Meet-Me Audio Conference

	EC Server	EM Server	EP Server
IPedge Branded Server Conference Users	32	48	4
IPedge Virtual Server Conference Channels	100	200	20

Notes

- The number of conference users does effect the station capacity of EM servers.
- The number of conference users does not effect the station capacity EP or EC servers.

Table 20 Call Manager

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

Table 21 Messaging

	EC App Server	EM App Server	EP App Server
Departments	999	999	999
Mailboxes (basic or UM)	5,000	10,000	1,000
Script Mailboxes	20	20	20
Simultaneous Calls	24[1]	24[1]	8 or 24[2]
Hours of Storage	4,000 hours	7,000 hours	4,000 hours

1. Limit of 24 on IPedge App Server for Strata CIX configuration due to requirement for single MIPU on Strata CIX.
2. If 24 channels are enabled, the system cannot be upgraded to include call processing using the I-APP-UP-EP-DSC.

Part Numbers

The IPedge Application Server only requires one license per platform.

Table 22 IPedge Application Server for Strata CIX Part Numbers

Platform	Part Numbers Required	Description
EP Application Server	I-EP-1A	IPedge EP server with AC adaptor. Factory equipped with single 250GB hard drive, 4GB RAM, and all the necessary software to support IPedge features.
	I-APP-EP-DISC	IPedge EP Application Server Discount Bundle. Includes system license, recovery DVD, and 6 free call manager standard licenses. One system license required per Server. Includes 1 x I-APP-EP, 6 x I-CM-STD1 and 1 x I-RCVY-DVD-VF. Limit 1 per system. I-APP-EP includes 1 X SYS-PLTFM-EP, 1 X I-MSG-BASE, 6 X I-MSG-ADV.
	I-APP-UP-EP-DSC	The IPedge Application Server Upgrade License for the EP at a special discount. Includes I-APP-UP-EP [Includes 6 x I-CP-USR-EP, 3 x I-CP-TRUNK, 1 x I-MS-BASE, and 8 x I-MSG-CH. Requires I-APP-EP]. Requires I-APP-EP
	I-APP-EP-VM24	Upgrade to 24 Messaging channels for the IPedge EP. **Important!** An IPedge EP App Server with 24 Messaging channels can not be upgraded to include call processing with the I-APP-UP-EP license. To upgrade, migrate to an IPedge EC server. Limit 1 per system.
EC Application Server	1-EC-1A	IPedge EC model rack mount server. Factory equipped with single 160GB SATA hard drive, 4GB RAM, and all the necessary software to support IPedge features. Mounting rails are required, see the 4-post (I-EC-RL4-1A) or 2-post (I-EC-RL2-1A) rail kits sold separately.
	I-APP-EC-DISC	IPedge EC Application Server Discount Bundle. Includes system license, recovery DVD, and 24 free call manager standard licenses. One system license required per Server. Includes 1 x I-APP-EC, 24 x I-CM-STD1 and 1 x I-RCVY-DVD-VF. Limit 1 per system. I-APP-EC includes 1 X SYS-PLTFM-EC, 1 X I-MSG-BASE, 24 X I-MSG-ADV.
	I-APP-UP-EC-DSC	The IPedge Application Server Upgrade License for the EC at a special discount. Includes I-APP-UP-EC [which includes 24 x I-CP-USR-EC, 12 x I-CP-TRUNK, 1 x I-MS-BASE, and 32 x I-MSG-CH] Requires I-APP-EC.

Table 22 IPedge Application Server for Strata CIX Part Numbers *(continued)*

Platform	Part Numbers Required	Description
EM Application Server	I-EM-1A	IPedge EM model rack mount server. Factory equipped with 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.
	I-APP-EM-DISC	IPedge EM Application Server Discount Bundle. Includes system license, recovery DVD, and 32 free call manager standard licenses. One system license required per Server. Includes 1 x I-SYS-EM, 32 x I-CM-STD1 and 1 x I-RCVY-DVD-VF. Limit 1 per system. I-APP-EM includes 1 X SYS-PLTFM-EM, 1 X I-MSG-BASE, 32 X I-MSG-ADV.
	I-APP-UP-EM-DSC	The IPedge Application Server Upgrade License for the EM at a special discount. Includes I-APP-UP-EM [Includes 32 x I-CP-USR-EM, 16 x I-CP-TRUNK, 1 x I-MS-BASE, and 60 x I-MSG-CH] Requires I-APP-EM

Additional Application Licenses

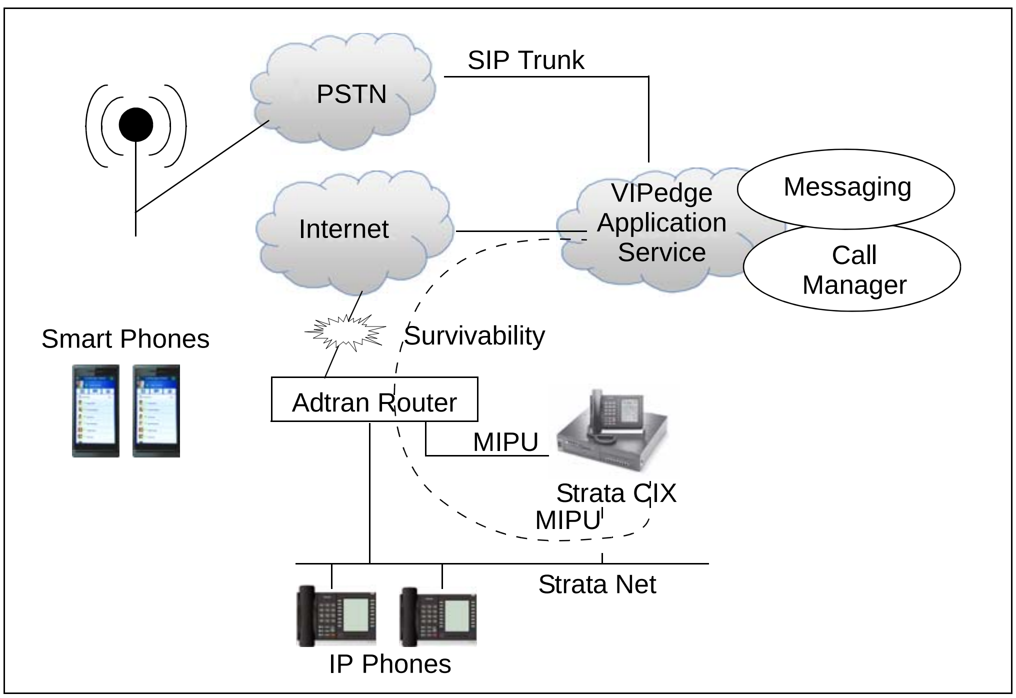
License Part Number	Description
I-MSG-ADV	IP*edge* IP Messaging Advanced User - per user. This license includes basic voicemail features plus unified messaging.
I-CM-1	IPedge Call Manager Advanced, 1 user license. Provides desktop call control. PC phone functionality, and chat text messaging capabilities. One license is required for each user. VoIP voice plug-in is sold separately per user.
I-CM-STD1	Single client license for 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 plug-in license to add VoIP for Strata Call Manager. Requires one IPedge Call Manager License (I-CM-1) as well as one IPedge user license (I-CP-USR-XXX).
I-MT-A	IP*edge* Meeting meet-me conferencing audio channel License. One required for each simultaneous meet-me audio conferencing participant. Minimum 4.
I-MT-RCD	IP*edge* Meeting Audio Conference Record License. One required for each simultaneous channel of audio conference recording.
I-MT-V	IPedge Meeting Video Channel license. This is used to upgrade a system to video. The number of video channels must equal the number of web channels.
I-MT-W	IP*edge* Meeting Web Conference Application - per concurrent user IP*edge* Meeting meet-me conference web collaboration channel license. One required for each simultaneous web collaboration session participant.

VIPedge Application Service 9

VIPedge Application Services for Strata CIX allow premise based Strata® CIX systems to take advantage of unified messaging and unified communications applications and survivability in the cloud. Strata CIX 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 MIPU02_12 software.

SIP Trunks, Voicemail and Call Manager in the cloud

VIPedge Application Service can be used with Strata CIX – creating a fully supported SIP Trunk, voicemail and unified communications for premise telephone solution. This offers unified communications features, support, and the reliability of a cloud based solution with the ability to use IP or digital phones as required. The Strata CIX system is connected to the internet behind a router, and it is networked to the VIPedge Application service using Strata 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. The VIPedge Application Service is connected by SIP Trunks to the PSTN.



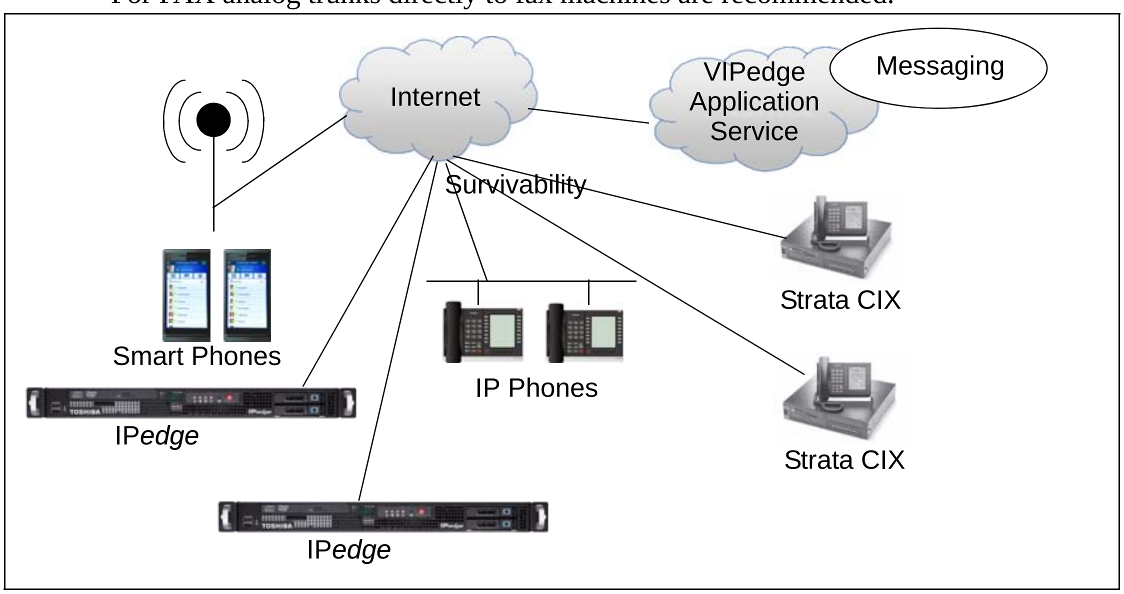
VIPedge Application Services offer the following features:

- **Messaging** – Calls that are not answered by a 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 Strata CIX system on 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 is lost, station-to-station to calling is still available on 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.
- **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.
  - For smaller sites using the CIX40, a single MIPU card with up to 24 channels can be given a public IP address and used to connect to VIPedge Application Server, and IP Phones. For larger sites one or two MIPU’s can be used for the VIPedge application server. Additional MIPU cards can be used for IP Phones.
  - 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 Strata CIX business telephone solution.

Strata CIX 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

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.

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Features 10

This chapter contains the Strata CIX features. They are presented in alphabetical order to make it easy to locate each feature. System availability below a feature title informs you of the Systems that support the particular feature.

Account Codes

System Availability: All systems

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. An account code feature button can be programmed on a digital telephone to make voluntary account code entry convenient and easy.

Add-on Module (ADM)

System Availability: All systems

One to two LM5110's (10 button) can be attached to DP5000-series digital telephones (except DP5008) and IP5000-series telephones (except IP5522-SD,5622-SD) to provide an additional 20 programmable buttons.

One to two KM5020's (20 button) can be attached to the DP5000-series digital telephone (except DP5008) and IP5000-series telephones (except IP5522-SD,5622-SD) to provide an additional 40 flexible buttons.

One to two IADMs can be attached to an IPT2000-series telephone to provide an additional 20 or 40 flexible buttons.

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

Advisory Messages

System Availability: All systems

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

Alarm Notification

System Availability: CIX100, CIX200, CIX670, and CIX1200

The Strata CIX can send alarm notifications to a Monitoring PC/Server or send an alarm notification to a telephone. The Strata CIX Network eMonitor software application provides system alarm monitoring functionality, either remotely or locally over TCP/IP.

System alarms can be sent to up to 11 unique eMonitor PC consoles IP addresses from Strata CIX SNMP traps. Alarms can be sent from multiple networked Strata CIX nodes to one or more eMonitor consoles.

Alarms include trunk failures on ISDN PRI, T1, or IP interfaces. System resource alarms include cooling fan failure (CIX200 only), MIPU card data set problem, SMDR memory buffer full, SMDR link down (LAN/RS-232c), SMDI link down (LAN only), CTI link down (Attendant Console, ACD), and Expansion cabinet power supply failures.

Alternate Answer Point

System Availability: All systems

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

System Availability: All systems

Automatic Busy Redial (ABR) enables a digital or standard telephone user to automatically redial a busy outside number multiple times at programmed intervals. Strata CIX supports a maximum of 16, 32 or 48 simultaneous registrations of ABR (limited by the number of busy tone detectors in the system). Each station may only have at most one call registered with ABR at any time.

Automatic Call Distribution (ACD) Server

System Availability: All systems

An external ACD software option with the Strata CIX is provided by connection of an external PC-based CTI application server. The CTI server runs the ACD call processing application and 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, refer to the *Strata CIX and CTX Call Center Solutions 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 by higher priority calls.

Automatic Callback (ACB)

System Availability: All systems

When a station user dials a busy station [DN] or outside line access code and receives busy tone, 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 dial tone from an outgoing line.

When ACB is activated, the calling station receives success tone followed by 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

System Availability: All systems

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 will be answered as a priority, then the longest ringing call in that call type will be 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.

Automatic Release

System Availability: All systems

The system will automatically release line connections under certain conditions.

Automatic Release from Hold

When a line is on hold and the held party hangs up, the line is automatically released. Individual loop start CO lines can be programmed to detect disconnect supervision signals from the CO and to respond by releasing the line. If the CO does not provide Disconnect Supervision, the user must manually retrieve the held line and then hang up.

Automatic Release of Incoming Calls

An outside caller may be placed in a queue waiting for an external application to handle the call such as an Auto Attendant, IVR, ACD or other device. If the CO line for that call does not offer "disconnect supervision," that call may remain in a holding position until forced to release the connection.

This feature provides full use of all CO lines at all times. A CO line is not tied up if the call goes unanswered and no alternative call handling is provided. This is very useful for disconnect supervision in voice mail and built-in auto attendant applications, but availability and reliability of the signaling from the CO must be confirmed.

Station Automatic Release

When the distant party disconnects from a call, the remaining digital telephone is automatically made idle, busy tone is not sent to the speakerphone or handset, and the digital telephone is automatically released. A digital telephone is released and returned to idle state. A standard telephone is simply released and returned to standard dial tone. The programming choice for this feature is system wide.

Background Music (BGM)

System Availability: All systems

Background music audio can be played through the speakers of digital telephones and external paging equipment. The Strata CIX supports up to 15 BGM audio input interfaces. Selection of which BGM source is played can be done individually by each telephone user and for each external page zone through the System Administrator's telephone.

Call Completion

System Availability: All systems

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 separately to permit the feature to be activated when called.

Call Forward

System Availability: All systems

Call Forward diverts internal and external calls intended for a Directory Number [DN] to a destination specified for that [DN], under calling conditions specified for that [DN]. Call Forward may be activated from the station that owns the [DN] or remotely from another station or from outside the system from a DISA line. Call Forward may be applied to any [DN] ([PDN], [PhDN], or Pilot [DN]).

There are two types of Call Forward options: System Call Forward and Station Call Forward. Each type may be activated independently or simultaneously for each telephone. If Station CF is activated, it will override System Call Forward on some or all calls.

Station Call Forward

Station users can set their individual call forwarding conditions and destinations as they choose (see "Call Forward Conditions" and "Call Forward Destination" in this section). Station forwarding has priority over System Call Forwarding, if set.

Station Call Forward provides two types of Call Forward (Any Call and Incoming line calls only). One type directs any type of a call to a designated destination; the other type directs only private or DID lines to a designated destination. Both types can be set on a telephone simultaneously with each type having a unique destination.

This allows the user to forward incoming calls on private or DID lines to a different destination than internal or transferred calls. If private and DID line calls are set to forward independently to an alternate destination, then internal and transferred calls will forward to another destination per Station Call Forward (any call) or System Call Forward.

System Call Forward

A system option is available to forward unanswered calls to voice mail or some other pre-determined destination. This option is set up for each station by the System Administrator using Network eManager. This ensures efficient call handling and better service to callers even when station users do not have Station Call Forward set at their telephone. Call Forwarding can also be set up by department with a special mailbox or destination with the use of Phantom Directory Numbers [PhDNs].

There are 32 or 48 different System Call Forward patterns that can flexibly forward calls with unique call type, condition and destination settings. Each pattern can be set up and assigned to individual stations by a System Administrator using Network eManager. Any pattern can be applied independently to each station's [PDN] or [PhDN]. System Call Forward patterns applied to stations can be changed automatically per Day/Night CO assignments.

Although System Call Forward is set up and assigned to individual telephones by a System Administrator, each telephone user can turn the feature On/Off from their telephone using a One Touch button or access code. Station Call Forwarding always overrides System Call Forward.

With Release 1.3 and higher, you can enable/disable System Call Forward Cascade, which means that a call that forwards to a destination that is also forwarded will follow the destination's call forward.

Call Forward Conditions

Call Forward (CF) conditions refer to the status of the [DN] that causes a call to forward. Whether using Station or System Call Forward, the CF conditions include: Busy, No Answer, Busy-No Answer, and All Calls (station CF only). Call Forward No Answer times are set individually for each station in Station Call Forward and system wide for all System Call Forward Patterns.

Note OCA and Voice First Calls will not Call Forward-No Answer unless the caller presses 1 to switch the call to tone ringing.

Call Forward Destination

Whether using Station or System Call Forward, the CF destination can be an internal Directory Number, a Hunt or ACD Group, Voice Mail, or a public or private network telephone number.

In the last case, the forwarded call will access an outside line or line group and dial an external telephone number. Both the line access code and the telephone number are set in the Call Forward destination during the Call Forward setup operation.

Station Call Forward allows one destination per each type of Station Call Forward (Any Call or Incoming Line Calls) set on a telephone. System Call Forward allows two destinations per Call Forward pattern: the Primary Destination and an alternate, in case a call cannot forward to the Primary Destination (e.g., the Primary Destination has been unplugged or malfunctions).

Call Forward – Call Types

In each System Call Forward pattern, the Call Forward conditions and destinations can be set independently.

For Station Call Forward, the destination and condition for each station can be different for incoming CO line calls, and internal and transferred calls.

Call Forward Remote

A station’s personal call forwarding destination can be cancelled or changed to another outside number or an internal voice mailbox either remotely via DISA or from another user’s telephone. Changing Call Forward remotely is password protected. System Call Forward can be changed locally or remotely using Network eManager.

Call Forward Override

See “Call Forward Override” on page 141.

Call History

System Availability: All systems

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

When calls ring a button (Line or [DN]) that appears on multiple stations, the number is stored on the telephone that is designated as the owner of the Line or [DN] and on the telephone that answers the call. If an incoming call is directed to a telephone, but the call is not answered by that telephone because it hunts or forwards to another destination, the call record will still be 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. The number of call records allowed per station and the total number of call records per system is provided in Table 56 on page 187.

Call Monitoring and Transfer

System Availability: All systems

This feature is available on the following Voice Mail systems. This optional feature enables a mailbox user to monitor a message while it is recorded in his mailbox from his telephone and optionally transfer the message to a caller. This feature is active when the User’s telephone is idle or for calls that are forwarded to Voice Mail, and when a message recording begins.

To operate, this feature must be initially enabled by the mailbox user. The default is disabled. The Call Monitor feature works only with real extensions or PDNs.

When a call is forwarded to a voice mail system and recording starts, the mailbox owner’s telephone displays a message indicating that a voice message is being recorded with the appropriate Caller ID information. Also there could be a beep tone or audible alert at this point. If the mailbox owner has DND enabled, there should be no beep. If the mailbox owner is present at this moment, he can press the “Call Monitor Button” to hear the caller leaving a message. The message plays in real time. When the caller stops the recording, the monitoring terminates, and the mailbox user will hear the prompt “The caller has finished. Good bye.”

You cannot cancel or rerecord from the voice mail system when Call Monitoring and Retrieve feature is enabled and invoked.

The mailbox user has an option to interrupt message recording and speak to the caller. In this case, the mailbox owner presses the “Call Monitor Button” button on the telephone. The caller leaving a message will be interrupted with a prompt such as “Bill Jones is now available, please hold while your call is being transferred”. The system will connect the caller to the mailbox owner, which will hear an audible tone once the connection is complete. Other callers are not allowed to ring the mailbox owners phone once the “Call Monitor Button” button is pressed.

When more than one caller is leaving messages at the same time, then the mailbox user is able to monitor the last caller.

Call Park

System Availability: All systems

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 called Auto Park.

Call Park Orbits

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

Refer to Table 56 on page 187 for the number of General Park and Personal Park Orbits, depending on the system processor.

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

System Availability: All systems

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. Station users can also pick up CO/DID/Tie line calls ringing or on hold at other stations, CO lines ringing during Night Mode to External Page or night bell, tandem CO line connections and Door Phone calls. Call pickup can be performed through programmable buttons (Directed Pickup, Group Pickup), or with an access code.

Call Waiting

System Availability: All systems

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

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

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

Digital telephones can be adjusted to receive or not receive Call Waiting tone over the handset or headset receiver, as well as the speaker. Standard telephones will receive Call Waiting tone twice from the handset receiver. Call Waiting tones can also be turned off on each station by a System Administrator.

When a station is busy with a call and another call is being received, 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.

This feature works with both digital and analog single-line telephones. The tone (two beeps) signaling Call Waiting tone is provided through the speaker of the digital phone. For standard analog telephones, the tone is inserted into the speech path. Caller ID display is not available with standard telephones.

Note Type II call waiting with Caller ID is not available with the R4.1 release of the BSTCIU or BCOCIU/BCOCIS interface cards.

Caller Identification

System Availability: All systems

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 (see Call History). Caller ID service from the carrier must be subscribed on analog CO lines or T1 ANI in order to receive calling number and name into the Strata system.

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 the Call History. See Table 56 on page 187 for the number of call records per system.

ISDN Calling ID Name and Number

System Availability: Strata CIX100, CIX200, CIX670, and CIX1200

Both Caller ID name, if available from service provider, and number are supported for incoming calls using either ISDN NI-1 or NI-2 PRI lines. Caller ID names can be accepted by the Strata system from a CO with NI-1 or NI-2. NI-2 is the only way a 5ESS CO delivers Caller ID names on PRI. Nortel DMS CO with NI-2 installed also uses NI-2 to deliver Caller ID names. Only a Nortel DMS CO can provide Calling Names for NI-1.

Camp on Busy

System Availability: All systems

Automatic Camp On

When a call comes in to a busy station from an outside line and that station does not have an idle button for the call to ring in on, and Station Hunting or Call Forward is not applied, the call automatically camps on to the busy station. This permits incoming calls to be accepted even if the station is busy. The outside caller will receive ring-back-tone immediately and the called station will receive two bursts of Call Waiting tone.

If the calling line has Caller ID, ANI, or DNIS information, it will be displayed on the called station's LCD for 10 seconds. Auto Camp On also applies to incoming line calls directed to Hunt Groups, Voice Mail systems, etc.

Various types of internal calls from one station to a busy station, voice mail system or hunt group can also Camp On automatically with system programming options. For details on these types of calls see the Camp On-Busy and Station Hunting descriptions.

Off-hook Camp On

A station caller who dials a busy station or line access code can remain off-hook to be automatically connected when the station or line becomes idle. After dialing a busy [DN] and receiving busy tone, the caller can just remain off-hook and Camp On will be initiated automatically after a predetermined time or the user can dial a `1` and remain off-hook to initiate Camp On immediately. When camp-on is activated, the caller will receive success tone followed by Ring Back Tone. The station that is the object of a camped-on call will receive two bursts of call waiting tone (see Call Waiting).

Even if Voice First is set at the called [DN], the station will be called by tone ringing when it is connected by Camp On. Internal and external stations can be the object of a Camp On. Calls may be camped on to the pilot number of Station Hunting groups and will be delivered to the first station in the group to become idle. ACD pilot numbers cannot be the object of a Camp On.

Incoming calls from outside lines to busy DNs camp-on automatically (see "Automatic Camp On," previous section). When a station dials the access code for an outside line and receives busy tone because all lines are busy, the user can remain off-hook and dial `1` to camp on to the busy line group. When a line becomes available, the station will connect to the line and receive dial tone.

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

Calls will camp on to hunt groups when all members of the group are busy (see "Station Hunting" on page 145 for more details).

Cancel Button

System Availability: All systems

The `Cancel` button voids the last entry or step in a procedure. This enables the station user to correct an error and then continue without having to starting over.

It is important to consider the consequences of this button in regards to the overall task. For example, during a conference call, `Cancel` will disconnect the last party added to the conference.

Centrex/PBX Compatible

System Availability: All systems

All system features are compatible with Centrex/PBX operation, including repeat of Centrex/PBX ringing cadence, one-button access to Centrex/PBX features, a two- to five-digit station numbering plan, and Delayed Ringing to selected stations.

Centrex Ringing Repeat

System Availability: All systems

The system can mimic CO/Centrex/PBX ringing cadences received from outside lines when it rings a called station.

Classes of Service (COS)

System Availability: All systems

Classes of Service are the mechanisms for assigning features and services to lines and stations within the system. The Class of Service for a given device, such as a station, is defined using 42 parameters. There are 32 Class of Service patterns available, each pattern can be set up to allow a unique combination of features. Each station and line group can be assigned independently to one of the 32 COS patterns.

Computer Telephony Integration (CTI)

System Availability: All systems

CTI combines the capabilities of the Strata CIX digital business telephone system with custom functionality provided by computer applications. This can be provided through the LAN connection.

Conference Calls

System Availability: All systems

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

Conference On-Hold

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

Join Button

Join allows an attendant or digital station user to connect two established calls to each other.

Split/Join/Drop

This feature enables the conference master to add (Join) other phones to a conference. The conference master and another member of the conference can leave (Split) the conference for a private conversation. During this time, other conference members remain connected. The conference master can then Join both of the Split callers back into the conference, or the master can Drop (disconnect) the Split member he/she is connected to. This feature requires an LCD phone with Soft Keys. A flexible Split button can be added to a 3000- or 2000-series telephone to use this feature.

Releasing from Conference Tandem CO Line Connections

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

Standard telephones and/or VM devices can establish tandem analog CO line connections and then release from them without disconnecting the tandem connection in the Strata system. After releasing from a tandem call, reconnecting to the call can be accomplished by dialing an access code. This reconnect feature does not work if one or both of the CO lines are digital.

Whether or not tandem line buttons appear on a telephone, the telephone user can enter the connection and release the line that was connected to the original line or release both lines by pressing the Cancel button. For details on 2-B channel release from conference transfer, see "Integrated Services Digital Network (ISDN)" on page 137.

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.

Continuous DTMF Tone

System Availability: All systems

Dual-Tone Multi-Frequency (DTMF) dial signal is transmitted to the CO line or voice mail/Auto Attendant device for as long as the telephone user presses a button on the dial pad. This feature may be selected for each digital telephone. Standard telephones always provide continuous DTMF tone operation.

Credit Card Calling

System Availability: All systems

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 Strata CIX CO line. The "0+" credit card calling feature can be enabled, selectively, or assigned to stations and CO lines capable of supporting this service.

Data Privacy

System Availability: All systems

This option blocks calls to data devices that are in use. This prevents override calls and warning tones from interfering with data devices such as modems and ISDN data terminals.

Day/Night Mode – Auto Schedule

System Availability: All systems

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 Day, 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 routing of individual DID and DNIS numbers or ground/loop start lines 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 38 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.

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 button can be set on telephones for manually switching at any time from one mode to another. The button's LED flash rate indicates the system's operating mode.

If used with the System Auto Schedule operation, the Night Transfer button 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 button (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.

Delayed Ringing

System Availability: All systems

If an incoming external or internal [DN] call rings a station [DN] and is unanswered, alternate DPs 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 DP.

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 to voice mail, auto attendant and/ or standard telephones 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.

Destination (Toll) Restriction

System Availability: All systems

Strata CIX offers Destination Restriction as a major expansion of traditional Toll Restriction. Historically, Toll Restriction was used to prevent the unauthorized use of toll prefixes to the PSTN: long distance (1), operator assistance (0) or international (011).

Strata CIX has expanded this to include restriction based on any string of dialed digits. A true, international business telephone system, Strata CIX can restrict any string of up to 11 dialed digits, including ⋆ and #. Eleven-digit screening allows control of access to individual telephone numbers in remote Area Codes. Restriction of ⋆ and # controls users' access to service codes from the CO, such as Camp On and Call Forwarding.

A stations's Destination Restriction level can be changed automatically with Day/Night mode Auto Scheduling. One use of this feature is to allow a telephone to make outside calls during the day, but to restrict them at night.

Through Dialing

A telephone user or an attendant can connect a destination-restricted station to a trunk enabling temporary access to an outside line. The connected station can then use external dial tone to complete the call, and revert back to destination-restricted status after the call is completed. This maintains the integrity of toll restriction, while still extending outgoing calling privileges when necessary.

Dial Directory

System Availability: All systems

Station users can dial by name using Toshiba's DP5000-series digital and IP5000-/IPT2000-series 10- and 20-button LCD telephones. The Dial by Name feature searches for names much like a cell phone directory and then allows the user to press on 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.

Note Dial Directory is not compatible with DKT2000-series digital telephones.

Direct Inward Dialing (DID)

System Availability: Strata CIX100, CIX200, CIX670, and CIX1200

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, which is sent to the CIX on a DID line from the CO, can be routed individually to an extension, pooled or group line button, ACD group, maintenance modem, external page, night bell, voice mail box, or back out over the public or private telephone network. DID routing assignments can change automatically when the system switches between the Day and Night Modes.

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

DID service is provided by DID analog, T1 or ISDN line interfaces.

Dialed Number Identification Service (DNIS)

System Availability: Strata CIX100, CIX200, CIX670, and CIX1200

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

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

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

Digital Pad

System Availability: All systems

The Strata system digital pad (decibel loss) is activated for the receiving path of the terminal, external line or resource.

The system adjusts for differing transmission levels between internal and external devices. This is very useful for conference calls when external parties have difficulty hearing due to public network loss. The Strata CIX can be programmed to insert the appropriate digital pad for each terminal and call type when establishing speech paths between telephones, external lines, and resources such as conference circuits, external paging devices, and external sound sources. This minimizes volume loss in conference calls.

The system recognizes these devices:

- Standard analog telephone (Type 500, Type 2500 and the equivalent)
- Toshiba digital telephone, cordless, wireless, door phone
- Analog trunk
- T1 trunk
- ISDN extension terminal (Audio and Speech)
- ISDN trunk/Tie line (Audio and Speech)
- Conference circuit
- Holding music source
- External paging device

Direct Inward System Access (DISA)

System Availability: All systems

Direct Inward System Access (DISA) allows outside callers to connect to the Strata CIX and make station or trunk calls as if they were stations within the system. An incoming call may be directed to DISA by Direct Inward Dialing lines, ground/loop start lines or Automated Attendant.

Note DISA lines require DTMF receivers.

DISA security code is changeable from a specific station. The station to change the security code needs to be allowed by Class of Service. This security code can also be changed using the Strata Network eManager administration software.

DISA provides access to the features listed below:

- Station Calls
- Station Calls over Private Network
- Attendant Access
- Account Codes
- Tie lines

DISA also provides access to these features, which require a security code:

- LCR
- Direct line access
- Outgoing line group access
- Emergency Call
- Call Forward Remote Control

Directory Numbers

System Availability: All systems

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

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

Primary [DN] Buttons

Primary Directory Number buttons [PDNs] are needed to make calls 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 [SDNs].

Phantom [DN] Buttons

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

[PhDNs] can be used to make a telephone appear to have multiple [PDN] extension buttons. When assigning a [PhDN] for use as another appearance of the [PDN], the display properties are set the same as the [PDN] and a hunting sequence is set up to roll the calls from the [PDN] over to the other [PhDN]s associated with the [PDN]. With this arrangement, Call Forward will send 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 Interface (CTI) can control the call. To ensure calls do not get lost in the Strata CIX, 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 Strata CIX.

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

Distinctive LED Indicator

System Availability: All systems

Each feature button on a digital telephone has a Light Emitting Diode (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

System Availability: All systems

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

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

Do Not Disturb (DND)

System Availability: All systems

Station users with digital 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. With Release 1.3 and higher, users can disable DND stutter dial tone (1/2 sec. burst of busy tone before dial tone) in programming.

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 (see "Do Not Disturb (DND) Override" on page 141).

Direct Station Selection (DSS) Buttons

System Availability: All systems

[DSS] buttons can be placed on digital and 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 dial tone when pressed. The [DSS] button LED shows the status of the station and [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 (see "[DSS] Button Status Display."

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] itself. 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

DTMF Receivers

System Availability: All systems

DTMF receivers are used when receiving incoming DNIS DID, Tie or DISA line calls and when originating calls with standard tone-dial telephones. Voice mail systems also require DTMF receivers for a number of VM features, even if using SMDI or Toshiba Proprietary VM integration. Four circuits are automatically active with the initial basic processor. Activation of more than four receivers requires the purchase of a DTMF software license, in four-circuit increments.

DTMF receivers are built into the Strata CIX100, CIX200, CIX670, and CIX1200 processors. For the number of receiver circuits, refer to Table 56 on page 187.

DTMF Back Tone

The system can be programmed to allow or prevent Dual-tone Multi-frequency (DTMF) tones from being returned to digital telephones when a user dials on outside lines or sends DTMF digits to a voice mail device.

DTMF and Dial Pulse CO Line Compatibility

When making outside calls, signals generated by pressing the dial pad buttons of a digital telephone are neither DTMF nor rotary dial signals – they are digital signals. The system can be programmed to translate these signals to either DTMF or rotary dial signals as required by the serving CO. Once the connection has been made, any further digits sent will always be sent as DTMF or rotary to allow the operation of devices at the other end of the connection.

DTMF Signal Time

DTMF tones that are sent via Speed Dial to lines and via automatic dialing to voice mail devices can be set to 80 or 160 milliseconds, or continuously. The time can be set system wide independently for line out-dialing and for voice mail automatic dialing.

Emergency Call

System Availability: All systems

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

Feature Prompting with Soft Keys

System Availability: All systems

As an alternative to dialing access codes and using feature buttons, station users with LCD digital telephones use Soft Keys (shown on their LCD) to access features. Abbreviated feature names appear during a call (when the telephone is in the ring or talk state) on the LCD above 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.

Enhanced E911

System Availability: All systems

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 to which emergency services are to be sent. The Strata CIX can use two types of trunks to deliver E911 calls: ISDN Primary Rate Interface and CAMA (Centralized Automatic Message Accounting) trunks. Each 911 call generates an SMDR record at the beginning of the call to enable the business to initiate its own emergency response. Internal emergency destinations can also be automatically included in an emergency call. E911 calls can be routed across a Strata Net network for connection to the public network.

External Amplified Speaker

System Availability: All systems

The External Amplified Speaker (BESCB) is a six-inch, three-watt speaker with a three-watt amplifier built into a wooden speaker box. The amplified ringer can be used to:

- Amplify the ringing on a digital telephone.
- Provide a paging amplifier/speaker.
- Create an amplified talk-back speaker arrangement in an area where a telephone is not needed. The BESCB is installed as a speaker and connected to a door phone unit that is used as the talk-back microphone.

Amplified ringing can improve call handling in noisy areas where non-amplified ringing on a phone may not be heard.

A paging speaker ensures that paging announcements can be clearly heard throughout an area. In an area where a DP is not needed, a talk-back speaker provides a cost-effective communications solution.

The number of BESCBs that can be installed per system depends on the function of the BESCB. Any number of BESCBs can provide loud ringers for digital telephones. Only one BESCB can be installed if it is used as a paging or an amplified talk-back speaker.

Note A BHEU interface and an HESC-65A cable are required for each digital phone that has a loud ringing bell. IP telephones require HESC-65A, but not BHEU to support External Amplified Speaker. A 3000- or 2000-series digital telephone that has been upgraded with a data interface unit can be upgraded with the BHEU options, but older telephone models cannot.

Flash Button

System Availability: All systems

This is an optional button that can be assigned on digital telephones. It can be used either to disconnect a line and regain CO dial tone, or to gain access to Centrex features. The timing choice is set system wide through system programming. Standard telephones can dial an access code to flash Centrex lines.

Flexible Line Ringing

System Availability: All systems

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

System Availability: All systems

The Strata CIX allows the system-numbering plan to 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.

Handsfree Answerback

System Availability: All systems

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

Headset

System Availability: All systems

3000-series digital telephones may be optionally equipped with a modular headset jack by installing a BHEU circuit card.

IP telephones and DP5000-series digital telephones have a built-in headset adapter and therefore require only the headset.

Hearing Aid Compatible

System Availability: All systems

All Toshiba digital telephones are hearing aid compatible.

High Call Volume Buttons

System Availability: All systems

Release, Release/Answer, and Cancel buttons can be assigned to digital 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

System Availability: All systems

There are several variations of Hold:

Automatic Hold

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

Analog Hold

This option enables a user to place a CO Line or [DN] call on Hold and the Line LED will flash on other DP telephones when the call is parked. This enables the call to be picked up from other telephones. This feature must be set in programming.

Call Hold

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

Consultation Hold

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

Exclusive Hold

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

Hold Recall

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

Hot Dialing

System Availability: All systems

Hot dialing enables the digital 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

System Availability: All systems

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.

IP Telephony

Data Network Assessment for Voice Traffic

System Availability: All systems

Important! *Voice over Internet Protocol (VoIP) can be affected by numerous factors related to network structure and design. To prevent delays, jitter and voice data packet loss, and achieve optimum VoIP traffic performance, your network must be designed or redesigned for voice traffic with priority over data.*

Toshiba highly recommends that the installing dealer assess the existing network for proper handling of voice traffic and make sure it provides the required bandwidth per the number of deployed VoIP instruments. Toshiba is not responsible for: network assessment for voice traffic, network design/redesign or network support outside the topology it provides. Nor is Toshiba responsible for diminished Quality of Service (QoS) caused by networks not capable of providing necessary VoIP bandwidth.

Expected voice QoS as it relates to network parameters is shown in Table 46 on page 180.

The CIX supports Toshiba proprietary IP telephones, enhancing the CIX VoIP capabilities, and providing powerful IP telephony add-on solutions for remote users. The Toshiba IPT2000- and IP5000-series telephones are compatible with all Strata CIX systems.

The Toshiba IP telephony strategy is based upon a transitional approach. For most enterprises, the migration path to IP telephony will be a gradual process rather than an event in time. Instead of acquiring IP technology benefits through complete system replacement at higher cost and higher risk, most enterprises prefer to integrate voice and data IP traffic into their existing telephone systems as the need arises. This protects their investment in existing voice and data networks and represents a low risk migration path. Toshiba has transformed Strata CIX systems into IP-enabled communication systems to achieve this very important objective.

IP Interface Unit

System Availability: All systems

The MIPU and GIPU / GIPH IP interface card works in conjunction with the system processor to provide distributed IP processing capabilities that include peer-to-peer IP telephone communication and IP Strata Net multi-system networking. MIPU and GIPU / GIPH cards install in card slots in Strata CIX base or expansion cabinets.

The MIPU comes in two flavors: MIPU16 and MIPU24. The MIPU supports up to 24 channels even when the echo tail length canceller is set up to 64ms. The GIPU / GIPH installs into the Strata CIX40 system. The GIPU / GIPH card on the CIX40 supports up to 8 IP channels. Each IP card type has a Network Interface Card (NIC).

When programmed for G.711 codec, or G.729A with 16 ms echo cancel delay the MIPU supports 16 IP channels and GIPU / GIPH supports 8 IP channels. Refer to the *Strata CIX Programming manual* for more details.

The MIPU, GIPU and GIPH supports all "IP" functionality in the Strata CIX. This includes the following:

- Multiple protocols supported:
    - MEGACO+ delivers a consistent user interface over all Toshiba IP endpoint devices, except SIP telephones. Toshiba has enabled the MEGACO protocol for call control to provide more stimulus/response capabilities that allows all the feature functionality of digital desktop telephones to be supported on IP telephones. The MEGACO+ protocol is able to implement this feature functionality consistency much better than could be done using other VoIP protocols.
    - SIP (Session Initiation Protocol) telephones provides basic functionality that can be used for very basic user capabilities. Toshiba supports only the Uniden telephone that Toshiba has approved for interoperability.
    - SIP Trunking is an application layer protocol used for establishing sessions in an IP network. SIP Trunking allows the CIX to get PRI like services from an Internet Telephony Service Provider using SIP.
- MIPU and GIPU / GIPH IP endpoint devices and interfaces (requires IP Endpoint license):
    - Toshiba IP telephones
    - SoftIPT on PDA, Notebook and TabletPC's
    - Wireless IP Telephone
    - SIP IP telephones
- MIPU Port devices and interfaces (requires Basic Port license on all CIX systems, except with GIPU / GIPH because all Basic Ports are fully licensed on CIX40).
    - IP Attendant Console
    - StrataNet IP Network resources
- In addition to the above features, the MIPU also includes the following:
    - Compatible in the CIX40, CIX100, 200, 670, and 1200
      The MIPU16 is compatible with the CIX40 with CIX Release 5.1 software.
    - Log files collected remotely from the MIPU
    - MIPU card uses one IP address
    - Tail-length echo cancellation increased from 32 ms to 64 ms (G.711 and G.729A)
    - Quality of Service (QoS) threshold alarm notification and measurement
    - IP Mobility support
    - Connects between a node using G.711 A-Law codecs and a node using G.711 Mu-Law codecs.
    - The maximum transmission interval of voice packet at using G.711 codec is 40ms
    - RTP ports used by IPTs can be user modified
    - Continuous rebooting (even after six failed attempts)
    - For a MIPU feature matrix, refer to Table 23.

Miscellaneous

- The MIPU/GIPU supports both G.711 and G.729A standard codec compressions simultaneously. The type of compression used is set independently for each telephone in system programming.
- With the MIPU/GIPU, the IP telephone works with DSL and cable routers.

  The MIPU/ GIPU operates on the network at 100 Mbps and can be connected to a fast switch router, LAN, WAN, etc. When connecting remote IP telephones to the MIPU / GIPU over the Internet, a VPN router is needed to circumvent Network Address Translation (NAT) and firewall issues by tunneling.
- MIPU/GIPU provides MEGACO+ mobility to enable roaming. The GIPU enables remote IP telephones to be connected over VPN and non-VPN IP networks.
- MIPU /GIPU firmware can be updated locally or remotely using Network eManager. 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).

Table 23 MIPU / GIPU Features

Feature	MIPU / GIPU
Channels	MIPU16:16 channels MIPU24:24 channels GIPU8:8 channels
Channel Limitation	No Limitation
Voice Codec	Same
Echo Canceller	G.168 (08/2004) Tail Length is Max 64 ms
IP Address to use	1
RTP Packet Transmission Interval	G.711: 20/ 30/ 40 ms G.729A: 20/ 30/ 40/ 80 ms
VoIP Protocol Support	Megaco+, SIP (Terminal), Strata Net IP, and SIP Trunking
VLAN	All channels are set the same.
QoS Measurement	Between MIPU/GIPU and IPT Between IPT and IPT Between MIPU/GIPU
SIP Trunking	Supported[1]

1. Use Toshiba's SIP Trunking I-VoIP Service or contact the Toshiba Sales Applications Desk for a list of approved providers.

Network Address Translation (NAT)

The Strata CIX supports the use of IP telephones that are behind NAT firewalls. NAT allows multiple devices, such as personal computers and IP telephones to share a single public IP address. This is very common in home and small office broadband networks. A Strata CIX in the main office is able to support IP telephones at remote offices that are behind broadband routers and not connected by VPN – something not supported by some VoIP vendors. Of course using a VPN is more secure, but there may be instances where it is not practical. If you would like to support a mix of devices on the private network, and devices on the public network, see the guidelines below.

NAT Firewall Guidelines

The use of a firewall between the Strata CIX, LAN or Media Application Server and the Public Internet is highly recommended. However, the use of some firewalls or routers with NAT requires proper configuration to avoid problems with IPTs registered to an MIPU card.

NAT General Guidelines

The following are general guide lines for successful IPT configuration.

- All MIPU / GIPU cards must have a Public IP Address if there are any public network IPTs. This allows the MIPU/ GIPU to resolve the NAT translated IP addresses of both IPTs involved in the call.
  This also allows you to configure an IP phone or soft IPT to work on the LAN and have the same configuration work when you take the phone or laptop to another network.
- There must not be more than one NAT router between an IPT and the internet.
- The MIPU/GIPU must have a Gateway Address assigned.
- The MIPU/GIPU should be connected to the public network, through a firewall. The firewall should use a one-to-one NAT to give the MIPU/GIPU the same IP address on the public side and the safe side of the firewall.
- The MIPU/GIPU cannot be located behind a NAT (given a private IP address). However, it can, and should be behind a firewall.

NAT Router Guidelines

There are three types of NAT routers available at this time.

- Full Cone — These are compatible with Strata CIX and Strata Net IP configurations.
- Partial/Restricted Cone — These are compatible with Strata CIX and Strata Net IP configurations.
- Symmetric — At this time these routers may cause unreliable service or cause unwanted symptoms. These routers are not compatible with Strata CIX and Strata Net IP configurations.

IP User Mobility

IP User Mobility is a set of features designed to give the user more flexibility in 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 IPT phone 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 phone to another.
- Allows the Administrator to “oversubscribe” when building IP extensions. The Administrator can create and build more IP stations using Network eManager than there are physical ports.

Advantages

- Multiple users can share one IPT (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 IPT.
- 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 IPU 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.

Requirements

- CIX Software: R5.00 MR14 or higher
- SoftIPT: 02.00.0000
- MIPU: MIPU01_01 or higher
- GIPU: MIPU01_12 or higher
- IP5000-series: All versions

Note IP User Mobility will run on an MIPU.

IPT Anywhere

System Availability: All systems

IPT Anywhere enables you to connect IP telephones remotely through the Internet and use all Strata CIX telephone features. IPT 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 IPT remote connections. When moving IPT 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 IPTs may be connected because of xDSL and cable bandwidth limitations. High speed T1, fiber, or Asynchronous Transfer Mode (ATM)-type connections are required when installing more than two IPTs 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 IPTs over the Internet, using third party or Microsoft VPN software residing on a DHCP gateway server, see Figure 18.
- 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 Strata CIX. The diagram below shows IPT Anywhere connections using the optional VPN conection.

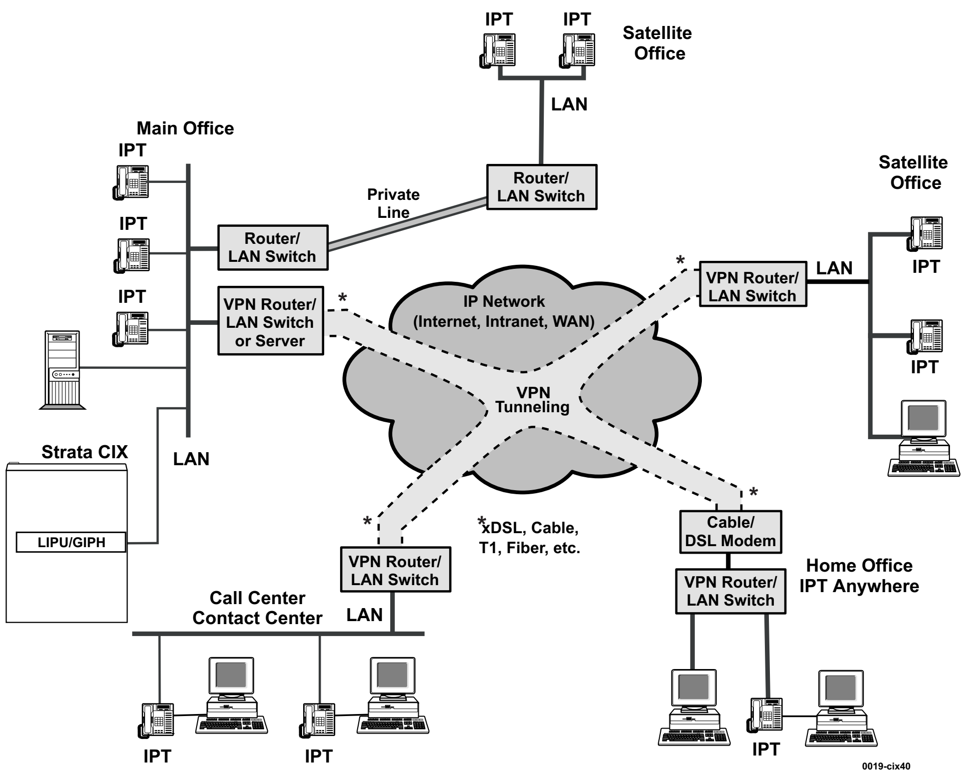

Figure 18 IPT Anywhere Example

Quality of Service (QoS) and Bandwidth

System Availability: All systems

In any telephone system, deploying many IP telephones on a data LAN can have some unexpected pitfalls if the network does not have the bandwidth and speed required to handle VoIP traffic. To prevent delay, jitter, and data loss for VoIP traffic and retain the performance of your other business-critical network applications a Network Voice Readiness Assessment must be completed before installing VoIP. Toshiba is not responsible for Network Voice Readiness Assessments.

More information about Voice Readiness Assessment can be found at Apparent Networks http://www.apparentnetworks.com/solutions/assess/overview.aspx.

CIX IP provides a number of adjustable tuning parameters dealing with sharing of network resources, collectively referred to as Quality of Service (QoS). Expected voice QoS as it relates to network parameters, including bandwidth, is shown in Table 46 on page 180.

Some CIX IP voice quality adjustable parameters are listed below:

System Wide Parameters

- Software selectable G.711 or G.729A codecs with variable interval timing
- Type of Service (TOS) precedence, delay, throughput and reliability types can be individually selected
- Differentiated Services (Diffserv) can be enabled

IP Telephone Group Parameters

- IP Telephone Groups (CIX1200: 256 groups; CIX670: 256 groups; CIX200, CIX100 and CIX40: 128 groups)
- Voice Packet Transmission Interval
- Jitter buffer type
- Jitter buffer length
- Packet loss threshold

Priority Control

Priority Control can be enabled system wide. It provides a framework in which voice traffic flowing on an IP network is given priority for processing. The CIX supports IEEE802.1p and Differentiated Services "Diffserv" priority control protocols- selectable. In order to have priority control processing work accurately, network equipment (router, switch, etc.) must support this function and appropriate service must be ordered from the ISP provider or carrier.

QoS Measurement

Network eManager can measure the IPT QoS parameters listed below:

- Packets (sent/received)
- Delay (msec.)
- Jitter (msec.)
- Loss (%)

Analog CO Line Interface Compatibility

To provide optimum voice quality of IP telephones and Strata Net IP networks on Strata CIX systems, there are some compatibility requirements that must be followed when using analog CO line cards in the system.

- Toshiba highly recommends only using RCOU3A, RCOS3A, BCOCIU1A, BCOCIS1A, REMU2A (two-wire/four-wire), REMU1A (four-wire), RDDU2A, and RGLU3A analog CO line interfaces in IP phone and Strata Net IP applications. These circuit cards provide optimum speech quality for Toshiba IP telephones.
- Do not use RCOU1A, RCOS1A, REMU1A (two-wire), RDDU1A, RGLU1A, or RGLU2A analog CO line interfaces in IP telephone applications. These circuit cards will work, but will cause IP telephone users to experience unacceptable voice quality and echo return loss.
- CIX40 Base cabinet CO lines, GCDU1A, GCDU2A and GCOCIH1A circuit cards provide optimum speech quality for Toshiba IP telephones.

Power Over Ethernet

System Availability: All systems

Toshiba has a strategic relationship with SMC to re-sell the 24-port, 802.3af certified SMC6824MPE PoE switch (shown right) through authorized Toshiba dealers.

Figure 19 SMC6824MPE PoE Switch

The SMC6824MPE PoE switch is sized and priced to fit well with Strata CIX systems to serve applications in which customers want to power their IP telephones over Ethernet, instead of using local power with each IP telephone. This switch works with the IP5000-series and IPT 2000-series telephones.

Notes

- The IP5522-SD, IP5622-SD, and the IP5631-SDL does not support plugging in the power adaptor and being connected to a POE switch at the same time.
- The 5000-series and 2000-series IP telephones require local power for operation unless connected to a LAN that has been equipped with special equipment to provide telephone PoE. The AC adapter supplied with it powers the telephone and is included in the price.

Toshiba SoftIPT IP Telephone

System Availability: All systems

The Toshiba SoftIPT™ is an IP telephony client that works with a wired or wireless (Wi-Fi) tablet, laptop or desktop PC, and PDA. The Toshiba SoftIPT integrates the power of a PC with all of the features available on a IPT3000-series telephone, except background music.

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 2000 / 2003 operating systems (OS). When using an Intel PC, the echo cancellation feature is supported on all these operating systems.

The SoftIPT Version 3.0 has a look similar to 5000-series telephones (shown below) and has some very useful enhancements. Note that version 3.0 software is only for PCs and not PDAs and does not support Windows 2000. SoftIPT on a PC integrates the power of the PC with most of the features available on an IP5000-series telephone (see Figure 20).

The features supported by SoftIPT Version 3.0 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. The Call button operates the same as the digital-series telephone Spkr button. Additionally, there are multiple feature buttons that can be customized from telephone programming mode.

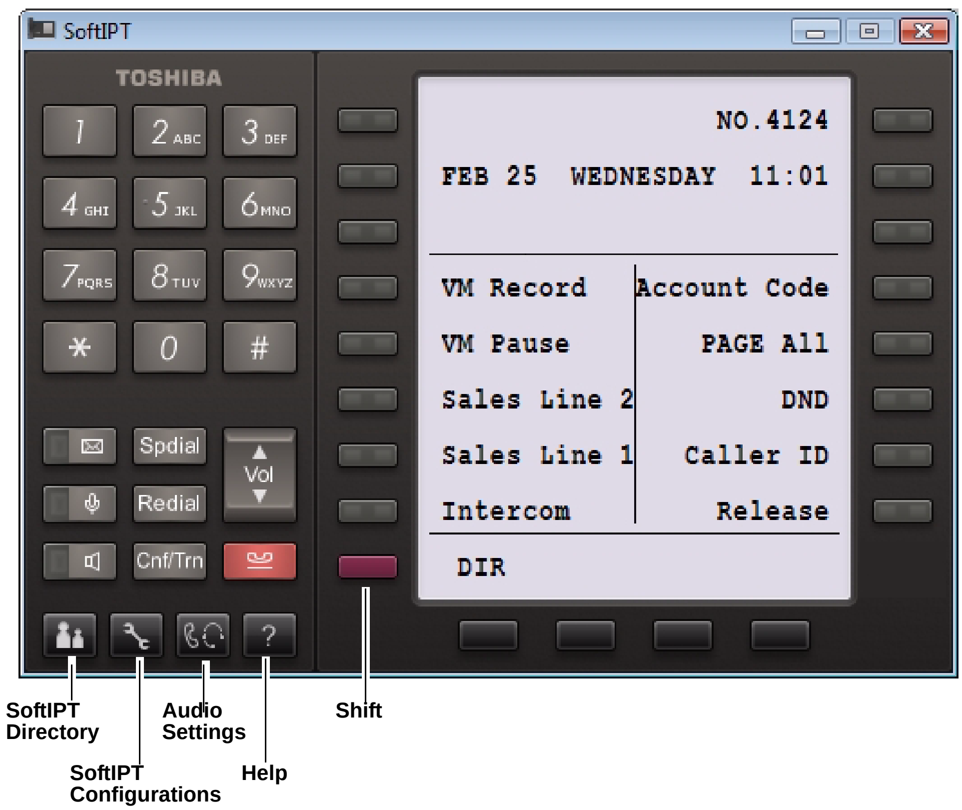

Figure 20 Toshiba SoftIPT 3.0 Sample Screen

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 21).

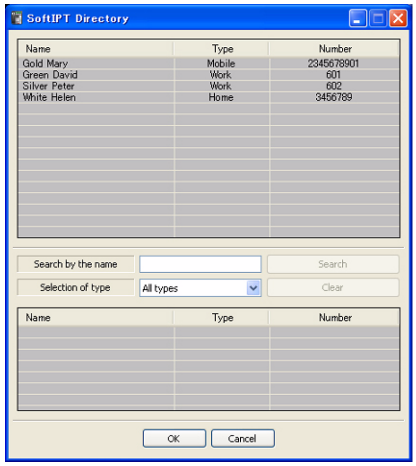

Figure 21 Example of SoftIPT Directory

The SoftIPT can be connected to the Strata CIX system several different ways:

- Intranet – A wired or wireless PC can connect to the office LAN that connects to an IP telephone that connects to a MIPU/GIPU in the Strata CIX.
- 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, to the MIPU/GIPU in the CIX (see Figure 22). (For additional connection examples, refer to the *Strata CIX I&M manual – IPT Chapter.)*
- Wireless – The wireless PCs, such as the Toshiba Pocket PC or Toshiba Tablet PC need a Wi-Fi system that uses the 802.11b standard. The SoftIPT wireless units can operate within 300 feet of an access point (dealer-supplied or use existing).

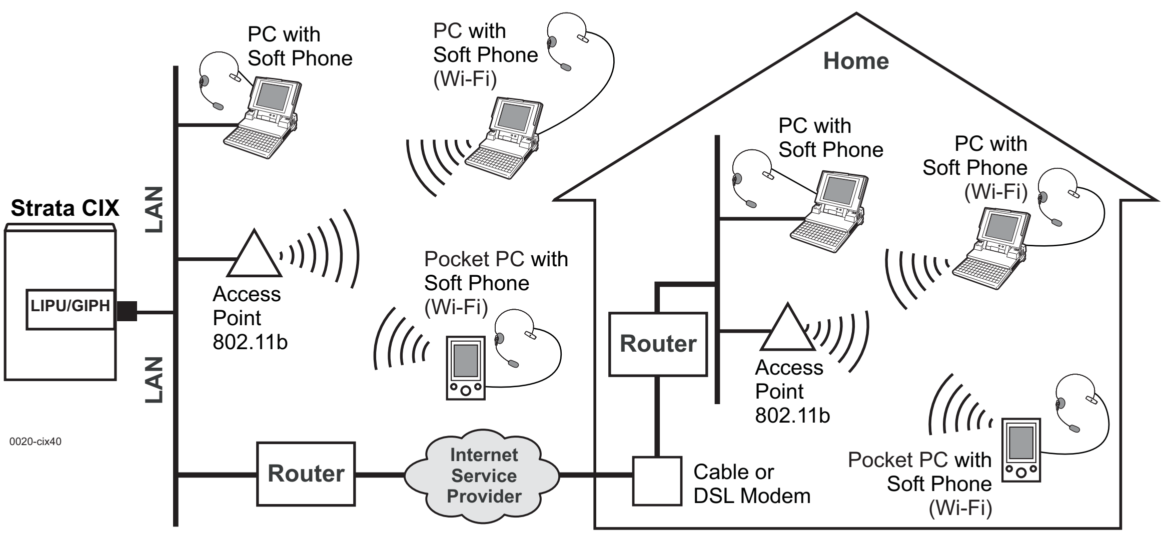

Figure 22 SoftIPT Internet Connection

Licensing

Refer to "Strata CIX Software License Requirements" on page 182.

Private Networking Over IP to the Strata CIX

System Availability: Strata CIX Systems

Strata Net CIX multi-system networking can be implemented over an IP network using Strata CIX systems with MIPU/GIPU IP interface circuit cards. This feature offers the same connection service as ISDN dedicated lines with the Strata Net protocol on the public network. Strata Net IP, does not support modemized data signals, such as modem signal and G3 fax because these signals require very low jitter and low delay on the networks.

The MIPU16 can be configured for up to 16 channels in system programming. The MIPU24 can be configured to 24 channels. The GIPU can be configured to 8 channels. CIX Strata Net private networking over IP can support up to 128 separate nodes.

For bandwidth requirements, refer to the section "Strata Net IP and IPT Bandwidth Requirements" on page 180.

Refer to "Analog CO Line Interface Compatibility" on page 130 if you are planning to mix analog and Strata Net IP circuit cards in the same Strata system.

MIPU/GIPU interface parameters include:

- 100Base-TX: Automatic recognition and switch
- Transmission: TCP/IP, UDP/IP
- Protocol: Based on IP QSIG (ECMA-336), NAT compatible
- Protocol: RTP/RTCP for voice transport
- Voice coding: G.711, G.729A, selectable
- Priority process: Diffserv/IEEE802.1p
- For networking between CIX systems the MIPU must be on a public IP address with recommended VPN

An example of Strata Net IP networking is shown below.

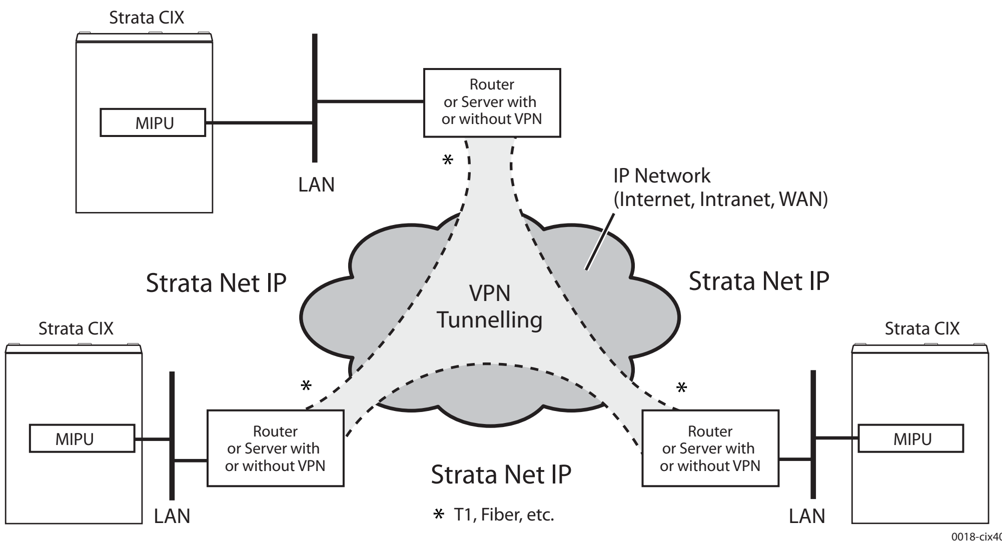

Figure 23 Strata Net IP Example

Private Networking Over IP to the IPedge System

System Availability: Strata CIX systems to IPedge systems

Strata Net CIX multi-system networking can be implemented over an IP network using Strata CIX systems with MIPU / GIPU IP interface circuit cards. The Strata CIX can Strata Net to an IPedge with IPedge Net. Strata Net IP to IPedge Net, does not support modemized data signals, such as modem signal and G3 fax because these signals require very low jitter and low delay on the networks.

The MIPU16 / MIPU24 can be configured for up to 16 / 24 channels in system programming. The GIPU can be configured to 8 channels. IPedge Net private networking over IP can support up to 128 separate nodes.

For bandwidth requirements, refer to the section "Strata Net IP and IPT Bandwidth Requirements" on page 180.

Refer to "Analog CO Line Interface Compatibility" on page 130 if you are planning to mix analog and Strata Net IP circuit cards in the same Strata system.

MIPU / GIPU interface parameters include:

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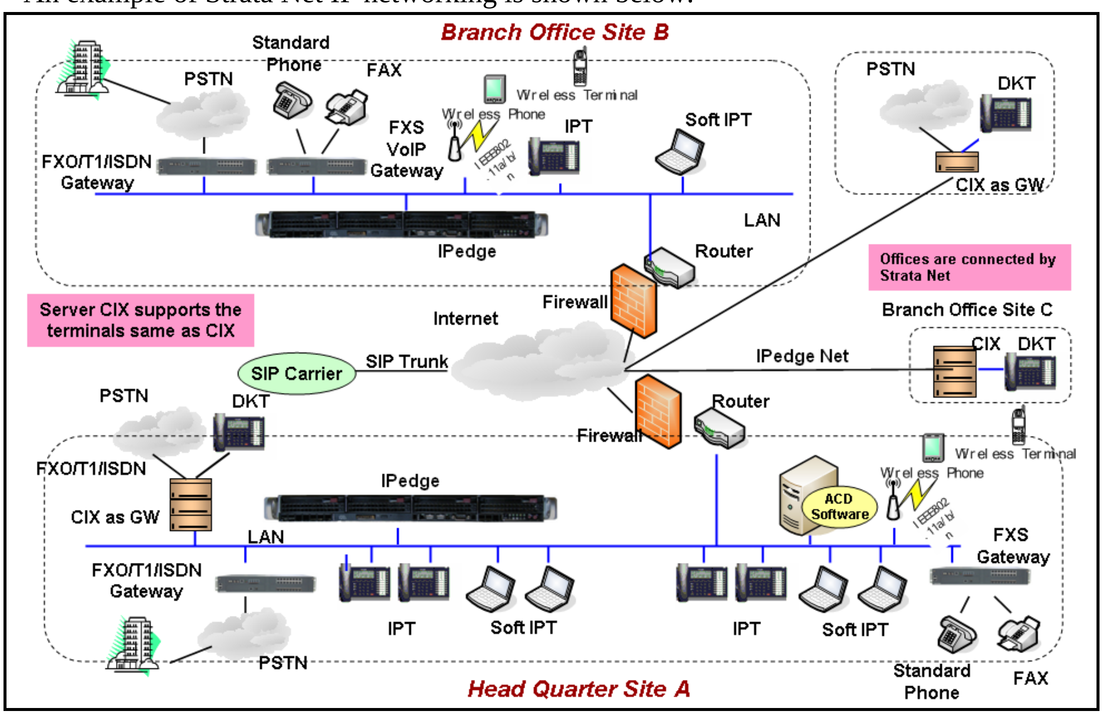
- Protocol: Based on IP QSIG (ECMA-336), NAT compatible
- Protocol: RTP/RTCP for voice transport
- Voice coding: G.711, G.729A, selectable
- Priority process: Diffserv/IEEE802.1p
- For networking between CIX systems the MIPU must be on a public IP address with recommended VPN

An example of Strata Net IP networking is shown below.



Note To network a Strata CIX system to an IP*edge* system via IP*edge* Net the Strata CIX system must use MIPU or GIPU interface cards.

Private Networking Over IP to the VIPedge

System Availability: Strata CIX systems to VIPedge systems

Strata Net CIX multi-system networking can be implemented over an IP network using Strata CIX systems with MIPU / GIPU IP interface circuit cards to VIPedge. This feature offers the same connection service as ISDN dedicated lines with the Strata Net protocol on the public network. Strata Net IP, does not support modemized data signals, such as modem signal and G3 fax because these signals require very low jitter and low delay on the networks.

The MIPU16 can be configured for up to 16 channels in system programming. The MIPU24 can be configured to 24 channels. The GIPU can be configured to 8 channels. CIX Strata Net private networking over IP can support up to 128 separate nodes.

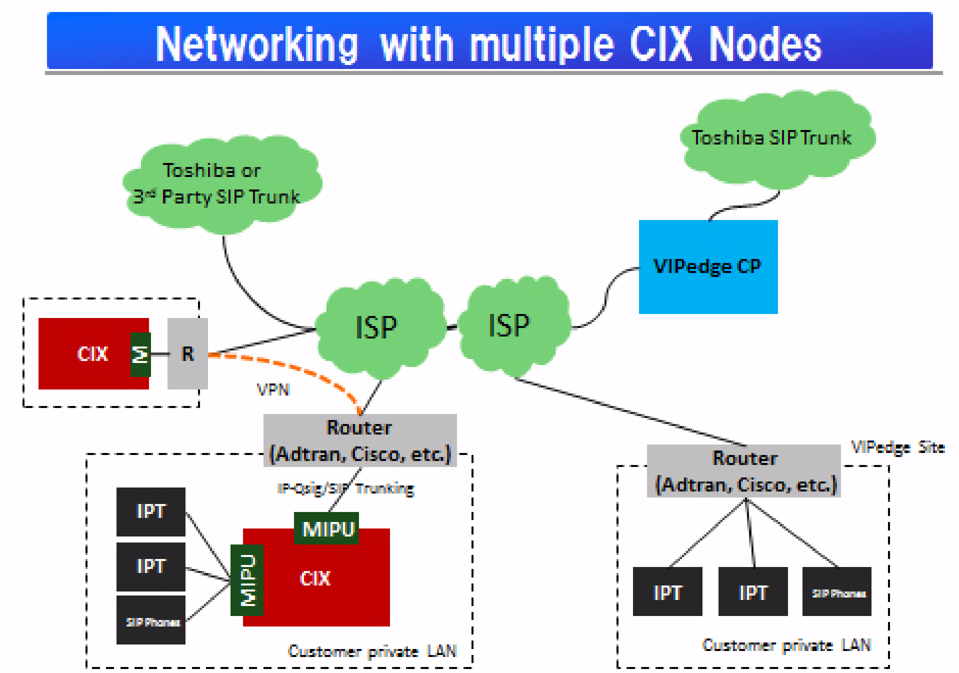
For bandwidth requirements, refer to the section "Strata Net IP and IPT Bandwidth Requirements" on page 180.

Refer to "Analog CO Line Interface Compatibility" on page 130 if you are planning to mix analog and Strata Net IP circuit cards in the same Strata system.

MIPU / GIPU interface parameters include:

- 100Base-TX: Automatic recognition and switch
- Transmission: TCP/IP, UDP/IP
- Protocol: Based on IP QSIG (ECMA-336), NAT compatible
- Protocol: RTP/RTCP for voice transport
- Voice coding: G.711, G.729A, selectable
- Priority process: Diffserv/IEEE802.1p
- For networking between CIX and VIPedge, the MIPU can be on a private IP address without VPN since the VIPedge supports Strata Net/VIPedge Net NAT Traversal.

An example of Strata Net IP to VIPedge networking is shown below.



Note To network a Strata CIX system to a VIP*edge* container via VIP*edge* Net, the Strata CIX system must use MIPU or GIPU interface cards.

SIP Trunking

SIP Trunking allows the CIX to get PRI like services from an Internet Telephony Service Provider using Session Initiation Protocol.

Toshiba implements SIP trunking using the MIPU/GIPU card. The MIPU/GIPU is designed from the ground up as a VoIP card able to support IP stations, Strata Net IP, and SIP trunking. With the SIP Trunking capability of the new MIPU card, companies do not have to purchase different types trunk cards and the bundles of physical wires to host Analog, PRI and BRI trunks. Companies are able to leverage their existing Toshiba CIX R4.x (and later) PBX systems with just the purchase of MIPU/GIPU cards, a software update, and corresponding license. SIP Trunking simplifies IP PBX trunking capability by replacing all the traditional PSTN lines with one SIP Trunking device hosted by SIP Trunking provider on the internet.

Available with the Strata CIX is Toshiba's SIP Trunking service. Toshiba's integrated SIP Trunking service and equipment enable businesses to choose a single-vendor business telephone service and equipment solution.

MIPU interface parameters include:

- CIX Hardware: CIX40, CIX100, CIX200, CIX670, and CIX1200
- CIX Software
- MIPU
- Network eManager
- Service provider: Contact Toshiba Sales Applications Desk
- Soft Switch: Contact Toshiba Sales Applications Desk
- License: LIC-CIX-SIPT-CH

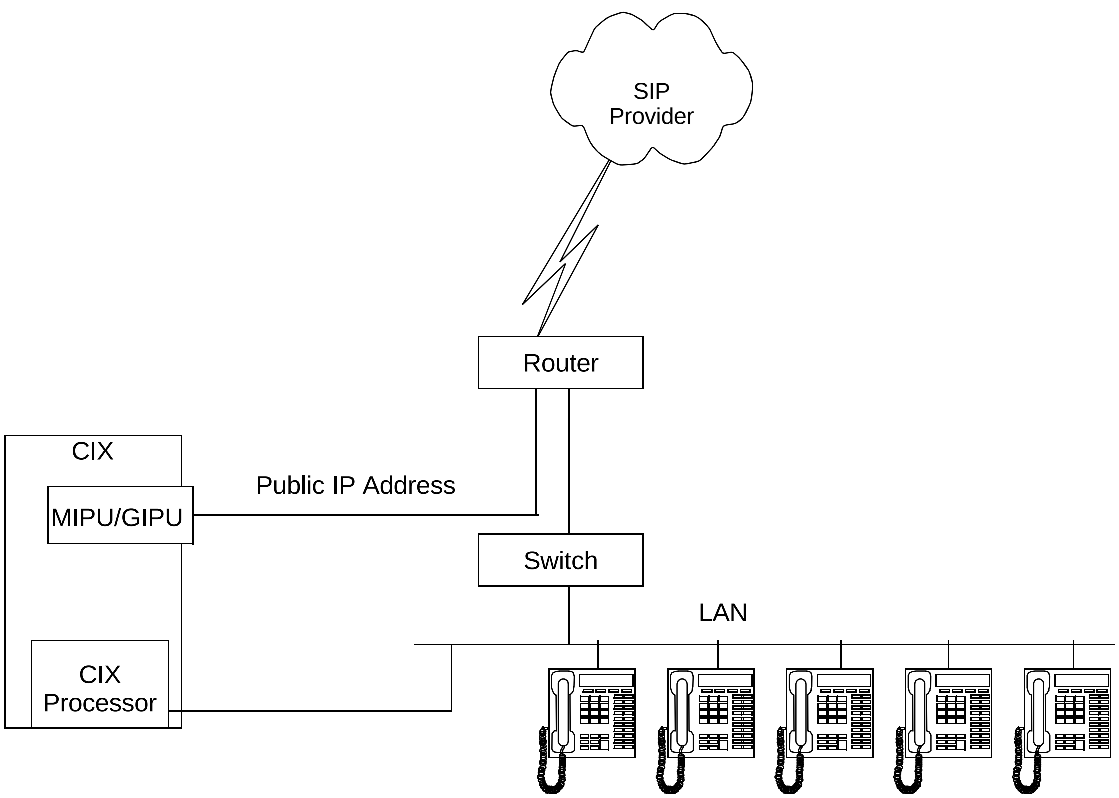


Figure 24 SIP Trunking Example

Integrated Services Digital Network (ISDN)

System Availability: All systems

ISDN is a set of integrated telecommunications services, available over the public telecommunications networks. ISDN makes it possible to send, receive and modify information using telephone lines in ways that were not previously possible, such as:

- Dynamic use of individual or groups of standard (POTS), DID, Tie, FX, WATS, 800 lines on an as-needed basis
- Much faster call setup and data transfer up to 128Kbps
- Multi-purpose line use, including sharing lines for voice, data, fax, and video
- DID functionality based on the number dialed; without needing to reserve a block of numbers
- 2-B channel transfer: Allow two external PRI line calls connected to a station conference to be released from the CIX and reconnected by the PSTN when the station drops out of the conference. Requires special ordering from the ISDN provider. Requires National ISDN protocol.

ISDN service comes in two forms:

- **Primary Rate Interface (PRI)** supports simultaneous voice or data connections (eight, 16 or 23). PRI is similar to digital T1 service and uses two pairs of wires from your phone company. The RPTU circuit card supports PRI on the Strata CIX100, CIX200, CIX670, and CIX1200.

Least Cost Routing (LCR)

System Availability: All systems

Least Cost Routing chooses the most appropriate route over which to connect an outgoing call based on the following:

- Dialed Digits
- Time of Day
- Type of Day (Business, Weekend, Holiday)
- LCR group of the caller

The combination of routing tables, indices, route definitions and time-of-day qualifiers can produce up to 75 million combinations. Routing changes automatically for each type of day, according to the time of day. This schedule is independent of the Day/Night mode schedule which applies to ringing and CO assignments.

Line Buttons

Telephone buttons that are used for making and receiving outside calls are referred to as `Line` (or `CO Line`) buttons. (For information on various [DN] buttons, refer to "Directory Numbers" on page 117.) The Strata CIX supports the following types of line buttons:

CO Line Buttons

System Availability: All systems

Smaller systems have traditionally provided the direct appearance of the CO lines on the telephones where maximum visibility of the line status, flexible ringing assignments, and informal call transfers associated with key telephone systems may be implemented.

Pooled CO Line Button

System Availability: All systems

Pooled Line Group buttons enable a group of CO lines to "appear" under one button. Pooled and single appearing line buttons are designed for use with loop and ground start lines, not Tie, DID, DNIS or ANI lines.

Group CO Line Button

System Availability: All systems

Group CO line buttons are like individual CO line buttons except these buttons represent all the lines for a particular ISDN Channel Group. This enables ISDN 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.

Live System Programming

System Availability: All systems

Programming the Strata CIX 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.

Lost Call Treatment

System Availability: All systems

Lost Call Treatment provides the CIX a mechanism for terminating calls that cannot be terminated with the usual calling patterns. One scenario would be a call that is recalled to a station, the station user is no longer there to answer the recall and no forwarding pattern is programmed. The call will ring at the recalled station until the Lost Call Timer has expired after which the system will direct the call to the Lost Call Destination.

Message Waiting

System Availability: All systems

Any station and most voice mail devices can turn on a message waiting indicator for a designated digital or standard telephone station.

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 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. (See Table 43 on page 177 for details of each type of stutter dial tone.) With Release 1.3 and higher, 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 normal dial tone when you go off hook.

Microphone (External Unit)

System Availability: All systems

An external microphone (RFDM) can be connected to the DKT2020-FDSP digital telephone enhancing “full-duplex” operation by virtue of the “superdirectional” characteristic of the microphone. When this option is on, the internal microphone is disabled on all but Voice First Handsfree Answerback calls and OCA calls. The external microphone is powered by the DKT and does not need to be turned off when not in use.

Music-on-hold

System Availability: All systems

Music-on-hold can be derived from a customer-supplied radio, tape player, tuner, CD player or other device to provide music or announcements to parties on hold on CO lines or on [DNs]. With the Strata CIX, you can have up to 15 MOH/BGM sources. Each CO line group and each DID/DNIS number may be assigned a specific MOH source. Stations and network Tie lines can also share a unique MOH source.

Multiple Call/Delayed Ringing

System Availability: All systems

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. Multiple calls to digital telephone is supported; however, delayed ring to digital telephones is not supported as of this printing.

Off-Hook Call Announce (OCA)

System Availability: All systems

Station users may announce a call when the station they call is busy talking with the handset off-hook. The announcement is only audible to the telephone user receiving the OCA call, not to the other party in the original conversation.

Two different methods of operation are provided – handset or speaker OCA. With handset OCA, the OCA caller's announcement comes in on the telephone handset. With speaker OCA, the announcement comes in on the speaker. Handset or speaker OCA can be set individually for each telephone that must receive OCA calls. Speaker OCA on digital DPs require a BVSU option for DKT3200-series telephones or a DOCA-1A option for DP5000 telephones. Handset OCA has no optional hardware requirement. Speaker OCA on IP5000-series telephones requires an MIPU/GIPU installed in the Strata CIX.

Note Each IP5000-series telephone enabled with Speaker OCA requires two IP channels on the same IP interface card but only one IP endpoint license. The OCA IP channel can be shared by all IP5000-series telephones and is only used while the OCA call is active.

When a busy telephone receives a handset OCA call, replying confidentially to the OCA caller can be accomplished by pressing the `Msg` button (toggle) to place the original call on hold or holding down the `Mic` button to make a short reply. In either case, the original caller will not hear the reply to the OCA caller. To reply to a speaker OCA call, the user covers the handset mouth piece and talks back through the telephone microphone.

Any type of telephone can be enabled to originate OCA when calling a busy digital telephone. The feature is activated automatically (optional setting) or manually (Call Completion code – digit 5). Stations receiving OCA must be proprietary digital telephones assigned with OCA-receiving capability in system programming. Standard single-line telephones cannot receive OCA.

Any type of station can make an OCA call, as long as the station has this option enabled in system programming. OCA to DND telephones is allowed only if DND Override is allowed on the called and calling telephones.

Note The CIX40 supports Headset OCA on digital and IP telephones, but only supports Speaker OCA on IP5000-series telephones.

Off-Premise Stations

System Availability: All systems

Off-premise stations are supported using either standard analog telephones or Toshiba digital telephones. This can accommodate both individual telephones and branch office connections. Off-site standard analog telephones can be part of the system, having access to many of the features offered by the Strata CIX. Each off-site standard telephone station requires a special OPX line from the CO and the RS48 subassembly on the CIX standard telephone interface card. This enables groups of remote workers to use Toshiba digital telephones to have seamless access to the main location's telephone system and voice mail system. Off-premise stations require secondary lightening protectors.

Remote employees have transparent access to all the same capabilities as if they were locally connected to the Strata CIX system. They have can the same ability transmitting voice traffic and digital telephone signaling over the customer's existing Local Area Network (LAN) Wide Area Network (WAN) private IP packet network or the public Internet.

Note CIX40 standard telephone ports do not support RS48 which limits OPS telephones to OLI3B (600 ohm) loops.

Override

System Availability: All systems

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 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

System Availability: All systems

The Strata CIX has a paging interface that supports a Toshiba External Speaker (BESCB) or a customer-supplied amplifiers and speakers for Paging, Night Ringing over Page, and BGM applications. Users can access any of the Paging options by dialing access codes or by using a programmed One Touch button.

Telephone Group Paging

Paging is activated from an extension by specifying a Paging Group. Paging can be broadcast through digital telephone speakers and external paging devices simultaneously. The system supports up to 16 or 24 telephone page groups with up to 120 (CIX200), 72 (CIX100), 120 (CIX670) or 120 (CIX1200) telephones per group. Standard telephones cannot be members of a page group.

External Speaker Page Zones

The Strata CIX supports eight different paging zones for external speakers. Users can access zones by dialing an access code plus the zone or pressing a One Touch button. The zones are composed of customer-supplied speaker(s) and amplifier(s). One BIOU supports up to four page zone interfaces, a second BIOU is required for 5~8 zones.

Emergency Page

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

Each of the 16 or 24 Paging Groups supports up to 32 devices. Emergency Page groups follow the regular Group Paging. The list for Emergency All Call Paging is a separately defined list from regular paging. An emergency page may be answered in the same manner as a regular page.

Night Ringing Over Selected Page Zones

Lines can be programmed to night ring over eight selected Page zones via customer-supplied paging equipment. Up to two BIOU circuit cards can be installed to connect external paging or night ringing equipment. Each BIOU supports up to four page zones.

Power Failure Protection

System Availability: All systems

The Strata CIX has important optional capabilities that keep the system operating when commercial AC power is interrupted.

Power Failure Transfer

Some Strata CIX systems can immediately switch loop start analog CO lines directly to dedicated standard telephones (customer-provided 2500- or 500-type) for incoming and outgoing calls in the case of a commercial AC power failure. The transfer is automatic with no manual transfer procedure required. During normal operation with AC power, the Power Failure telephones function with all Strata CIX features available to a normal standard telephone. This feature requires a Toshiba external unit called the Power Failure Transfer Unit (LFPU1A) for the CIX200. The CIX40 provides one built-in PFCT relay for CO Line 1 and the base unit standard telephone circuit. Strata CIX100, CIX670 and CIX1200 systems require third-party devices to accomplish power failure transfer directly to CO lines.

Reserve Power Battery Backup

An Uninterruptible Power Supply (UPS) is required for power backup on a CIX200. The UPS is similar to the ones used for Computer systems and Networking equipment. Two or four 12-volt gel-cell, maintenance-free batteries can be connected to the CIX40, CIX100, CIX670, and CIX1200 system power supplies for system battery backup (80 amps./hours max.). The CIX670 and CIX1200 system power supply is standard-equipped with a battery charger and the batteries continuously trickle charge to capacity while electrical power is present. The CIX100 power supplies must be equipped with the optional ABCS battery charger to charge reserve power batteries.

If the AC power fails, the Strata system automatically switches over to battery power without any interruption in operation. Calls in progress are not interrupted. Battery operation duration depends upon the condition and ampere hour rating of the batteries and the system load. However, the minimum battery operation time would be several hours. Connection of reserve power batteries must be made when commercial AC power is available.

Privacy

System Availability: All systems

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.

Remote Update

The remote program update is administered using the Network eManager Administrator tool to update the Strata CIX software remotely over a TCP/IP or Modem connection. The SD or SM card must be installed on the Strata CIX processor to allow remote updates.

Repeat Last Number Dialed

System Availability: All systems

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. Digital key telephones have a fixed `Redial` button for automatic redialing of the last number dialed.

Ringing

System Availability: All systems

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.

Ringing Cadence

With Release 1.3 and higher, you can choose between two different ringing cadences for standard telephone circuits in programming. This option is necessary to accommodate some voice mail and/or auto attendant devices. This is a system wide option that allows external calls to ring with the traditional one sec. on./three sec. off cadence or optionally with a faster cadence of 0.4 sec. on./.2 sec. off. This option does not apply to digital telephones. If ringing cadence is used, Centrex ring repeat must not be used.

Delayed Ringing

See "Delayed Ringing" on page 114.

Distinctive Ringing

See "Distinctive Ringing" on page 118.

Speed Dial

System Availability: All systems

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. Strata CIX offers 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.

CIX Network eManager 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 CIX Network eManager. This name appears in the DP5000-series 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 a `Cnf\Trn`, [DN], CO line, etc., can be set on One Touch buttons to allow multiple button presses to be stored under one button. This enables tandem line connections and other call setup sequences to be dialed easily by pressing one button.

Station Hunting

System Availability: All systems

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

Serial Hunting

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

Circular Hunting

In this type of hunt group, calls hunt [DNs] in a series in a specific order; however, the series forms 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 in such a way so as to distribute the calls 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 busy tone. The caller can then dial a digit to initiate Camp On-Busy to the called, busy [DN] or the Pilot [DN], if used. As an option, for each hunt group that uses a Pilot [DN], calls will automatically camp on to the called Pilot [DN].

With the Automatic Camp On option, the caller does not get busy tone, instead the caller receives confirmation tone followed by 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 be connected first when the destination becomes available. If the parties have the same QPL, the longest waiting call will be connected first.

Station Message Detail Recording (SMDR)

System Availability: All systems

For each incoming, outgoing or tandem call, the Strata CIX 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 Strata CIX 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 an optional BSIS or network interface circuit card and a connected Call Accounting system.

For Network SMDR and Centralized SMDR for Systems in a Strata Net Network, refer to page 151.

Call Manager

The Call Manager (CM) is a powerful unified communications tool that provides the following and much more:

- Desktop call control from your PC
- Customized call handling
- Outbound dialing from any application
- CRM integration with screen pops
- Presence Viewer to display the status of other users
- Instant Messaging / Chat

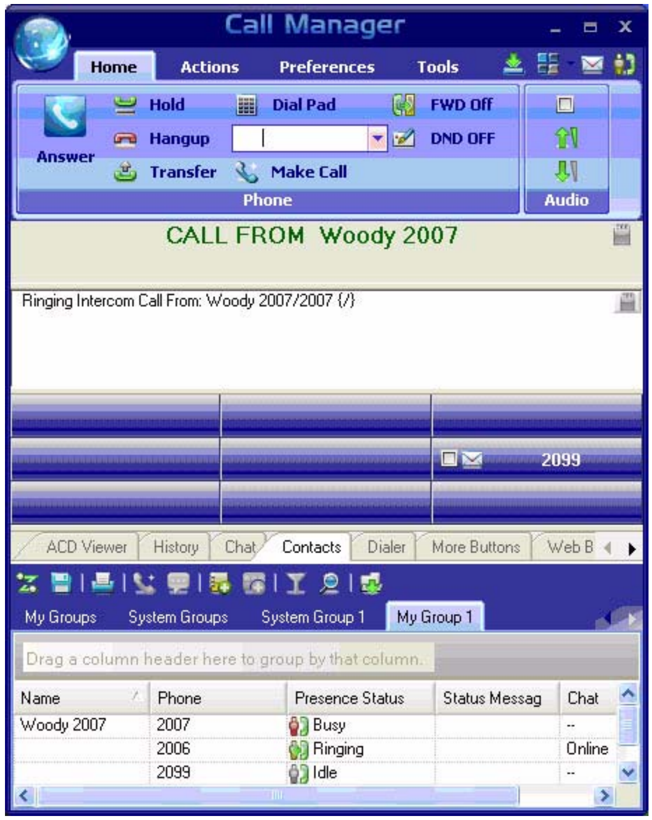


Figure 25 Call Manager Main Page

The Call Manager is built from the ground up using .Net Framework 3.5 and modeled against the Microsoft Fluent Ribbon design. Combined with this powerful GUI, Call Manager is easy to use and manage.

The Call Manager buttons are created equal so that any button can be programmed as any other button. Call Manager can use any of the buttons including other buttons from the Buttons companion application for ACD functionality. Each Buttons companion application can hold eight banks of 64 buttons per bank. You can have eight instances of the More Buttons companion applications to ensure that you have enough buttons in the Call Manager. All buttons can be programmed by the user for the following: Feature Code, Speed Dial, System/PBS Commands, User Defined Actions, ACD keys, DSS Extension Key, Web Key, and Run Programs.

Call Manager provides a powerful open architecture that allows companion applications to add features and functions. These companion applications are provided as tabs on the Main Screen.

The companion applications includes ACD Viewer, Buttons, History, Contacts, Chat, Dialer, and Web Browser.

The companion applications are docked in the lower panel of the SCM screen and are accessed using the tabs. Users may also undock the companion applications and place them anywhere on the desktop. Refer to the Call Manager User Guide for instructions on using these applications.

Microsoft® Lync® Integration

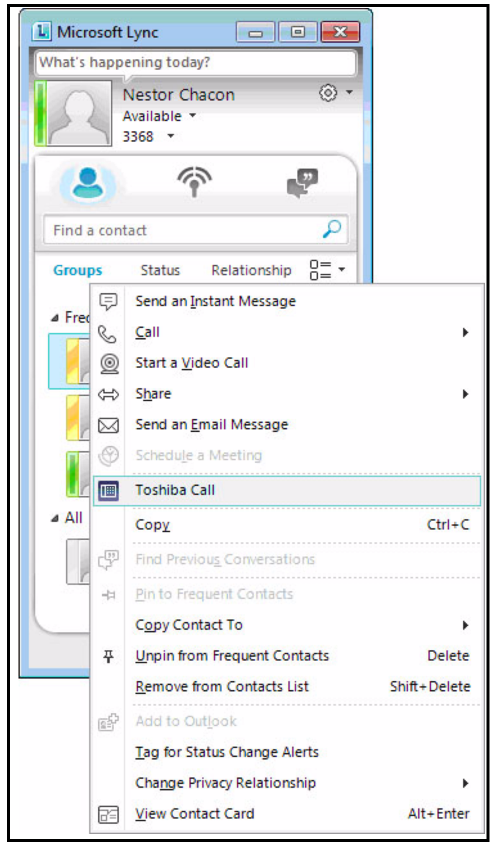
Toshiba has a plug-in that is installed on a customer's PC to integrate with the Microsoft Lync client. This eliminates the complex server configuration that is required for server integration. This integration enables customers who adopt Lync as the Instant Messaging/Presence application to integrate with the Strata CIX system telephone features.

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 Plug-in 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 Plug-in 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 Plug-in main window.

- Optional built-in softphone

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

  Telephones to be used with Toshiba Plug-in 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 Plug-in or popup notification may not work properly.

Strata Net Multi-system Networking

System Availability: All systems

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

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.

Strata CIX systems are interconnected with DS1 (T1) circuits to provide ISDN-type interconnectivity.

Strata Net IP also provides full Strata Net 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 Strata Net numbering plan. Up to four nodes connected in tandem can give satisfactory performance with regard to latency. 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.

Refer to “Private Networking Over IP to the Strata CIX” on page 133, “Private Networking Over IP to the IPedge System” on page 134 and “Private Networking Over IP to the VIPedge” on page 135 for more details.

Coordinated Numbering Plan

System Availability: All systems

Strata Net can 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.

Strata Net Basic Call Control

System Availability: All systems

The Strata CIX conforms to the QSIG standard for Basic Call Control. This is the basis for all Strata Net connectivity and interoperability with PBXs from other manufacturers. Basic Call Control provides for connection, dialing, identification of calling and called parties’ names and numbers and message waiting indications among other features.

Important! *Toshiba does not guarantee interoperability with other manufacturer’s products: only conformance to the standard.*

Alternate Routing

System Availability: All systems

Each Strata CIX can be programmed for thousands of routing patterns for Strata Net alone. This allows the creation of networks in which calls can be automatically re-routed around network disruptions. Centralized facilities and features can continue to work and users will be unaware of problems while they are being repaired.

Alternate Routing also permits Toll Bypass in which Strata Net can be used to deliver a public call from a point in the network where toll charges are minimized. Such a scheme is known as “Hop Off” for the ability of the private network to determine the point at which the call will hop off to the public network.

Centralized Attendant

System Availability: All systems

One attendant can serve an entire Strata Net. Station users only need to dial “0” to reach the centralized attendant regardless of the node in which they reside. The attendant can reach any station in the network using its Network Directory Number. Trunks attached to any network node can be programmed to terminate to the centralized attendant and their source and calling party information will be delivered to the attendant’s display. The BLF appearances of all stations from all nodes can appear on the centralized Attendant Console.

Telephone DSS Buttons

System Availability: All systems

Telephone DSS buttons can appear across the Strata Net network. This enables a user’s DSS button to function in all nodes in a CIX network. The DSS function works within or across a network.

Centralized Voice Mail

System Availability: All systems

A voice mail system attached to any Strata Net network node can serve users throughout the enterprise. Unanswered calls will be forwarded to the voice mail, the source and calling conditions identified and the appropriate voice mailbox greeting will be played. The voice mail system can control message waiting indications throughout the network as messages are left and retrieved. A single network can even support multiple centralized voice mail systems with each station being programmed for the appropriate system.

Network SMDR

System Availability: All systems

Distributed Network SMDR for Systems in a Strata Net Network

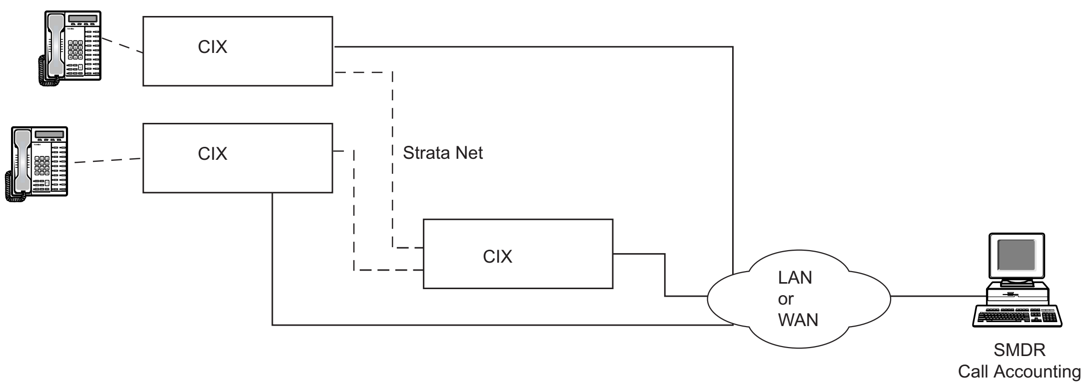
An external Strata Net call will generate a call record at the terminating node for that call. Transit nodes will not generate records. The records can be stored in customer-supplied external buffers at each node. Third-party polling call accounting software can gather and organize the data from multiple nodes. Local buffering provides survivability in the event of network disruption.

Centralized SMDR for Systems in a Strata Net Network

System Availability: All systems

The SMDR information from each Strata CIX system, connected over a TCP/IP Strata Net network via the CIX processor LAN interface jack can be sent to a Centralized SMDR system. Separate SMDR equipment is not required at each node.

Dealer supplied third-party Call Accounting software can be located on a single Call Accounting Server that can receive SMDR call data from each Strata CIX node. Users having dealer supplied Call Accounting client software can retrieve reports from the server from any location.



SMDR RS-232 interface (BSIS) continues to remain an SMDR interface option; so customers that use Call Accounting software that only supports RS-232 interface can continue to use it with Strata CIX Release 4.1.

Survivability

The Toshiba Strata CIX system achieves survivability by combining technologies from Strata Net multi-route programming, IP Mobility and SIP Trunking. By combining these technologies, the Toshiba IP telephones and the SoftIPT phone are able to survive by automatically connecting to a secondary Strata CIX system and still maintain station-to-station calling, inter-node calling, outbound dialing, and access to voicemail resources. Incoming call routing can be achieved by having the service provider manually re-route all the incoming calls to the secondary Strata CIX system. Optionally, using the Audiocodes gateway, incoming call routing can now be achieved without many manual changes or involvement from the telephone service provider.

Note There will not be any special survivability license required to achieve Strata CIX survivability.

Because the IPT and SoftIPT phones will be re-registering to the secondary backup Strata CIX system, there must be enough Strata CIX IP licenses and Strata CIX hardware resources to support the surviving IP phones on the secondary Strata CIX system. For more details on this feature, refer to the Strata CIX Information Bulletin: “Strata CIX System Survivability for IP Telephones.”

System Fault Finding and Diagnostics

System Availability: All systems

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

Alarm Indication of System Faults

Visual Alarms are presented to CIX Network eManager.

Secure Digital Card

This is a flash memory card that is commercially available in retail stores. The Strata CIX uses the Secure Digital card to store all error, trace logs and a backup copy of the system operating software and the customer database. Toshiba ships 512MD Secure Digital cards with each CIX40, CIX100, CIX670, and CIX1200 system package. The Secure Digital card is inserted into a socket on the CIX processor. The CIX processor creates directories and files onto the card for maintenance functions. Using Network eManager enables moving, copying, or deleting these files without having to remove the Secure Digital card from the CIX processor. With Network eManager, this works locally or remotely. Secure Digital files can also be managed by removing the Secure Digital card from the CIX processor and inserting it into a PC Secure Digital card read/write adapter. The CIX200 processor, the LCTU1A, uses a SmartMedia flash memory card.

The Strata CIX Release 3.10 and below processors and all CIX200 LCTU1A processors use the SmartMedia card.

Fault Detection and Error Logs

The Strata CIX detects and logs abnormalities that it encounters during operation. All error and trace logs are stored on the Secure Digital or SmartMedia card on the system processor and are monitored by Network eManager. Examples are trunk failure detection and auto busy-out, digital telephone port failure detection and auto busy-out plus error log, Expansion Cabinet power supply failure alarm and 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.

System Trace

The system records telephone key strokes and other high level events and presents the data in a format understandable and useful to the field technician for troubleshooting purposes. The system also records more detailed data useful to a software support engineer.

Manual Test

The maintenance technician can perform certain test functions using Network eManager to determine proper operation of the system.

Backup and Restore

The customer database can be backed up and restored using the SD flash memory card. The customer database is a file that can be stored on a SD card, transferred to the PC hard drive, e-mailed, etc. The backup and restore functions can be performed locally or remotely.

Maintenance and Administration

The Network eManager terminal can be connected directly to the Strata CIX or via the customer's LAN as well as remotely over the Internet and via modem over the public network. The Strata CIX processor comes standard with a network interface port and a built-in modem.

Software Upgrade

The Strata CIX operating software can be upgraded using the Secure Digital or SmartMedia card or by downloading it from a remote location. The operating software is a file that can be stored on the Secure Digital/SmartMedia card, transferred to the PC hard drive, e-mailed, etc.

Transfer

System Availability: All systems

Transfer is the ability to redirect a connected call to new destination. The Strata CIX 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 to which 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 to Voice Mail

See “Direct Transfer to Voice Mailbox” on page 160.

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.

Tenant Services

System Availability: All systems

This feature enables one CIX 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

System Availability: All systems

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 CIX system programming commands. Approximately five days of Traffic Measurement reports can be stored on the system SD flash memory card.

The Traffic Measurement is system based. The measurement cannot be set to report per tenant.

Traffic Reports

New traffic reports include outgoing and incoming trunk group usage, “all circuits busy” reporting DTMF and conference circuits. The reports are stored on the processor’s Secure Digital (SD) or Smart Media flash memory card locally, and reports can also be sent to a remote device over a TCP/IP or RS-232 connection. 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 DTMF and Conference circuit usage. Reports can measure traffic in Centum Call Seconds (CCS) or Erlangs. All circuits busy and Abandoned calls are also reported.

Unified Communications

Most businesses have several ways to communicate with their customers, suppliers, and each other by using office telephones, cell phones, voice mail, email, video conferencing, instant messaging, and more. In a non-unified approach, these various forms of communication work independently of each other, sometimes causing you to try multiple methods before reaching someone.

Toshiba’s Unified Communications (UC) suite includes the structure and intelligence to enable these various forms of communication to work together, so information reaches recipients quicker and through the most appropriate medium. This type of Communications-Enabled Business Process (CEBP) integration requires business applications and information databases to have imbedded communications capabilities that become part of the business application.

With the IPedge/VIPedge Application Servers, Toshiba Unified Communications capabilities integrate with the Strata CIX IP PBX voice features to form one comprehensive, integrated communication solution. These solutions work through a seamless user interface that is independent of the device you use and works from any location.

Uniform Call Distribution

System Availability: All systems

Strata CIX systems have built-in Uniform Call Distribution (UCD) functionality, which 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 to find the next available agent. Callers in queue can receive music and announcements imbedded in one of the systems music-on-hold sources, and each UCD group can share or have a separate music source. 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.

The built-in UCD standard feature is ideal for basic call processing applications not requiring the more robust optional ACD and reporting capabilities available with Strata CIX systems.

Interaction with Other Features

Call Forward, System Call Forward

UCD calls do not forward per the station's call forward assignments. Calls routed directly to the agent's station, not through the UCD pilot, will forward when System Call Forward is set.

Do Not Disturb

If DND is activated while the call is ringing the agent, the ringing tone is stopped. However, queuing or overflow is not executed. On the other hand, if DND is activated on the overflow destination, the call does not overflow.

Dialed Number Identification Service

If the overflow destination is voice mail and the overflowed call is a DNIS call and associated with VM-ID, VM-ID sent to Voice Mail follows DNIS specifications.

Station Hunting

The overflow destination can be a station hunting pilot or member. The system selects an idle station from the Station Hunting group and terminates a call. If no idle station is found in the station hunting group, the system camps on the call according to the station hunting specifications.

Offhook Call Announce (OCA), Handset Offhook Call Announce

Even if automatic OCA is set at the originator terminal and the overflow destination is set to allow OCA, the overflowed call does not use OCA.

Private Networking

The overflow destination can be a station in another node. However, if the overflow destination is an UCD pilot in another node and all agents are in logout, DND or Make Busy state, the call does not overflow and stays in the queue.

Multiple Calling

If a UCD pilot is a member of a Multiple Calling Group (MCG), the call does not terminate at the UCD agent. If all members are UCD pilots in a MCG, the caller hears ROT.

Door Phone

If a door phone call comes to UCD, it terminates to an UCD agent according to the hunting rule even if the agent is not logged in.

Class of Service

Call is queued to UCD, even if caller's Class of Service does not allow the camp on.

Lost Call Treatment

If a Lost call comes to UCD, it terminates to an UCD agent according to the hunting rule even if the agent is not logged in.

Intercept

If UCD is assigned as the intercept position, the call terminates to UCD agent according to the hunting rule even if the agent is not logged in.

Phantom DN

If the phantom DN is assigned to the UCD group, Login/Logout and DND are controlled by the owner of the phantom DN. When the owner logs in, the call can be terminated at phantom DN. If this DN appears on multiple phones, it will ring all phones. However, if no owner is assigned to the phantom DN, the call cannot be terminated at the phantom DN.

Music on Hold

If a UCD call is camped on to the overflow destination, the caller hears MOH programmed for the overflow destination. When a UCD call is in the UCD queue, the caller hears MOH programmed for the UCD group.

User Programming Mode

System Availability: All systems

Digital telephone users can use the programming mode for customizing their Toshiba telephones without the aid of an Administrator or Service Technician. The User Programming mode is accessed with a `Program` button assigned to a flexible button or through an access code. User Programming enables users to customize these features:

- **Flexible Buttons** – Toshiba telephones have 10, 14, or 20 flexible buttons to which the user can assign any one of approximately 50 different features (DND, ACB, Release, etc.). Once assigned to a button, the feature is accessed by pressing that button. Some buttons have parameters that users can set. These include:
  - **Call Forward** – Users can set the Call Forward (CF) destination and CF-No Answer Timer for the CF buttons.
  - **One Touch** – Users can set speed dial and custom feature access code sequences for One Touch buttons.
  - **Background Music** – Users can select the music source (up to 15 sources) that will play on their telephone's speaker when they activate the `BGM` button.
  - **Ring tones** – `Line` and [DN] button ringing tones can be changed to one of four different tones. These tones apply to direct or transferred incoming calls from outside lines, not internal calls.

Note Directory number and external line buttons cannot be added or deleted, but their ring tones can be individually changed.

In addition to the Programming Mode, an advanced programming function enables administrators to individually turn On/Off the telephone's beep tone, handset call waiting tone, and microphone background noise cancellation option.

VLAN Tagging

System Availability: All systems

The MIPU and IP5000-series phones support 802.1Q Virtual Local Area Network (VLAN) technologies. For sites with LANs that have 1000's of IP devices, VLANs can be used to separate the network virtually rather than physically, to prevent the broadcast and other traffic from one virtual LAN (typically a data lan) from impairing the performance of equipment on another virtual LAN (for example a VoIP LAN) even though the devices are plugged into the same physical network. VLAN for the IPT phone and the phone's data port can each be programmed manually in the phone, or remotely through Network eManager. And within or without VLANs, 802.1P and Diffserv can be used to provide Quality of Service for voice by allowing voice packets to be prioritized over data packets. Note that when using 802.1Q or 802.1P it is important to ensure that all the network of the ethernet switches and routers are capable of supporting this.

Reasons why a company might want VLANs:

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

Voice or Tone Signaling

System Availability: All systems

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. The pitch and sound of internal tone signaling is always the same. Incoming CO line ringing is uniquely different, with up to four optional ring tone sounds.

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.

Voice Mail Integration

System Availability: All systems

The CIX670 and CIX1200 supports Dual-tone Multi-frequency (DTMF) integration, Simplified Message Desk Interface (SMDI) integration, and Toshiba Proprietary integration.

DTMF Integration

DTMF integration uses DTMF strings to pass and interpret information between the Strata CIX and a voice mail system. DTMF integration can be used with any compatible voice mail system. It does not require optional hardware interface.

Simplified Message Desk Interface (SMDI)

SMDI is an industry standard method of integrating a telephone system with voice mail or other peripheral systems. This interconnection is made via an RS-232 data connection. SMDI requires the BSIS interface in the Strata CIX. SMDI integration can be used with any compatible voice mail system.

Toshiba Proprietary Integration

Toshiba proprietary integration provides the highest functionality between the Strata CIX and a Toshiba voice processing system. Toshiba proprietary integration requires the BSIS interface for control signaling between Stratagy and Strata CIX. Toshiba proprietary integration is required to use Stratagy voice processing system's support the features of Call Record to Voice Mail and Voice Mail Soft Keys.

Call Record to Voice Mail

While on an active call, a station user can record the conversation and store it in a Stratagy ES voice mailbox by pressing Record on the digital telephone. To end the recording, they can press Record again. Station users can also stop and start recording by pressing PS/RES.

Users can replay recorded messages by calling the voice mailbox that has the stored recording and play it back as any other message. The "record to" mailbox can be any mailbox number and can be accessed automatically when Record is pressed or dialed after Record is pressed.

Recording to Voice Mail (VM) is available on two-party and multi-party conference calls.

Voice Mail Soft Keys (LVMU / GVPH only)

Voice Mail Soft Keys provide LCD telephone users with an active set of Soft Keys that prompt the user with available commands to play Voice Mail messages and to manage their mail boxes.

The LCD shows the number of New and Saved messages in the user's mailbox.

The number of New/Saved messages displays on the LCD when the telephone is idle and has at least one new message.

Soft Keys requires the Toshiba Proprietary VM integration and connection to the Strata CIX LAN.

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 Strata CIX. 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.

Volume Control

System Availability: All systems

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

Appendix – Specifications

This appendix includes detailed information on the items listed below. The sections in this appendix apply to the Strata CIX1200, CIX670, CIX200 and CIX100, unless otherwise stated. Refer to page 13 for CIX40 requirements.

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

Environmental Characteristics

The environmental requirements for the Strata CIX40, CIX100, CIX200, CIX670, and CIX1200 systems are shown in Table 24.

Table 24 Environmental Characteristics for the Strata CIX systems

Environmental Specifications	
Operating temperature Operating humidity Storage temperature	32~104° F (0~40° C) 20~80% relative humidity without condensation -4~140° F (-20~60° C)
BTU Rating	
ACTU (1) or HCTU/HEXxU BCTU/BEXU (1 installed) BDKU (5 installed) RCOU/RCOS (1 installed) Digital Telephones (40 installed)	CIX100: 105 BTUs (31 watt hours) per cabinet. CIX670 and CIX1200: 190 BTUs (56 watt hours) per cabinet.

Power Considerations

CIX40

Table 25 CIX40 Electrical Characteristics

CIX40 Primary Power	
Input AC (Power Supply Specification) AC frequency Power AC input current	120~240VAC 50/60 Hz CIX40 - 100 watts maximum 1.3A maximum (100 VAC)
Power	
Input DC	15V use the factory-shipped AC adapter
Power Converter	
DC voltage output specification	-24VDC (-26.3 ~ -28.3VDC) +5VDC (+4.5 ~ +5.5VDC)
Standard Telephone Ring Circuit (GMAU2 and GSTU1)	
Ring Voltage	180V p-p square wave
Ringing capability	1 REN, 1 circuit - one telephone per circuit

CIX100

The power supply in each CIX100 Base and Expansion Cabinet furnishes power to all of the stations and some of the interface peripherals (see Table 26). The primary AC power for each cabinet is 120VAC.

Table 26 CIX100 Electrical Characteristics

CIX100 Primary AC Power Voltage	
Input AC AC Frequency Watts per cabinet (maximum)	120VAC 60 Hz, Single-phase (48Hz~62Hz) 100 watts (maximum)
CIX100 Primary Power Current Consumption (Rating in Amperes)	
 1 cabinet 2 cabinets	120VAC 1.8 amps 3.6 amps
Power Supply Unit (APSU112A)	
DC voltage output specification	-24VDC (-26.3~-27.8VDC, 3.2 DC amps) +5VDC (+4.5~5.5VDC, 2.0 DC amps) -5VDC (-4.5~-5.5VDC, 0.2 DC amps) +3.3VDC (+3.0~3.6VDC, 0.5 DC amps

CIX200

Table 27 CIX200 Electrical Characteristics

CIX200 Primary AC Power Voltage	
Input AC AC Frequency Watts per cabinet (maximum)	120VAC, 4.0 amps max. Single-phase (45Hz~65Hz) 480 watts (maximum)

CIX670 and CIX1200

The power supply in each CIX670 and CIX1200 Base and Expansion Cabinet furnishes power to all of the stations and some of the interface peripherals (see Table 28). The primary AC power can be 120VAC, 208VAC or 240VAC. Systems containing six or seven cabinets require 208VAC or 240VAC. See page 13 for CIX40 requirements.

Table 28 CIX670 and CIX1200 Electrical Characteristics

CIX670 and CIX1200 Primary AC Power Voltage			
Input AC AC Frequency Watts per cabinet (continuous) Watts for five cabinet system	115±10VAC or 208±20VAC or 240±20VAC 50/60 Hz, Single-phase (48~62Hz) 180 900		
CIX670 and CIX1200 Primary Power Current Consumption (Rating in Amperes)			
Number of Cabinets:	120VAC	208VAC	240VAC
1	3.2 amps	2.2 amps	2.0 amps
2	6.4 amps	4.4 amps	4.0 amps
3	9.6 amps	6.6 amps	6.0 amps
4	12.8 amps	8.8 amps	8.0 amps
5	16.0 amps	11.0 amps	10.0 amps
6	N/A	13.2 amps	12.0 amps
7	N/A	15.4 amps	14.0 amps
CIX1200 only Primary Power Current Consumption (Rating in Amperes)			
Number of Cabinets:	120VAC	208VAC	240VAC
8	3.2 amps	2.2 amps	2.0 amps
9	6.4 amps	4.4 amps	4.0 amps
10	9.6 amps	6.6 amps	6.0 amps
11	12.8 amps	8.8 amps	8.0 amps
12	16.0 amps	11.0 amps	10.0 amps
Power Supply Unit (BPSU672)			
DC voltage output specification	-24VDC (-26.3~-27.8VDC, 6.0 DC amps) +5VDC (+4.5~5.5VDC, 4.0 DC amps) -5VDC (-4.5~-5.5VDC, 0.8 DC amps)		

Note AC Power treatment for CIX1200 systems should be configured as two separate systems – one 7-cabinet stack and one 5-cabinet stack. This will require two 20 amp, 240VAC, single phase power circuits.

Reserve Power

Two or four customer-supplied 12VDC reserve batteries (80 ampere-hours max.) can be connected to either system to maintain normal operation during a power failure (see Tables 29~33). The batteries are kept in a highly-charged state by the power supply's battery charger and must be connected when the system is operating normally. Fully charged batteries must be connected when normal AC power is available, batteries cannot be connected after/during an actual power failure.

The Strata CIX200 does not have a battery; make sure you use Uninterruptible Power Supply (UPS). The battery charger is standard on the CIX670 and CIX1200 power supply. An optional ABCS battery charger must be used in the CIX100 power supply.

CIX40 Reserve Power

Refer to "Reserve Power" on page 11.

Table 29 CIX100 Reserve Power Characteristics

Battery Charger Characteristics	Maximum Battery Charger Drain (-24VDC)	
Charger: current limiting Nominal float voltage: 2.275 volts/cell Charge current: 280mA amps maximum Battery discharge cut-off voltage: 20.5 ±0.5VDC	Base Cabinet Base + Expansion Cabinets	3.15 amps 6.30 amps

Table 30 CIX100 Typical Reserve Power Duration Estimates[1]

Number of Cabinets	1	2
Estimated operation time: two-battery configuration	25 hr.	12.5 hr.
Estimated operation time: four-battery configuration	50 hr.	25 hr.
DC Current Drain (-24VDC)	3.15 amps.	6.30 amps.

1. Assumes 80 ampere-hours with 12VDC batteries.

Table 31 CIX670 Reserve Power Characteristics

Battery Charger Characteristics	Maximum Battery Charger Drain (-24VDC)			
Charger: current limiting Nominal float voltage: 2.275 volts/cell Charge current: 0.7 amps maximum Battery discharge cut-off voltage: 20.5 ±0.5VDC	1 cabinet 2 cabinets 3 cabinets 4 cabinets	6.0 amps 12.0 amps 18.0 amps 24.0 amps	5 cabinets 6 cabinets 7 cabinets	30.0 amps 36.0 amps 42.0 amps

Table 32 CIX1200 Reserve Power Characteristics

Battery Charger Characteristics	Maximum Battery Charger Drain (-24VDC)			
Charger: current limiting Nominal float voltage: 2.275 volts/cell Charge current: 0.7 amps maximum Battery discharge cut-off voltage: 20.5 ±0.5VDC	1 cabinet 2 cabinets 3 cabinets 4 cabinets 5 cabinets 6 cabinets 7 cabinets	6.0 amps 12.0 amps 18.0 amps 24.0 amps 30.0 amps 36.0 amps 42.0 amps	8 cabinets 9 cabinets 10 cabinets 11 cabinets 12 cabinets	6.0 amps 12.0 amps 18.0 amps 24.0 amps 30.0 amps

Note Reserve power treatment for CIX1200 should be configured as two separate systems – one 7-cabinet stack and one 5-cabinet stack.

Table 33 CIX670 and CIX1200 Typical Reserve Power Duration Estimates[1]

Number of Cabinets	1	2	3	4	5	6	7
Estimated operation time Two-battery configuration	12.0 hr.	6.0 hr.	4.0 hr.	3.0 hr.	2.5 hr.	2.0 hr.	1.8 hr.
Estimated operation time Four-battery configuration	24.0 hr.	12.0 hr.	8.0 hr.	6.0 hr.	5.0 hr.	4.0 hr.	3.5 hr.
DC Current Drain (-24VDC)	4.6 amps.	8.7 amps.	12.8 amps.	16.9 amps.	21.0 amps.	25.1 amps.	29.2 amps.

1. Assumes 80 ampere-hours with 12VDC batteries.

Table 34 CIX1200 Typical Reserve Power Duration Estimates[1,2]

Number of Cabinets	8	9	10	11	12
Estimated operation time Two-battery configuration	12.0 hr.	6.0 hr.	4.0 hr.	3.0 hr.	2.5 hr.
Estimated operation time Four-battery configuration	24.0 hr.	12.0 hr.	8.0 hr.	6.0 hr.	5.0 hr.
DC Current Drain (-24VDC)	4.6 amps.	8.7 amps.	12.8 amps.	16.9 amps.	21.0 amps.

1. Assumes 80 ampere-hours with 12VDC batteries.
2. Reserve power treatment for CIX1200 should be configured as two separate systems – one 7-cabinet stack and one 5-cabinet stack.

Strata CIX Component Compatibility

Because the Strata CIX supports all types of Toshiba IP and digital 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. That's why the Strata CIX is much more than just an IP system. It's a unified communications environment that supports many types of client devices.

Table 35 Component Compatibility

System Software and Features	CIX40	CIX100	CIX200	CIX670	CIX1200
Strata CIX Software and features	X	X	X	X	X
FeatureFlex Adaptability features	X	X	X	X	X
ACD / MIS	**CIX40**	**CIX100**	**CIX200**	**CIX670**	**CIX1200**
Strata ACD	X	X	X	X	X
TASKE Call Center Reporting	X	X	X	X	X
Administration	**CIX40**	**CIX100**	**CIX200**	**CIX670**	**CIX1200**
Network eManager [Browser-based Unified (CIX & SES) Admin]	X	X	X	X	X
My Phone Manager ™	X	X	X	X	X
WinCIX (Off-line Programming)	NC	X	X	X	X
Processor Cards	**CIX40**	**CIX100**	**CIX200**	**CIX670**	**CIX1200**
BECU/BBCU with optional BBMS, BEXS, BSIS	NC	NC	NC	X	NC
B_CAU/B_CBU cards for DK424i	NC	NC	NC	NC	NC
RCTU cards for DK424	NC	NC	NC	NC	NC
ACTU and subassemblies	NC	X	NC	NC	NC
LCTU and subassemblies	NC	NC	X	NC	NC
HCTU1[1]	NC	NC	NC	NC	X
HEXAU1	NC	NC	NC	NC	X
HEXBU1	NC	NC	NC	NC	X
BCTU2[2] and BEXU2	NC	NC	NC	X	NC
GCTU2A	X	NC	NC	NC	NC
DTMF Receiver Unit	**CIX40**	**CIX100**	**CIX200**	**CIX670**	**CIX1200**
BRCS-4/8/12	NC	NC	NC	NC	NC
RRCS-4/8/12	NC	NC	NC	NC	NC
ARCS (16) - (Used for the ACTU1 on the CIX 100, built-in on the other ACTU)	NC	ACTU1	Built-in (16)	Built-in (16)	NC
BEXU (Adds 16 DTMF receivers to the BCTU)	NC	NC	NC	BCTU	NC
Optional Interface Unit	**CIX40**	**CIX100**	**CIX200**	**CIX670**	**CIX1200**
BIOU	NC	X	X	X	X
BSIS	NC	X	X	X	X
Standard Telephone Interface	**CIX40**	**CIX100**	**CIX200**	**CIX670**	**CIX1200**
BSTU, BSTU2A, RSTU3, RDSU/RSTS	NC	X	X	X	X
BSTCIU (Requires R4.1 software)	NC	X (R4.1)	X (R4.1)	X (R4.1)	X
BSLU / BSLU (Requires R3.1 or later software)	NC	X (R3.1)	X (R3.1)	X (R3.1)	X
ASTU (R1.3 and higher)	NC	X	NC	NC	NC
LSLU	NC	NC	X	NC	NC
GSTU	X	NC	NC	NC	NC
Digital Telephone Interface	**CIX40**	**CIX100**	**CIX200**	**CIX670**	**CIX1200**
ADKU	NC	X	NC	NC	NC
BDKU	NC	X	X	X	X
BWDKU1A	NC	X	X	X	X
BDKS	NC	X	X	X	X

Table 35 Component Compatibility *(continued)*

PDKU2 (DKT2000-series phones only)	NC	X	X	X	X
RDSU, RSTS (DKT2000-series only)	NC	X	X	X	X
GCDU	X	NC	NC	NC	NC
IP Telephone Interface	**CIX40**	**CIX100**	**CIX200**	**CIX670**	**CIX1200**
MIPU	MIPU only	X	X	X	X
CO Line Interface	**CIX40**	**CIX100**	**CIX200**	**CIX670**	**CIX1200**
BVPU	NC	X	X	X	X
RCIU/RCIS	NC	X	X	X	X
RCMU/RCMS	NC	X	X	X	X
RCOU/RCOS[3]	NC	X	X	X	X
BCOCIU / BCOCIS (Requires R4.1 software)	NC	X (R4.1)	X (R4.1)	X (R4.1)	X
RDDU	NC	X	X	X	X
RDTU2, 3	NC	X	X	X	X
REMU	NC	X	X	X	X
RGLU2, RGLU3	NC	X	X	X	X
GCDU	X	NC	NC	NC	NC
GCOCIU	X	NC	NC	NC	NC
ISDN Interface	**CIX40**	**CIX100**	**CIX200**	**CIX670**	**CIX1200**
BPTU1, RPTU2, RPTU	NC	X	X	X[4]	X
Remote Exp. Cab.Interface	**CIX40**	**CIX100**	**CIX200**	**CIX670**	**CIX1200**
RRCU	NC	NC	NC	X	X
Strata Net over IP Interface	**CIX40**	**CIX100**	**CIX200**	**CIX670**	**CIX1200**
MIPU [3]	MIPU only	X	X	X	X
GIPU / GIPH	X	NC	NC	NC	NC
Stations and Terminal Equipment	**CIX40**	**CIX100**	**CIX200**	**CIX670**	**CIX1200**
Strata CIX PC Attendant Console, BATI	X	X[5]	X	X	X
BPCI (USB) - Data or Voice Record TAPI	X	X[4]	X	X	X
DP5000	X	X[4]	X	X	X
DKT1000[6]	X	X	X	X	X
DKT2000	X	X[4]	X	X	X
DKT3200/3000	X	X[4]	X	X	X
IPT1020-SD	X	X[4]	X	X	X
IPT2010-SD	X	X[4]	X	X	X
IPT2020-SD	X	X[4]	X	X	X
IPT2008-SDL	X	X[4,6]	X[6]	X[7]	X
IPT2010-SDC	X	X[4]	X	X	X
IP5000-series telephones	X	X[4]	X	X	X
SoftIPT softphone client for Laptop	X	X	X	X	X
SoftIPT softphone client for PDA	X	X	X	X	X
5000/3000/3200-series digital telephones	X	X[8]	X[1]	X[1]	X
DKT3207-SD (W) and DKT3207-SD 7-button digital telephone	X	NC	NC	NC	X
SIP (3rd party) IP telephones (Requires MIPU[9])	X	X	X	X	X
SIP (3rd party) IP telephones (Requires GIPU)	X	NC	NC	NC	X
2000-series digital telephones	X	X	X	X	X
Voice Mail	**CIX40**	**CIX100**	**CIX200**	**CIX670**	**CIX1200**
LVMU1A	NC	X	X	X	X
GVPH1A	X	NC	NC	NC	NC
Attendant Consoles	**CIX40**	**CIX100**	**CIX200**	**CIX670**	**CIX1200**
Strata CIX IP Attendant Console (Requires MIPU/GIPU)	X	X	X	X	X
Strata CIX Digital Attendant Console (Digital Telephone Port connection)	X	X	X	X	X

Table 35 Component Compatibility *(continued)*

Maximum Capacities	CIX40	CIX100	CIX200	CIX670	CIX1200
KSU Cabinets (base plus expansion)	1	2	2	7	12
Base Cabinet PCB Slots	4	4	3	8	8
Expansion Cabinet PCB Slots	0	4	4	10	10
Total System PCB Slots	4	8	7	68	118
Trunk/Station/Voice mail Ports	57	112	192	672	1152
Trunk Lines	11	64	96	264	440
IP Telephones (IP5000-series/IPT2000-series)[10]	24	72	160	560	1000
Strata Net Channels	24	48	96	264	440
Digital Telephones (DP5000-series, DKT2000, 3000/3200)	16	72	112	560	952
Analog Standard Telephones	2	56	58	544	944
Ethernet LAN	**CIX40**	**CIX100**	**CIX200**	**CIX670**	**CIX1200**
AETS (Used for the ACTU1 on the CIX 100, built-in on the other ACTU processors)	Built-in	ACTU1	Built-in	Built-in	Built-in
V.34 Admin Modem	**CIX40**	**CIX100**	**CIX200**	**CIX670**	**CIX1200**
AMDS	NC	X	X	X	X
Base Cabinet	**CIX40**	**CIX100**	**CIX200**	**CIX670**	**CIX1200**
CHSUB672	NC	NC	NC	X	X
CHSUB112	NC	X	NC	NC	NC
CHSUB192A	NC	NC	X	NC	NC
CHSU40A	X	NC	NC	NC	NC
CHSU40A2	X	NC	NC	NC	NC
Expansion Cabinet	**CIX40**	**CIX100**	**CIX200**	**CIX670**	**CIX1200**
CHSUE672	NC	NC	NC	X	X
CHSUE112	NC	X	NC	NC	NC
CHSUE192A	NC	NC	X	NC	NC
Data Cable for CIX670 Expansion Cabinet	NC	NC	NC	X	X
Data Cable for CIX100 Expansion Cabinet	NC	X	NC	NC	NC
Data Cable for CIX200 Expansion Cabinet	NC	NC	X	NC	NC
Power Supply Unit	**CIX40**	**CIX100**	**CIX200**	**CIX670**	**CIX1200**
BPSU672 (120VAC/208VAC/240VAC power supply)	NC	NC	NC	X	X
APSU112 (120VAC)	NC	X	NC	NC	NC
Conduit Connection Box	**CIX40**	**CIX100**	**CIX200**	**CIX670**	**CIX1200**
RCCB2	NC	NC	NC	NC	NC
BCCB120 (120V box)	NC	NC	NC	X	X
BCCB240 (240V box)	NC	NC	NC	X	X
Battery Distribution Box	**CIX40**	**CIX100**	**CIX200**	**CIX670**	**CIX1200**
BBDB1 (new Battery Dist. Box, 7 BBTC2A-2.0M)	NC	NC	X	X	X

Table 35 Component Compatibility *(continued)*

Power Strip	CIX40	CIX100	CIX200	CIX670	CIX1200
RPSB2 (120VAC power strip)	NC	NC	NC	X	X
BPSB240 (240VAC power strip)	NC	NC	NC	X	X
Battery Cable	**CIX40**	**CIX100**	**CIX200**	**CIX670**	**CIX1200**
PBTC-3M	NC	NC	NC	X	X
BBTC1A-2.0M	NC	NC	NC	X	X
ABTC-3M	NC	X	NC	NC	NC
Battery Charger	**CIX40**	**CIX100**	**CIX200**	**CIX670**	**CIX1200**
ABCS1	NC	X	NC	Built-in	Built-in
HPFB-6[11]	X	NC	NC	NC	NC

X= Compatible NC = Not Compatible

1. When the HCTU processor is installed in the base cabinet (CHSUB672A or CRSUB672A), the system is a Strata CIX1200.
2. When the BCTU processor is installed in the base cabinet (CHSUB672A or CRSUB672A), the system is a Strata CIX670.
3. The RCOS1A cannot be installed on the RCOU3A. The RCOS3 can be installed on the RCOU1A.
4. BPTU1, RPTU2, MIPU is required for Strata Net Networking.
5. Compatible with CIX40
6. DKT1000-series telephones do not support continuous DTMF tones.
7. Requires an MIPU / GIPU on CIX40.
8. DP5022-SDM requires LIC-1-DP5022SDM license to operate on the CIX100/200/670/1200 systems.
9. Release 5.1 hardware is required to support the MIPU, GIPU and GCOCIH1A on the CIX40.
10. Refer to “Telephones and Peripherals” on page 45 for numbers of IP5000-series phones supported in the system.
11. Provides battery with built-in charger.

Public Network Requirements

The Circuit Card requirements for connecting to the public network are shown in Table 36.

Table 36 Circuit Card Network Requirements

<table>
<tr><th>Circuit Card Interface</th><th>Facility Interface Code</th><th>Network Jack</th><th>Ringer Equivalence</th><th>Universal Service Order Code</th></tr>
<tr><td>BSTU[1]/RSTU3/RDSU[2]
(Off-premises Station)</td><td>OL13B (RSTU3, –24V)
OL13C (RSTU3, RDSU with R48S-48V)</td><td>RJ21X</td><td>N/A</td><td>9.0F</td></tr>
<tr><td>BSLU/BSLS[3]
CIX40 Standard Telephone Interfaces</td><td>OL13B (–24V)</td><td>RJ21X</td><td>N/A</td><td>9.0F</td></tr>
<tr><td>BSTCIU (Analog Station[4] with Caller ID)</td><td>OL13B (–24V)
OL13C (with R48S-48V)</td><td>RJ21X / RJ11C</td><td>N/A</td><td>9.0F</td></tr>
<tr><td>RCOU/RCOS[5]
(loop start line)</td><td>02LS2</td><td>RJ14C/RJ21X
(all others)</td><td>0.3B</td><td>N/A</td></tr>
<tr><td>RCIU2/RCIS (Caller ID)</td><td>N/A</td><td>RJ21X/RJ14C</td><td>0.3B</td><td>N/A</td></tr>
<tr><td>BCOCIU/BCOCIS
CIX40 CO Line Interfaces</td><td>02LS2</td><td>RJ21X / RJ11C / RJ14C</td><td>0.3B</td><td>N/A</td></tr>
<tr><td>RDDU</td><td>02RV2-T (Dealer-supplied CSU)</td><td>RJ14C/RJ21X</td><td>0.0B</td><td>AS.2</td></tr>
<tr><td>REMU type 1 or type 2</td><td>TL11M, 2-wire
TL31M, 4-wire
TL12M, type 2, 2-wire
TL32M, type 2, 4-wire</td><td>RJ2EX
RJ2GX
RJ2FX
RJ2HX</td><td>Not Available
(N/A)</td><td>9.0F</td></tr>
<tr><td>RGLU3 (ground or loop start line)[2]</td><td>02GS2 (ground)
02LS2 (loop)</td><td>RJ14C/RJ11CX</td><td>0.3B</td><td>N/A</td></tr>
<tr><td>RDTU (DS-1/T1)[6]</td><td>(See last bullet note on Note 2 below.)</td><td>RJ48C/RJ48X/ RJ48M</td><td>N/A</td><td>6.0P</td></tr>
<tr><td>BPTU, RPTU (PRI)[7, 8]</td><td>04DU9-1SN (Dealer-supplied CSU)</td><td>RJ48C/RJ48M</td><td rowspan="3">N/A</td><td rowspan="3">6.0P</td></tr>
<tr><td>BPTU, RPTU (Strata Net)</td><td>04DU9-1SN (Dealer-supplied CSU)</td><td>RJ48C/RJ48M</td></tr>
<tr><td>RMCU/RCMS (CAMA)</td><td>02RV2-O</td><td>RJ11C/RJ21-X</td></tr>
</table>

1. BSTU parameters: Loop current 25mA to 35mA.
   Maximum loop resistance allowed:
      600ohm each without R48S1A - Class B (FCC Part 68)
      1200ohm each with R48S1A - Class C (FCC Part 68)
   Impedance: 600ohm
2. Only RDSU ckts. 1~4 provide Off-premises Station (OPS) ability. RDSU must use OL13A or OL13B if providing –24 volt loop voltage. If equipped with the –48 volt loop option circuit card (R48S), OL13A, OL13B, or OL13C may be used for OPS connection.
3. Supports one ringer per circuit. Does not support Message Waiting Lamp or OPS.
4. BSTCIU parameters: Loop current 25mA to 35mA.
   Maximum loop resistance allowed:
      600ohm each without R48S1A - Class B (FCC Part 68)
      1200ohm each with R48S1A - Class C (FCC Part 68)
   Impedance: 600ohm
   Ring cadence: On 2 sec., Off 4 sec. (Use for Caller ID) or On 1 sec., Off 3 sec.
5. Loop current requirements for Strata loop and ground start lines: 20 milliamperes (mA) min./120 mA max.

6. When ordering DS-1/T1 circuits, six items must be specified:

   The number of channels per T1 circuit, fractional increments are normally 8, 12, or 16 channels, full service is 24 channels. Unused channels must be bit-stuffed.

   CO line types assigned to each channel: Loop Start, Ground Start, Tie (Wink or Immediate Start), DID (Wink or Immediate).

   Frame Format Type: Super Frame (SF) or Extended Super Frame (ESF). The T1 provider normally specifies the Frame Format to be used, either is adequate for CO digital voice lines. ESF provides a higher level of performance monitoring, but requires trained personnel and the ESF CSU normally costs more than an SF only CSU.

   Line Code Type: Alternate Mark Inversion (AMI) or Bipolar 8 Zero Substitution (B8ZS). The T1 provider normally specified the Line Code to be used, either is adequate for T1 CO digital voice lines.

   The customer may have to provide the Channel Service Unit (CSU) to interface the CIX T1 circuit to the Telco T1 circuit. (CSUs are a Telco requirement.)

   RDTU Network Channel Interface Codes: 04DU9-BN, 04DU9-DNZZ, 04DU9-1SN, 04DU9-1KN, 04DU9-1ZN.
7. For information on how to order ISDN PRI circuits, you should refer to the Toshiba ISDN Training CBT. ISDN circuits may require a customer-provided CSU for PRI and/or Terminal Adapter or Network Terminal units for BRI. In U.S. CSU/TAs must be UL-listed in the U.S. In Canada, they must be CSA certified.
8. RPTU2 is required for Strata Net private networking.

Station Loop Lengths

In a single site installation, the Base and optional Expansion cabinets must be placed within the allowed maximum distance of each other as designated by Table 37.

Table 37 Station Loop Lengths[1]

<table>
<tr><th rowspan="2">Mode</th><th colspan="3">Maximum line length (24 AWG)</th></tr>
<tr><th>1 Pair[2]</th><th>2 Pair (Not Available for CIX40 or BDKS)</th><th>1 Pair plus external power [3]</th></tr>
<tr><td>DP5000, DKT3000/3200 or DKT2000-series</td><td rowspan="6">1000 ft. (303m)</td><td rowspan="8">1000 ft. (303m)</td><td rowspan="8">1000 ft. (303m)</td></tr>
<tr><td>DP with DOCA
DKT with BVSU or DVSU</td></tr>
<tr><td>DP with DOCA-1A
DKT with BHEU or HHEU</td></tr>
<tr><td>DKT with BPCI</td></tr>
<tr><td>DKT with BPCI + BHEU[4]</td></tr>
<tr><td>DP with DOCA
DKT with BVSU + BHEU[4]
or DVSU + HHEU[4]</td></tr>
<tr><td>DP with KM5020 or LM5110
DKT with DADM3020 or DADM2020 (1 ADM)[2, 5]</td><td>675 ft. (204m)</td></tr>
<tr><td>DKT with DADM3020 or DADM2020 (2 ADMs)[2, 5]</td><td>500 ft. (151m)</td></tr>
<tr><td>DDM5060 or DDSS3060, DDSS3060 or DDSS2060</td><td rowspan="3">1000 ft. (303m)</td><td>1000 ft. (303m)</td><td>1000 ft. (303m)</td></tr>
<tr><td>BATI, RATI</td><td>n/a</td><td>n/a</td></tr>
<tr><td>DDCB3</td><td>1000 ft. (303m)</td><td>1000 ft. (303m)</td></tr>
<tr><td rowspan="3">Standard telephones, voice mail, AA, etc.</td><td>Approx. 3000 ft. (909 m) with 150 ohm device. [6]</td><td rowspan="3">n/a</td><td rowspan="3">n/a</td></tr>
<tr><td>Approx. 9000 ft. (2727 m) with 150 ohm device.[6]</td></tr>
<tr><td>Approx. 21000 ft. (6363 m) with 150 ohm device.[6]</td></tr>
<tr><td>IPT2010-SD
IPT2020-SD
IPT2008-SD
IPT1020-SD
IPT2010-SDC</td><td colspan="3">The IP telephone interface is 10Base-T/100Base-TX and requires
CAT5/5e/6 twisted pair cabling.
The maximum distance between the IP telephone jack and the ethernet device it connects to is 100 meters (328 ft.). This includes the 3 meter (9.84 ft.) straight-through CAT5 cable (black) supplied with the IP telephone. Ethernet devices include
MIPU servers, routers, etc.</td></tr>
<tr><td>IP5022-SD, IP5122-SD, IP5132-SD, IP5131-SDL, and IP5122-SDC</td><td colspan="3">10 Base-T / 100 Base-T / 1000 Base-T</td></tr>
<tr><td>IP5522-SD, IP5622-SD and IP5631</td><td colspan="3">10 Base-T / 100 Base-T</td></tr>
</table>

1. When the system is powered by backup battery, range may be less as the backup battery is discharged.
2. One-pair wiring must be used with BWDKU and BDKS and CIX40 digital circuits (see Figure 2). The BWDKU and BDKS and CIX40 digital circuits do not support two pair wiring.
3. Two-pair wiring or optional telephone power supply is required to achieve maximum range in all cases.
4. DP5000-series does not require BHEU for headset and does not support BPCI.
5. BDKS and BWDKU do not provide the power wire pair; an external power supply is required to achieve maximum range (see “Digital Telephone DSS and DDCB External Power Connection” in Chapter 8 – MDF circuit card Wiring of the *Strata CIX I&M Manual)*.
6. See manufacturer’s product specifications for exact resistance of device.

Standard Telephone Ringer Specifications

Specifications for standard telephone ringers appear in Table 38.

Table 38 Standard Telephone Ringer Specifications

Interface Card	Ring Voltage	Ring Capacity	MW Voltage	Modem Data Rate	Ring Cadence
RSTU3 or RDSU	75 Vrms@Ren1, 60 Vrms Ren 3, 20Hz	RSTU3: 3.0 ringers per circuit RDSU: 1.5 ringers per circuit	RSTU3 and BSTU: -120VDC~-85VDC 0.9 sec. high/ 9.1 sec. low 1 telephone per circuit (max.)	14,400 bps maximum	Ring Cadence 1: (RSTU, RDSU, BSTU, BSLU, BSLS) External Ring: 20Hz, 1 sec. ON - 3 sec. Off Internal Ring: 20Hz, 0.4 sec. On - 0.2 sec. Off - 0.4 sec. On - 3 sec. Off Recall: 20Hz, 1 sec. On - 1 sec. Off Ring Cadence 2: (RSTU, RDSU, BSTU) External Ring: 20Hz, 0.4 sec. On - 0.2 sec. Off - 0.4 sec. On - 3 sec. Off11 Internal Ring: 20Hz, 1 sec. On - 3 sec. Off Recall: 20Hz, 1 sec. On - 3 sec. Off
BSTU		3 ringers per circuit			
BSLU/BSLS ASTU LSLU CIX40 mother board GSTU		1 ringer per circuit	None		
BSTCIU		3 ringers per circuit	110 V +/- 10 1 Hz		Ring: 20Hz, 2 sec. ON - 4 sec. Off or 1 sec. ON - 3 sec. Off

Standard Telephone Interface Options

The table below summarizes the standard telephone interface cards and their capacities and capabilities.

Table 39 Standard Telephone Interface Options

Interface Card	CIX Software Release Requirement	Interface Circuits per Card	Message Waiting Lamp Control	Message Waiting Off-Hook Audible Alert (stutter dial tone)	Off-Premise Station support (R48S card)	Standard telephone Caller ID and MW display.	Voice Mail Port, FAX machine, Modem, Standard 2500 telephone
BSTU1A	All Releases	8	Yes	Yes	Yes	No	Yes
BSTU2A[1,2]	R520MT034 and above						
BSTCIU1A	R4.1 and above	8	Yes	Yes	Yes	Yes	Yes
BSTCIU2A[1,2]	R520MT034 and above						
BSLU1A/ BSLS1A	R3.1 and above	8/8 total 16	No	Yes	No	No	Yes
ASTU, LSLU	CIX100 CIX200	2	No	Yes	No	No	Yes
Motherboard and GSTU	CIX40	1	No	Yes	No	No	Yes
Note: All standard telephone interface cards, including ASTU and LSLU, support off-hook selection of an outside line in CIX R4.1 and later.							

1. BSTU2A / BSTCIU2A support disconnect supervision.
2. BSTU2A / BSTCIU2A also require Network eManager R520B10 and above.

5000/3000/3200-series Telephone Option Circuit Cards

DKT3000/3200-series telephones can be upgraded with option circuit cards to add a number of features. Each of these upgrades shares a circuit with the telephone that it is connected to and is not considered a station. See Table 40 for more information.

Table 40 3000/3200 Telephone Subassembly Upgrades

Subassembly	No. per Phone	Function
BVSU[1] (DKT) DOCA (DP)	1	**Speaker Off-hook Call Announce (OCA):** Provides interface for digital telephone to receive Speaker OCA. Not required for Handset/Headset OCA. (Built into IP telephones).
BHEU or HHEU[2]	1	**Headset and external ringer telephone interface:** Can be installed with BVSU, BPCI or DADM. (Built into IP telephones).
BPCI[1,3]	1	Desktop PC Interface for CTI applications.

1. Telephones with the BPCI cannot have Speaker OCA (BVSU) or Add-on modules. Also, DKT3001, DKT3201 telephones cannot have CTI (BPCI), Speaker OCA (BVSU) or Add-on modules.
2. DP5000-series telephones have a built-in headset interface and do not require BHEU or HHEU.
3. DP5000-series do not support the BPCI interface.

Station Dimensions

Dimensions for the 3000-series, IPT telephones and related equipment are listed in Table 41.

Table 41 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], 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
DP5018-S, DP5022-SDM, DP5022-SD, DP5122-SD, DP5032-SD, DP5132-SD, DP5130-SDL, DP5130-FSDL [15-degree tilt]	5.1	129	10.16	258	6.10	155
Single Line Telephone (DP5008) [15-degree tilt]	3.86	98	5.9	150		
Add-on Module (KM5020, LM5110) [15-degree tilt]	4.0	102	3.54	90		
Direct Station Selection (DSS) Console (DDM5060) [15-degree tilt]			10.16	258		
10-button IP Telephone with LCD (IPT2010-SD) 20-button IP Telephone with LCD (IPT2020-SD) 8-button IP Telephone with large LCD (IPT2008-SDL)	4.5	115	7.8	198	9.6	245
Add-on Module (IADM2020	3.8	96.5	2.8	70		
Direct Station Selection (DSS) Console (IDSS2060)	3.8	96.5	7.8	198		
20-button IP Telephone with LCD (IPT1020-SD)	4.0	101.5	8.1	205	9.3 in.	235
Digital Single Line Telephone (DKT3001)	4.0	101.5	5.9	150		
Add-on Module (DADM3020, DADM3120)	3.5	88	2.8	70		
Direct Station Selection (DSS) Console (DDSS3060)	3.5	88	8.1	205		
Handset with Handset Cradle (BATHC)	2.9	73	2.8	71	9.6	244
10-button DKT with Handsfree Answerback (DKT2010-S)	4.1	104	7.7	195	9.1	230
10-button DKT with LCD (DKT2010-SD)						
20-button DKT (DKT2020-S)						
20-button DKT with LCD (DKT2020-SD)						
20-button DKT with LCD (DKT2020-FDSP)						
Digital Single Line Telephone (DKT2001)	4.2	107	5.5	140	9.1	230
Add-on Module (DADM2020)	3.3	85	2.8	70	9.1	230
Direct Station Selection (DSSS2060) Console	3.3	85	7.8	199	9.1	230
External Speaker Amplifier (BESCB)	10.2	260	10.2	260	4.9	125
Door Phone/Lock Control Unit (DDCB)	4.7	120	6.5	165	1.5	38
Door Phone (MDFB)	5.5	140	3.1	80	1.3	32
Attendant Console Interface (BATI)	1.7	42.4	5.0	126	7.3	185
DKT2404-DECT Digital Cordless Telephone	**Height**		**Width**		**Depth**	
	Inches	**mm**	**Inches**	**mm**	**Inches**	**mm**
Base	5	127	8.5	216	2.25	57.15
Handset	6.5	165	2.25	57.15	1.0	25.4

IP Telephone Power Consumption

The power consumption for the IP5000-series telephones and the Add-on modules is shown in Table 42. Use this information to calculate the Power over Ethernet (PoE) requirements and UPS capacity.

Table 42 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

System Tones

Tones which can be heard from speaker or handset are described in Table 43.

Table 43 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	Ringing 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 44. Due to the limitation in the tone generation algorithm, the listed tone duration is slightly different from the actual one.

Table 44 Ring Tones

<table>
<tr><th>Tone Name</th><th>Description</th><th>Ringing Cadence</th></tr>
<tr><td>Internal/External Ring 1</td><td rowspan="10">Incoming call from internal or external party to DP or IPT. (10 different ring tones are available with R1.3 or higher software.)</td><td>500 Hz 1 sec. On, 3 sec. Off, repeat</td></tr>
<tr><td>Internal/External Ring 2</td><td>1300 Hz 1 sec. On, 1 sec. Off, repeat</td></tr>
<tr><td>Internal/External Ring 11</td><td>500/640 Hz 1 sec. On, 3 sec. Off, repeat</td></tr>
<tr><td>Internal/External Ring 12</td><td>500/640 Hz 1 sec. On, 1 sec. Off, repeat</td></tr>
<tr><td>Internal/External Ring 13</td><td>860/1180 Hz 1 sec. On, 3 sec. Off, repeat</td></tr>
<tr><td>Internal/External Ring 14</td><td>860/1180 Hz 1 sec. On, 1 sec. Off, repeat</td></tr>
<tr><td>Internal/External Ring 15</td><td>1300/1780 Hz 1 sec. On, 3 sec. Off, repeat</td></tr>
<tr><td>Internal/External Ring 16</td><td>1300/1780 Hz 1 sec. On, 1 sec. Off, repeat</td></tr>
<tr><td>Internal/External Ring 17</td><td>860/1180 Hz 0.5 sec. On, 1300/1780 Hz 3 sec. Off, repeat</td></tr>
<tr><td>Internal/External Ring 18</td><td>860/1180 Hz 0.5 sec. On, 1300/1780 Hz 1 sec. Off, repeat</td></tr>
<tr><td rowspan="2">External/Internal Ring for Standard Telephones</td><td rowspan="2">Internal and External Ringing Cadence: For Release 1.3 and higher, two types of ringing cadences can be selected in system programming.</td><td>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</td></tr>
<tr><td>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</td></tr>
<tr><td>Recall</td><td>A call is returned & needs to be answered.</td><td>2 kHz interrupted at 10 Hz, 1 sec. On, 1 sec. Off, repeat.</td></tr>
<tr><td>Recall for Standard Telephone</td><td>A call is returned & needs to be answered.</td><td>20 Hz, 1 sec. On, 1 sec. Off, repeat.</td></tr>
<tr><td>Ring Over Busy (Internal)</td><td>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.</td><td rowspan="2">2 kHz interrupted at 10 Hz, 1 sec. On, 3 sec. Off, twice or repeat (For Call Waiting, twice only).</td></tr>
<tr><td>Call Waiting (Internal)</td><td>Internal call is waiting for the busy button. A call is camped-on to a busy [DN] or CO line button.</td></tr>
<tr><td>Ring Over Busy (External)</td><td>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.</td><td rowspan="2">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</td></tr>
<tr><td>Call Waiting (External)</td><td>External call is waiting for busy station. A call is camped-on to a busy [DN] or CO line button.</td></tr>
<tr><td>Volume Control - Ringing Speaker</td><td>Adjusts speaker volume for ringing state.</td><td>500/640 Hz continuous.</td></tr>
</table>

Other types of tones that do not fit in the previous categories are listed in Table 45.

Table 45 Administration/Programming Tones

Tone Name	Description	Ringing 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.

Strata Net IP and IPT 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 Strata Net IP channels and IP telephones depends heavily on the LAN parameters as shown in Table 46. This table shows the amount of bandwidth required for each Strata Net IP voice call (without data traffic) based on the interval and the CODEC.

Table 46 Strata Net IP and IPT 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] IPT 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 Prg 250-07 Prg 152-01	88kbps[3]	Excellent	Good	Fair	Poor
G.711 at 40ms i Prg 250-07 Prg 152-01	76kbps[3]	Excellent	Good	Fair	Poor
G.729A at 40ms Prg 250-07 Prg 152-01	20kbps[3]	Good	Good	Fair	Poor
G.729A at 80ms Prg 250-07 Prg 152-01	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 47 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 47 Strata 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 48.

Table 48 Strata Net IP Class Definitions

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

Table 49 shows the amount of bandwidth required for setting up and tearing down calls independent of the amount of voice traffic.

Table 49 Strata 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 IPT telephones, keep alive packets are exchanged between the IPU and the phones. This traffic amounts to 3 kbps per phone.

Strata CIX Software License Requirements

Table 50 describes the content and use for each Strata CIX license.

Table 50 Licenses for all Strata CIX Systems

New License	Description	Comments
LIC-4 BASIC[1]	4-port CO Line/Station License for CIX Systems	One required for every four ports used for CO lines, DP or SLT stations, Attendant Consoles, voice mail ports, Strata Net and ACD announcement ports beyond the ports bundled with the system processor.
LIC-4 DTMF[1]	4-port DTMF Receiver License for CIX Systems	Required for activation of four DTMF receiver ports number 5 and above on system. Maximum is determined by the CTU card in use.
LIC100-STRATA N[1]	CIX100 Strata® Net Networking Application License.	One per CIX100 system (node) required to network multiple systems using Strata® Net networking.
LIC200-STRATA N[1]	CIX200 Strata® Net Networking Application License.	One per CIX200 system (node) required to network multiple systems using Strata® Net networking.
LIC670-STRATA N[1,2]	CIX670 Strata® Net Networking Application License.	One per CIX670 system (node) required to network multiple systems using Strata® Net networking.
LIC1200STRATA N[1,2]	CIX1200 Strata® Net Networking Application License.	One per CIX1200 system (node) required to network multiple systems using Strata® Net networking.
LIC-CIX-STRN-CH	Strata Net channel license LIC-CIX-STRN-CH is only available in R4.1 version MN023 and above.	This license allows you to license a specific number of Strata Net channels on Strata CIX systems. The same license part number, LIC-CIX-STRN-CH can be used on any R4.10 (MN023) and above Strata CIX system. This license makes Strata CIX networking more competitive for the CIX system applications that require 12 or less Strata Net channels. The Strata Net system licenses LIC100-STRATA N, LIC200-STRATA N, LIC670-STRATA N, and LIC1200STRATA N are still available for applications that require more than 12 Strata Net channels. The CIX40 uses the Strata Net channel license and does not require a Strata Net system license.
LIC-CIX-SIPT-CH	SIP Trunking License, only available with R5.1MS17 and above; MIPU01-06 firmware.	This license allows you to license a specific number of SIP Trunks on Strata CIX systems. The same license part number, LIC-CIX-SIPT-CH can be used on any R5.1MS17 and above Strata CIX system.
LIC-SER PORT[1]	Serial Port License required for BSIS Ports 2, 3 and 4 for CIX.	Required to activate serial port 2, 3 and 4 in Strata CIX systems (license for serial port one included with BSIS).
(Sheet 1 of 2)		

New License	Description	Comments *(continued)*
LIC-ACD[1]	CIX ACD Server License	Required to activate ACD support in a Strata CIX system. (This license is included with ACD turnkey packages and software packages.)
LIC-SOFTIPT	License for one SoftIPT user on Strata CIX.	One required for each SoftIPT application interfaced to the Strata CIX. One MIPU port is required for each SoftIPT.
LIC-2 LVMU	LVMU two port Voice Mail license requires R4.2 MP026 or higher software.	R4.2MP026 software provides two LVMU Voice Mail port licenses by default. LIC-2 LVMU is required for 4, 6, or 8 VM ports. One Basic Port License is also required for each voice mail port.
LIC-2 GVPH	GVPH two port VM licenses requires R5.1 software or higher.	CIX40 R5.1software provides four GVPH voice mail port licenses by default. LIC-2 GVPH is required for six or eight voice mail ports.
LIC-1 DP5022SDM	DP5022-SDM telephone migration license.	One license is required on CIX100, CIX100-S, CIX200, CIX670, and CIX1200 for each DP5022-SDM. LIC-DP5022-SDM is not required for DP5022-SDM or CIX40.
LIC-CIX-IP PORT	IP Endpoint License for Strata CIX.	If the CIX software version is R5.0 or later, the IP endpoint license is used when the IPT or SoftIPT is registered. If the CIX software version is R4.x or 3.x, the IP endpoint license will be used as soon as the DN for the IPT or SoftIPT has been created.
LIC-CIX-FF[1]	FeatureFlex license for CIX.	Strata CIX license to activate FeatureFlex. Requires one per CIX system to run FeatureFlex applications.
The following license is included in some of the license packages, they are not sold separately.		
LIC-ATT[1]	CIX Attendant Console License	Required to activate Attendant Console support in a Strata CIX system. This license is included in the Attendant Console Part Number.
(Sheet 2 of 2)		

1. This license is not required on the CIX40 for Feature or Port activation, but is required for all other CIX systems.
2. All licenses applied to Strata CIX670 processors can be transferred to the CIX1200 processor.

Table 51 Type and Number of Licenses Required by Feature

		IP Interface Card	Basic Port License[1]	Strata Net System License[2]	Strata Net Channel License[3]	SoftIPT License[4]	IP Port License[5]
CIX R3.1 and above	IP ACD (App Server) announcement ports	MIPU	Each Port				
	IP Attendant Console	MIPU	Each Port				
	IP Voice Mail (App Server)	MIPU	Each Port				
	Strata Net IP System[6]	MIPU	Each Channel	per system			
	Strata Net IP Channel[3]	MIPU			Each Channel		
	IP5000-series telephones	MIPU					Each Active Station
	SIP Phone	MIPU					Each Active Station
	SoftIPT	MIPU				Each Active Station	Each Active Station

1. LIC-4 BASIC (Not required on CIX40, but is required on all other CIX systems).
2. LIC100 STRATA N; LIC200 STRATA N; or LIC670 STRATA N; Not required on CIX40
3. LIC-CIX-STRN-CH (not required when using LICxxx-STRATA N system license).
4. LIC-SOFTIPT
5. LIC-CIX-IP PORT
6. Not required when using LIC-CIX-STRN-CH for Strata Net

Capacities

The following tables contain Strata CIX40, CIX100, CIX200, CIX670, and CIX1200 capacities for stations and peripherals, CO lines, station buttons and system features. Tables are divided based on the small Strata CIX systems (CIX40, CIX100 and CIX200) and large Strata CIX systems (CIX670 and CIX1200).

Table 52 Cabinet and Slot Capacities for Strata CIX40, CIX100 and CIX200

Cabinets/Slots/Ports	CIX40 GCTU2A	CIX100 ACTU	CIX200 LCTU
Cabinets	1	1 to 2	1 to 2
Universal slots	Dedicated slots	4 or 8	3 or 7
Maximum capacity of ports including Voice Mail ports (lines + stations)	57	112	192

Table 53 Station/Peripherals System Capacities for Strata CIX40, CIX100 and CIX200

Stations	CIX40	CIX100 Base & Expansion	CIX200 Base and Expansion
Digital Add-on modules per Base Cabinet[1]	16 System 1 or 2 per DP	30 DPs with 1 DADM 23 DPs with 2 DADMs	46 DPs with 1 DADM 36 DPs with 2 DADMs
Digital Add-on modules per Expansion Cabinet[1]	N/A[2]	31 DPs with 1 DADM 24 DPs with 2 DADMs	46 DPs with 1 DADM 36 DPs with 2 DADMs
CIX Attendant consoles	2	2	2
DP5000, DKT3200, 3000 and 2000-series DKTs[1]	16	72 (40 Base Cabinet) (40/Expan. Cab.)	112
IP5000-series/IPT2000-series telephones[3]	24	64 per cabinet 72 per system	160 per System 80 per cabinet
KM5020, LM5110 and IADM2020 on IPTs[4]	48 per system 2 IADM2020 per IPT	53 per cabinet 53 per system	58 per System 58 per cabinet
DKT2204-CT or DKT2304-CT Cordless Telephone[1]	16	72	112
Door locks[5]	3	4	5
Door phone control boxes (DDCB)	2	2	3
Door phones	6	6	9
DSS consoles (DSS)	3 per system 3 per station	3 per system 3 per station	5 per system 5 per station
Off-premise stations	2[6]	56	58
BPCI used for TAPI only: per cabinet[1]	16[7]	35	48 Base Cabinet 54 Expansion Cabinet
Total Stations (Digital/Analog)	18	72	160
Standard stations[8]	2	56	58
DTMF Receivers[9]	16	16	16
Calls existing at the same time	unlimited	unlimited	96

1. Limit is based on cabinet Power Factor (PF).
2. N/A means Not Available.
3. Limited by system bus traffic.
4. Limited by the number of flexible buttons per system.
5. Each Door lock reduces the number of Door Phones by one Door Phone and vice versa.
6. OL13A and OL13B only.
7. CIX40 cannot use 2B channels on digital telephones.
8. Capacity includes standard telephones with or without Caller ID interface. Caller ID interface requires R4.1 or later software.
9. DTMF receivers are required for standard touch tone telephones, voice mail integration, Tie, DID and DISA lines.

Table 54 Line Capacities and Universal Circuit Card Slots for Strata CIX40, CIX100 and CIX200

Lines	CIX40	CIX100 Base & Expansion	CIX200 Base and Expansion
CO lines – loop start (analog - 8 lines/slot)	11 with built-in CLID	64	56
CO lines – ground start (analog - 4 lines/slot)	N/A[1]	32	28
DID lines (analog - 4 lines/slot)	N/A[1]	32	28
Tie lines (analog - 4 lines/slot)	N/A[1]	32	28
T1 lines (DS-1)[2]	N/A[1]	64	96
ISDN PRI B channel lines[3]	N/A[1]	48	96
Strata Net IP Channels[4]	24	48	96
SIP URIs per system	72	72	160
SIP Trunking	24	48	96
Total lines (Analog, T1, and ISDN PRI B channels combined)	11	64	96
Channel Groups	1 (IP)	32	48
Number of groups w/ GCO Line buttons	8	32	50
Busy Tone Detectors	16	16	16

1. N/A means Not Available.
2. T1 lines can be loop start, ground start, Tie or DID (maximum 24 lines per unit, any type or combination).
3. PRI lines provide CO line services, including Strata Net Networking, Calling Party Number/Name, DID, Tie, POTS, FX and DIT.
4. IP Strata Net channels provides CIX networking functionality. When connecting to the IPedge or VIPedge, use MIPU/GIPU only.

Table 55 Digital and IP Telephone Station Buttons for Strata CIX40, CIX100 and CIX200

Station Buttons per System	CIX40	CIX100 Base & Expansion	CIX200 Base and Expansion
Call Forward, Personal CF Buttons	24	72	160
Caller ID (CLID) button (DP or IPT only)	24	72	160
CO Line Buttons[1]	6	64	56
Group CO Line Buttons[2]	N/A	64	96
Pooled CO Line Buttons[2]	32	32	50
CO Group and Pooled Line Buttons[2]	N/A, 32	64	96
Door Unlock Buttons	24	32	64
Flexible Telephone Buttons	3200	3200	7000
Line and DN Buttons in use at the same time	96	1440	3360
Message Waiting Registration (DNs with MW)	130	130	230
Multiple Appearances of DNs on Telephones	2300	2300	4200
Night Transfer Buttons	24	32	64
One Touch Buttons	1600	1600	3500
Primary Directory Numbers [PDNs] per system	40	72	160
Phantom Directory Numbers [PhDNs] per system	288	288	640
[PhDNs] with Message Waiting Indication LED	18	18	38

1. This is the number of unique CO Line Buttons (i.e., Line 1, Line 2, etc.). The total number of CO Line Buttons can not exceed the Flexible Telephone Button limit. Example: If Line 1 button appears on 10 telephones, it counts as one button.
2. 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 56 System Feature Capacities for Strata CIX40, CIX100 and CIX200

Features	CIX40	CIX100 Base & Expansion	CIX200 Base and Expansion
Pilot DNs	90	90	200
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	4	4	10
Call Park Orbits (General)	14	14	32
Call Park Orbits (Individual)	56	56	96
Minimum / Maximum Caller ID per Station	Minimum: 10; Maximum: 100		
Maximum number of Stations that can have Caller ID/ANI/ CNIS Numbers stored (Call History records)	66	66	100
	Up to 660/system	Up to 660/ system	Up to 1000/ system
CO Line Groups - Incoming Line Groups (ILG)	11	32	50
CO Line Groups - Outgoing Line Groups (OLG)	11	32	50
Outgoing Line Groups (OLG) Members per system (Trunks + ISDN Line Service Index)	96 (No ISDN)	96	144

Table 56 System Feature Capacities for Strata CIX40, CIX100 and CIX200 *(continued)*

Features	CIX40	CIX100 Base & Expansion	CIX200 Base and Expansion
Conference Circuits	64	64	64
Conferencing (three-parties simultaneously)[2]	20	20	20
Conferencing (eight-parties simultaneously)[2]	8	8	8
Conference Party types (up to 8 total lines + stations)[2]	6 lines max. 8 stations max.	6 lines max. 8 stations max.	6 lines max. 8 stations max.
Two-CO Line simultaneous Connection[2] (Two party only, no telephone or VM port)	5	32	48
Conference/Line Volume Adjustment (PAD) Groups	6	6	10
DID Numbers for Calling Number ID/system	N/A[3]	225	500
DNIS/DID Network Routing Numbers (8~32 digits)	N/A[3]	200	400
DNIS/DID Numbers (total 4~7 digits)	N/A[3]	450	1000
DNIS/DID Numbers (4~7 and 8~32 digits)	N/A	450	1000
Network DNs	3000	3000	3000
DTMF Receivers[4]	16	16	16
E911 Groups	8	8	8
Emergency Call Groups	8	8	8
Hunt Groups (Serial/Circular/Distributed combined)	16	90	200
Hunt Group Size (DNs per group)	18	72	160
Hunt Group Stations (per system)	18	360	800
ISDN Line Service Indexes	N/A	32	48
Multiple Call Ring Group	16	16	32
Night Bell Control Relay per tenant[5]	1	1	1
Night Transfer Control Relay per tenant[5]	1	1	1
Off-hook Call Announce Handsets (simultaneous)	20	20	20
Off-hook Call Announce to Telephone Speakers[6]	23 IPT shared 12 IPT dedicated	72	112
Page Mute External BGM Control Relay[5]	1	1	1
Page Zone Relays[5]	N/A	4	8
Page Groups (Phones with or without External Zones)	4	4	8
Paging – (Group Page – simultaneous stations paged)	24	72	120
Pickup Groups	5	5	10
Ring Tones (External Call Ring Tones for DPs and IPTs)	10	10	10
Ring Tones (Internal Call Ring Tones for DPs and IPTs)	10	10	10
Speed Dial - Station SD numbers per system[7]	1080	1080	2400
Speed Dial - System SD numbers per system	800	800	800
GVPH / LVMU systems per system[8]	1	1	1
Tenants	8	8	8
Destination Restriction Level (DRL) Classes	16	16	16
Verified Account Codes	135	135	300
Voice Mail SMDI Interface[1]	1	1	1
CSTA Device Monitors	512	512	512
CSTA Call Monitors	320	320	320

1. SMDI and SMDR may require BSIS serial port or LAN interface.
2. Conference circuits 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 circuits. Each conference can have up to eight members. Two CO line connections do not require a conference circuit.
3. N/A means Not Available
4. DTMF receivers are required for standard touch tone telephones, voice mail integration, Tie, DID and DISA lines.
5. An option BIOU is required for up to four zone page relays and four control relays on the CIX100, CIX200 and CIX670 processor. One control relay and one 600 ohm page output is provided on board the CIXCIX100 and CIX200 processor but not the CIX670 processor.
6. On Digital telephones Speaker OCA capacity is determined by 2B channel slot availability and power supply. CIX40 supports Speaker OCA (S-OCA) on IP5000-series and IPT2000-series telephones only and not on digital telephones. S-OCA requires the DOCA-1A option in DP5000-series digital telephones, but not in IP telephones. Speaker OCA on IP telephones require an MIPU / GIPU installed in the Strata CIX. Each IP telephone with Speaker OCA requires two IP channels on the MIPU / GIPU but only requires one IP endpoint license. The IP OCA channel is only used while the S-OCA call is in progress so it can be reserved or shared with other telephones for S-OCA but cannot be used as a dedicated IP channel for another telephone or Strata Net channel. DP5008 and IP2000-series telephones do not support Speaker OCA; all other current Toshiba telephones support S-OCA.
7. Up to 100 Station SD numbers, allocated in increments of 10, can be programmed per station.
8. CIX40 supports only the GVPH voice mail systems internally.

Table 57 Strata CIX1200 and CIX670 Cabinet and Slot Capacities

Cabinets/Slots/Ports	CIX1200 EXP. B HCTU+HEXBU	CIX1200 EXP. A HCTU+HEXAU[1]	CIX1200 BASIC HCTU	CIX670 Expanded Processor BCTU + BEXU	CIX670 Basic Processor BCTU
Cabinets	1 to 12	1 to 7	1 to 2	1 to 7	1 to 2
Universal slots	8 to 118 [2]	8 to 68	8 to 18	8 to 68	8 or 18
Maximum capacity of ports including Voice Mail ports (lines + stations)	1152	672	192	672	192

1. The port and cabinet capacity of HCTU/HEXAU is identical to CIX670 BCTU/BEXU, however, the memory capacity for numerous features has been expanded with the HCTU/HEXAU - for details see the following tables.
2. For 8 to 12 cabinet systems, RRCU cards will require up to 3 card slots in a base cabinet stack and up to three slots in expansion cabinet stack. Up to a system total of six card slots.

Table 58 Strata CIX1200 and CIX670 Station/Peripherals System Capacities

Stations	CIX1200 EXP. B **HCTU+HEXBU**	CIX1200 EXP. A **HCTU+HEXA**	CIX1200 BASIC **HCTU**	**CIX670 Expanded Processor BCTU + BEXU**	**CIX670 Basic Processor BCTU**
DKT2000, 3000, 3200, 5000 series[1]	952	552	152	552	152
DKT2000, 3000, 3200, 5000 series per cabinet	72 (Base) 80 (Exp)	72 (Base) 80 (Exp)	72 (Base) 80 (Exp)	72 (Base) 80 (Exp)	72 (Base) 80 (Exp)
IPT telephones[2]	80 (Base) 80 (Exp) 1000 System	80 (Base) 80 (Exp) 560 System	80 (Base) 80 (Exp) 160 System	80 (Base) 80 (Exp) 560 System	80 (Base) 80 (Exp) 160 System
DAMA per Base cabinet[1]	55 DPs with 1 ADM 43 DPs with 2 ADMs	55 DPs with 1 ADM 43 DPs with 2 ADMs	57 DPs with 1 ADM 43 DPs with 2 ADMs	55 DPs with 1 DADM 43 DPs with 2 DADMs	55 DPs with 1 DADM 43 DPs with 2 DADMs
DADM per Expansion cabinet[1,‘]	57 DPs with 1 ADM 45 DPs with 2 ADMs	57 DPs with 1 ADM 45 DPs with 2 ADMs	55 DPs with 1 ADM 45 DPs with 2 ADMs	57 DPs with 1 DADM 45 DPs with 2 DADMs	57 DPs with 1 DADM 45 DPs with 2 DADMs
IADM on IPT[3]	84/cabinet	80/cabinet	80/cabinet	80/cabinet	80/cabinet
IADM System Capacity	800/system	400/system	116/system	400/system	116/system
DKT2204-CT or 2304-CT	952	552	152	552	152
CIX Attendant Console	6	6	6	4	2
Door locks[4]	10	10	5	10	5
Door phone control boxes (DDCB)	8	8	3	8	3
Door phones	24	24	9	24	9
DSS consoles (DSS)	24 per System 8 per Station	24 per System 8 per Station	5 per System 5 per Station	15 per System 8 per Station	5 per System 5 per Station
Off-premise stations	944	544	144	544	144
BPCI used for TAPI only per cabinet	66	66	66	66	66
Total Stations (Digital/ Analog/IP combined)	1000	560	160	560	160
Standard Station	944	544	144	544	144
DTMF Receivers[5]	48	32	16	32	16
Calls existing at the same time	576	336	96	336	96

1. Limited by power factor
2. Limited by system traffic
3. Limited by the number of flexible buttons per system
4. Each Door lock reduces the number of Door Phones by one Door Phone and vice versa.
5. DTMF receivers are required for standard touch tone telephones, voice mail integration, Tie, DID and DISA lines.

Table 59 Strata CIX1200 and CIX670 Line Capacities and Universal Circuit Card Slots

Lines	CIX1200 EXP. B **HEXBU**	CIX1200 EXP. A **HEXAU**	CIX1200 BASIC **HCTU Only**	**CIX670 Expanded Processor BCTU + BEXU**	**CIX670 Basic Processor BCTU**
CO lines – loop start (analog - 8 lines/slot)	440	264	96	264	96
CO lines – ground start (analog - 4 lines/slot)	440	264	72	264	72
DID lines (analog - 4 lines/slot)	440	264	72	264	72
Tie lines (analog - 4 lines/slot)	440	264	72	264	72
T1 lines (DS-1)[1]	440	264	96	264	96
ISDN PRI B channel lines[2]	440	264	96	264	96
Strata Net IP Channels[3]	440	264	96	264	96
SIP Trunks	440	264	96	264	96
SIP URI per system[4]	1000	560	160	560	160
SIP Trunk Service Index	128	128	48	128	48
Total lines (Analog, T1, SIP Trunks, Strata Net, and ISDN PRI B channels combined)	440	264	96	264	96
Channel Groups	220	128	48	128	48
Number of groups w/ GCO Line buttons	220	128	50	128	50
Busy Tone Detectors	48	32	16	32	16

1. T1 lines can be loop start, ground start, Tie or DID (maximum 24 lines per unit, any type or combination).
2. PRI lines provide CO line services, including Strata Net Networking, Calling Party Number/Name, DID, Tie, POTS, FX and DIT.
3. IP Strata Net channels provides Strata CIX networking functionality.
4. 160 URIs per MIPU.

Table 60 Strata CIX1200 and CIX670 Digital and IP Telephone Station Buttons

Station Buttons per System	CIX1200 EXP. B **HEXBU**	CIX1200 EXP. A **HEXAU**	CIX1200 BASIC **HCTU Only**	**CIX670 Expanded Processor BCTU + BEXU**	**CIX670 Basic Processor BCTU**
Call Forward, Personal CF Buttons	1000	560	160	560	160
Caller ID (CLID) button (DP or IPT only)	1000	560	160	560	160
CO Line Buttons[1]	440	264	96	264	96
Group CO Line Buttons[2]	440	264	96	264	96
Pooled CO Line Buttons[2]	220	128	50	128	50
CO Group and Pooled Line Buttons[2]	440	264	96	264	96
Door Unlock Buttons	96	96	64	64	64
Flexible Telephone Buttons	48000	48000	7000	24000	7000

Table 60 Strata CIX1200 and CIX670 Digital and IP Telephone Station Buttons *(continued)*

Station Buttons per System	CIX1200 EXP. B **HEXBU**	CIX1200 EXP. A **HEXAU**	CIX1200 BASIC **HCTU Only**	**CIX670 Expanded Processor BCTU + BEXU**	**CIX670 Basic Processor BCTU**
Line and DN Buttons in use at the same time	6000	6000	3360	3360	3360
Message Waiting Registration (DNs with MW)	1344	1344	230	800	230
Multiple Appearances of DNs on Telephones	27000	27000	4200	15000	4200
Night Transfer Buttons	192	192	64	128	64
One Touch Buttons	24000	24000	3500	12000	3500
Primary Directory Numbers [PDNs] per system	1000	560	160	560	160
Phantom Directory Numbers [PhDNs] per system	4000	4000	640	2240	640
[PhDNs] with Message Waiting Indication LED	192	192	38	128	38
ISDN DNs - System Capacity	768	768	224	768	224
ISDN DNs - Station Capacity (per station)	8	8	8	8	8

1. This is the number of unique CO Line Buttons (i.e., Line 1, Line 2, etc.). The total number of CO Line Buttons can not exceed the Flexible Telephone Button limit. Example: If Line 1 button appears on 10 telephones, it counts as one button.
2. 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 61 Strata CIX1200 and CIX670 System Feature Capacities

Features	CIX1200 EXP. B HEXBU	CIX1200 EXP. A HEXAU	CIX1200 Basic HCTU Only	CIX670 Expanded Processor BCTU + BEXU	CIX670 Basic Processor BCTU
Pilot DNs	256	256	200	256	100
Advisory LCD Messages (Set on a Telephone)	1	1	1	1	1
Advisory LCD Messages Lists (per System)	10	10	10	10	10
Attendant Groups	1	1	1	1	1
Call Accounting SMDR Interface[1]	1	1	1	1	1
Call Forward, System CF Patterns	48	48	10	32	10
Call Park Orbits (General)	96	96	32	64	32
Call Park Orbits (Individual)	576	336	96	336	96
Minimum / Maximum Caller ID per Station	Minimum: 10 Maximum: 100				
Maximum number of Stations that can have Caller ID/ANI/CNIS Numbers stored (Call History records)	600	600	200	200	100
	3000	3000	1000	Up to 2000/ system	Up to 1000/ system
CO Line Groups - Incoming Line Groups (ILG)	220	128	50	128	50

Table 61 Strata CIX1200 and CIX670 System Feature Capacities *(continued)*

Features	CIX1200 EXP. B HEXBU	CIX1200 EXP. A HEXAU	CIX1200 Basic HCTU Only	CIX670 Expanded Processor BCTU + BEXU	CIX670 Basic Processor BCTU
CO Line Groups - Outgoing Line Groups (OLG)	220	128	50	128	50
Outgoing Line Groups (OLG) Members per system (Trunks + ISDN Line Service Index)	660	392	144	392	144
Conference Circuits	128	96	64	96	64
Conferencing (three-parties simultaneously)[2]	40	30	20	30	20
Conferencing (eight-parties simultaneously)[2]	16	12	8	12	8
Conference Party types (up to 8 total lines + stations)[2]	6 lines max. 8 stations max	6 lines max. 8 stations max	6 lines max. 8 stations max	6 lines max. 8 stations max.	6 lines max. 8 stations max.
Two-CO Line simultaneous Connection[2] (Two party only, no telephone or VM port)	220	132	48	132	48
Conference/Line Volume Adjustment (PAD) Groups	32	32	10	32	10
DID Numbers for Calling Number ID/system	1500	1500	500	1000	500
DNIS/DID Network Routing Numbers (8~32 digits)	1500	1500	400	1000	400
DNIS/DID Numbers (4~7 digits)	3000	3000	1000	2000	1000
Network DNs	6000	6000	6000	3000	3000
DTMF Receivers[3] and Busy Tone Detectors	48	32	16	32	16
E911 Groups	8	8	8	8	8
Emergency Call Groups	8	8	8	8	8
Hunt Groups (Serial/Circular/Distributed combined)	1100	1100	200	640	200
Hunt Group Size (DNs per group)	1000	560	160	560	160
Hunt Group Stations (per system)	5000	5000	800	2800	800
ISDN Line Service Indexes	220	128	48	128	48
Multiple Call Ring Group	96	96	32	64	32
Night Bell Control Relay per tenant[4]	1	1	1	1	1
Night Transfer Control Relay per tenant[4]	1	1	1	1	1
Off-hook Call Announce Handsets (simultaneous)	40	30	20	30	20
Off-hook Call Announce to Telephone Speakers[5]	592	352	112	352	112
Page Mute External BGM Control Relay[4]	1	1	1	1	1
Page Zone Relays[4]	8	8	8	8	8
Page Groups (Phones with or without External Zones)	24	24	8	8	8
Paging – (Group Page – simultaneous stations paged)	120	120	120	120	120
Pickup Groups	48	48	10	32	10
Ring Tones (External Call Ring Tones for DPs and IPTs)	10	10	10	10	10

Table 61 Strata CIX1200 and CIX670 System Feature Capacities *(continued)*

Features	CIX1200 EXP. B HEXBU	CIX1200 EXP. A HEXAU	CIX1200 Basic HCTU Only	CIX670 Expanded Processor BCTU + BEXU	CIX670 Basic Processor BCTU
Ring Tones (Internal Call Ring Tones for DPs and IPTs)	10	10	10	10	10
Speed Dial - Station SD numbers per system[6]	8400	8400	2400	5600	2400
Speed Dial - Station SD numbers per Station (allocated in increments of 10)	Min: 0 Max: 100	Min: 0 Max: 100	Min: 0 Max: 100	Min: 0 Max: 100	Min: 0 Max: 100
Maximum stations with SD per system	840	840	240	560	240
Speed Dial - System SD numbers per system	800	800	800	800	800
Tenants	8	8	8	8	8
Destination Restriction Level (DRL) Classes	16	16	16	16	16
Verified Account Codes	4000	4000	3000	1000	300
Voice Mail SMDI Interface[1]	1	1	1	1	1
External Ring Repeat - to available stations	256	256	All	All	All
LCR Exception Table Size (Programs 521 and 522)	2500	2500	2500	1280	1280
Maximum number of digits in the LCR Modified Digits Table	19	19	19	19	19
Maximum number of LCR Route Plans	128	128	128	64	64
CSTA Device Monitors[7]	1152	1152	1152	512	512
CSTA Call Monitors[8]	560	560	560	320	320

1. SMDI and SMDR may require BSIS serial port or LAN interface.
2. Conference circuits 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 circuits. Each conference can have up to eight members. Two CO line connections do not require a conference circuit.
3. DTMF receivers are required for standard touch tone telephones, voice mail integration, Tie, DID and DISA lines.
4. An option BIOU is required for up to four zone page relays and four control relays on the CIX670 and CIX1200 processor.
5. On Digital telephones Speaker OCA capacity is determined by 2B channel slot availability and power supply. S-OCA requires the DOCA-1A option in DP5000-series digital telephones, but not in IP telephones. Speaker OCA on IP telephones require an MIPU installed in the Strata CIX. Each IP telephone with Speaker OCA requires two IP channels on the MIPU but only requires one IP endpoint license. The IP OCA channel is only used while the S-OCA call is in progress so it can be reserved or shared with other telephones for S-OCA but cannot be used as a dedicated IP channel for another telephone or Strata Net channel. DP5008 and IPT1020-SD and IP2000-series telephones do not support Speaker OCA; all other current Toshiba telephones support S-OCT.
6. Up to 100 Station SD numbers, allocated in increments of 10, can be programmed per station.
7. See "Device Monitor Capacities for Strata CIX Systems" on page 195.
8. See "Device Monitor Capacities for Strata CIX Systems" on page 195.

Device Monitor Capacities for Strata CIX Systems

Applications including Strata ACD, SCM, Tracer, Taske, VCS, FeatureFlex, and System TAPI send requests to the Strata CIX 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 Strata CIX system. These requests can produce a heavy load on the CIX processor 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 expanded and non-expanded CIX processor configurations.

- Total number of devices that can be monitored: CIX1200 = 1152; CIX40/100/200/670 = 512
- Total number of simultaneous device monitor calls: CIX1200 = 560; CIX40/100/200/670 = 320

Table 62 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 SCM and/or Tracer or both.	1 CSTA device monitor per agent or supervisor.
3	Normal User with SCM and/or Tracer or both.	1 CSTA device monitor per user.
4	ACD Groups.	1 CSTA device monitor per group.
5	ACD Queue Announcement ports.	1 CSTA device monitor per port.
6	Extensions, Analog and T1 trunks/channels to be monitored by Taske.	1 CSTA device monitor each.
7	Attendant Consoles	1 CSTA device monitor per console.
8	VCS users	1 CSTA device monitor per VCS user.
9	Feature Flex users	1 CSTA device monitor when user is using one or more FF application.
10	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.		

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Index

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