



# Testing Report

## Pingtel SIPxchange SIP PBX Test for Internet Telephony Magazine A TMC Publication

August, 2005

### Overview

CT Labs was asked by Technology Marketing Corporation (TMC) ([www.tmcnet.com](http://www.tmcnet.com)) to evaluate the Pingtel SIPxchange PBX product. CT Labs performed the tests and evaluation on the Pingtel SIPxchange PBX in the CT Labs Rocklin, CA facility, concluding the effort with this report. This report is intended for publication in Internet Telephony Magazine (in abbreviated form) as well as the popular TMC website. This report summarizes the specific tests and evaluations that were conducted for the Pingtel SIPxchange PBX.

### 1 Executive Summary

The SIPxchange PBX product was found to be easy to install as supplied to us in its typical form as a turn-key solution. CT Labs was able to set the system up and get it working with the pre-configured desktop phones in about 30 minutes. It was slightly more difficult, however, to get the SIP softphone to register with the SIPxchange server. Overall, though, the installation was found to be straight-forward.

The SIPxchange features were found to be comprehensive for a basic SIP PBX. While this was not the focus of this test, we understand that the next release (due out in October) will include many new features, including Meet-Me Conferencing and ACD features.

The documentation and integrated help were found to be well-organized, easy to comprehend, and provided much helpful information. The administrative and client GUIs were easy to use with a simple structure and clearly-labeled options.

Our manual testing of features and functionality went well, with calls between the variety of supplied desktop and softphones connecting properly and providing a high level of voice quality.

The excellent performance results from our automated call-handling load tests showed that the SIPxchange PBX can easily handle heavy call loads of station-to-station as well as trunk-to-station calls.

Overall, the CT Labs found the Pingtel SIPxchange to be a solid, dependable system that is easy to install and use, and provides a solid level of performance.

## 1.1 Product Review Results Grid

The following are the results of the CT Labs evaluation of the Pingtel SIPxchange PBX:

**Table 1: Review Results Grid**

<b>Product Evaluation Score</b> (each rated out of a possible 10)	<b>Product Score</b>	<b>Relative Weight (%)</b>	<b>Weighted Score</b>
Ease of installation & configuration	8.5	10	0.85
Features evaluation	8.7	10	0.87
Product documentation, online help	8.9	10	0.89
Graphical user interfaces (admin and end-user)	8.9	20	1.78
Manual product exercise	9.4	25	2.35
Automated call-handling load test (no VM)	9.6	20	1.92
Technical Support	9.6	5	0.48
<b>Total</b>		<b>100</b>	<b>9.14</b>

The weighted score is calculated by multiplying the score for a given category by the relative weight (e.g. a score of 6 times a weight of 10% would yield a weighted score of 0.6). Note that the maximum weighted score for a given product for the sum of all evaluation categories is 10.

## 2 General Review Test Information

### 2.1 General goals of the evaluation

This section lists the "guiding light" questions that we are attempting to answer as a result of performing the test procedures and analysis for this product:

1. How easy is it to install and set up (a) the PBX server components, (b) a client workstation, and (c) an administrative console?
2. How easy is this system to administer and maintain from (a) a local console, and (b) a remote console?
3. How easy-to-use are the product's graphical user interface programs (administrative as well as client programs, if available)?
4. How do the integrated voice messaging and auto attendant features compare with typical enterprise voice mail systems in terms of feature depth?
5. How do the supported SIP Phone devices perform with respect to perceived speech quality and call handling?
6. How effective and helpful is the product's documentation package?
7. How effective and helpful is the product's on-line help for the product's graphical user interface programs?

### 2.2 Vendor equipment provided

The vendor was asked to provide CT Labs with the following:

- A fully turnkey solution with processor and memory support chosen to correctly match the production requirements for the trunk and station line density of the unit provided.
- Support for standard auto attendant and voice mail features.
- At least three desktop SIP phones and one Pingtel softphone.
- **Telephone line support requirements:**
  - **Trunk-side lines.** Systems with trunk line support for T-1 CAS (robbed bit).
  - **Station-side lines.** Support for a minimum of 32 stations.
- Provide CT Labs with optional installation and setup assistance by a trained system installer for one to two business days.

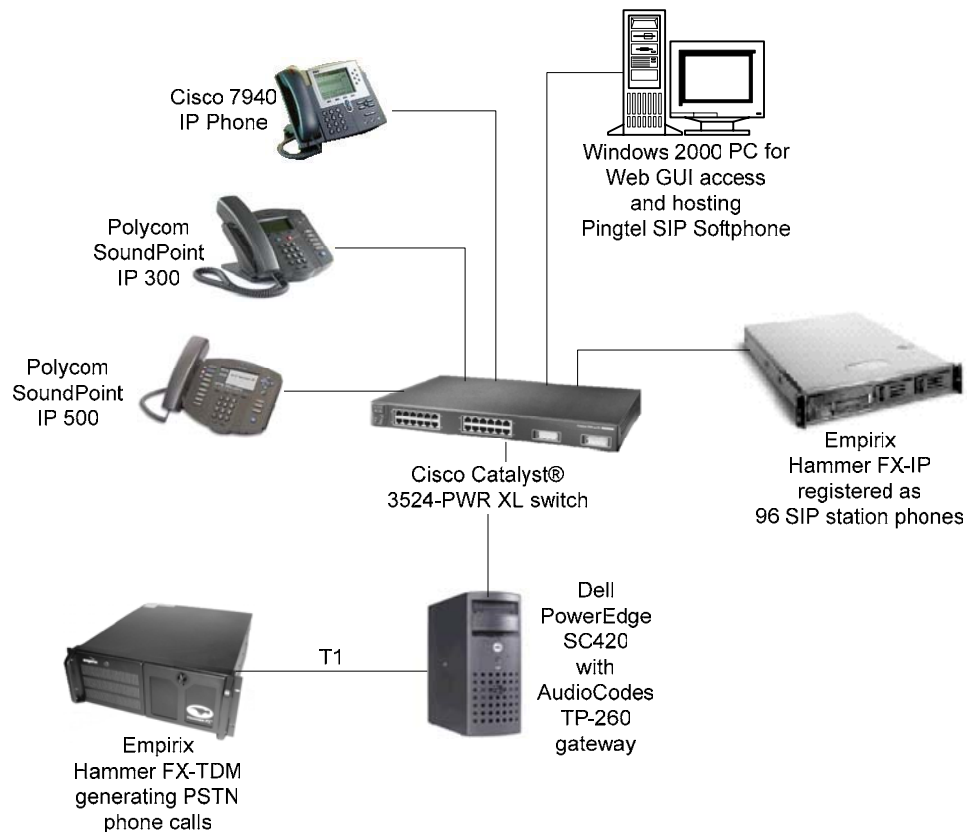
### 2.3 The vendor was invited to the lab

The vendor was invited to bring their experts here to install and configure their system. Prior to this test the vendor was allowed to review the test and scoring methodology document that defined the tests that CT Labs intended to perform for this review. Pingtel decided not to send a representative to CT Labs, since they provided us with a turnkey system as well as a dedicated support contact available by phone.

### 3 Test Setup Description

Pingtel supplied CT Labs with a turnkey SIPxchange system installed on a Dell PowerEdge SC420. This was connected to a Cisco Catalyst 3524-PWR XL switch which was used for connecting all the internal phones and our workstation (used to access the administrative and client web-based GUIs). An AudioCodes TP260 gateway card was already installed in the Dell server to provide us with T1 trunks using ISDN PRI signaling.

For the automated call-handling tests, an Empirix Hammer FX-TDM was connected via T1 to the AudioCodes gateway on the Dell server so that the Hammer FX-TDM could place PSTN calls into the system. An Empirix Hammer FX-IP was also connected to the system via Ethernet through the Cisco switch and registered as 96 SIP station phones. For the station-to-station tests, the Hammer FX-IP used 48 of its registered SIP station channels to call the other 48 registered SIP station channels on the Hammer. For the trunk-to-station tests, the Hammer FX-TDM placed PSTN calls into 23 of the T1 trunk lines on the system and placed calls to the 96 registered SIP station channels on the Hammer FX-IP.



**Figure 1 – Pingtel SIPxchange PBX Test Setup**

### 3.1 Equipment Tested

The following is model number and revision information about the components that were provided to CT Labs for this test:

**Table 2: Equipment Tested**

<b>Item</b>	<b>Details</b>
<b>Product Name</b>	SIPxchange
<b>Product Server</b>	Dell PowerEdge SC420 Intel P4 CPU, 2.80GHz 1GB RAM 30GB hard drive
<b>Software Version Number</b>	2.8.0 Patches 1-4, 10-12, 20-21, 30-38, 51-52, 60-61, 71-72, 80-81
<b>Desktop Phones</b>	1 – Cisco 7940 IP Phone 1 – Polycom SoundPoint IP500 SIP 1 – Polycom SoundPoint IP300 SIP
<b>Softphones</b>	1 – Pingtel SIP Softphone Version 2.4.3
<b>PSTN – SIP Gateway (Trunk)</b>	AudioCodes TP-260

## 4 Evaluation Results

The remainder of this report presents the results of tests and analysis performed on the Pingtel SIPxchange PBX.

### 4.1 Installation and Configuration Evaluation

Pingtel provided a turnkey system with all the components of the SIPxchange product installed on a single Dell PowerEdge SC 420 server. A total of 30 minutes was spent connecting up all the items shown in the test setup diagram (refer to Figure 1 in Section 3 above). From there the server was started up and station calls were placed between the pre-assigned Cisco and Polycom desktop phones. About an hour was then spent installing the Pingtel SIP softphone and registering it with the Pingtel server. While it was simple to install and set up the Pingtel SIP softphone for peer-to-peer mode, it took longer to get it set up as a registered phone on the Pingtel system. After trying to accomplish this using the Pingtel SIP Softphone documentation, CT Labs had to ultimately contact our Pingtel support representative. The rep then walked us through the registration process using the web-based interface on the softphone itself, which was not an option that we saw in the documentation.

Pingtel noted that the Installation Guide is mainly used by resellers and system integrators, although some large enterprise customers prefer to install the software on their own networked Linux servers.

Overall, we rated the installation and configuration of the SIPxchange an “8.5.”

### 4.2 Product Features

This section lists a series of key features as submitted by Pingtel for their system. Not all features were tested by CT Labs due to time limitations. For information on features that *were* tested, please see the Manual Product Exercise Results section of this report. The feature score is based on CT Labs’ overall experience and expectations with SIP PBX products. Additional credit is given for features that were considered unique or innovative. If features are missing that are considered “core”, the score is decreased proportional to that feature’s importance.

**Table 3: Product Feature Grid**

Features	Pingtel SIPxchange PBX
<b>System</b>	
Aliasing facility	<ul style="list-style-type: none"> <li>• Associates alternate names or extensions with primary extensions. (Similar to email aliases.)</li> <li>• Functions as part of the address lookup &amp; routing function within the Comm Server</li> <li>• Configured via the Config Server browser interface</li> </ul>
Automatic Route Selection	<ul style="list-style-type: none"> <li>• Re-writes SIP URLs to specify the “next hop” (or destination) for a SIP message that has been received by the Comm Server</li> <li>• Can be used to direct messages to different IP / PSTN trunk gateways (either on premises or at remote premise location) based on any portion of SIP URL (including E.164 number)</li> <li>• Can be used to route messages to different PSTN trunks if IP/PSTN trunk gateway provides capability to map URL patterns to specific PSTN trunks (3P)</li> <li>• Implemented by Comm Server (URI mapping rules)</li> </ul>

Features	Pingtel SIPxchange PBX
Auto-restart SIPxchange services after power failure using watchdog facility	<ul style="list-style-type: none"> <li>• Single SIPxchange application can start all other application processes associated with starting up SIPxchange (including “dependent” processes that must be started in a particular order)</li> <li>• Accessed via Comm Server browser, or automatically invoked as part of rc.* startup files in Linux</li> </ul>
Call Admission Control	<ul style="list-style-type: none"> <li>• Controls whether a given user or phone is permitted complete the call being dialed</li> <li>• Success can depend on destination (E.164 number or SIP URL) or on message containing valid SIP credentials</li> <li>• Implemented within the Comm Server (Auth Proxy)</li> </ul>
Dynamic call forwarding	<ul style="list-style-type: none"> <li>• End-users use browser interface designate another system station to which a call can be forwarded on busy/no-answer</li> <li>• Comm Server settings indicated via browser interface provided by Config Server</li> </ul>
Flexible automated bandwidth selection	<ul style="list-style-type: none"> <li>• Phones can be configured to utilize a preferred, ordered list of audio bandwidth compression codecs on a per-phone and per-group basis</li> <li>• Done by setting the preferred order of voice encoding methods (codec) via the Config server; stations negotiate codecs in indicated preferential order. (3P)</li> </ul>
Hotline / ring-down	<ul style="list-style-type: none"> <li>• Lifting a handset off the cradle causes an immediate connection to a preconfigured E.164 (or system) number</li> <li>• Implemented on SIP Phone</li> </ul>
Hunt group	<ul style="list-style-type: none"> <li>• Enables busy/no-answer forwarding to a preconfigured, sequential list of station numbers</li> <li>• Implemented by Comm Server (URI mapping rules)</li> </ul>
Message waiting indication	<ul style="list-style-type: none"> <li>• Provides visual indication on phones of voicemail messages waiting</li> <li>• Phones provide both MWI light as well as LCD display of message count (3P). Some 3rd-party station gateways may support stutter dial-tone for message waiting indication (3P).</li> </ul>
Multi-site / multi-location station and gateway	<ul style="list-style-type: none"> <li>• Individual or groups of phones configurable with separate dial plans to permit single server cluster to support multiple offices and separate gateways in several locations</li> <li>• Normal multi-gateway operation can also direct call to PSTN to “local” or “remote” gateways as determined by dialed string, enabling toll avoidance using private IP network</li> <li>• Station to station calls between separate geographical locations yet within private IP network “stay on” IP network</li> <li>• Implemented by Comm Server (URI mapping rules)</li> </ul>
Multi-station appearance	<ul style="list-style-type: none"> <li>• Calls to one station number can ring at multiple stations</li> <li>• Also known as “Boss / admin” feature</li> <li>• Display indicates the number that is ringing or in use (3P)</li> <li>• After answering, calls can be transferred to other stations, including station number originally dialed (3P)</li> <li>• Implemented by Comm Server</li> </ul>
Off-premises stations	<ul style="list-style-type: none"> <li>• Centralized server can provide call setup services for stations in any geographical location connected via IP</li> <li>• Off-premises stations can use either centralized IP/PSTN trunk gateways or gateways local to that station</li> <li>• Implemented by Comm Server; station configuration by Config Server; gateway routing by URI mapping rules</li> </ul>
Outbound call blocking	<ul style="list-style-type: none"> <li>• Calls from stations to PSTN numbers, or classes of numbers, can be prevented based on:</li> <li>• Destination of call (e.g. user or device cannot initiate an international long distance call)</li> <li>• Source of call (e.g. lobby phone can only initiate calls to internal numbers)</li> <li>• Implemented by Comm Server</li> </ul>
<b>Auto-Attendant</b>	
Confirmation of alpha-dialed number via .wav file for each users name	<ul style="list-style-type: none"> <li>• Plays either name recorded by end user, or confirms the extension number that was dialed (if no name recorded by end user)</li> </ul>
Customizable auto attendant message for main greeting	
Custom IVR Menus with VXML	
Dial by extension	
Dial by name	
Operator escape from anywhere	

Features	Pingtel SIPxchange PBX
<b>Voice Messaging</b>	
.wav file messages	<ul style="list-style-type: none"> <li>• Audio messages stored in standard encoding format</li> <li>• Played on PC using user's normal .wav file player (i.e. Windows Media Player, WinAmp, iTunes) when reviewing via browser</li> <li>• Implemented in Media server and end-user PC desktop</li> </ul>
Browser-based interface	<ul style="list-style-type: none"> <li>• Used by end-users for retrieving, and managing messages, mailbox options and greetings</li> <li>• Supported browsers:               <ul style="list-style-type: none"> <li>• Microsoft Internet Explorer 6.0 and higher</li> <li>• Netscape 6.2 and higher</li> </ul> </li> <li>• Implemented in Config and Media servers</li> </ul>
Editable message headers	<ul style="list-style-type: none"> <li>• Subject line can be edited by users and saved</li> <li>• Implemented in Config and Media servers</li> </ul>
Email notification of new voicemail messages	<ul style="list-style-type: none"> <li>• User-settable option to include message as .wav file attachment to email</li> <li>• Web links in email for reviewing message, deleting message, or launching browser to file message in a specified folder</li> </ul>
Folders for message organization	<ul style="list-style-type: none"> <li>• Messages can be organized into folders using browser interface</li> <li>• User created folders in folder hierarchy</li> <li>• Implemented in Config and Media servers</li> </ul>
https-based message storage	<ul style="list-style-type: none"> <li>• Stores and retrieves messages to/from a HTTPS server</li> <li>• Storage size limited only by capacity of HTTPS server disk</li> </ul>
Multiple user customizable voicemail greetings	<ul style="list-style-type: none"> <li>• Users can record separate greetings for, and change active greeting to, any of the following cases:               <ul style="list-style-type: none"> <li>• End-user standard greeting</li> <li>• Out of office greeting</li> <li>• Extended absence greeting</li> <li>• Recording of personal name for use with default system greeting for voicemail</li> </ul> </li> <li>• Implemented in Config and Media servers</li> </ul>
Operator escape from anywhere	<ul style="list-style-type: none"> <li>• Escapes to pre-designated extension</li> <li>• Implemented in Media server</li> </ul>
<b>Administrative Interface</b>	
Browser-based configuration system	<ul style="list-style-type: none"> <li>• Used to configure phones, users, and system functions</li> <li>• Also provides system start/stop/status controls and views</li> <li>• Config Server supplies config files for use by other system elements. Any endpoint (station, gateway) that supports the IETF draft proposal for SIP configuration protocol standard can be managed via this interface. (3P)</li> </ul>
Web services APIs for Config server	<ul style="list-style-type: none"> <li>• Initial, and ongoing user and device configuration can be performed through web service APIs</li> <li>• APIs provide mechanism to link Config server with corporate directories, e.g. LDAP, custom SQL servers</li> <li>• Sample Java code provided for "batch import" of CSV file containing user and device information</li> </ul>
Manual Back-up and restore	<ul style="list-style-type: none"> <li>• Back up configuration, users and voicemail through web interface.</li> <li>• Each component can be backed up individually</li> </ul>
<b>Client Interface</b>	
Answer / release	<ul style="list-style-type: none"> <li>• Enables call handling by pressing soft key to enter speakerphone mode to answer incoming call, and to hang up on call using soft button. (No handset use required.)</li> <li>• Implemented by phones (3P)</li> </ul>
Browser-based user interface	<ul style="list-style-type: none"> <li>• Call handling settings and voicemail can be set using PC browser</li> <li>• Supported browsers:</li> </ul>
Call coverage	<ul style="list-style-type: none"> <li>• Busy / no answer transfer; implemented in Comm server</li> <li>• Call waiting indication (tone, visual); implemented in phones (3P)</li> <li>• Multiple incoming line support; Implemented by phones (3P)</li> </ul>
Call forward	<ul style="list-style-type: none"> <li>• Call forwarding of all calls to other (on-net) stations; implemented in Comm server via Config Server interface</li> <li>• Implemented in Comm server</li> </ul>
Call history	<ul style="list-style-type: none"> <li>• Shows calls placed, received, missed, failed</li> <li>• Implemented as phone capability (3P)</li> </ul>
Call hold / retrieve	<ul style="list-style-type: none"> <li>• Call status per-line (state, duration, number; shown on call status screen when call is taken off hold)</li> <li>• Implemented by stations &amp; trunk gateways (3P)</li> </ul>
Call waiting / retrieve	<ul style="list-style-type: none"> <li>• Answer 2nd/3rd/... incoming call while one is in progress; retrieve 1st call after answering subsequent calls</li> <li>• Implemented as phonetop capability (3P)</li> </ul>

Features	Pingtel SIPxchange PBX
Calling line identification	<ul style="list-style-type: none"> <li>• Number of remote end displayed while ringing; CLID</li> <li>• Implemented in gateway and phone (3P)</li> </ul>
Calling party name identification	<ul style="list-style-type: none"> <li>• Name of remote end displayed while ringing; Caller-ID</li> <li>• Implemented in trunk gateway and phone (3P)</li> </ul>
Conferencing	<ul style="list-style-type: none"> <li>• 3-way conference calls</li> <li>• Implemented on phone</li> </ul>
Direct inward dial (DID)	<ul style="list-style-type: none"> <li>• DID call block directs calls to dialed number without going through automated or manual attendant</li> <li>• Implemented in Comm server and phone</li> </ul>
Do Not Disturb	
Hands-free, full-duplex speaker-phone	<ul style="list-style-type: none"> <li>• Implemented by phones (3P)</li> </ul>
Initiate call from call log	
Message waiting indication as part of Media Server	<ul style="list-style-type: none"> <li>• Message waiting light on phone</li> <li>• Message count indication on display of phone</li> <li>• Implemented in Comm server, Media server, and phone (3P)</li> </ul>
Multiple call appearance	<ul style="list-style-type: none"> <li>• More than one call to same phone number rings, and answerable at same phone</li> <li>• Implemented in Comm server and phone (3P)</li> </ul>
Multiple line appearance per phone	<ul style="list-style-type: none"> <li>• Phone can be assigned more than 1 phone number (up to 12)</li> <li>• Implemented in Comm server and phone (3P)</li> <li>• Mute capability from handset, speakerphone, and headset</li> <li>• Hard-cap button for mute operates context-sensitive</li> <li>• Implemented by phones (3P)</li> </ul>
Multi-station appearance	<ul style="list-style-type: none"> <li>• Single number rings at multiple phones. Call is completed to first phone to answer &amp; terminated at others</li> <li>• Implemented by Comm server.</li> </ul>
On-hook dialing	<ul style="list-style-type: none"> <li>• Instant off-hook when DTMF digits are pressed</li> <li>• Implemented by phones (3P)</li> </ul>
Station to Station Dialing	
Transfer Call to Voice Mail	<ul style="list-style-type: none"> <li>• By prefixing an extension with an 8, a call can be transferred directly to a voice mail box.</li> </ul>
Transfer – blind	<ul style="list-style-type: none"> <li>• Transfer to a different station without speaking to the other station prior to transfer</li> <li>• Transfers initiated by hard-cap transfer button on phone. Wizard on LCD screen assists user with process of transfer. (3P)</li> <li>• Implemented by stations (3P)</li> </ul>
Transfer – with consultation	<ul style="list-style-type: none"> <li>• Transfer to a different station after speaking to the other station prior to completing the transfer</li> <li>• Transfers initiated by hard-cap transfer button on phone. Wizard on LCD screen assists user with process of transfer. (3P)</li> <li>• Implemented by stations (3P)</li> </ul>
<b>Standard Supported</b>	
	<ul style="list-style-type: none"> <li>• RFC 3261 Session Initiation Protocol using both UDP and TCP transports</li> </ul>
	<ul style="list-style-type: none"> <li>• Advanced call control using RFCs <ul style="list-style-type: none"> <li>• 3515 Refer Method</li> <li>• 3891 Referred-By header</li> <li>• 3892 Replaces header</li> </ul> </li> </ul>
	<ul style="list-style-type: none"> <li>• Provide for consultative and blind transfer and third party call controls</li> </ul>
	<ul style="list-style-type: none"> <li>• RFC 3263 Locating SIP Servers - use of DNS SRV records for call routing control and server redundancy.</li> </ul>
<b>SIP Implementation</b>	<ul style="list-style-type: none"> <li>• RFC 3581 Symmetric Response Routing (rport)</li> </ul>
	<ul style="list-style-type: none"> <li>• RFC 3265 SIP Event Notification - for phone configuration and</li> </ul>
	<ul style="list-style-type: none"> <li>• RFC 3842 Voice mail message waiting indication (MWI)</li> </ul>
	<ul style="list-style-type: none"> <li>• RFC 3262 Reliable Provisional Responses</li> </ul>
	<ul style="list-style-type: none"> <li>• RFC 2833 Out-of-band DTMF tones</li> </ul>
	<ul style="list-style-type: none"> <li>• RFC 3264 Offer/Answer model for SDP for Codec Negotiation</li> </ul>
	<ul style="list-style-type: none"> <li>• Early media (SDP in 180/183)</li> </ul>
	<ul style="list-style-type: none"> <li>• Delayed SDP (SDP in ACK)</li> </ul>
	<ul style="list-style-type: none"> <li>• Re-INVITE: Codec change, hold, off-hold</li> </ul>
	<ul style="list-style-type: none"> <li>• Route/Record-Route header fields</li> </ul>
	<ul style="list-style-type: none"> <li>• Configurable RTP/RTCP ports</li> </ul>
	<ul style="list-style-type: none"> <li>• Configurable SIP ports</li> </ul>

Features	Pingtel SIPxchange PBX
<b>Diagnostic and Troubleshooting Features</b>	
Registrations	• View of all user/phone registrations and current status
Job Status	• View of all jobs being run on Config Server – will show status of updating managed phones.
View of Server status	• View of server status (Comm, Media, Config) and all services
Patches	• View of all patches and ability to apply new patches.
<b>Security Features</b>	
System Security	<ul style="list-style-type: none"> <li>• HTTPS between SIPxchange server components</li> <li>• HTTPS between admin and user consoles and the SIPxchange servers</li> <li>• Secure channel for retrieval voice messages from voicemail repository</li> <li>• HTTP digest authentication for SIP signaling (3P)</li> </ul>
URI mapping engine for call routing and inter-company (/domain) SIP calls	<ul style="list-style-type: none"> <li>• XML-described rules for directing SIP messages flowing through call servers</li> <li>• Enables server-to-server direct routing over IP, either intra-organizationally or inter-organizationally</li> <li>• Inter-organizational calls can be challenged for authorized credentials</li> <li>• Implemented in Comm server.</li> </ul>
<b>SIP Softphone</b>	
Soft phone	<ul style="list-style-type: none"> <li>• Works on Windows 2000 and later</li> <li>• Direct support for Clarisys USB handset &amp; headset for the following: <ul style="list-style-type: none"> <li>• Dialing from handset (including “send”)</li> <li>• Volume up / down from handset buttons</li> <li>• “Ring” played through laptop speakers; audio switches to handset after answer</li> </ul> </li> </ul>
<b>Unique Features</b>	
User Programmable Follow-me	• User can forward calls from User Web portal. Calls can ring forwarding destination(s) simultaneously or in a linear hunting format.
Browser-based Voice Mail management	• View and manage Voice mail folders from User Web portal.
Manage 3 <sup>rd</sup> party phones	• Config Server can manage Polycom and Cisco SIP phones. Other phones will be added in the next version.

CT Labs found this product to have a good feature set comprising all the features that most small businesses would need in a SIP PBX (with the exception of call park/pickup). The system now interoperates with Cisco and Polycom phones – a definite improvement since the last time CT Labs reviewed this system.

Overall, the current set of Pingtel SIPxchange features was rated an “8.7.”

**Upcoming release feature alert:**

In addition to the above list of current features, Pingtel also submitted a list of features that will be added to the SIPxchange system in their next release (version 3.0 – due out in October 2005). These include call park/pickup, multiple auto-attendants, voicemail distribution lists, a software-based Attendant Console, Meet-me conferencing, and an ACD feature set. Please see Appendix A for the full list. These new features should move the Pingtel system up a notch from “basic PBX solution” to “full-featured PBX solution.”

## 4.3 Product Documentation and Integrated Help Evaluations

### 4.3.1 User Guides

Pingtel provided CT Labs with on-line access to the PDF documentation for the SIPxchange system and the SIP Softphone, the same way they provide this information to every customer. The following guides were evaluated:

- ◆ **Administrative Guides**
  - SIPxchange Administration
  - Installing and Configuring the Pingtel SIP Softphone
- ◆ **User Guides**
  - SIPxchange Voicemail Options
  - Using Cisco 7940 and 7960 Phones with SIPxchange
  - Using Polycom SoundPoint 300 Phones with SIPxchange
  - Using Polycom SoundPoint 500 and 600 Phones with SIPxchange

The 43-page **SIPxchange Administration** guide provided information on the system modules and interfaces, user settings, endpoints, network services, administering the system, and diagnostic tools. There were screenshots and graphics liberally available throughout the guide to help the reader identify with the concepts presented in the text. There were also plenty of useful hot-links provided.

The 231-page **Installing and Configuring the Pingtel SIP Softphone** guide included information on installing the softphone, administration and configuration topics, and troubleshooting. Since the softphones can be set up peer-to-peer or registered with a SIP server, these options are covered. While these many configuration options make the softphone extremely versatile, CT Labs found that the variety of options made it challenging to read through the guide and perform the softphone configuration necessary to register the softphone with the SIPxchange server. We would like to see more concise sections of instructions for setting up the softphone in order to have it registered with a SIPxchange server.

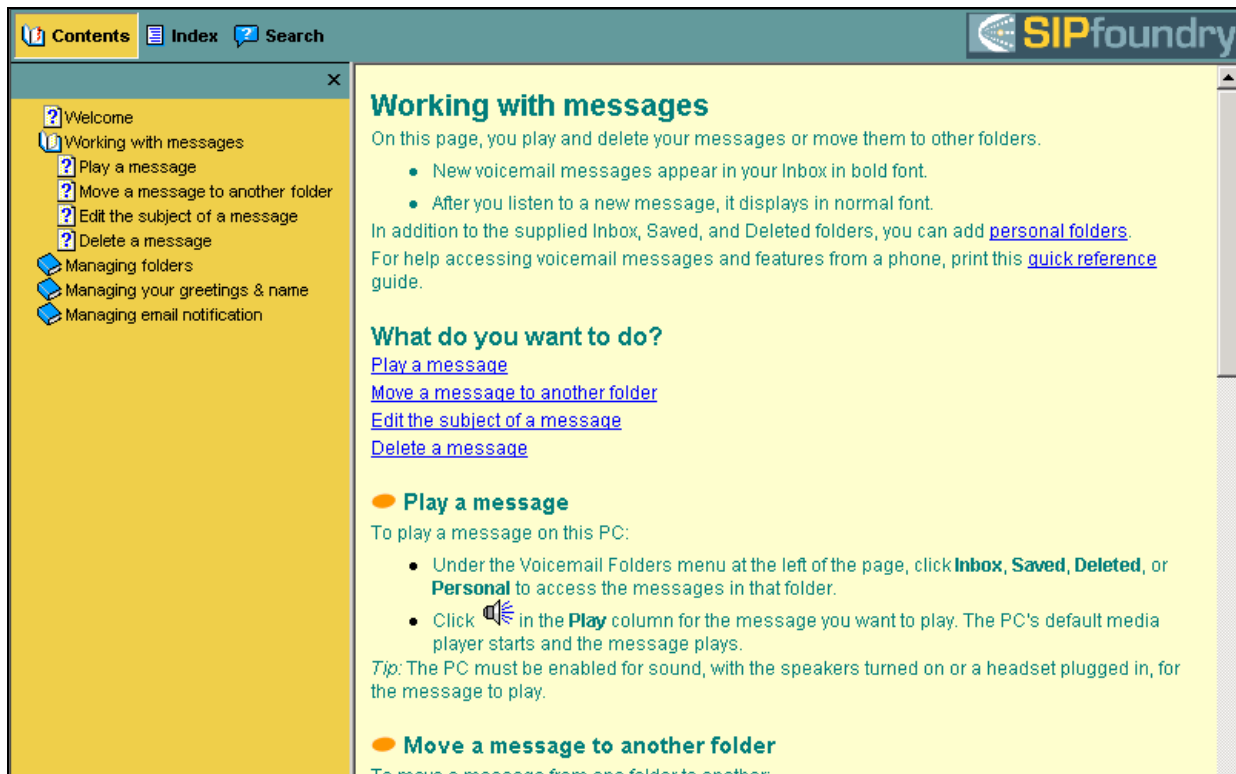
The **SIPxchange Voicemail Options** guide was provided on a single page showing a map of the voicemail system TUI, helpful for familiarizing the voicemail user with the menu structure. It also shows instructions on how to perform the basic functions available within voicemail.

The **Using “XXX” Phones with SIPxchange** guides were each two-page guides, meant to be printed and folded into thirds, providing a handy phone-specific reference. Each is designed to help the user with basic dialing instructions and setting up their options and includes a flowchart of the voicemail system. These should be very helpful to novice users.

### 4.3.2 Integrated Help

The administrative and client web-based GUIs both contained integrated help files that were organized by categories of Contents, Index, and Search. The Contents tab showed an expandable “tree” of options in the left-hand frame, and the Index Tab showed a list of words from A to Z. One limitation of the integrated help: while most Search tabs of integrated help products allow the user to type in a word which is then searched in the help files, the Pingtel SIPxchange online help Search tab only allowed us to search from a pre-defined list of words.

The administrative and client integrated help links provided context-sensitive GUI help. Both contained plenty of hot links to related topics, and both used illustrative icons (see Screenshot 1 below for an example of a topic in the voicemail client GUI).



**Screenshot 1: Client Voicemail Integrated Help**

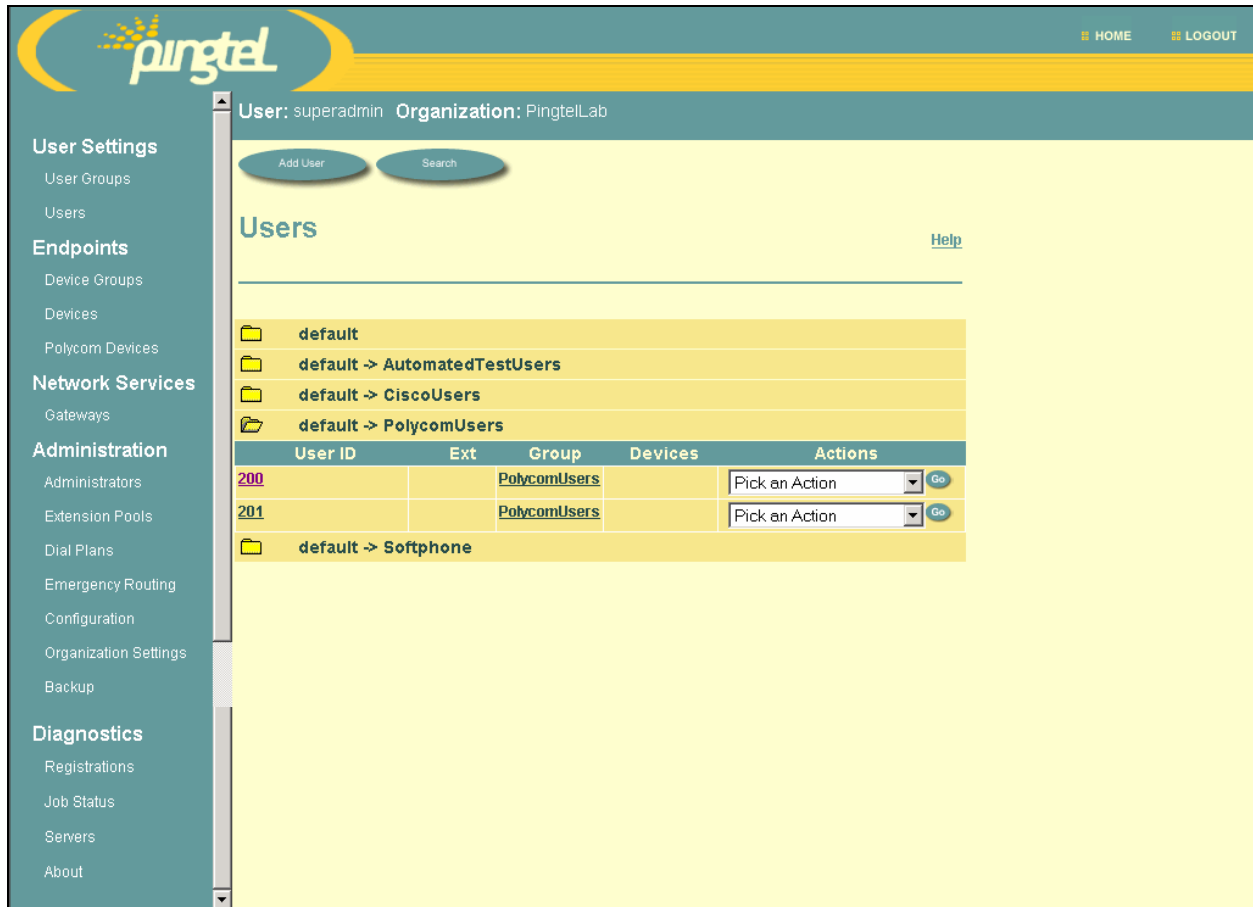
Overall, CT Labs found the Pingtel documentation and integrated help files to be well-organized with helpful information that is presented in a straight-forward and intuitive manner. CT Labs rates the documentation and integrated help with a score of “8.9.”

#### **4.4 Graphical User Interface Ease of Use Evaluation**

CT Labs reviewed the administrative and client web-based GUIs provided with the SIPxchange product. All are similar in design, with the main window containing a menu of options on the left, and options related to the chosen topic displayed on the right. These GUIs are logically designed and easy to navigate. Options were found to be clearly labeled, and the Help links provided useful, context-sensitive help.

The **administrative GUI** allows the user to define individual user and phone device profiles, as well as user and device groups. The ability to create groups is handy because it allows you to create a template of group settings (which saves much time as compared to individually configuring each user). After a group of users is created, individual user’s settings can still be modified.

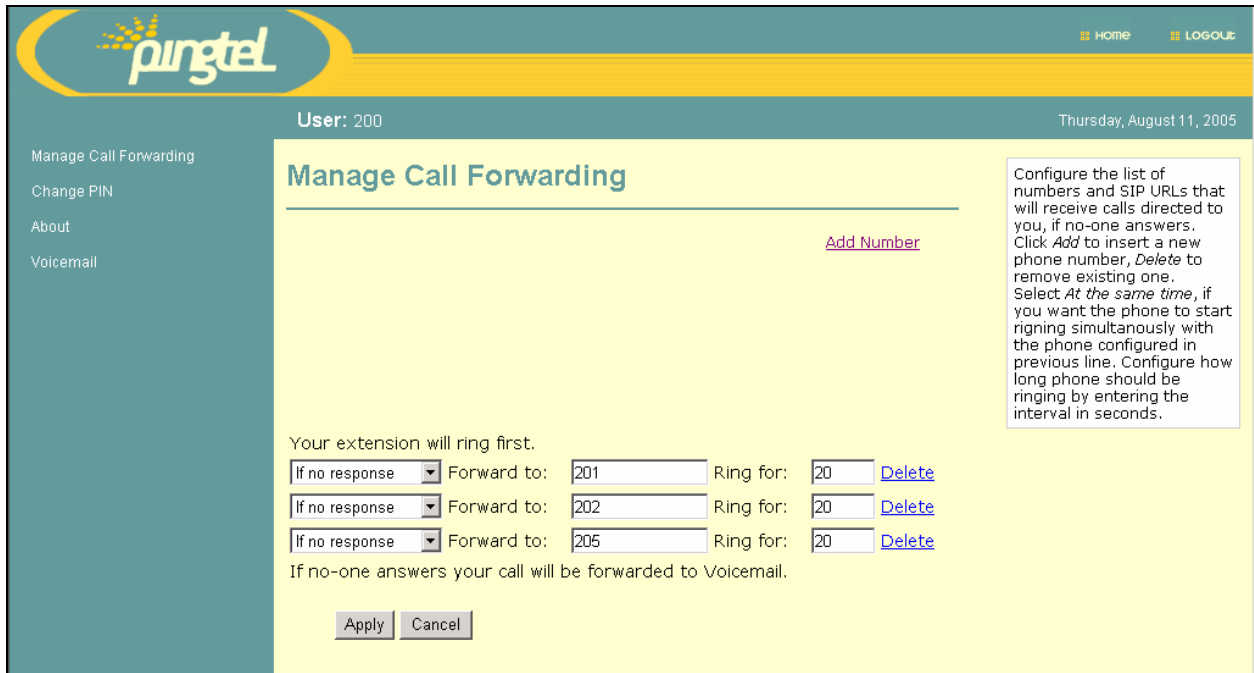
See Screenshot 2 below for an example of the administrative GUI.



**Screenshot 2: Administrative GUI – Users Screen**

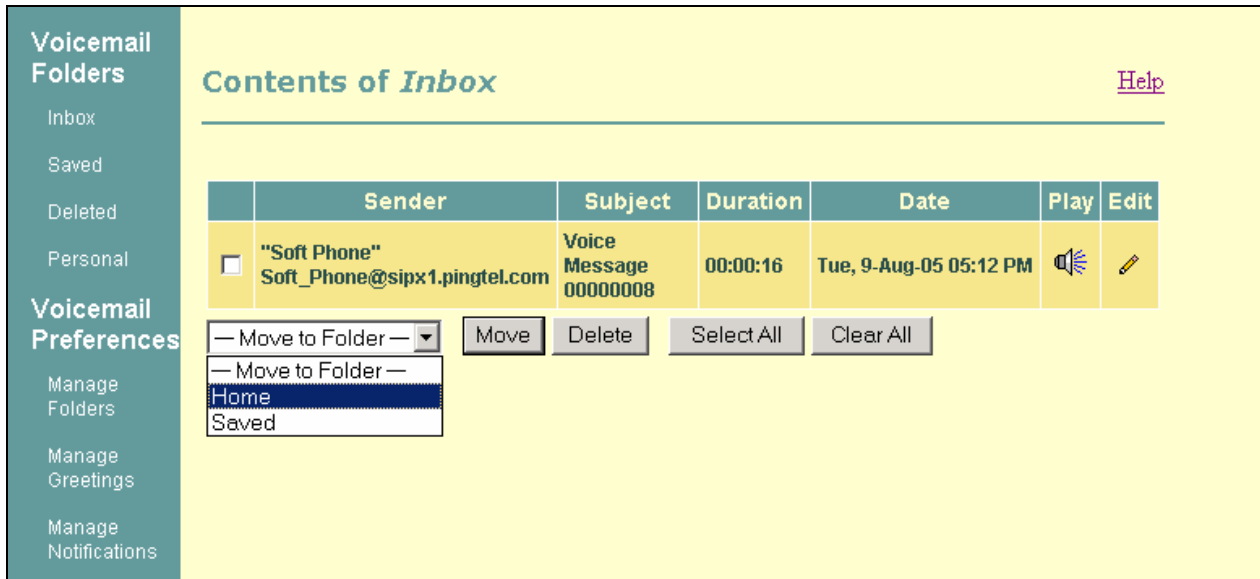
One thing that CT Labs found a bit puzzling in the administrative GUI was the “Administrators” section. This comes predefined with a superadmin administrator, but does not allow you to add any other administrators... all you can do is change the superadmin password. Pingtel confirmed that currently there is only a superadmin administrator allowed on the system. However, they did indicate that they plan to add the ability to have multiple administrators in Q1 2006.

There are two client user GUIs – one for managing user preferences, and the other for voicemail. The **user management GUI** (shown below in Screenshot 3) allows the user to change their call forwarding preferences (with one or more forwarding destinations) and also to change their PIN (for logging into the GUI and the TUI).



**Screenshot 3: Client Management GUI**

The **voicemail GUI** (shown in Screenshot 4 below) allows the user to listen to and manage their voicemail messages, as well as managing folders, greetings, and email notifications.



**Screenshot 4: Client Voicemail GUI**

CT Labs also used the **Pingtel SIP Softphone GUI** (shown in Screenshot 5 below) while placing calls to and from the desktop SIP phones used in this test. This GUI was found to be easy to use.



Screenshot 5: Pingtel SIP Softphone GUI

Overall, CT Labs rated the Pingtel SIPxchange GUIs with a score of “8.9.”

#### 4.5 Manual Product Exercise Results

For the manual product exercise, CT Labs product analysts spent several hours exercising as many features and options as possible from the list of features provided by Pingtel. The following table shows features that were tested.

Table 4: Manual Feature Exercise Results

Feature Tested	Result
<b>System</b>	
Dynamic call forwarding	Using the client web interface, we were able to successfully designating other stations as forwarding destination(s) for a phone.
Flexible automated bandwidth selection	We were able to successfully choose preferred codecs for our test phones based on individual phones and phone groups.

Feature Tested	Result
Hunt group	As noted with call forwarding, we were able to successfully set up a list of phones where we would like to have our calls forwarded. We could choose to have them ring together, or sequentially.
Message waiting indication	We were able to successfully view MWI indicators on each phone used, as well as being able to hear stutter-dial tone (when enabled on the phone device).
<b>Auto-Attendant</b>	
Confirmation of alpha-dialed number via .wav file for each users name	We confirmed that if we did not have a name recorded for a user, the auto attendant would simply read the extension number, but if we had a name recorded for the user, the auto attendant would play the recorded file.
Dial by extension	We were able to successfully dial by extension through the auto-attendant.
Dial by name	We were able to successfully dial by name through the auto-attendant. We also verified that if there were multiple matches for an entry we made, that the system would list all of the matches as choices to dial.
<b>Voice Messaging</b>	
.wav file messages	We were able to review the messages through the client user web GUI using the default media player on the workstation we used.
Browser-based interface	We were able to use the browser-based interface successfully.
Editable message headers	We were able to successfully edit message headers.
Folders for message organization	We were able to successfully create and use other folders in the GUI, and access them later through the voicemail TUI.
Multiple user customizable voicemail greetings	We were able to record and manage multiple greetings.
<b>Administrative Interface</b>	
Browser-based configuration system	We were able to use the browser-based interface successfully.
<b>Client Interface</b>	
Answer / release	We were able to answer and release calls successfully.
Browser-based user interface	We were able to use the client user browser-based interface successfully for managing the options for each phone.
Call coverage	We were able to successfully set up and use busy/no-answer forwarding to another extension.
Call forward	We were able to successfully forward calls to one or more destination extensions.
Call history	We were able to successfully view our call history.
Call hold / retrieve	We were able to successfully put calls on hold, and then retrieve them.
Calling line identification	We were able to successfully receive the number of the calling party on our phones.
Calling party name identification	We were able to successfully receive the name of the calling party on our phones.
Conferencing	We were able to successfully place on-demand conference calls with three phones.
Do Not Disturb	We were able to successfully use Do Not Disturb and found that incoming calls would automatically be directed to voicemail.
Initiate call from call log	We were able to successfully view our call history and place calls by clicking on the call record in the log.
Message waiting indication as part of Media Server	We were able to successfully view MWI indicators on each phone used, as well as being able to hear stutter-dial tone (when enabled on the phone device).
On-hook dialing	We were able to successfully dial without going off-hook first. Depending on the phone settings, we sometimes had to press a "dial" button to complete the call.

Feature Tested	Result
Station to Station Dialing	We were able to successfully call from station to station.
Transfer Call to Voice Mail	We were able to successfully transfer a call to voicemail by dialing 8 before the station.
Transfer – blind	We were able to successfully blind-transfer a call.
Transfer – with consultation	We were able to successfully transfer a call after consultation.
<b>Diagnostic and Troubleshooting Features</b>	
Registrations	We were able to successfully view all SIP phones registered with the server.
Job Status	We were able to successfully view all jobs being run.
View of Server status	We were able to successfully view the status of the server components.
Patches	We were able to successfully view all applied patches.
<b>SIP Softphone</b>	
Soft phone	We were able to successfully install and use the SIP softphone on our Windows 2000 workstation with a USB headset.
<b>Unique Features</b>	
User Programmable Follow-me	Using the client web interface, we were able to successfully designating other stations as forwarding destination(s) for a phone.
Browser-based Voice Mail management	We were able to successfully use the browser-based GUI to hear voicemail messages and manage our voicemail messages and settings.
Manage 3 <sup>rd</sup> party phones	We were able to use the SIPxchange administrative GUI to manage the Cisco and Polycom phone settings, instead of having to do them through the phone.

CT Labs also manually exercised the SIPxchange product with a variety of phone types and call types. During these calls, CT Labs testers noted perceived voice quality, call connection capability, and any problems with call handling.

**Table 5: Manual Call Connectivity and Perceived Voice Quality**

		Terminating Phone			
		Polycom IP 300 Phone	Polycom IP 500 Phone	Cisco 7940 Phone	SIP Softphone
Originating Phone	Polycom IP 300 Phone	Not Tested	All Calls Successful High Call Quality	All Calls Successful High Call Quality	All Calls Successful High Call Quality
	Polycom IP 500 Phone	All Calls Successful High Call Quality	Not Tested	All Calls Successful High Call Quality	All Calls Successful High Call Quality
	Cisco 7940 Phone	All Calls Successful High Call Quality	All Calls Successful High Call Quality	Not Tested	All Calls Successful High Call Quality
	SIP Softphone	All Calls Successful High Call Quality	All Calls Successful High Call Quality	All Calls Successful High Call Quality	Not Tested

All calls placed in this manual functionality evaluation were completed successfully with high perceived voice quality. No problems were noted with call connectivity or voice quality.

Overall, CT Labs rated the SIPxchange product with a score of “9.4” for the manual product exercise.

#### 4.6 Automated Call-Handling Load Test

CT Labs performed both station-to-station and trunk-to-station automated call-handling tests with the purpose of evaluating the ability of the SIPxchange system to handle multi-line call loads.

The **station-to-station test** was performed with a single Empirix Hammer FX-IP call generator system, which was registered as 96 station phones with the SIPxchange system. The Hammer FX-IP then used 48 of the “stations” to call the other 48 Hammer stations and complete a short (approximately 6 second) phone call. After going off-hook, the originating station waited two seconds before placing another call. CT Labs performed this test as an overnight test run, logging 17 ½ hours of calls<sup>1</sup>.

The **trunk-to-station test** was performed with one Empirix Hammer FX-TDM call generator system placing trunk calls into the T1 interface on the SIPxchange system. The SIPxchange system then routed the calls directly to the appropriate SIP extension, each of which was answered by the Hammer FX-IP. Each call in this test lasted approximately 9 seconds; after going on-hook, the originating trunk line would wait 4 seconds before placing another call. This test was run for 6 hours.

Table 6 below shows the results for each of the two automated call-handling load tests.

**Table 6: Automated Call-Handling Load Test Results**

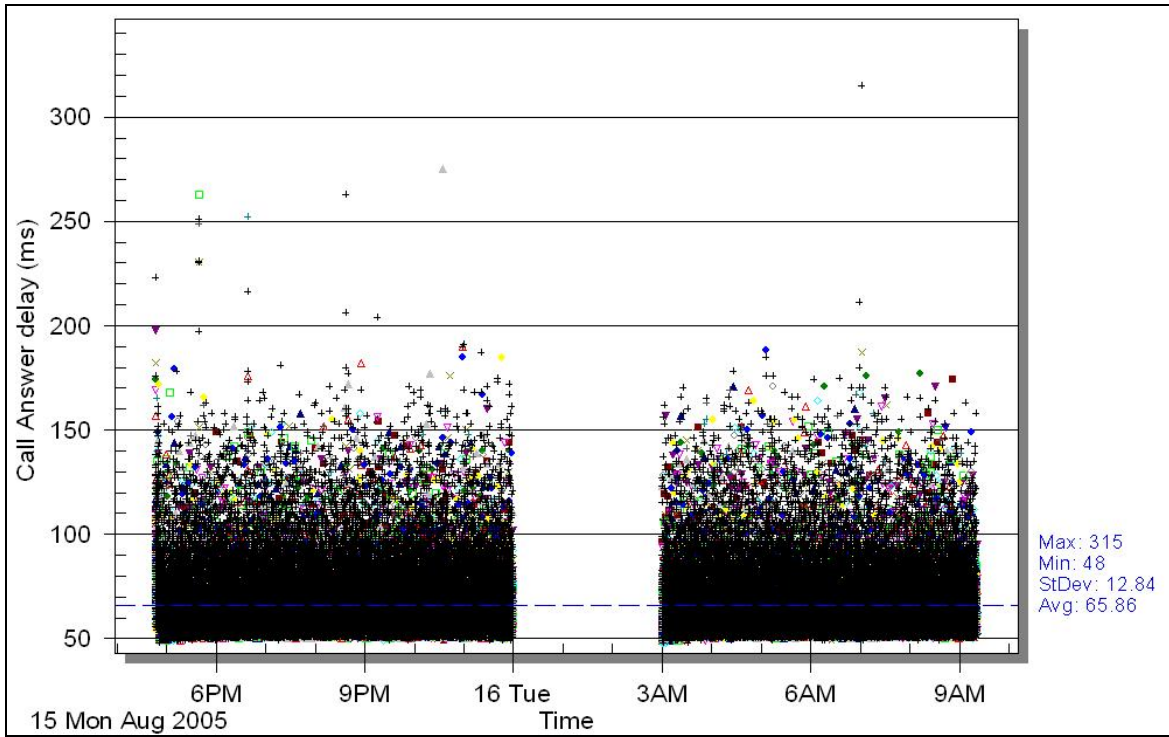
	Station-to-Station Test	Trunk-to-Station Test
Number of calls placed	214,523	41,368
Number of calls successfully answered	214,523	41,368
Number of call failures detected	0	0
Percentage of successful calls	100%	100%
Call Answer performance <sup>2</sup>		
Minimum (mSec)	48	50
Maximum (mSec)	315	1262
Average (mSec)	65.86	131.66

The Pingtel SIPxchange system was found to perform very well under these test conditions. There were no failed calls during either test. During the station-to-station test, the call answer delay was consistently very good. During the trunk-to-station test, the call answer delay was also found to be very good for most of the calls, with only 12 calls that took longer than 200 milliseconds to answer.

The Call Answer performance results for these tests are also shown in graphical form in Figures 2 and 3 below.

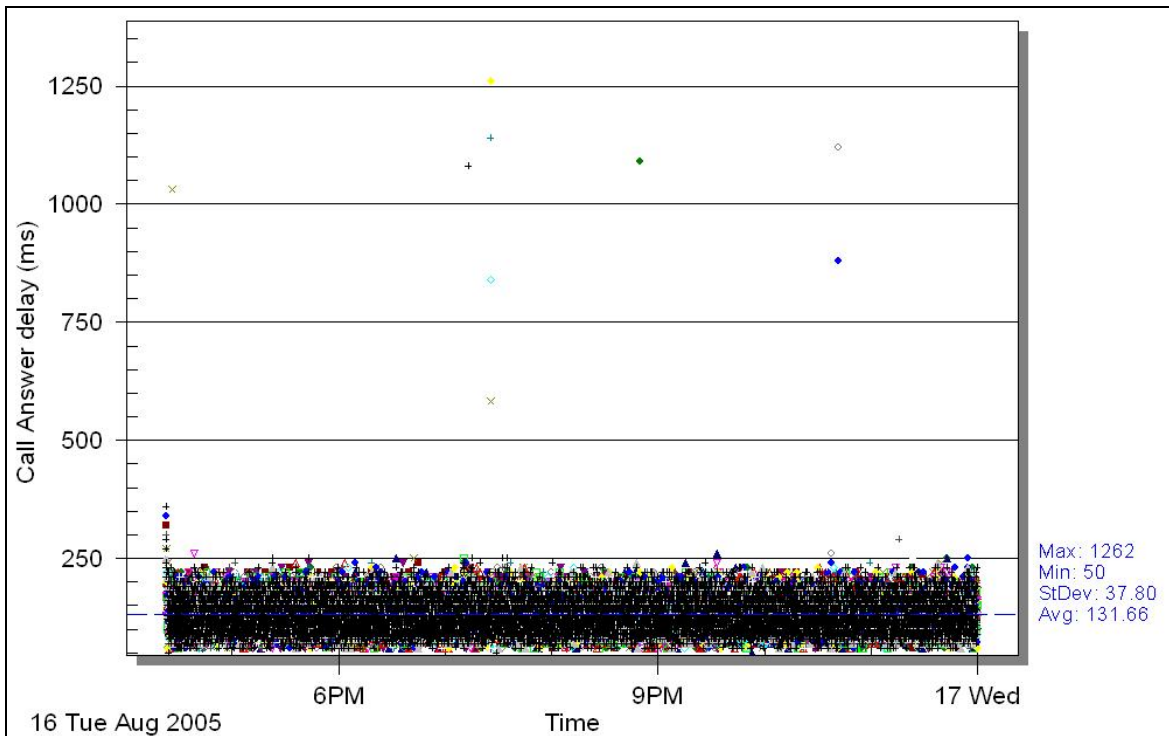
<sup>1</sup> Note that the test scripts were designed to stop calling between 12 midnight and 3 AM, since we have found that frequently systems that we test are set up to run maintenance jobs during that time, which can significantly impact performance.

<sup>2</sup> Call Answer performance is measured from the time that the automated call generator initiates a call into the SIPxchange system until the automated call generator “hears” the first prompt being played by the SIPxchange system.



**Figure 2: Station-to-Station Call Answer Delay Results**

(Note: the midnight to 3AM gap in calls was intentional)



**Figure 3: Trunk-to-Station Call Answer Delay Results**

As illustrated in the results above, the SIPxchange system provided consistent handling for both station-to-station calls and trunk-to-station calls throughout the duration of the CT Labs tests.

Overall, CT Labs rates the call-handling ability of the Pingtel SIPxchange system with a score of “9.6.”

#### **4.7 Technical Support**

Throughout the evaluation process, each tester was instructed to keep track of when questions were posed to the technical support department, and when a callback or email response was received. The status of each reported bug or product defect, if any, was maintained in a log. The relative responsiveness of the vendor as well as the quality of the answers received was also noted, if any.

During this test CT Labs testers found that they needed some help to configure the softphone properly to register with the SIPxchange server, and to configure the Empirix Hammer FX-IP channels to register with the SIPxchange server. A CT Labs analyst also called once to get clarification on a couple of minor questions. Our assigned technical representative was found to be responsive, and was usually even reachable even during the evening hours on his cell phone. The one time that we were unable to reach hi during the evening, we sent our question via email, and it was answered the next morning. While evening access to a technical representative may not be guaranteed for every customer, CT Labs takes this as a good sign that Pingtel is trying hard to keep their customers happy.

Overall, CT Labs received knowledgeable, prompt responses to our questions. As such, CT Labs rates the Pingtel technical support experience with a score of “9.6.”

*End of report*

## Appendix A Pingtel Version 3.0 New Features

The following list of features was provided by Pingtel as new features that will be implemented in the SIPxchange system in their next release (version 3.0, due out in October, 2005).

Features	Pingtel SIPxchange PBX
<b>PBX Features</b>	
Call pick-up	<ul style="list-style-type: none"> <li>The call pick-up feature allows a user to dial a feature code and retrieve a call that is ringing at another phone or group of phones.</li> </ul>
Call park/retrieve	<ul style="list-style-type: none"> <li>This feature will allow users to park calls. The user would then page or notify a user which extension the call is parked on. The call can be retrieved by dialing the call park extension. Each call park extension can have a unique music on hold source or greeting that the administrator can import.</li> </ul>
Red Hat Enterprise Linux 4.0 Support	<ul style="list-style-type: none"> <li>SIPxchange will support RHEL 4 with this release.</li> </ul>
Scheduled Back-ups	<ul style="list-style-type: none"> <li>Administrator can schedule back-ups of system configurations and/or voice mail messages daily or weekly.</li> </ul>
<b>Auto Attendant Features</b>	
Multiple Auto Attendants	<ul style="list-style-type: none"> <li>Administrator can create up to 100 auto attendant menus. Menus can answer calls with unique greetings and call routing rules. Menus can be nested together to create custom call flows.</li> </ul>
<b>Voice Messaging Features</b>	
Voice Mail Distribution Lists	<ul style="list-style-type: none"> <li>Voice Mail users can create custom distribution lists.</li> </ul>
Change PIN with TUI	<ul style="list-style-type: none"> <li>User can now change their voicemail and web portal PIN through the TUI. Previous versions would not allow this.</li> </ul>
<b>Administrative Interface Features</b>	
SIP Phone provisioning	<ul style="list-style-type: none"> <li>This release adds provisioning support for Snom and Grandstream phones</li> </ul>
User Configuration	<ul style="list-style-type: none"> <li>Enhanced user management by sorting by User ID, First Name, Last Name, Alias or Group.</li> </ul>
<b>Client Interface Features</b>	
Soft Attendant Console	<ul style="list-style-type: none"> <li>The Soft Attendant Console is a PC application that is used for managing calls. The application includes softphone capabilities and can operate on a MS Windows platform with a USB headset.</li> </ul>
<b>Meet-Me Conferencing Bridge Features</b>	
Secure Conferencing	<ul style="list-style-type: none"> <li>Administrator can set up a PIN for the Conference Organizer and the Participant</li> <li>Each conference can be configured to start or not start without the Organizer.</li> </ul>
Entry and Exit Tones	<ul style="list-style-type: none"> <li>Tones are played when each caller enters or exits</li> </ul>
Mute	<ul style="list-style-type: none"> <li>Each participant and organizer can mute and un-mute by dialing a code</li> </ul>
Configurable with SIPxchange PBX or stand-alone	<ul style="list-style-type: none"> <li>Bridge can be installed on same server with SIPxchange PBX or installed by itself with a 3<sup>rd</sup> party proxy server.</li> </ul>

Features	Pingtel SIPxchange PBX
<b>ACD Features</b>	
Intelligent Routing	<ul style="list-style-type: none"><li>• Ring All, Linear, Circular and Longest Idle is supported. This is programmable on a per queue basis.</li></ul>
Welcome Greeting	<ul style="list-style-type: none"><li>• A greeting is played to caller before they enter the queue. This greeting can be disabled or can be interrupted as soon as an agent is available. This is programmable on a per queue basis.</li></ul>
Queue Message	<ul style="list-style-type: none"><li>• A message or music is played to callers that are in queue. This is programmable on a per queue basis.</li></ul>
Agent Sign in/Sign out	<ul style="list-style-type: none"><li>• Agent can sign in or out through a feature code, web interface or custom API</li></ul>
View of all callers in queue	<ul style="list-style-type: none"><li>• An API is provided to view all calls in queue. Caller information like ANI, DNIS, time in queue, ringing etc is available.</li></ul>
Overflow	<ul style="list-style-type: none"><li>• Call can overflow to a secondary destination. The destination can be any extension or SIP URL. Each queue can overflow the last call in to queue or the call that has been queued the longest.</li></ul>